







B&H

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The ProfessionalAudio SourceBook

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INTRODUCTION TO

SELECTING A MICROPHONE—THE BASICS

Dynamic or Condenser?

One of the first things that you need to know before purchasing is whether you need a dynamic or condenser microphone. Inside each and every microphone is a transducer. This is the component which captures sound by changing one form of energy into another. In this process acoustic energy is transformed into electrical energy. Mic transducers utilize either dynamic or condenser transducer elements.

Dynamic Microphones:

These fall into two main groups, ribbon and moving coil. All dynamic microphones utilize the more rugged moving coil principle. This is like a loudspeaker in reverse: a coil of wire connected to the microphone diaphragm is held in a magnetic field. Sound waves change the pressure of the air they travel through. This causes the diaphragm to vibrate, thus, the coil moves in the magnetic field and an audio frequency voltage is induced. This signal is balanced, grounded and of low resistance and can be connected directly to the microphone input of a tape recorder or mixing console without any electronics or supply voltage being needed in the microphone. Moving coil mics are extremely rugged and reliable. Consequently, they can be used in difficult working situations where more fragile microphones would not be suitable. The main uses are in broadcast reporting situations and in live stage work.

Condenser and Electret Condenser Microphones:

The condenser transducer is made up of two plates in a capacitor arrangement. The front plate is the diaphragm and responds to acoustic pressure changes. The back plate is fixed. An electrical charge is applied to either the front diaphragm or back plate depending on the design. As acoustic pressure moves the diaphragm, the capacitance changes between the plates and a voltage proportionate to the acoustic signal is produced.

The electrodes used in electret (pre-polarized) condensers are manufactured from a special synthetic material. This material has a unique quality, being able to permanently store an electric charge; the charge is thus 'frozen' in the electret material. No external polarizing voltage is required for the capsule and a low voltage power source can power the remainder of the mic's circuitry. In contrast to dynamic mic, electret condenser mics can be very compact. This makes it possible to design ultra-small clip-on lavalier mics, and small modular capsules. Back-electret condenser mics use 48v phantom power (P 48) for external power.

Dynamic	vs.	Condensers
Pros:		**Pros:**
• Rugged		• More sensitive
• Low self-noise		• Better transients and frequency response
• Lower Cost		• Useful small or large diaphragms
• No power required		• Multiple patterns in one housing
• Less prone to weather and RF interference		**Cons:**
Cons:		• Higher self-noise
• Lower sensitivity/output level		• More fragile, more expensive
• Slower transient response		• Require power source
• Usually larger and heavier		• Prone to weather/RF interference

Reading Microphone Specifications

The most common and useful specs for the microphone selection process are Polar (or pick-up) Pattern, Polar Response Graphs and Frequency Response Graphs. These specs, in addition to, the most important decision factor of all, your ear, can tell you how appropriate a specific mic will be for your application.

Polar Patterns:

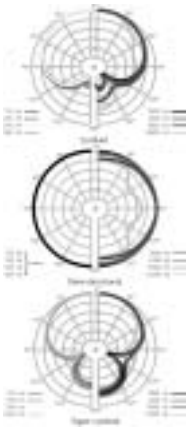
One of the goals in microphone design is to make the transducer element more sensitive to sounds coming from a certain direction. Its directionality plays a key role the application it will be selected for — whether it's capturing live vocals or a kick drum. A polar pattern illustrates this directionality by showing the general sensitivity of a microphone from different directions using a 1 kHz (kilohertz) test tone. This is sometimes referred to as "Rejection @ 1 kHz." There are 5 common polar patterns: Omni-directional, Bi-directional or Figure-8, Cardioid, Super Cardioid and Lobar.

Omni-directional	360° coverage
Bi-directional	Coverage from the front and rear of capsule
Cardioid	180° coverage. No sound from rear.
Super Cardioid	160° coverage. Slight pick-up from the rear.
Lobar	140° coverage. Off axis sound is highly attenuated



As you look at the representation of a microphone's polar pattern you can see its ability to reject sounds from a given direction. The area where a mic completely rejects sound is its "null" point. A cardioid mic has a null point that is directly to the rear of the mic. A super-cardioid mic or lobar mic has null points at its rear sides. Knowledge of a microphone's null point is paramount for proper mic placement relative to loudspeaker arrays and other microphones.

Polar Response Graphs:

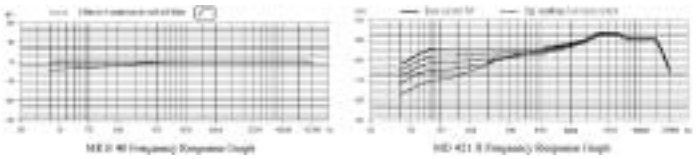


Polar response charts, like those shown at right, illustrate a mic's pickup pattern at various frequencies. By comparing the size and shape of the various frequency plots, you can get a good sense of a mic's off-axis response or, how the mic discriminates frequencies hitting its capsule from different angles. The graph is drawn as if the mic capsule was positioned in the center of the graph facing 0 degrees. To make polar plots less cluttered, only one side of each frequency plot is shown even though its pattern is symmetrical. Generally, directional mics become more omni-directional at lower frequencies. This is because higher frequencies have shorter wavelengths than lower frequencies and are easier to control in terms of their directivity.

SELECTING A MICROPHONE—THE BASICS

Frequency Response Graphs:

Polar response graphs are often accompanied by frequency response graphs, which illustrate output levels of the microphone over its operating range from lowest to highest frequency. The limits of human hearing are contained within the frequency spectrum from 20 Hertz (cycles per second) to 20,000 Hertz. As you look at the graph notice that these are shown from left to right along the bottom. The vertical figure on the left can either be a measurement of the relative response measure in decibels (dB) [see MKH 40 Graph below]. In this scale, a 3 dB increase or decrease is considered to be noticeable to most individuals and a 10 dB change is considered to be twice as soft or twice as loud as the original sound. Other graphs will use a dBv scale as a means of expressing the output voltage of a microphone at different frequencies [see MD 421 II Graph below. The flat response of the MKH 40 will have a more neutral character and is suitable for a wide range of applications. The peaked response of the MD 421 II in contrast shows a mic with a "presence peak" beginning at around 2 kHz to add emphasis to upper mid-range frequencies and increase vocal intelligibility. Also, certain mics feature bass roll-off switches which attenuate low-end rumble without altering the general character of the microphone. These roll-off effects will also be illustrated on the graph by dashed lines.



Sensitivity:

The rating of a microphone's output voltage for a particular SPL input. Sennheiser mics are subjected to 1 kHz sine wave at a level of 1 Pascal which is the same as saying 94 dB SPL. The resulting output voltage of the microphone is given in millivolts and expressed as "1.8 mV/Pa" for example. This provides a means of comparison between microphones.

Maximum SPL:

A measure of how much Sound Pressure Level (SPL) the microphone can tolerate before distortion. A 1 kHz tone is presented to the mic and the maximum output prior to distortion is measured. The spec sometimes written as a two-part figure, for example: "140 dB SPL, 1 kHz @ 0.5% T.H.D.

Impedance (Nominal):

The lower the better. You want an impedance of at least five times lower than that of the console you're connecting the microphone to. The term "nominal" refers to an average since impedance changes slightly over the frequency range. A lower impedance value also means you can run a longer cable without loss of signal integrity.

Remember: Regardless of the number of mics you use and whichever miking techniques you employ, experimentation is the key to optimizing sound.

Microphone introduction courtesy of Sennheiser USA.

Microphone Setup and Miking Techniques

3-to -1:

Whenever multiple mics are used in a stage setup, the 3-to-1 rule may be employed to minimize phase cancellation. The rule states that the distance between mics must be three times the distance from each microphone and it's sound source. The sound picked up by the more distant mic will be at least 12db less than the sound picked up by the closer mic. Generally, the 3-to-1 rule is based on use with omni-directional mics It can be relaxed to some extent for unidirectional mic use.

Stereo Mic Techniques:

Stereo miking refers to a number of miking setups where we position and aim usually two microphones to capture left-to-right soundscapes of one or more sound sources. The auditory nerves inside our ears can detect phase and amplitude differences from played back sounds recorded in stereo. This allows us to localize sounds that are wider, spacious and more pronounced in the mix. Stereo miking is sometimes used when the size of an ensemble makes close-miking impractical.

There are four common miking arrangements for stereo recording: XY or Coincident Pair, Near Coincident Pair, Spaced Pair and MS (mid-side):

XY or Coincident Pair: XY miking uses two directional or cardioid mics angled between 90 and 135° apart and placed as close together as possible. Sound hits both mic capsules at the same time, hence the term coincident to describe the technique. However, sounds to the left or right of center hit one mic on-axis and one mic off-axis resulting in an audible level discrepancy between both capsules, creating a stereo image.

Near Coincident Pair: The only difference between XY and Near Coincident Pair stereo miking is the introduction of space between both mic capsules. This effectively creates a time difference and level difference between both receiving capsules. The result is a well-defined stereo image that has more depth and sharpness than a coincident pair. There are several industry references for this type of setup. For instance, French Broadcasting utilize the ORTF technique which specifies angling cardioid mics at 110° with a capsule spacing of approximately 7 inches.

Spaced Pair: This stereo miking arrangement points two mics in the same direction with a large space between them. Only sounds emanating from the direct center arrive at both mics at the same time. Sounds to the left or right of center arrive sooner or later than the other. As a result, time delay cues occur between capsules while no significant amplitude differences occur. Usually, two omni-directional mics are positioned at least two feet from each other or as distant as 10 to 12-ft. from each other. Sometime a very large spacing between two mics produces a sonic hole and a third microphone may be added.

MS (Mid-Side) Pair: Mid-side stereo miking is a specific type of coincident pair miking using one cardioid and one bi-directional mic. This arrangement positions the bi-directional mic at 90° to the sound source while the cardioid is pointed directly at the sound source. This miking technique is more complex because it requires a decoding process for playback. Basically, the bi-directional mics output is split and one side is phase-reversed. Upon playback, the two mic outputs must be panned hard left and hard right. The resulting stereo image has outstanding center stage sharpness and localization accuracy.

STAGE & STUDIO CONDENSER MICS

C 451 B Classic Condenser Microphone



Tapping the sound of a legendary capsule, AKG has created a modern version of the classic microphone. Combining modern design with a classic sound, the C 451 B is the modern interpretation of AKG's former C451 EB + CK1 modular microphone. Back by popular demand, the C 451 B duplicates the C 451 EB + CK1's acoustics, while incorporating state-of-the-art electronics in an integrated mechanical design (not a modular one). The result is dramatically improved specifications with outstanding price/performance while the legendary response of the CK1 capsule—including the characteristic slight high-frequency boost —is maintained. At home anywhere from commercial recording facilities to the personal studio and live touring sound, the cardioid C 451 B offers a switchable 10dB/20dB pad and a 12 dB/octave bass low frequency cut filter. Transformerless output ensures a flat uncolored low frequency response and eliminates potential distortion from low frequencies at high SPL's. Ideal for any application where sounds, particularly their transient content, need to be captured with maximum accuracy, such as overhead miking of drums, percussive instruments, or guitar. The C 451 B's airy sound, high overload limit of the capsule and electronic circuitry, and improved engineering are a solid foundation for creating another legend.

In the 1960s, US sound engineers regarded big, heavy microphones as the epitome of acoustic quality while their European colleagues demanded much smaller microphones. Large-diaphragm transducers have a characteristic response that is perfectly suited for the natural sound of certain instruments and provides a pleasant, warm sound. However, if a sound is to be captured with maximum accuracy small mics are superior because the vibrations of a small diaphragm are much easier to control. In addition, small mics cause much less turbulence within a sound field.

In 1969 a compromise was reached and the C 451 was launched. The C 451 system (also called the CMS Condenser Modular Series) comprised a preamp and several interchangeable screw-on capsules. It was made until 1993 and then replaced with the AKG Blue Line. However, the specific response of the CK1 capsule has remained in demand until this day.

- Built-in switchable 10 dB/20 dB pre-attenuation pad increases SPL capability when close-miking high-energy sound sources or driving preamps or mixers with limited headroom.
- Switchable highpass filter at 75 Hz or 150 Hz prevents low-end distortion that may be caused by rumble or wind noise. Sensitivity is 9 mV/Pa and maximum SPL is 135 dB.
- Highly insensitive to handling noise. The all-metal body provides efficient protection from RF interference and makes it rugged enough to give excellent results even under very harsh conditions.
- Like the original C 451 EB + CK 1, the C 451B features a sandblasted, nickel plated all-metal body with engraved lettering (not the printed lettering on most current AKG microphones).
- Comes standard with an individually traced frequency plot, SA 40 stand adapter and W 90 foam windscreen.
- Optional computer matched pairs are available in a special carrying case that includes all accessories plus an H 50 stereo bar for stereo miking applications.

Microphones That Sound Better

Designing microphones and headphones is both a science and an art. This is a concept well understood at AKG. Over the past 50 years, they have consistently married the two to create some of the truest and most natural-sounding microphones in history. AKG was founded in 1947 in the city of Vienna, Austria, home to some of the finest composers of all time—Beethoven, Mozart, Brahms, Haydn, Schubert, Mahler, Schoenberg and, of course, the Strauss family, to name a few. When it comes to crafting products such as theirs, the critical judgements of a finely trained ear—something Vienna has no shortage of—can make the crucial difference between a good microphone and a legendary one. The recipient of over 1400 patents, AKG's research and development laboratories are the largest, best-equipped and most productive of their kind in the world. They employ more than 50 engineers in full-time research and development, and provide them with the most advanced tools available for computerized design, testing and performance simulation.

STAGE & STUDIO CONDENSER MICS

C 1000S Dual Pattern Microphone



A "do-it-all" microphone famous for its flexibility, the C 1000S is a condenser mic designed for assignment on stage or in the studio. By placing the supplied acoustical converter over the end of the mic capsule— its polar pattern can be quickly converted from cardioid to hypercardioid —without affecting it's on-axis response. The on-axis response of the C 1000S has a gentle (-6 db/octave) roll-off below 200 Hz which has been tailored for the normal close-in use of cardioid and hypercardioid microphones in studio recording and sound reinforcement applications. The mic also includes an attachable Presence Boost Adapter (PB1000) for increased intelligibility.

- Rugged construction qualifies the C 1000S for demanding road work, yet pristine condenser response and low noise make it ideal for recording in home/project studios.
- High sensitivity, high SPL handling with low distortion (137 dB for 1.0 % THD).
- Extended low and high frequency response is tailored for close proximity to sound sources.
- Presence Boost Adapter (PB1000) can be attached to create an emphasis of the higher frequencies around 5 to 9 kHz by 3 to 5 dB. This not only improves intelligibility but is also advantageous when recording instruments with high frequency content.
- PPC1000 Polar Pattern Converter allows quick change from cardioid to hypercardioid when a more directional pickup pattern is required.
- Powered by 9-volt battery or 9 to 52v DC phantom power.
- Silent on/off switch is recessed to prevent inadvertent shutoff.
- Includes PB1000, PC1000S, A43 stand adapter, foam windscreen, and gig bag.



C 2000 B Cardioid with Roll-Off and Pad

An affordable high quality recording microphone, the C 2000 B is a small diaphragm condenser that captures the high frequency harmonic detail and true transient nature of the sound source while at the same time offers the higher sensitivity, low noise and extended bass response (flat to 100Hz) of a large diaphragm condenser. Ideal for acoustic instruments and percussion on stage or in the studio, the C 2000 B's almost ruler-flat response provides a crystal-clear, "up-front" sound. A switchable bass rolloff filter for eliminating proximity effect and a switchable 10dB pre-attenuation pad let you adjust the mic's response to all recording or live miking situations. Built-in pop screen reduces unwanted noise that may be caused by plosive sounds when you use the mic extremely close-in. Rugged construction, an elegantly styled die-cast metal housing, and silver-gray finish add to the professional level quality of the C 2000 B.

- Versatile, side-address cardioid condenser mic designed for professional and home studio applications or on-stage.
- Cardioid pattern provides an extremely flat on-axis frequency response with a 3 to 4 dB upper high-frequency boost between 8 and 12 kHz that adds clarity, even with ambient miking.
- One-half inch capsule element placed in a large baffle ring gives the C 2000 B more forward directionality at high frequencies.
- An integral bass cut switch provides a gentle 6 dB per octave roll-off below 500 Hz, compensating for the proximity effect (low frequency boost) associated with close miking.
- Added presence at very high frequencies is an asset in many miking situations, especially home recording studios.
- Self-noise floor is 20 dB (A), giving the mic an effective dynamic range of 120 dB.
- High SPL level 140 dB (150 dB with the integral 10 dB pre-attenuation pad engaged).
- Internal wiremesh grills and a massive metal housing protect against unwanted electrostatic interference.
- Includes SA41/1 stand adapter. Optional accessories include the B18E battery supply, and H100 external elastic shock suspension.

1″ LARGE DIAPHRAGM CONDENSER MICS

C 3000 B Cardioid with Roll-Off and Pad



Combining innovative design and efficient manufacturing, the all-purpose C 3000 B offers premium sound quality and durability for stage and studio—but at a price far below that of a premium studio mic. Ideal for cost-conscious performers and home/project studio owners, the C 3000 B's capsule design features a gold-sputtered 1″ mylar diaphragm to capture the distinctive character and purity of sound that is pure AKG. Sound is smooth, clear and open, with a vibrant presence that projects to the front of a mix. Bass roll-off and pre-attenuation switches let you tailor the mic for stage or studio. An internal suspension supports the capsule assembly, an integrated windscreen allows outdoor use, and a safety flange provides secure connections in hanging applications. Rugged die-cast construction and scratch-resistant silver-gray matte finish make the C 3000 B well suited to the rigors of touring as well.

- Gold-sputtered 1″ (25mm) diameter mylar foil offers the warmth and transparency typically found in AKG mics.
- Capsule is internally shock mounted for low handling noise, while an internal foam sheath minimizes breath noise/pop with minimal attenuation of high frequencies.
- Mic is transformerless and accepts phantom power from 9 to 52v DC.
- Tailored for the modern studio, the C3000B has the typical AKG large diaphragm "signature contour," a slight dip at 2.5 kHz followed by a rise at 5 - 6 kHz and 9 - 11 kHz.
- Very low self noise floor of 14-dB-A and maximum level capability of 140 dB, produces an overall dynamic range of 126 dB. Via the 10 dB pad, mic range can be shifted up to 150 dB for close instrument miking.
- Cardioid pickup pattern is extremely well controlled, providing excellent off-axis rejection across the entire frequency band.
- A bass cut switch provides a gentle 6-dB per octave rolloff below 500 Hz, compensating for proximity effect in vocal recording.
- Supplied with H100 elastic suspension mount. Optional accessories include B18 battery, pop filter, and wind screen.



C 4000 B Multi-Pattern Microphone

Designed for studio use, the multi-pattern (omnidirectional, cardioid, and hypercardioid) C 4000 B condenser mic is equally at home in broadcast operations or on stage for the highest quality vocals. A true 1″ large diaphragm premium microphone, the C 4000 B's capsule has been carefully tuned to preserve the integrity of the source while offering mild compliments in the high frequencies.

Open, clear and natural highs coupled with smooth, accurate lows make the C 4000 B an exemplary choice when employed on vocals and on a varying array of string instruments, brass and woodwinds. With all the on-board options the C4000B has the versatility to take on many tasks in commercial, project and personal studios as well as on a stage and in the broadcast studio.

- 1″ thick gold-sputtered diaphragms provide extended high frequency performance.
- Transformerless output ensures low frequency accuracy and nearly non-existent self-noise of 8 dB-A.
- Switchable 12-dB/octave low cut filter at 100 Hz removes rumble or excessive proximity effect.
- Maximum SPL is 145 dB and overall dynamic range is an incredible 137dB—enough headroom for all day-to-day recording situations.
- Switchable 10 dB pad extends operating range even more. Use for distortion-free reproduction of extremely high sound pressure levels (up to 155 dB SPL)
- Mic is transformerless and accepts phantom power from 9 to 52v DC.
- Capsule is internally shock mounted for low handling noise, while an integrated windscreen reduces wind and breath noise.
- Includes the H 100 spider-type shock mount and W 4000 windscreen for demanding situations.

1″ LARGE DIAPHRAGM CONDENSER MICS

C 4500 B-BC "On Air" Broadcast Microphone



The C 4500 B-BC was developed to meet the demands associated with digital broadcasting. A front-address, large-diaphragm condenser microphone, the C 4500 B-BC is immune to electrostatic and magnetic fields and provides an extremely wide dynamic range and low self-noise. With the ever-increasing number of broadcast stations becoming "digital"—at least with their automatic control facilities"—the on-air studios are more and more clustered with control computers. These computers and their associated monitors radiate electro-magnetic RF fields which can be a serious problem for nearby microphones. The C 4500 B-BC solves this problem with an all-metal housing and double screening of all acoustically open sections of the housing. Additional features include internal shock mounting, switchable -20dB pre-attenuation pad, and a switchable low-frequency roll-off (6 dB/octave below 120Hz).

- Main axis for sound entry is in the front of the mic. This allows the C 4500 B-BC to be positioned in front of the talent's mouth without impeding their field of vision
- Multi-layer pop filter between the front grille and the capsule provides excellent suppression of vibrations and pop noise
- 6 dB per octave below 120 Hz bass-cut filter for frequency roll off and additional reduction of proximity effect
- Switchable -20 dB pre-attenuation pad for undistorted reproduction of extremely high sound pressure levels
- Operates on phantom power (9 to 52v DC)
- Extremely "voice-friendly" acoustics, requires no further processing after the mic
- Supplied with the H 100 spider-type shock mount and W 4000 windscreen for demanding situations

SOLIDTUBE 'Experience the Warmth'



It has been said that the SOLIDTUBE is what a modern large diaphragm condenser microphone would look like if the transistor had never been developed. Everything about it is in fact modern, except the use of the ECC 83 (12AX7) vacuum tube. Designed from the ground up as a new expression of the tube philosophy—sound warmth and the generous margin of overload that tube/transformer output stages are known for—only the mic's power supply makes use of solid state components. Much of the "signature" of large format condenser microphones lies in the geometry that surrounds the diaphragm assembly.

The SOLIDTUBE has a large housing/screen assembly which reacts minimally with the capsule, creating a broad +6 dB peak in response only above 10 kHz. Over most of the range the response lies within a window of ±2 dB. Midband pattern rejection at 180° is in excess of 20 dB. The microphone is response tailored for solo vocal and instrumental pickup in the studio, and as such has only a dedicated cardioid pattern.

- Dedicated cardioid pattern with response tailored to solo vocals and instrumentals in the modern studio
- Sensitivity is 20 mV/Pa and maximum SPL is 130 dB.
- Switchable -20 dB pre-attenuation pad for undistorted reproduction of extremely high sound pressure levels (up to 145dB SPL)
- Wide dynamic range of 110 dB in the studio
- Switchable 12-dB/octave low cut filter at 100 Hz removes rumble or excessive proximity effect
- Includes N-SOLIDTUBE power supply, all necessary cabling , an isolation shock mount, foam windscreen and a custom aluminum flight case

C-414 B-ULS / C-414 B-TLII

Multi-Pattern Large-Diaphragm Reference Microphones — The Classic Twins



The worldwide studio standard, the C 414 B-ULS delivers flawless performance and uncommon flexibility in the most demanding studio and concert sound applications. The crisp, full-bodied, and naturally open character of the C 414 B-ULS makes it a preferred choice for critical recording or reinforcement of vocals, piano, percussion, and other instruments with delicate transient content. The most ubiquitous studio condenser microphone in the world, the C 414 B-ULS is designed around dual back-to-back large one-inch, gold-sputtered diaphragms for multiple pickup patterns. The four selectable polar patterns (omni-directional, cardioid, hypercardioid, figure-8) precisely tailor response for maximum versatility, while two bass filter settings suppress unwanted low frequencies. Extremely low self-noise (14 dB-A) and high overload points combine for a remarkably wide 126 dB dynamic range. Switchable 10 or 20 dB internal pre-attenuation pads allow use on instruments producing over 150 dB SPL.

Both mics include windscreen, spider suspension, and individually tested frequency response graphs.



The C 414 B-TLII combines legendary C 12 acoustics and the latest generation of transformerless FET electronics. Although similar to the C 414 B-ULS in shape and features, the C 414 B-TL II differs in two important respects. First, the capsule is a faithful sonic recreation of the one used in AKG's classic C12 tube microphone. Above 3 kHz, the C 414 B-TL II takes on vintage C12 characteristics, while below this point the response is ruler flat. Additionally, AKG's computer-aided manufacturing techniques assure greater uniformity in response from microphone to microphone (early CK12's were all slightly different in response, a consequence of individual hand-crafting). Second, the "Vintage TL" incorporates transformerless electronics for extended low-end reproduction with extremely low distortion. The result is a remarkable microphone that exceeds all requirements for digital recording, yet also captures the warmth, smoothness, and delicate openness cherished by artists the world over. Switchable polar patterns, pre-attenuation pads, and bass roll-off filters are the same as in the C 414 B-ULS.

	C-2000	C-3000	C 4000 B	C 4500	C 414 B/ULS	C 414 B/TLII
Polar Pattern	◡	◡	◡, ◔, ○	◡	◡, ◔, ○, ⧖	◡, ◔, ○, ⧖
Frequency Range	30 to 20,000 Hz	20 to 20,000 Hz	20 to 20,000 Hz	30 to 20,000 Hz	20 to 20,000 Hz	20 to 20,000 Hz
Sensitivity	20 mV/Pa (-34 dBV)	25 mV/Pa (-32 dBV)	25 mV/Pa (-32 dBV)	20 mV/Pa (-34 dBV)	12.5 mV/Pa (-38 dBV)	12.5 mV/Pa (-38 dBV)
Max SPL (THD=0.5%)	140 dB	140 dB	145 dB	145 dB	140 dB	140 dB
Equivalent Noise Level *	20 dB	14 dB-A	8 dB-A	8 dB-A	14 dB-A	14 dB-A
S/N Ratio*	74 dB	80 dB	86 dB	86 dB	14 dB-A	80 dB
Pre-attenuation Pad	10 dB, switchable	10 dB, switchable	10 dB, switchable	20 dB, switchable	10/20 dB, selectable	10/20 dB, selectable
Bass Rolloff Filter	6 dB/octave below 500 Hz	6 dB/octave below 500 Hz	12 dB/octave at 100 Hz	6 dB/octave at 120 Hz	12 dB/octave at 75 or 150 Hz	12 dB/octave at 75 or 150 Hz
Impedance	200 ohms	200 ohms	200 ohms	200 ohms	180 ohms	180 ohms
Recommended Loading	>1000 ohms	>1000 ohms	>1000 ohms	>1000 ohms	>1000 ohms	>600 ohms

** A-Weighted*

C12VR

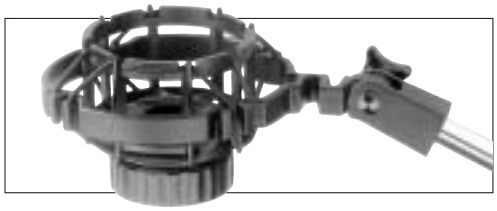
Vintage Re-Issue Tube Microphone



Production of the original C 12 ceased in 1963, and over the years it gradually became a sought-after collectors' item—a true legendary recording instrument acclaimed worldwide for its exquisite sound. The new C 12 VR is true to the original C 12's electronic and acoustic design with only subtle refinements (made possible by modern manufacturing techniques and improvements in basic circuit components) made to increase overall robustness and reduce susceptibility to hum and noise.

With the recreated 1˝ twin diaphragm capsule design and the same 6072 vacuum tube at the heart of its circuitry, the C 12 VR ensures that even the most discriminating users will enjoy the trademark warmth, clarity and presence that characterized the original. Nine different polar patterns are available, any of which may be remotely selected from the N12 VR powering unit. Other user control features include two bass roll-off filters (also remotely selectable), switchable pre-attenuation pads, and a switchable 10 dB boost in nominal sensitivity. A large core-section output transformer minimizes low frequency distortion, and special shock mounting elements protect circuitry and provide acoustic isolation.

Supplied with power/remote unit, H15/T elastic shock mount suspension, 30´ cable, W42 foam windscreen, individually tested frequency response graph and a rugged aluminum carrying case.



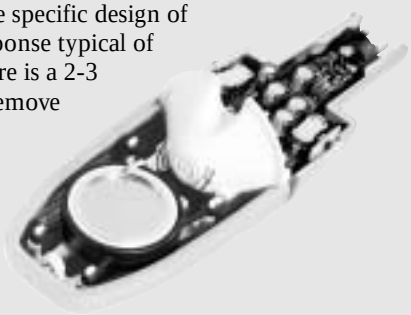
H100 Spider Suspension

Supplied with the C 2000 B, C 3000 B, C 4000 B and C 414 B-TL II, the H100 is a universal shock mount that can be used for all microphones with shafts from 0.85 to 1˝. The H100 Spider Suspension accepts most microphones, fits any standard mic stand, and is adjustable to any angle.

AKG Large-Diaphragm Technology



The heart of any high quality studio condenser microphone is its transducer capsule. The specific sound of a microphone depends on the way the transducer diaphragm is made and installed. For over 50 years, AKG has been making large-diaphragm condenser microphones with a vibrating diaphragm diameter of one inch (25mm). Electrical contact is established by the clamping ring so the entire diaphragm can vibrate freely.



The secret behind AKG's pristine sound is the specific design of the diaphragms that shape the frequency response typical of AGK microphones. Between 2 and 3 kHz there is a 2-3 dB dip in sensitivity that is wide enough to remove any harshness and aggressiveness from the human voice. In addition, the diaphragm is designed to boost the frequency range from 4 kHz to 15 kHz. This high frequency emphasis contributes to actual and perceived "up-front localization" as demonstrated by measurements and listening tests. This is why AKG microphones are so popular worldwide. They will cut through the mix naturally and the signal will be up front, even at low levels.

ULTRA LINEAR SERIES (ULS)

Modular Condenser Microphone System

Based around the C 480 B preamplifier powering unit, the ULS system combines unparalleled performance with exceptional flexibility. Offering switchable cardioid, hypercardioid, omnidirectional, and highly directional shotgun elements, the ULS Series is designed for studio, broadcast and a variety of live sound and speech reinforcement applications in which state-of-the-art performance is mandated.
Patented electronic circuitry coupled to a transformerless output stage assures completely linear transfer characteristics across the entire audio bandwidth, while achieving very low self-noise. This, in combination with a typical SPL handling capability of 140 dB means a dynamic range exceeding 125 dB, making the system suitable for the most critical digital recording applications. Gold plated switch and connector contacts guarantee consistent performance. A specially designed output stage drives excessively long cables with no signal degradation. A complete line of accessories is available.



C 480 B

C 480 B Preamplifier

The C 480 B accommodates the CK 61-ULS, CK 62-ULS, CK 63-ULS and the CK 69-ULS capsule. Newly designed, the preamp has a vastly improved noise floor, permitting self noise performance with the four capsules in the range of 9 to 11 dB-A. Maximum sound pressure levels in the range of 144 dB (with the -10 dB pad) can be attained with no greater than 0.5 % distortion. It has an integral low cut filter (12 dB/octave) can be set for either 150 or 75 Hz, and a three position control for normal (0 dB), elevated sensitivity (+6 dB), and attenuated (-10 dB) sensitivity. The C 480 B operates on 48v phantom powering.

CK 60 SERIES CONDENSER CAPSULES

Each of the "ULS" series capsules must surpass the strictest tolerances for accurate presentation of both on and off-axis source material. A three micron, gold sputtered diaphragm utilizes AKG's uni-tension, double-stretched mylar for superb transient and frequency response unaltered by temperature and humidity shifts.







CK 61-ULS Cardioid Capsule

The broadband, ultra linear frequency response and uniform polar pattern of the CK 61-ULS make it ideal for critical recording/reinforcement applications where suppression of off-axis sound is desired. Includes the W 32 foam windscreen.

CK 62-ULS Omnidirectional Capsule

Same as CK 61-ULS, except for omnidirectional polar pattern. For more reverberant sound recordings. Includes W 32 foam windscreen.

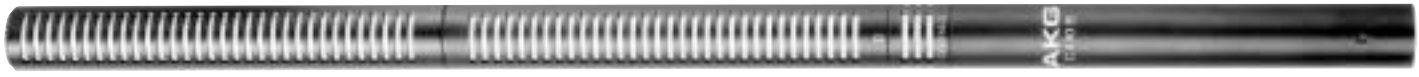
CK 63-ULS Hypercardioid Capsule

Same as CK 61-ULS, except for hypercardioid polar pattern. Better off-axis rejection provides better channel separation and in many cases, higher gain before feedback. Includes W 32 foam windscreen.

CK 69-ULS 2-in-1 Shotgun Capsule

Two shotgun capsules in one, with quick and easy conversion from long to short shotgun applications without compromising sonic integrity. In its long shotgun configuration, the CK 69-ULS delivers perfect results for outdoor applications—getting sounds in sync with picture—or indoor front of stage recording/reinforcement from approximately 20 to 24´.. Unscrew the front half of the interference tube and the same capsule becomes a short shotgun for use in TV, film and video close-up work. Additionally, the short shotgun mode resolves the problems associated with performing interviews in noisy environments.

The CK 69-ULS operates flawlessly in high humidity environments, and windscreens are provided for both long and short configurations. Convenient, quiver-type carrying case accommodates capsules, powering module, and W 48 and W 49 windscreens. Includes original frequency response curves. Matte black finish.



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EQUIPMENT LEASING AVAILABLE

BLUELINE SERIES

Modular Condenser Mic System

The Blue Line Series consists of directional prepolarized condenser capsules and microphone configurations that are all compatible with a single powering/output assembly. With their unique snap-on attachment and modular construction, you can change capsules with different characteristics in seconds. Eight different capsules and a complete line of optional accessories —including an extension cable for operating the mic capsule up to 10´ from the powering/output module—offer a solution for every recording situation. Ideal for studio, live sound, and a variety of speech reinforcement applications —especially where the facility for switching capsules is a major advantage.

- The positive Modulock bayonet design allows changes to be made quickly
- Capsule designs employ gold sputtered diaphragms
- Self noise and maximum operating levels are exemplary, creating a very wide usable dynamic range for the most demanding operations. They also exhibit accurate axial response and uniform pattern control



SE 300 B Transformerless Powering/Output Module

The SE 300 B powering/ output assembly is for use with all AKG Blue Line modular capsules. The SE 300 B operates from any 9-52V phantom power supply, and incorporates a 3-position switch offering linear response, -10dB pre-attenuation and 12dB/octave 75Hz bass-cut filter. Because of the general popularity of the cardioid pattern, the CK 91 cardioid capsule and SE 300 B powering/output assembly are sold as a single unit: the C 391 B microphone.

CK 91 Cardioid

The flat frequency response and uniform polar pattern of the CK 91 make it ideal for general applications where suppression of off-axis sound is desired.

CK 92 Omnidirectional

The CK 92 offers exceptionally neutral presentation of a source in its ambient sound field, providing uniform results regardless of the distance between microphone and source. This characteristic, combined with linear frequency response across the entire audio band, makes the CK 92 an excellent choice for capturing the natural acoustic subtleties of a recording environment.

CK 93 Hypercardioid

Similar to the CK 91, the CK 93 provides additional attenuation of 90° off-axis sounds. In addition to a higher ratio of direct to ambient sound, it also offers a virtually ruler-flat frequency response.

CK 94 Figure-Eight

Single diaphragm membrane construction provides precise reproduction of information from opposing directions. Excellent on-axis response combined with linear rejection of side information make the CK 94 ideal for Blumlein and MS stereo recording, studio vocals, and broadcast interviews.

CK 97-O Omnidirectional Lavalier

This tiny lavalier capsule has linear bass and midrange response, while a gentle rise around 8 kHz crisply accentuates voices. Omnidirectional pattern allows natural sound reproduction with no proximity build-up, while high sensitivity and low self-noise combine for exceptional S/N performance.

CK 97-C Cardioid Lavalier

The CK 97-C is ideal for a variety of sound reinforcement and film set applications. It offers a bass roll-off at 400 Hz to reduce chest resonance, gentle 10 kHz rise to improve vocal clarity, and uniform cardioid pickup pattern for excellent gain-before-feedback.

CK 97-CVR

Acoustically identical to the CK 97-C, the CK 97-CVR incorporates the same capsule into a 15-inch mini gooseneck to create an unobtrusive, high quality microphone for podium, pulpit or boardroom use.

CK 98 Short Shotgun Capsule

The CK 98 combines high sensitivity and controlled directivity. It offers excellent reach, thanks to a very tight polar pattern and exceptionally low self noise. Small size and light weight make the CK 98 ideal for boom applications, while the smooth, wideband frequency response makes it a superb choice for edge-of-stage and ceiling placements.

SPECIFICATIONS

ULTRA LINEAR SERIES (ULS)

Specifications (measured with C 480 B preamp)

	CK 61-ULS	CK 62-ULS	CK 63-ULS	CK 69-ULS
Transducer Principle	Externally polarized condenser capsules			
Frequency Range	20–20k Hz ± 1 dB	20–20k Hz ± 1 dB	20–20k Hz ± 1 dB	20–18k Hz ± 2 dB
Polar Pattern				Shotgun (2-section)
Sensitivity at 1000 Hz				
(0 dB setting)	20 mV/Pa; -34 dBV	20 mV/Pa; -34 dBV	20 mV/Pa; -34 dBV	27 mV/Pa; -31 dBV
(+6 dB setting)	40 mV/Pa; -28 dBV	40 mV/Pa; -28 dBV	40 mV/Pa; -28 dBV	54 mV/Pa; -25 dBV
(-10 dB setting)	6.3 mV/Pa; -44 dBV	6.3 mV/Pa; -44 dBV	6.3 mV/Pa; -44 dBV	8.5 mV/Pa; -41 dBV
Equivalent Noise Level				
(0 dB setting)	11 dB-A	11 dB-A	11 dB-A	9 dB-A
(+6 dB setting)	13 dB-A	13 dB-A	13 dB-A	11 dB-A
(-10 dB setting)	17 dB-A	17 dB-A	17 dB-A	15 dB-A
Max. SPL for 0.5 % THD				
(0 dB setting)	140 dB	140 dB	140 dB	140 dB
(+6 dB setting)	134 dB	134 dB	134 dB	134 dB
(-10 dB setting)	144 dB	144 dB	144 dB	142 dB
S/N Ratio				
(0 dB setting)	83 dB	83 dB	83 dB	85 dB
(+6 dB setting)	81 dB	81 dB	81 dB	83 dB
(-10 dB setting)	77 dB	77 dB	77 dB	79 dB
Finish	Non-reflective black	Non-reflective black	Non-reflective black	Non-reflective black
Capsule Dimensions	.83 x 1″	.83 x 1″	.83 x 1″	.83 x 6.9″ (single section) .83 x 12.5″ (dual section)

BLUELINE SERIES

Specifications with SE300B

	CK91 B	CK 92	CK 93	CK 94	CK 97-0	CK 97-C / 97-CVR	CK 98
Transducer Principle	Prepolarized condenser capsules						
Frequency Range	20–20kHz	20–20kHz	20–20kHz	20–20kHz	20–18kHz	150–18kHz	20–20kHz
Polar Pattern							
Sensitivity at 1000 Hz	10 mV/Pa (-40 dBV)	10 mV/Pa (-40 dBV)	10 mV/Pa (-40 dBV)	10 mV/Pa (-40 dBV)	10 mV/Pa (-40 dBV)	12 mV/Pa (-38 dBV)	25 mV/Pa (-32 dBV)
Equivalent Noise Level	17 dB-A	17 dB-A	17 dB-A	22 dB-A	24 dB-A	26 dB-A	17 dB-A
Max SPL for 1 % THD	132 dB	132 dB	132 dB	132 dB	120 dB	120 dB	124 dB
S/N Ratio (A-weighted)	77 dB	77 dB	77 dB	72 dB	70 dB	68 dB	77 dB
Electrical Impedance	< 200 ohms	< 200 ohms	< 200 ohms	< 200 ohms	< 200 ohms	< 200 ohms	< 200 ohms
Load Impedance	> 1,000 ohms	> 1,000 ohms	> 1,000 ohms	> 1,000 ohms	> 1,000 ohms	> 1,000 ohms	> 1,000 ohms
Power Requirement	9 - 52 Vdc Phantom powering for SE 300 B						
Connector	All models XLR male output when connected to SE 300 B						
Cable Length	—	—	—	—	10′	10′/—	—
Finish	Dark gray matte	Dark gray matte	Dark gray matte	Dark gray matte	Dark gray matte	Dark gray matte	Dark gray matte
Capsule Dimensions	.7 x 2″	.7 x 2″	.7 x 2″	.7 x 2.3″	.3 x 0.7″	.4 x 1.0″	.4 x 10.2″

With the 10-dB pad engaged, all SPL values for 1.0 % THD can be raised 10 dB. Specifications measured with capsules attached to SE 300 B.

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SPECIAL APPLICATION MICS

C 426 B Stereo Microphone



A compact and highly refined mic system designed for the most critical single-point stereo recording applications. A direct descendent of AKG's renowned C 422 and C 24 large diaphragm stereo condensers, the C 426 B incorporates new technology for significant improvements across virtually all performance parameters: dynamic range, self-noise, sensitivity, linearity, frequency response between channels, and crosstalk. Two gold-sputtered twin-diaphragm capsules are arranged in a coincident pair; the microphone head may be rotated 45° against the shaft and the upper system rotated 270° against the lower to facilitate quick changeover from M/S to X/Y techniques. Two tightly focused LEDs mounted on each rotating capsule provide quick visual verification of capsule angle from a distance. Switches on the mic housing select bass roll-off (12 dB/octave at 75 Hz) and –10 dB pre-attenuation. Included remote control system allows completely noiseless selection among the nine polar patterns available for each capsule. Complete system also includes shock mount/stand adapter, foam windscreen, 66´ connecting cable, and rugged carrying case.



C 568 EB Shotgun Microphone

High directivity and wide frequency response in a relatively short integrated unit make this an ideal microphone for theater, sound reinforcement, and film/video recording. Due to its compact and lightweight design, the C 568 EB is ideal for camera mount applications. The C 568 EB is a backplate electret condenser design which operates on both pressure gradient and interference principles. Below 500 Hz it functions as a hypercardioid microphone; above 500 Hz the acoustic interference tube introduces increasing directivity. A switch selectable 12 dB/octave 150 Hz roll-off filter effectively suppresses rumble and wind noise. Robust, all-metal housing contains the built-in preamp/impedance converter. Phantom powering (9-52v) required. Foam windscreen and stand adapter included.

D230 Rugged Dynamic ENG Microphone

A robust ENG (electronic news gathering) mic that incorporates a high output rare earth magnet structure designed to deliver exceptional sensitivity and clarity in the demanding news gathering field. Its response has been tailored for maximum intelligibility of close-in speech, and its omnidirectional pattern is immune to proximity effects. An internal hum compensation coil allows microphone use in environments where ac hum fields may be significant. It has an integral windscreen and is designed for very low handling noise. The extended shaft allows for easy placement of station flags without typical handling problems. The housing is made of an extremely rugged die cast metal and has a dark matte finish for minimum reflection or glare on television. Comes complete with stand adapter.

	C426 B	C568 EB	D230
Frequency Range	20–20kHz	20–20kHz	40–20kHz
Polar Pattern	∞, ○, ∞ (with 6 intermediate positions)	∞/∞	○
Sensitivity	11 mV/Pa (-39 dBV) (each channel)	11 mV/Pa (-39 dBV)	2.5 mV/Pa (-52 dBV)
Bass Cut Filter	12 dB/octave roll-off at 75Hz	12 dB/octave below 120 Hz	—
Equivalent Noise Level	14 dB-A	18 dB-A	—
Max SPL for 0.5 % THD	132 dB (142 dB at-10 dB)	128 dB	—
Impedance	200Ω	600Ω	320Ω
Power Requirement	9-52 V phantom power	9 to 52 V phantom power	N/A
Current Consumption	approx. 2 mA per channel	<2 mA	N/A
Size; Net Weight	1.65/1.3 Ø x 9.25˝; 1.04 lbs.	0.8 Ø x 10˝; 5.6 oz.	2.0Ø x 8.5˝; 8 oz.

EMOTION SERIES

'The Sound That Rocks'

AKG's Emotion series incorporate a unique design and the revolutionary Varimotion process to far surpass the performance parameters of other microphones. In addition, handling and cable noise is all but eliminated by AKG's Doubleflex Anti-Vibration System. The end result are mics with the performance characteristics of an expensive studio mic at the price of a music store standard. Patented technology, the Tiefzieh Varimotion System allows AKG to produce a diaphragm of varying thickness in a single press. By doing so AKG almost eliminates the need for tuning resonators so that the microphone's sonic characteristics can now be tuned at the capsule, effectively reducing the cost of production by more than half. The microphones come in a rugged all metal housing, some utilizing 24 karat gold plated connectors. Last, they feature a neodymium magnet system to provide an extremely powerful output signal. This will make your vocals cut through the loudest of rhythm sections. For vocal or instrumental music, for moderators, lead, or backing vocalists, Emotion Series mics are the ideal tool on stage.



D660

Varimotion Technology

Varimotion refers to the patented process used to create the diaphragm. For the first time, diaphragms can be made whose thickness varies along their diameter. Using this technology, diaphragm thickness can be varied during the manufacturing process resulting in a more accurate and natural reproduction of the sound while also reducing the need for expensive mechanical tuning resonators.

The Doubleflex Shock Mount

A patented dual shock mount system, the extremely efficient Doubleflex shock mount system comprises two separate shock absorbing suspension rings for the transducer. These work together to provide optimum suppression of all kinds of unwanted noise such as handling and cable noise—especially important on stage. Allows the mic to "focus" on your voice rather than picking up unwanted noise.

D 660 S

Dynamic Microphone

The D 660 S is a unidirectional dynamic microphone with rugged die-cast metal body developed specifically for miking speech, vocals, and instruments on stage, and for Karaoke and home recording applications. A professional entry-level vocal microphone, the D 660 S offers a price/performance ratio that is unbeatable.

- The transducer element of the D 660 S has been designed for use with both sound reinforcement systems and cassette or home studio recorders. Neutral low and mid frequency response combined with a 6 dB presence boost between 4 and 8 kHz ensures smooth sound and excellent intelligibility
- Extremely rugged protective basket ensures incredible durability and keeps the highly sensitive transducer out of harms way. It is easy to remove and the internal windscreen is easy to clean
- Uses a dual shock mount to suppress of unwanted noise
- Integrated on/off switch indicates microphone status
- Includes stand adapter and carrying case

D 770

Multipurpose Instrument/Vocal Microphone

The D770 is a rugged dynamic cardioid microphone intended for on-stage instrumental and vocal pickup. Its cardioid pattern is uniform with frequency, and response has been boosted at mid and high frequencies for crisp instrumental pickup and improved speech intelligibility. The microphone has a heavy duty steel screen with an integral diecast support frame. The capsule's elastic Doubleflex suspension eliminates handling and cable noise. Outstanding performance is obtained by means of AKG Varimotion diaphragm technology, a high output magnet, and a super-cardioid polar pattern for excellent gain-before-feedback and superb off axis rejection.



D770

- Frequency response is shaped especially for instrumental/vocal performance. The wide frequency response is slightly boosted in the mid and high frequency ranges for a crisp instrumental sound (wind instruments, mouth-organ, guitar amps, percussion instruments and backing vocals) and for better speech intelligibility
- Utilizing AKG's Varimotion technology, diaphragm thickness varies from 20 microns at the edge to 40 microns at the center. This produces extended bass response, virtually eliminating the need for a tuning resonator
- XLR plug with 24 carat gold-plated pins
- Doubleflex shock mounted transducer system used with the capsule assembly places the primary resonance of the moving system to well below the audible range, resulting in excellent isolation of handling and cable induced noises
- Duraflex spring steel wire-mesh cap with integrated wind an pop screen is resistant to deep denting providing optimum protection for the transducer capsule beneath it
- The light-weight body is made of die cast zinc alloy. Has a dark matte finish for minimum reflection and glare
- Includes a vinyl carrying case and SA 44 stand adapter

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D 880/D 880S

Dynamic Vocal Microphone

Performance microphone with a frequency response designed specifically for lead and backing vocals. Wind noise and sibilance are eliminated by an integrated wind and pop screen. Its outstanding performance is obtained by means of the newly developed AKG Varimotion diaphragm technology and a frequency-independent supercardioid polar pattern that provides very effective feedback suppression. Handling noise is all but eliminated with the Doubleflex anti-vibration system. The D 880/D 880 S comes complete with a vinyl carrying case and SA 44 stand adapter. S version has on/off switch.

C900

Condenser Vocal Microphone

The key to getting your emotions across to your audience is in your hand—the microphone. Knowing this, AKG left conventional wisdom behind and set the standard with the Emotion line of dynamic microphones. Now AKG is raising the standard again with the C 900 condenser microphone.



- A professional condenser vocal microphone, the C900 will make your voice cut through the loudest mix - on any stage
- Features a frequency independent cardioid polar pattern that has not only been optimized for exceptionally high gain before feedback, but also for use with in-ear monitor systems
- The transducer is 24-carat gold plated for optimum protection from corrosion and humidity. The entire transducer element is shock mounted in a special elastic ring
- In addition to the transducer case and decorative ring, the XLR connector is also 24-carat hard gold plated for exceptionally reliable electrical contact
- The C 900 is extremely insensitive to handling noise. A resonance weight tunes the transducer resonance frequency to a value where handling noise is almost totally eliminated
- Presence Boost Adapter can be installed to boost the 5 to 9 kHz range by 2 to 4 dB for even more brilliant high-frequency sound and better intelligibility. This improvement adds to the mic's versatility in vocal performance as well as overhead, cymbal, or hi-hat miking
- The impact resistant die-cast metal body is finished with "soft-touch" enamel that feels pleasant to the touch, while the wire-mesh grill is made of first-rate spring steel to provide added protection from the hardships of touring
- The C 900 package includes the microphone with its rugged die-cast metal body, a stand adapter, PB 1000 Presence Boost Adapter and carrying case

PERFORMER SERIES

AKG's low cost Performer Series includes three mic each designed and manufactured in Austria. Equipped with high output Neodymium capsules and precision AKG engineering these mics meet the needs of professionals at a price accessible to amateurs and beginning musicians. Each mic comes with removable grills for easy washing of windscreens, on/off switches, and extremely rugged housings designed for the rigors of live performance.



D 40 S

Unidirectional dynamic microphone with rugged plastic case. Includes fixed 5m (16´) cable with gold plated mini and 1/4˝ plugs.

D 50 S

Unidirectional dynamic microphone with rugged die-cast metal case and on/off switch. Supplied with fixed 5m (16´) cable with gold plated mini and 1/4˝ plugs.

D 60 S

Unidirectional dynamic microphone with rugged die-cast metal case and on/off switch. Supplied with separate 5m (16´) cable with XLR connector, gold plated mini and 1/4˝ plugs.

C 5900

Tri-Power Condenser Vocal Microphone



The C 5900 combines the detailed transparency of a premium AKG studio condenser with the road tough construction, low-handling noise, and feedback rejection required of a true stage performance microphone. At the heart of the C 5900 is AKG's condenser capsule with an ultra-thin, gold-sputtered, double-stretched diaphragm which precisely captures every subtle nuance of a vocal performance. Transient response is superb, yet vocal quality remains full and warm with no trace of harshness. Vocal contoured response projects to the front of the mix, and a 3-way switch allows performers to customize proximity effect by choosing flat response or one of two bass roll-off curves. A tightly controlled hypercardioid pattern and uniform polar response ensure maximum gain-before-feedback, with consistently predictable performance when used close-in on monitors. A four-layer filter assembly effectively suppresses P-pops and wind noise, while the integrated and hardened protective basket of AKG's DuraShell system safeguards all internal components. AKG's InterSpider three-point internal elastomer suspension isolates the capsule from handling noise so energetic performers can work the stage without worry.

FEATURES

- Rugged condenser mic designed for on-stage instrumental and vocal pickup
- Hypercardioid pattern is uniform with frequency, and response has been boosted slightly at mid and high frequencies for crisp instrumental pickup and improved speech intelligibility
- Mic capsule is shock mounted with a removable wire mesh grill. The smooth triangular shape of the mic is easy to hold and is stable when placed on a surface
- Switchable 6 dB/octave bass rolloff at 150Hz and 12 dB/octave bass cut at 100Hz provide flexibility in dealing with proximity effect
- Light-weight die cast metal body with dark matte finish for minimum reflection and glare
- Broad high frequency rise (4 dB) for added presence
- Internal elastic suspension for reduced handling noise
- Universal phantom powering

C 535 EB Cardioid Mic



The C 535 EB has the clean, open characteristic of AKG's best studio condensers, enhanced by a 2dB rise between in the 7 to 12 kHz range to give added presence to vocalists. An integral four-position switch tailors output level and bass roll-off for a variety of uses. A sharp LF cut at the full output level reduces pops and handling noise in vocal applications, while a more gradual roll-off at the reduced output is ideal for most instrument applications.

- Versatile condenser microphone for studio and vocal applications
- Operates on phantom powering from 9 to 52v DC

	C5900	C535B	D3700 (S)	D3800	D660	D770	D880 (S)	C900
Transducer Type	Condenser	Condenser	Dynamic	Dynamic	Dynamic	Dynamic	Dynamic	Condenser
Frequency Response	20–22kHz	20–20kHz	40–20kHz	40–21kHz	70–20kHz	60–20kHz	60–20kHz	20–20kHz
Polar Pattern								
Impedance	200 Ω	200 Ω	600 Ω	600 Ω	500 Ω	600 Ω	600 Ω	200 Ω
Recommended Load Ω	> 2000 Ω	—	> 2000 Ω	> 2000 Ω	> 1200 Ω	> 2000 Ω	> 2000 Ω	> 2000 Ω
Output Connector	XLR-M	XLR	XLR-M	XLR-M	XLR-M	XLR-M	XLR-M	XLR-M
Sensitivity at 1000 Hz (re 1V/Pa)	6 mV/Pa (-45 dBV)	7 mV/Pa (-43 dBV)	2.5 mV/Pa (-52 dB)	2.8 mV/Pa (-51 dBV)	2.0 mV/Pa (-54 dBV)	1.8 mV/Pa (-75 dBV)	2.5 mV/Pa (-52 dBV)	6 mV/Pa (-44 dBV)
Max. SPL for 1% THD	140 dB	137 dB	147 dB	147 dB	140 dB	147 dB	147 dB	139 dB
Size (Max. dia./length)	2.1 x 7.1″	1.8 x 7.2″	1.8 x 7.3″	2.1 x 7.4″	1.97 x 7.13″	1.97 x 7.13″	1.97 x 7.13″	2 x 7.1″

D3700/D3700S/D3800

Tri-Power Dynamic Vocal Microphones

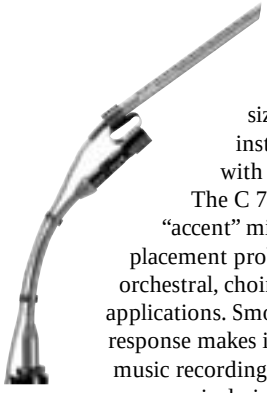
Rugged construction, distinctive Tri-Power styling, and outstanding sonic performance for on-stage instrumental and vocal pickup. The D3700 and D 3800 offer a combination of unsurpassed performance and exclusive features unmatched by any other dynamic vocal mics. At the heart of both mics is AKG's Maximum Flux neodymium/iron-boron rare earth transducer. This high output element, coupled with a uniformly tight hypercardioid pickup pattern, provides exceptional gain-before-feedback right on top of high level stage monitors.



- Vocal contoured response rises gently starting at 2 kHz to pull vocals in the front of the mix, while a smooth low end boost adds fullness and body.
- Extremely high SPL levels are handled easily without harshness or boominess.
- Smoothly triangulated PowerGrip body balances perfectly in the hand, and rugged die-cast body with Iso-Damp internal suspension guarantees reliability under the most severe touring conditions.
- Built-in windscreen/pop filter removes easily for cleaning
- They are made of die cast zinc alloy and have a dark matte finish for minimum reflection and glare.
- The D3700S is identical to the D3700 except for the addition of a noiseless, recessed on/off switch. The D3800 offers even more sensitivity than the D3700.

SPECIALTY MICROPHONES

C 747 Podium Condenser Mic



Practically invisible, the pencil-sized C 747 picks up voices and instruments at surprising distances with phenomenal fullness and clarity. The C 747 is a premium quality "spot" or "accent" mic designed to solve difficult placement problems in a variety of exacting orchestral, choir, podium, and boardroom applications. Smooth, extended frequency response makes it suitable for high quality live music recording, while it's hypercardioid response is designed to be "forgiving", making it an excellent choice for critical podium/pulpit or boardroom applications where speakers tend to wander off-axis.

A tuned interference tube provides extended bass response, high sensitivity, and excellent S/N ratio. A LF roll-off switch filters out low-frequency rumble. Mounting accessories include shock mount/adapter, windscreen, mini gooseneck, swivel stand adapter, mounting clamp, and thread link. Operates on 9-52v phantom power.

HSC200SR Broadcast Headset



The HSC200SR is a combination of semi-open stereo headphones and an electret microphone that is both mechanically and electrically adjustable. The mic elements rests in a shock mount that suppresses both cable noise and acoustic headphone leakage. Soldered jumpers in the phantom powering adapter allow you to optimally adjust the mic's frequency response to individual requirements. Operates on 9-52v DC phantom power. Mic sensitivity is 7 mV/Pa at 1 kHz. Has a XLR mic connector and headphone stereo jack.

HSD200SR Communications Headset

The HSD200SR is a combination stereo headset and close-talking dynamic microphone intended for hands-free operation in a communications environment. Response is tailored with a gentle rolloff below 200Hz and a broad peak centered at 3 kHz to increase speech intelligibility. The semi-open, supra-aural headphone provides a comfortable fit around the ears, while the semi-open design enables the user to directly hear sounds and speech originating near by.

MICROMIC III SERIES

Plug in the Giants

The MicroMic III Series gives you a wide choice of dynamic or condenser miniature microphones as well as accessories.

D 409 Wind Instruments, Drums and Percussions MicroMic

A dynamic hypercardioid, cost-efficient alternative to the C419 condenser version. Its smooth frequency response ensures uncolored audio. A neutral sound with a little proximity effect and a slight presence rise between 3 and 7 kHz will make your instrument cut through the loudest mix on stage. Clips onto the bell of the trumpet, saxophone, trombone, tuba, etc.

D 409 with standard XLR connector

D409 L with mini XLR connector

C 411 Acoustic Pickup MicroMic

Suitable for all types of string instruments (acoustic guitar, banjo, mandolin, violin, etc.), C 411 comes with a reusable adhesive compound which can be removed without marring the finish. Placing it on or near the bridge of the instrument will reproduce a clear and uncolored sound without changing the balance of the instrument.

C 411 PP for phantom powering, with preamp/XLR connector

C 411 L with 1/8″ gold-plated lockable mini jack

C 416 Instrument MicroMic

The C 416 PP can be used for all types of low profile miking, including accordion, piano, guitar cabinets, auto-harp, hammered dulcimer and more. It incorporates the flat response, hypercardioid capsule of the C 419 in a format with a longer gooseneck, a mounting bracket and a full complement of mounting accessories.

C 416 PP for phantom powering, preamp/XLR connector

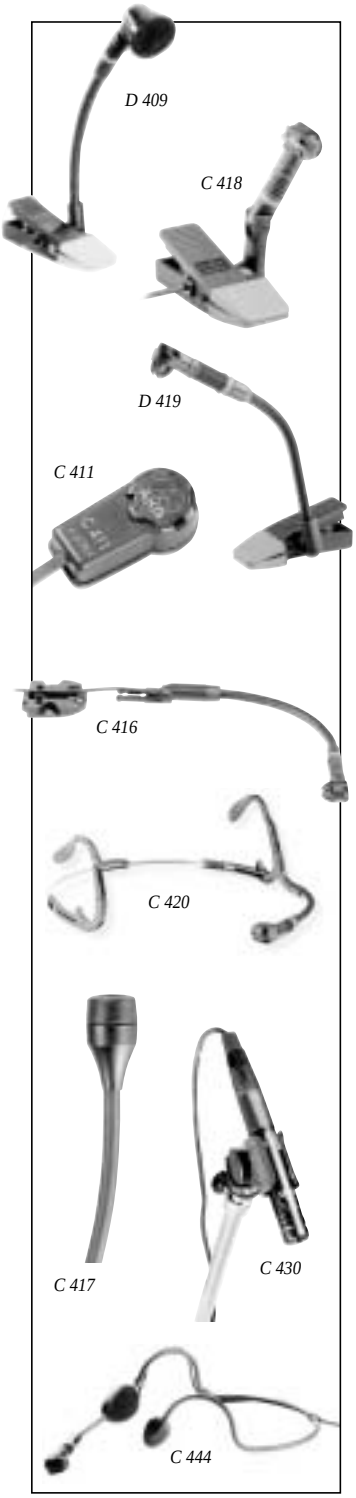
C 416 L with 1/8″ gold-plated lockable mini jack

C 417 Lapel MicroMic

One of the smallest lavalier microphones available, the C417 is an extremely smooth omnidirectional mic designed for speakers who demand the highest audio quality and complete freedom of movement. Available in both black and beige versions for ultimate concealment.

C 417 PP for phantom powering, with preamp/XLR connector

C 417 L with 1/8″ gold-plated lockable mini jack



C 418 Drum and Percussion MicroMic

The frequency response of the C 418 PP has been specifically tailored for drum and percussion miking. Compensating for the bass boost which is inherent when putting the microphone so close to the drum, the microphone reproduces a punchy sound that suppresses rumble and unwanted upper head ringing.

C 418 PP for phantom powering, with preamp/XLR connector

C 418 L with 1/8″ gold-plated lockable mini jack

C 419 Wind Instruments, Drums and Percussions MicroMic

Clips onto the bell of a trumpet, saxophone, trombone, tuba, etc. giving freedom to horn players. Its tight hypercardioid pattern makes it problem-free for stage work near floor monitors. The natural sound of the horn is retained via the flat response with a gentle high frequency peak of the transducer element, allowing the horn player to reach the audience with power and clarity.

C 419 PP for phantom powering, with preamp/XLR connector

C 419 L with 1/8″ gold-plated lockable mini jack

C 420 Headset MicroMic for Vocals and Flute

The C 420III is a condenser design with a full bandwidth 20 - 20 kHz frequency response. It captures the voice with a detailed, brilliant sound quality. The mic provides the performer with a pure, powerful sound that carries the punch and presence to cut through to the audience. Ideal for singing keyboard players, drummers, guitarists, dancers and anyone who needs high quality vocals in a hands-free application.

C 420 PP for phantom powering, with preamp/XLR connector

C 420 L with 1/8″ gold-plated lockable mini jack

C 430 Cymbal and Overhead MicroMic

Designed for cymbal and overhead miking, the frequency response of the C 430 ensures a unique crisp sound. With its extremely compact external dimensions, this microphone will keep a low profile on the drum kit without compromising audio quality.

C 430 with W32 windscreen and SA 40 stand adapter

C 444 Headset MicroMic

The C 444 L is extremely easy to use, rests securely and comfortably on the head, and is perspiration-proof. A dedicated moisture shield prevents perspiration and water from penetrating into the transducer capsule.

C 444 L with mini XLR connector and W444 windscreen

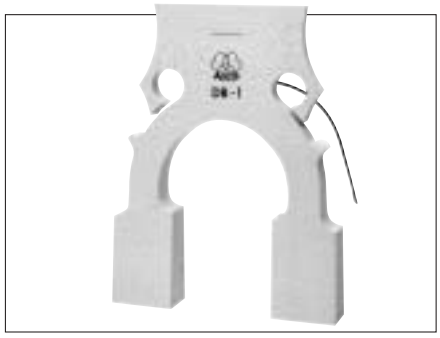
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MICROMIC III SERIES

DB 1
Double Bass Bridge Pickup



The piezoelectric transducer element is incorporated into a conventional, wooden double bass bridge. It therefore permits a non compromising, natural audio quality. Providing seamless performance and quality in acoustical bass sound reproduction, the DB-1 is designed to fit every conventional standard double bass. The pickup is an easy-to-install piezoelectric transducer element integrated into a conventional wooden double bass bridge. The DB 1 installs on the double bass like a standard bridge is mounted on the instrument.
DB-1 with a preamp/XLR connector

MICROMIC ACCESSORIES

B 15 MicroMic In-line Battery Supply



The B 15 is an in-line battery supply for all C 400 PP versions of MicroMic III Series microphones that also connects directly to the C 430 overhead MicroMic. An integrated on/off switch helps prolong battery life. The B 15 uses a single AAA size 1.5-V battery (average battery life is 30 hours). The B 15 can also be used for all commercial condenser microphones operating off 12v phantom power.



B 29 L Power Supply and Mini Mixer

The B 29 L powers one or two C 400 MicroMic III Series microphones. It features an on/off switch and status LED as well as a mic/line selector for matching the output level to the input of the connected mixer or recorder, etc.
The B 29 L operates on a 9v battery, with an average battery life of 50 hours. Two volume controls allow you to balance the levels of the two microphones. A supplied clip makes it easy to attach the B 29 L to a belt or guitar strap, etc.

MPA III L MicroMic
Phantom Power Adapter



Adapter cable with 1/8″ lockable jack socket and male 3-pin XLR connector. The MPA III L reduces the usual phantom powering voltage (9 to 52v) supplied by mixing consoles to the operating voltage (9v maximum) of AKG MicroMics.

	D409	C411	C416	C417	C418	C419	C420	C430	C444	DB1
Microphone Element Type	Dynamic	Condenser	???	???						
Frequency Response (Hz)	100-17k	10-18k	20-20k	20-20k	50-20k	20-20k	20-20k	20-20k	20-20k	5-5k
Polar Pattern	◌	vibration	◌	○	◌	◌	◡	◡	◡	vibration
Maximum SPL at 1kHz	124 dB	96 dB	121 dB	118 dB	131 dB	126 dB	126 dB	126 dB	126 dB	130 dB
Impedance	600Ω	200Ω	200Ω	200Ω	200Ω	200Ω	200Ω	200Ω	200Ω	200Ω
Sensitivity at 1 kHz	1 mV/Pa	1mV/Pa	7mV/Pa	10mV/Pa	4mV/Pa	7mV/Pa	7mV/Pa	4mV/Pa	4mV/Pa	2..5mV/Pa

INSTRUMENT MICROPHONES

D440



A cardioid dynamic mic, designed primarily as an instrument microphone for rough onstage use on percussion instruments, guitar amps, wind instruments etc. The wide frequency response of the D 440 that slightly favors the mid frequency and treble regions ensures good intelligibility.

D550



A cardioid dynamic microphone, designed primarily as a bass instrument microphone for rough onstage use on bass drum, tuba, trombone, bass amps, etc. The wide frequency response of the D 550 that slightly favors the low frequency and treble ranges ensures a very powerful sound and good intelligibility.

D112



Developed for clean kick drum and bass guitar performance with a powerful, punchy sound. The D112 is absolutely free of distortion even at high sound pressure levels, and offers very low diaphragm resonance, a relatively narrow-band rising high frequency response at 4 kHz and an extremely robust construction.

AT4040



Cardioid Condenser Microphone

A versatile large-diaphragm microphone, the AT4040 features a true condenser design (externally polarized) and utilizes A-T's latest transducer technology with state-of-the-art surface-mount electronics and transformerless circuitry. It also features a new large-diaphragm element designed specifically to provide smooth, natural sonic characteristics with a wide dynamic range and the ability to handle extremely high SPLs. The AT4040 has a cardioid polar pattern and utilizes an aged, vapor-deposited-gold large-diaphragm condenser element. A precision-machined, nickel-plated brass element baffle provides enhanced element stability and optimal sensitivity. An open acoustical environment of the symmetrical housing assembly minimizes unwanted internal reflections. In addition, the AT4040's 2-micron-thick diaphragm provides accurate reproduction of even the subtlest sounds without sacrificing high-frequency performance.

- Has an extended flat frequency response (20-20,000 Hz), wide dynamic range (133 dB, 1 kHz at max SPL), and can handle high SPLs (up to 155 dB with the 10 dB pad on).
- Distinctive transformerless circuitry virtually eliminates low-frequency distortion and enables it to deliver high-speed transients.
- Operates on 48V phantom power and has a switchable low-frequency roll-off (at 80 Hz, 12 dB/octave).
- Includes the heavy-duty AT8449 shock mount to provide mechanical isolation and secure mounting. Also includes a dust cover and protective microphone carrying case.
- Like all 40 Series microphones, each AT4040 is individually tested to ensure it maintains optimum performance characteristics and consistency from model to model.

AT4033/CL

In 1992, Audio-Technica rocked the audio world with the introduction of the legendary AT4033 Cardioid Condenser Microphone. Now, a decade later, A-T pays tribute to this classic transducer technology with the AT4033/CL. Embraced by professionals enamored with its strong up-front presence and low-noise circuitry, the AT4033/CL offers exceptional transient response and clean output signal even under the most adverse conditions. A specially contoured, vapor-deposited gold diaphragm provides accurate reproduction of even the subtlest sounds, especially high-frequency signals not easily reproduced by conventional large-diaphragm mics. The symmetrical housing assembly surrounds the mic capsule, creating an open acoustical environment ideally suited for studio work where natural reproduction of off-axis sound is paramount.





AT4047/SV

Cardioid Condenser Microphone

A large-diaphragm, cardioid condenser microphone, the AT4047/SV incorporates a transformer-coupled output and specially-tuned element to provide sonic characteristics reminiscent of early F.E.T. studio microphone designs, while utilizing modern manufacturing techniques to ensure consistency and reliability. The AT4047/SV features a dual-diaphragm configuration that allows undistorted reproduction of high sound pressure level signals at all frequencies. The 2-micron-thick, vapor-deposited gold diaphragms provide accurate reproduction of even the most subtle sounds, especially high-frequency signals.

- Has a wide dynamic range of 140 dB without the built-in attenuator, and it accepts up to 159 dB SPL with the 10dB pad on.
- Coupled with exceptionally low self-noise (9dB), the AT4047/SV excels under the stringent demands of digital recording systems.
- Floating construction mounting of the element provides isolation from noise and vibration, while an integral 80 Hz hi-pass filter provides easy switching from a flat frequency response to a low-end roll-off.

AT4050/CM5

Multi-Pattern Condenser Microphone



The AT4050/CM5 is a transformerless, multiple-pattern studio microphone designed for use in the most demanding applications. The condenser element utilizes dual, gold-plated and aged, large diaphragms for extended frequency response. The mic's polar pattern can be switched for cardioid, omnidirectional or figure-of- eight operation. This, coupled with low-noise transformerless electronics, makes this microphone ideal for the most critical digital recordings. The dynamic range of the AT4050/CM5 is 132 dB without the built-in attenuator and it accepts up to 149 dB SPL without capsule or electronic-system distortion above 1% T.H.D.

- The AT4050/CM5's low-noise circuitry is symmetrical and transformerless, resulting in exceptional transient response and clean output signals, even under high-output conditions.
- To ensure optimum performance, floating-construction mounting of the element provides isolation from noise and vibration.
- Handles very high sound pressure levels with ease. A switchable 10 dB pad is built-in, increasing the SPL capabilities to 159 dB.
- An integral 80 Hz hi-pass filter provides easy switching from a flat frequency response to a low-end roll-off. The hi-pass switch is recessed to prevent accidental activation.

	AT4033/CL	AT4040	AT4047/SV	AT4050/CM5	AT4060
Element	Condenser	Condenser	Condenser	Condenser	Condenser
Polar Pattern	◡	◡	◡	◡, ○, 8	◡
Frequency Response	30 - 20k Hz	20 - 20k Hz	20 - 18k Hz	20 - 18k Hz	20 - 20k Hz
Open Circuit Sensitivity[1]	-32 dB (25.1 mV)	-32 dB (25.1 mV)	-35 dB (17.7 mV)	-36 dB (15.8 mV)	-34 dB (19.9 mV)
Impedance	100Ω	100Ω	250Ω	100Ω	200Ω
Maximum Input Sound Level[2]	- 145 dB SPL	145 dB SPL	149 dB SPL	149 dB SPL	150 dB SPL
Signal to Noise Ratio	77 dB, 1 kHz at 1 Pa	82 dB, 1 kHz at 1 Pa	85 dB, 1 kHz at 1 Pa	77 dB, 1 kHz at 1 Pa	75 dB, 1 kHz at 1 Pa
Dynamic Range[3]	128 dB	133 dB	140 dB	132 dB	131 dB
Phantom Power Requirements	48V (±4V)	48V	48V (±4V)	48V (±4V)	—
Weight	13.4 oz	12.7 oz	14.5 oz	18.0 oz	22.6 oz

[1] *±2 dB re 1V at 1 Pa* [2] *1 kHz at 1% T.H.D.* [3] *1 kHz at Max. SPL*

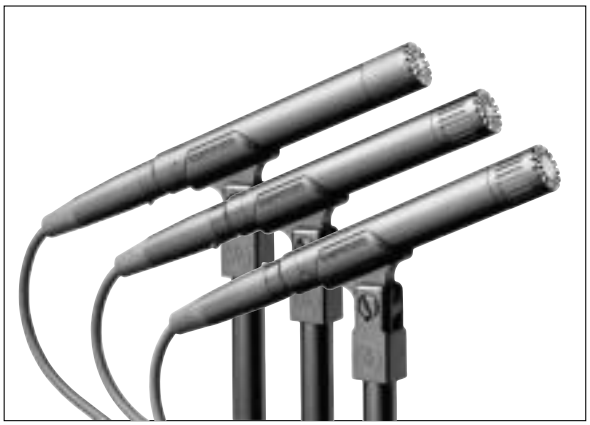


AT4060

Cardioid Vacuum Tube Condenser Studio Microphone

The AT4060 is a cardioid vacuum tube condenser microphone designed for use in the most demanding studio applications. It utilizes dual, gold-plated and aged, large-diaphragm elements for extended frequency response. The AT4060 has a wide dynamic range of 131 dB and the ability to accept up to 150 dB SPL without capsule or electronic-system distortion above 1% T.H.D. These capabilities, coupled with the benefits of exceptionally low self-noise and very high output, combine in a versatile microphone that meets the diverse requirements of contemporary recording situations, while still providing the warm sound of vintage valve design. Each hand-selected tube is individually aged and tested, then employed in a uniquely-configured circuit for enhanced tube performance. Specially-tuned elements improve dynamic range and the ability to handle high SPLs without sacrificing high-frequency characteristics. Floating-construction mounting of the elements provides isolation from noise and vibration.

AT-4049A • AT-4051A • AT-4053A



40-Series Small Diaphragm, Modular Condenser Microphones

The AT4049a, AT4051a and AT-4053a are transformerless phantom-powered condenser studio microphones engineered for professional recording and broadcast. They offer smooth, flat frequency response and a clean signal, even in high-output conditions. They also feature exceptionally low self noise, high output and fast transient response. But what separates them from other mics, is that the head capsules may be easily removed, stored and replaced with an optional omnidirectional, cardioid or hypercardioid capsule to meet the needs of the widest range of miking applications.

They combine the benefits of exceptionally low self-noise with very high output. Equally important, they handle very high sound pressure levels with ease. An integral 80 Hz hi-pass filter provides easy switching from a flat frequency response to a low-end roll-off. The hi-pass position reduces mic sensitivity to "popping" in close vocal use and to handling noise. It also attenuates unwanted low-frequency ambient noise. Use of the supplied foam windscreen helps reduce wind and "popping" noise. Their housings are constructed of turned brass, rather than thin-wall aluminum. The surface is plated in black chrome for durability and low reflectivity. They operate on 48v DC phantom power. A compartmentalized protective carrying case accommodates the microphone, stand adapter, windscreen and two optional head capsules.

- They handle high sound pressure levels with ease—up to 146 dB SPL while producing no greater than 1% T.H.D.
- Integral 80 Hz hi-pass filter provides easy switching from a flat frequency response to a low-end roll-off. The switch is recessed to prevent accidental activation
- Included foam windscreen helps reduce wind and "popping" noise
- Constructed of turned brass and is plated in black chrome for durability and low reflectivity
- Requires remote "phantom" or "simplex" power source supplying 48 volts DC
- DC biased condenser element design
- Omnidirectional polar pattern with optional capsules offering cardioid or hypercardioid polar patterns
- Output impedance of 100 ohms and output shall be transformerless balanced.
- Length of 6.10″ a diameter of 0.83″ (21.0 mm), and a weight of 4.2 oz. (120g)
- A compartmentalized protective carrying case accommodates the microphone, stand adapter, windscreen and two optional head capsules

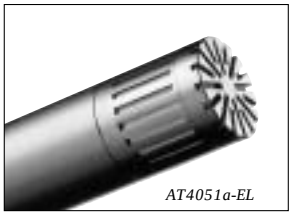
MODULAR EXPANSION CAPABILITY

AT-40 series microphones consist of two modular sub assemblies which are threaded together: the AT4900a-48 handle containing the mic's electronics, and either the AT4049a-EL (omnidirectional), AT4051a-EL (cardioid) or the AT4053a-EL (hypercardioid) condenser head capsules. The head capsules may be easily removed, stored and replaced with an optional omnidirectional, cardioid or hypercardioid capsule to provide the widest range of miking capability.



AT4049a-EL

The **AT4049a-EL** capsule provides the precision recording and audio measurement advantages of a flat, wide-response omnidirectional side pattern.



AT4051a-EL

The **AT4051a-EL** capsule provides the advantages of a smooth, wide-response cardioid pattern.



AT4053a-EL

The **AT4053a-EL** hypercardioid element offers excellent side rejection for increased working distance, and a highly uniform polar pattern that virtually eliminates off-axis coloration.

AT-4041

Small Diaphragm Cardioid Condenser Microphone

The AT4041 is a pressure-gradient condenser microphone with a uniform cardioid polar pattern that is ideal for professional recording and broadcast applications. Very similar to the AT4051a except for interchangeable capsules. The AT4041 features a transformerless output for low self-noise, high output and very high SPL handling capability (up to 145dB). The balanced output is direct-coupled, resulting in a clean signal even under high-output conditions.



Integral high-pass filter provides easy switching from a flat frequency response to a low-end roll-off. In addition to preventing the pickup of low-frequency ambient noise (such as air-conditioning) the high-pass filter may also be used to compensate for proximity effect in close-miking situations.

Use of the supplied foam windscreen further reduces wind and "popping" noise. Ruggedly built, the AT4041 is housed in a case made of turned brass, not thin-wall aluminum. The surface is painted black for low reflectivity. Compatible with any remote "phantom" or "simplex" power source supplying 48v DC.

AT-4071a/AT-4073a

Line + Gradient Condenser Microphones



The AT4071a and AT4073a are transformerless line + gradient condenser microphones designed to meet the critical long-distance pickup demands of broadcasting, film/TV sound, professional recording and theater sound reinforcement. They feature a broad-band, linear on-axis frequency response, with maximum rejection of sounds from their sides and the rear. Rear polar lobing, common in shotgun microphones is greatly reduced and their off-axis response remains highly uniform. The resulting lack of sound coloration on- and off-axis makes them particularly suited for miking dynamic action in film/TV audio as well as in "spot" miking techniques in the music studio or theater.

- The effective working distance of a directional mic is determined in great part by its polar pattern and the uniformity of its off-axis response. However, the S/N ratio also affects the usable miking distance. The distance may be less if the system S/N ratio is low because of low mic sensitivity or noisy electronics. The AT4071a and AT4073a provide extremely high output and a noise floor that is hardly measurable, much less audible. Use them with confidence in distant miking applications and even under the stringent demands of modern digital recording systems.
- They set new standards in small size and light weight. Overall length is 9.13″ (AT4073a) and 15.55″ (AT4071a) and at 4 and 5.5 oz. respectively, they add practically no noticeable weight to the end of a fish pole or the top of a minicam.
- Through the use of an advanced design, both the diaphragm and the side ports are exposed to the same acoustic environment. One result of this is that they are less sensitive to noise caused by wind turbulence or the "encounter" noise of panning action. And there is a marked reduction in proximity effect. Recordings made at varying distances remain more consistent in response, making both production and editing quicker, easier and less costly.
- They are exceptionally resistant to mechanical or handling noise. An integral 150 Hz hi-pass filter is available to "roll-off" the low-frequency response, thereby attenuating unwanted low-frequency ambient noise (such as air conditioning). The switch is recessed to prevent accidental activation. Built to withstand rigorous use in the field, they are powered with any remote "phantom" or "simplex" power source supplying from 11 to 52v DC.

AT-3031/3032/3035/3060

Cardioid and Omnidirectional Condenser and Tube Microphones



Audio-Technica's 30 Series offer all the advantages of high-quality microphones at an affordable price. They feature rugged construction and high performance specifications as well as AT's legendary engineering and manufacturing technology. This provides an exceptional level of consistency from model to model, critical when any of the 30 Series are purchased in pairs. The 30 Series consists of four models: the AT3031 cardioid and AT3032 omnidirectional small-diaphragm condenser mics, the AT3035 large-diaphragm side-address cardioid condenser mic and the large-diaphragm AT3060 cardioid tube mic.

AT3031 and AT3032

The low-profile AT3031 and AT3032 condenser mics offer outstanding performance in critical recording applications. They have a frequency response of 30Hz to 20kHz; high SPL capability of 148 dB (158 dB with 10 dB pad); 48v phantom power; and a switchable roll-off (at 80 Hz, 12 dB/octave). The AT3031's cardioid pattern reduces pickup of sounds from the sides and rear, improving isolation of desired sound source. The AT3032's omnidirectional pattern provides maximum ambient pickup.

AT3060

To offer that warm classic tube sound, the AT3060 uses hand-selected tubes that are individually tested and aged to maintain peak performance. It requires no dedicated power supply or special cable—operates exclusively on 48v phantom power. Shock-mounted tube assembly dampens mechanically-coupled vibrations, while the precision nickel-plated brass, acoustic element baffle provides enhanced element stability and optimal sensitivity.

AT3035

The AT3035 is a precision-crafted, large-diaphragm condenser microphone that represents a value while offering all the performance of a high-quality studio microphone. Designed to provide versatility in the most demanding recording applications, the AT3035 has a fixed cardioid polar pattern and features excellent performance specifications, including:

- 20Hz-20kHz flat extended frequency response ; SPL capability of 148 dB (158 dB with the 10dB pad); and an extremely low self-noise (12dB).
- Every aspect of the AT3035 has been tailored for a very smooth, natural sound while ensuring maximum accuracy, making it a perfect complement for today's digital recording equipment.
- To ensure optimum performance and isolation from noise and vibration, the AT3035 includes a professional shock mount. It uses 11-52v phantom power
- Switchable low-frequency roll-off (at 80 Hz, 12 dB/octave) reduces sensitivity to pops and the pickup of low-frequency ambient noise and mechanically-coupled vibrations.

	AT4049a	AT4051a	AT4053a	AT-4041	AT-4071a	4073a	AT-3031 / AT-3032	AT-3035
Element	Condenser	Condenser	Condenser	Condenser	Condenser	Condenser	Condenser	Condenser
Polar Pattern	◯	ᔓ	ᔕ	ᔓ	Line + Gradient	Line + Gradient	AT-3031: ᔓ AT-3035: ◯	ᔓ
Freq. Response	20 - 20k Hz	20 - 20k Hz	20 - 20k Hz	20 - 20k Hz	30 - 20k Hz	30 - 20k Hz	30 - 20k Hz	20 - 20k Hz
Open Circuit Sensitivity[1]	-33dB (22.4 mV), ±2 dB	-34dB (19.9 mV), ±2 dB	-33dB (22.4 mV), ±2 dB	-36dB (15.8 mV), ±2 dB	-21 dB (89.1 mV) ±1 dB	-23 dB (70.8 mV ±1 dB	-34 dB (19.9 mV)	-32 dB (25.1 mV)
Impedance	100 Ω	100 Ω	100 Ω	100 Ω	100 Ω	100 Ω	250 Ω	250 Ω
Maximum SPL[2]	146 dB SPL	146 dB SPL	146 dB SPL	145 dB SPL	124 dB SPL	126 dB SPL	148 dB SPL	148 dB SPL
S/N Ratio[3]	75 dB	75 dB	75 dB	70 dB	82 dB	80 dB	78 dB	82 dB
Dynamic Range[4]	127 dB	127 dB	127 dB	121 dB	112 dB	112 dB	132 dB	136 dB
Phantom Power	48V	48V	48V	48V	11 - 52V	11 - 52V	11 - 52V	11 - 52V
Pad	–	–	–	–			10 dB pad	10 dB pad
Low Freq. Roll-off	12dB/oct @ 80Hz	12dB/oct @ 80Hz	12dB/oct @ 80Hz	12dB/oct @ 80Hz	12dB/oct @ 150Hz	12dB/oct @ 150Hz	12dB/oct @ 80Hz	12dB/oct @ 80Hz
Weight	4.2 oz	4.2 oz	4.2 oz	4.2 oz	5.5 oz	4.0 oz	AT-3031: 4.1 oz (116g)	13.8 oz
Price	**499.95**	**479.95**	**509.95**	**296.50**	**629.95**	**529.95**	**168.95 / 168.95**	**199.95**

ARTIST ELITE SERIES

Live Performances Mics

Recent advances in the quality and sophistication of professional live-sound systems have been nothing short of revolutionary. Tours, clubs, broadcast events, corporate facilities and worship venues sound better than ever, utilizing better system design and better components in the audio chain. So Audio-Technica partnered with top touring companies, FOH and monitor engineers, audio consultants and artists – to learn what it takes to make the best-sounding, most reliable and consistent microphones for the live-sound industry. They listened carefully. Then they applied this knowledge to the creation of a high performance line —the Artist Elite Series.

AE 5400 Cardioid Condenser Mic

True condenser, large-diaphragm element design delivers pristine sound for the most discriminating vocalist. The AE5400 utilizes an aged, vapor-deposited gold diaphragm for a flat frequency response with improved dynamic range and easily handles high SPLs. Featuring a robust design to combat the rigors of the road, the mic also has a switchable 80Hz high-pass filter and a 10dB pad. It also provides excellent natural vocals with off-axis response that is linear and uniform across the entire frequency range. To ensure optimum performance, the AE5400 features anti-shock engineering for superior isolation from noise and vibration, a multi-level windscreen design, and includes AT's AT8470 Quiet-Flex stand clamp.



AE 3300 Cardioid Condenser Mic

Offers exceptional performance for venues ranging from the largest arenas to the most intimate club settings. Featuring a well-tempered cardioid polar pattern with outstanding rejection qualities, the AE3300 provides exacting detail, brilliant clarity and uncolored realism. Has a switchable 80Hz high-pass filter and 10dB pad, superior anti-shock engineering, multi-level windscreen and includes the AT8470 stand clamp.

AE 6100/AE 4100 Hypercardioid and Cardioid Dynamic Mics

The AE6100 delivers clean articulation for great on-stage presence and excellent monitor cut. The AE4100 provides focused vocal pickup while offering superior gain-before-feedback. AT tailored the performance of the AE4100 for a more aggressive sound quality that keeps the vocals up-front in the mix. Both mics have superior anti-shock engineering for low handling noise and include the AT8470 Quiet-Flex stand clamp.

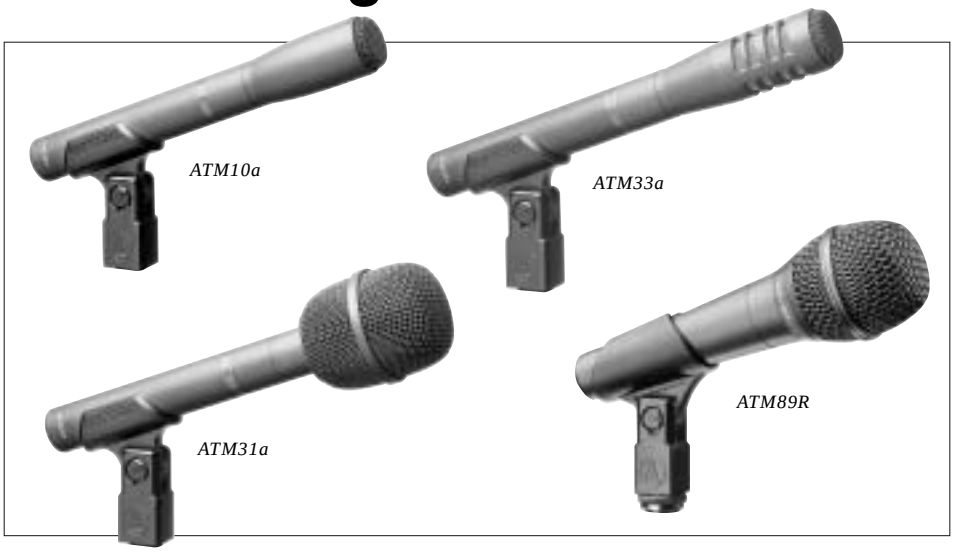


artist elite series • instrument mics

The Artist Elite Series instrument mics combine robustness, versatility and consistent performance night after night in the most demanding areas of live sound instrument miking. They have an integral 80 Hz high-pass filter switch, 10 dB pad and include the new AT8471 isolation clamp. The AE5100 is a large-diaphragm end-fire cardioid condenser; the AE3000 is a large-diaphragm side-address cardioid condenser; and the revolutionary AE2500, a combination condenser/dynamic model, is the ultimate mic for kick drums.

- The **AE5100** is a live-sound instrument mic designed for drum overheads and percussion, acoustic guitar, strings and other acoustic instruments. The large diaphragm delivers a natural sound with accurate response, while the end-fire design allows it to be placed where other large diaphragm mics are unable to go.
- The **AE3000** is designed to take the highest sound pressure levels encountered on stage from sources such as guitar cabinets, toms, snare, tympani and overheads while still delivering uncolored performance, thanks to an open-architecture headcase and large-diaphragm condenser capsule. The mic features a small-format side-address design with tapered grille for optimum placement versatility.
- The **AE2500** features a revolutionary dual-element design that combines condenser and dynamic capsules within a single housing and positioned in a perfect phase relationship – nearly impossible to accomplish with separate microphones. Ideal for kick/bass drum applications, the mic will survive extreme SPLs while providing uncompromising audio quality. The AE2500's dynamic element cleanly captures the attack transients of the beater, while the condenser element renders the internal tonalities of the shell and the instrument as a whole.

ARTIST SERIES – Studio/Stage Condenser Mics



Great stage and studio sound starts with the microphone. Today, more musicians and sound engineers find that their best sound starts with Artist Series microphones. The wide range of condenser mics is ideal for both studio and sound reinforcement applications. Unique, specialized models solve the most difficult pickup problems with innovative engineering and ultra-precise manufacturing. In addition, a select group of rugged Artist Series dynamic microphones are the first choice of many touring musicians as well as major recording studios. Carefully tailored response assures effective sound reinforcement. Every Artist Series microphone has withstood the close scrutiny of industry professionals in extensive field testing. Each also benefits from some of the most rigorous production testing available anywhere in the world.

ATM10a

Omnidirectional condenser microphone for vocal/instrumental use when maximum ambient pickup is desired. Excellent for hi-hat, strings, guitar and piano. Extremely smooth extended response on and off axis. Low sensitivity to popping and overload. Internal AA battery or phantom power operation **136.50**

ATM33a

Cardioid condenser microphone for more critical studio/stage use. Has outstanding linearity both on and off axis, high SPL capability, wide frequency response and low-mass diaphragm. Provides balanced sound for strings and solo vocals, plus crisp reproduction of pianos, reeds, brass and drums. Internal AA battery or phantom power operation.....**188.50**

ATM31a

Cardioid condenser microphone for general studio/stage use. Has wide frequency response, and added warmth when used close up. Large protective screen. Excellent for close-up vocals, piano, reeds and brass. Low-mass diaphragm ensures minimum distortion. Sound is similar to far more expensive studio microphones. Operates on internal AA battery or 9-52v phantom power **162.50**

ATM89R

Hypercardioid condenser microphone delivers smooth, accurate vocal reproduction for studio-quality sound on stage. Special internal shock mounting eliminates virtually all handling and cable noise. Accepts interchangeable elements for a variety of polar patterns available in cardioid, subcardioid and omni. Features high sensitivity and SPL capability with a rugged design for reliable performance. Operates on 48v phantom power.....**206.50**

	ATM10a	ATM31a	ATM33a	ATM89R
Element	Condenser	Condenser	Condenser	Condenser
Polar Pattern	◯	◡	◡	◌
Frequency Response	20 - 18k Hz	30 - 20k Hz	30 - 20k Hz	70 - 20k Hz
Open Circuit Sensitivity	-44 dB (6.3 mV)	-44 dB (6.3 mV)	-44 dB (6.3 mV)	-42 dB (7.9 mV)
Impedance (Phantom/Battery)	200 Ω / 270 Ω	200 Ω / 270 Ω	200 Ω / 270 Ω	100 Ω
Maximum SPL Input[1]	137 dB	137 dB	137 dB	138 dB
Signal to Noise Ratio[2]	70 dB	70 dB	70 dB	67 dB
Dynamic Range (Typical)[3]	113 dB	113 dB	113 dB	111 dB
Phantom Power	9 - 52V DC	9 - 52V DC	9 - 52V DC	48V DC
Battery Type	1.5V AA	1.5V AA	1.5V AA	—
Battery Current/Life (Typical)	0.4 mA/1200 hrs.	0.4 mA/1200 hrs.	0.4 mA/1200 hrs.	—
Weight	7.1 oz (200 g)	7.1 oz (200 g)	5.3 oz (150g)	10.2 oz (290 g)

1. re 1V at 1 Pa 2. 1kHz @ 1Pa 3. 1 kHz @ Max SPL

ARTIST SERIES

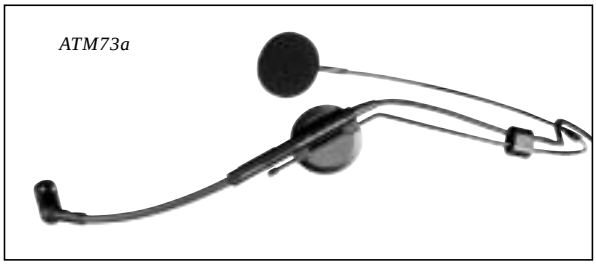
SPECIAL PURPOSE CONDENSER MICROPHONES

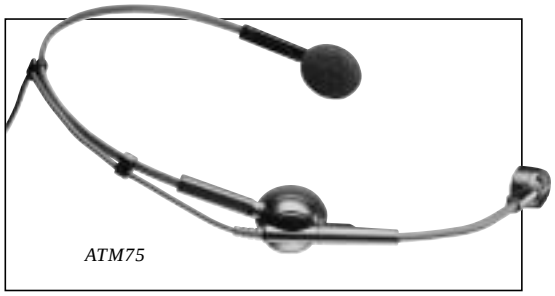
ATM73a

Headworn cardioid condenser microphone ideal for drummers, keyboardists or anyone who needs to keep their hands free. Unobtrusive headband. Flexible mic boom. Belt-mounted power. Includes battery and windscreen.

ATM79ac: Same as ATM73a less battery/remote power module. Unterminated for wireless and custom applications.....**181.95**

ATM73cW: Same as ATM73a less battery/remote power module. Terminated for use with Audio-Technica professional wireless systems.....**109.95**





ATM75

Headworn cardioid condenser microphone provides superior vocal performance with hands-free operation. Pivot-mounted flexible mic boom descends from left or right side of comfortable, secure headband. Moisture-resistant protective coating on headband and boom. Belt-mounted power module. Includes battery and windscreens.

ATM75c: Same as ATM75 less battery/remote power module. Unterminated for wireless and custom applications.....**142.95**

ATM75cW: Same as ATM75 less battery/remote power module. Terminated for use with Audio-Technica professional wireless systems.....**76.95**

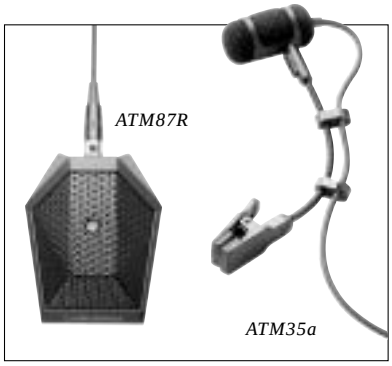
	ATM35	ATM87R	ATM73a	ATM75
Element	Condenser	Condenser	Condenser	Condenser
Polar Pattern	◠	Unidirectional	◠	◠
Frequency Response	20 - 20k Hz	30 - 20k Hz	60 - 15k Hz	60 - 15k Hz
Open Circuit Sensitivity	-51 dB (2.8 mV)	-36 dB (15.8 mV)	-55 dB (1.7 mV)	-51 dB (2.8 mV)
Impedance	200 Ω	100 Ω	200 Ω	200 Ω
Maximum SPL Input[1]	145 dB	151 dB	135 dB	132 dB
Signal to Noise Ratio[2]	67 dB	66 dB	60 dB	58 dB
Dynamic Range (Typical)[3]	113 dB	123 dB	101 dB	96 dB
Phantom Power	11 - 52V DC	48V DC	5 - 52V DC	9 - 52V DC
Battery Type	9v	—	1.5V AA	1.5V AA
Battery Current/Life (Typical)	1 mA/1000 hrs.	—	1 mA/1000 hrs.	0.4 mA/2000 hours typical
Weight	0.3 oz (8.5 g)	4.8 oz (135 g)	1.1 oz (31 g)	2.1 oz (60 g)

1. re 1V at 1 Pa 2. 1kHz @ 1Pa 3. 1 kHz @ Max SPL

HEADWORN CONDENSER MICROPHONES

ATM35

High-intensity cardioid condenser microphone includes unique UniMount that attaches directly to brass or reed instruments. Ideal for active active stage performances. UniMount foam windscreen holds microphone for accurate positioning while protecting element. Also clips to piano, string bass, or to drum kit hardware for clean reproduction of toms and snares. Crisp, clear, well-balanced response, even at high SPLs.....**222.50**



ATM87R

A high SPL cardioid condenser boundary microphone, the ATM87R excels in picking up highly dynamic instruments such as kick drum, piano, etc. Low-profile design is ideal for applications requiring minimum visibility. Heavy die-cast case and rubber non-slip bottom pad minimize mechanical coupling of surface vibrations to the microphone, while low-roll-off switch helps control undesired ambient noise. Operates on 48v DC phantom power. A 25-ft. cable with TA3F and XLRM-type connectors is included.....**194.50**

ARTIST SERIES - Hi-Energy Studio/Stage Dynamic Mics

These dynamic handheld and instrument mics feature Hi-Energy neodymium magnets and low-mass, copper-clad aluminum wire voicecoils to offer high SPL output capabilities with exceptional output sensitivity.

ATM23HE
Hypercardioid instrument microphone combines excellent performance characteristics and rugged dependability for superior results on snare drum, as well as other highly dynamic instruments such as floor and rack toms, piano and electric guitar cabinets. Low profile design permits versatile placement around drum kit hardware **152.95**

ATM27HE
This handheld hypercardioid microphone offers superb value and outstanding performance for vocal applications. The double-dome diaphragm, low-mass voice coil and close-tolerance neodymium magnet structure provide crisp high-frequency performance. The element is double-isolated to greatly reduce handling noise.....**89.95**

ATM29HE
Designed for instrumental pickup and general stage use, this hypercardioid microphone is a versatile live-sound producer. Low-mass voice coil and neodymium magnet structure ensure excellent transient response. Double-dome diaphragm captures acoustic and electronic instrument sound with accuracy and ease. Double-isolation of the element assures low mechanical noise.....**84.95**



ATM41HE
This handheld hypercardioid mic, designed for close-up vocals, features excellent off-axis rejection for minimizing feedback and a tailored response that provides natural, full-range sound reproduction. Patented floating diaphragm reduces handling noise.....**129.95**

ATM61HE
A hypercardioid vocal microphone that utilizes a unique high-output element and an advanced, ultra-precise diaphragm that extends response and reduces coloration of off-axis sounds — similar to much more expensive stage condenser mic's. Patented floating diaphragm reduces handling noise. Offers excellent gain before feedback and best stage presence.....**148.95**

ATM61HE/S Same as above with an integral on/off switch and non-removable grille .**149.95**

ATM63HE
Close-working hypercardioid microphone for both vocal and instrumental pickup. Extended frequency response is ideal for pickup of toms, snares, electric guitar and piano with excellent isolation and peak-free response. Wide-range uniform polar pattern ensures natural, balanced pickup both on and off axis. Extremely rugged with all-metal grille.....**119.95**

artist series • dynamic mics



ATM25
Rugged dynamic hypercardioid microphone responds cleanly in high SPL environments. Big, warm low-frequency response with excellent presence is ideal for close-up vocals and dialog as well as kick drums, toms, acoustic bass, piano or any high-output low-frequency instrument.....**178.95**

ATM41a
Close-working, field-proven rugged dynamic cardioid microphone sets a high standard for handheld vocal use. Also useful for pickup or amplified instrument speaker output. Comfortable shape and good balance is helpful during extended sets. Excellent off-axis rejection for minimum feedback.....**69.95**

	ATM23HE	ATM27HE	ATM29HE	ATM41HE	ATM61HE	ATM63HE	ATM25	ATM41a
Element	Dynamic	Dynamic	Dynamic	Dynamic	Dynamic	Dynamic	Dynamic	Dynamic
Neodymium Magnet	✓	✓	✓	✓	✓	✓	—	—
Polar Pattern	Hypercardioid	Hypercardioid	Hypercardioid	Hypercardioid	Hypercardioid	Hypercardioid	Hypercardioid	Cardioid
Frequency Response	70 - 16k Hz	60 - 15k Hz	70 - 16k Hz	50 - 17k Hz	50 - 18k Hz	50 - 18k Hz	30 - 15k Hz	50 - 16k Hz
Open Circuit Sensitivity[1]	-53 dB (2.2 mV)	-56 dB (1.5 mV)	-56 dB (1.5 mV)	-51 dB (2.8 mV)	-51 dB (2.8 mV)	-52 dB (2.5 mV)	-54 dB (1.9 mV)	-56 dB (1.5 mV)
Impedance	600 ohms	200 ohms	200 ohms	600 ohms	600 ohms	600 ohms	600 ohms	250 ohms
Weight	10.6 oz (300 g)	9.9 oz (280 g)	10.2 oz (290 g)	10.3 oz (292 g)	9.7 oz (275 g)	9.3 oz (265 g)	13.8 oz (390 g)	11.6 oz (328 g)

1. re 1V at 1 Pa

B&H PHOTO - VIDEO - PRO AUDIO

MIDNIGHT BLUE SERIES

Hard-Rockin' Cardioid Microphones

With their cardioid pickup pattern, the MB1000, MB2000 and MB3000 (Dynamic) and MB4000 (Condenser) mics reduce unwanted background noise and control feedback, especially when used extremely close. Miking instruments or vocalists at very close range also effectively reduces "leakage" from one performer to the next. Low-frequency noise is also reduced by a low-frequency roll-off of sounds more than one foot away from the microphone. Enclosed in a rugged die-cast housing, they feature a special, baked-on "Midnight Blues" finish. In addition to being attractive and comfortable to hold, this special finish further reduces handling noise and increases durability.
A multi-stage windscreen reduces wind noise and "popping" when performers work extremely close. The supplied microphone clamp permits mounting on any mic stand with 5/8˝-27 threads.



MB1000L and MB1000H

Available in low-and high-impedance versions, the MB1000L and MB1000H are wide-range dynamic mics designed specifically for close-up vocal and instrumental use. A special internal shock mount system minimizes handling and case noise. The microphone also features the MagnaLock on/off switch for silent operation. The switch may by locked in the "on" position to prevent accidental turnoff during use. High-output Neodymium design and low-mass diaphragm and voice coil assure excellent signal-to-noise ratios with mic mixers, tape recorders, and other electronic inputs. The MB1000 is capable of undistorted output even in very intense sound fields.....**44.50**

MB2000L

Like the MB1000L, the MB2000L is designed for close-up vocal and instrumental use. Crisp high-frequency response and extended low-frequency response makes it an ideal choice for lifelike instrument pickup. The MB2000L also adds a foam-lined carrying case to protect the mic during transport and storage.....**52.50**

MB3000L

The MB3000L is specifically designed for use in high-quality recording and sound reinforcement applications to meet the needs of professional musicians and engineers. Extended frequency response, high sensitivity and extremely fast transient response make the mic ideal for high-quality vocal and instrumental pickup as well. Includes a foam-lined carrying case.....**69.50**

MB4000C

The MB4000C is a wide-range condenser microphone designed for professional musicians and sound engineers in live performance, recording and sound reinforcement applications. Smooth response fast attack and quiet operation ensure high-quality vocal and instrumental pickup. The MB4000C is powered by a 1.5V AA battery or 9-48V DC phantom power. Includes a foam-lined carrying case.....**69.95**

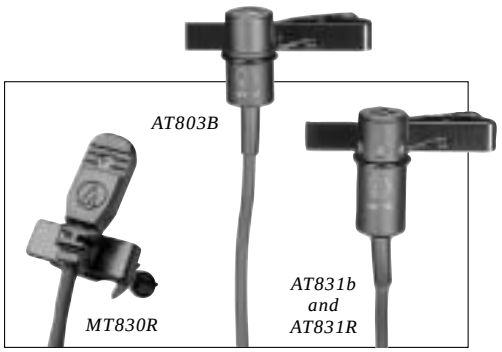
	MB1000L	MB1000H	MB2000L	MB3000L	MB4000C
Element	Neodymium Dynamic	Neodymium Dynamic	Neodymium Dynamic	Neodymium Dynamic	Condenser
Polar Pattern	◠	◠	◠	◠	◠
Frequency Response	70 - 15k Hz	70 - 15k Hz	60 - 15k Hz	50 - 16k Hz	100 - 18k Hz
Open Circuit Sensitivity	-49 dB (3.5 mV)	-38 dB (12.5 mV)	-49 dB (3.5 mV	-52 dB (2.5 mV)	-48 dB (3.9 mV)
Impedance	500 Ω	10,000 Ω	500 Ω	250 Ω	200 Ω
Maximum SPL Input[1]	—	—	—	—	128 dB
Signal to Noise Ratio[2]	—	—	—	—	66 dB
Dynamic Range (Typical)[3]	—	—	—	—	100 dB
Phantom Power (2.0mA Typical)	—	—	—	—	9 - 48V DC
Battery Type	—	—	—	—	1.5V AA
Battery Current/Life (Typical)	—	—	—	—	1 mA/1000 hrs.
Weight	9.3 oz (264 g)	9.6 oz (272 g)	8.6 oz (244 g)	12.0 oz (340 g)	4.7 oz (132 g)

1. 1 kHz at 1% T.H.D. 2. 1kHz @ 1 Pa 3. 1 kHz @ Max SPL

UNIPOINT SERIES

Miniature Condenser Microphones

Designed to be worn on the clothing of performers, Audio-Technica's miniature condenser microphones provide excellent yet unobtrusive sound pickup. Their mic element is enclosed in a rugged housing with low-reflectance finish and internal construction minimizes handling and clothing noise. The wide-range capability of these mics ensure clean, accurate reproduction with high intelligibility for lecturers, singers, stage and TV performers, as well as musical instruments.



	AT-803b	AT-831b	AT-831R	MT830R
Element	Condenser	Condenser	Condenser	Condenser
Polar Pattern	○	ᔓ	ᔓ	○
Frequency Response	30Hz - 20kHz	40Hz - 20kHz	40Hz-20kHz	30Hz - 20kHz
Open Circuit Sensitivity[1] (Phantom/Battery)	-45 dB (5.6 mV)/-46 dB (5.0 mV)	-44 dB (6.3 mV) / -45 dB (5.6 mV)	-42 dB (7.9 mV)	-34 dB (19.9 mV)
Maximum SPL Input[2] (Phantom/Battery)	135 dB / 121 dB	135 dB / 121 dB	141 dB	131 dB
Dynamic Range-Typical[4] (Phantom/Battery)	106 dB/92dB	106 dB/92dB	112 dB	107 dB
Phantom Power	9 - 52V	9 - 52V	9 - 52V	9 - 52V
Battery Type	1.5V AA	1.5V AA	—	—
Low-Frequency Roll-off	18dB/oct @ 150 Hz	18dB/oct @ 80 Hz	—	—
Price	**139.95**	**149.95**	**174.95**	**155.95**

AT803b

Omnidirectional Condenser Microphone

The AT803b provides excellent yet unobtrusive sound pickup. The wide-range capability of the AT803b ensures clean, accurate reproduction with high intelligibility for lecturers, singers, stage and TV performers.

- Powered by AA battery (up to 1200 hours) or 9-52v phantom power
- 6′ (1.8 m) cable is provided between the mic and power module
- A built-in 3-position switch on the power module allows selection of off, on/flat response, or on/low-roll-off
- Includes AT8417 clothing clip, AT8116 windscreen and carrying case

MT830R

Subminiature Omnidirectional Condenser Mic

The MT830R's wide-range capability ensures clean, accurate reproduction with high intelligibility for lecturers, singers, stage and TV performers, as well as instruments. Runs on 9-52v phantom power only.

- 25′ cable is provided between the mic and power module
- Includes AT8417 clothing clip and AT8130 windscreen

AT831b/AT831R

Cardioid Condenser Microphone

Cardioid versions of the AT803b, the miniature AT831b and AT831R (phantom power only) are designed for hands-free applications in sound reinforcement systems, and for use by musicians, especially for pickup of acoustic guitar. Cardioid pattern provides improved gain before feedback. Close-up voice and instrument pickup are crisp and clean, yet full sounding. Suppression of background noise is significantly better than that of full-size microphones. Same features and accessories as the AT803b plus they adds the AT8444 guitar adapter.

New! AT899

Subminiature Omnidirectional Condenser Mic



Designed for high-quality, unobtrusive operation, the AT899 provides maximum intelligibility and clean, accurate reproduction for vocalists, lecturers, stage and television talent, and houses of worship. A mere 5mm in diameter the AT899 is ideal for applications requiring minimum visibility. The omni element is enclosed in a rugged, low-profile housing to minimize handling and wind noise.

Supplied accessory kit includes single and double mounts, clothing clip, viper clip, magnetic clip, two open-pore foam windscreens, and two metal mesh element covers to protect the condenser element from moisture, makeup, and other contaminant's. Also includes a convenient carrying case to hold the microphone and accessories.

Available in four terminations and two colors (low-reflectance black or "theater" beige)

AT899: Wired model with TA3F connection to the supplied AT8537 battery/phantom power module. The module features battery/phantom power operation, low-frequency roll-off switch, and an XLR3M connector supplying a low-impedance balanced output.

AT899cW: 55″ cable terminated with professional 4-pin locking connector for use with AT's UniPak wireless transmitters

AT899cT5: 55″ cable terminated with 5-pin connector for ATW-U101 wireless body-pack systems

AT899c: 55″ cable unterminated for custom applications

UNIPOINT SERIES

Boundary Condenser Microphones

Audio-Technica's wide range boundary condenser microphones are designed for surface-mounted applications such as high-quality sound reinforcement, professional recording and conferencing, television, and other demanding sound pickup situations. Their small size makes them ideal for use in applications where minimum visibility is required. Available in omnidirectional (half space) or hemi-cardioid (half-space cardioid) polar patterns, the microphones are enclosed in a rugged die-cast case and protected by two layers of perforated steel. The combination of heavy die-cast case and rubber non-slip bottom pad minimizes mechanical coupling of surface vibrations to the microphone.

AT841a Omnidirectional (Half Space) Mic



- 25´ cable with TA3F-type connectors for use between the microphone and the AT8531 power module is provided
- Built-in 3-position switch on the power module allows selection of off, on/flat response or on/low-roll-off
- Powered by a AA battery (1200 hrs) or 9-52V phantom power

AT841a Black finish.....**225.95** **AT841Wa** White finish.....**254.95**

AT851a Hemi-Cardioid Mic



- 25´ cable with TA3F-type connectors for use between the microphone and the AT8531 power module is provided
- Built-in 3-position switch on the power module allows selection of off, on/flat response or on/low-roll-off
- Powered by a AA battery (1200 hrs) or 9-52V phantom power

AT851a Black finish.....**225.95** **AT851Wa** White finish.....**254.95**

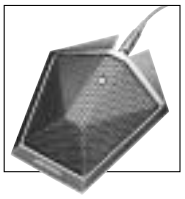
AT851RX Hemi-Cardioid Mic



- Requires 9V to 52V DC phantom power
- Recessed switch in the power module permits choice of flat response or low-frequency roll-off to help control undesired ambient noise
- Includes 25´ miniature cable with a TA3F plug for connecting to the microphone
- Output end connects to the provided AT8533x power module via internal solderless screw terminals for simple cable-length adjustment in the field

AT851RX Black finish ...**188.50** **AT851RWx** White finish...**216.95**

AT871R Hemi-Cardioid Mic



- Requires 9V to 52V DC phantom power
- A 25´ (7.6 m) cable is provided with TA3F and XLRM-type connectors for use between the microphone and electronics input

AT871R Black finish **235.95**
AT871RW White finish **262.50**

AT891R Hemi-Cardioid Mic with Programmable Switch

- Programmable three mode touch-sensitive on/off switch: touch-on/touch-off, touch-to-talk (momentary on) and touch-to-mute (momentary off)
- Red LED on the unit indicates when the microphone is "on."
- Powered from any external 11-52V DC phantom power supply



- A switch on the bottom of the unit permits choice of flat response or low-frequency roll-off to help control undesired ambient noise
- A 25´ (7.6 m) shielded cable with TA3F and XLRM-type connectors is provided for use between the microphone and electronics inputs

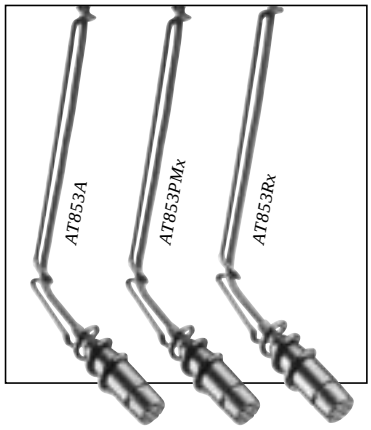
	AT-841a	AT-851a	AT-851RX	AT871R	AT891R
Polar Pattern	○	◠	◠	◠	◠
Frequency Response	30 - 20k Hz	30 - 20k Hz	30-20k Hz	30 - 20k Hz	30 - 10k Hz
Open Circuit Sensitivity[1] (Phantom/Battery)	-42 dB (7.9 mV)/-44 dB (6.3 mV)	-44 dB (6.3 mV) / -45 dB (5.6 mV)	-43 dB (7.0 mV)	-33 dB (22.4 mV)	-33 dB (22.4 mV)
Impedance (Phantom/Battery)	200 Ω / 270 Ω	200 Ω / 270 Ω	200 Ω	200 Ω	200 Ω
Maximum SPL Input[2] (Phantom/Battery)	133 dB SPL / 119 dB SPL	135 dB SPL / 121 dB SPL	138 dB SPL	130 dB SPL	123 dB SPL
Signal to Noise Ratio[3]	65 dB	67 dB	67 dB	67 dB	64 dB
Dynamic Range-Typical[4] (Phantom/Battery)	104 dB/90dB	108 dB/94dB	111 dB	103 dB	93 dB
Phantom Power (2.0mA Typical)	9 - 52V	9 - 52V	9 - 52V	9 - 52V	11 - 52V
Battery Type	1.5V AA (up to 1200 hrs.)	1.5V AA (up to 1200 hrs.)	—	—	—
Low-Frequency Roll-off	18dB/oct @ 80 Hz	18dB/oct @ 80 Hz	6dB/oct @ 150 Hz	—	18dB/oct @ 80 Hz

1. re 1V @ 1 Pa 2. 1 kHz at 1% T.H.D. 3. 1kHz @ 1 Pa 4. 1 kHz @ Max SPL

UNIPOINT SERIES

Hanging Condenser Microphones

Audio-Technica's wide-range of miniature cardioid condenser hanging microphones are designed for use in high-quality sound reinforcement, professional recording, television, and other demanding sound pickup applications. They are furnished with a vinyl-coated steel hanger that allows them to be positioned inconspicuously over a choir, orchestra, stage, etc., for very low-profile situations. Supplied as cardioid mics, they also accept easily interchangeable elements to permit selection of angle of acceptance from 100° to 360°. They all include an AT8102 windscreen, AT8451 steel hangar, power module, and a permanently attached 25´ cable. Elements are enclosed in either a low-reflectance black finish or a white-finished housing, cable and steel hanger.



AT853a

- Powered by a single AA battery for up to 1200 hours or 9-52V DC phantom power
- Built-in 3-position switch on the power module allows selection of off, on/flat response, or on/low-roll-off
- Roll-off is 18 dB/octave at 80 Hz
- 25´ permanently-attached cable to mic with TA3F connector at power module. Module output is XLRM-type

AT853a Black finish.....**197.50**
AT853Wa White finish.....**227.95**

AT853PMx

- Accepts 9-52v DC phantom power
- Roll-off of 6 dB/octave at 150 Hz is enabled by cutting jumper on PC board
- 25´ cable may be cut to any length and connected to screw terminals on the supplied wall/ceiling plate power module
- Power module features a white-finished standard electrical cover plate for easy, secure installation. Output is low impedance balanced via screw terminals

AT853PMx Black finish.....**179.50**
AT853PMWx White finish **204.95**

AT853Rx

- Accepts 9-52v DC phantom power
- Recessed switch in the power module permits choice of flat response, or a low-frequency roll-off of 6 dB/octave at 150 Hz
- 25´ cable connects to the power module via internal solderless screw terminals for simple length adjustment in the field. Module output is XLRM-type

AT853Rx Black finish **186.50**
AT853RWx White finish **213.50**

REPLACEMENT ELEMENTS

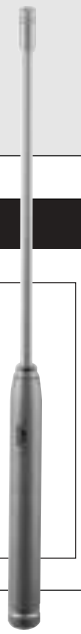
*Supplied as cardioid mics, the **AT851**, **AT853**, **AT857**, **AT871** and **AT873R** easily accept interchangeable elements to permit selection of angle of acceptance from 100° to 360°. Optional elements include:*

AT853C-ELE Cardioid
Supplied with AT851, AT853, AT857 and AT871. 120° frontal pickup pattern **54.50**

AT853H-ELE Hypercardioid
Supplied with AT873R. Increases directionality to provide better gain-before-feedback or reduction of background noise over other patterns. 100° pickup **54.50**

AT853O-ELE Omnidirectional
Allows operation in all directions with a 360° pickup pattern.....**54.50**

AT853SC-ELE Subcardioid
Allows a wide 170° frontal working area than a standard cardioid....**54.50**



HANDHELD CONDENSER MICS

AT873R Handheld Hypercardioid Condenser Mic



- Relatively small vocal mic well suited for lectern and stand applications
- Superior internal shock mounting eliminates handling and cable noise
- Balanced low impedance output
- Easily accepts optional interchangeable elements
- Requires 48V phantom power
- Rugged housing; low-reflectance black finish

AT859b Miniature Interview Mic

- Elegant, long and slender (16˝) handheld interview mic
- Cardioid condenser element
- Response is specifically tailored for crisp speech
- Battery or phantom power
- Includes stand clamp.....**197.95**

UNIPOINT SERIES

	AT-857AMa	AT-857AMx	AT-857QMa	AT-857QMLx	AT-853a	AT-853PMx	AT-853Rx
Polar Pattern	∩	∩	∩	∩	∩	∩	∩
Frequency Response	30 - 20k Hz	30 - 20k Hz	30-20k Hz	50-18k Hz	30 - 20k Hz	30 - 20k Hz	30-20k Hz
Open Circuit Sensitivity Phantom[1]	-44 dB (6.3 mV)	-43 dB (7.0 mV)	-43 dB (7.0 mV)	-46 dB (5.0 mV)	-44 dB (6.3 mV)	-33 dB (22.4 mV)	-43 dB (7.0 mV)
Open Circuit Sensitivity Battery[1]	-45 dB (5.6 mV)	—	—	—	-45 dB (5.6 mV)	—	—
Impedance (Phantom/Battery)	200Ω / 270Ω	200 Ω	200 Ω	200 Ω	200 Ω / 270 Ω	200 Ω	200 Ω
Maximum SPL Input[2] (Phantom/Battery)	135 dB/121 dB	138 dB	139 dB SPL	140 dB SPL	135 dB / 121 dB	130 dB	138 dB
Signal to Noise Ratio[3]	67 dB	67 dB	67 dB	65 dB	67 dB	67 dB	67 dB
Dynamic Range-Typical[4] (Phantom/Battery)	108 dB/94dB	111 dB	112 dB	111 dB	108 dB/94dB	103 dB	111 dB
Phantom Power (2.0mA Typical)	9 - 52V	9 - 52V	9 - 52V	9 - 52V	9 - 52V	9 - 52V	9 - 52V
Battery Type	1.5V AA	—	—	—	1.5V AA	—	—
Battery Current/Life (Typical)	0.4 mA	—	—	—	0.4 mA	—	—
Low-Frequency Roll-off	18dB/oct @ 80 Hz	6dB/oct @ 150 Hz			18dB/oct @ 80 Hz	6dB/oct @ 150 Hz	

1. re 1V @ 1 Pa 2. 1 kHz at 1% T.H.D. 3. 1kHz @ 1 Pa 4. 1 kHz @ Max SPL

Gooseneck Condenser Microphones

Audio-Technica's gooseneck wide-range condenser microphones are designed for quality sound reinforcement, professional recording, television, and other demanding sound pickup applications. Standing at either 12 or 19˝ above the mounting surface, their small-diameter alternating gooseneck design permits highly flexible positioning while maintaining a smooth, well-contoured appearance. Microphone elements are enclosed in a rugged housing with a low-reflectance black finish. Supplied as cardioid microphones, all except the AT859QMLx accept easily interchangeable elements for selection of angle of acceptance from 100° to 360°. Switchable low frequency roll-off filters minimize background ambiance, air conditioning rumble and mechanical noise, ensuring the highest sound quality, even under adverse conditions.



AT857AMa/AT857AMLa

- Built-in 3-position switch on the power module allows selection of off, on/flat response, or on/low-roll-off
- 9.8′ permanently-attached cable to mic with TA3F connector at power module. Module output is XLRM-type
- A 5/8˝-27 threaded flange allows the microphone to mate with the threads of a standard mic stand and can be permanently mounted using the included 5/8˝ - 27 threaded stud.

AT857AMa 12.2˝ long ..**235.95** **AT857AMLa** 19˝ long ..**249.95**

AT857AMx/AT857AMLx

- 9.8′ cable connects the mic to the power module via internal solderless screw terminals for simple length adjustment in the field. Module output is XLRM-type
- Recessed switch in the power module permits choice of flat response or a low-frequency roll-off
- A 5/8˝-27 threaded flange allows the microphone to mate with the threads of a standard mic stand and can be permanently mounted using the included 5/8˝ - 27 threaded stud.

AT857AMx 12.2˝ long .**219.95** **AT857AMLx** 19˝ long .**235.95**

AT857QMa/AT857QMLa

- A quick-mount version of the AT857AMx, the mic can be instantly installed and removed. 3-pin XLR-M connector insert at the base allows the AT857QMa to be plugged directly into any XLRF-type surface or cable connector
- Recessed switch in the power module permits choice of flat response or a low-frequency roll-off

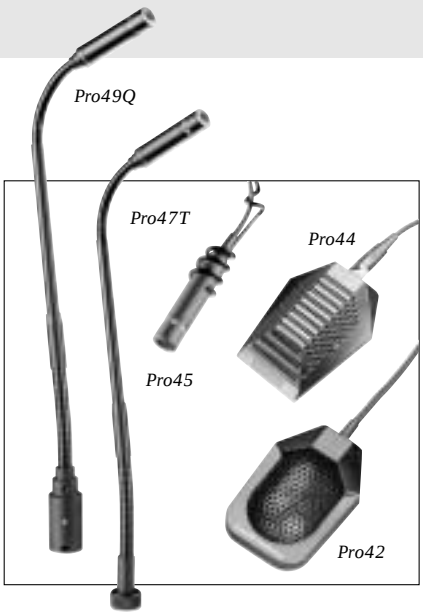
AT857QMa 14.5˝ long.....**197.95**
AT857QMLa 19˝ long **209.50**

AT859QMLx

- Quick-mounting, the AT859QMLx has a 3-pin XLR-M connector insert at the base allowing it to be instantly installed and removed. Plugs directly into any XLRF-type surface or cable connector
- Recessed switch in the power module permits choice of flat response, or a low-frequency roll-off of 6 dB/octave at 150 Hz

AT857QMLx 18.5˝ long **136.50**

PRO SERIES— ProPoint Mics



Installed-Sound Condenser Microphones

From ceilings to stage floors, from podiums to tabletops, ProPoint microphones have your miking needs covered.

- ➤ Wide-range condenser elements with low-mass diaphragms for superior performance
- ➤ Uniform cardioid polar pattern for maximum rejection of unwanted sounds
- ➤ Flat, smooth frequency response, high-SPL handling for voice, choir and orchestra
- ➤ They operate on 9-52v DC phantom power
- ➤ Self-contained electronics eliminate the need for external power modules
- ➤ Podium mics feature quiet, fixed-placement, long-life goosenecks
- ➤ Low-profile boundary mics for minimum visibility
- ➤ Rugged design and construction for reliable service

	PRO 42	PRO 44	PRO 45	PRO 47T / TL	PRO 49Q / QL
Element	Condenser	Condenser	Condenser	Condenser	Condenser
Polar Pattern	◠	◠	◠	◠	◠
Frequency Response	70 - 14k Hz	70 - 16k Hz	70 - 16k Hz	70 - 16k Hz	70 - 16k Hz
Open Circuit Sensitivity	-38 dB (12.5 mV)	-25 dB (56.2 mV)	-37 dB (14.1 mV	-37 dB (14.1 mV)	-37 dB (14.1 mV)
Impedance	100 Ω	100 Ω	100 Ω	100 Ω	100 Ω
Maximum SPL Input[1]	134 dB SPL	114 dB SPL	134 dB SPL	134 dB SPL	134 dB
Signal to Noise Ratio[2]	65 dB	66 dB	66 dB	66 dB	66 dB
Dynamic Range (Typical)[3]	105 dB	86 dB	106 dB	106 dB	106 dB
Phantom Power (2.0mA Typical)	9 - 52V	9 - 52V	9 - 52V	9 - 52V	9 - 52V
Price	**84.95**	**92.95**	**84.95**	**106.95 / 114.95**	**92.95 / 96.95**

1. 1 kHz at 1% T.H.D. 2. 1kHz @ 1 Pa 3. 1 kHz @ Max SPL

Pro42/Pro44

Cardioid Boundary Mics

The miniature PRO42 and low-profile Pro44 are wide-range condenser microphones with a hemi-cardioid (half-space cardioid) polar pattern designed for surface-mounted applications in high-quality sound reinforcement and other demanding sound pickup situations.

- ◆ Miniature PRO42 measures 1.44 x 2.14 x .61″ (W x D xH) while the low-profile PRO44 measures 2.87 x 3.56 x .59″ (W x D xH)
- ◆ The PRO42 (permanently-attached) and PRO44 (detachable) include a 25' cable with XLRM-type connector
- ◆ Mic elements are enclosed in a a rugged case and protected by heavy perforated steel. The rubber non-slip bottom pad minimizes mechanical coupling of surface vibrations to the microphone

Pro45/Pro45W

Cardioid Hanging Mics

Designed for high-quality sound reinforcement, the PRO45 is furnished with a vinyl-coated steel hanger that allows it to be positioned inconspicuously over a choir, orchestra, stage, etc., for low-profile situations.

- ◆ Has a 25′ permanently- attached cable with XLRM-type connector
- ◆ Enclosed in a rugged housing with a low-reflectance black finish. Also available with a white-finished (PRO45W) mic housing, cable and steel hanger for the times you need the mic hung against a light background

Pro47T/Pro47TL

Cardioid Thread-Mount Gooseneck Mics

The PRO47T and PRO47TL are designed for high-quality sound reinforcement and other demanding sound pickup situations. Standing 12.4″ (PRO47T) and 15.7″ (PRO47TL) above the mounting surface, their small-diameter alternating gooseneck design permits highly flexible positioning while maintaining a smooth, well-contoured surface.

- ◆ 9.8′ permanently-attached cable with an XLRM-type connectors
- ◆ Mic element is enclosed in a rugged housing with a low-reflectance black finish. A 5 /8″-27 threaded flange allows the mic to mate with the threads of a standard microphone desk or floor stand. A 5/8″ -27 threaded stud is also provided so it can be secured to a permanent mounting surface

Pro49Q/Pro49QL

Cardioid Quick-Mount Gooseneck Mics

With an overall length of 13.07″ (PRO49Q) and 16.46″ (PRO49QL), their small-diameter alternating goose-neck design permits highly flexible positioning while maintaining a smooth, well-contoured appearance.

- ◆ XLRM-type connector insert at their base allows them to be plugged directly into an XLRF-type jack or cable connector

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ONE HOUR FREE PARKING
AT 349 W. 34th STREET (with purchase of $100 or more)

PRO SERIES – Handheld/Stand Mics

With an incredibly wide selection of offerings, the Pro Series has a microphone to suit any budget and any application. Includes a hefty assortment of handheld, lavalier, instrument, gooseneck and headphone microphones.

PRO 2ax Handheld Dynamic Cardioid Mic


PRO 2ax

The PRO 2ax is dynamic microphone designed for close-up vocal and instrumental use. Very affordable, the PRO 2ax offers unusually smooth response and uniform off-axis rejection of unwanted sound due to its cardioid pickup pattern. Cardioid pattern reduces unwanted background noise and controls feedback, especially when used extremely close. Miking instruments or vocalists at very close range also effectively reduces "leakage" from one performer to the next. Low-frequency noise is also reduced by a low-frequency roll-off of sounds more than one foot away from the microphone.
When used close-up, bass response increases because of proximity effect. This characteristic adds a mellow sound to both voice and instruments.
Enclosed in a rugged housing with a low-reflectance finish and a conveniently located on/off switch, the mic has a 2-stage ball-type screen to reduce wind noise and "popping" when performers work really close. Also includes a 5 meter (16.5′) cable with an XLR-F connector at the mic end, and a standard 1/4″ phone plug at the equipment end **34.95**

PRO 4L Handheld Dynamic Cardioid Mic


PRO 4L

Dynamic microphone designed for close-up vocal and instrumental use, the PRO 4L is also ideal for general stage use and other sound reinforcement applications. Unusually smooth response, uniform off-axis rejection of unwanted sounds, and dependable performance have made the PRO 4L the industry standard in its class. When used ultra-close, the PRO 4L provides proximity effect (extra bass emphasis) which can be used to achieve a fuller sound, or to further reduce feedback in conjunction with equalization of the microphone input.
The PRO 4L is enclosed in a rugged housing with a low-reflectance finish and a conveniently located on/off switch. Internally shock-mounted capsule reduces handling and cable noise while the multi-stage windscreen effectively controls "popping" and wind noise. There is also a conveniently located on/off switch. Includes a 5 meter (16.5′) XLR cable and soft case **79.50**

PRO 5x Handheld Dynamic Cardioid Mic


PRO 5x

PRO 10HE

Designed for use in high-quality sound reinforcement systems and to meet the demands of professional musicians, the PRO 5X is equally suited to any application requiring absolute dependability and where quality sound reproduction cannot be sacrificed. Response is tailored to provide natural, full-range reproduction when used by vocals and instrumental performers at very close distances. The PRO 5X is enclosed in a rugged housing with a low-reflectance finish. Superior internal shock mounting provides effective isolation from handling and cable noise while the multi-stage windscreen effectively controls "popping", distortion, and wind noise. Includes a 20′ XLR cable and soft case.....**87.95**

PRO 10HE Handheld Dynamic Cardioid Mic

A dynamic hypercardioid microphone, the PRO 10HE features a Hi-ENERGY neodymium magnet with almi-chromate magnet coating to ensure excellent transient response coupled with high efficiency. Designed for high-quality recording and sound reinforcement applications, the PRO 10HE has a double-dome diaphragm that unlike most dynamic mics provides both high sensitivity and extremely fast transient response for musical instrument and orchestral pickup **109.95**

	PRO 2ax	PRO 4L	PRO 5x	PRO 10HE
Element	Dynamic	Dynamic	Dynamic	Neodymium Dynamic
Polar Pattern	◠	◠	◠	◠
Frequency Response	90 - 13k Hz	60 - 14k Hz	60 - 16k Hz	60 - 15k Hz
Open Circuit Sensitivity[1]	-40 dB (10.0 mV)	-58 dB (1.2 mV)	-57 dB (1.4 mV)	-53 dB (2.2 mV)
Impedance	8000 Ω	250 Ω	250 Ω	600 Ω
Weight	5.1 oz (145 g)	10.1 oz (287 g)	11.5 oz (328 g)	8.6 oz (244 g)

1. re 1V at 1 Pa

PRO SERIES - Specialty Instrument Mics

PRO 7a

Miniature Cardioid Condenser Lavalier/Instrument Microphone

A miniature cardioid condenser mic, the PRO 7a is designed for pickup of acoustic guitar and other acoustic instruments, and for hands-free applications in sound reinforcement systems. Close-up voice and instrument pickup are crisp and clean, yet full sounding, while suppression of background noise is significantly improved over that of stand-mounted full-size cardioid microphones.

- Mic element is enclosed in a rugged housing with low-reflectance finish
- Internal construction minimizes handling and clothing noise
- A built-in 3-position switch on the power module allows selection of off/flat response/low-roll-off
- Integrated 6-ft. cable with XLR-M connector between the mic and power module
- Runs 3500 hours on a single AA battery
- Includes clothing clip, guitar adapter and windscreen



PRO 7a



PRO 25

PRO 25

High-SPL Hypercardioid Dynamic Instrument Microphone

With high SPL capability, the PRO 25 meets the critical requirements of professional musicians and high-quality sound reinforcement systems. Efficient motor system, large diaphragm and well-controlled polar pattern make it very effective in picking up highly dynamic instruments such as drums (kick, tom, snare), timpani, piano, acoustic and electric bass, and trombone. Also suitable for vocal pickup where low-frequency emphasis is desirable. When used ultra-close, the PRO 25 provides proximity effect (extra bass boost) which can be used to achieve a fuller sound.

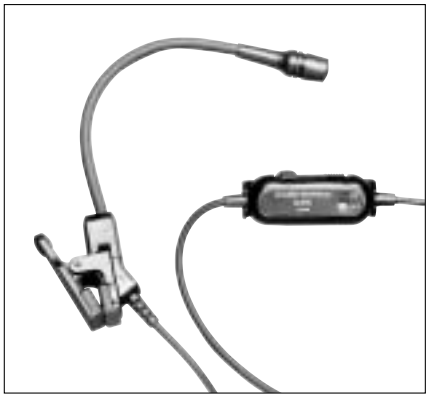
- Rugged housing with a low-reflectance finish
- Internal shock mount provides effective isolation of the mic element from handling and cable noise
- Multi-level steel grille is extremely sturdy and provides "pop" protection for vocals
- Has an XLR output (no cable) and includes an integral stand clamp and soft protective pouch

PRO 35

High-Intensity Cardioid Clip-on Instrument Microphone

A miniature cardioid condenser mic, the PRO 35 is uniquely capable of picking up high-intensity sound without distortion. With a very small diaphragm (for excellent transient response) and the ability to handle high sound pressure levels, it is an effective tool in recording and sound reinforcement applications. Ideal for closely-miked instruments such as woodwinds, brass, strings, piano and specifically, snare and toms. The supplied AT8418 UniMount surrounds the mic with foam, shielding the element against accidental damage, isolating the mic from noise, and providing an efficient windscreen. The flexible boom lets you aim the microphone in any direction, while cable strain reliefs reduce cable-transmitted noise and cable damage. The "rubber grip" spring clamp holds the microphone securely to the instrument without scratching the surface. A 5.9′ cable is provided between the mic and power module. A switch in the power module permits choice of flat response or low-frequency roll-off to help control undesired ambient noise. Also available as the PRO 35xcW: Same as above less power module and with a 55″ (1.4 m) cable terminated with 4-pin HRS connector for Audio-Technica UniPak wireless system.





PRO 95

Acoustic Instrument Condenser Microphone

The PRO 95 is designed for professional musicians, especially for reproduction of acoustic guitar and other string instruments. Uniquely capable of being mounted inside a guitar, the mic provides sound pickup that is crisp and clean, yet full sounding.

A "rubber-grip" spring clamp holds the microphone securely to the instrument without marring the surface. The flexible boom allows you to aim the mic in any direction, while a cable strain relief reduces cable-transmitted noise and the possibility of cable damage. The mic element is enclosed in a rugged housing with a low-reflectance finish. The PRO 95 is permanently connected to an ultra-flexible 24' cable and in-line power module with an on/off switch. A clip for the power module and a battery (runs for about 400 hours on a single 1.5v LR44-type battery) are provided as well as a wind screen. Also includes an end-pin jack and 1/4″ phone plug for optional permanent installation inside a guitar.

PRO SERIES Mics

PRO 8HEx

Hi-ENERGY Headworn Hypercardioid Dynamic Microphone



The PRO 8HEx is a light-weight, headworn dynamic mic ideal for performing musicians, aerobics instructors and for anyone who needs a professional-quality mic with hands-free operation. Featuring a Hi-ENERGY Neodymium magnet with almi-chromate coating, the motor structure ensures excellent transient and frequency response, reduces distortion, and provides a high output level not normally achieved by other headworn dynamic microphones. The result is crisp, clean vocal pickup.

The combination of hypercardioid polar pattern and headworn design provides suppression of background noise that is significantly improved over that of full-size, stand-mounted cardioid microphones.

Features include a mic capsule moisture barrier, a durable cable assembly, and a covered stainless steel headband which can be gently flexed to adjust for a comfortable, stable fit around the back of any head. The Pro 8HEx includes a permanently attached 7.2′ cable with XLR-M connector, two wind screens (large and small) and a clothing clip.



PRO 37R

Small-Diaphragm Cardioid Condenser Recording Microphone

A wide-range cardioid condenser mic, the PRO 37R is an excellent choice for high-quality sound reinforcement systems and for pickup of instruments such as drums, piano and brass. It can also be used for professional recording, broadcasting and situations demanding the highest-quality sound performance coupled with excellent reliability in a small package.

With the addition of and AT8106 metal-frame pop filter, the PRO 37R becomes a studio-grade vocal and dialog mic suitable for professional studio and sound reinforcement applications. The PRO 37R is enclosed in a rugged housing with a low-reflectance finish. Internal shock mounting is designed to minimize handling and cable noise. Includes snap-in clamp for standard 5/8″-27 threaded stands and a foam windscreen. Accepts external phantom power from 9v to 52v DC, output is via XLR-M.

PRO 11

Single -Gooseneck Unidirectional Dynamic Console Microphone



A uni-directional dynamic console mic designed for use as a quality talk-back microphone, the PRO 11 provides excellent intelligibility and dependable performance in entertainment, commercial and industrial voice applications. Features include a conveniently-located on/off switch, balanced low-impedance output, and an integral 3-pin XLR connector that plugs directly into a panel jack or mic cable. A protective screen reduces wind noise and “popping” when used extremely close, while internal construction reduces handling and shock noise.

	PRO 7a	PRO 25	PRO 35x	PRO 95	PRO 8HEx	PRO 37R	PRO 11
Element	Condenser	Dynamic	Condenser	Condenser	Neodymium Dynamic	Condenser	Dynamic
Polar Pattern	◠	◠	◠	◠	◠	◠	◠
Frequency Response	300 - 15k Hz	30 - 12k Hz	50 - 17k Hz	80 - 13k Hz	200 - 18k Hz	30 - 15k Hz	150 - 13k Hz
Open Circuit Sensitivity	-45 dB (5.6 mV)	-54 dB (1.9 mV)	-54 dB (1.9 mV)	-40 dB (10.0 mV)	-55 dB (1.7 mV)	-42 dB (7.9 mV)	-55 dB (1.7 mV)
Impedance	600 Ω	600 Ω	200 Ω	2000 Ω	600 Ω	200 Ω	500 Ω
Maximum SPL Input[1]	124 dB SPL	—	149 dB SPL	105 dB SPL	—	141 dB	—
Signal to Noise Ratio[2]	59 dB	—	60 dB	56 dB	—	65 dB	—
Dynamic Range (Typical)[3]	89 dB	—	115 dB	67 dB	—	112 dB	—
Phantom Power (2.0mA Typical)	—	—	9 - 52V	—	—	9 - 52V	—
Battery Type	1.5V AA	—	—	1.5V LR44-type	—	—	—
Battery Current/Life (Typical)	0.2 mA/3500 hrs	—	—	0.4 mA/400 hrs.	—	—	—
Low-Frequency Roll-off[4]	150 Hz	—	150 Hz	—	—	—	—
Weight	0.1 oz (2.8 g)	9.5 oz (270 g)	0.25 oz (7 g)	3.4 oz (95 g)	2.1 oz (60 g)	2.1 oz (60 g)	9.9 oz (280 g)
Price	**89.95**	**118.95**	**CALL**	**117.50**	**69.95**	**136.50**	**52.50**

1. 1 kHz at 1% T.H.D. 2. 1kHz @ 1 Pa 3. 1 kHz @ Max SPL 4. 6dB per octave

SPECIAL PURPOSE MICS

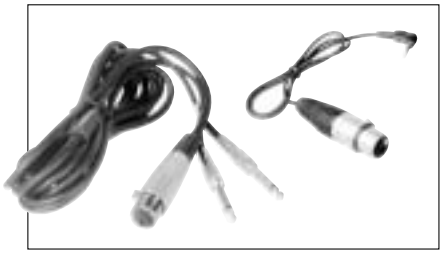
AT822/AT825 OnePoint X/Y Stereo Mics



The AT822 and AT825 are X/Y stereo condenser microphones designed for broadcast, professional recording and field applications. The AT822 features unbalanced connections ideally suited for use with portable MiniDisc and DAT recorders, or for mounting onto a prosumer camcorder. On the other hand, the AT825 features balanced XLR connections compatible with professional mic preamps, mixers and video cameras, as well as the more substantial digital field recorders. Both the AT822 and the AT825 provide excellent channel separation with full, natural stereo ambience in a compact, lightweight design and are equally at home mounted on a stand, fishpole or camera and can even be as hand-held mics for electronic news gathering. The two wide-range, closely-matched, optimally-positioned cardioid condenser elements exhibit a uniform frequency response throughout their polar range. This provides the spatial impact and realism of a live sound field—while simultaneously remaining fully mono (L+R) compatible.



- Consistent natural-sounding response over an arc of approximately 170° (AT822) and 220° (AT825)
- Excellent channel separation
- Switchable low-frequency roll-off
- Distortion-free signal even in sound fields as high as 125 dB SPL
- The AT822 runs on a single AA, the AT825 can run on phantom power or a AA battery



- AT822 includes a 1.7′ cable (XLR-female to stereo 3.5 mm mini-plug); and a 10′ cable (XLR-female- to two mono 1/4″ plugs); gun-grip/stand clamp; camera shoe-mount adapter; windscreen; battery; case
- AT825 includes a 16.5′ cable (5-pin XLR to two standard 3-pin XLR connectors), stand clamp; windscreen; battery; case.

AT849 Stereo Condenser Boundary Mic



The AT849 is a X/Y stereo condenser boundary microphone designed for surface-mounting. The two wide-range, optimally-positioned miniature condenser cardioid elements provide natural stereo ambience ideally suited for broadcast, professional recording and live sound reinforcement applications.

- Consistent, natural sound over an arc of approximately 220°
- Fully mono compatible L+R polar pattern is virtually free of lobes and comb effects.
- Uniform polar response over an extended, smooth and peak-free frequency response of 30Hz to 20kHz
- Requires 9V to 52V DC phantom power per channel
- 2-position switch selects flat or low-roll-off response
- Rugged low-profile housing protected by two layers of perforated steel. The heavy case and rubber non-slip bottom pad minimizes mechanical coupling of surface vibrations to the microphone.
- Low-reflectance black finish
- Includes a soft pouch and 25' shielded cable with a five-pin TA5F input connector and two standard three-pin male XLR output connectors

AT854R Multi-channel Condenser Boundary Mic



A multi-channel boundary microphone with four cardioid condenser elements mounted in a low-profile housing that ensures minimum visibility. Ideal for high-quality sound reinforcement and a variety of video and audio conferencing applications. The four independently phantom powered quadrants can be used individually or in any combination and integrated easily with a standard or automatic mic mixer.

- Four mic channels mounted in separate quadrants
- Rugged low-profile housing protected by two layers of perforated steel. The heavy case and rubber non-slip bottom pad minimizes mechanical coupling of surface vibrations to the microphone.
- Requires 9-52V DC phantom power per channel.
- Includes a rugged 30′ cable with a miniature multi-contact metal-body plug and four separate shielded and jacketed pairs terminating to four male XLR connectors
- Low-reflectance black finish

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SPECIAL PURPOSE MICS

	AT822	AT825	AT849	AT854R	AT804	AT813a
Element	Condenser	Condenser	Condenser	Condenser	Dynamic	Condenser
Polar Pattern	∩ (x2)	∩ (x2)	∩ (x2)	∩ (x4)	○	∩
Frequency Response	30 - 20k Hz	30 - 20k Hz	30-20k Hz	70-16k Hz	50 - 15k Hz	30 - 20k Hz
Open Circuit Sensitivity Phantom[1]	-45 dB (5.6 mV)	-47 dB (4.4 mV)	-41 dB (8.9 mV)	-37 dB (14.1 mV)	-49 dB (6.3 mV)	-44 dB (6.3 mV)
Open Circuit Sensitivity Battery[1]	—	—	—	—	—	-45 dB (5.6 mV)
Impedance (Phantom/Battery)	200Ω	200 Ω	200 Ω	100 Ω	600 Ω	200 Ω / 270 Ω
Maximum SPL Input[2] (Phantom/Battery)	125 dB	126 dB	137 dB	134 dB	—	137 dB / 123 dB
Signal to Noise Ratio[3]	70 dB	70 dB	67 dB	65 dB	—	70 dB
Dynamic Range-Typical[4] (Phantom/Battery)	101 dB	102 dB	110 dB	105 dB	—	113 dB / 99 dB
Phantom Power (2.0mA Typical)	—	5 - 52V	9 - 52V	9 - 52V	—	9 - 52V
Battery Type	1.5V AA	1.5V AA	—	—	—	1.5V AA
Battery Current/Life (Typical)	0.4 mA	0.4 mA	—	—	—	0.4 mA
Low-Frequency Roll-off	6dB/oct @ 150 Hz	6dB/oct @ 150 Hz	6dB/oct @ 150 Hz	—	—	—
Price	**239.95**	**339.95**	**329.95**	**417.50**	**88.95**	**149.95**

1. re 1V @ 1 Pa 2. 1 kHz at 1% T.H.D. 3. 1kHz @ 1 Pa 4. 1 kHz @ Max SPL

AT804 Omnidirectional Moving Coil Dynamic Mic

Designed for use in professional recording and broadcasting studios, high quality sound reinforcement, and other demanding sound pickup situations. Hardened-steel grille and die-cast case make the mic ideal for field applications where rugged construction is essential. The omnidirectional polar patterns ability to pick up sound equally from all directions means that with close-up miking, the mic won't accentuate bass frequencies due to proximity effect. The omnidirectional pick-up pattern is also ideal for distant miking where the reproduction of ambient environments is desired.



- Rugged die-cast housing with a low-reflectance finish
- A protective hardened-steel screen reduces wind noise and plosives
- Internal shock mounting reduces handling and cable noise.
- Includes a snap-in stand adapter with 5/8″-27 threads and a soft protective pouch are included

813a Battery/Phantom Cardioid Condenser Mic



Designed for use in high-quality sound reinforcement systems and for use by professional musicians, especially for vocal pickup. It can also be used for professional recording, broadcasting and other applications demanding the highest quality sound performance coupled with excellent reliability.

- Powered by a single AA battery for up to 1200 hours or 9-52V DC phantom power
- Cardioid polar pattern controls feedback and provides extra bass emphasis (proximity effect) when used ultra-close
- Low-mass diaphragm ensures minimum distortion
- Large windscreen is especially effective in reducing "popping" and distortion from sibilant sounds
- Includes stand clamp; battery and protective pouch

AT808G Subcardioid Dynamic Console Mic

Designed for use as a quality talk-back microphone.



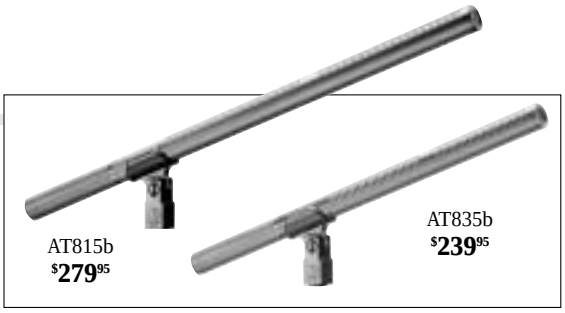
- Dependable performance for entertainment, commercial and industrial applications
- Custom-tailored frequency response ensures excellent intelligibility in environments with excessive ambient noise
- Protective windscreen and rugged construction
- Plugs directly into an XLRF-type surface or cable connector
- Overall length of 16.2″

SHOTGUN MICS

AT815b and AT835b

Line + Gradient Condenser Mics

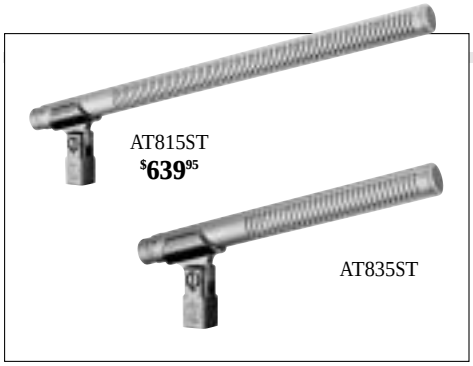
The 18˝ AT815b and 14.5˝ AT835b are wide-range condenser microphones with a line + gradient polar pattern specifically designed to provide the narrow acceptance angle desirable for long distance sound pickup. They are ideal for use in professional recording, broadcasting, film/ TV/video production, high-quality sound reinforcement and other demanding applications requiring excellent sound rejection from the sides and rear. The shorter AT835b is well suited for boom/fishpole use as well as mounting on professional video cameras, whereas the longer AT815B has a narrower acceptance angle and is particularly useful for applications that require a longer reach.



- Rugged housing with a low-reflectance matte satin nickel finish
- Recessed switch lets you choose either flat or a bass roll-off of 180 Hz at 12dB/octave
- Powered by a single AA battery or 9-52v DC phantom power
- Includes AA battery, foam wind-screen, microphone stand adapter with 5/8˝-27 threads, and a compartment carrying case designed to hold the microphone and accessories

	AT-815b	AT835b
Element	Condenser	
Polar Pattern	Line + Gradient	
Frequency Response (AT815B / AT835B)	30-20k Hz	40-20k Hz
Open Circuit Sensitivity[1] (Phantom/Batt)	-38dB (12.5mV) / -39dB (11.2mV)	
Impedance (Phantom/Batt)	500 Ω / 600 Ω	
Maximum SPL Input[2] (Phantom/Batt)	131 dB / 115 dB	
Signal to Noise Ratio[3]	70 dB	
Dynamic Range-Typical[4] (Phantom/Batt)	106 dB / 91dB	
Phantom Power (2.0mA Typical)	9 - 52V	
Battery Type (Life)	1.5V AA (up to 1200 hrs.)	
Low-Frequency Roll-off	12dB/oct @ 180 Hz	

1. re 1V @ 1 Pa 2. 1 kHz at 1% T.H.D. 3. 1kHz @ 1 Pa 4. 1 kHz @ Max SPL



AT815ST/AT835ST

Stereo Condenser Shotgun Microphones

Developed at the request of broadcasters for cost effective, compact stereo shotguns, the 9˝ AT815ST and 14˝ AT835ST combine professional audio quality, outstanding versatility, ease of use and robust construction, making them essential audio tools for today's high-quality stereo video production. Both mics feature independent line-cardioid and figure-of-eight condenser elements to provide a choice of three stereo modes including non-matrixed M-S (Mid-Side) stereo as well as two internally-matrixed left/right stereo modes.

- Two internally-matrixed left/right stereo modes and non-matrixed M-S mode are switch selectable
- Switchable low-frequency roll-off
- Compact, lightweight design provides exceptional mounting versatility including camera-mount
- Includes foam wind-screen, microphone stand adapter with 5/8˝-27 threads, and compartment carrying case

	AT-815ST	AT-835ST
Element	Condenser	Condenser
Polar Pattern	Line-Cardioid & Figure of 8	
Frequency Response (AT815B / AT835B)	30-20k Hz	40-20k Hz
Open Circuit Sensitivity[1] (Mid/Side/LR Stereo)	-30 dB (31.6 mV) /-34 dB (19.9 mV) /-36 dB (15.8 mV)	
Impedance	200 Ω	200 Ω
Maximum SPL Input[2] (Mid/Side/LR Stereo)	123 dB / 127 dB / 126 dB	
Signal to Noise Ratio[3]	72 dB / 68 dB / 70 dB	
Dynamic Range-Typical[4] (Mid/Side/LR Stereo)	101 dB / 101 dB / 102 dB	
Phantom Power (4mA Typical at 48v)	11 - 52V	11 - 52V
Low-Frequency Roll-off	12dB/oct @ 80 Hz	

1. re 1V @ 1 Pa 2. 1 kHz at 1% T.H.D. 3. 1kHz @ 1 Pa 4. 1 kHz @ Max SPL

THREE STEREO MODES

M-S mode provides an independent cardioid element (Mid) and figure-8 element (Side). The advantage of this design is that the stereo balance of the separate Mid-Side signals can be adjusted later on in post-production instead of in the field.

Two internally-matrixed modes provide traditional "left-right" stereo that accommodates varying acoustic environments. The LR-W pattern is wider, allowing increased ambient pickup, while the LR-N pattern is more narrow to offer greater rejection with less ambience.

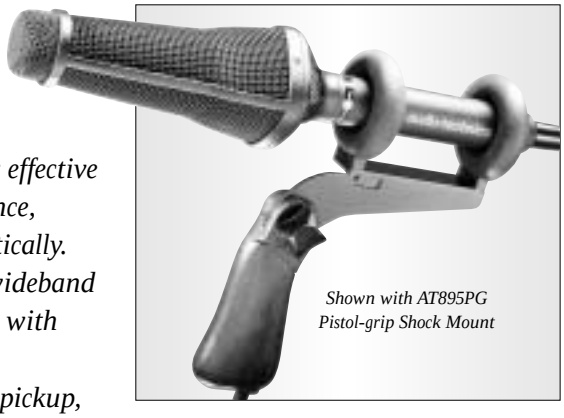
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ONE HOUR FREE PARKING
AT 349 W. 34th STREET (with purchase of $100 or more)

AT895

Adaptive-Array Microphone System

Utilizing AT's proprietary DeltaBeam technology, the AT895 system manipulates the amplitude and phase of its 5-element array by acoustical, analog and digital means, providing off-axis rejection of up to an astounding 80 dB.

The system continuously adapts to a changing soundscape, and is especially effective at suppressing unwanted low-frequency information. Background interference, mechanical noise, wind noise, proximity effect—all are minimized automatically. And, unlike some attempts at directive devices, the audio quality remains wideband and natural. The AT895 provides great operating versatility and ease of use with three polar-pattern choices, three EQ selections and three powering options. Functioning equally well for handheld interview use and long-range sound pickup, the AT895 excels at tasks from ENG to sports broadcast to film production to surveillance operations.



Shown with AT895PG Pistol-grip Shock Mount

About DeltaBeam Technology

DeltaBeam technology is a unique, groundbreaking optimization of acoustic, analog and digital design that produces unmatched directional performance, operating versatility and ease of use.

- Provides an adaptive array to substantially reduce background acoustic interference
- Improves sensitivity, pickup distance and gain-before-feedback as compared to currently-available professional microphone technology
- Provides a tool for the discrimination of specific sources, even in very high-noise environments
- Provides greatly increased off-axis rejection, even for very loud interference
- Provides outstanding low-frequency directionality (an amazing 78dB maximum rejection at 200 Hz!) in a mic only 14″ long
- Minimizes the audibility of proximity effect
- Minimizes nearfield effect on the low-frequency directionality of the array
- Reduces susceptibility to mechanical noise, wind noise and racking as compared to other professional mics

The Ultimate Audio Acquisition Solution

High-directivity, wideband audio quality, and the ability to suppress unwanted low-frequency information make the AT895 the best solution for many broadcast, film/video and surveillance/security applications.

Used in football or soccer, the AT895 delivers vivid audio of player contact, ball strikes and sideline conversations. Covering golf, it eliminates wind noise rumble while providing precise directivity. Also very effective in track & field, equestrian events and tennis.

The AT895's ability to attenuate audio clutter that masks desired sounds is invaluable for a multitude of field production situations such as interviewing, sound effects gathering and dialog acquisition.

The same capabilities that make the AT895 such an effective tool in broadcast also apply to security applications. Used alone or in conjunction with surveillance cameras, the AT895 provides exact audio pickup in locations such as hotels, casinos, banks, schools and parks.

AT895 vs. TYPICAL SHOTGUN PERFORMANCE*

Compare acceptance angles (at 3 dB down):

Microphone	200 Hz	400 Hz	1 kHz	2 kHz	4 kHz	8 kHz
Shotgun	60°	60°	60°	50°	30°	20°
AT895	20°	20°	20°	20°	60°	50°

Compare polar patterns at 200 Hz:
Microphone - 200Hz Rejection at 90°/270°
Shotgun.....15 dB AT895.....70 dB

Compare maximum off-axis rejection:
Microphone - Maximum Off-axis Rejection
Shotgun.....30 dB AT895.....80 dB

**Compared to typical performance of a professional-quality 15.5″ line + gradient microphone (measurement conditions: 1Pa at 0.5 m). Due to the adaptive nature of its digital processing, AT895 performance in actual field conditions will vary with the environment encountered.*

The AT895 is available in two system configurations: As the AT895/RK "Remote Kit" for field use, and as the AT895/MK "Mount Kit" for fixed-install applications.

AT895/RK Remote Kit

- **AT895MIC** Adaptive-array Microphone
- **AT895CP** Control Pack w/ attached belt clip
- **AT895BH** Click-on Battery Housing
- **AT895PG** Pistol-grip Shock Mount
- **AT895Z** Zeppelin-type Windscreen
- **AT895C-10** 6-conductor shielded 10′ cable
- **AT895CC** Carrying Case

AT895/MK Mount Kit

- **AT895MIC** Adaptive-array Microphone
- **AT895CP** Control Pack
- **AT895SC** Shock-mount Stand Clamp
- **AT895PS1** Multi-input AC Adapter with "120v" IEC detachable power cord
- **AT895C-25** 6-conductor shielded 25′ cable

AT-MX341A / AT-MX351

4-and 5-Channel SmartMixers



The ATMX-341A and AT-MX351 are automatic mixers ideal for meetings, seminars, teleconferencing, worship services, courtroom proceedings, radio and TV broadcasting and A/V applications. All switching is under microprocessor control with proprietary algorithms for instant, seamless operation. An individual priority pre-select switch for each mic/line channel allows choice of one-channel-at-a-time operation, all active, or any mix of moderator override. The last microphone on stays on so that ambient sound is never muted. Plus, any number of SmartMixers can be linked to create a system for large multi-mic installations. The priority control system extends throughout the entire chain.



FEATURES

What is a SmartMixer?

Field-proven to be the easiest, most flexible way to automatically control a sound system, SmartMixers deliver fast, quiet and incredibly "smart" performance in a variety of applications. By keeping the number of open channels to a minimum, they reduce background noise and other distractions. And they provide instant, completely transparent switching between channels. Now available in two models, SmartMixers offer a host of innovative features that ensure easy setup, effortless operation and unrivaled versatility.

Flexible Operation

SmartMixers feature priority pre-select switches for each channel. If all channels are priority "off" (no one has priority), the system opens just one microphone at a time, switching to the next only after the controlling channel falls quiet. The operation is so fast and silent that this is the recommended mode. A single channel set to priority "on" will override all the others, useful for moderators and chairpersons. If all channels are priority "on," any number of people can talk at once. Regardless of the priority switch settings, all mics not in use are attenuated by the mixer. SmartMixers also have a switchable manual mode, permitting them to function as conventional mixers.

AT-MX341a

- Four Mic/Line inputs with individual priority pre-select switch for each channel
- Balanced inputs and output are individually switchable between mic and line-level
- 12v phantom power individually selectable for each channel
- Single threshold setting for all channels
- TTL logic output signals available to control speaker or camera switching, tally lights, etc.
- -20 dB adjustable attenuation for inputs
- Mounts in a single 19″ half-rack space, includes rack mount adapters

They Both Feature:

- Master output level control
- Switchable manual mode overrides automatic functions
- Individual gain controls for each channel
- Output level meter with Peak and RMS modes

AT-MX351

- Four automatic mic/line inputs, plus Aux in
- Automatic threshold setting
- 48v phantom power individually selectable for each channel
- Limiters on each mic channel prevent individual channel overload
- Balanced inputs and output, selectable Mic/Line-level
- Pre- or post-controller audio outputs from Mic/Line channels
- Sub-D connector for TTL output, plus closure-control input for external control of each channel
- Off attenuation for each input adjustable to -40 dB
- NOMA operation (can be turned on/off)
- Headphone output with adjustable level
- Internal AC supply
- Mounts in a single 19″ rack space, includes rack mount adapters

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MICROPHONE ACCESSORIES

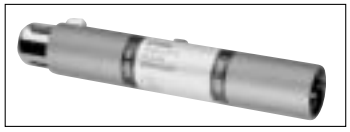
AT8801/CP8506
Single- and Four-Channel 48v Phantom Power Supplies



The AT8801 and CP8506 are 120v AC powered devices that provide 48v phantom power to remote-powered microphones such as those in the 40 Series, UniPoint, and Artist Series.

The AT8801 powers a single microphone while the CP8506 can power up to four. Their rugged steel case is finished in enamel and provides shielding from electrostatic interference.

AT8202 Adjustable In-Line Attenuator



The AT8202 provides -10dB, -20dB or -30dB attenuation, preventing overload of sensitive input stages. For use with balanced Lo-Z microphones, it is compatible for use in phantom power applications. Durable steel case provides optimum shielding from hum. Input impedance is 1000 ohms (-10 dB), output impedance is 200 ohms (-10 dB). Has a XLR-F input and XLR-M output **43.50**

CP8201 Line Matching Transformer



The CP8201 matches low-impedance microphones to high-impedance electronic inputs. Reduces noise pickup, maintains high frequency response and permits use of long mic cables. Input impedance is 250 ohms , output impedance is 50,000 ohms. Has a XLR-F input, 1/4″ plug out.....**19.95**

CP8305 Line Matching Transformer Cable

Same as the CP8201 except the CP8305 adds 16′ of cable. XLRF in, 1/4″ plug out.....**29.95**

MIC CLIPS

Plastic Tie Clip *(AT8411)*.....**9.50**

Plastic Double Tie Clip *(AT8412)*.....**14.50**

Wire Tie Clip *(AT8413)*.....**6.50**

Wire Tie Clip *(AT8417)*.....**9.50**

Metal Tie Clip *(AT8420)*.....**6.50**

Plastic Tie Clip *(AT8434)*.....**6.50**

Plastic Tie Clip *(AT8435)*.....**6.50**

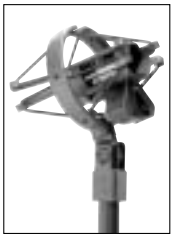
Clothing clip for cable *(AT8439)*.....**4.95**

AT8601/AT8614/AT8615/AT8615RS
Microphone Desk Stands

The AT8601 is compatible with any 5/8″-27 threaded stand clamps. The AT8614 is the same, plus it has a thru-hole in 5/8″-27 stud for cable passage. The AT8615 is a quick-mount plug-in cast-metal stand for any gooseneck mic with XLRM output. The AT8615 is the same plus it has a press-on/off mute switch for phantom-powered mics.

AT8601..**17.50** **AT8614**..**29.95** **AT8615**..**49.95** **AT8615RS**..**57.50**

AT8410a & AT8415 Shock Mounts



These shock mounts provide excellent mechanical isolation while offering unusual versatility and ease of use. Equally adapted to cylindrical and tapered mics, they attenuate noise, shock and vibration transmitted through stands, booms and fixed mic mounts. The AT8410a permits easy mic insertion and removal without detaching the mic cable; its spring-loaded clamp holds even heavy mics securely. The AT8415's light weight and low profile make it an excellent choice for "fishpole" booms and for video use.



The AT8410a is compatible with 5/8″-27 threaded stands and includes a 180° swivel for easy, positive mic adjustment. Isolation is accomplished with elastic shock cord. The AT8415 is compatible with 3/8″-16 and 5/8″-27 threaded stands, and features a constant-tension clutch and 360° rotation. Isolation is accomplished with four chloroprene bands.

AT8410a.....**49.95** **AT8415**.....**49.95**

AT8416 Shock Mount



Designed for UniPoint and similar gooseneck mics mounted on lecterns, pulpits, conference tables and similar surface, the AT8416 effectively isolates them from impact vibration and shock handling. A slot is provided in the mounting flange to allow a small-diameter cable to pass down the mounting hole. A cable channel secures the cable in the flush mount assembly and helps to further deaden the transmission of vibration.....**49.95**

AT8418 UniMount Microphone Instrument Mount



Rubber grip spring clamp holds mic securely to instrument without marring finish. Flexible boom permits aiming the microphone in any direction. Cable strain reliefs reduce cable-transmitted noise and cable damage. Foam windscreen entirely surrounds mic for best performance and protection **37.50**

AT8646AM/AT8646QM/AT8647QM/S
Shock Mount Plates



Designed for UniPoint and similar gooseneck mics mounted on lecterns, pulpits, conference tables and similar surface, these shock-mount plates effectively isolate mounted microphones from impact vibration and mechanical noise normally transmitted from the surface. All three models incorporate a sturdy yet flexible suspension panel which provides the mechanical isolation. The AT8646AM has a 5/8″-27 threaded adapter-mount, while the AT8646QM has an XLR-F connector. The AT8647QM/S accepts plug-in mics and features a phantom-powered mute switch with a lighted "on" indicator.

AT8646AM.....**27.95** **AT8646QM**.....**36.95** **AT8647QM/S**.....**66.50**

AUDIO-TECHNICA

UniTools

Phantom-powered in-line microphone accessories to overcome commonly-encountered sound installation challenges quickly, easily and effectively. Available in two function-specific models, they are intended primarily for use with UniPoint and similar condenser microphones. Both have XLR input/output connectors, and require balanced cables and 24-48v phantom power for operation.

- Rugged steel housings
- Phantom power passes thru to the mic
- Can be installed anywhere in-line, either at the mixer input or near the mic

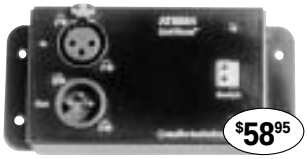
AT8681 UniMix

2-to-1 Mic Combiner with Balance Control

Combines the output from two microphones into one channel. The balance control attenuates each microphone by 6 dB (typical) at midpoint.



$46.95



$58.95

AT8684 UniMute

Mic Attenuator with User-Definable Switch Function

Versatile attenuator permits easy in-line switch installation and flexible application. Type of external single-pole switch the user supplies determines muting function; closed contact causes muting. Use for on/off, push-to-talk, push-to-mute, etc.

SPECIFICATIONS

Frequency Response .....20-20,000 Hz
Noise .....(A-weighted) -115 dBV typical
Weight .....8.8 oz
Size .....5.12 x 2.36 x 1.73″ (WxDxH)

UniMix

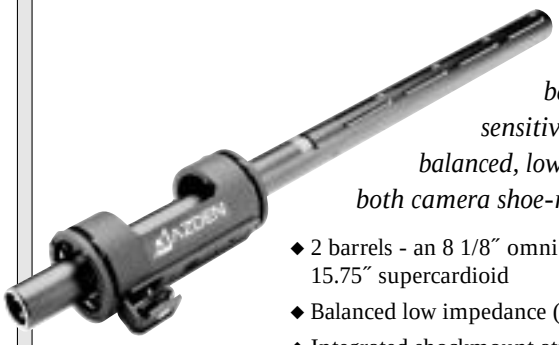
Input Impedance .....10,000 Ω
Output Impedance .....250 Ω
Max. Input Level* .....+15 dBV
Max Attenuation .....33 dB at 1 kHz

UniMute

Input Impedance .....5,300 Ω
Output Impedance .....500 Ω
Max. Input Level* .....+17 dBV
Max Attenuation .....57 dB at 1 kHz

AZDEN

SGM-2X *Professional Shotgun Microphone*

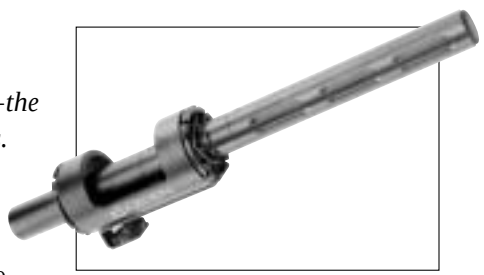


Designed for pro-sound and broadcast applications, the SGM-2X comes with two barrels, an omni (8.11″ long) or, by adding the extension barrel, a supercardioid (15.75″ long). Either configuration provides a sensitive (-50dB), low-noise, wide bandwidth signal (40Hz - 20kHz) through its balanced, low-impedance XLR output. Comes complete with an integrated shock-mount, both camera shoe-mount and mic stand-mount, and two foam windscreens.

- 2 barrels - an 8 1/8″ omni and extension barrel – make it a 15.75″ supercardioid
- Balanced low impedance (680 ohm) XLR output
- Integrated shockmount attaches to camera shoe or mic stand
- Plugs into the 41XT transmitter to make a wireless shotgun mic
- Switchable low-cut filter
- Runs 1000 hours on a single AAA battery

SGM-1X • SGM-X

Following up on the success of the SGM-2X, Azden offers two shorter versions—the SGM-1X and SGM-X—specifically designed for today's smaller DV camcorders.



Ideal for the Canon GL-2, Sony DCR-VX2000 and other similar type camcorders, the 8.5″ SGM-X is a complete system, including windscreen, shoe-mount holder and mini cable. Has a mini output.

At 11.75″ the SGM-1X is designed for camcorders like the Canon XL-1s and Sony PD-150. It features an XLR output and comes supplied with Azden's unique shockmount holder with integrated shoe mount.

M1245 / M1290 'MICROS'

World's Smallest Condenser Mics with Integrated Preamp

The M1245 and M1290 'Micros' are the world's small condenser microphones with an integrated preamp and detachable cable. The M1245 is less than 2″ long and weighs a mere .6 oz while the M1290 is 3.5″ in length and weighs in at 1 oz. Characterized with precision machined brass body and black E-coat finish, each is laser etched with its model and serial number. Individual models include a variety of polar patterns to choose from including cardioid, hypercardioid, omni-directional and shotgun capsules.



Other miniature mics require external power supplies that are bulky, cumbersome, and require extra cabling. Not only do the Micros eliminate the additional power supply, but the preamp they use is very special. The circuitry is the same topology used in Audix's SCX series. The Micros are studio quality mics with very low noise, wide dynamic range, and high quality performance. And with a sound is that is absolutely astounding, just think of the possibilities for live sound, location recording, and installations.

The M1245 has a smooth uniform response frequency range is 80Hz-20kHz, while the M1290 has a wider response of 40Hz-20kHz. Phantom powered, the mics are low impedance, fully balanced, and allow cable runs of up to 150′'. The mics are packaged in a classic rosewood case, and include 12′ cable, mic stand adapter, and snap on foam windscreen. Optional accessories include cables, insulated shock mount clip, and a clip for for overhead hanging applications. They are also compatible with the D-Vice and D-Clamp for a wide variety of instrument miking.



Rosewood Case

SMT-Micro

M1245 shown on a Hanger-Micro

Supplied Accessories

CBL-M12: 12′ high quality cable with Mini XLRF-XLRM connectors
MC-Micro: Mic stand adapter
Rosewood Case

Optional Accessories

CBL-M25: 25′ high quality cable with Mini XLRF-XLRM
CBL-M50: 50′ high quality cable with Mini XLRF-XLRM
Hanger-Micro: Low profile hanger clip allows precise positioning
SMT-Micro: Stand adapter with rubber insulated shockmount

	M1245	M1245-HC	M1290	M1290-HC	M1290-O	M1290-S
Pattern	Cardioid	Hypercardioid	Cardioid	Hypercardioid	Omni	Supercardioid
Frequency Response (Hz)	80 to 20k	80 to 20k	40 to 20k	40 to 20k	20 to 20k	40 to 20k
mV/Pascal Sensitivity	14 mV/Pa	14 mV/Pa	14 mV/Pa	14 mV/Pa	14 mV/Pa	14 mV/Pa
Impedance	250Ω	250Ω	250Ω	250Ω	250Ω	250Ω
Rejection Off Axis	>20dB	>25dB	>20dB	>25dB	—	>20dB
Maximum SPL	140 dB	140 dB	140 dB	140 dB	140 dB	140 dB

AUDIX

OM-Series

Dynamic Hypercardioid Vocal Microphones

OM-Series microphones are designed for live vocal performances. Combining high performance, high levels of gain and very low feedback, they challenge the best of much more expensive brands. When it comes to handling high sound pressure levels, off-axis rejection and purity of sound, these mics will simply blow you away! They are also designed for artists and engineers who refuse to compromise and, as such are manufactured to exacting standards and built to last a lifetime.
"Application specific", you can choose a microphone based on the type of music you play, vocal style, size of the group, stage volume, and type of PA equipment. Audix mics will give you the freedom to optimize your performance each and every time. So whether you 're on a world tour, playing a small club, or recording your next demo, there is an Audix mic for you.



OM-1 Vocal Mic

First produced in 1985 and re-introduced for the new millennium, the OM-1 provides original cutting edge sound with a stylish two-stage, machined brass body. Updated composite capsule housing and mounting system allows very high output with extremely low handling noise.

- Frequency response of 50 Hz-15 kHz frequency response, 140 dB SPL handling, electronically cut blast filter and a durable unique two-piece brass body
- Supplied with a cordura carrying case and nylon mic clip



OM-2 Vocal/Instrument Mic

Designed to provide high quality sound for a wide variety of vocal styles, the highly affordable OM-2 is an all purpose mic that sounds great on just about everything. It incorporates hallmark Audix features and characteristics including rich, natural vocal sound with exceptional clarity and detail; superior off-axis rejection; and the ability to perform at high levels without feedback or distortion. A worthwhile addition to any live sound or home studio application.

- Frequency response of 50Hz-16kHz, 140 db SPL handling, durable zinc alloy construction, electronically cut blast filter with black E-coat finish
- Comes with cordura carrying case and nylon mic clip

- Handheld dynamic hypercardioid mics
- Handle at least 140 dB SPL
- Proprietary anti-feedback design
- Low-reflective black durable e-coat finish
- Vocals can be heard above the stage mix
- Dent and scratch-resistant spring-steel grill ball —10 times the tensile strength of typical grille balls
- Gold-plated contacts
- Superior transient response
- Individually tested and hand assembled
- Each includes pouch and clip

OM-3 Vocal/Instrument Mic



An excellent choice for any size stage, the OM3 produces highly accurate, natural sound with minimal bass proximity. It provides over 25dB of feedback rejection off-axis and handles up to 144dB SPL without distortion. A dent-resistant spring-steel grill adds to the OM3's road-worthiness. Also available with an on/off switch (OM3-S).

- Uniform frequency response of 50-18kHz
- Clear, natural sound for mid to large PA. Compared to the OM-2, the OM3 has slightly less bass proximity and will provide higher gain before feedback in the monitors
- Overall natural sound quality and consistency, make the OM3 the mic of choice for many sound companies and high profile vocalists
- Transformerless design, low impedance, and balanced output allow interference-free performance even with long cable runs

OM-5 PA/Vocal Mic



A truly remarkable performance vocal microphone, the OM-5 has appeared in many high level concert performances including Alanis Morissette, Foo Fighters and Bonnie Raitt as well as becoming a standard on many TV shows and music videos.

- The OM-5 is attenuated in the lower mid-bass with a slight presence peak in the upper mid-range. It sounds excellent on full range PA systems and is characterized as being natural, articulate, and able to cut through the stage mix without sounding harsh or shrill
- An exceptionally tight pattern mic, the OM-5 allows for very high level monitor gain without feedback
- Also successful at reducing the sound of guitar and drums coming through the vocal mix, a real plus for the performer and engineer

OM-Series, Fusion Series, CD-Series

OM-6

Full, rich sound. Ideal for in-ear monitor systems. The OM-6 has also met with critical acclaim in the world of pro sound. Unlike the OM-5 which has a natural attenuation of bass frequencies (below 100 Hz), the OM-6 is a very full range microphone capable of reproducing tones down to 63Hz. Characterized as being extremely pure and accurate, the OM-6 has a more "studio-like" sound and appeals to vocalists who want minimal coloration.



OM-7

There is no competition when it comes to gain before feedback. Having become a standard on the touring rock concert scene, the OM-7 is considered to provide the highest gain before feedback of any microphone on the market, especially in the monitor system. The unconventional low output gain stage of the OM-7 acts as a natural pad and allows for incredible sound pressure levels to be achieved during live performances, without sacrificing tonal quality.

	OM1	OM2	OM3	OM5	OM6	OM7
Microphone Element Type	Dynamic	Dynamic	Dynamic	Dynamic	Dynamic	Dynamic
Frequency Response (Hz)	50 to 18k	50 to 16k	50 to 18k	48 to 19k	40 to 18k	45 to 19k
Polar Pattern	♤	♤	♤	♤	♤	♤
Output Impedance	200Ω	250Ω	250Ω	200Ω	200Ω	150Ω
Sensitivity	2.45 mV/Pascal	2.3 mV/Pascal	2.4 mV/Pascal	2.4mV/Pascal	2.3 mV/Pascal	0.9 mV/Pascal
Maximum SPL	140 dB	140 dB	144 dB	144 dB	144 dB	144 dB
Off-axis Rejection	>30 dB	>25 dB	>25 dB	>30 dB	>25 dB	>30 dB

Fusion Series

Affordable dynamic microphones, the Fusion series offer exceptional performance at a break-through price. They are designed for a wide variety of applications including music, speech, corporate AV presentations, karaoke, worship and eduction.



- An excellent all purpose mic for both live sound reinforcement and recording. The F50's cardioid pattern helps minimize feedback and unwanted off-axis noise from other mics or instruments on stage.
- The F50 is a dynamic mic designed to handle a wide variety of vocal and instrument applications for both live sound and recording. Characterized with a wide cardioid pickup pattern and a 50 Hz to 16 kHz frequency response, the F50 is designed with a slight bass proximity and mid-range presence boost. This results in a warm natural response for nearly every type of voice.
- The F50 is engineered to handle very high levels of gain without distortion and provides over 20 dB of off-axis feedback rejection.
- Solidly built with a cast zinc alloy body and steel mesh grille ball with an internal pop screen. The F50-s includes an on/off switch.

F50-BP Band Pack includes 3 F-50 and 1 F50-S in an aluminum case

CD-Series

The CD series mics are cost effective solutions designed for a wide variety of applications including music, karaoke, speech, presentations, worship and education. Rugged and road worthy, they each feature a cardioid polar pattern to minimize feedback, on/off switch, and an internal pop filter to control breath and wind noise. Each mic comes with 20´ mic cable, stand adapter and carry case.

CD-11: Low impedance dynamic mic characterized with a very clear midrange vocal presence to suit portable sound systems. Includes XLR-XLR mic cable.

CD-11H: A high impedance version of the CD-11 allowing it to be used with equipment such as guitar amps, recording tape decks and karaoke systems with high impedance inputs. Includes a XLR – 1/4″ mic cable.

CD-21: A step up to the next level of vocal performance, the stylish and compact CD-21 offers a wider frequency response and a rich, warm sound. Includes XLR-XLR mic cable.



CONDENSER MICS

VX-10



The VX-10 is a true condenser microphone designed to set new performance standards in live sound and broadcast applications—where sonic performance is everything.

- With a smooth uniform response over a frequency range of 40Hz -to -20 kHz, the VX-10 is highly sensitive to transient response and will reproduce vocals and speech with exceptional detail and realism.
- Cardioid pattern provides excellent gain before feedback
- Runs on 48-52v phantom power
- While it provides high sensitivity and studio quality sound reproduction, the mic can also handle the very high sound pressure levels (up to 140 dB) of instruments.
- The VX-10 handles close miking with minimal proximity effect and will not lose signal if the user is slightly off-axis from the front of the microphone.
- Even with its extended low end response, the VX-10 offers sonic clarity without "boominess" in the lower mids-usually difficult to attain with live vocals.

SCX-25

Studio Condenser Microphone



The SCX-25 is a compact, brass condenser microphone with an elegant design and a revolutionary patented capsule suspension system. Featuring a low-noise, high-output preamp and 1˝ gold sputtered diaphragm, this innovative mic delivers a pure, open-air sound that places it among the world's finest—regardless of size or cost. The patented suspension-mounting system isolates the capsule from the housing and electronics, which minimizes acoustic reflections and diffraction. This phenomenal feat of engineering distinguishes the SCX-25 from all other mics— in fact its the closest thing to hanging the capsule in mid-air. Expensive shock-mount clips are unnecessary and, with the mic's compact size, setup time is significantly reduced. Ideal for recording a wide variety of acoustic instruments, vocals, and piano, the SCX-25 also thrives as an ambient room and field recording mic.

- With a smooth uniform cardioid frequency response from 20Hz- 20kHz, the SCX-25 reproduces vocals and acoustic instruments with exceptional detail and realism.
- Employs miniaturized low noise components (preamp circuitry is housed in a brass microphone body 4˝ long). The result is the smallest footprint of any large diaphragm mic—allowing more convenience and freedom for mic placement.
- Very consistent when responding to incoming off-axis signals, exhibiting excellent phase coherence and minimal proximity effect.
- Operates on 48 - 52v phantom power

D-SERIES *The Ultimate Drum Mics*

With the success and acceptance of the D-Series microphones, Audix is now the recognized leader in percussion microphones. Compact and lightweight yet featuring full size diaphragms, the D-Series mics employ exclusive Sub-Impulse Technology which allow them to pick up transients and nuances which substantially contribute to the accuracy and richness of the sound. The low mass of the diaphragm, coupled with a sensitive internal air suspension, allows the outer wall of the diaphragm to react to sound waves, harmonics, and sub- harmonics more quickly than other microphones. This results in a more precise sampling of the original sound. Furthermore, their tight hypercardioid pickup pattern allows clear definition of sound without the feedback problem commonly associated with placing multiple mics in close proximity.



B&H PHOTO - VIDEO - PRO AUDIO

D-SERIES

- Hypercardioid Dynamic Instrument Mics
- 144dB SPL
- Proprietary anti-feedback design
- Durable scratch resistant spring steel grille
- Black hard coat finish
- Isolates each instruments' "sweet spot"
- Gold-plated contacts
- VLM (Very Low Mass) diaphragm
- Reduced phase shift
- Superior transient response
- Highest levels of gain before feedback
- Exceptional off-axis rejection
- Individually tested and hand assembled
- Built (in the U.S.A) to last a lifetime
- Pouch and clip included

D1

Snare Drum, Bongos, Flute, Hi-Hat, Clarinet, Cowbell

The D1 is characterized with natural articulation and attack in the upper mid- range and will complement instruments requiring crisp, clean reproduction. The D1 requires little or no EQ and provides great results when close miking techniques are employed.

D2

Rack Toms, Floor Toms, Congas

Designed to capture the warmth and punch of instruments with upper mid bass, the D2 is the perfect combination of attack and decay and can be successfully used with very little or no gating. With a wide dynamic range, the D2 can also be used on guitar cabs and saxophone.

D3

Timbales, Piccolo Snare, Trumpet, Electric Guitar

The D3 employs a unique low output gain stage which acts as a natural pad in order to compensate for instruments that would normally overload most mics. Characterized with a very flat and accurate frequency response, the D3 is perfect for extremely high SPL instruments with short blasts of percussive sound.



D4

Floor Toms, Kick Drum, Djembe, Bass Cabs, Piano, Saxophone

The D4 utilizes a new capsule design enabling it to capture instruments with extended frequencies below 100Hz. In fact, the D4 extends down to 38Hz! A remarkable feat for a microphone of this size and versatility. The D4 has not only great low end response, but is a full-range microphone making it an ideal choice for a wide variety of instruments. It is important to note that when using the D4 to mic a kick drum, it should be placed just slightly inside the hole on the front head. This will provide a warm, fat sound with plenty of punch.

D6

Kick Drum, Cajon

The D6 is designed for anyone looking for a huge, clean, undistorted kick drum sound. About twice the size of the normal D series microphone, it is still extremely easy to use and position. It utilizes the same low profile mounting clip as the D series mics, and it will work with the Dvice and D-flex clips, as well as a variety of other popular clips on the market.

The D6 has a frequency response of 30 Hz - 15 kHz and is characterized with a cardioid pick-up pattern. The D6 is designed to sound good in any position and it is not dependent on finding the "sweet spot" of the drum. The D6 capsule features the same legendary VLM technology that has made the D series percussion and instrument microphones very popular for today's live stages and recording studios. Includes a cordura pouch and nylon mic stand adapter.

	D1	D2	D3	D4	D6
Microphone Element Type	Dynamic	Dynamic	Dynamic	Dynamic	Dynamic
Frequency Response (Hz)	46 to 18k	44 to 18k	50 to 19k	38 to 19k	30 to 15k
Polar Pattern	Cardioid	Cardioid	Cardioid	Cardioid	Cardioid
Rated Impedance	250Ω	250Ω	150Ω	200Ω	200Ω
Open Circuit Sensitivity[1]	2.4 mV	2.5 mV	1 mV	2.1 mV	2.2 mV
Maximum SPL	144 dB	144 dB	150 dB	144 dB	144 dB
Off-axis Rejection	>30 dB	>30 dB	>30 dB	>30 dB	>20 dB

[1] *1 Pa + 94 dB SPL*

INSTRUMENT MICS

ADX-51

Overheads, Hi-Hat, Cymbals, Toys, Acoustic Instruments, Group Vocals



The ADX-51 is an affordable pre-polarized condenser microphone designed to handle a wide variety of live and studio applications. Characterized with a smooth uniform response over a frequency range of 40Hz - 20kHz, the ADX-51 is very versatile and can be used for a variety of instruments as well as overheads. The ADX-51 has the added value of a 10dB pad and a bass roll-off switch. Capable of 142dB sound pressure levels, the ADX-51 is ideal for hi-hat and applications with very high stage volumes.

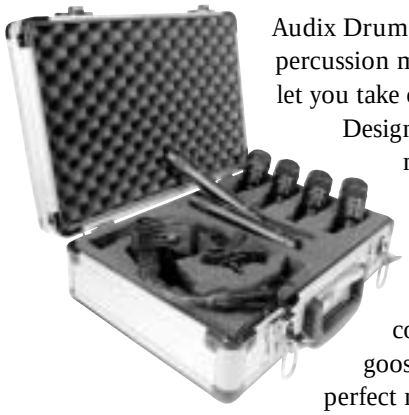
ADX-60

Kick Drum, Piano, Plays



The ADX-60 is a pre-polarized condenser boundary microphone capable of a very wide variety of applications.With a frequency range of 50Hz -18kHz,the ADX-60 provides a warm,f full-bodied sound not typical of microphones this size. Housed in a heavy-duty zinc die-cast base with a low reflective black finish, the ADX-60 operates on phantom power and is supplied with an in-line preamplifier and 25´ mic cable.A boundary microphone is unique in that it picks up sounds that are parallel to the surface. This allows the ADX-60 to be placed on surfaces such as floors, the inside of a piano lid, or the inside of a bass drum.

D-Series Drum Packs



Audix Drum Packs are a matched collection of drum and percussion mics and clamps in an aluminum road case that let you take control of your sound at substantial savings. Designed for both studio and live applications, Audix mics get you right to the heart of the mix with consistent, high-quality sound.

Each microphone consists of a VLM (very low mass) capsule housed in a precision-machined lightweight aluminum body. Their compact size, in conjunction with the D-Vice gooseneck clip, allows for quick and easy set-up and perfect mic placement. And there are four packages to choose from (the DP3 and DP Elite also include condenser mics). So whether you are a performing artist or an engineer, there is a package designed for you.

DP2

1x **D-1**
2x **D-2**
1x **D-4**
4x **D-Vice**
1x **Durable Road Case**

DP3

1x **D-1**
2x **D-2**
1x **D-4**
2x **ADX-51**
4x **D-Vice**
1x **Durable Road Case**

DP5

1x **D-1**
2x **D-2**
1x **D-4**
1x **D-6**
4x **D-Vice**
1x **Durable Road Case**

DP ELITE

1x **D-1** 2x **D-2** 1x **D-4** 1x **D-6**
2x **SCX1-c** (cardioid) 1x **SCX1-hc** (hypercardioid) 4x **D-Vice** 1x **Durable Road Case**

Fusion Series Drum Packs

Fusion series mics effectively capture the sound of any drum or percussion instrument. Built to withstand the rigors of live stage applications, they can also be successfully used in a studio recording environment. The two packages are available complete with a foam padded road case. The F-10 and F-12 mics come with integral mic clips, and the F-15 mics come with mic stand adapters.



Fusion 4

3x **F-10 snare/tom mics**
1x **F-12 kick/floor tom mic**
1x **Durable road case**

Fusion 6

3x **F-10 snare/tom mics**
1x **F-12 kick/floor tom mic**
2x **F-15 overhead mic plus 2 clips**
1x **Durable road case**

INSTRUMENT MICS

Micro-D



The Micro-D is a miniature electret condenser microphone that is very low profile and is exceptionally easy to use. The unique design works for a wide variety of applications including drums and percussion, guitar cabinets, cymbals, speech, vocals, and acoustic instruments. Available with three different types of specialized clips and two different types of capsules, the Micro-D becomes a modular system that can fulfill a multitude of creative miking needs.

The Micro-D comes equipped with a phantom power adapter, D-vice spring tension gooseneck clip, windscreen, carrying pouch, and thread adapter. The D-vice is ideally suited for mounting to the rims of most drums. The thread adapter allows the Micro-D to be used with normal mic and boom stands.

- Basic version has a hypercardioid pick-up pattern and is capable of handling sound pressure levels of 140 dB
- The mic is housed in a protective aluminum ring and isolated by means of a rubber shock mount system
- Smooth uniform response over a frequency range of 40 Hz - 20 kHz
- Optional accessories include the D-clamp tension clip for percussion instruments, and the D-flex clip for drums, percussion, guitar cabinets, goodie table, and cymbal stands

SCX1 Series



The SCX1 Series high quality transformerless condenser microphones have exceptional performance value in both studio and sound reinforcement applications. The SCX1-c has a cardioid pattern and is designed for overheads, acoustic instruments, choir, and room miking.

If more isolation is required, the SCX1-hc, a hypercardioid mic, will provide excellent focus and gain before feedback.

The SCX1-o is an omnidirectional mic, optimized for recording, room miking, and measurement.

A true condenser, requiring 48-52v phantom power, the SCX1 Series will provide audio detail beyond your expectations.

ADX-20i

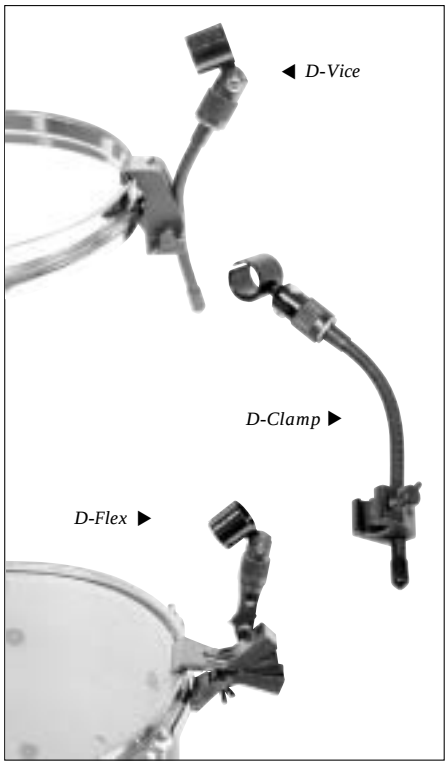


The ADX-20i (cardioid) is an electret condenser mic that is designed to handle a wide variety of clip-on applications. Sonically superior as well as aesthetically pleasing, the ADX20i is perfect for saxophone, trombone, trumpet and other brass instruments.

- Smooth frequency range of 40Hz -20kHz provides a rich, warm sound not typical of a microphone this size.
- Machined from solid brass with a low reflective black e-coat finish.
- Operates on phantom power (9-52v) with the preamp module provided (APS-910). It can also be used in battery mode with the APS-911 adapter.

ADX-20i Includes shock-mount adapter

MICROPHONE MOUNTS



D-Vice

A new solution for drum miking, D-Vice will eliminate drummer's concerns about microphone positioning. The D-Vice has a low profile and unique mounting system that allows precise positioning and attachment (using only one hand) to the rim of most drums and rimmed percussion. Precision molded out of composite material, the patent pending D-Vice can easily accommodate any 5/8˝ microphone clip. Works with the Micro-D and D-series mics.....**24.99**

D-Clamp

A tension clamp for the Micro-D and D-Series, the extra long metal gooseneck allows precise mic placement on congas, bongos and percussion lugs. Also interchangeable with D-Vice gooseneck for modular use **24.99**

D-Flex

A dual pivot clip with extra wide jaws, the D-Flex insures that your Micro-D or D-Series mic is positioned perfectly. It also minimizes vibration and unwanted resonances **13.95**

BEYERDYNAMIC

DYNAMIC VOCAL MICS

Founded in 1924 by Eugene Beyer, Beyerdynamic is the oldest audio company in the world that remains under its original ownership. Interestingly, Eugene was best friends with George Neumann, and the two legends actually started together in business. But they soon discovered that each had his own ideas with respect to transducer designs. So shaking hands, it was agreed that George Neumann would only produce condenser transducers, while Eugene Beyer would produce only dynamic capsules. This agreement remained in place until the mid-1970's, after they both passed away. Today of course, Beyerdynamic is known not only for the quality of its dynamic mics but condenser and ribbon mics, wireless products, and headsets.

TG-X 58

Hypercardioid Dynamic Vocal Mic

The TG-X 58 is designed for vocals as well as the miking of instruments. It can be used as a handheld or used on a stand. Hypercardioid pattern provides high gain before feedback. The mic barrel is finished with a soft lacquer coating which makes the microphone very comfortable to hold in the hand. The TG-X 58 also features an integrated pop shield which reduces "pop" noise associated with close vocal miking. The On/Off switch can be locked into position to avoid accidental switching off. Built to withstand serious abuse during live performances. Includes mic clamp and bag.....**67.50**

TG-X 60

Hypercardioid Dynamic Vocal Mic

The TG-X 60 is a very high quality dynamic microphone intended for use handheld or stand-mounted by lead vocalists, particularly in rock bands where maximum volume before feedback is required. It also has the ability to withstand high sound pressure levels, making it suitable for close miking of musical instruments or speaker cabinets. Dubbed the ultimate "Rock n' Roll" microphone, the TG-X60 is ruggedly constructed and has a shock absorbing rubber ring to withstand touring damage.....**279.95**

M 69 TG

Hypercardioid Dynamic Vocal Mic

In service for almost thirty years, the M 69 TG has been used in just about every application possible. A classic all-around mic with a superb response characteristic, the M 69 TG makes the human voice sound real, be it spoken or sung, chanted or shouted. The unique shape and sound of the M 69 TG coupled with high SPL capability, makes it a natural choice for the recording of guitars, brass and woodwind as well. Ideal for live applications, it has an excellent hypercardioid pickup pattern that guarantees high gain before feedback, while a reinforced basket allows it to withstand rough handling. Also well suited for sound contractors for use in churches or as an announcers microphones.....**189.95**



	TG-X 58	TG-X 60	M 69
Microphone Element Type	Dynamic	Dynamic	Dynamic
Operating Principal	Pressure Gradient	144 dB	144 dB
Frequency Response (Hz)	50 to 15k	44 to 18k	50 to 19k
Polar Pattern	◌	◌	◌
Nominal Impedance	600Ω	250Ω	150Ω
Open Circuit Sensitivity at 1 kHz	2 mV/Pa ± 1.5db	2.5 mV	1 mVV

OPUS SERIES

High Performance Instrument and Vocal Mics

The Opus range of microphones is a cohesive family of products optimized for music performance. Combining cutting-edge microphone technology, 21st century materials and over 75 years of transducer design and development, beyerdynamic has created a complementary range of microphones that capture the dynamism of music performance, new music styles and spontaneous creativity— on stage or in the studio. The Opus series includes a full range of popular microphone types and is designed to take the mystery and luck out of choosing the right type. All microphones are made in Germany from the highest quality materials and tested to the most rigorous standards.

Opus 39

High Performance Dynamic Mic

A high performance, general purpose dynamic mic, the ruggedly constructed Opus 39 is ideal for live-performance vocals, close musical instrument pickup and club PA. It offers a flat, wide range frequency response for uncolored sound reproduction, while its super-cardioid pattern provides excellent isolation from unwanted sound. Mic handle is soft- lacquered for comfort and grip.



Opus 59

High Performance Dynamic Mic

A high performance dynamic microphone for professional vocal use in live performance, the Opus 59 features a high output Neodymium magnet and extended high-frequency response tailored for vocals.
Supercardioid pattern provides high gain before feedback and excellent isolation from unwanted sources. Soft-lacquered for comfort and grip. Rugged all metal body and legendary Beyerdynamic quality guarantee years of service.



	OPUS 39	OPUS 59	OPUS 69	OPUS 81
Microphone Element Type	Dynamic	Dynamic	Dynamic	Condenser
Operating Principal	Pressure gradient	Pressure gradient	Pressure gradient	Pressure gradient
Frequency Response (Hz)	50 to 16k	45 to 16k	35 to 16k	50 to 18k
Polar Pattern				
Maximum SPL at 1kHz	—	—	—	138 dB
Signal-to-Noise Ratio rel. to 1 Pa	—	—	—	60 dB
Nominal Impedance	600Ω	600Ω	290Ω	190Ω
Open Circuit Sensitivity at 1 kHz	2.4 mV/Pa	2.6 mV/Pa	2.5mV/Pa ± 1.5 dB	3.2mV/Pa



Opus 69

High Performance Dynamic Mic

A singer's dream, the Opus 69 delivers spectacular vocal reproduction. A wide, uncolored frequency response and warm proximity effect, gives the performer the ability to create a sonically unique style.

Remarkable response and tightly controlled hypercardioid pattern enhances usable gain before feedback while rejecting off-axis sound sources, producing a crisp, natural sound.



Opus 81

High Performance Condenser Mic

Advanced technology and extraordinary engineering has produced this state of the art condenser condenser microphone. The Opus 81 can be used on the stage or in the studio. The condenser element withstands sound pressure up to 138 dB SPL without a loss of performance.

OPUS SERIES

Opus 53

High-Quality Condenser Instrument Mic

The Opus 53 is a high-quality condenser microphone at a reasonable price. It consist of a light-weight, but rugged, plastic casing strengthened with carbon fibers. Its small and short design allow flexible and unobtrusive positioning. The frequency response has been optimized for universal miking of instruments. In particular it is suitable for miking drums such as snare or hi-hat. But it is also a great microphone for miking brass and wood winds instruments.



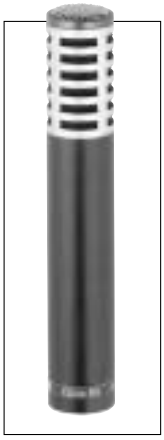
Opus 51

Acoustical Boundary Condenser Mic

Designed for instrument sound reinforcement. Using its boundary effect, the Opus 51 captures direct and reflected sounds from an instrument in perfect balance and at an equal level, taking the guess work out of mic placement. Ideal for capturing the full sound within an instrument, such as the piano. The Opus 51 captures all the of the piano's subtle overtones and nuances, producing a natural sound and conveying the emotion of the performance.



	OPUS 53	OPUS 51	OPUS 83	OPUS 65	OPUS 64
Microphone Element Type	Condenser	Condenser	Condenser	Dynamic	Dynamic
Operating Principal	Pressure gradient	Pressure	Pressure gradient	Pressure gradient	Pressure gradient
Frequency Response	30 to 20k	30 to 20k	40 to 20k	15 to 18k	40 to 12k
Polar Pattern	◡	◯	◡	◌	◌
Maximum SPL at 1kHz	136 dB (@ 48v)	131 dB	138 dB	—	
Signal-to-Noise Ratio rel. to 1 Pa	64 dB	26 dB	60 dB	—	
Nominal Impedance	200Ω	250Ω	190Ω	280Ω	200Ω
Open Circuit Sensitivity at 1 kHz	2.4 mV/Pa	7 mV/Pa	3.2mV/Pa	2mV/Pa	1.5mV/Pa



Opus 83

High Performance Condenser Instrument Mic

The Opus 83 is a condenser mic designed to capture drums and other acoustic instruments with precision and feeling. Built from solid brass, the casing of the mic protects the sensitive element from the rigors of the road. The Opus 83's natural sound and ability to reproduce depth of field is ideal for drum overheads, cymbals, percussion overheads, and acoustic instruments.



Opus 65

Kick Drum Microphone

Only a special large diaphragm dynamic mic can reproduce the low frequency energy created by kick drums, congas and timpani. The Opus 65 combines a 1.5″ diameter diaphragm with a powerful neodymium magnet to effortlessly capture and reproduce low frequency signals and harmonics — requiring little or no EQ. Though primarily designed for close miking of the kick drum, it is also well-suited for floor toms, bass drum etc. Ability to withstand extremely high sound pressure levels allows it to be used for close miking of musical instruments or speaker cabinets.

Opus 64

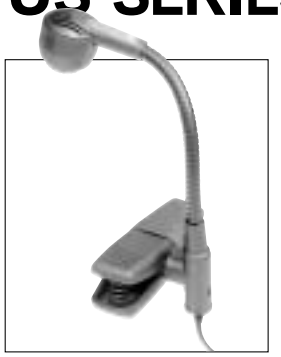
Headworn Microphone

Designed for "hands free" applications that require a high quality and lightweight microphone, the Opus 64 is ideal for entertainers, dancers, drummers and keyboard players etc. It interfaces with wireless systems to provide a low profile neckworn microphone for aerobics instructors, sales and product presentations. Its hypercardioid polar pattern allows a high gain before feedback. The Opus 64 features a rugged yet adjustable headband that can be shaped for personal comfort with a secure fit. The microphone can be worn on the left or right side of the headband. To interface with the TS 100 beltpack transmitter the Opus 64 has been equipped with a mini jack plug (3.5 mm).



OPUS SERIES

Opus 62 Hypercardioid Dynamic Wind Instrument Microphone



The Opus 62 is a high performance, low profile clip-on dynamic mic designed for miking wind instruments such as trumpets, trombones and saxophones. The capsule is attached to a 4″ gooseneck and is externally shock isolated to prevent excessive transient response. Neodymium element provides increased output, while a tight polar pattern enhances off-axis noise rejection. The flexible gooseneck ensures optimal positioning.

- For miking trumpets, trombones, and saxophones
- Clip ensures solid and easy mounting to the instrument
- Hypercardioid pattern ensures high gain before feedback
- Flexible 4″ gooseneck for optimum positioning
- Small and rugged construction
- Supplied with bag
- Frequency range: 40 Hz to 12 kHz



Opus 67 Hypercardioid Dynamic Drum & Percussion Microphone

The Opus 67 is a dynamic microphone designed for close miking of drums such as snare and toms and percussion such as congas and bongos. The mic's hypercardioid polar pattern achieves very high gain before feedback while the flexible gooseneck ensures optimal positioning to suit both musician and sound engineer.

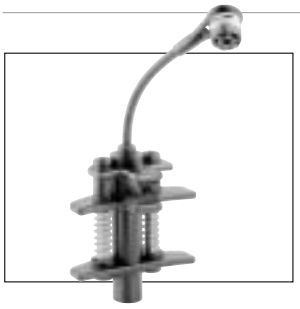
- For miking toms, snare, congas and bongos
- Very high gain before feedback
- High SPL capability
- Flexible gooseneck for optimal positioning
- Small, ruggedly constructed for on-tour applications
- Frequency response: 40 Hz to 12 kHz

Opus 66 Dynamic Supercardioid Drum & Percussion Microphone



The Opus 66 is a small and ruggedly constructed percussion microphone perfect for miking snares, hi-hats and rack- toms. It rejects mechanical and handling noise making it ideal for mounting on drum hardware. The mic mount ensures optimal positioning to suit both musician and sound engineer. Also has a 3/8″ (5/8″) thread for mounting on a conventional mic stand if required.

- For miking snare, hi-hat and rack-toms
- Extra stiff Macrolon diaphragm ensures exact reproduction or percussion and drums
- Excellent transient behavior
- Extremely high SPL (140 dB)
- Supplied with special mic mount and bag
- Frequency response: 40 Hz to 12 kHz



Opus 87 Cardioid Condenser Drum & Percussion Microphone

The Opus 87 is a superb electret back condenser mic for miking drums. It features a patented microphone clamp with integrated pre-amp that is an absolute innovation. The microphone capsule is mounted on a short gooseneck allowing optimum positioning. Ruggedly constructed the mic can withstand the worst abuse. Includes a case.

- Ideal for snare and toms
- Cardioid pattern with high gain before feedback
- Innovative mic clamp with integrated preamp
- Runs on 12-48v phantom power
- Maximum SPL of 136 dB
- Frequency response: 30 Hz to 20 kHz

Opus 82 Cardioid Condenser Wind Instrument Microphone



Ruggedly constructed, the Opus 82 is an electret condenser microphone for miking brass instruments. Its clip ensures a solid mounting to the instrument, while the flexible 4″ gooseneck allows optimum positioning. Unwieldy mic stands are no longer needed. Cardioid polar pattern ensures the Opus 82 achieves high-gain-before feedback.

- For miking trumpets, trombones, and saxophones
- Cardioid pattern ensures high gain before feedback
- High SPL capability (135 dB)
- Flexible 4″ gooseneck for optimum positioning
- Runs on 12-48v phantom power
- Supplied with CVU 82 fixed preamp and bag
- Frequency range: 30 Hz to 20 kHz

STUDIO MICROPHONES

MCD-100/MCD-101

Cardioid and Omnidirectional Digital Microphones



Equipped with a 24-bit analog-to-digital converter placed immediately behind their capsule, the MCD-100 and MCD-101 offer the optimum conversion of an analog signal into the digital domain. As a result they are able to provide a flawless reproduction of the mic capsule's dynamic range and frequency response. To accept 6-10v digital phantom powering and to facilitate connection to a digital console, a separate power supply (MPD-200) is available as an option. For high SPL applications up to 140 or 150dB a remote attenuator (-10/-20 dB) controlled from the MPD-200 is provided. And since they are digital, they can allow very long cable runs (up to 1000´) without signal degradation. Digital output is AES/EBU.

MCD100 Digital Cardioid Condenser Mic with MKV-11 mic clamp

MCD100 Set (includes MCD100, MPD200 Power Supply, EA37 Elastic Suspension and ZMK-6 Case)

MCD100 Stereo Set (includes 2 MCD100's, MPD200 Power Supply, 2 EA37 Elastic Suspensions and ZMK-6 Case)

MCD101 Digital Omnidirectional Condenser Mic with MKV-11 mic clamp

	MCD-100	MCD-101	MC-740	MC-834
Operating Principal	Pressure gradient	Pressure gradient	—	—
Frequency Response	20 to 20k	20 to 20k	40 to 20k	20 to 20k
Polar Pattern	◠	◯	◠/◠(wide)/◠/◯/8	◠
Maximum SPL at 1kHz	130 dB	130 dB	144 dB	—
Dynamic Range	113 dB	113 dB	—	—
Signal-to-Noise Ratio rel. to 1 Pa	—	—	—	—
Nominal Impedance	—	—	150Ω	180Ω
Open Circuit Sensitivity at 1 kHz	-34.5 mV/Pa	-34.5 mV/Pa	10mV/Pa	20mV/Pa



MC-740

5-Pattern Condenser Mic

With five polar patterns (omni, cardioid, wide cardioid, hypercardioid, figure-8), the MC-740 can be adapted to a variety of recording situations in the studio—while capturing every sound detail with absolute clarity and purity. The MC-740's double diaphragm capsule is characterized by a frequency response that remains constant in all patterns. Switchable 10 dB pad allows close miking of instruments up to 144 dB SPL. Also has a two-position low-frequency roll-off (80 and 160 Hz). Use for choir recordings, for vocals, wood and brass wind instruments, piano and string instruments, even as an overhead mic for percussion instruments. Includes MKV-11 mic clamp, EA-37 external suspension mount and case.



MC-834

Condenser Mic

A true condenser large diaphragm cardioid mic, the MC-834 has a wide range, natural frequency response that makes it ideal for a host of applications. Equipped with a large diameter gold vaporized membrane element, it features exceptional S/N ratio and high SPL capability (up to 150dB with the switchable -10/-20 dB pad). Cardioid pattern allows high gain before feedback. To adjust for proximity effect or extraneous low frequency sounds, the mic offers a three position low-frequency roll-off: LIN (flat, with no roll-off) and 6 dB per octave below 80 or 160 Hz. Use for lead and back up vocals, voice-overs, miking of pianos, strings and brass instruments, and home recording. Includes EA 37 ext. suspension mount.

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STUDIO MICROPHONES

M 130
Bi-Directional Ribbon Mic

The M130's unique double ribbon element has superb transient response, creating a highly detailed sound with unequalled accuracy and transparency. The bi-directional figure-8 pattern effectively suppresses unwanted interference from the sides. The M130's balanced, uncolored sound handles a wide variety of applications. In conjunction with the hypercardioid M 160, the M 130 is ideal for creating an authentic stereo image through the use of the M-S (Mid-Side) technique. Use for digital recordings, ambient miking, piano, harpsichord, harp and strings.



MCE 91
Condenser Vocal Microphone

A handheld cardioid condenser microphone, the MCE 91 delivers high output with very low self-noise making it ideal for recording in the studio or at home. A faithful purveyor of the human voice, the MCE 91 has a linear frequency response curve that has been optimized for clear and neutral voice reproduction. It also has a sophisticated elastic transducer suspension system that ignores heavy-handed vocalists, and can reproduce high pressure sound levels of up to 139 dB to handle even the most passionate arias. Includes MKV-8 mic clamp and padded bag.



M 160
Hypercardioid Ribbon Mic

The M 160 is a dynamic ribbon microphone with an extremely tight polar pattern and extended frequency response that is ideal for stringed instruments. The M 160's hypercardioid characteristic with a noise cancellation of 25 dB at 110° effectively suppresses lateral sound reflections and external noise. Recommended for picking up the sound of stringed instruments such as violins, violas and cellos as well as for zithers and dulcimers. The mic's black chromium plating and compact size makes it very popular in TV and film studios. Use with the M 130 to create an authentic stereo image via the M-S (Mid-Side) technique.



MCE 90
Condenser Vocal Mic

A project studio multi-purpose microphone, the MCE 90 features a linear frequency response, switchable controls to vary functionality, and is constructed in a rugged die-cast housing. Easily handles high sound pressure levels up to 139 dB without pre-attenuation and up to 154 dB with the pad. If necessary, low frequency noise can be attenuated with switchable LO-CUT filter.



	M 130	M 160	MCE 91	MCE 90
Microphone Element Type	Dual dynamic ribbon	Dual dynamic ribbon	Electret Condenser	Condenser
Operating Principal	Pressure Gradient	Pressure Gradient	Pressure Gradient	Pressure Gradient
Frequency Response	40 to 18k	40 to 18k	30 to 20k	30 to 20k
Polar Pattern	8	♡	◠	◠
Maximum SPL at 1kHz	—	—	139 dB	139 dB
Dynamic Range	—	—	—	—
Signal-to-Noise Ratio rel. to 1 Pa	—	—	68 dB	70 dB
Nominal Impedance	200Ω	200Ω	190Ω	190Ω
Open Circuit Sensitivity at 1 kHz	1.0 mV/Pa	1.0 mV/Pa	7mV/Pa	10mV/Pa
Price	$529.95	$529.95	$469.50	$469.50

STUDIO MICS, *continued*

MC 833
Stereo Condenser Mic



The MC 833 is a general purpose stereo condenser microphone that offers true MS, XY, and mono recording without an additional matrix. The mic is equipped with three true condenser systems. The polar pattern of each cartridge is cardioid and almost independent of the frequency.

The middle system supplies the mono signal or the M-signal with MS stereophony. The system located to the left and right of the middle system are switched together to form a figure-8 polar pattern and supply the S-signal.

At XY sterephony only the two side system are used. They can be mechanically adjusted. Therefore, at XY recording the recording angle can continually varied over a large range. Includes EA-833 elastic suspension, MKV-11 mic clamp and a case.

STUDIO / INTERVIEW MIC

MCE 82
Stereo Mic for X-Y Recording



The MCE 82 is a stereo microphone for X-Y recording. It is used for stereo recording in ENG/EFP applications, reporting, film, video and interviewing. The MCE 82 has two condenser capsules with cardioid polar patterns, which are mounted at an angle of ±45° of the microphone axis. This allows a wide recording range and an accurate reproduction of the sound source.

Has a flat linear frequency response for neutral miking with the highest sound quality.

Due to its compact and rugged metal housing, the MCE 82 is ideal for mobile applications. The elastic suspension of the capsule suppresses handling and cable noise. A switchable bass filter eliminates wind noise and rumble. Runs on 12 to 48v phantom power or a 1.5-volt AA battery. Also has a LED battery indicator. Includes MVK-82 mic cable.

INTERVIEW MICROPHONES



M58 Omnidirectional Dynamic Microphone

The M 58 is designed specifically for electric news gathering (ENG) and electronic field production (EFP) applications. Its sophisticated internal shockmount dramatically reduces handling noise, while the mic's frequency response is tailored to provide broadcasters with a very accurate and intelligible sound. The M 58's weight-balanced design provides journalists with a high degree of comfort during lengthy interviews. Its rugged construction enables the mic to withstand physical and environmental punishment typically encountered during field production operations. Slim profile and non-reflective Nextel finish result in a low profile on camera. Includes WS-58 windscreen.

M59 Hypercardioid Dynamic Broadcast Mic

The M 59 is a rugged, vocal quality dynamic mic designed to withstand the critical demands of in-studio or on-location ENG/EFP speech and interview work. In addition, it has a low mass, large diaphragm Macrolon element and hypercardioid pattern to provide very fast transient response, high output and high sensitivity, with startling accurate reproduction of the sound source while also providing excellent off-axis rejection.

The M 59 uses a sophisticated internal shock-mount isolation system to reduce stand, boom, or handheld noise and vibration. Durable construction and non-glare Nextel finish provide consistent protection against the rigors of ENG/EFP work. It also incorporates an internal multi-stage blast filter to reduce undesirable wind or pop noise. An optional windscreen is also available. With high gain before feedback, it is ideal for lecterns or miking of wind instruments. Includes MKV-8 mic clamp.

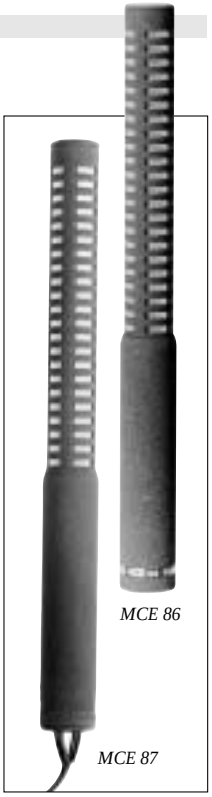
MCE58 Omnidirectional Condenser Microphone

The MCE 58 condenser microphone has been specially developed for interviewing and broadcasting. Its unbalanced output supplies a very high level which allows direct connection to mobile DAT and MiniDisc machines. The elasticated suspension of the transducer element eliminates unwanted handling noise and built-in active filters suppress unwanted low frequency noise such as traffic rumble. The MCE 58 has a frequency response curve which has been optimized for excellent intelligibility of speech. The non-glare Nextel surface is scratch-resistant and will withstand the rough use often associated with news gathering events. Includes AA battery and MKV-8 mic clamp.

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SHOTGUN MICROPHONES

MCE 86/86S • MCE 87/87VS
Short Shotgun Mics



The MCE 86 and MCE 87 are short shotgun microphones designed to meet the rigors of studio and on-location assignments in broadcasting. They can be used with a pistol grip, on a fishpole boom or on cameras and camcorders. Their condenser element and hypercardioid/lobe polar pattern provide high sensitivity, excellent off-axis rejection and accurate reproduction of each sound source.

Their rugged design and integrated filter make them insensitive to wind and vibrations. For greater flexibility, they can be fed with 8-48v phantom power. If phantom power isn't available, the battery/phantom powered version MCE 86S or the battery-powered MCE 87S can be used. The MCE 87VS can be powered directly by the camcorder, if an output for external devices is provided (5- 9v, 2.5 mm jack plug). The balanced output of the MCE 86 enables the use of long cables. The output of the MCE 87 is unbalanced. The MCE 87 is extremely insensitive to interference caused by view finders.

Available with two elastic suspensions: the EA 19/25 with the MKV 11 microphone clamp for fishpole booms and stands or EA 86 with horseshoe connection for the use on video cameras.

MCE86 with EA86 elastic suspension and mounting kit **364.95**

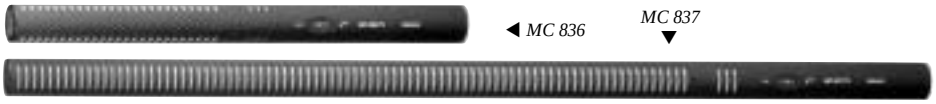
MCE86S (same as above, plus On/Off switch).....**374.95**

MCE86 with EA19/25 suspension and MKV-11 mic clamp **394.95**

MCE87S Battery powered (switchable); with EA86 and mounting kit ..**348.95**

MCE87VS Powered by camera; with EA86 and WS716 windscreen **279.95**

MC 836 / MC 837 Shotgun Mics



The rugged and lightweight MC 836 (Lobe/Cardioid polar pattern) and MC 837 (Lobe pattern) shotgun microphones have been designed to meet the demands of ENG and EFP applications. Due to their acoustical characteristics they are also suited for studio, film and theatre applications. The directional polar pattern helps to suppress lateral and background noise.

Both mics operate according to the pressure gradient/line transducer technique. A switchable low frequency roll-off eliminates unwanted low frequencies at a cut-off frequency of 90Hz. An integrated footfall sound insulation avoids the transmission of noise below 40 Hz. (To avoid wind noise when used outdoors, a wind shield is needed). Both run on 11-52v phantom power.

MIC ACCESSORIES

Fishpole Booms

MZA-716: 4-piece fishpole boom in matte black with swivel joint MZG-2 for mounting the mic at any desired angle. The pickup direction can be varied within 360° simply by turning the fiber glass tubes. Fitted with 3.8″ thread for microphone clamps of flexible mounts (with EA 19/25 shock mount). Working length (extended) is 10′, collapsed length is 3′. Weighs 1.3 lbs.....**258.95**

MZA-717: 2-piece light fishpole boom made of matte aluminum. Adjustable from 2″ to 5′ **108.95**

MZA-718: 3-piece light fishpole boom made of matte aluminum. Adjustable from 3′ to 10.5′.....**163.50**

Shock Mounts

EA 19/25 Elastic Suspension for the M59 ...**49.95**

EA-37 Elastic Suspension for the MCD-100 and MCD-101.....**59.95**

Windscreens

WS-59 Charcoal-gray. For M59.....**28.95**

WS-91 Charcoal-gray. For MCE-58.....**Call**

WS-716 Charcoal-gray. For MCE-86, MCE-87, MC836 .....**35.95**

WS-717 Charcoal-gray. For MC837 .....**64.95**

WS-740 Charcoal-gray. For MCD-100, MCD-101, MC-740, MC-834.....**34.95**

PS-740 Black Popscreen for MCD-100, MCD-101, MC-740, MC-834 .....**67.50**

WJ-726 Windjammer for MCE-86/87.....**Call**

Mic Accessory Kits

KWS-833 Set: Includes basket-type windscreen, elastic suspension and pistol grip for the MC-833.....**Call**

KWS736/836 Set: Consists of KWS726/826 basket-type windscreen, EA736/836 elastic suspension and MZP767/867 Pistol Grip for the MC836/MC837 .....**Call**

	MCE 86 / MCE 86S	MCE 87S	MCE 87 VS	MC 836	MC 837
Transducer Type	Condenser	Condenser	Condenser	Condenser	Condenser
Frequency Response	50 to 18k	50 to 18k	50 to 18k	40 to 20k	40 to 20k
Polar Pattern	Hypercardiod/Lobe	Hypercardiod/Lobe	Hypercardiod/Lobe	Lobe/Cardioid	Lobe
Maximum SPL at 1kHz	138 dB	138 dB	122 dB	130dB	130dB
Signal-to-Noise Ratio rel. to 1 Pa	61 dB	61 dB	61 dB		
Nominal Impedance	190Ω	190Ω	50Ω	180Ω	180Ω
Load Impedance	1000Ω	1000Ω	1000Ω	1000Ω	1000Ω

BLUE

BOTTLE

Tube Condenser Mic with Interchangeable Capsules

A classic modern vacuum tube microphone, the Bottle is a precision recording tool that combines the low noise and superb transient response of top grade electronics with the sonic magic of legendary vintage vocal microphones. With eight different interchangeable capsules the Bottle is not just a microphone, but a complete mic system. In an instant, capsules can be changed to deliver different tonality, patterns and uses while the mic is still turned on—eliminating the need for multiple mics. The Bottle mic incorporates the EF86 pentode (used in triode mode) and is packaged with one B6 Lollipop capsule, a power supply with Soft Start and a Champagne tube mic cable all housed in a Blue velvet lined ATA approved flight case. So if there is a large orchestra tuning up, a legendary singer about to record, or an impromptu gathering of famous musicians and there is no room for failure—you want the Bottle— you want only the best.



- The ultimate in hand built craftsmanship, Class A discrete circuitry, integrated capsule shock mounting, custom wound mic transformer, hand-tuned and tested.
- The Bottle is huge — 14.75″ and 4 lbs.
- EF86 pentode (used in triode mode) and a custom-built output transformer can
- Choice of eight interchangeable capsules, completely hand-built and individually voiced. The eight lolly popshaped capsule heads for the Bottle mic are:

B0: Cardioid, large diaphragm—superior bright, silky, and intimate presence

B1: Cardioid, small diaphragm—accurate sound with detailed high end

B2: Figure 8, large diaphragm—big, warm sound of vintage ribbon mic

B3: Cardioid, midsize diaphragm—very transparent, neutral sound

B4: Perspex sphere pressure omni (like on the Neumann M50)—sweet highs, robust lows, unique design and sound perfect for ambient recordings

B5: Presence Omni, large diaphragm—very unique compressed quality, accentuated lows and highs

B6: Cardioid, large diaphragm—dual backplate (like an AGK CK-12). "Larger than life" sound with enhanced lows and high frequency presence

B7: Cardioid, large diaphragm—single backplate (like a Neumann M7). Classic vintage tube detail in mids and airy presence.

- Power supply has a SOFT START feature to assure the longevity of the vacuum tube, and the stability of the mic circuitry. First it turns on the heater. After approx. 80 seconds — when the cathode is fully heated — the plate voltage is gradually applied and the microphone output is muted. After about three (long) minutes, when the mute has settled into its correct operating mode, the muting is disabled and audio is output from the supply.
- Although optimized for 60v, one can step it up to 90v or down to a minimum of 34v. This is useful for matching the sensitivity requirement of different program material; it also changes the sound of the mic from most aggressive (at higher voltages) to mellow and laid-back (at lower voltages).

CACTUS

Multi-Pattern "Vintage Sound" Tube Mic



A blend of the past with the cutting-edge of the future. With detailed information in the mid-range area and a soft silky response in the top-end, Cactus fills the need of detailing and mellowing the program material. Ideal for vintage pop vocal, brass, and electric guitar.

- Multi-pattern condenser-omni, cardioid, figure-8
- Vintage tube and Class A discrete circuitry
- Blue's famous hand-built large diaphragm capsule
- Vintage sound—rich lows, detailed mid-range, silky presence in the highs
- Integrated capsule shock mounting; mic hand-tuned and tested
- Includes premium quality Cactus power supply with SOFT START, Champaigne tube mic cable, Series 1 shock/pop integrated shock mount and pop filter, and ATA-style storage case

KIWI

Multi-Pattern Studio Condenser Mic



Blue's top of the line Class A Discrete model, the Kiwi (like the Cactus) has three capsule patterns (Cardioid, Figure- 8, Omni) and selections in between for a total of nine different choices. This is achieved in a careful, unique manner using the BLUE double backplate capsule. Mounting to the Kiwi microphone amplifier is provided by a mold injected rubber stem to help isolate unwanted rumble. Additional protection from low end rumble is provided by the use of the elastic spider type shockmount. This microphone provides the recordist's with many options for the most critical of all recordings. Ideal for silky lead vocal, piano, and multi-pattern recording.

- The capsule is modeled after the B6 capsule used in the Bottle mic but in a multi-pattern version using the same large-diaphragm, dual-backplate design. A switch on the back provides settings for cardioid, omni and figure-8, as well as three intermediate settings for each (nine patterns total).
- A wide, smooth presence boost that starts at about 6.5 kHz with a +3dB peak around 12 kHz that's particularly nice on male lead vocals and adds a nice airiness to male and female background tracks.
- Kiwi's attractive green body houses modern, transformer-less electronics — Class A, all-discrete design (no ICs), using high-quality components such as metal film resistors. The approach is minimalist; the signal path has no attenuation pads or low-cut filters that could compromise the signal.
- Modern sound—rich lows, contoured mid-range, extra airy presence in the highs. Adds punch, clarity and a low-end warmth on spoken and singing tracks — a trait that male and female vocalists will appreciate.
- In addition to the outboard shock-mount, Kiwi's capsule is mounted on a rubber stem, providing near-total isolation from external vibration and rumblings.
- Includes "The Shock," a robust elastic shock-mount and a cherry-wood storage box. Kiwi Cable and a stainless steel pop filter are available as options.

BLUE (stands for Baltic Latvian Universal Electronics) builds and tunes each mic capsule by hand. The finished capsules are measured in their anechoic chamber for the optimum performance. Each mic capsule is mounted with BLUE's own molded, injected rubber stem to isolate unwanted rumble. The mylar film of the capsules are sputtered with BLUE's special formula of 24-carat gold of absolute pure (99.99%) quality and aluminum. This allows the mics to achieve the fastest transient response without the sacrifice of low-end frequencies. And insured by the use of high-end quality components, BLUE applies strict electronic and construction tolerances that guarantee unsurpassed reliability and low noise. That is how they are able to handle high SPLs and still capture the crucial dynamics that make recordings stand out.



MOUSE *Cardioid Studio Condenser Mic*

The Mouse mic amplifier consists of top grade quality components such as polystrene condensers and metal film resistors. The single pattern, cardioid handbuilt capsule uses only the highest quality mylar film and is sputtered with a special mixture of 24 karat gold (99.99%) of absolute pure quality and aluminum. The mic capsule is shockmounted in two areas, internally in the grill assembly and within the rotating grill armatures.
Finished in a dark royal blue, the Mouse uses a transformerless circuit design. However, a transformer-based (matte black finish) Mouse is available as well, allowing the use of extra long mic cables for special applications while providing the utmost protection from outside interference such as RF. Both include a cherrywood storage box.

- Multi-purpose mic ideal for broadcast vocal, lead vocal, voice-over, bass amp, and kick drum.
- Big sound—robust lows, contoured mid-range, and an airy presence in the highs.
- Integrated suspension shockmount and rotating head for optimal placement.
- Optional accessories include "The Shock" shock mount, "The Pop" pop filter and Cranberry cable

BLUEBERRY



The Blueberry consists of a Class A discrete mic amplifier perfectly chosen to the handbuilt, large-diaphragm, cardioid pattern capsule. BLUE's entry level microphone, The Blueberry is designed to fill the needs of the home musician and professional alike. The timbre of the microphone capsule has been tuned to provide the end user with a silky bright and airy top end reminiscent of the best vintage microphones. This provides the engineer with a recorded signal, such as a vocal, minus unwanted low end resonates that can cause a compressor to breathe or pump. The final results are vocals and instruments that sit and blend with the basic track of recorded program material.

- Intimate sound—tight lows and mids, contoured top end for superior airy presence in the highs
- Multi-purpose mic that is ideal for lead vocal, acoustic guitar and backing vocal
- Hand tuned and tested
- Includes cherrywood storage box, optional accessories include the Series 1 Shock/Pop, Series 2 Shockmount and Blueberry cable



DRAGONFLY

Complete with an integrated elastic shockmount, this class A discrete, transformerless microphone offers a rotating capsule grill that can be positioned in the most difficult of areas. Countless hours were spent to test and select the most suitable active and passive components. This no compromises process ensures the low noise, high dynamic range and unappreciable distortion of the microphone circuitry. The end result give the Dragonfly a complete sonic neutral electronics with a low output impedance and no side effects from cable capacitance and load impedance. This way only the capsule determines the sound. The Dragonfly is packaged in an attractive linen box. Available in a matched set.

- Cardioid condenser with Class A discrete circuitry
- Versatile sound—rich lows, contoured mid-range, silky presence in the highs
- Integrated elastic shockmount and rotating head for optimum placement
- Multi-purpose mic—ideal for pop vocal, electric guitar and drum overheads
- Includes linen storage box
- Optional accessories include the Cranberry Cable

	Bottle w/B6	Cactus	Kiwi	Mouse	Blueberry	Dragonfly	Baby Bottle
Directional Pattern	Cardioid	Multi-pattern	Multi-pattern	Cardioid	Cardioid	Cardioid	Cardioid
Frequency Range	20Hz-20Hz	20Hz–20kHz	20Hz–20kHz	20Hz–20kHz	20Hz-20Hz	20Hz–20kHz	20Hz–20kHz
Sensitivity *	20mV/Pa	18mV/Pa	19mV/Pa	21mV/Pa	20mV/Pa	21mV/Pa	33.5mV/Pa
Rated Impedance	200Ω	150Ω	150Ω	150Ω	150Ω	50Ω	50Ω
S/N Ratio CCIR468-3	76dB	71dB	76dB	76dB	76dB	76dB	76dB
S/N Ratio DIN/IEC 651	87dB-A	82dB-A	87dB-A	87dB-A	86dB-A	87dB-A	87dB-A
Noise Level DIN/IEC 651	7,5dB-A	10dB-A	8dB-A	7dB-A	8dB-A	7dB-A	5.5dB-A
Max. SPL for THD 0.5%	134dB	130dB	133dB	1334dB	133dB	132dB	133dB
Mic. Amp Dynamic Range	78dB	77dB	75dB	76dB	75dB	76dB	128dB
Dimensions	15.2 x 3.5″	90 x 1.96 x 1.18″	8.6 x 2.4″	6.5 x 2.6″	9.3 x 1.96 x 1.2″	6.5 x 2.4″	8.7 x 1.8″
Weight	3.7 lbs.	1.76 lbs.	1.94 lbs.	2.16 lbs.	1.14 lbs.	1.38 lbs.	12.3 oz.

**@1kHz into 1 kohm*

BABY BOTTLE

BLUE's most affordable microphone to date, the Baby Bottle employs the styling, attention to detail, and hand-crafted quality for which BLUE has become famous. Each and every part of this solid-state, class A discrete condenser has been hand selected to insure nothing short of a stellar performance. Enclosed in a Lollipop spherical grille is a precision-machined, gold-sputtered capsule with a fixed cardioid pattern. Ideal for recording vocals, percussion, or any acoustic instruments. Comes nestled inside a velvet pouch protected by a beautiful cherrywood box.



- Cardioid condenser Class A discrete circuitry
- Upfront sound—robust lows, detailed mid-range and an accurate presence
- Multi-purpose mic ideal for classic lead vocals, brass and string instruments
- Hand tuned and tested
- Includes cherrywood storage box, velvet bag and mic clip. Optional accessories include the Baby Pop pop filter, Baby Shock shock mount and Blueberry Mic cable

Microphone Cables

BLUE's high definition microphone cables gives you noticeable sonically improvement. Champagne is the ultimate tube mic cable with nothing comparably available elsewhere. The audio signal conductors are separately shielded from all the other conductors. With this is achieved additional clarity to your mic signal. The Blueberry, Cranberry and Kiwi accompany the Class A Discrete microphones of the same name. Built to BLUE specifications, as provided below, these mic cables offer the recordist and performer, a signal path of sonic quality and high durability.

Champagne Tube Mic Cable

A perfect blend of the best materials, the Champagne cable will tickle the finest taste buds, as well as warm the coldest critic. The perfect compliment to the ultimate tube microphones, Champagne is the only cable designed to reproduce the quality sound you would expect to hear out of the best tube microphones in the industry.
A 2 conductor 22 awg with 60 strands of 40 awg using only the finest of BLUE's virgin proprietary materials available for the microphones transmission line. BLUE then twist the pair to totally eliminate noise and covers then with the finest tinned copper braid to insure your listening environment. To guarantee voltage and power, BLUE incorporates 5 more conductors or 24 gauge. When you listen to the bouquet, it could only come from BLUE's classic Champagne. Supplied with the Bottle and Cactus tube mics.

Blueberry (BB)

A two conductor 22 Awg cable using only BLUE's virgin proprietary materials. The twisted pair construction eliminates any noise caused by electromagnetic interference emanating from equipment used in the studio or stage environments. Furthermore, the tinned copper 95% braided shield makes this cable a dream to handle in any temperature condition where microphonic pick-up might come into play. When it comes to maximum frequency response, this is definitely a fruit of a different color.

Cranberry (CB)

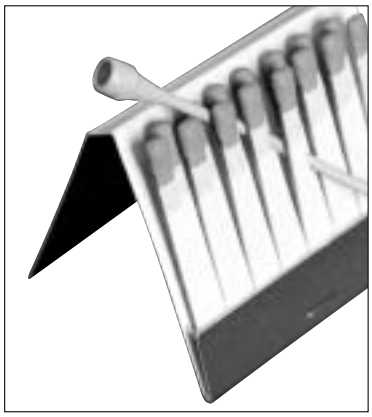
A rugged two conductor 20Awg twisted pair mic cable using BLUE's virgin proprietary materials. This cable is the perfect compliment for any type of radio, TV, or sound reinforcement applications. The twisted pairs reject any electromagnetic interference that may be caused by surrounding equipment. Also, this cable has a 95% tinned copper braided shield. The braided shield is excellent in eliminating microphonic pick-up which will allow a noise free listening environment. Ultra-flexible and ultra quiet, this cable might taste sour, but it will never be sour to your ears.

Kiwi (KB)

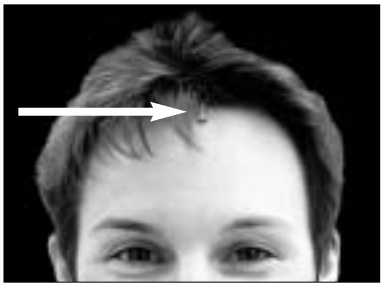
The ultimate in quad mic cable design, the Kiwi cable uses BLUE's hand picked virgin proprietary materials. The quad design gives the cable an advantage when trying to achieve the highest quality recording where exquisite tonal quality is the utmost importance. By using 4-22 Awg twisted pair conductors and a 95% tinned copper braided shield, this will eliminate any matters of electromagnetic interference and microphonic pick-up. When you're looking for the sweetest in sound, pick a Kiwi!

B3

Broadway Series Lavalier Microphone



The B3 is a very small, round, omnidirectional electret microphone specially-equipped to withstand moisture and makeup when placed in the hair or on the body. It will even survive an overnight bath in your favorite cola beverage. Extremely small, the B3 looks great when you want to see it, hides easily when you don't. Hide it in hair of actors, news anchors and lecturers. Ideal for use in churches, schools and general lavalier applications, the B3 features up to 150dB SPL on 48v phantom power, has field-selectable high-frequency response, and with Kevlar reinforced cable and an ultra thin diaphragm, sets a new standard for low handling noise. Available in five colors (Black, White, Light Skin, Cocoa, Gray).



- Excellent choice for hair miking in theater work. Features include moisture resistance, small size, and flat frequency response.
- The protective cap on the B3 is easily removed for cleaning or replacement when clogged with makeup.
- Includes tie clip, wind screen, belt clip for connector and two High-Frequency response control protective caps.

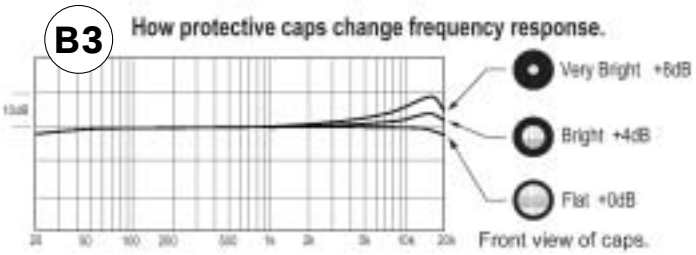
OPTIONAL ACCESSORIES FOR THE B3 AND B6:		
Single Tie Clip **14.95**	**Magnetic Clip** **16.95**	**Windscreen** **7.95**
Double Tie Clip **21.95**	**Viper Clip** **13.75**	**Protective Cap** **9.95**

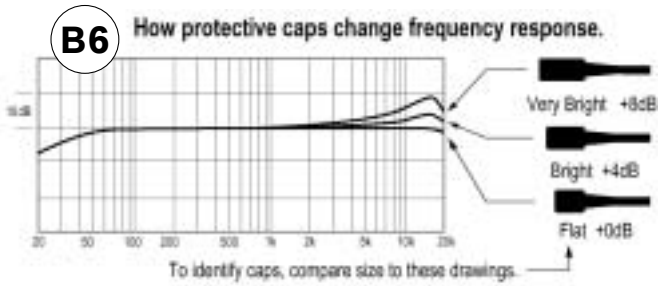
B6 *Incredibly Tiny Lavalier Microphone*



Only one tenth of an inch (2.5mm) in diameter, the B6 is smaller than the cable of other lavalier microphones, yet out performs microphones many times it's size. The B6 is effectively hidden when it's in plain sight. For example choose a protective cap near the color of a shirt and poke the B6 out through a button hole. Tape the cable to the back of the shirt to secure it and your done. Say goodbye to the hassle and degraded sound quality you get burying mics under clothing! Hide it in hair or glue on the faces of actors, news anchors and lecturers. t is also designed to survive.

Highly resistant to moisture, makeup and other hazards found in a production environment, the B6 will even survive an overnight soak in your favorite cola beverage. The outer screen in the B6's replaceable cap stops makeup and sweat before it enters the mike element. If it clogs you can simply remove the cap for cleaning or replacement. The B6 includes a tie clip, wind screen, belt clip for connector and a set of three high frequency protective caps so you can match your application.





ISOMAX E6 EARSET

Ultra Light Headset Microphone

The E6 EarSet is truly revolutionary combination of beautiful form and outstanding sound a package so small, it's all but invisible. Perfect on stage or in church for clergy, performers, trade show presenters, auctioneers, teachers and lecturers, the Isomax E6 EarSets are the answer when you need the highest quality voice pickup without the bulk and appearance of a conventional headset. Light and springy, Isomax E6 EarSets clip around your ear, not around your head so they never interfere with hairstyles, headphones or hats. Available in Light Beige, Tan, Cocoa and Black colors.



- The industry's smallest, lightest and least visible headworn microphones.
- Clips directly to your ear. No conflict with hairstyles. No adjustment needed for the width of your head. The cable attaches with a quick connect connector for easy service.
- Hhas the wide 50Hz to 20kHz response of a world class vocal microphone.
- Cardioid and hypercardioid patterns available for tough feedback situations.
- Handles high sound levels (up to 145dB)
- Available for use with most wireless transmitters or hard-wired phantom power with adapter.
- Good rejection of feedback and unwanted noise
- Resistant to the sweat and moisture on stage
- High frequency response can be adjusted by changing the provided protective caps.
- Available in four colors (light beige, tan, cocoa and black) and three sensitivities:
  E6OW5 for normal speaking (135dB SPL),
  E6OW6 for general theater (145dB SPL),
  E6OW7 for very loud vocals (155dB SPL)

MICROPHONES FOR WIRELESS

Manufacture	Connector Type	B3 Omni Lav Mic	B6 Omni Lav Mic	E6 Headset Mic	E6 EarSet Cables
AKG	Switchcraft TA3F	B3W[ ]FF05[ ]AK	B6W[ ]FF05[ ]AK	E60W[ ] [ ]AK	E60CABLE[ ]AK
	B-LOC (3.5mm)	B3W[ ]FF05[ ]AG	B6W[ ]FF05[ ]AG	E60W[ ] [ ]AG	E60CABLE[ ]AG
Audio-Technica	Special 3.5mm Plug	B3W[ ]FF05[ ]AP	B6W[ ]FF05[ ]AP	E60W[ ] [ ]AP	E60CABLE[ ]AP
	Switchcraft TA5F	B3W[ ]FF05[ ]AW	B6W[ ]FF05[ ]AW	E60W[ ] [ ]AW	E60CABLE[ ]AW
Azden	Hirose 4-pin	B3W[ ]FF05[ ]ZU	B6W[ ]FF05[ ]ZU	E60W[ ] [ ]ZU	E60CABLE[ ]ZU
	Non-locking 3.5mm	B3W[ ]FF05[ ]ZP	B6W[ ]FF05[ ]ZP	E60W[ ] [ ]ZP	E60CABLE[ ]ZP
Beyer	Beyer 4-pin	B3W[ ]FF05[ ]BS	B6W[ ]FF05[ ]BS	E60W[ ] [ ]BS	E60CABLE[ ]BS
Lectrosonics	Switchcraft 851	B3W[ ]FF05[ ]L2	B6W[ ]FF05[ ]L2	E60W[ ] [ ]L2	E60CABLE[ ]L2
	Switchcraft TA5F	B3W[ ]FF05[ ]LS	B6W[ ]FF05[ ]LS	E60W[ ] [ ]LS	E60CABLE[ ]LS
	Switchcraft A3M	B3W[ ]FF05[ ]LX	B6W[ ]FF05[ ]LX	E60W[ ] [ ]LX	E60CABLE[ ]LX
Nady	Lemo 4-pin	B3W[ ]FF05[ ]NL	B6W[ ]FF05[ ]NL	E60W[ ] [ ]NL	E60CABLE[ ]NL
	Switchcraft TA3F	B3W[ ]FF05[ ]NS	B6W[ ]FF05[ ]NS	E60W[ ] [ ]NS	E60CABLE[ ]NS
Samson	Hirose 6-pin (gray or black)	B3W[ ]FF05[ ]SA (X)	B6W[ ]FF05[ ]SA (X)	E60W[ ] [ ]SA (X)	E60CABLE[ ]SA (X)
	Switchcraft TA3F	B3W[ ]FF05[ ]SV	B6W[ ]FF05[ ]SV	E60W[ ] [ ]SV	E60CABLE[ ]SV
Sennheiser	Microdot	B3W[ ]FF05[ ]SE	B6W[ ]FF05[ ]SE	E60W[ ] [ ]SE	E60CABLE[ ]SE
	Lemo 3-pin	B3W[ ]FF05[ ]S3	B6W[ ]FF05[ ]S3	E60W[ ] [ ]S3	E60CABLE[ ]S3
	Lemo 1-pin	B3W[ ]FF05[ ]S1	B6W[ ]FF05[ ]S1	E60W[ ] [ ]S1	E60CABLE[ ]S1
	3.5mm locking plug	B3W[ ]FF05[ ]SD	B6W[ ]FF05[ ]SD	E60W[ ] [ ]SD	E60CABLE[ ]SD
Shure	Switchcraft TA4F	B3W[ ]FF05[ ]SL	B6W[ ]FF05[ ]SL	E60W[ ] [ ]SL	E60CABLE[ ]SL
	Lemo 4-pin	B3W[ ]FF05[ ]SU	B6W[ ]FF05[ ]SU	E60W[ ] [ ]SU	E60CABLE[ ]SU
Sony	Twist-lock type 4-pin	B3W[ ]FF05[ ]SO	B6W[ ]FF05[ ]SO	E60W[ ] [ ]SO	E60CABLE[ ]SO
	3.5mm locking plug	B3W[ ]FF05[ ]SN	B6W[ ]FF05[ ]SN	E60W[ ] [ ]SN	E60CABLE[ ]SN

If your wireless system is not listed here, give us a call or check our website for additional models (many more are available).
When ordering, please specify desired color and microphone sensitivity.

MB SERIES

Mini-Boundary Microphone System



With their incredibly small size, Crown's MB Series mics offer a wide range of options and flexibility—making boardroom installations much easier and a lot more affordable. Using a miniature, low-profile designs these mini-boundary, supercardioid microphones virtually disappear when installed, yet provide the same great sound as Crown's highly-acclaimed PCC microphone series while providing substantial savings compared to traditional microphones.

Designed with the help of leading audio consultants and contractors, the MB Series is a comprehensive system of microphones and electronics that's perfect for a wide range of applications. Comprised of five microphones—three that work with the innovative MB-100 and MB-200 interfaces and two that operate in a traditional stand-alone manner—the MB System lets you mix and match microphones as needed for multiple configurations.

Designed to accommodate up to four microphones, the MB interfaces provide four separate inputs, outputs and programmable switching connections. With easy-hookup Phoenix connectors and a low profile, the MB interface mounts quickly and conveniently beneath a tabletop. All MB System microphones feature a surface-mounted supercardioid capsule for extended reach and clarity. Each microphone also features a built-in red LED that illuminates when the microphone is active.

The MB-1, MB-2, MB-3 and MB-4 are each designed for multi-miking applications such as teleconferencing, distance learning, boardrooms, and courtrooms. The MB-3 can also be used for surveillance in security applications, and the MB-4 in TV or theatrical applications.

FEATURES

- Removable grille assembly can be painted
- Remote sensing of switch closure (in MB-200)
- A common interface for up to four mics reduces the size and cost of the system
- Electret condenser mic elements deliver clean, clear, natural sound
- Low-Z balanced output prevents hum and high-frequency loss
- On/off switches can be added
- Switches can be internally programmed for touch-on/touch-off, momentary on, or momentary off (not on MB-3)
- Defeatable LED (not on MB-3)
- They operate on 12-48v DC phantom power
- Three year warranty

MB SYSTEM ELECTRONICS INTERFACE



The MB-100 and MB-200 are electronic interfaces that power up to four Crown mini-boundary MB-1, MB-2, or MB-4/E microphones. Interfacing up to four mics reduces the size and cost of the mic system and reduces installation time.

The MB-100 provides programmable switching to turn the mic on or off if desired (switches not provided). The MB-200 is the same plus it also has remote switch sensing via an optical coupler.

Two microphones can be wired to each pluggable terminal block, which plugs into the MB-100 or MB-200. Pluggable terminal blocks are also provided for the balanced outputs on the interface, and for remote switch sensing. They are powered by 18 to 48v phantom power from a mixer or a phantom power supply.

- Powers and equalizes up to four mics
- Provides balanced outputs from the interface
- Common interface for up to four microphones reduces the size and cost of the system
- Pluggable microphone inputs

MB-100.....**229.95** **MB-200**.....**307.95**

MB SERIES

MB-1
Removable Aluminum Insert



A low-profile circular microphone just 1.3″ wide, the MB-1 features a bottom-mounted 1/4″ stereo phone plug for quick insertion and removal from the supplied table insert.

A table insert with a phone jack that accepts the mic is included. Featuring a black anodized aluminum finish, the insert flush-mounts in a hole drilled in the table. It can be permanently installed, and includes a black dust cap for use after meetings when the mic is removed.

Thanks to its supercardioid pick-up pattern, the MB-1 greatly reduces ambient noise and pickup of room reverberation. The result is a clearer sound.

In combination with an MB-100 or MB-200 Interface, the MB-1 can be programmed for multiple switching functions.

Available in black anodized (MB-1), solid brass (MB-1BR) and paintable aluminum (MB-1AL) finishes.

MB-2
Removable Phone Jack



The tiny MB-2 (less than 1x1½″) is a removable low-profile rectangular microphone featuring a 1/4″ stereo phone plug for quick insertion and removal from a standard contractor-supplied jack (Switchcraft 152B Thick Panel Phone Jack or equivalent). A locating pin (removable) prevents the microphone from being rotated.

Thanks to its supercardioid pickup pattern, the MB-2 greatly reduces ambient noise and pickup of room reverberation. The MB-2 features multiple switching modes when used with an MB Interface.

MB-3
Built-in Electronics



A stand-alone unit with a built-in electronics package, the low-profile MB-3 mounts easily into tabletops, lecterns, walls or ceilings for permanent installation.

On the bottom of the mic is a tubular power module with a 3-pin XLR-type output connector. It is powered by 12-48V phantom power, and has a low-impedance balanced mic-level output.

Thanks to its supercardioid pickup pattern, the MB-3 greatly reduces ambient noise and pickup of room reverberation. The result is a clearer, more intelligible sound.

MB-4 / MB-4E
For Temporary Installations



The MB-4 is a mini rectangular microphone designed for stand-alone use. A power module with an attached 15′ XLR cable has all the required mic electronics built-in. The mic is powered by 12-48v DC phantom power, and is low-impedance balanced. The cable may be positioned to exit either the bottom or rear of the microphone.

Double-sided sticky tape is provided for attaching the unit under shelves, on walls, etc.

Thanks to its supercardioid pickup pattern, the MB-4 greatly reduces ambient noise and pickup of room reverberation. The result is a clearer sound.

The MB-4 is also available in a pigtail version. The MB-4E is the same, but without the electronics connector. Its cable fits through a small hole in the mounting surface and is intended for use with the MB Interfaces. (Must be used with the MB-100 or MB-200).

MB SERIES SPECIFICATIONS

Freq. Resp.: 50 Hz to 15 kHz
Sensitivity: 22 mV/Pa
Impedance: 150 ohms
Polar Pattern: Half-Supercardioid

MB SERIES PRICING

MB-1 Black finish.....**113.95**
MB-1AL Aluminum finish**113.95**
MB-3 .....**152.50**
MB-4E without an electronics connector.....**101.95**
MB-1BR Brass finish.....**113.95**
MB-2 .....**98.95**
MB-4 .....**118.50**

...for Boardrooms, Teleconferencing, Distance Learning, Theater and Surveillance

PCC SERIES

Phase Coherent Cardioid Microphones for Stage Floor, Altar Tables, Boardrooms and Teleconferencing

Crown's PCC (Phase Coherent Cardioid) Series microphones offer exceptional acoustic performance in handsome, low-profile enclosures. Designed for high-end theatrical work, the PCC Series has become the standard for stage miking applications. These mics also provide an elegant solution for discrete multi-mic applications, such as lecterns and boardroom tables as well as altar, teleconferencing and distance-learning applications.

Each mic employs Crown's Phase Coherent Cardioid boundary mic technology to provide impressive gain before feedback and directional pickup without sacrificing tonal quality. This results in microphones with a wide, smooth frequency response, free of phase interference from surface sound reflections.

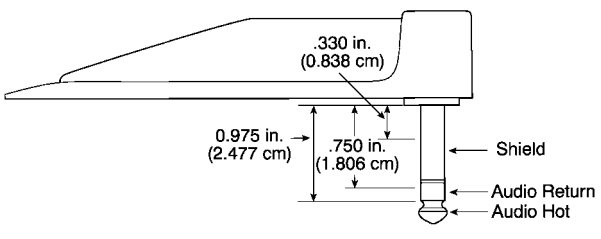
Designed to withstand the rigors of the stage, each microphone has been finely crafted using the very best materials while maintaining affordability. Their low profile enables them to be placed inconspicuously on the stage floor, on a lectern, or any other appropriate surface.

Flexibility is further enhanced by a membrane switch control option for boardroom and podium applications. This unique, silent operating switch can be configured three ways: touch on/off, momentary on, or momentary off operation.

FEATURES

- They are surface-mounted miniature cardioid microphone of professional quality. Handsomely styled, they are appropriate for use on the most elegant boardroom table or lectern. Other applications include churches, courtrooms and teleconferencing
- The microphones use the principle of phase coherency achieved by mounting a small-diameter mic capsule very near a surface or boundary. Direct and reflected sound waves arrive at the mic in-phase.This increases sensitivity 6 dB and ensures phase coherency up to the highest frequencies in the audible spectrum, resulting in a wide, smooth frequency response free of phase interference. Clarity and reach are also enhanced
- Electret-condenser capsules provides a wide, smooth frequency response from 50 Hz to 20 kHz. RFI suppression is included. Self-noise is low and sensitivity is very high
- Capable of withstanding up to 120 dB SPL without distortion, they will never overload in practical use
- All include three-position (Flat, Cut and Boost) bass-tilt switch on the bottom of the microphones allows you to tailor the low-end frequency response for particular applications. On the bottom of the microphones is a three- position switch allowing you to adjust low-frequency response:
  - The FLAT position provides a flat low-frequency response, for normal usage. They are shipped from the factory in flat position
  - The CUT position rolls off the bass,useful in noisy or boomy surroundings
  - The BOOST position boosts the bass for a more natural sound when the mic is used on a small surface such as a lectern
- Self-contained electronics eliminate the need for an in-line preamp. Powered by 12-48v DC phantom power, they have a low-impedance balanced output which permits long cable runs without hum pick-up or high-frequency loss.
- They use a Switchcraft TB3M mini-XLR as their standard connection. In addition, all except the PCC-160 can be ordered with a 1/4˝ stereo phone plug on the bottom of the base plate (they have the suffix "SPP" added to the model number)
- All mics (except SPP versions) include a black 15´ cable with Switchcraft TA3F (mic end) and an XLR-M connector (output).
- Because of their highly directional pickup pattern, they minimize background noise and feedback.
- Permanent mounting is enabled by screw holes in the base
- Engineering attention-to-detail assures years of reliable use.
- All except the PCC-160 (all steel construction) are made of high-impact molded plastics and steel mesh grille.
- They each include a three-year warranty.
- All mics are available in an off-white version to blend with church altars or any situation where a lighter color is called for.

All except the PCC-160 are available with a 1/4˝ stereo phone plug on the bottom of the base plate instead of a cable. They have the suffix "SPP" added to the model number. For example, PCC-130SPP or PCC-170SW/SPP.



PCC-130/130W Mighty Mini Boundary



Sometimes, unobtrusive just isn't enough. Enter the PCC-130, featuring exceptional performance with the appearance of the PCC-170 in a package only about half the size. When the situation simply demands that a mic be heard but not seen, the PCC-130 with its small size, gives you an obvious choice. Its cardioid pattern reduces background noise, reverb, and feedback.

PCC-130 Black color .....**211.95** **PCC-130W** White color....**211.95**

PCC-130SPP Base-mounted, 1/4″ stereo phone plug version. Black color **211.95**

PCC-130WSPP Base-mounted, 1/4″ stereo phone plug version. White color **211.95**

PCC-130SW Mighty Mini Boundary



Same as the PCC-130, the PCC-130SW also offers a membrane switch control option. This makes it ideal for multiple-microphone use on a conference table where each person wants control of his or her microphone. A bottom-mounted slide switch configures the silent-operating membrane switch for touch on/off, momentary on or momentary off. A high-intensity LED lights when the unit is on. In the off position, the mic is attenuated 70 dB.

PCC-130SW **233.50**

PCC-130SW/SPP 1/4″ (6.35-mm) stereo phone-plug version **233.50**

PCC-160/160W Industry-Standard Boundary Mic



Long recognized as an industry-standard stage-floor microphone, the supercardioid PCC-160 is also great for lecterns, conference tables and news desks—anywhere maximum gain and articulation are essential. Thanks to its low profile and black finish, the microphone becomes almost invisible in use. A side-mounted connector complements the form factor of the PCC-160, allowing the unit to be placed effectively at the stage edge, at the top of a lectern or in other tight spots. If desired, the cable can be hard-wired for bottom entry. The heavy-gauge, all steel body protects the unit from abuse. Available in off-white finish (PCC-160W) for blending with church altars and other surroundings where a lighter color is appropriate.

PCC-160.....**249.95** **PCC-160W** white color **249.95**

PCC-170/170W The Elegant Boundary



A larger, more elegant version of the PCC-130, the PCC-170 features a frequency response carefully tailored to create extremely natural voice reproduction. Equally adept in the boardroom, at the lectern, or for teleconferencing. Available in off-white finish (PCC-170W) for blending with church altars and other surroundings where a lighter color is appropriate.

PCC-170 Black.....**224.95**
PCC-170W White.....**224.95**

PCC-170SW The Elegant Boundary



Same as the PCC-170, the PCC-170SW also offers a membrane switch control option. This makes it ideal for multiple-microphone use on a conference table where each person wants control of his or her microphone. A bottom-mounted DIP switch configures the membrane switch for touch on/off, momentary on or momentary off. A high-intensity LED lights when the unit is on. In the off position, the mic is attenuated 70 dB.

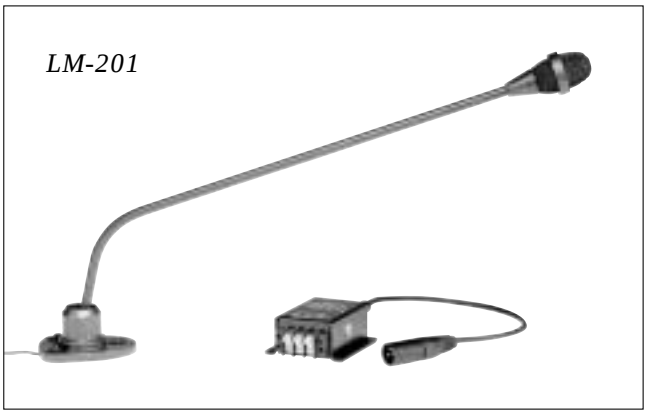
PCC-170SWSPP base-mounted, 1/4″ stereo phone plug version **254.95**

PCC-170SWO Includes all the features of the PCC-170SW, and adds remote sensing via a patented opto-coupled MOS switching circuit. A pair of extra leads in the mic cable allows remote sensing of the switch closure.....**286.50**

CROWN

LM SERIES

Lectern Microphones for Schools, Government, Boardrooms, and Houses of Worship



Crown's LM (lectern microphones) series is designed to provide clear, precise vocal reproduction with superior noise rejection in an elegant, functional package. Simple installation and stylish looks complement any lectern application, from the corporate boardroom to the bully pulpit, while their durable high-quality design provide years of consistent, reliable performance. Featuring a supercardioid element, they virtually eliminate the potential for undesirable noise to enter the system, even at high gain levels. Performance is further enhanced by specially designed pop filters that effectively reduce explosive breath sounds. Add their silent-operating gooseneck and shock mount designs, and you understand why LM Series microphones are so easy to position and use. Designed for the most demanding applications, they are manufactured with top-quality materials and feature reinforced joints and connections. Each microphone is individually tested to ensure perfect performance out of the box.

LM-201/LM-201L

Supercardioid Condenser Lectern Microphones

Slim, rugged and unobtrusive, the LM-201 (16″ arm) and LM-201L (24″ arm) are professional-quality, supercardioid condenser microphones ideal for legislature, city council and other demanding lectern applications. They easily install with the included mounting screws. When swiveled to the desired position, adjustment is far quieter than conventional gooseneck microphones.

- ◆ Rugged and is built to withstand daily use.
- ◆ Ball-and-socket joint combines with a rugged mic arm to provide quiet, long-term operation.
- ◆ Wire-screen grille with two-stage pop filter reduces pickup of explosive breath sounds.
- ◆ An external foam windscreen is supplied for extra pop rejection or for outdoor use.
- ◆ The base and mic capsule are shock-mounted to attenuate handling noise and lectern thumps.
- ◆ Powered by 9-48v DC phantom power

LM-201.....**264.95**
LM-201B (Black) **264.95**
LM-201L.....**Call**
LM-201BL.....**Call**

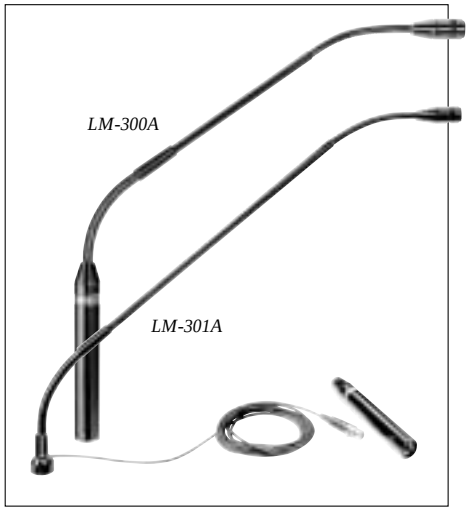
LM-300A/300AL

Supercardioid Condenser Dual Gooseneck Lectern Microphones

No matter the situation, the elegant LM-300A, LM-300AL and LM-301A can meet the need. Dual-gooseneck microphones for lecterns, pulpits, or similar applications, each is a professional-quality, electret-condenser mic with superb supercardioid performance and noise rejection. Explosive breath sounds are effectively reduced with the supplied foam 2-stage pop filter, while a flat/low-cut switch further reduces breath pops.

The LM-300A (17.25″) and the 22″ LM-300AL (extended) plug directly to an XLR female panel connector. The microphone screws onto a standard 5/8″-27 thread. Includes LM-SM shock mount for situations requiring even more isolation from noise.

LM-300A .....**219.95** **LM-300AL**....**229.95**



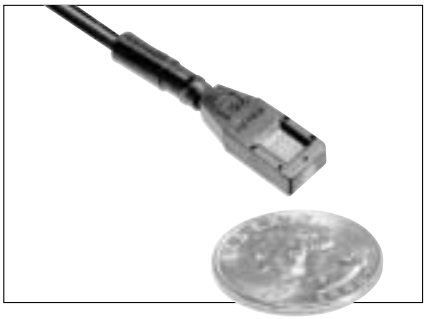
LM-301A

Supercardioid Condenser, Miniature Dual Gooseneck Microphone

The LM-301A is designed for professional use on lecterns as well as inconspicuous miking of musical instruments. The LM-301A comes in two parts: microphone cable and a cylindrical electronics housing. The microphone has a built-in 5/8″-27 threaded collar that can be screwed onto a mic stand, mic boom or an Atlas AD-12 male flange. Because of the unique design of the mounting collar, the mic cable can be routed either outside or inside the mounting collar. Includes the LM-301SMA shock mount adapter for use with the LM-SM shock mount **279.95**

GLM SERIES

GLM-100/GLM-200
Miniature Omni and Hypercardioid Condenser Mics



The GLM ("Great Little Mic") series are miniature omnidirectional (GLM-100) and hypercardioid (GLM-200) condenser microphones designed for vocals and musical instruments in professional recording, broadcast and sound reinforcement applications. They offer all the quality of large studio microphones, yet are nearly invisible in use. Built to the exacting standards of the film and broadcast industry, they incorporate miniature electronics in a rugged housing that resists damage from demanding professional use and provides lasting performance under the most rigorous of conditions.

GLM-100
MINIATURE FLEXIBILITY

For instrument applications, the GLM-100's miniature size and gigantic performance make it a natural for recording, sound reinforcement, broadcast and motion pictures. An omnidirectional condenser mic, it features a smooth, wide-range frequency response for truly professional sound quality.

GLM-200
EXTENDED RESPONSE

Also for instrument applications, the GLM-200 offers the same size advantages as the GLM-100 but provides extra gain-before-feedback for critical sound reinforcement applications. Another feature is uniform off-axis rejection. Ideal for musical instrument reinforcement or recording.

- Their output is balanced, low impedance, which allows long cable runs without high-frequency loss
- The frequency response of large microphone capsules is often degraded by the microphone housing (due to sound diffraction). In contrast, their housing is designed to actually improve the high-frequency re-response of the microphone capsule
- Used as a lavalier microphone, suspended over a choir or orchestra, or hidden on a film set, the GLM-100/200 disappear in use yet provides excellent sound quality
- The GLM-100/200 can be clipped or taped onto an acoustic guitar, sax or flute to allow the performer freedom of movement. They can also be attached to drum rims to pick up a drum set without the unsightly clutter of boom stands
- Permanently attached 8′ XLR cable
- Frequency response is wide and smooth, with emphasized high frequencies for presence and brilliance
- They accept 12-48v phantom power supply
- Excellent transient response, minimal off-axis coloration and high overload levels (GLM-100/150 dB SPL, GLM-200/131 dB SPL)
- The GLM-100E version of the mic comes without electronics for use with a wireless transmitter
- GLM-100 includes the GLM-TB Tie Bar, GLM-BC Belt Clip and WS-2 windscreen.
- GLM-200 includes the GLM-UM Universal Mount, GLM-TB Tie Bar, and GLM-BC Belt Clip and WS-2 windscreen

Optional Accessories

GLM-SM Surface Mount
Mounts a GLM to a surface, PZM-style)

GLM-DM Drum Mount
Provides permanent mounting on drums

GLM-HM Horn Mount
Includes a large, soft clip to prevent marring

GLM-CH Choir Hanger
For overhead miking

GLM-OHM Overhead Mount
Boom stand mount for overhead miking of drum sets. Screws onto a standard mic stand with 5/8″-27 thread

GLM-CM Cymbal Mount
Includes clamp, rod/elbow assembly and vertical rod)

LM and GLM SERIES SPECIFICATIONS

	LM-201	LM-300A	GLM-100	GLM-200
Element	Condenser	Condenser	Condenser	Condenser
Polar Pattern	◌	◌	○	◌
Frequency Response	100 - 15k Hz	80 - 15k Hz	20 - 20k Hz	60 - 20k Hz
Impedance	150 Ω	150 Ω	150 Ω	150 Ω
Sensitivity	6mV/Pa	6mV/Pa	3.2mV/Pa	6mV/Pa

CROWN

PZM SERIES

Crown's PZM (Pressure Zone Microphones) are serious tools for the sound reinforcement and recording professional, as well as cost-effective solutions for a variety of other applications. The first licensed manufacturer of the Pressure Recording Process (the foundation of PZM technology), Crown offers the most successful line of PZM microphones available today. With the PZM advantages of a 6 dB increase in sensitivity and S/N ratio, Crown's PZM series provide you with the ability to optimally place microphones in the performance venue. Designed for use on flat surfaces, PZM's can be placed where conventional mics do not work well. Small size and low profile also make them an excellent choice for recording projects, conference rooms and security applications. Their wide, smooth frequency response makes them ideal for miking pianos, harps and other orchestral instruments. In fact, top recording engineers all over the world are using Crown's professional PZM mics to improve the quality of their recordings.



Sound Grabber II

The Sound Grabber II is an affordable PZM mic designed to meet the needs of the home or business recordist and bring them the benefits of PZM technology. Ideal for conferences, group discussions, interviews, home video production, lectures, and recordings, the Sound Grabber II can be placed on a large surface such as a table, floor, wall, or lectern.

- Offers the opportunity for vast improvement in the quality of recordings
- Picks up sounds at distances you never thought possible with a clarity that other mics, because of the limitations of their construction, cannot match
- Sound Grabber II does not need to follow the action, since it has a wide-angle pickup pattern. So long as the mic "sees" the sound, in whatever direction or angle, it will pick it up clearly; it does not need to be between you and the other person
- Speakers or vocalists can move freely around the Sound Grabber II without their tone quality changing

Pressure Zone Microphone Technology

PZM mics use Pressure Zone Microphone technology in which sound waves reinforce themselves in the area nearest a flat surface, usually referred to as the boundary. Crown's PZM mics use a miniature mic capsule mounted very near a sound-reflecting plate or boundary in the Pressure Zone. In this zone, direct sound from the sound source combines in phase at all frequencies with reflected sound off the boundary. The benefits are many: 6 dB more sensitivity, 6 dB less noise, a wide smooth frequency response free of phase interference, excellent clarity and "reach," and consistent pickup anywhere around the microphone.

PZM-185 ***General Purpose Mic***

An affordable, all-around performer, the PZM-185 is ideal for applications such as conferences, group discussions, interviews, broadcast, home video production, lectures, and recordings. The mic can be placed on a large surface such as a table, floor, wall, or lectern.



- Powered either by an internal AAA 1.5V battery or 12-48v phantom power
- The boundary "paddle "can be detached so the mic will fit in your pocket. Attached to the mic is an 8´ cable leading to a tubular power module.
- Low-profile reduces "mic fright" and minimizes clutter. Nearly invisible on camera
- The PZM-185 minimizes pick up of vibration and handling noise. Its electret condenser capsule provides a wide, smooth frequency response from 50 Hz to 16 kHz
- Hemispherical polar pattern allows clear pickup of everyone surrounding the mic
- The mic is protected against static and RFI. Output is XLR-balanced, low impedance, which allows long cable runs without hum pickup or high-frequency loss

PZM-20R ***Special Purpose Pro***



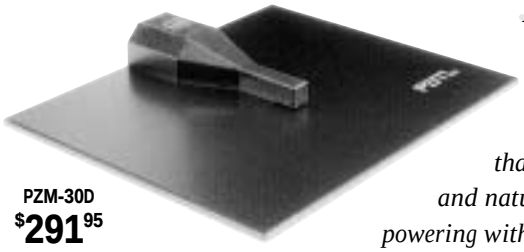
Perfect for permanent flush-mounting to conference tables, lecterns, pulpits, walls, you name it. A hemispherical pickup pattern, rising high-frequency response and low-frequency roll-off makes the PZM-20R a great choice for clear speech pickup. It also mounts in shallow cavities or a standard "double gang" electrical wall box and can be powered directly from phantom power via screw terminals. A white version also available.

- Designed for permanent flush-mounting in tables, lecterns, pulpits, and judicial benches
- Unobtrusive appearance and hemispherical pickup pattern makes it ideal for conference table use.
- Three small guard-posts prevent papers from sliding underneath the cantilever arm and altering the frequency response.
- It fits into a 4.59″ square hole or in a standard 4 x 4″ electrical outlet box. No external power-supply interface is required as electronics in the base of the PZM-20R adapt it for 12-48v phantom powering.
- The wide-range frequency response of the PZM-20R is emphasized at high frequencies for clarity and articulation. Low frequencies below the voice range are filtered out to reduce rumble.

PZM-20R (Black).....**279.95**

PZM-20RW (White) **279.95**

PZM-30D ***Switchable Performance***



PZM-30D
$291.95

Added acoustical gain. Clean, clear high frequencies. Uniform polar response. The PZM-30D offers all of these in addition to the PZM advantages in a rugged, durable package. A switchable dual-frequency response provides a "rising" position that adds brilliance and articulation, while the "flat" position maintains a smooth, flat and natural high-frequency response. Miniaturized electronics allow direct 12-48v phantom powering without external interface.

- The PZM-30D is designed for professional recording, sound reinforcement and broadcasting. It has many applications, from miking full orchestras or individual musical instruments to security or teleconferencing
- Switchable dual frequency response: rising (R) or flat (F). The "rising" position adds brilliance. This makes it useful wherever a crisp attack is desired, such as on percussion, drums, or piano. You can get a bright sound without boosting high frequencies on the recording console; the result is lower noise. The "flat" position provides a smooth, flat, high-frequency response for natural sound reproduction.
- The sturdy PZM-30D is especially reliable because it can be used with a rugged detachable cable. The output is 3-pin XLR-M balanced and low impedance, which permits long cable runs without hum pickup or high-frequency loss

PZM-6D ***Indisputably Inconspicuous***



PZM-6D
$279.95

With 100 percent of the performance but only one-quarter the footprint of the PZM-30D, the PZM-6D is the smallest high-performance PZM available. Designed for professional recording, sound reinforcement, teleconferencing and broadcasting, the PZM-6D is ideal for miking grand pianos, mounting overhead or for placement on conference tables. Like the PZM-30 it offers switchable dual-frequency "rising" or "flat" response.

- Unobtrusive appearance is great for conference rooms, as well as on TV/film and video productions
- Suspended over an orchestra on a clear panel, the PZM-6D practically disappears. Miniature, permanently attached cable also reduces visual clutter
- Capable of withstanding up to 150 dB SPL without distortion. Self noise is low, and sensitivity is high enough to override mixer noise in distant-miking applications

CROWN

PZM SERIES

PZM-10/10LL Security and Surveillance

Designed for security, surveillance, and conference table use, the PZM-10 is a low-profile omni pressure zone mic that doesn't look like a microphone, so it is inconspicuous. Featuring a cylindrical design for ceiling, wall or tabletop installations, it is practically invisible. The mic's tubular housing mounts in a 25/32˝ diameter hole in a ceiling panel, wall or table.



- In the PZM-10, low frequencies below the voice range are rolled off to reduce pickup of air conditioning rumble. The high-frequency response is boosted slightly to help articulation.
- Because of its tailored response and PZM construction, the PZM-10 will pick up conversations or other desired sounds with extra clarity and definition
- Unique mounting system allows quick installation
- Output is XLR-balanced, low impedance, which allows long cable runs without hum pickup or high-frequency loss. Powering is by 12 to 48 V phantom power
- The mic's humbucking transformer and steel case prevent hum pickup, even near fluorescent lights

PZM-10 ..... **99.95**

PZM-10LL The same as the PZM-10 with a line-level output from its balanced but unterminated cable **139.95**



PZM-11
$83.95

PZM-11 Security and Surveillance Mic

The PZM-11 presents a low-cost solution for security and surveillance applications. Mounted on a standard wall plate, the PZM-11 doesn't look like a microphone but rather an ordinary light switch. Intelligibility is excellent, with low frequencies below the voice range rolled off to reduce the pickup of HVAC rumble. Electronics on the rear of the plate offer screw terminals, eliminating the need for connectors. The PZM-11 uses 12-48 volts phantom power, while the PZM-11LL features line-level output and is powerable by 12-24V DC or 24V AC.

- Ideal for security, surveillance, and conference-room applications, the PZM-11 can be mounted in the ceiling or wall in a standard electrical outlet box. Providing excellent intelligibility,the PZM-11 is designed to look like something other than a microphone so as not to draw attention
- In the PZM-11, low frequencies below the voice range are rolled off to reduce pickup of air-conditioning rumble.The high-frequency response is boosted slightly to aid clarity and articulation
- The microphone connector is a row of screw terminals for easy installation.The output is balanced, low impedance,which allows long cable runs without hum pickup or high-frequency loss.Powering is via 12 to 48 volts phantom power
- The PZM-11LL looks and sounds like the PZM-11, but has a line-level output and is powered by 24v DC or phantom power. And since it has a high output, the PZM-11LL can be plugged directly into a VCR line input—no costly mic preamp required

PZM-11LL – Same as the PZM-11, with a line-level output and is powered by 24v DC or phantom power. The PZM-11LL can be installed without a box because its electronics are shielded inside a removable metal housing **106.50**

PZM SERIES SPECIFICATIONS

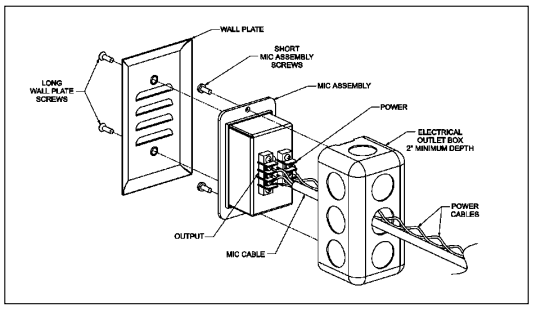
	Sound Grabber II	PZM-185	PZM-20R	PZM-30D	PZM-6D	PZM-10	PZM-11	PZM-11LL	SASS-P MKII
Element	Condenser	Condenser	Condenser	Condenser	Condenser	Condenser	Condenser	Condenser	Condenser
Polar Pattern	Hemi	Hemi	Hemi	Hemi	Hemi	Hemi	Hemi	Hemi	◯*
Frequency Response	50 - 16k Hz	50 - 16k Hz	80 - 20k Hz	20 - 20k Hz	20 - 20k Hz	80 - 20k Hz	80 - 20k Hz	80 - 20k Hz	20 - 18k Hz
Impedance	1600 ohms	150 ohms	150 ohms	150 ohms	150 ohms	150 ohms	150 ohms	75 ohms	150 ohms
Sensitivity	20mV/Pa	3.2mV/Pa	7mV/Pa	7mV/Pa	7mV/Pa	5mV/Pa	5mV/Pa	1.4mV/Pa	6mV/Pa

** Omnidirectional at lower frequencies; Unidirectional at higher frequencies*

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PZM-11LWR ***Security & Surveillance***

Similar to the PZM-11LL, the PZM-11LLWR is a weather-resistant PZM microphone with a balanced line-level output designed for outdoor use. A plastic membrane inside the microphone protects the mic capsule from water damage. The mic mounts in a standard electrical outlet box. Applications include fast-food restaurants, outdoor intercoms, toll booths, bridges, theme park security and home automation.



- ◆ Can be plugged directly into a VCR line input — no costly mic preamp is needed
- ◆ Output (via screw terminals) can be balanced or unbalanced.
- ◆ Powering is by 12-24V AC or DC
- ◆ Frequencies below the voice range are rolled off to reduce traffic rumble. The high-frequency response is boosted to help articulation. Tailored response and PZM construction picks up conversations or other desired sounds with extra clarity
- ◆ Mic is "weather resistant," which means:
  - ➢ Works during and after rainfall
  - ➢ Frequency response changes temporarily if the membrane gets wet or if water freezes on the membrane. Frequency response returns to normal when the membrane dries
  - ➢ High-pressure water sprayed directly on the membrane (through the louvered wall plate) may damage the membrane



SASS-P Mk II ***Natural Stereo Imaging***



The SASS-P MKII is a patented stereo condenser microphone using PZM technology. A mono-compatible, near-coincident array design makes it perfect for professional recording, broadcast and sampling. It uses two premium-quality PZM mics mounted on boundaries to make each directional. The carefully controlled polar patterns and human-head-sized spacing between the capsules create well-focused natural stereo imaging without a "hole in the middle." Reproduction is precise and realistic, with the SASS-P Mk II accurately conveying ambient environment for location recording, film sound or synced audio ambience. Separate left and right outputs are balanced. A low-frequency cut switch attenuates frequencies below 100 Hz by 12 dB-per-octave, reducing wind noise and low-end rumble. Powered by two 9-volt batteries or phantom power.

- ◆ Provides a very warm smooth sound with more high-frequency brilliance ("air") and less midrange "hardness."
- ◆ Reproduction of the sound field is precise and realistic. The SASS MKII accurately conveys the ambient environment for location recording film-wide sound, or synced audio ambience (movement of traffic) where movement heard must precisely match movements in the screen image
- ◆ A foam barrier/baffle between the capsules shapes the pickup angle of each capsule in the front, and limits overlap of the two sides at higher frequencies. Although the mic capsules are spaced a few centimeters apart, there is little phase cancellation when both channels are combined to mono because of the shadowing effect of the baffle.
- ◆ Powered in the field by two 9-volt batteries or 12-48V phantom power. Separate left and right outputs are low- impedance balanced
- ◆ Professional applications include:
  - ➢ Classical-music recording (ensembles)
  - ➢ Stereo electronic news gathering (ENG)
  - ➢ Sports ambience and crowd reaction
  - ➢ Televised theatre and dance
  - ➢ Film ambience
  - ➢ Stereo sampling for keyboards
  - ➢ Stereo sound effects

CROWN

CM SERIES



CM-150 Recording/Acoustical Measurement Microphone

The CM-150 is a 1/2˝-diameter omni-directional electret-condenser microphone designed for professional recording applications. It is also ideal for accurate acoustical measurement (free-field measurement and sound-level measurement) applications. The CM-150 is extremely stable over a wide range of environmental conditions and features a flat, wide-range frequency response. It includes a calibrated frequency response curve, WS-15 foam windscreen, ASA-4 stand clamp and a male XLR output.

- Pre-aged titanium diaphragm makes the CM-150 extremely stable over a wide range of environmental conditions.
- Very flat, wide-range frequency response.
- Powered by 18-48v DC phantom power.
- Supplied with a proof-of-performance sheet, which shows frequency response and sensitivity. This information can be used to calibrate the microphone.
- Mic capsule grille fits into a Bruel & Kjaer 4220 pistonphone or similar mic calibrator.
- When used for recording, the CM-150 sounds natural and accurate. It preserves the delicate timbre of acoustic instru-ments, yet can reproduce all the power of a pipe organ.
- Can handle very loud sounds (127 dB SPL) without distortion. It is protected against static and RFI.
- The off-axis response is also smooth, so any leakage picked up has little coloration. Self-noise is low, permitting clean recordings — especially with close miking.
- Output is balanced, low impedance, which allows long cable runs without hum pickup or high-frequency loss.
- Supplied foam windscreen reduces wind noise outdoors.

Crown's CM Series of condenser mics offer exceptional quality at truly affordable prices. With application specific microphones, ranging from professional-grade headworn microphones to critical response microphones for the most demanding applications, the CM series can fit almost any miking requirement. With innovations such as Crown's patented Differoid technology, these microphones combine years of technical know-how with durable, high-quality materials and workmanship.

CM-200A/CM-200ASW Vocal and Instrumental Mics

The CM-200A is a cardioid condenser microphone that delivers studio-quality sound in a rugged package able to withstand the toughest field conditions. Its smooth, articulate performance combined with a wide frequency response make it an excellent choice for professional applications. Its outstanding performance is complemented with handsome styling and a comfortable balance. The CM-200ASW is the same plus it adds an on/off switch.



- They offer studio quality sound, yet are rugged enough to withstand harsh professional use in the field.
- Sound is smooth and non-harsh – yet very articulate – with a wider frequency range than standard vocal microphones.
- Used up-close, their proximity effect boosts the low-frequency response for a warm, robust sound.
- Upper-midrange "presence peak" adds intelligibility.
- Cardioid pattern increases gain-before-feedback and ejects unwanted background noise.
- Built to withstand extremely loud sound pressure levels without distortion as well as repeated drops and abuse.
- They accept 12-48v DC phantom power
- Multi-stage pop filter effectively suppresses explosive breath sounds. Handling noise is extremely low.
- Balanced, low impedance output allows long cable runs without hum pickup or high-frequency loss.
- They include carrying case, adjustable mic stand adapter and a foam windscreen to further reduce breath popping and wind noise

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CM-310A/CM-310ASW Noise-Cancelling Condenser Mics

A hand-held noise-cancelling microphone for singers, sporting events or other applications with high ambient sound level, the CM-310A features Crown's patented Differoid technology to provide sizzling performance combined with the best gain before feedback in the industry. Used by top touring professionals worldwide, the CM-310A is ideal for high SPL sound reinforcement applications. And while it offers studio-quality sound, it is still rugged enough to withstand hard use on the road. The CM-310ASW adds switches for on/off operation and rising/flat high-frequency response.

- Differoid technology is differential or noise cancelling, so it rejects sounds that are not close to the microphone. It also discriminates against distant sound sources such as monitor speakers or instruments. An announcer can sit in the stands with the audience, with the P.A. loudspeakers playing loudly and not experience feedback
- Built to withstand extremely loud sound without distortion as well as repeated drops
- Cardioid pattern rejects sound from the rear, giving them exceptional gain before feedback
- They accept 12-48v DC phantom power
- Flat (sounds warm and smooth) or rising (sounds clear and crisp) frequency response can be selected via DIP switch inside the mic handle. Flat position has slightly more gain before feed-back. Announcers may prefer the rising position for its extra intelligibility
- Built-in pop filter suppresses explosive breath sounds. Handling noise is very low
- Balanced, low impedance output allows long cable runs
- Outstanding performance is complemented with handsome styling and a comfortable balance in the hand
- They include pouch case, mic stand adapter, and foam windscreen for outdoor use

CM-700/CM-700MP Cardioid Condenser Mics

The CM-700 is a condenser microphone intended for use in professional recording and sound reinforcement applications. With a cardioid pickup pattern and extremely low self-noise, this mic is perfect for clean, noise-free recordings of acoustic instruments, drum overheads and studio vocals. Able to handle up to 151 dB SPL, it features a smooth, wide frequency response and a three-position bass roll-off switch.



- Rugged enough for the road, the CM-700 is well suited for acoustic instruments, drum overheads and studio vocals. It works equally well for popular music (multi-miking) or classical music (stereo and spot miking). Small and inconspicuous, it is a good choice for miking a lectern
- Very smooth, wide-range frequency response gives it a natural sound. It preserves the delicate timbre of acoustic instruments, yet it can reproduce all the power of a pipe organ
- Cardioid pickup is uniform with frequency and reduces background noise, room reverb and feedback. Self-noise is also very low, allowing clean, noise-free recordings
- Handles extremely loud sounds without distortion (151 dB maximum SPL) and is protected against static and RFI
- To achieve pristine sound quality, the CM-700 incorporates an ultralight diaphragm, humbucking transformer, polycarbonate capacitors and a gold-plated 3-pin XLR connector
- Output is balanced, low impedance, which allows long cable runs without hum pickup or high-frequency loss
- Supplied 2-stage foam pop filter softens breath pops and a foam windscreen reduces wind noise outdoors
- Accepts 12-48v DC phantom power
- Three positions bass-tilt switch (flat, low-cut and rolloff)

ALSO AVAILABLE IN STEREO MATCHED PAIRS AS THE CM700MP

The CM-700MP is a matched pair of CM-700 cardioid condenser microphones. They are matched in sensitivity and frequency response within 1.5 dB. A pair of the CM-700's are an excellent choice for stereo recording of a classical music ensemble, folk group, drum set, piano, vocals, percussion and so on. Includes carrying pouch, ASA-4 swivel mount, foam pop filter and foam windscreen.

CM SERIES

CM311A Differoid Headworn Cardioid Condenser Microphone



Offering Differoid technology in a comfortable but rugged headworn package, the CM-311A sounds like the best handheld mic—even with the loudest singers. Designed for demanding touring and live sound applications, cardioid pattern rejects sounds from the rear, such as floor monitors while noise-canceling ability rejects sounds at a distance, such as instruments on stage and crowd noise. The result is full, clear, distortion-free sound with exceptional isolation and gain before feedback—the ideal choice for hands-free work directly in front of loudspeakers. Also available in a version that mounts on Sony MDR-7506 headphones (CM-311AHS) making it ideal for broadcast applications, and with a pigtail output (CM-311AE) for connection to the wireless transmitter of your choice. (CM-311A and CM-311AHS have a 3-pin XLR output).

Used by world-renowned artists like Garth Brooks and Janet Jackson, the CM-311A is the headworn mic of choice for the live-performance industry. Featuring clear, crisp vocal performance in a lightweight package, this microphone signifies the quality and performance value of the CM series.

- The headband and mic boom are adjustable to any vocalist, and because of the boom's unique behind-the-ear design, it does not cover up the singer's face.
- Supplied with a pop filter to greatly reduce breath noise and pops, and a 9v battery/phantom-powered beltpack
- The beltpack has an on/off switch, a programmable mute switch for hardwired use, and a "Mic On" LED

CM-311A .....**259.95** **CM-311AE** connects directly to any wireless transmitter.....**192.95**
CM-312AHS mounts on a Sony MDR-7506 headphone.....**244.95**

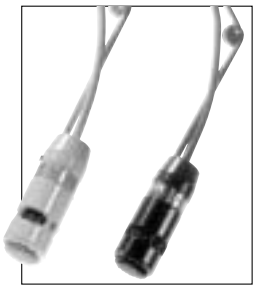
CM312A Headworn Hypercardioid Condenser Microphone



The CM-312A is a headworn mic with the fit and comfort of the CM-311A–but without Differoid technology. However, its hypercardioid pattern reduces feedback and aids isolation, making it ideal for live sound, aerobics and broadcast situations where a little less discrimination against high stage levels is required. (For more difficult situations though, the CM-311A is recommended). Sound can be tailored by moving the mic closer or farther from the corner of the mouth. And with the mic to the side of the mouth, breath pops are eliminated. Two other versions are available: The CM-312AHS mounts on a Sony MDR-7506 headphone making it ideal for broadcast applications. The CM-312AE offers pigtail output for connecting to the wireless transmitter of your choice (the CM-312A and CM-312AHS have a 3-pin XLR output).

CM-312 .....**Call** **CM-312AE** connects directly to any wireless transmitter.....**177.95**
CM-312AHS mounts on a Sony MDR-7506 headphone.....**244.95**

CM-30/CM-31 Miniature Supercardioid Condenser Mics



Professional-quality, supercardioid condenser microphones, the CM-30 is designed for inconspicuous miking of a choir, orchestra or stage. Virtually invisible and with a built-in (removable) hanger for easy positioning, it is the mic of choice for church choir and overhead stage miking. It can also hang over conference tables or discussion groups for inconspicuous isolated miking. Accepts 12-48v phantom power and includes 30´ mic cable which connects to the electronics/faceplate which fits a standard electrical outlet box.
Matching the CM-30's tiny size, outstanding performance and ease of use, the CM-31 is attached via a 30´ cable to an in-line power supply with a low-frequency cut switch and XLR output.

- Smooth, wide-range response makes them ideal for natural reinforcement or recording of the singing voice, organ or other musical instruments
- Ultra-tight supercardioid pattern provides impressive gain-before-feedback while rejecting unwanted noise and reverb
- Extreme low frequencies are filtered out to reduce pickup of room rumble and are protected against static and radio frequency interference

CM-30 Black .....**182.50** **CM-30W** White .....**184.95** **CM-31** Black .....**182.50** **CM-31W** White **182.50**

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CM SERIES

CM SERIES SPECIFICATIONS									
	CM150	**CM200A/ASW**	**CM310A**	**CM310ASW**	**CM700/MP**	**CM311A**	**CM312A**	**CM30**	**CM31**
Element	Electret Condenser	Electret Condenser	Electret Condenser	Electret Condenser	Electret Condenser	Electret Condenser	Electret Condenser	Electret Condenser	Electret Condenser
Frequency Response	20 to 20k ±1.5 dB	80 to 15k	60 to 17k	80 to 15 k	30 to 20k	50 to 17k	50 to 17k	40 to 20k	40 to 20k
Polar Pattern	○	ᔓ	ᔓ	ᔓ	ᔓ	ᔓ	ᔕ	Supercardioid	Supercardioid
Impedance	150 Ω	150 Ω	150 Ω	200 Ω	190 Ω	75 Ω	75 Ω	150 Ω	150 Ω
Open Circuit Sensitivity	10.5 mV/Pa	3 mV/Pa	1.4 mV/Pa	1.4 mV/Pa	2.5mV/Pa	4mV/Pa	3.8mV/Pa	10 mV/Pa	10 mV/Pa
Power Sensitivity	-39.5dB re 1 mW/Pa	-52dBm/Pa*	–54.5dB re 1 mW/Pa	–54.5dB re 1 mW/Pa	–52dB re 1 mW/Pa	–42.5dB re 1 mW/Pa	–43dB re 1 mW/Pa	–37.5dB re 1 mW/Pa	–37.5dB re 1 mW/Pa
Equiv. Noise Level	19dB SPL	21.5dB SPL	21.5dB SPL	21.5dB SPL	21dB SPL	26dB SPL	31dB SPL	28dB SPL	28dB SPL
S/N ratio	75dB at 94dB SPL	72.5dB at 94dB SPL	72.5dB at 94dB SPL	72.5dB at 94dB SPL	73dB at 94dB SPL	68dB at 94dB SPL	63dB at 94dB SPL	66dB at 94dB SPL	66dB at 94dB SPL
Max. SPL (3% THD) 12v	—	143dB SPL	143dB SPL	143dB SPL	142dB SPL	148dB SPL	148dB SPL	120dB SPL	120dB SPL
Max. SPL (3% THD) 48v	127dB	151dB SPL	151dB SPL	151dB SPL	151dB SPL	produces 3% THD			
Current Drain	2.7 mA	1.7 mA	1.7 mA	1.5 mA	3.2 mA	6.5 mA	6 mA	4.5 mA	
Weight	4.48 oz.	7 oz.	7.1 oz.	7.1 oz.	4 oz.	2.1 oz.	1.5 oz.	4 oz.	4 oz.

**1 pascal = 10 dynes/cm2 = 10 microbars = 94 dB SPL*

Crown Microphone Accessories

PH-1A Single Channel Battery Supply



A single-channel phantom power supply that operates from two standard 9v batteries or PS-24 AC/DC adapter. (Battery power provides 17v phantom and AC power provides 24v phantom.) The PH-1A powers all condenser mics that operate with phantom power of 18 to 24 v. It includes sufficient hum filtering for extremely quiet operation and is transformer isolated. If AC power fails,the batteries automatically take over. The rugged steel and aluminum chassis is designed to withstand daily abuse in professional applications **93.95**

PS-24 AC Adapter
A plug-in AC/DC adapter for the PH-1A. It will also provide power for the LM-200A and LM-201 microphones **29.50**

PH-4B Phantom Power Supply



The PH-4B is an AC powered 4-channel power supply providing 48V DC phantom power for up to four condenser and electret-condenser microphones. It uses XLR-type connectors for each channel of input and output. DC blocking is provided for easy coupling to input circuits **25.95**

CM-30-SA
Adapts the CM-30 and CM-31 for use on a mic stand or boom **29.95**


CM-30-SA

CM-SM
Shock mount for CM-700 and CM-150.....**74.95**

CM-SB
A stereo bar for the CM-700MP matched-pair mics .....**Call**


LM-SM

LM-SM
A fully adjustable shock mount for the LM-300a and LM-300aL, the LM-SM reduces mechanically induced noise. The rubber isolator is highly stable and securely holds the mic in place. **39.95**

LM-301-SMA
Adapts the LM-301a for use with the LM-SM shock mount.....**Call**

ASA-2
A stand adapter for the CM-200a and CM-310a hand-held microphones.....**29.95**

ASA-4
A stand adapter for the CM-700 and CM-150 microphone **29.95**


LM-301-SMA

DPA

4060/4061/4062

Miniature Microphones



Designed for close-miking applications, the 4060, 4061 and 4062 are almost invisible prepolarized omnidirectional miniature condenser microphones with a 5.4mm vertical diaphragm that deliver unsurpassed audio quality. Specially designed for maximum performance under the often difficult conditions when mounted directly on the human body, the utmost attention has been paid to the microphones' resistance to humidity. A double vent protection system together with water resistant materials inside the microphones make it almost impossible for moisture to cause the mics to fail. Furthermore, by changing the protection grids, the mics can be acoustically modified according to the placement of the microphone on the body. A wide range of connection adapters makes it possible to use the mics with almost any professional wireless systems available —even on 48v phantom power.

The three mic types are acoustically identical, but differ in their sensitivity and self-noise:
The 4060 has the highest sensitivity (20mV/Pa), and is able to handle sound pressure levels up to 134dB before clipping occurs. The 4061's sensitivity is 10dB lower to match some of the more sensitive transmitters on the market and offers a maximum SPL of 144dB . The 4062 is the high-SPL specialist, capable of handling up to 154dB before clipping occurs.

- Two different protection grids are supplied with mic for acoustic equalization. If the mic is mounted on a performer's head, the short plastic protection grid can be used to provide a 3dB soft boost at 8-20kHz. When the placement is on the performer's chest, a 10dB boost at 12kHz is required for the microphone to sound neutral. This unique acoustic equalization is noiseless and is provided by the longer protection grid. the grids can be cleaned if they become clogged with make-up, etc.
- The self-noise of the 4060 is only 23dB(A), absolutely unique for a microphone of this size. Effective dynamic range is 97dB, and there is 14dB headroom before clipping occurs.
- The 4060 and 4061 are primarily designed for use with wireless systems in theater or TV. A variety of ultra-compact adapters make it possible to match any of the miniature mics to virtually any wireless system —even on 48v phantom power.



Step 1



Step 2



Step 3

4065/4066

Miniature Microphone Headband Solutions



The 4065 and 4066 are headband mounted condenser microphone with the same prepolarized omnidirectional cartridge element as in the 4061. They are specially designed for maximum performance when mounted directly on the human body. A double vent protection system, a drop stopper on the tube, and water resistant materials inside the microphone, make it almost impossible for the 4065 or 4066 to fail because of humidity.

The 4065 offers maximum performance for stage, broadcast and home studio applications. It is designed for everyday use on the same performer. The headband is exceptionally lightweight and easily formed to the individual shape of each performer's head, providing a firm yet comfortable fit. Any adjustment in the bend is easily done with a soft touch, around a wide curved object.

The 4066 is primarily for head mounting on actors, public speakers, broadcast hosts, musicians and singers. The system is quickly and simply adjustable—and extremely versatile. It is designed for a different user everyday. The headband features very robust construction, choice of left or right mounting and color change (simply by changing to a boom in another color). It even enables two mics to be used on the same headband for situations in which continuity can never be compromised. The boom can be used separately when the headband isn't needed.

4071

Miniature Microphone with Presence Boost



A miniature omnidirectional condenser microphone, the 4071 features a specially-designed, fixed microphone grid with a soft presence boost of typically 5B at 4-6kHz—the area of speech intelligibility, clarity and definition of voices and instruments. While clarity seems to be missing from chest-worn microphone placement, the 4071 will make voices cut through. Ideal for ENG/EFP use on the reporter in studio or in the field. Or for film production use, with hidden microphone placements on the actors.

Otherwise identical to the 4061 (uses the same prepolarized omnidirectional miniature condenser capsule), the 4071 does add an acoustic low-cut filter so the built-in pre amp (or wireless system) will have much larger headroom.

When recording voices from mic placement on a performers body, no frequency below 100Hz is attractive. For this reason, the airing vent in the capsule has been specially designed to make the frequency response 2dB attenuated at 100Hz, with a 3dB per octave roll-off. Also reduces low-frequency rumble from wind noise when used outside.

DAK4060 Accessory Kit



This is a complete accessory kit with room for a Miniature Mic and one or more connection adaptors. The box is extremely rugged so it gives both microphone and accessories perfect protection. The box has a transparent lid, so you can make sure you have a complete kit without opening the box all the time.

Includes: 5 colored (mix) miniature windscreens (DUA0565), Black Vampire Pin, Black Magnet, single clip lapel and 3 single tie pins

DAK4070 Accessory Kits

Designed for use with the 4071 Microphone, the DAK4071-E Accessory kit is made for ENG/EFP when the mic can be visible, while the DAK4071-F is for film or TV when the mic can't be visible. Each kit includes a hard case with a transparent lid and one or more connection adapters.

DAK4071-E: Black Double Pin, Single Pin Holder, Double Lock Clip, windscreens (7 colors), windjammer and mesh **84.95**

DAK4071-F: Miniature concealer, Tie Pod, Mic Tape, Double-sided Tape for Concealer and Tie Pod, windjammer and mesh....**119.95**

4060 High Sensitivity Miniature Mic *(specify Black, Beige or White)*.....**Call**
4061 Low Sensitivity Miniature Mic *(specify Black, Beige or White)*.....**Call**
4062 X-Low Sensitivity Miniature Mic *(specify Black, Beige or White)*.....**Call**
4065 Miniature Mic Headset, Microdot *(specify Black or Beige and Left or Right)*.....**Call**
4066 Miniature Mic Headset, Microdot *(specify Black or Beige and Left or Right)*.....**Call**
4071 Miniature Microphone with Presence Boost.....**Call**

IMK4061 Instrument Microphone Kit

Designed for close miking a wide variety of musical instruments, the IMK4061 consists of a 4061 Miniature Mic, and an XLR adapter, together with a selection of unique mounting accessories including holders for attachment to stringed instruments, universal surface mounts with double sided tape for use with guitars, wind instruments, drums and percussion, a magnet holder for attaching the mic to drum rims or the frames of pianos, and foam windscreens.

Single Clip *(DMM0001)*.....**17.95**
Single Pin, Black *(DMM0002-B)*.....**16.95**
Single Pin, White *(DMM0002-W)*.....**16.95**
Miniature Magnet, Black *(DMM0003-B)*..**27.95**
Miniature Magnet, White *(DMM0003-W)***27.95**
Lapel Single Clip *(DMM0004)*.....**10.95**
Single Tie Pin, 3pc. *(DMM0005)*.....**14.95**
Cleavage Mount*(CLV-1)*.....**14.95**
Miniature Windscreens, 5 pc. *(specify black, red, blue, yellow, green, white, beige)*.....**17.95**

Acoustic Modification Kits

Soft Boost



These kits consist of 5 protection grids. When fitted the grids introduces a 3dB soft boost around 8-20kHz for applications where the mics need to be placed on the head of the performer. The metal grid is anodized stainless steel.

DUA6001 (Black).....**56.95**
DUA6003 (Beige).....**56.95**
DUA6005 (White).....**56.95**

High Boost



These kits consist of 5 protection grids. When fitted the grids introduces a 10dB soft boost around 12kHz for applications where the mics need to be placed on the chest of the performer. The metal grid is anodized stainless steel.

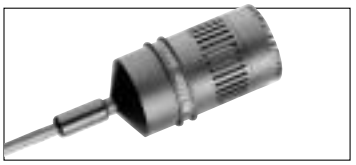
DUA6002 (Black).....**56.95**
DUA6004 (Beige).....**56.95**
DUA6006 (White).....**56.95**

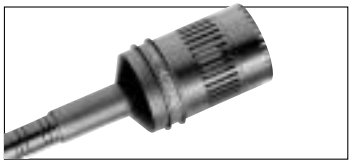
DPA

4021/4022/4023

Compact Cardioid Microphones







When microphones have to be out of sight, but maximum clarity and signal are essential, there is only one choice— DPA's high performance Compact Cardioid microphones. Phantom-powered condenser mics, the 4021, 4022 and 4023 use the same type of cartridge as the 4011 Standard, but are preamplified using a built-in, ultra-small, thick-film mounted FET-preamplifier. The result is exceptionally compact, high performance mics ideally suited for the the unobtrusive miking of all types of television, film, theater and live performances—as well ad for studio applications. In fact, their linear frequency response from 40Hz to 20kHz (±2dB) together with their ultra-small design makes them the perfect choice whenever high definition audio and insignificant size is paramount.

The 1.2″ **4021** has its cable (5m) side-mounted, which mechanically is perfect for situations where the mic needs to be mounted directly on string or wind instruments or on a stand.

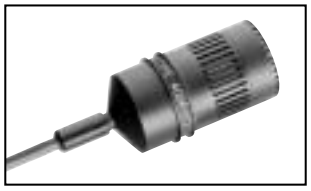
The 1.38″ **4022** has its cable (5m) rear-mounted, for situations where the microphone needs to be suspended from its own cable or mounted on a stand.

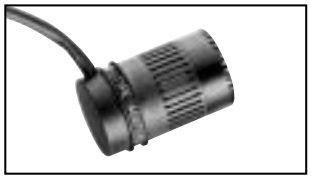
The 1.18″ **4023** has a standard LEMO-connector for simple and safe mounting and removal of the microphone and to provide maximum ruggedness when in use.

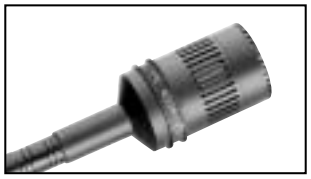
- For on-camera recordings with live audiences and artists, their matte black finish gives them a classic yet unobtrusive appearance.
- For sound reinforcement in concert halls, theaters and churches, they provide clean, natural sound, with a high gain-to-feedback ratio that limits the effect of reflections.
- In film and TV studios, they are excellent for off-camera recording of voice or Foley work.
- They include the SM4000 Suspension Mount and DUA0020 windscreen.

3521 Compact Cardioid Stereo Kit

The 3521 is a complete stereo kit with two 4021 Compact Cardioid Mics carefully matched within 2dB on frequency response, sensitivity and self-noise. Designed especially for low-profile mounting inside a piano, with the supplied XY/ORTF holder the kit is also exceptional as a compact stereo pair for applications such as overhead for drums, horn or strings sections, choirs, main or additional spot pair for small or big acoustic ensembles. In addition to the two 4021 mics, the kit includes two GM4000 Gooseneck Mounts, two MB4000 Magnet Bases and the CX04000 compact XY and ORTF holder.







4026/4027/4028

Compact Wide Cardioid Microphones

The 4026, 4027 and 4028 are phantom-powered compact condenser microphones with the same type of cartridge as the 4015 Standard Series wide cardioid microphone, but are preamplified using a built-in, ultra-small, thick-film mounted FET-preamplifier. Their linear frequency response together with a nice and subtle high frequency lift give that brilliant sparkle of purity, which will bring your sound to a higher level. And with unbelievable depth and precision, they will exceed your expectations on choirs, percussion, grand pianos, reed and brass instruments and much more. They include the SM4000 Suspension Mount and DUA0020 windscreen.

The ***4026*** has its cable (5m) rear-mounted, for situations where the microphone needs to be suspended from its own cable or mounted on a stand.

The ***4027*** has its cable (5m) side-mounted, which mechanically is perfect for situations where the microphone needs to be mounted directly on the musical instrument or on a stand.

The ***4028's*** 5-meter cable is attached with a standard LEMO-connector for easy mounting and removal of the microphone capsule, and for maximum ruggedness when in use.

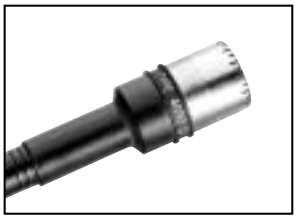
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EQUIPMENT LEASING AVAILABLE

4033/4036/4037 • 4051/4052/4053

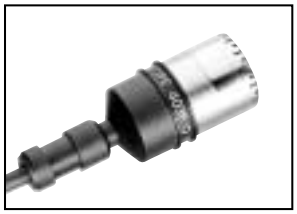
Compact Omni (12mm Cartridge) Microphones

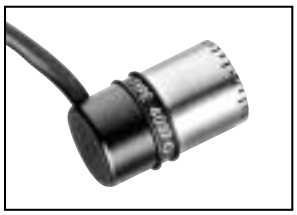
These phantom-powered omnidirectional condenser mics use the same type of mic cartridge as the 4007 Standard reference mic but are preamplified using a built-in, ultra-small, thick-film mounted FET-preamplifier. This 12mm (0.417˝) diameter cartridge gives them impressive SPL handling capabilities, as well as extended frequency range and excellent phase response. They are also insensitive to handling noise and humidity. Extremely linear frequency response from 20Hz to 40kHz (±2dB) and 144dB SPL capability makes them the professional choice in close miking situations with percussion, saxes, horns, wind instruments or grand piano. They include the DUA0030 windscreen.



Compact Omni (16mm Cartridge) Microphones

When mics need to be small, lightweight and inconspicuous, but quality is paramount, the 4051, 4052 and 4053's compact design and clean transparent sound make them the perfect choice for theater opera and concert halls where there is a live audience, as well as film applications—both in the studio or on location. They use the same type of mic cartridges as the 4006 Standard where their 16mm (0.63˝) cartridge provides impressive sensitivity characteristics, an extremely linear frequency response and a low-noise floor. This makes them ideally suited for all types of low SPL recordings while their wide dynamic range also enables recording at higher SPLs without worrying about clipping or distortion. Additionally, the mics are insensitive to handling noise, pop-noise and humidity, and exhibit no proximity effects. The 16mm Compact Omni's include the DD0251 Near Field Grid and DD0297 Diffuse Field Grid.





The ***4033***'s and ***4053***' 5-meter cable is attached with a standard LEMO-connector for easy mounting and removal of the microphone capsule, and for maximum ruggedness when in use.

The ***4036*** and ***4052's*** 5m cable is mounted behind the mic axis for situations where the mic needs to be suspended from its own cable, or concealed off camera. They can also be mounted on the supplied tie-pin clip.

The ***4037*** and ***4051*** have their 5m cable mounted at 90° to the microphone axis which mechanically is perfect for situations where the microphone needs to be mounted directly on the musical instrument.

3552 Compact Omni Stereo Kit



The 3552 is a complete stereo kit with two carefully matched 4052 Compact Omni Mics. Designed especially for low-profile mounting inside a piano, the supplied goosenecks can also be mounted directly on mic stands for various A-B stereo applications such as a full orchestra, choirs and other ambient recordings. In the A-B stereo principle, time differences will act as stereo clues to enable the listener to "capture the space" in the recording, and experience a vivid stereo image of the complete sound-field with a full bass response.

In addition to the two 4052 mics, the kit includes two (of each) DD0251 Near Field Silver Grids, DD0297 Black Diffuse Field Grids, UA0777 Nose Cones, GM4050 Gooseneck Mounts and MB4000 Magnet Bases.

Optional Accessories for Compact Mics



Mic Holder for Cello *(CH4000)* **249.95**

Mic Holder for Violin/Bass *(VH4000)***249.95**

Mic Holder for Sax/Horn *(SAX4000)*..**229.95**

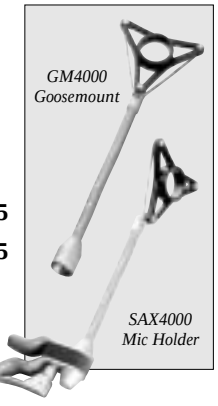
Goosemount 0.75˝ (19mm) *(GM4000)***119.95**

Goosemount 0.63˝ (16mm) *(GM4050)***119.95**

Suspension Mount *(SM4000)* **15.50**

Compact XY Holder *(CXY4000)* **79.95**

Extension for CXY4000 *(EXY4000)*....**24.95**

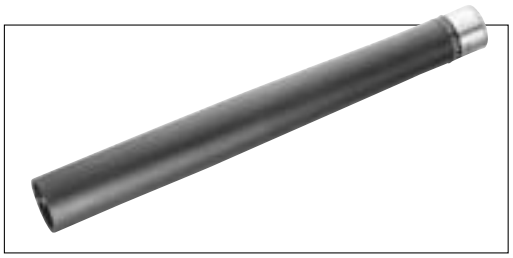


	Cardioid	Wide Cardioid	12mm Omni	16mm Omni
Sensitivity	7mV/Pa (at 1kHz)	Nominally 7mV/Pa (at 1kHz)	Nominally 8 mV/Pa (at 250Hz)	Nominally 30 mV/Pa (at 250Hz)
Equivalent noise level*	Typ. 20dB(A) re. 20 µPa	Typ. 20dB(A) re. 20µPa	Typ. 24dB(A) re. 20 µPa	Typ. 15dB(A) re. 20 µPa (max. 17dB(A))
Maximum SPL	145dB SPL peak	145dB SPL peak	144dB before clipping	135dB SPL peak
Output Impedance	<300 Ω	<300 Ω	<300 Ω	<300 Ω

**A-weighted*

4006

Low Noise Omnidirectional Microphone

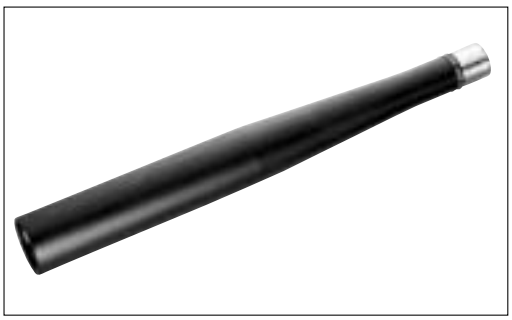


Using a German silver diaphragm and a pre-aged capsule, the 4006 is in use all over the world with professional studios, broadcasting facilities, and anywhere the utmost in reliability and repeatability are required. The 4006 exhibits no proximity effect and an inherent insensitivity to popping, making it ideal for close-miking vocals, strings, woodwind or plucked instruments.

The 4006 is an omnidirectional condenser microphone with an extremely linear frequency response from 20Hz to 20kHz, high sensitivity and a low noise-floor. The 4006 is phantom powered (P48), which together with the extended range of acoustic pressure equalizers and different protection grids turns the 4006 into an extremely cost effective solution. All in all the 4006 can be changed into 7 different microphone versions acoustically.

- Superior sensitivity characteristics, an extremely linear frequency response and a low noise floor make the 4006 the ideal choice in most miking applications.
- The capsule holds a 16mm (0.63-inch) diaphragm on a housing of German silver. During production capsules are pre-aged and rigorously tested, ensuring that the mic remains extremely stable over time, in all kinds of environments.
- Use at high SPL levels (up to 124dB) without worrying about clipping or distortion
- Inherently insensitive to popping and exhibiting no proximity effect, the 4006 is equally well-suited for close-miking vocals, strings, woodwind or plucked instruments.
- In addition to the silvered protection grid that gives it the greatest possible accuracy on-axis, the 4006 also includes a Diffuse-Field Grid (DD0297) as standard. The grid guarantees excellent linear frequency response in the diffuse field, and in addition, provides a high-frequency boost when miking closer to a sound source.
- Includes a windscreen, mic clip, calibration chart and a rugged case. The chart contains individually-measured on-axis frequency response, sensitivity and A-weighted equivalent noise level with other essential data.
- Optional accessories include a Nose Cone (UA0777) which gives a true omnidirectional response, even at higher frequencies, and Acoustic Pressure Equalization (APE) Kits that modify the directivity and frequency response characteristics of the mic without compromising audio integrity.

4007 *High SPL Omnidirectional Microphone*



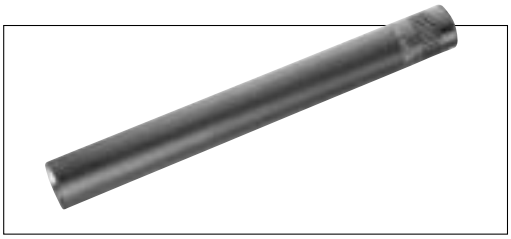
A high-SPL omnidirectional microphone, the 4007 is a phantom powered (P48) close miking specialist with an extremely linear on-axis frequency response from 20Hz to 40kHz (±2dB). This high-performance condenser microphone has an unbeatable dynamic range of 124dB from the noise floor to 1% THD and is capable of handling up to 155dB SPL peak before clipping occurs. The 4007 has through its total acoustical transparency and ultra-high SPL handling capability gained a reputation as being the first choice for picking up drums, percussion and brass instruments, whenever the natural and non-distorted result is paramount. Last, due to its linear qualities, the 4007 can also be used as reference mic in both microphone comparisons and acoustic test equipment.

- Specially designed to handle extremely high SPLs with optimum performance, the 4007's 12mm cartridge can handle peak sound pressure level fo 155dB before clipping.
- Small cartridge diameter also results in an extended on-axis frequency response, and excellent phase response and a high degree of omnidirectivity.
- The ultra-high pre-polarization voltage allows for a larger diaphragm-to-backplate distance, lowering distortion levels to an absolute minimum.
- Close-mike drums, percussion and brass with total accuracy for a clean, undistorted dynamic sound as never before.
- Includes windscreen, mic clip and a calibration chart.

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4011

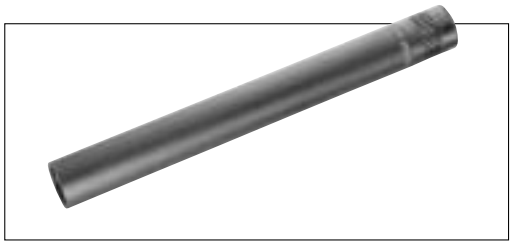
Cardioid Microphone



The 4011 is a cardioid condenser microphone with a linear frequency response from 40Hz to 20kHz (±2dB), totally flat off-axis qualities, so any leakage that occurs from off-axis sound sources is still a faithful reproduction of these sounds, though attenuated according to the true first order cardioid pick-up pattern.
The 4011 is phantom powered (P48), transformerless and has an extremely high SPL handling capability of 158dB before clipping occurs. These special qualities have made the 4011 one of the most popular microphones in the recordist's toolbox and the span of applications is limited only by your imagination.

- First order cardioid pick-up pattern offers superb separation between sound sources. Ideal for multi-track recording applications, classical recording or live situations.
- Able to handle extremely high SPLs before clipping (up to 158dB), the 4011 is equally well-suited for close-up work as well.
- Flat on-axis frequency response, together with an excellent phase response, delivers clean, uncolored sound that is totally faithful to the original.
- The off-axis response is similarly smooth so that any leakage is likewise an accurate reflection of the original.
- Second and third order distortion characteristics have been optimized, resulting in very low harmonic and intermodulation distortion. The 4006 is also unaffected by amplitude, phase and distortion problems.
- Includes windscreen, mic holder and individual calibration chart.

4015 *Wide Cardioid Microphone*



The 4015's polar pattern is a wide cardioid (also known as hemisphere), attenuating less to the sides and back than a traditional first order cardioid. It is designed with a linear on-axis frequency response from 40Hz (±2dB) to 20 kHz (+3/-1dB) at 60 cm, with a soft high frequency lift from 10 to 15 kHz of max. 3dB. The 4015 is the perfect fill-in when an omni is too ambient and a cardioid is too narrow.
The 4015 features the supreme accuracy, definition and resolution of the legendary 4011 to offer natural sounding recordings, however it has a richer bass response than the 4011 and will therefore serve you well both in close-up work as well as diffuse field tasks. The 4015 is phantom powered (P48), transformerless and can handle 158dB SPL peak before clipping occurs. Ideal for recording vocals, supporting sections of an orchestra, miking a conference panel, a percussion set-up or a grand piano.

4003/4004/4012/4016

130v Omnidirectional and Cardioid Microphones

DPA also offers the 130v high-voltage equivalents of the 4006, 4007, 4011 and 4015. Acoustically identical, these models differ in the powering system. The high-voltage system enables them to handle approximately 10dB higher SPL. The extra headroom, together with the excellent phase, transient and distortion performance offered by the 130v powering system, offers a true investment in versatility.
To power a 130v microphone, you need the optional HMA4000 Hi-Voltage Microphone Amplifier. A high quality power supply and mic amp, the HMA4000 has a transformerless, high-level, single-ended or electronically balanced signal path that preserves the integrity of the audio signal and delivers excellent phase and distortion characteristics. These powerful features make the 4003 (Omnidirectional), 4004 (Hi-SPL Omnidirectional), 4012 (Cardioid) and 4016 (Wide Cardioid) with the HMA4000 perfect for direct-to-track recordings.



3530-A

A complete A-B Stereo Kit with two factory-matched phantom-powered 4006 condenser mics, stereo boom and a complete range of acoustic modification accessories, in an attractive but sturdy carrying case with a foam insert. The hand selected 4006 mics are guaranteed to match within 1dB on both frequency range, sensitivity and self-noise. The ultimate portable solution for making sharply focused, realistically ambient stereo recordings—from acoustic soloist to full orchestra. In addition to two 4006 Omni Mics, the kit includes:

UA0836 Stereo Boom with Holders

2 UA0777 Nose Cones

2 DD0251 Near Field Grids, Silver

2 DD0297 Diffuse Field Grids, Black

2 L110C 110mm Acoustic Pressure Equalizer Cylinders

2 each **L30B** (30mm), **L40B** (40mm) and **L50B** (50mm) Acoustic Pressure Equalizer Balls (see next page for full description)



3511

The 3511 is a complete stereo kit for a broad range of recording applications using two 4011 Cardioid Microphones, carefully matched within 2dB on frequency response, sensitivity and self-noise. Several standard set-ups such as ORTF and XY are possible. Often used for piano recordings of jazz or pop with a placement closer to the hammers or as overheads for drums, the 4011s handle very high SPL and will therefore allow close placement as well as more distant. In addition to two 4011 Cardioid Mics, the kit includes:

2 **UA0897** Shock Mounts

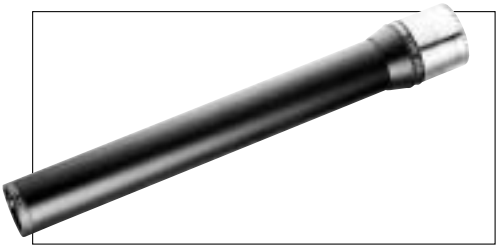
UA0836 Stereo Boom with Holders. Precision-crafted stereo boom for secure mounting of microphone pairs for A-B stereo recordings

DUA0019 Spacer for Stereo Boom. Allows XY or ORTF stereo set-ups on the boom

DUA0075 XY Windscreen

2 **UA0896** Windscreens for 19mm

4041 *Large Diaphragm Microphones*



The 4041 series are 1˝ omnidirectional microphones with the totally transparent MMP4000 preamp (except the 4041-SP) to deliver outstanding performance. They have extremely low self-noise of 6dB (typical) and 7dB (maximum). With a THD less 0.5% at 120dB SPL peak, they are a top choice with a dynamic range of 113dB. The specially treated stainless steel diaphragm and housing give the capsule unique immunity against ambient temperature differences. The backplate is polarized with 200v providing extremely high sensitivity and SPL handling of up to 144dB (134dB on the 4041-SP) before clipping. The frequency response is 10Hz to 20kHz with a 4-6 dB soft boost around 8kHz .

The 4041 mics have a unique modular design enabling the capsule to be unscrewed from the preamp and offering you the choice between a 130v or 48v phantom-powered solid-state preamplifier and a 130v tube technology preamp. The solid state is used for the most transparent and faithful reproduction, whereas the tube system adds a slight musical coloration to the recording.

4041-SP

The 4041-SP uses 48v phantom power. It offers the same superior sound quality as the high voltage versions, since the capsule is the same, with very slight changes in the specifications. Because it is part of the modular system, it is possible to upgrade to the 130v solid-state or tube preamp.

4041-S

The 4041-S uses the MMP4000-S solid-state preamp which offers a total linear and transparent signal path. The low noise output stage is driven as a class A unity gain impedance converter. HMA4000 Mic Amplifier supplies 130v power allowing extremely high audio signals without adding any distortion.

4041-T

The 4041-T uses the MMP4000-T mic preamp module and incorporates a pentode vacuum tube driven as a cathode follower in a class A unity gain output stage. The high-quality subminiature vacuum tube is heated by the 6v supplied with the HMA4000 Mic Amplifier.

4000 SERIES ACCESSORIES

UA0897 Shock Mount



Ultra-compact, robust design significantly reduces the amount of vibration, that reaches the microphone without having adverse acoustical effects. Unlike other shock mounts, the vibration reduction is clearly specified. It can be used stand mounted in the studio, or in hand-held boom applications. Fitted with a thread adapter, which allows usage on up to three of the most popular international standard threads used on microphone stands.

UA0836/UA0837 Stereo Booms



UA0836 Stereo Boom

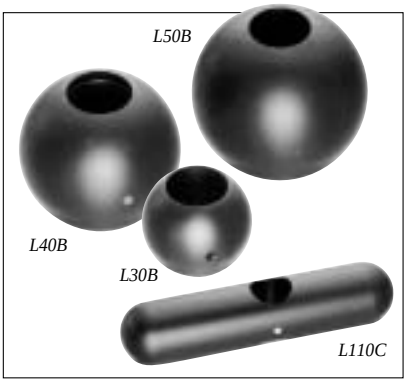
The UA0836 and UA0837 are precision-crafted microphone booms for secure mounting of microphone pairs for A-B stereo recordings. They have adjustment possibilities for both mic spacing and angle. The boom can be mounted on a stand, or suspended from wires, and has centimeter graduations on the boom and angle graduations on the microphone holders for quick and easy configuration of the recording set-up. (The UA0836 includes microphone holders, the UA0837 does not). The stereo boom can be fitted with the DUA0019 19mm spacer, which allows you to assemble an XY stereo or ORTF stereo setup with any two 4000 series mics.

UA0777 Nose Cone



Designed for the 4003,4006, 4051, 4052, and 4053 mics. The Nose Cone gives your mic a perfect omnidirectional response across the whole audio frequency range. It counteracts the directional characteristics that omnidirectional mics exhibit at higher frequencies, and guarantees an even tonal balance of sound arriving at all angles of incidence, but with a high-frequency boost on-axis. Especially suited for increasing the ambiance to your recordings, or for miking a variety of sound sources that are positioned around the microphone.

Acoustic Pressure Equalizers (for 4003 and 4006)



The 30mm sphere-shaped L30B gives a mild modification of the mic's on- and off-axis frequency responses, and a slight change in the direction-to-reverberation ratio. This gives recordings greater presence, nearness and clarity on-axis (especially around 4kHz). An on-axis boost, between 2 and 8kHz, and high-frequency roll-off off-axis, will sometimes help you capture the subtle audible details of both musical instruments and voices.

The L40B (40mm) offers a more noticeable on-axis boost between 2 and 8kHz, and a stronger off-axis attenuation than the L30B. This gives a marked increase in the directivity of your recordings (at 4kHz, the front-to-back sensitivity ratio is 8dB), and decreases the effects of reverberation and off-axis sources. Modify the spectral characteristic of a source by simply altering the angle of incidence to give a frequency boost, a flat response or an attenuation.

The L50B (50mm) offers a stronger on-axis boost and off-axis attenuation than the L30B and L40B. Its larger dimensions accentuate the on-axis boost at lower frequencies and allow off-axis sources to be strongly attenuated above 1kHz. This gives recordings a more intense sensation of source clarity and presence, a marked increase in brightness, and extend the effective reach of your mics.

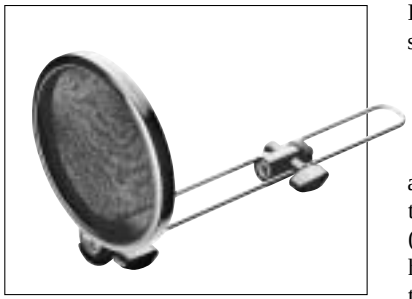
The 110mm cylinder-shaped L110C creates two distinct directional responses - or a combination of both - depending on the position of the sound source. Vertically oriented, there is maximum attenuation between 150° and 250°; horizontally oriented, there is a stronger high-frequency roll-off in the entire rear hemisphere. Plus, a strong on-axis boost between 3 and 4kHz, and powerful attenuation off-axis, allows excellent highlighting and close-miking source-separation without the problems of proximity boost.

UA0639 Mic Clip



Of a traditional design, but with additional modifications, that ensure a tighter and firmer grip on the microphone, while still enabling it to be secured and removed repeatedly with ease. Ideal for 4000 Series Standard Mics. Thread adapter allows use on three of the most popular international standard threads used on mic stands.

DUA0090 Pop-filter



Designed for the everyday use in the recording studio where acoustic performance and mechanical reliability is of the utmost importance. Elegant yet sturdy mechanical design allows a quick and easy positioning of the pop-screen in front of the mic. Thread adapter allows attachment to three of the most popular international standard threads used on mic stands. The (5.12-in) pop-screen does not attenuate the higher frequencies beyond the tolerance field of the microphone's frequency response and can be replaced with new filters if necessary. Can be used with any microphone.

QTC1

Omni-Directional Studio Microphone



The QTC1 provides flat, extended frequency response, superb impulse response, very high sensitivity and that great Earthworks sound with lower self noise than other omni models. It is an extraordinarily accurate microphone, typically within ±0.5dB from 20Hz to 20kHz. More importantly, the QTC1's impulse response is super clean. (Impulse information embedded within the tones of music is the detail needed to capture the experience of being there.) Clean impulse performance allows the QTC1 to transduce wave forms exactly. The QTC1 will provide a true and accurate representation of sonic events as they happened in the air, the experience of being there.

- The QTC1 is ideal for ambient miking and for quiet sources. It is also great for close miking.
- Capture the full size and natural texture of many difficult and complex sources.
- An excellent choice for all applications where natural, uncolored pick up of sound is the desired result.
- Excellent mic for Percussion, Piano, Guitar, Upright Bass, Strings, Vocals, Brass, Woodwinds, Location Recording of Classical, Jazz, Ethnic, Folk and Pop music. Also useful for Foley, ADR, sound effects and sampling.
- If you prefer transparency to coloration this is the microphone for you.



TC30MK *Omni-Directional Studio Microphone*

The TC30K is a cost effective general purpose omni which is accurate yet has a warm, life-like sound. It is optimized for fast, clean impulse response. Using a very small, very accurate omnidirectional capsule, a carefully crafted stainless steel body and innovative circuitry the TC30K delivers flat, Time Coherent response from 9Hz to 30kHz. It effortlessly handles explosive transients and simultaneously reveals subtle detail, delivering all of the sound as it happen in the air. By keeping time relationships coherent and transducing wave forms exactly, Earthworks TC30K accurately captures the experience of being there.

- The TC30K is superb for a surprising range of applications, capturing the full size and texture of many difficult sources ranging from moderate to extremely loud.
- Use for sources with significant impulse information, such as Percussion (Overheads, kick drum, hand drums, timbales & etc), Piano, Guitar (Acoustic and Electric), Upright Bass, Brass, Strings, recording live bands and many other applications . The TC30K is even useful for live sound in certain circumstances.
- The TC30K can be positioned very close to a source because it doesn't exhibit the bass proximity effect. This allows you to apply the inverse square law to great effect when isolation is desired and to capture the range of up close sounds which are simply unavailable to mics which get boomy when you get too close. And although it delivers startling bass performance there is almost no handling noise.



The TC30MK and QTC1 are both available as matched pairs. All Earthwork matched pairs include beautiful solid cherry boxes.

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TC30K

Time Coherent Omni Microphones

Lifelike realism through time accurate performance

Earthworks Omnis are flat from well below 20Hz to well above 20kHz. However, flat and extended frequency response is not the sole reason for Earthworks great sound. It results from the speed and accuracy of Earthworks Time Coherent impulse response. Accuracy in the time domain allows you to capture the performance as it sounded in the room.

The QTC1 and TC30K are optimized to deliver time accurate performance, they don't smear the time domain as much as most other microphones do. The impulse time settling time of 85 uses (TC30K) and 45 uses (QTC1) is faster than any other mic available. The QTC1 and TC30K are true and accurate but not in a cold, sterile way. They reveal all the warmth and texture of the source. They can be positioned close enough to a source to provide tremendous separation without boominess and to capture the incredible range of up close, detailed sounds which are simply unavailable to directional mics due to the bass proximity effect. And yet when used at a distance they capture the entire sound complete with a feel of being there. They also handle massive SPL (TC30K 150dB, QTC1 142dB), so it is unlikely that you will ever distort them even when close miking a kick drum. And despite having flat bass response to below 9Hz, Earthworks omnis have effectively no handling noise.

SRO *Sound Reinforcement Omni*

The SRO is a general purpose omni for live sound and recording. It handles huge SPL, has no proximity effect and no handling noise. It has fast clean impulse response and delivers accurate uncolored sound. Omnidirectional microphones are limited by feedback and leakage for many amplification applications. However, there are several types of applications for which they will deliver superior results.

For live sound try using the SRO to close mic loud stationary sources, like guitar amps, pianos and drums. Try it also for picking up on stage or room ambiance. This can be useful as a real world reference for in-ear monitors, as a reference for monitoring on-stage sound or to give the FOH (front of house) mixer a real world reference of what it sounds like in the room An affordable SRO Box Set (not matched but close) is available for recording gigs and rehearsals as well as for general studio use.



The Earthworks Philosophy

Earthworks believes that there is more to human hearing than the range of frequencies which are heard as tones. Most recordings don't live up to the sonic experience of being there with the live music. That is why Earthworks is convinced that audio equipment with extended frequency response and faster, cleaner time domain performance will yield more life-like results than with the slower more colored equipment of yesterday.

The time resolution of human hearing is 5 microseconds or better. This corresponds to frequencies beyond 80kHz. Most microphones smear frequencies in relation to time to such a degree that the relationships which existed in the air are audibly degraded. When you choose a microphone you get its time domain imprint on the track. Mics with poor time resolution will distort the signal beyond any practical recovery of the original.

Earthworks supports the concept that you can and should record acoustic events accurately enough to capture the sound as it happened in the room. This approach allows you to make your sonic coloration and texture choices later when you can consider them carefully and you still have recourse.

Traditional audio methodology is based on working around the limitations, personalities and quirks of the equipment—with Earthworks you never have to settle for less. That's because Earthworks is dedicated to an audio nirvana they call 'Perceptual Perfection.'

SR Series

Using carefully machined aircraft grade aluminum alloys plated with handsome hard coat finishes, the SR series mics are rugged and beautiful. They offer Earthworks being there quality in durable light-weight packages. They effortlessly handle high SPL transients while simultaneously revealing subtle details which lesser mics may miss. Bass response is flat at about 6-inches. Moving closer gives a little more warmth. They also have low handling noise. The cardioid models have an open easy to use pickup pattern with smooth uncolored rejection of off-axis arrivals. Every SR mic offers superb value and doubles for recording applications as well. Matched pairs of the SR77 and SR78 are available in a beautiful solid cherry box.

SR77

Precision Cardioid Studio Mic

The SR77 is a directional condenser microphone which delivers a natural, uncolored representation of the sound as it arrives at the front of the mic—maintains most of the being there quality of Earthwork's omni microphones. On-axis performance is smooth and open. Its sonic timbre is uniform across its entire front hemisphere of pick up. On-axis, its frequency response is flat from 30Hz to 30kHz at 15cm; closer there is a little more bass, at ambient distances there is a little less. The SR77's impulse response is fast and smooth. It can handle very hot signals. It has very low handling noise. The off-axis rejection is very thorough and uncolored.

The SR77 offers superior performance for a wide variety of applications, especially guitar, sax, flute, drums and voice. It is also rugged enough for high quality sound reinforcement (there are no peaks to encourage feedback and the rear rejection is unusually good), and powerful enough for broadcast and vocal production where its lifelike clarity and detail enhance communication through realism.



SR78

Precision Hypercardioid Studio Mic

An exceptionally directional condenser microphone, the SR78 delivers a natural uncolored representation of sounds as they arrive at the front of the mic. Its impulse response is fast and smooth so it maintains most of the being there quality of Earthwork omnis.

The SR78 cleanly picks up what it is aimed at while just as cleanly rejecting arrivals from off-axis. And yet, the sonic timbre is uniform across the entire frontal hemisphere. The SR78 effortlessly handles quite hot signals while simultaneously revealing subtle detail. In the studio or on the stage the SR78 can pick a real acoustic instrument, like guitar, bass, sax, flute, drums or voice, out of an ensemble and still sound real. It is usable for any situation where you want natural uncolored sound but need supernatural rejection.

Concert tapers love the SR78 when they want to avoid bad sounding rooms. It is a powerful tool for broadcast and vocal production applications where its lifelike clarity and detail will enhance intelligibility and communications through realism.

Its light weight, low handling noise and awesome ratio of open on-axis pick up to uncolored off-axis rejection make it an excellent choice for boom work.

1021/1022/1024

Single-, Two- and Four-Channel Zero Distortion Technology (ZDT) Preamps



The 1021(single channel), 1022 (dual channel) and 1024 (4-channel) have no whistles and bells and no hum or distortion either. They provide exceptionally accurate gain. In the time domain they are picture perfect. Their frequency response is flatter and more wideband than most people dream of. In the real world, what you can hear with your ears, they are perceptually perfect. The primary gain path (XLR input- stepped gain- XLR output) is entirely differential. The total distortion is unmeasureable by normal measurement equipment. They are cleaner than most wire, but they have gain. Lots of it!

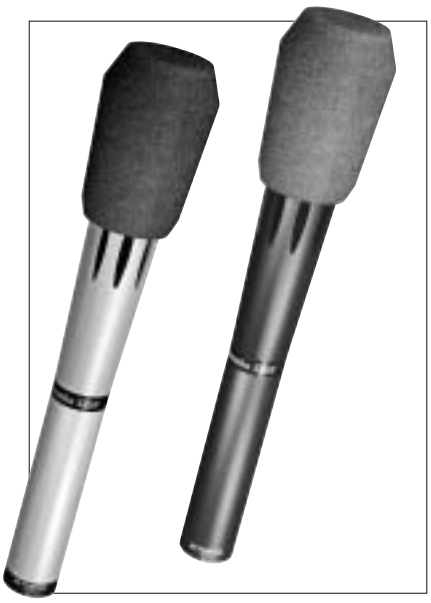
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EQUIPMENT LEASING AVAILABLE

SR Series

SR69

Cardioid Mic for Stage and Studio



Specs alone don't convey the quality of this mic's sound. When you hear it you will understand. As a vocal microphone the SR69 is spectacular. It accurately captures voice with just a bit of bass warmth available if you need it but no presence peak. It offers clean easy to use on-axis pick up and uncolored off-axis rejection. It is well protected from all popping of P's and has low handling noise. It sounds great. As with other Earthworks mics you will need to use less EQ because what is there is all there and it is done right.

Superb performance makes the SR69 a good choice for many other live sound and studio applications as well. It covers a wide range of real acoustic sources, like choir, guitars, horns, flute and especially, drums and other percussion. The SR69 will cleanly handle kick drum. For hand drums alone every stage and every studio will benefit from using the SR69.



SR68

Hypercardioid Mic for Stage and Studio

As a vocal mic the SR68 provides an accurate representation of voice with excellent gain before feedback. It is light weight and user friendly. It handles massive acoustic input levels and is well protected from popping Ps and wind noise. Up close the SR68 has punchy bottom end yet its handling noise is low. It is unsurpassed for choirs, as well as horns & woodwinds.

Removing the windscreen assembly reveals the SR68's alternate persona. The SR68 delivers the sound of any instrument with uncanny realism and separation. Frequency response chart shows that it has flatter response above 10kHz with the windscreen off. For acoustic guitar and other stringed instruments the SR68 sounds perfect at 6˝ yet rejects off-axis arrivals, especially bass. For drums the SR68 will spot a snare or high hat, kick drum or even overheads. It is also good for miking piano.

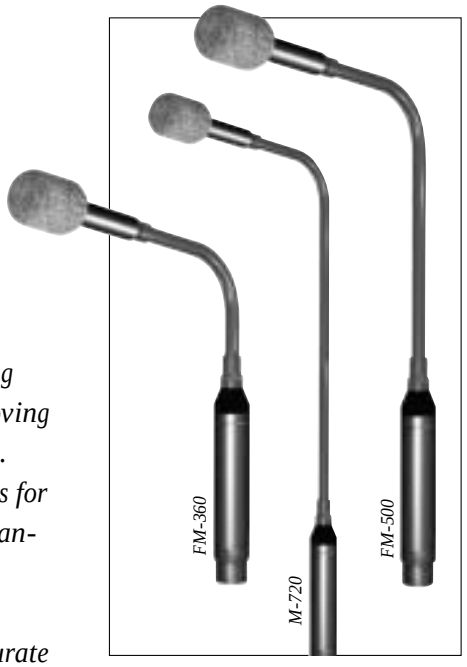
	QTC1	TC30MK	SRO	SR77	SR78	SR69	SR68
Frequency Response	4Hz to 40kHz[1]	9Hz to 25kHz[1]	10Hz to 20kHz[1]	30Hz to 30kHz[2]	30Hz to 30kHz[3]	50Hz to >30kHz[4]	50Hz to >30kHz[4]
Polar Pattern	○	○	○	ᔓ	ᔕ	ᔓ	ᔕ
Sensitivity	30mV/Pa	8mV/Pa	8mV/Pa	10mV/Pa	10mV/Pa	10mV/Pa	10mV/Pa
Power Requirements	48v, 10mA	48v, 10mA	48v, 10mA	48v, 10mA	48v, 10mA	48v, 10mA	48v, 10mA
Max Acoustic Input	142dB SPL	150dB SPL	150dB SPL	145dB SPL	145dB SPL	142dB SPL	142dB SPL
Min. Output Load	600Ω	600Ω	600Ω	600Ω	600Ω	600Ω	600Ω
Noise (A weighted)	22dB SPL	27dB SPL	27dB SPL	22dB SPL	22dB SPL	22dB SPL	22dB SPL
Dimensions/Weight	9 x .86˝; .5 lb.	9 x .86˝; .5 lb.	6.25 x .86˝; .2 lb.	9.3 x .86˝; .5 lb.	9.25 x .86˝; .3 lb.	9 x .86˝; .3 lb.	9 x .86˝; .3 lb.

[1]±1dB [2]±1.5dB @ 6˝ [3]±1.5dB @15cm [4]±2dB @ 6˝

EARTHWORKS

FLEX SERIES

Gooseneck Mics for Podiums, Choirs and Instruments



Flex Series microphones offer the same great sound and open easy to use cardioid pickup as the SR69 and SR71 but with a range of quiet, flexible neck lengths. These allow real time and real world adjustment of microphone positioning. Earthworks Flex mics are intended for podium use, hanging choir mics and for reaching into hard to reach places such as drums and hammer dulcimers. Their on-axis sound quality is natural and uniform all the way out to 90° and yet arrivals from off-axis are rejected smoothly with very little coloration. This is a strong plus for church, civic and corporate podium vocal applications because an orator moving about in front of the mic will be picked up more evenly than with most microphones. The sound quality of the Flex mics is every bit as good as Earthworks' other cardioids for applications ranging from acoustic guitar to choirs. The Flex mics will also cleanly handle very loud sources like drums and guitar amps. Flex mics are flat from 50Hz to 20kHz within ±2 dB at 6 inches. This level of accuracy is unusual in flexible microphones. The Flex mics facilitate intelligibility and communication through time accurate response. They are optimized for fast clean impulse response. This results in uncolored sound because the relationships between frequency and time are less smeared than with normal mics. They sound true to life and are user friendly.

M 720

The extra long M720 is the best hanging choir mic. It can also be used to pick up a group of mixed musicians with minimal visibility. Its flexible neck allows it to stand cobra-like on a desk top without a stand. For extra flexible reach the FM720 is superb.

FM 500

Half a meter is just the right length for most podium and church applications. The FM 500's swan-like neck can be adjusted with very little handling noise and it will stay where you put it. It is a good size to position unobtrusively into an instrumental ensemble or in front of a choir for sound reinforcement or recording.

FM 360

Don't let its diminutive size fool you. The FM 360 is the Napoleon of the bunch. It is powerfully equipped to totally dominate in situations which require a very low profile in order to get it right in tight places. The FM360 will effortlessly handle applications ranging from tall podiums for short people to piano, drums and guitar amplifiers.

SIGMA 6.2

40kHz Time Coherent Reference Monitors



Available in black or cherry, Earthwork's Sigma 6.2 system provides a reality check, delivering precision, time accurate sound-field reproduction. Accurate in time and frequency, their near perfect performance lets you hear all of the detail in your recordings. Their sonic accuracy is beyond question, the frequency response is flat and wideband, impulse response is fast and well behaved, and their step function chart looks like the theoretical text book ideal. The impedance vs. frequency chart shows them to have remarkably stable impedance which means the Sigma 6.2 is less subject to differences in amplifier quality.

Imaging is pristine. Their front to back depth of image is unparalleled. The way they sit in the room is more natural and less problematic than most loudspeaker boxes because care has been taken to avoid putting acoustic energy into the air beside and behind the cabinet. Their unique port design provides accurate bass reproduction and doubles as a convenient handle. The low frequencies are put into the room in the correct relationship to the rest of the signal. In conjunction with Earthworks microphones and preamps you will be able to hear the sound of the source through the loud-speaker for the very first time. The only way to hear it with better clarity is to be there when it is recorded.

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MEASUREMENT MICROPHONES

Earthworks has become the standard for affordable, reliable reference microphones. These mics are accurate in the time domain as well as in frequency. Most microphones smear time information to such a degree that the relationships which existed in the air are degraded beyond any practical recovery of the original. Earthworks measurement mics are optimized for fast clean impulse performance. Having such fast impulse response dictates that these mics must have accurate wideband frequency response. For measurement the M30 is a very cost effective reference for SMAART, MLSSA, TEF, RTA and all audio band measurements. It has very clean and fast impulse response. Typically the M30 is within 1dB from 20Hz to 20kHz. The M550 is astonishingly accurate over an even wider frequency range.



M30

The M30 is a very accurate microphone. It utilizes a very small, very accurate omni directional capsule, a carefully crafted stainless steel body and innovative circuitry to deliver time coherent response from 9Hz to 30kHz. 20 to 20kHz is not enough! This extended frequency response is essential for capturing sonic events accurately. In terms of the standard audio range the M30 is typically well within 1dB from 20Hz to 20kHz.

Operating the M30 is extremely simple. Using a good quality standard XLR mic cable connect the M30 to an active electronic microphone preamp which provides 48 volt phantom power with at least 10mA of current available. Adjust the preamp gain to the level required for your test equipment. The M30 has extremely low handling noise and does not require a shock mount. All windscreens effect the higher frequency performance, so they are to be avoided. When needed, a small ball type intended for a 1/4″ lavalier will do the job.

M30BX

The M30BX delivers the same calibrated 30kHz performance as the M30 reference mic—without the need for phantom power. It is powered by a single AA cell which will last 1,500 hours. An internal switch allows14dB of gain. If your sound card has any gain at all, and the sound level is even moderately loud, the output of the M30BX will be enough that you won't need an external preamp. The M30BX is a handy and powerful tool for laboratory, field and production measurements. It also sounds great for recording.

M550

Extraordinary impulse response. 30mV/Pa sensitivity. 4 Hz to 40 kHz ±1 dB, to 50 kHz +1/-3db. Laboratory grade accuracy for research beyond the brick wall. Omnidirectional. 48 volt phantom power.

	M30	M30BX	M550
Polar Pattern	Omnidirectional	Omnidirectional	Omnidirectional
Power Requirements	48V Phantom,10mA	AA 1.5V battery (2 months- 1,500 hours)	48V Phantom, 10mA
Min. Output Load	600 Ω at pin 2 and pin 3	600 Ω at pin 2 and pin 3	600 Ω between pins 2 and 3
Peak Output Voltage	3V into 1K Ω, 10V into 5K		3V into 1k Ω,10V into 5k ohm
Noise	27dB SPL equivalent (A weighted)	27dB SPL equivalent (A weighted)	22dB SPL equivalent (A weighted)
Peak Acoustic Input	150 db SPL with 5K capacitor coupled load	132dB SPL with 5k ohm load, or with 14dB gain 118dB SPL	142 dB SPL
Sensitivity	8 mV/Pa (-42 dBV/Pa)	8 mV/Pa (-42 dBV/Pa), or with 14dB gain 40mV/Pa (-28dbV/Pa)	30mV/Pa(-30.5 dBV/Pa)
Frequency Response on Axis	±1dB 9Hz to 27kHz, +1/-3 dB 5Hz to 30kHz	±1dB 9Hz to 27kHz, +1/-3dB 5Hz to 30kHz	+1/-3 dB 5 Hz to 50 kHz, ±1 dB 4 Hz to 40 kHz

N/DYM SERIES

Vocal and Instrument Microphones

15 years ago EV started a revolution in mic technology when they introduced a radical neodymium-based magnet structure to microphone design. For the first time, engineers, musicians and audio professionals had available to them mics with extended frequency response and increased power input, at very affordable prices. Now EV continues the revolution with VOB ("vocally optimized bass") technology. This cutting-edge innovation provides the performer with reduced resonant distortion at low frequencies. In other words, it efficiently controls the "proximity effect," or bass boost you hear in close-up microphone use, which tends to create "boomy" or "muddy" vocals, and replaces it with greater warmth and increased vocal clarity. The increased clarity permits greater working distances and ensures a clean, clear consistent sound that "cuts through the mix."

N/D167

Handheld Vocal Microphone

Your entry into the world of high-performance N/DYM microphones, the N/D167's innovative design harnesses the increased power of a neodymium based magnet design, allowing for dynamic, efficient microphone performance. With the power and clarity to "cut through the mix", the N/D167's acoustic response offers optimal performance in a live sound environment.

- Close proximity frequency response of 50Hz to 12kHz with a far range response of 100Hz to 12kHz
- Cardioid pattern for superior feedback rejection and acoustic isolation
- Neodymium-based magnet structure provides greater sensitivity and S/N ratio
- Comes complete with a stand adapter, zippered gig bag and EV's exclusive Warm Grip handle a for more comfortable feel

N/D267a and N/D267as

Handheld Vocal Microphones

Designed for live performances, the N/D267a and N/D267 are equipped with a controlled proximity effect that eliminates the boomy effects found in typical vocal microphones. Provides superior vocal clarity across all frequencies and in any environment. Also available in a switchable version (N/D267as).

- Close proximity frequency response is 45Hz to 12kHz with a far response of 100Hz to 12kHz.
- VOB technology provides tailored bass response for controlled "proximity effect" and exceptional vocal clarity
- Cardioid pattern for superior feedback rejection and acoustic isolation. Has an integral microphone element shock mount.
- Neodymium based magnet structure provides greater sensitivity and signal-to-noise ratio
- Comes complete with a stand adapter, zippered gig bag and EV's exclusive Warm Grip handle a for more comfortable feel.
- N/D267a and N/D267as are the same except the latter features an On/Off switch



N/D767a

Dynamic Vocal Microphone

The mic of choice among singers and engineers for concert and touring sound environments, the N/D767a provides outstanding vocal clarity in any application. VOB design provides the N/D767a with a tailored bass response for proximity control and excellent gain-before-feedback while supercardioid polar pattern provides superior feedback rejection and acoustic isolation. The mic also features multistage shock mounting for extremely low handling noise.

- Neodymium based magnet structure provides greater sensitivity and S/N ratio
- VOB technology provides tailored bass response for controlled "proximity effect" and exceptional vocal clarity
- Supercardioid pattern for superior feedback rejection and acoustic isolation
- Close frequency response is 25Hz to 22kHz, far response is 50Hz to 22kHz
- Multistage shock mount for unmatched low-handling noise
- Warm Grip handle for comfortable feel and lower handling noise
- Includes stand adapter and gig bag

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ONE HOUR FREE PARKING
AT 349 W. 34th STREET (with purchase of $100 or more)

N/DYM SERIES

N/D468 Instrument Microphone



Designed for miking horns, drums, and acoustic and electric guitar, the N/D468's unique pivoting head insures perfect placement providing smooth, natural sound while capturing the inherent sonic excitement produced by any instrument. Supercardioid pattern provides superior rejection and acoustic isolation in any application. Innovative EV design harnesses the increased power of a neodymium based magnet design, allowing a large-diameter voice coil (up to 50% larger than other mics) for dynamic, efficient microphone performance. N/DYM offers the power and clarity to "cut through the mix." Whatever your instrument application the N/D468 will be a top performer.

- Provides a smooth frequency response independent of proximity. Easily withstands the high sound pressure levels and sharp transients generated by the close miking of musical instruments
- Supercardioid pattern for exceptional acoustic isolation and feedback rejection
- Close range frequency response is 30Hz to 22kHz, far range is 60Hz to 22kHz
- Flexible mechanical design for optimum positioning. Rugged steel construction for exceptional durability
- Includes stand adapter and gig bag

N/D478 Instrument Microphone



The N/D478 is a high performance instrument microphone that offers outstanding vocal performance as well. Designed for miking of acoustical and electric instruments as well as vocals, the N/D478's acoustic response presents optimal performance in both a live sound or studio environment. A "must have" in any microphone mix.

- Neodymium based magnet provides greater sensitivity and signal to noise ratio
- Close frequency response is 45Hz to 15kHz, far response is 100Hz to 15kHz
- Cardioid pattern for superior feedback rejection and acoustic isolation
- VOB technology provides tailored bass response for controlled "proximity effect" and exceptional sound clarity
- Warm Grip handle for comfortable feel and lower handling noise
- Includes stand adapter and gig bag

N/D868 Kick Drum/Bass Drum Microphone



The N/D868 is truly a top performer in any application. Whether in a live sound or studio environment, the N/D868 is able to handle incredibly high sound pressure levels without distortion or overriding the input mixer. With a response specifically designed to "kick" the N/D868 can be used is "as is"—no additional EQ required.

- Designed specifically for kick drum applications in live or studio environments, sensitivity is optimized for the high sound pressure levels found in bass drum miking
- No additional EQ required
- Frequency response is 20Hz to 10kHz
- Rugged steel construction for exceptional durability
- Includes stand adapter and gig bag

	N/D167	N/D267a/as	N/D767	N/D468	N/D478	N/D868
Microphone Element Type	Dynamic	Dynamic	Dynamic	Dynamic	Dynamic	Dynamic
Polar Pattern	◡	◡	◡	◡	◡	◡
Frequency Response (Hz)	50 - 12k	45 - 15k	35 - 22k	30 - 22k	45 - 15k	20 - 10k
Output Level Sensitivity[1]	-52 dB	-52 dB	-51 dB	-51 dB	-52 dB	-52 dB
Open Circuit Voltage[2]	2.3 mV/Pascal	2.9 mV/Pascal	3.1 mV/Pascal	3.1 mV/Pascal	2.9 mV/Pascal	1.0 mV/Pascal
Input Impedance	600 Ω	300 Ω	300 Ω	150 Ω	300 Ω	150 Ω
Dimension	181 x 52mm	181 x 52mm	181 x 52mm	115 x 52 mm	181 x 52 mm	133 x 60 mm
Weight	238 g	238 g	260 g	190 g	247 g	295 g

[1]0 dB = 1mW/Pascal (@1000Hz) [2] at 1000Hz

COBALT SERIES

Professional Vocal and Instrument Mics



Designed for the working musician, the Cobalt series sets the standard for performance and value. A perfect blend of aesthetics, high performance and durability, Cobalt mics offer excellent vocal intelligibility, EV's radical neodymium-based magnet structure, exceptional feedback rejection at high SPLs, low handling noise, extended frequency and controlled proximity effect—professional-quality performance at a price that working musicians can afford. Whether you're a lead vocalist or miking an antique dobro, a vintage guitar amp, brass or percussion, Cobalt mics keep your tone "intact." Additionally, each includes a rugged carrying case and gig bag — perfect for the rigors of the road.

Co4 Instrument Mic

The Co4 yields outstanding performance when miking acoustic and electric instruments, as well as vocals. It uses a dynamic neodymium element with a cardioid pattern (excellent for stage or studio) providing a frequency response of 50Hz – 18kHz. The Co4 also features a unique "floating head" construction. The mic capsule is encased in a die cast zinc inner housing that is shock mounted inside an outer zinc housing. This gives it tremendous durability combined with superb shock isolation. Yet, the Co4 maintains a small profile that makes it perfect for tight miking applications. Excellent for stage or studio, the Co4's versatility make it a "must have" in any mic mix.

Co7 Crossroad Vocal Mic

The perfect choice for the performer who wants hassle free sound—whether on tour or getting serious in the studio, the Co7 performs without the feedback problems of competitive models. Built to the same standards as EV's N/DYM microphones, the Co7's low handling noise and versatility make it perfectly suited for critical acoustic settings as well. Features include a dynamic neodymium element, cardioid pattern for superior feedback rejection and a frequency response of 50Hz -18kHz.



Co5 Classic Vocal Microphone

Designed for the performer who prefers a careful balance of highs, midrange and bass, the Co5 offers exceptional vocal intelligibility with controlled proximity effect. Got the blues? Pull out your harp. With the Co5 you get clarity and power– when you want it. Dynamic cardioid mic with a neodymium magnet structure and a frequency response of 50Hz-15 kHz.

Co9 Premium Vocal Mic

A favorite among vocalists across musical genres. A slight bass roll-off and accentuated mid range make this a superb, world-class sound transducer. Features a dynamic neodymium element, cardioid pattern, and a frequency response of 50Hz -18kHz.

Co11 Condenser Vocal Mic

A high quality handheld condenser microphone with extended frequency response and wide dynamic range, the Co11 is ideal for anyone that appreciates the crisp high end and warm proximity effect that comes from a top condenser mic. Features a high compliance shock mount that makes handling noise "go away", and comes in a rugged die cast housing with the famous EV Memraflex dent resistant grille screen.

	Co4	Co5	Co7	Co9	Co11
Microphone Element Type	Dynamic	Dynamic	Dynamic	Dynamic	Self-Biased Condenser
Polar Pattern	◠	◠	◠	◠	◠
Frequency Response (Hz)	50 to 18k	50 to 18k	50 to 18k	50 to 18k	50 to 20k
Open Circuit Voltage (1000Hz)	2.2 mV/PA	2.8 mV/PA	3.2 mV/PA	3.2 mV/PA	4.0 mV/PA
Input Impedance	600 Ω	300 Ω	600 Ω	600 Ω	250 Ω
Included Accessories	Stand adapter, zippered vinyl carrying pouch and/or hard-shell case				
Dimensions	5.8 x 0.9″	6.7 x 2.08″	6.7 x 2.08″	6.7 x 2.08″	6.8 x 2.1″
Weight Net	12 oz.	10.8 oz.	11.7 oz.	11.8 oz.	10.3 oz.

LIVE INTERVIEW MICS

635A Dynamic Omnidirectional ENG/EFP Mic

Probably the most popular and well-known ENG/EFP microphone in the world, the 635A has survived being dropped, run over, thrown and even being shot while on location for film, TV and radio remotes. The 635A's uniform frequency response (80Hz-13kHz) is specially shaped for up-close vocals. Also available in black (635A/B) and with a Neodymium element (635ND-B).



635A • 635A/B

- Completely pop-free performance
- Four-stage pop and dust filter
- Internal effective shock absorber
- Includes stand adapter

635N/D-B

- N/DYM element offers higher output
- Acoustalloy diaphragm material for very smooth response over a wide frequency range
- Integral windscreen and blast filter

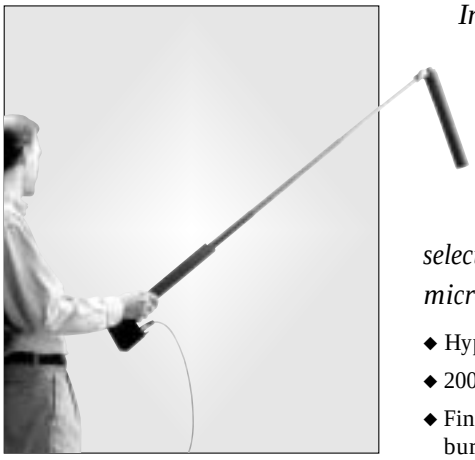


RE50/B • RE50N/D Omnidirectional Dynamic Mics

The RE50/B and RE50N/D are specially shock-isolated, omnidirectional, dynamic microphones created for the most exacting professional applications. Incorporating advanced shock-mounting techniques, they are designed particularly for hand-held broadcast and sound reinforcement use. Handling and clothing noise, even cord shock are isolated from the mic element so that mechanical induced noise is greatly reduced.

- No muddy lows when used near lips
- Acoustalloy diaphragm material for very smooth response over a wide frequency range
- Dyna-Damp for extremely effective handling noise isolation
- Four-stage pop filter prevents dust and magnetic particles from reaching the diaphragm
- RE50/B's attractive, non-reflecting, semi-gloss black finish is ideal for on-camera use
- Integral windscreen and blast filter provides excellent protection from wind noise, excessive sibilance, and "popping"
- They include a stand adapter, vinyl carrying pouch and hard shell case

ENG-618 Integrated Boom Pole/Shotgun Mic



Integrating a shotgun microphone and a boom pole with built-in headphone amplifier, the ENG-618 is ideal for ENG work. It features a perfectly balanced, ergonomically designed graphite fiber K-Tek boom pole that provides great strength with minimum weight and a permanently attached back electret shotgun microphone with superior selective pickup performance. The six foot boom pole and microphone collapses to a mere 18˝ for easy transport.





- Hypercardioid-line (shotgun), back-electret condenser microphone
- 200 Hz roll-off circuit lets you precisely tailor the sound quality
- Finely crafted, high-density K-Tek graphite fiber boom-pole with a burnished finish
- Low-battery LED indicator
- 50Hz to 20kHz on-axis frequency response that reduces handling noise
- Balanced XLR connector
- Supplied with windscreen and pouch
- Powered by external 12- 52v phantom power or by the 9v internal battery that is used to power the headphone amplifier

RE SERIES

RE20 and RE27N/D Dynamic Cardioid Microphones

The RE20 and RE27N/D are professional-quality dynamic microphones designed for broadcast, high-quality recording, and sound reinforcement applications. Exceptional sensitivity and wide frequency response, coupled with excellent transient response and inherently low noise of the dynamic transducer, all combine to ensure a superior S/N ratio— comparable to the finest condenser cardioid microphones. Plus they incorporate EV's Continuously Variable-D design making them virtually free of the "proximity effect" when used close. True cardioid mics, they offer the greatest rejection at 180° off axis - directly to the rear of the microphone.



RE20



RE27N/D

- Directional control is so effective that the frequency response is nearly independent of angular location of sound source, creating virtually no off-axis coloration yet providing greatest possible rejection of unwanted sounds
- An integral blast and wind filter covers each acoustic opening on the mics. In the studio or on stage, singers can "close talk" the microphone, singing with their lips almost touching the grille screen with no worry of "P-pop," breath sounds or excessive sibilance
- Part of the filter also shock mounts the internal mic element, reducing vibrations. This, along with a rugged exterior steel casing allows them to withstand all the rigors of professional use
- The RE20 has a "bass tilt down" switch, while the RE27N/D has two bass rolloff switches (sharp low-frequency cut at 200 Hz or a gentle rolloff) for situations where bass attenuation is required. A third switch is available for reducing the high-frequency treble boost
- Physically identical, the RE27N/D steps up with a neodymium alloy magnet and a reinforced diaphragm dome to offer increased sensitivity (up to 6 dB more output), undistorted output at high SPLs nd an extended high-frequency response. To further reduce noise, a highly effective hum-bucking coil is used to attenuate hum from lighting and other sources

The RE Series is the first choice of microphones for smooth and accurate sound reproduction. Its unique and famous performance make it a favorite in the broadcast, studio and professional touring business. RE20 and RE27N/D incorporated EV's Variable-D design to ensure true and accurate response across all frequencies without the upclose boominess associated with proximity effect. As a result, they have become the industry standard for radio studios worldwide. And when it comes to studio recording or concerts, the RE20, RE200, RE510 and RE1000 condenser mics provide you with the tools you need to get the sound you're after when recording vocals, guitars, drums or brass.

RE16 Dynamic Supercardioid Mic

The RE16 is designed for professionals who need an exceptional super-cardioid mic for ENG interviews, voice-overs and music production recording. It has an excellent, built-in heavy duty pop/blast filter, switchable bass roll-off, hum-bucking coil and a Memraflex grille screen that retains its shape if dropped.



- Utilizing the Variable-D design the RE16 features a directional characteristic independent of frequency. The result is a mic that generates little off-axis coloration while providing the greatest rejection of unwanted sounds
- Transducer is a nearly solid mechanical structure that is highly resistant to damage from mechanical shock
- Blast filter permits hand-held and out-door use without "P-popping" or excessive wind noise
- Easily operated "bass-tilt" switch corrects spectrum balance for boom use and other longer reach situations.
- Humbuck coil and screw-machined steel outer casing provide 25 dB of hum rejection
- Memraflex grille screen and carefully designed steel outer case provide additional protection. Finish is non-reflecting fawn beige micromatte

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RE SERIES

RE200 Condenser Instrument Mic



The RE200 is a cardioid condenser instrument microphone designed for the exacting acoustic reproduction of choirs, strings, percussion and brass instruments in live performance and sound recordings. It is designed to be used in professional remote recordings, sound reinforcement systems, sound studios and motion picture/TV scoring stages. The RE200's frequency response has a continuous presence rise to enhance the sound quality. The excellent transient response and smooth frequency response is gently lifted above 1000 Hz, lending an open, transparent detail to the recorded sound.

- Ultra-thin, gold-laminated, and ultra-low mass diaphragm
- AcoustiDYM shock-mount system for mechanical isolation
- Runs on 12-52v DC phantom power
- Includes stand adapter and external foam windscreen
- Small, light and rugged, it can be used for variety of applications without being obtrusive.

RE510 Hand Held Condenser Microphone



The RE510 is a professional quality hand-held condenser microphone designed especially for vocalists. It features a 5/8˝ diameter capsule, advanced electronic design, and best of all, a sound that will enhance any vocal performance. The high frequency response provides just the right amount of definition and "air" without the stridency so common in other condenser vocal microphones. The wide bandwidth, high SPL handling capability, and low frequency selector switch of the RE510 also make it a versatile instrument microphone for most any application.

- 5/8˝ diameter self biased condenser capsule offers the warmth of a larger capsule design without sacrificing off-axis performance or transient response.
- Warm Grip handle for more comfortable feel.
- Low-frequency roll-off switch enables tailoring of low-frequency response
- Multistage pop filter eliminates breath pop noises
- Modern high-speed electronic design for extremely low distortion, and outstanding transient response.
- High current differential output stage can drive long microphone cables without loss of performance.
- Supercardioid polar pattern for superior feedback rejection and acoustic isolation.

RE1000 Supercardioid Condenser Microphone

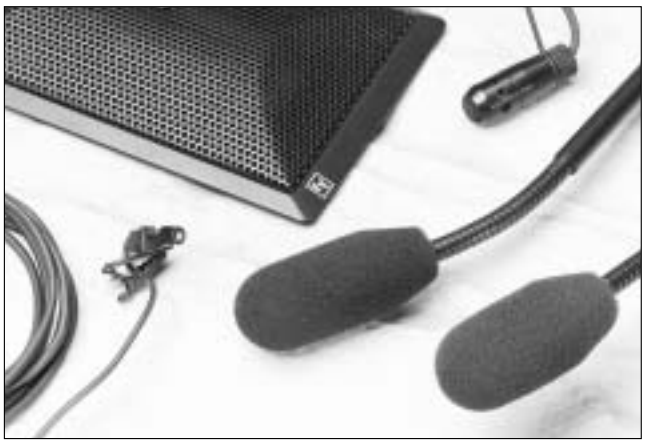


A condenser studio microphone, the RE1000 provides flat frequency response, low self-noise and an externally biased transducer. It is designed with rugged and reliable EV durability while providing excellent performance. Designed for high-quality recording and sound reinforcement, the combination of very low self-noise, excellent transient response and smooth frequency response makes the RE1000 an ideal choice for use in the most demanding of situations.

- Transducer is a short-D, high-efficiency unit, with the high output level of a large diaphragm and the tighter polar pattern and better transient response of a smaller diaphragm.
- The diaphragm itself is an ultra-thin, gold-laminate, environmentally stabilized, low-mass design.
- 12dB-per-octave low-frequency roll-off at 130 Hz
- The output electronics provide no voltage gain to ensure the lowest self-noise.
- Includes a hard-shell case for storage and transport.
- Internal two-stage pop filter reduces explosive breath blasts and wind noise

RE90 SERIES

Architectural Microphone Series



Using an EV RE90 Series architectural microphone is easy and enjoyable for everyone. The presenter will sound his or her natural best, without fear of feedback, cut-out, noise or distortion – or of sounding like a different person when switching microphones. These miniature electret condenser microphones are available in podium, hanging, lapel and boundary models. Fixed. Portable. Omnidirectional. Unidirectional. An EV family of microphones suitable for many applications and venues. Each with the same great sensitivity to sound pick-up and a remarkable clarity and consistency. Even when shifting from one RE90 to another — they have the same frequency response characteristics. No need to remix. It's all in the family. But there is more. Streamlined, they blend seamlessly into the architecture. For easy installation and use, the power module is integrated within each microphone — no need to install a separate unit. Yet, the lapel mic is smaller, the hanging mics more shapely, the podium mics more slender, and the boundary mics lower in profile, than others on the market.

RE90P-12/RE90P-18

Unidirectional Podium Microphone



RE90P-12

Acoustically designed for high-quality sound reinforcement and public address applications, this slim-profile podium microphone picks up and reproduces sound naturally for either distant or close-up delivery. The gooseneck, 12″ (RE90P-12) or 18″ (RE90P-18), can be precisely positioned without excessive twisting and turning. Small in diameter, it has two supple joints and a rigid center tube. The electronics module is housed in the gooseneck base, which makes installation and service quick and easy. The RE90P mounts to a lectern, pulpit, or podium equipped with the proper receptacle. Or install with the optional flange-mount and shock-mount kits. Unobtrusive, however it is used.

- Small, streamlined appearance
- Quick-connect to female XLR-type, 3-pin receptacle
- Permanent installation with optional FMK flange-mount kit and CPSM shock-mount kit
- Choice of 12″ or 18″ gooseneck
- Microphone can be exactly positioned without unsightly twisting and turning
- Uniform cardioid polar pattern
- Steel construction reduces noise from magnetic field and RFI

RE90P-12 12″ gooseneck **RE90P-18** 18″ gooseneck

RE90H/RE90HW

Unidirectional Hanging Microphone



RE90H

Ideal for choir, instrumental and vocal groups, and live theatre, this hanging microphone is so sensitive it catch-es the smallest sigh, the softest note. So faithful to the performance, so natural in delivery, everyone in the audience can hear every word, experience every nuance. It's ideally suited for distant sound pick-up, with a precisely controlled polar response designed to suppress feedback without coloring the sound. The RE90H/RE90HW features an exclusive braided, shielded cable which prevents the microphone from rotating, keeping it firmly in place, even with wide swings in temperature or humidity. There's no need for awkward "fish-line" tethering.

- Non-reflecting black (RE90H) or low-gloss white (RE90HW) with matching cable
- Shielded pre-amp is highly-resistant to electrical noise and radio frequency interference
- High gain electronics for added reach
- 25′ braided, shielded cable maintains microphone's position (no rotation)
- Cardioid polar pattern
- Three-pin male XLR-type output connector

RE90H non-reflecting black **RE90HW** low-gloss white

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RE90 SERIES

RE90L

Ultra Miniature Omnidirectional Lapel Microphone

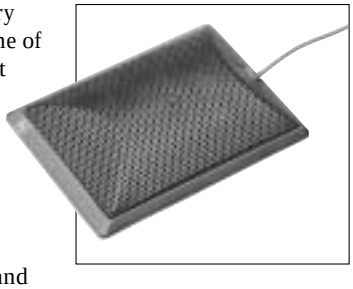


Hearing is believing (you have to look closely to see it). Ultraminiature and ultra lightweight, anyone wearing the RE90L can easily forget he or she is wired. Designed for sound reinforcement in applications such as houses of worship, schools and business presentations, this smallest, lightest of all microphones on the market, has the highest quality sound characteristics. It is well-balanced throughout its full range. Omnidirectional output is uniform and natural, no matter where the microphone is placed. The compact electronics housing is made of a high-strength polycarbonate that is very lightweight but very durable. A side clip attaches the unit to belt, waistband or pocket.

- Ultraminiature and lightweight (the smallest available).
- Omnidirectional
- Tiny but sturdy 6´ shielded cable
- Alligator and tie-tac clips included
- Lightweight, clip-on electronics pack

RE90B/RE90BW

Boundary Microphone



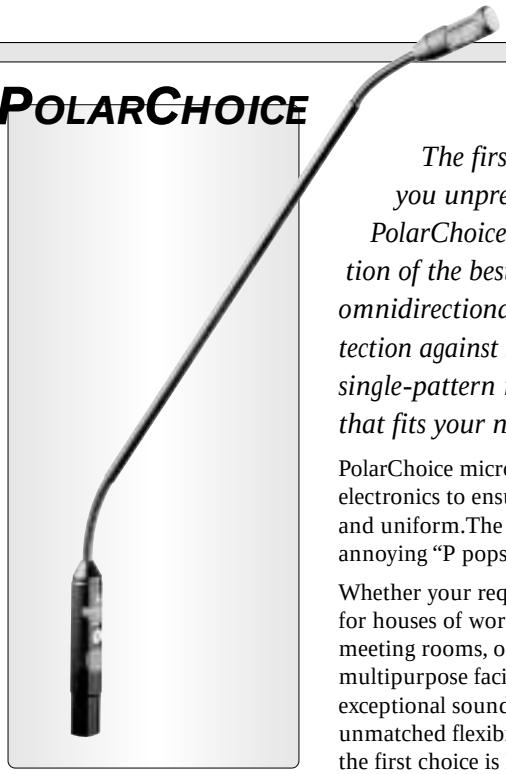
With the lowest profile of any boundary microphone, the RE90B/RE90BW is one of the smallest available. Unidirectional, it picks up every word and delivers it fully and naturally, without the distraction of ambient noise.

The RE90B is specifically designed for surface-mounted applications such as distance learning centers, video- or teleconferencing systems, classrooms, and houses of worship. It can be placed on a surface for temporary use, routing its miniature cable from the rear where it's barely visible. Or installed permanently, with the hidden cable routed through a hole in the surface.

The mic is equipped with internal padding, printed circuit board, fiber washers for the mounting screws, and a rubber, non-slip pad covering the entire base. These elements combine to give you the advantage of extremely low surface-to-microphone vibration. And built-in equalization provides a fuller, more natural sound than other typical boundary microphones.

- Non-reflecting black (RE90B)or low gloss white (RE90BW)
- Heavy-duty zinc die-cast body and strong, perforated steel screen
- Low profile
- Half-cardioid polar response promotes excellent gain

POLARCHOICE



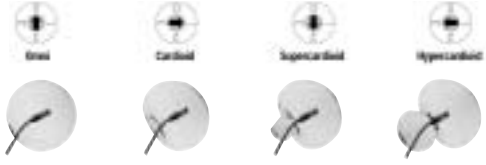
All-in-One Podium Microphone

The first podium microphone with four polar patterns, the PolarChoice gives you unprecedented performance and flexibility. It's like four microphones in one! PolarChoice offers the option of four different pickup patterns, allowing the selection of the best pattern for the application.The ability to conveniently select from omnidirectional, cardioid, supercardioid or hypercardioid assures the maximum protection against system feedback and ambient background noise. No need to inventory single-pattern microphones or expensive interchangeable heads. Just select the pattern that fits your needs with a convenient and easy turn of a rotary switch.

PolarChoice microphones feature EV's exclusive dual-mounted condenser transducer and sophisticated electronics to ensure that regardless of the pattern selected, the response will remain smooth, accurate and uniform.The use of EV's patented triple-layer Acoustifoam windscreen minimizes the level of annoying "P pops" commonly associated with other podium microphones.

Whether your requirement is for houses of worship, hotels, meeting rooms, or other multipurpose facilities where exceptional sound quality and unmatched flexibility are critical, the first choice is PolarChoice.

Selector switch at different positions



NEUMANN

M 147 TUBE

Cardioid Vacuum Tube Studio Microphone



Introduced three years ago, the M 147 Tube is an affordable top-of-the-line studio cardioid microphone designed for project, home, and professional recording studios. At the heart of the M 147 is the K 47 capsule— made famous over 50 years ago in the legendary U 47 microphone and then later by the U 47 FET. Employing the K 47 pressure-gradient capsule ensures that the M 147 delivers the same warmth and clarity produced by its legendary predecessors.

Exceeding traditional designs, Neumann uses a modern tube complemented by a transformerless output circuit. This innovative combination of current tube technology with the most advanced solid-state circuitry results is a more controlled, uncolored audio signal from capsule to output. Bass response extends quite low while avoiding phase problems or low level signal losses typical of transformers. Similar to all Neumann mics, the response characteristics are achieved acoustically. The mic also maintains excellent impulse response and faithfully reproduces transient signals found in music and speech.

The M 147 has a fixed cardioid pattern that becomes more directional as frequency increases — giving the microphone more of a supercardioid pattern in the high frequency range. The M 147 also features extremely low self noise for a tube microphone (12 dB-A) and handles high pressure acoustic signals up to 134 dB SPL without distortion. An excellent microphone for acoustic guitar or bass, horns, instrument amps, and any kind of vocals, the M 147 like all Neumann tube mics, comes with an elegant satin nickel finish.

FEATURES

Polar Pattern

- The M 147 has a cardioid characteristic, leaning more toward super-cardioid due to its distinctive capsule design. At higher frequencies the pattern becomes more directional. This is very similar to the U 47 and U 47 FET.

Acoustic Features

- The M 147 is addressed from the side where the body has the diamond-shaped Neumann logo. The black color identifies tube mics.
- The K 47 capsule (equivalent to the one in the U 47) is the deciding factor in determining the sound characteristic. The M 147 is flat from 60Hz to2 kHz. It has a small, wide 1 dB peak from 3 to 5 kHz, and a 2dB peak at about 10 kHz.
- Reproduces extremely low frequency signals without distortion.
- Headgrille design protects the capsule effectively against popping and wind noise.
- Capsule and the entire circuitry are shock mounted to prevent structure-borne noise.

Common Applications

- Use as announcer's mic for broadcasting and voice overs or as a vocalist mic where its warm and yet transparent sound gives volume and presence. Use as a spot mic for close miking of solo instruments especially strings, wind instruments, and piano.

Electronics

- Compared to other tube mics the M 147 is distinguished by its extremely low self-noise level of only 12 dB-A / 24 dB CCIR weighted.
- Combines a specifically selected vacuum tube (triode) with modern circuitry. This technique takes full advantage of the special transfer characteristics of the tube and passes the processed audio signal of the capsule to the microphone output, without any coloration or unwanted side effects.
- The tube amplifies the capsule's signal by approximately 10 dB, thus preventing any possible influences from subsequent electronics. The M 147 delivers a high output voltage, and can feed mic cables up to 300m without signal degradation.

N 149 A Power Supply

- To maintain optimum anode current and heater voltage for the tube regardless of cable length, a built-in sensor lead monitors voltage levels. If it detects any voltage drop at the heater, the N149 A power supply compensates for the difference. The N149 A also employs a slow start-up to maximize the life of the tube.

Supplied Accessories

- Includes N149 A universal AC power supply (100-240v, 50-60 Hz) and 30-ft. cable, SG-1 metal swivel stand adapter, and attractive aluminum case.

Optional Accessories

- To reject unwanted low-frequency interference from structure-borne or wind noise the invincible EA 1 elastic suspension and the WS 87 windscreen are available as optional accessories.
- For close miking of vocals the optional PS 30 pop screen is also available.

M 149 TUBE

Multi-Pattern Tube Studio Microphone



Like the M 147 Tube, the M 149 Tube is the perfect combination of vintage sound and modern technology. A transformerless tube microphone optimized to have the sound "character" of the classic U 47 and the multi-pattern capability of the M 49 microphones, the M 149 gives you the unmistakable tube warmth that audiences love, while at the same time delivering the clarity and precision of a modern capsule. The ideal choice for a wide range of recording situations, the M 149 has nine polar patterns to choose from as well as a seven-position high-pass filter.

Same features as the M 147 Tube EXCEPT—



Polar Patterns

- Easily switch between 9 directional patterns: Omnidirectional, wide-angle-cardioid, cardioid, hypercardioid, figure-8, and one additional intermediate pattern between each major position.

Acoustic Features

- The M 149 is addressed from the front, marked with the Neumann logo.
- Large headgrille surrounds the capsule. Acoustically very open, it therefore increases the sonic realism. In addition, the wedge-shaped grille reduces internal reflections.
- The M 149 is distinguished by its extremely low self-noise level of only 13 dB-A / 28 dB CCIR weighted (in cardioid).

Filter

- Seven-position slide switch on the back of the mic selects a high-pass filter, advancing in half-octave steps between 20 Hz and 160 Hz (-3dB). This filter suppresses rumble from air-conditioning and is an effective tool to control proximity effect when the mic is used up close.

Applications

- With nine polar patterns to choose from, this mic is ideal for a wide range of recording situations. As its ancestors, the M 149 Tube is superb as a vocalist microphone, for piano, strings and orchestral recording.

Supplied Accessories

- Includes N 149 universal AC power supply (100-240v, 50-60 Hz), EA 170 elastic suspension, dust cover, and a wooden box for protecting and storing the mic safely. Optional PS30 pop screen is available when close miking vocals.

75 Years of Microphone Greatness

It all began in 1928, when George Neumann split with Eugen Reiz, with whom he had collaborated to design the historic “Reisz Microphone”, and founded his own firm, George Neumann & Co., in Berlin, Germany. That year saw the company’s first product, and what is now recognized as the first commercial condenser microphone: the CMV3. With its omni capsule (and later with directional capsules) and innovated tube amplifier, this ground breaking mic is known as the “Neumann Bottle” and is still today sought after for its sound.

In the late 1940s, Neumann began the period in which it would become known the world over for making the benchmark recording studio mic. The long reign of the ribbon microphone came to an end with the widespread acceptance of Neumann’s U 47, introduced in 1949. This mic was the first with variable polar patterns (omni and cardioid) and has remained one of the most famous microphones ever produced. In fact, the majority of the Beatles vocals were recorded with the U 47.

Other legends from Neumann’s great history include the M 49 and its omni relative, the M 50, both introduced in the early 1950s. Like the U 47, the M 49 used the M7 capsule, while the M 50 used a unique pressure transducer with a pure metal membrane. Practically all “Decca Tree” orchestra recordings from the 1950s to the present rely on the M 50 or its new version, the M 150 Tube, for their unique polar and frequency response.

Neumann’s small diaphragm mics have earned their keep in innumerable studios for their clear sound, excellent off-axis characteristics, and tremendous value. The KM 54, introduced as a small microphone for television and broadcast use in the mid 1950s, remains in service today as a premier mic for acoustic instruments. The KM 84, introduced in 1966 as the first phantom 48V powered condenser mic, was an enormously popular mic right up through its discontinuation in 1988 and is already considered a “classic”. Today, scarcely a studio would consider itself serious without at least one Neumann mic on hand, often the U 87, which was introduced in its original form in 1967. The current product line, including such microphones as the M 149 Tube, TLM 103, KMS 105 and KM 184, pays homage to the company's rich history while pushing the limits of what is possible in microphone design and application.

NEUMANN

M 150 TUBE

Omnidirectional Tube Studio Microphone



Since the 1950s, the Neumann M 50 has been heralded as the ideal microphone for orchestral recording and string scoring. With its phenomenal transient response and unique directional characteristic, this classic mic has endeared many fans, both in the control room and on the soundstage. Now using modern materials and technology Neumann has recreated the beauty of the vintage M 50.

The M 150 Tube delivers the same unique omni-directional characteristics—however, that's where the similarity ends. With low self-noise, a Titanium membrane and capsule, transformerless tube amplifier and sophisticated power supply, the M 150 Tube is not a re-issue but an entirely new microphone in its own right. Destined to join the ranks of Neumann's classic microphones, the M 150 Tube is also excellent for full-bodied vocals and ensemble stereo recording. Ideal for any stereo, 5.1 or 7.1 surround recording, particularly DECCA Tree technique.

FEATURES

- Much lauded for its desirable mechanical characteristics, titanium was, until recently, difficult to machine as well as to obtain the quality necessary for use in microphones. Now with the necessary materials at hand, Neumann is using this lightweight element for the M 150's thin, 12mm capsule to deliver low self noise, a smooth extended frequency response, and unprecedented transient response.
- The headgrill is formed just like the original M 50, and the capsule mounting employs the same 40mm sphere that rendered the M 50's famous pick-up pattern: circular at low frequencies and increasingly narrow up the spectrum. This means, at the lowest frequencies, the system is a pure omni pressure transducer with a perfectly circular polar pattern. But in the mid- and upper frequencies, pickup becomes more narrow.
- Wide 119 dB dynamic range allows reproduction of the full musical expression, without restraint. Self noise is low, only 15 dB-A so more gain can be used without risk of adding noise to the final product.
- The transformerless output circuit of this microphone allows for extremely fine reproduction of small signals and low frequency information. Also, long cable runs can be used with no loss of signal quality.

Includes N 149V Vintage AC power supply (100-240v, 50-60 Hz), EA 170 elastic suspension and attractive aluminum case.

"Decca Tree" Recording

The specialized Decca Tree configuration is characterized by having three omnidirectional microphones in a "T" shaped setup. Two of the microphones are positioned about two meters apart. The third microphone is positioned between the first two, but about 1.5 meters forward (closer to the source) of them. This configuration is often used for orchestral recordings and film scoring due to its natural sound with good separation. It is useful in film because the image doesn't usually cause problems with Dolby or other surround processes. In most cases a trio of Neumann M50 microphones (or now, the new M 150 Tube) are used due to their unique directional and frequency response characteristics and smooth sound.

	M 147	M 149	M 150
Microphone Element Type	Pressure gradient transducer	Pressure gradient transducer	DC-polarized condenser
Polar Pattern	◡	○, ◡, ◡, ∞	Modified omni
Frequency Range	20 Hz - 20 kHz	20 Hz - 20 kHz	20 Hz - 20 kHz
Sensitivity at 1 kHz into 1 kohm	20 mV/Pa	34/47/62 mV/Pa*	20 mV/Pa
Rated impedance	50 Ω	50 Ω	50 Ω
Rated load impedance	1000 Ω	1000 Ω	1000 Ω
Equivalent SPL CCIR 468-3	24 dB	28/25/23 dB*	28 dB
Equivalent SPL DIN/IEC 651	12 dB-A	16/13/11 dB-A*	15 dB-A
S/N ratio CCIR 468-3	70 dB	66/69/71 dB*	66 dB
S/N ratio DIN/IEC 651	82 dB	78/81/83 dB*	15 dB-A
Maximum SPL THD < 0.5 %	114 dB	120 dB	114 dB
Maximum SPL THD < 5 %	134 dB	136 dB	134 dB
Maximum output voltage	8 dBu	8 dBu	8 dBu
Diameter	5.6 x 2.24″	7.9 x 2.76″	6.5 x 3.07″
Weight	16.2 oz.	19.8 oz.	28.2 oz.

** Omnidirectional/cardioid/figure-8*

TLM 103

Large Diaphragm Microphone



With a universal cardioid pattern, straightforward handling, extremely low self-noise level, and an affordable price, the large diaphragm TLM 103 is designed for applications that requires the utmost in sound quality—but on a limited budget. Utilizing the tried and true transformerless circuit found in numerous Neumann microphones, the TLM 103 achieves incredibly low self-noise and the highest sound pressure level transmission. The K 103 capsule, derived from the used in the U 87, has a cardioid pattern, is acoustically well-balanced and provides extraordinary attenuation of signals from the rear. The TLM 103 is available in satin nickel and matte black. Includes an SG 1 metal swivel mount and a wooden jeweler's box.

FEATURES

Polar Pattern

- Equipped with a large diaphragm capsule with cardioid pattern. By focusing on this pattern – used in most recording situations– the attenuation of unwanted rear sound is optimized. Off-axis sounds are rendered naturally while isolation is increased. This also leads to a high feedback suppression when the mic is used in live situations or where loud-speaker playback is a factor.

Electronics

- Just 7 dB-A / 17.5 dB CCIR self-noise level allows even the smallest signals to be reproduced noise-free. And with SPL capability up to 138 dB without distortion, the TLM 103 provides a dynamic range of 131 dB.

Acoustic Features

- The TLM 103 is addressed from the front, marked with the red Neumann logo on the microphone body. The K 103 large diaphragm capsule is based on the K 87, well known from the U 87 microphones.
- The capsule has a flat frequency response up to about 5 kHz, and above that, a wide flat 4 dB presence boost.
- The large wire mesh headgrille protects the capsule from plosive sounds and effectively prevents pop noises. These characteristics are achieved without resorting to corrective resonance effects. Therefore, the microphone maintains an excellent impulse response and reproduces the finest details of music and speech without coloration.

Applications

- Use as announcer's mic for broadcasting and voice overs or as a vocalist mic. With minimal self-noise, the TLM-103 is ideal as on-air mic for radio/broadcast, sampling, foley/sound effect. Use as a spot mic for wind instruments, strings, percussion, pianos, guitar amps and drum overheads.

Optional Accessories

- EA 1 elastic suspension and WS 87 windscreen for rejection of unwanted low-frequency interference from structural or windborne noise. The optional PS 30 pop screen is for close miking of vocals.



TLM 170 R

Multi-Pattern Large Diaphragm Microphone

Introduced 20 years ago, the TLM 170 R is a large diaphragm condenser mic with multiple polar patterns, as well as the first Neumann microphone to use FET 100 technology. Its sound has a very transparent characteristic, in contrast to other Neumann microphones that have a distinct personality. Therefore it can be used for many diverse applications in professional recording studios, in broadcasting, film and television, and for semiprofessional productions. Along with a balanced, transformerless output stage it features extremely low self-noise and an impressive dynamic range. Five directional characteristics are selectable by means of a rotary switch. In the sixth position, marked "R", the directional patterns can be remote controlled with the optional N 48 R-2 power supply. There is no special cable necessary for this purpose. The microphone has at its rear a 10dB attenuation switch for extremely high sound pressure levels, and a high-pass filter to suppress structure born noise.

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TLM 170 R

Polar Patterns

- The TLM 170 R offers five polar patterns: omnidirectional, cardioid, hypercardioid, wide-angle cardioid and figure-8.
- Compared to the standard cardioid pattern, the hypercardioid characteristic suppresses sound from the side more efficiently. The wide-angle polar pattern is especially useful to record large sound sources.

Applications

- For universal use, very transparent without coloration. Ideal announcer's mic for broadcasting, dubbing, voice-overs. For close miking of high SPL instruments such as kick drums and guitar amps, or as a spot mic for wind instruments, strings and pianos.
- Optional remote control of polar patterns, allows the mic to be placed anywhere, even suspended from the ceiling.

Acoustic Features

- The mic is addressed from the front, marked with the Neumann logo. The large diaphragm capsule inside the headgrille has a very smooth frequency response for all polar patterns over a wide acceptance angle.
- The curves are flat and parallel to the 0° frequency curve up to 10 kHz within an angle of ± 100° the result of which is a very even diffuse-field response for all polar patterns.
- This ensures that even in reverberant environments–as more reflections arrive at the mic from different directions–the mic maintains an excellent impulse response and reproduces all transient phenomena of music and speech without any coloration.
- Low self-noise level of 14 dB-A / 26 dB CCIR weighted. SPL capability up to 144 dB without distortion, dynamic range of 130dB.

Side Bracket

- The TLM 170 R has a tilting side bracket to attach the mic to booms or stands. The bracket is equipped with rubber elements to protect the mic from mechanical shock. If necessary, it can be mounted on the other side of the microphone as well.

Optional Remote Control

- The N 48 R-2 provides remote control of the mic's polar patterns by varying the phantom voltage. The range is ± 3v of the nominal 48v value.

Filter and Attenuation

- A 10 dB attenuation switch prevents the input from being overloaded.A second switch at the rear lets you attenuate the frequency response below 100 Hz to suppress undesired structure borne noise.

The U 87 Ai, "The Standard"



Since the late 1960s, the Neumann U 87 has been the standard by which all other studio microphones have been measured. Not only is it a legendary audio product, but in recent years, it has also become the most imitated mic in the world. This microphone got its start as a solid-state version of the older U 67, which was a tube mic. The U 67 was introduced in 1960 as a more modern, versatile, and capable microphone than the venerable U 47. In comparison to the U 47, the U 67 had three polar patterns (instead of two), a -14dB pad switch for louder sound sources, and a high pass filter to reduce low frequency sounds.

The capsule developed for the U 67 was a radical design and a departure from anything that had come before. During the process of creating this capsule, more than 40 prototypes were tested with broadcast and recording engineers throughout the world. The result was the K67, which as survived today in the thoroughly modern U 87 Ai studio microphone. This capsule, when compared to previous designs, exhibits very linear off-axis pickup (especially for such a large capsule), excellent transient response and a very even response across all polar patterns. In terms of sound quality, the K67 capsule provides a sublime capture of anything placed before it.

Over the last three decades, the U 87 has only undergone one significant change: from the original U 87 to the newer U 87 Ai. This change took place in 1986, and involved mostly the re-design of the audio and powering circuits. The addition of a DC-DC converter allows the modern U 87 Ai to operate at a very high level of sensitivity but with exceptionally low self noise.

Today, if you run a professional recording studio, the U 87 Ai should be a part of your collection. Since there are more of these mics throughout the world, and their legendary performance is well known by artists and engineers alike, clients looking to book your rooms will expect this microphone to be available. Whether you are recording strings, vocals, drums or piano, this is one microphone you simply can not do without.

TLM 103 Microphone Nickel color. With SGI mount and case

TLM 103 MT Microphone Black color. With SGI mount and case

TLM 103/Z ST Stereo Pair of TLM103 Nickel color. Includes EA1 suspension shock mounts, cables, windscreens, case

TLM 103MT/Z ST Stereo Pair of TLM103 Black color. Includes EA1 suspension shock mounts, cables, windscreens and case

TLM 170R Microphone (Nickel)

TLM 170RMT Microphone (Black)

TLM 170R/ST Stereo Pair of TLM170R Nickel color. Includes case

TLM 170RMT/ ST Stereo Pair of TLM170R Black color. Includes case

UA 87 Ai Microphone (Nickel)

UA 87 Ai MT Microphone (Black)

UA 87 Ai/Set A: Includes UA87Ai, SG-367 Swivel Mount, Windscreen, Cable

UA 87 Ai/Set A: Includes UA87Ai, SG-367 Swivel Mount, Windscreen, Cable

UA 87 Ai/Set Z: Includes UA87Ai, EA-87 Shock Mount, Windscreen, Cable

UA 87 Ai/Set A: Includes UA87Ai, EA-87 Shock Mount, Windscreen, Cable

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U 87 Ai

Multi-Pattern Large Diaphragm Studio Microphone



The U 87 Ai is the best known and most widely used Neumann studio microphone. It is equipped with a large dual-diaphragm capsule with three directional patterns (cardioid, omnidirectional and figure-8) and a unique frequency and transient response characteristic. Users recognize the microphone immediately by its distinctive design. The U87 Ai is ideal for most general purpose applications in studios, for broadcasting, film and television. It is used as a main microphone for orchestra recordings, as a spot mic for single instruments, and extensively as a vocal mic for all types of music and speech. Patterns are selectable with a switch below the headgrille. A 10dB pad enables the mic to handle sound pressure levels up to 127 dB without distortion. Low frequency response can be reduced to compensate for proximity effect. Available in black or nickel finish, standalone or in packaged with popular accessories for stand mounting.

Polar Patterns

- The dual-diaphragm capsule is elastically mounted and protected by a large head-grille. A switch below the grille selects the three patterns: omnidirectional, cardioid and figure-8. A window above the switch shows the symbol of the selected pattern.
- The cardioid characteristic maintains a smooth frequency response at a distance of 30 to 40 cm, the figure-8 characteristic even at a distance of 15 to 20 cm.

Acoustic Features

- The U 87 Ai is addressed from the front, marked with the Neumann logo. The frequency response of the cardioid and figure-8 directional characteristics are very flat for frontal sound incidence, even in the upper frequency range. The mic can be used very close to a sound source without the sound becoming unnaturally harsh
- High-pass filter practically eliminates interference from subsonic and low frequencies
- SPL capability of 117 dB without distortion and self-noise of 12dB-A gives the mic an overall dynamic range of 105 dB
- Switchable -10 dB pre-attenuation pad lets the mic handle up to 127dB SPL without distortion
- Low-frequency roll-off filter lets you change the mic's cutoff frequency thus reducing low frequency interference directly at the input of the mic amplifier. Also compensates for proximity effect

Applications

- For universal use
- The classical studio mic for vocalists (soloists & background vocalists)
- Announcer's mic for broadcasting, dubbing, voice-over; as overhead mic
- Spot mic for wind instruments, strings (especially cello & double bass), piano, and percussion

	TLM 103	TLM 170R	U 87 Ai
Microphone Element Type	Pressure gradient transducer	Pressure gradient transducer	Pressure gradient transducer
Polar Pattern	◡	○, ◡, ◡, 8	○, ◡, 8
Frequency Range	20 Hz - 20 kHz	20 Hz - 20 kHz	20 Hz - 20 kHz
Sensitivity @ 1 kHz into 1 kohm	21 mV/Pa	8 mV/Pa	20/28/22 mV/Pa*
Rated impedance	50 Ω	50 Ω	200 Ω
Rated load impedance	1000 Ω	1000 Ω	1000 Ω
Equivalent SPL CCIR 468-3	17.5 dB	26 dB	26/23/25 dB*
Equivalent SPL DIN/IEC 651	7dB-A	14 dB-A	15/12/14 dB-A*
S/N ratio CCIR 468-3	76.5 dB	68 dB	68/71/69 dB*
S/N ratio DIN/IEC 651	87 dB	80 dB	79/82/80 dB*
Maximum SPL THD < 0.5 %	138 dB	144 dB	117 dB (cardioid)
Maximum SPL THD with pre-attenuation		154 dB	127 dB
Dynamic range of the Mic Amp DIN/IEC 651	131 dB	130 dB	105 dB
Supply Voltage	48 V ± 4 V	48 V ± 4 V	48 V ± 4 V
Current Consumption	3 mA	3 mA	0.8 mA
Matching Connector	XLR 3F	XLR 3F	XLR3F
Diameter	5.2 x 2.4″	6 x 2.4″	7.9 x 2.2″
Weight	17.6 oz.	22 oz.	17.6 oz.

** Omni-directional / cardioid / figure-8*

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SOLUTION-D



Digital Microphone System

Setting new standards in microphone applications and recording techniques, Neumann's Solution-D digital microphone closes the final gap in the all-digital signal path. By transferring the dynamic range and fidelity of the best analog studio microphones into the digital domain, one of the last parts of the audio chain—the microphone—has become part of the digital revolution. Thanks to a revolutionary new A/D conversion process, a specially developed synchronization method, and remote control of typical microphone parameters and mixing console functions, Solution-D satisfies even the highest demands of professional audio production. It consists of three components: the Digital Microphone D-01, the Digital Microphone Interface DMI-2, and the software (RCS) that permits operation and remote control of the microphone. Solution-D also introduces a new AES standard. AES 42-2001 identifies the transmission of output signals, power supply for the microphones, and remote control of all typical microphone functions and parameters.

The wide variety of Solution-D features permits small productions in excellent quality with just a microphone and a recording device. Every sound designer who is familiar with digital recording technology, can begin using Solution-D immediately without a learning curve. As with analog microphones, the same acoustic principles and recording techniques apply. However, the additional features of digital microphones open up new possibilities during production.

FEATURES

Integrated Digital Audio Signal Processing

Neumann developed a completely new A/D converter specifically for the signal levels directly after the mic capsule. Necessary gain adjustments can be performed inside the mic in the digital domain. This eliminates traditional peripherals devices such as preamplifier and A/D converter, as well as reduces costs.

The front-end conversion of the analog signal provides a marked improvement in dynamic range. This is noticeable throughout the entire signal chain and makes level adjustments less critical. Other signal processing functions, such as mute and phase reverse, which are usually inside a mixing console, are integrated into the mic.

Other possibilities include the use of remote commands, such as an LED to indicate Red Light ("On Air"). Features include the transient limiter, which for the first time can be applied at the most efficient point, that is, at the signal source. This can reduce damaging transients that are very short, however exhibit a large amplitude. Analog microphones require extremely large headroom in the following signal path to accommodate such signals.

Additional features will include a noise gate function and switchable test signals inside the microphone.



D-01 Digital Microphone

At first glance, the D-01 mic looks no different than its analog counterparts. Right behind the mic capsule, however, is an A/D converter developed by Neumann and optimized for the signal level of the microphone capsule. The signal sent by the microphone capsule is converted directly into a digital signal. The result is an internal digital 28-bit signal with a dynamic range of more than 130 dB (A-weighted, including microphone capsule).

The digital signal will be processed by the help of an FPGA (Field Programmable Gate Array) built right into the microphone. This means that parameters such as polar pattern, pre-attenuation, low-cut, pre-amplification and various switching functions can be realized digitally and remote controlled. This eliminates previously required devices such as analog pre-amps and A/D converters. For positive self-identification, the microphone transmits information such as manufacturer name, type, serial number, and implemented software level to the connected receiver.

The microphone is equipped with a 3-pin XLR connector via which the supply voltage, phantom power, and remote control data stream are transmitted. The signal components for a control loop as required for synchronous transmission are also contained in this data protocol. The output signal of the microphone conforms to the AES 42-2001 format.

SOLUTION-D

DMI-2 Digital Microphone Interface

The Solution-D microphone connects directly to any equipment that supports the new AES 42-2001 Standard. In all other situations, the Neumann DMI-2 Digital Microphone Interface is necessary. This separate, two-channel device converts the AES 42-2001 data format of the microphone into an AES/EBU signal. The system will be operated by the Neumann remote control software with a computer connected to the DMI-2 interface via a USB connection. If a large number of microphones are used, it is possible to cascade multiple DMI-2's and still provide each interface with an individual address. Besides word-clock input and output, the unit also has an integral master word clock generator that, in the future, will support sample rates of up to 192 kHz. If there is no word clock at the input, then the internal master is used automatically to synchronize both microphone channels and is switched to the word-clock output. External commands—for example, a red "On Air" light—can be controlled via the 9-pin port.







RCS Remote Control Software

The Solution-D is controlled by Remote Control Software, which would ideally be part of a recording system, but can also run on a standalone desktop or laptop computer. All important parameters are shown on the screen and can be changed at any time. During production, the recording engineer has complete control over the status and parameters of all mic's and can change settings quickly and conveniently if necessary.
The parameters include polar pattern, pre-attenuation, low-cut, pre-amplification, various indicators of microphone status, command indicators, mute and phase reverse functions. It is also possible to follow signal levels on the screen. The information is transmitted by the mic, (manufacturer name, type, serial number, etc.) is displayed and used for positive self-identification of the microphone connected. In addition, it is possible to add self-written notes. For example, the name of the sound source and positioning of the microphone. Of course, it is also possible to store and retrieve the setup for the entire recording session.

D-01 SPECIFICATIONS

Acoustical Transducer	Double diaphragm capsule K 07; Large diaphragm, diameter 30 mm with protected internal electrodes
Sensitivity at 1 kHz, 0 dB gain	12 mV/Pa
Equivalent SPL DIN/IEC 651	7 dB-A
S/N ratio DIN/IEC 651	87 dB
Interface	AES 42-2001
Dynamic Range	130 dB (complete system incl. capsule) 140 dB (ADC input shorted)
Internal Resolution	28 Bit
Sampling rates	48 kHz / 96 kHz, alternatively 44.1 kHz / 88.2 kHz
Remote Controllable Functions	Polar pattern: 15 patterns, omni -cardioid-figure-8 Low Cut: Flat/40 Hz/80 Hz/160 Hz Pre-attenuation: 0 dB, - 6 dB, -12 dB, - 18 dB Gain: 0-63 dB in 1 dB steps, clickless
Switch functions	Soft muting, phase reverse, signal light (further functions in preparation)
Synchronization	AES 42 - Mode 2 (default) and AES 42 - Mode 1 (see DMI)
Signals Blue and red LED (switchable via control software and Aux User Port)	
Output	XLR 3 M, AES 42-2001 data format: transmission of audio and status data from the microphone, phantom powering and remote control data to the microphone
Parametric transient limiter	
Weight	approx. 700 g
Dimensions	63.5 x185mm (DxL)

DMI-2 SPECIFICATIONS

Inputs	XLR 3 F, AES 42-2001 data format: transmission of audio and status data from the microphone, phantom powering and remote control data to the microphone
Outputs	XLR 3 M, AES/EBU data format, switchable between one and two channel operation. Configuration via Remote Control Software RCS
Phantom Power	+ 10 V, max. 250 mA per channel, short circuit protected
Remote Control Data	750 Bit/s pulses overlaid on the phantom power.
Control Bus	2 RJ45 jacks, connecting with included Neumann interface to USB-Port (Windows 2000, Win 98SE, Win ME), cascading of up to 16 DMI-units (2 or 8 channel) possible.
Device Address	Selectable via coding switch on the rear side.
Aux User Port	9pol Sub D, 4 Bit per channel, 2 Bit each reserved for signal light control
Synchronization	**AES 42 -** Mode 2, clock control via PLL (Default setting). Without external word clock master, the internal master is used for synchronization of both microphone channels. **AES 42 -** Mode 1, clock free running at center frequency, necessitates Sample Rate Converter (SRC) on receiver side. Reduction of dynamic range depending on the quality of the SRC.
Word Clock Input	BNC, 75 Ohm.
Word Clock Output	BNC, 75 Ohm, set automatically to internal word clock master when no signal is present on word clock input.
Internal Word Clock Generator	Selectable sampling rates: 48 kHz, 96 kHz, 192 kHz, 44.1 kHz, 88.2 kHz, 176.4 kHz.
Indications	Data Valid (AES 42) and Sync Locked (Mode 2) per channel, Power On and Ext.Word Clock.
Power Supply	100-240 V, 50/60 Hz.

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KMS 105

Vocal Microphone

Designed with superlative vocal reproduction in mind, the KMS 105 brings Neumann quality to the demanding sound reinforcement environment. With its supercardioid polar pattern, low self noise and uncolored off-axis pickup, the KMS 105 complements in-ear monitor systems. By employing a unique 4-layer acoustic filter, the KMS 105 minimizes popping and wind noise, and with special mechanical and electrical filters, handling noise is virtually eliminated. Because of the superior resolution and linear frequency response, it is very easy for the artist using the KMS 105 to identify whether he or she is "on mic".



FEATURES

Highest Quality Vocals

- Compared to other handheld vocal mics, the KMS 105 offers a superior resolution of the voice, with extended frequency response and accurate transient detail. Sibilance problems common to other handheld condenser mics are eliminated.
- The K 105 capsule features a supercardioid pattern, achieved by specially modifying the response of the pressure gradient transducer. The frequency response characteristic developed in the acoustic realm, allows for accurate representation of transient details found in music and speech, thus transmitting a very "open" and detailed capture of the human voice.
- Due to the tight supercardioid polar pattern and the coloration-free off axis response, a very high level of gain before feedback is achieved. This polar pattern offers the highest degree of isolation while still allowing for a generous "sweet spot" in front of the microphone.
- The KMS 105 has a dynamic range of 132 dB, allowing reproduction of the artist's full expression, without restraint. Maximum SPL is 150 dB, and with low self noise (18 dB-A) greater gain can be used without risk of adding noise to the mix. This way, the artist is able to use the microphone at a greater distance therefore maintaining a wide range of creative freedom during the performance.
- And by employing a transformerless output circuit, this microphone can incorporate long cable runs with no loss of signal quality or high-frequency response.

The Microphone Basket

Most vocal mics rely on the attenuating effects of foam layers inside the basket. Although this is certainly a cost-effective procedure, the detrimental effects of this foam layer can be shown quite easily. Treble frequencies are attenuated, and furthermore the polar pattern can be deformed.

- No foam is used in the basket, thus eliminating any "clouding" or "muffling" of the sound. Also, by using a DC-polarized studio condenser capsule, all the nuances of the voice come through with subtlety, power and focus.
- These drawbacks can be eliminated by selecting open mesh constructions. With careful dimensioning and selection of different mesh sizes, excellent pop and wind attenuation can be delivered, as is well known from the standard studio-use pop screen PS30, made of two layers of spaced fabric.
- Naturally, the outer microphone basket is hardened steel, due to the required ruggedness for on-stage use.

	KMS 105	KMS 140	KMS 150
Element Type	DC-biased Condenser		
Operating Principle	Pressure gradient transducer		
Directional Pattern			
Frequency Range	50 Hz - 20 kHz	50 Hz - 20 kHz	50 Hz - 20 kHz
Sensitivity @ 1 kHz into 1 kohm	4.5 mV/Pa	4.7 mV/Pa ±1 dB	3.1 mV/Pa ±1 dB
Rated impedance	50 Ω	50 Ω	50 Ω
Rated load impedance	1000 Ω	1000 Ω	1000 Ω
Equivalent SPL DIN/IEC 651	18 dB-A	16/18 dB-A	16/18 dB-A
S/N ratio DIN/IEC 651	76 dB	68 dB	66 dB
Maximum SPL for THD 0.5%	150 dB	138/148 dB	142/152 dB
Maximum Output Voltage	12 dBu	10 dBu	10 dBu
Dynamic Range	132 dB	122 dB	124 dB
Supply voltage	48 V ± 4 V	48 V ± 4 V	48 V ± 4 V
Current consumption	3.5 mA	2 mA	2 mA
Weight	approx. 10.5 oz.	approx. 10 oz.	approx. 10 oz.
Dimensions	1.89 x 7.08″	1.89 x 6.89″	1.89 x 6.89″

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KMS 140/KMS 150

Condenser Vocalist Microphones



The KMS 140 (cardioid) and KMS 150 (hypercardioid) vocalist microphones have been developed for the use of instrumental and vocal soloists at very short range, and can also be hand-held. They incorporate a special acoustic filter and a transformerless, high-loadability amplifier to provide very effective protection against pop and other explosive sounds. Pop stability is excellent and sibilants and S-sounds are reproduced with all their natural accentuation. Although the acoustic filter effectively suppresses interference by plosive sounds, the distinctive directional characteristics of the capsules are retained all the way down to the bass region, lending the mics a high degree of feedback imperviousness when used for stage work. With two easily operated recessed switches, sensitivity can be reduced by 10 dB, and the transmission characteristics of the microphones matched to the vocal character of the soloist and the preferred pick-up distance (proximity effect).

The thick walled metal case of the microphones effectively blanket holding noises. The acoustic filters can be easily un-screwed and cleaned if necessary.



KU 100 *Dummy Head*

The KU 100 dummy head is a binaural stereo microphone. It resembles the human head and has two microphone capsules built into the ears. When listening through high-quality headphones it gives the illusion of being right at the scene of the acoustic events. When using the KU 100 the binaural stereo experience moves the listener into the scene of the original performance, in contrast to other space-related recording techniques, where the acoustic event is moved to the listener.

The dummy head is also used in industrial applications as a measuring device, for example in acoustic research. The KU 100 can be operated with 48v phantom powering, from an external power supply, or from the built-in battery. At the bottom of the unit is a switch for the different power supply modes, as well as connectors for balanced and unbalanced output signals. Inside the head are additional switches for 10dB attenuation and the high-pass filter.

- The KU 100 is a replica of the human head with a mic built into each ear. When the recorded audio signal is reproduced through high-quality headphones the listener perceives a sound image almost identical to the one he would have heard at the recording location of the dummy head (head-related stereophony).
- Balanced (XLR) and unbalanced (BNC) outputs
- When played back through loudspeakers, the sound matches that of stereo microphones, placed in the same position. However, a superior quality is added, that of a distinct spatial depth perception.
- The KU 100 is just as easy to use during creative radio drama productions, and music recordings where the room acoustics should be recorded at the same time
- It is also an important tool to preserve natural sounds of all kinds. In addition, the dummy head is used to examine and document the influence of noise in industrial applications under realistic conditions.
- Includes a robust aluminum carrying case, external power supply, 5-pin XLR cable, and an adapter cable that splits the 5-pin XLR output into two 3-pin XLR connectors

NEUMANN

SERIES 180



The "Series 180" consists of three compact miniature microphones with patterns that satisfy the demands of all common studio applications. Designed for economy-minded production, their slender shape and the transmission characteristics make them ideal for a wide range of applications in radio and television. The KM 183 (omni), KM 184 (cardioid), and KM 185 (hypercardioid) are available in either matte black or nickel finish. The mics come in a folding box with a windshield and two stand mounts that permit connection to the microphone body, or the XLR-connector.

The KM 183 is a pressure transducer with a boost of approximately 7dB at 10 kHz in the free field. In the diffuse sound field it has a flat frequency response. The pressure gradient transducers KM 184 and KM 185 feature very smooth frequency responses not only for the 0° axis, but also for lateral (off-axis) sound incidence. In typical usage, there is no coloration of sound over a wide pickup angle. The KM 185 features attenuation of sound incidence from the side or rear of about 10 dB, with minimum sensitivity at an angle of 120°. The KM 184 has a gentle rise at about 9 kHz.

The "Series 180" microphones have the same transformerless circuitry as is used in the KM 100 system, resulting in excellent technical specifications. Their output as in all Neumann microphones, is balanced and phantom (48v) powered. The KM 183 and KM 184 are available in stereo sets, packaged in wooden boxes.

KM 183

- For close miking of instruments when there is no need to attenuate extraneous noise, and in a balanced acoustic environment to record acoustic guitar, wind instruments, strings, percussion, drums
- Ideal as AB stereo pair because of the flat frequency response in the diffuse sound field
- As a main mic, especially for capturing room acoustics
- For stereo recordings with a baffle plate
- As a spot mic for piano, wind instruments, organ, choir

KM 184

- For universal use, especially for recording situations when it is necessary to attenuate off-axis sound (mainly from the rear) from other nearby instruments.
- As XY and ORTF stereo pair
- Announcer's mic for broadcasting
- Spot mic, overhead
- Close miking of strings, wind instruments, percussion, piano, Leslie speakers, guitar amps

KM 185

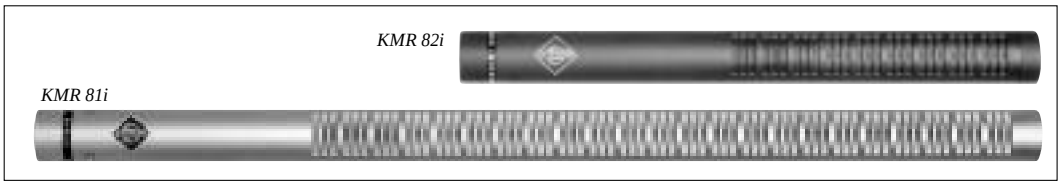
- Especially for recording situations when it is necessary to attenuate off-axis sound (lateral and rear) from other nearby instruments.
- As XY stereo pair
- Overhead, toms
- In situations that are susceptible to acoustic feedback
- To attenuate unwanted sound of nearby instruments
- Recording of speech, as in TV, movie and video productions, PA systems
- Produces especially warm and bass supporting sound for artists who perform in proximity effect range



	KM 183	KM 184	KM 185
Acoustical Operating Principle	Pressure/Pressure gradient transducer		
Directional Pattern	◯	◠	◠
Frequency Range	20 - 20 kHz	20 - 20 kHz	20 - 20 kHz
Sensitivity at 1 kHz into 1 kohm	12 mV/Pa	15 mV/Pa	10 mV/Pa
Rated Impedance	50 Ω	50 Ω	50 Ω
Rated Load Impedance	1000 Ω	1000 Ω	1000 Ω
Equivalent SPL DIN/IEC 651	16 dB-A	16 dB-A	18 dB-A
S/N Ratio DIN/IEC 651	78 dB	78 dB	76 dB
Maximum SPL for THD 0.5%	140 dB	138 dB	142 dB
Maximum Output Voltage	10 dBu	10 dBu	10 dBu
Dynamic Range of Mic Amp DIN/IEC 651	124 dB	122 dB	124 dB
Dimensions (DxL)	.87 x 4.2″	.87 x 4.2″	.87 x 4.2″

KMR 81i/ KMR 82i

Shotgun Microphones



The KMR 81i and KMR 82i are shotgun microphones with a high directivity that remains within the acceptance angle independent of the frequency. The advantage is that a sound source, for example an actor on stage, will not change its apparent tonal balance when moving within this area.

Shotgun microphones are particularly useful in recording situations where a microphone cannot be positioned within the desired distance of the sound source to produce a sufficiently loud signal level.

Typical applications are film and video recordings, where the microphone should not appear in the picture. The KMR 82i is very often used on stage. The KMR 81i has been specifically designed for electronic news gathering.

FEATURES

Acoustic Features

- These mics use a combination of a pressure gradient transducer and an interference tube. If the wavelength of the frequency is longer than the tube length, the mics work as pressure gradient transducers. At higher frequencies they operate as interference transducers for lateral sound.
- Off-axis sound sources are picked up with reduced level, but without coloration. Therefore, the microphones are well suited to record individual instruments of an orchestra. The pickup areas of several shotgun microphones may even overlap, like during recordings on a large stage, without causing any problem.
- They are less sensitive to wind and pop noise when compared to other mics with similar high directivity. The KMR 81i/82i both feature extremely low self noise, good impulse response, and high output level.

Polar Pattern

- These mics have a very directional characteristic. Their mic capsule is positioned inside a housing tube that is acoustically open but has a high flow resistance. The directional patterns of the mics are lobe shaped. The attenuation of lateral sound is practically independent of the frequency. The KMR 82i has a frequency independent directivity within a pickup angle of 45° for audio signals that determine the tonal balance of the program material. For the KMR 81i, this angle is 90°.

Filter and Attenuation (KMR 81i)

The microphone has a 10dB attenuation switch to prevent the input of the following unit from overloading. A second switch activates a 200 Hz high-pass filter. Toward the lower frequencies the sensitivity of the microphone is attenuated by approximately 15dB at 50Hz. The frequency range above 200 Hz is unaffected.

Using on Location

The shotgun microphones feature very high output capability and a remarkably low self-noise level. Their low power consumption, light weight, and low sensitivity to wind and handling noise, make them ideal tools for news gathering on location. Small dimensions, together with a balanced center of gravity, make handling easy without any whiplash effect.

However, when on location and during strong wind conditions, it is recommended you use the using an additional wind screen (included). The wind screen is made of polyurethane foam and also serves as soft padding of the microphone in its leather carrying case. For mobile use a handle and an elastic suspension are available. An active handle with a built-in battery power supply makes an external phantom supply unnecessary.

Filter KMR (82i)

Between 2 kHz and 15 kHz the KMR 82i has a boost to compensate for HF transmission losses in air when recording distant sound sources. This may overemphasize any sibilance if the microphone is used close-up. Therefore, a two-position slide switch allows to select the setting that is best for balanced upper frequencies. The KMR 82i has a high-pass filter to suppress subsonic interference. The cutoff frequency may be raised to 120 Hz (-3 dB) with a built-in two-position slide switch.

	KMR 81i	KMR 82i
Acoustical Operating Principle	Interference transd.	Interference transd.
Directional Pattern	Supercardioid/lobe	Supercardioid/lobe
Frequency Response (Hz)	20 - 20k	20 - 20k
Sensitivity[1]	-18 mV/Pa	21 mV/Pa
Rated Impedance	150 Ω	150 Ω

[1] *at 1 kHz into 1 kohm*

NEUMANN

SM 69 / USM 69

Stereo Microphones



The SM 69 and USM 69 are condenser studio stereo microphones suitable for XY and MS recordings. The construction of the capsules is the same in both microphones. Therefore, the acoustic features of the transducers are identical. The only difference is the method of switching the polar patterns and the way the microphones are connected to the system. The SM 69 and USM 69 stereo microphones each have two separate dual-diaphragm capsules. These are mounted vertically and rotate against each other. The directional polar patterns can be selected separately for each capsule. The capsules operate independently from each other.

They can also be used as completely independent mono microphones. There are many applications when it is important to have a second mono microphone as a backup, or when the outputs of microphones with different polar response characteristics must be available simultaneously. The outputs of the two microphone channels can be linked (cascaded). In addition to the individual directional patterns, other characteristics are available through the combination of both channels.

FEATURES

- Two completely separate microphone capsules are positioned closely above each other within the capsule head. Their diaphragms are made out of gold-sputtered polyester film. The upper capsule rotates against the lower one over a range of 270°. Color markings on the lower capsule system help to identify the angle by which the upper capsule has been rotated.
- When sound waves reach the microphone capsules from different directions they will generate audio signals with different intensity only, but not with time differences, since the capsules are in close proximity and the sound arrives at both capsules simultaneously. The result is an intensity stereo signal that can be summed together for excellent mono compatibility without causing interference.
- The amplifiers feature high output capability and extremely low self noise. Distant as well as very loud sound sources at close range are easily recorded.. Each amplifier has an active filter to suppress wind or structure borne noise.
- Mic systems operate independent from each other. The second amplifier will be unaffected, even in case of a faulty ground for one of the channels, or a short circuit in one of the outputs. The mic is reliable in mono usage, even if only one of the systems is operated and connected.

Polar Patterns SM 69

In addition to omnidirectional, cardioid and figure-8 polar patterns, this mic has six intermediate positions. The narrower characteristics suppress sound from the side more efficiently, while the wide-angle polar patterns are especially useful to record large sound sources. Patterns are selected remotely from the NS 69 i power supply. Two rotary switches are used to switch between nine different polar patterns for each system. The switches are marked in color and numbered corresponding to the respective microphone capsule.

Polar Patterns USM 69

Like the SM 69, the USM 69 has two built-in rotary switches. However, the five polar patterns of both capsules can be selected at the microphone itself. No special AC power supply or power adapters are necessary.

In addition to the omnidirectional, cardioid, and figure-8 pattern, Neumann added a hypercardioid and a wide-angle cardioid pattern. A built-in DC converter generates the required capsule polarizing voltages.

Boundary Layer Microphone



The GFM 132 is a boundary layer microphone. Through computer simulation Neumann optimized the design to be free of any comb filter effects due to reflections, typical of other such microphones. The smooth frequency response for all angles of incidence exhibits a rise in the upper frequency range. This assures that all sound sources, even distant ones, will be recorded with clarity and presence. Therefore, typical applications are for live recordings, such as in the orchestra pit of opera houses, theaters and on stage. The back of the microphone has non-slip pads for its use in a horizontal or inclined position, and holes for wall suspension. It is supplied with a wooden case and a wind screen.

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RSM 191 A-S

Stereo Shotgun Microphone



The RSM 191 is a stereo microphone system consisting of the microphone and the MTX 191 A matrix amplifier. It has an adjustable pick-up angle and high directivity. The microphone has two separate capsule systems, a hypercardioid element and a figure-8, both in a single, lightweight body. Together they generate the mid and the side signals. The matrix amplifier controls the width of the stereo image by changing the gain of the side signal relative to the middle signal in six 3dB steps.

The output signals of the matrix amplifier are either mid-side (MS) or left/right (XY). The signals are converted through transformerless sum-and-difference circuitry. The balanced, transformerless technology permits the use of long cables between the mic, the matrix amplifier and the following mic input of other signal processing equipment.

The RSM 191 consists of a capsule head containing two separate transducer systems and an amplifier section with two independent transformerless microphone amplifiers. An interference tube system with a hypercardioid characteristic generates the middle signal, while a horizontal figure-8 assembly provides the side signal. The RSM 191 is an ideal microphone for outdoor stereophonic news coverage, even in extremely noisy surroundings and stereo recordings for film and television where the actors are recorded simultaneously with acoustic ambiance, or an orchestra.

FEATURES

- The RSM 191 achieves its high directivity and special acoustic features through a unique construction. The microphone capsule assembly is mounted inside a housing tube that is acoustically open but has a high flow resistance.
- This results in a high driving force for the diaphragm, even with a low pressure gradient factor of the capsule. Therefore, the microphone can suppress unwanted off-axis interference to a considerably higher degree than other microphones without such an interference tube.
- The middle system (M-signal) combines high attenuation of lateral sound incidence, similar to a hypercardioid characteristic (which is about 10 dB), with the large front-to back ratio of the super-cardioid characteristic.
- Therefore, sound from the rear is likewise attenuated by approximately 10 dB. This design makes the mic less sensitive to wind or pop noise than other shotgun mics
- The side system (S-signal) consists of two single capsules arranged back to back, closely behind the interference tube (M-signal) capsule. Arrows engraved on the mic housing show their orientation.
- The uniform and balanced performance characteristic of the microphone system is achieved without resorting to corrective resonance effects. Therefore, the microphone maintains an excellent impulse response, reproducing all transient phenomena in music and speech without coloration. The entire internal construction is decoupled from the housing to avoid structure borne and handling noise.
- Output signals of the included matrix amplifier are in MS or XY format. The small capsules are mounted in close proximity to ensure transparent and coloration-free stereo sound with excellent mono-compatibility. The stereo acceptance angle is variable and controlled remotely.

Filter and Attenuation

- For the handling of very high sound pressure levels, a 10dB attenuation can be switched ON to avoid overloading of the following equipment.
- To suppress structure borne noise, a high-pass filter in the matrix box provides cutoff fre-quencies of 40 Hz, 80 Hz, and 200 Hz.

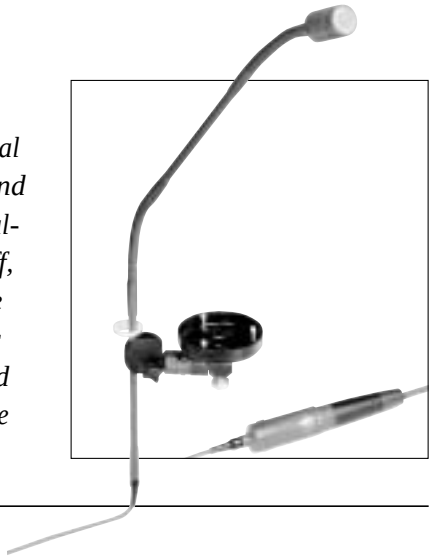
Battery Supply

- The RSM 191 system is fed from either 2 x P 48 phantom power or from a 9 V battery inside the matrix box. One alkaline battery provides 8 hours of operation.
- The complete microphone system comes in a robust aluminum case. Along with the microphone and the matrix amplifier the case contains a windscreen, the special interconnecting cable and an adapter cable that splits the 5-pin XLR output of the matrix amplifier into two 3-pin XLR connectors.

KM100

Miniature Microphone System

The KM100 is a variable condenser miniature microphone system that consists of several active microphone capsules with different directional characteristics, an output stage, and numerous accessories. There are seven active capsules available: omni diffuse-field equalized, omni free-field equalized, cardioid, wide-angle cardioid, cardioid with bass roll-off, hypercardioid, and figure-8. Through the modular construction of mic capsules and the output stage it is very easy to adapt the system to a wide range of applications. The mic becomes nearly invisible during work with cameras (film, video), on stage, or suspended from the ceiling in a concert hall. An active capsule can also be screwed directly onto the output stage. The result is a compact miniature microphone.



FEATURES

Construction



Microphones are approx. 4″ long (except the KM120 which is 5″) and 22mm in diameter. They consist of the condenser capsule and the output stage. Both parts can be unscrewed from each other. The system offers several condenser capsules with different directional characteristics. Numerous accessories can be mounted between the capsules and the output stage. The capsules attach to cables, capsule extensions, swivel mounts, table stands, goose-necks, stereo mounts, and hangers. This makes it very easy to adapt the system to a wide range of applications. The KM 100 powering stage and the active microphone capsule may be separated by up to 50m of interconnecting cable. These cables are 3mm in diameter, and therefore very inconspicuous.

Pre-attenuation

The output stage has a -10 dB switch. Attenuation is achieved by reducing the capsule voltage to one third. When the switch is on, the microphone is capable of accepting sound pressure levels up to 150 dB without being overloaded.

Connectors



LC 3 KA

To diminish the number of connectors within the KM 100 System some accessories were modified. They can now be screwed directly onto the KM 100 output stage without using the KA 100 cable adapter. The new accessories which include the cable adapter, were renamed adding the suffix KA. For example: LC 3 is now LC 3 KA. The separate KA 100 cable, needed for older accessories, will be available also in future. The KM 100 output stage has a 3-pin XLR connector.

Stereo Recordings

By means of the AC 30 adapter cable two active capsules, AK 20 and e.g. AK 40 can be connected as MS stereo pair directly with the MTX 191 (A) matrix amplifier. The XY or MS signal is then available at the 5-pin XLR output connector of the MTX 191 (A), and the recording angle can be electrically remote controlled. The output stages KM 100 are then not required.

Stereo Set

The cardioid and hypercardioid microphones are also available as complete stereo sets, SKM 140 and SKM 150, including all accessories in a single jeweler's box.



Sound Diffraction Sphere



The SBK 130 sound diffraction sphere slips onto the front of the KM 130/KM 131 pressure microphones. The diaphragm becomes an integral part of the surface of the sphere. This affects the frequency response of the micro-phones. While sounds coming from the front-half space are emphasized by up to 2.5 dB between 2 kHz and 10 kHz, sounds arriving from the rear-half space are attenuated by a maximum of 2.5 dB in the range above 5 kHz. Since the sound diffraction sphere causes the pressure buildup of the KM 130/KM 131 pres-sure microphones to begin earlier, the frequency response rises smoothly in the middle and upper range. This is similar to a typical pres-sure gradient microphone, where the directi-vity increases with rising frequencies. However, since the KM 130/KM 131 are pressure microphones, they maintain a linear sensitivity down to the lowest frequencies. This changing directivity allows to record at a greater distance from the sound source, and makes the KM 130/KM 131 microphones especially suited as stereo main microphones in A-B configurations.

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KM100

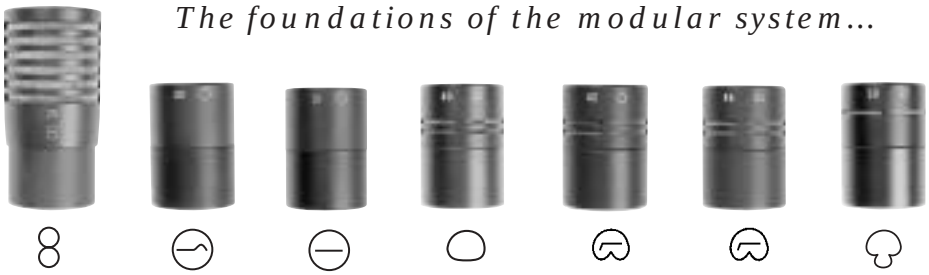
KM 100 Powering Module

- The KM 100 is 48v phantom-powered and uses transformerless output circuitry. This has several advantages:
  - —It features high output capability and extremely low self noise.
  - —It provides exceptionally clean sound, free of any coloration.
  - —As with traditional transformers, this circuit approach ensures good common mode rejection.
  - —The balanced output signal is protected against interference.
- All sensitive components are protected within the capsule. As a result, the quality of the audio signal is never compromised through the use of accessories, for example, when the capsule is detached from the output stage and mounted on a cable or a gooseneck. Even with a long cable between active capsule and output stage, the signal is immune to external interference.

KM 100F
Filtered Powering Module

The KM 100F is offered as an alternative to the KM 100. In contrast to the KM 100, which has a flat frequency response, the KM 100F attenuates frequencies below 80 Hz at 6 dB/octave. This eliminates or suppresses interference caused by wind or structure-borne noise. A built-in slide switch allows to raise the cut-off frequency (-3 dB point) from 80 Hz to 120 Hz. An additional slide switch lowers the sensitivity by 10 dB.

The foundations of the modular system...



AK 20 Figure-8

The AK 20 is a pressure gradient transducer with a figure-8 pattern in a single diaphragm (just 16mm in diameter). All sound field components reach the diaphragm directly. This results in identical frequency response curves and output levels at 0° and 180° sound incidence. Via optional accessories, the AK20 can also be combined with other active capsules or microphones to obtain an MS-Stereo setup.

AK 30 Omnidirectional

The AK 30 is a diffuse-field equalized pressure transducer with a flat frequency response up to 10 kHz (in the diffuse field). In the free sound field this microphone has a boost of approximately 7 dB at 10 kHz.

AK 31 Omnidirectional

AK 31 is a free-field equalized pressure transducer with a flat frequency response up to 20 kHz (in the free field). In the diffuse field it has a high frequency roll-off above 5 kHz.

AK 40 Cardioid

The AK 40 is a pressure gradient transducer with cardioid pattern. The frequency curves are very smooth and match 0° sound incidence. Sound from sources within a pickup angle of ± 135° is reproduced without any coloration.

AK 43 Cardioid Wide Angle

The AK 43 is a pressure gradient transducer with wide-angle cardioid characteristic. Attenuation at 90° is 4dB, at 135° it is 8dB and at 180° it is 11dB. The frequency response curves for sound sources within an angle of ± 90° are parallel up to 12 kHz.

AK 45 Cardioid with Low-Frequency Roll-Off

The AK 45 is a pressure gradient transducer with cardioid pattern similar to the AK 40. It also has a bass roll-off that is useful when low frequencies may cause difficulties. The AK 45 is optimized for a flat low frequency response at a recording distance of 15 cm (“speech cardioid”).

AK 50 Hypercardioid

AK 50 is a pressure gradient transducer with hypercardioid characteristic. Attenuation of sound incidence from the side or rear is about 10 dB, with minimum sensitivity at an angle of 120°.

	KM 120	KM 130	KM 131	KM 140	KM 143	KM 145	KM 150
Operating Principle (Transducer)	Press. grad.	Pressure	Pressure	Press. grad	Press. grad.	Press. grad.	Press. grad.
Directional Pattern	Side-fire figure-8	Omni diff. field eq	Omni free field eq	Cardioid	Wide Angle Cardioid	Cardioid	Hypercardioid
Frequency Range	20 - 20kHz	20 - 20kHz	20 - 20kHz	20 - 20kHz	20 - 20kHz	20 - 20kHz	20 - 20kHz
Sensitivity at 1 kHz into 1 kohm	12 mV/Pa	12 mV/Pa	12 mV/Pa	15 mV/Pa	15 mV/Pa	14 mV/Pa	10 mV/Pa
Rated Impedance	50 Ω	50 Ω	50 Ω	50 Ω	50 Ωs	50 Ω	50 Ω
Rated Load Impedance	1000 Ω	1000 Ω	1000 Ω	1000 Ω	1000 Ω	1000 Ω	1000 Ω
Equivalent SPL*	17.5 dB-A	16 dB-A	16 dB-A	16 dB-A	16 dB-A	17 dB-A	16 dB-A
S/N Ratio*	78 dB	78 dB	78 dB	78 dB	78 dB	77 dB	76 dB
Max SPL**	140 dB	140 dB	140 dB	138 dB	138 dB	138 dB	142 dB
Max SPL with pre-attenuation**	150 dB	150 dB	150 dB	148 dB	148 dB	148 dB	152 dB
Maximum Output Voltage	10 dBu	10 dBu	10 dBu	10 dBu	10 dBu	10 dBu	10 dBu
Dyn. Range of the Mic. Amp*	122.5 dB	124 dB	124 dB	122 dB	122 dB	121 dB	124 dB
Dimension (DxL)	.94/.87 x 4.3″	.87 x 3.6″	.87 x 3.6″	.87 x 3.6″	.87 x 93.6″	.87 x 3.6″	.87 x 3.6″

** DIN/IEC 651 **for THD 0.5%*

STUDIO CONDENSER MICROPHONES

NT1-A Cardioid



Today's musicians and recording engineers demand clarity and dynamic range and the NT1-A, a true large-capsule condenser mic, with state-of-the-art, transformerless FET circuitry meets these demands in both vocal and instrumental performances. The heart of the NT1-A, the capsule, features a full 1˝ gold sputtered diaphragm. This precision transducer is mounted in a proprietary internal shock mounting system. The electronic components are mounted on a revolutionary monocoque sub assembly which assures structural integrity. The microphone head is made of two layers of high-grade stainless steel. The outer layer is of an extremely heavy open weave to provide strength, the internal fine layer provides an effective 'pop' shield while still being acoustically transparent. Supplied stand mount with 3/8 - 5/8˝ thread adapter (RM-1) allows for easy mounting and positioning. Also includes a zippered pouch and windshield.



NT-3 Hypercardioid

The baby of the Rode family, the stylish and modern NT3 is a very capable performer with a wide range of applications in studio, stage, location and broadcast environments. Made from the finest components to deliver results demanded by engineers and musicians alike, the NT3 is a true condenser with transformerless output. The internal capsule is shock mounted and housed in a substantial cast-metal body with a satin-nickel finish. It features battery and phantom power and its low noise (under 16dB SPL A-weighted) makes it hard to match. Dynamic range is greater than 123dB with a maximum SPL of 140dB and sensitivity is a healthy -39dB (ref 1V/Pa).

- High immunity to RF interference
- Full frequency response
- High strength welded and heat treated mesh head
- Premium quality surface mount components
- Battery status LED indicator
- Stainless-steel mesh grille with a dense internal foam pop shield
- Includes a padded zipped pouch, a foam windshield and a rubber standmount clip with a thread adapter for both 3/8˝ and 5/8˝ microphone stands.



NT1000 Cardioid

With its ultra low-noise transformerless circuitry the NT 1000 brings new standards to the recording industry When today's musicians and recording engineers demand clarity and dynamic range,they demand large capsule performance. The NT1000 is superior in both vocal and instrumental performances. Made from high quality components, including internal capsule shock-mounting the NT1000 provides unmatched quality and sound.

- True condenser (externally biased)
- Full frequency response.
- Large capsule with gold-plated membrane
- Ultra low-noise transformerless circuitry
- Cardioid polar pattern
- Heavy-duty cast metal satin nickel body
- High-strength welded and heat-treated mesh head
- Internal capsule shock-mounting

STUDIO CONDENSER MICROPHONES

NT2 Dual-Pattern Condenser Mic



With tens of thousands in use around the world, the NT2 is a modern legend! It is the industry standard. With each component selected for its ability to provide the ultimate sound quality, the NT2 delivers full frequency response and exceptional sound. The NT2 is quality tested at every manufacturing stage. Before dispatch,e very NT2 is critically evaluated in a studio environment. NT2 -when an outstanding, multi-purpose professional microphone is needed.

- Dual gradient transducers with large diaphragm and gold-plated membrane to warmly receive and translate the full body of the performance
- Ultra low noise, transformer-less circuitry
- Switchable omni and cardioid polar patterns
- Switchable High pass filter switch
- Switchable -10dB pad switch
- Full frequency response
- Internal gold-plated connections insure warmth and precise signal transitions for years to come
- Includes shock mount, flight case and pop filter

NT4 Stereo Microphone



In the past, true stereo recording required the use of 'matched' microphones. More than simply using two of the same mic models, matched pairs had to be stringently manufactured so variables such as frequency plots, sensitivity ratings and SPL characteristics would be almost identical. However, even with two matched 'mono' microphones, achieving a successful stereo image still depends on how each mic is placed in relation to the other. Matched mic pairs also need a uniform angle between them to achieve the same effect. This all changes with the NT4.

A studio-grade go-anywhere stereo condenser, the NT4 is a totally flexible mic that enables direct connection to most recording devices. It utilizes two 1/2˝ matched capsules in a perfectly aligned -90°, XY arrangement. All you have to do is point and keep an eye on recording level. It requires 48V phantom power, but for flexibility, a 9v battery can also be used. Ideal for location recording for film and video. Includes Røde case with a stand mount, wind shield and custom stereo cables (XLR and minijack).

NT5 Stereo Microphones



Using the same capsule design as the NT4, the NT5 cardioid condenser mic comes as a matched pair in a plastic case complete with clips and foam windshields. The NT5 is a microphone along the lines of the Neumann KM184 or AKG C451 (but quite shorter). As with the NT4, there are no bass roll-off or pad switches on the microphone. The build quality is good and because the capsule is the same, the sound is similar to that of the NT4.

There is no battery option (requires phantom power), but you do get the extra flexibility of having two separate microphones. For stereo work this means you can vary the angle between them, which will affect the stereo spread and coverage. You can also adjust the spacing of the mics in order to use other stereo techniques, rather than just coincident pair recordings (ORTF for example). Of course, if you don't need to work in stereo, you have two good mics to use individually.

BROADCASTER

True Large Diaphragm Condenser Capsule

Developed using only the finest components, the Broadcaster is a large capsule condenser microphone that gives a warm vocal sound, wide dynamic range and extended frequency response and has a unique LED 'ON AIR' indicator. The mic features a cardioid polar pattern with high 180° rejection, a voice-tailored low-cut filter and ultra low-noise. Designed to provide performance and reliability, it offers a rugged stainless steel body, has an internal shock- mounted capsule to reduce structural borne noise and a fine mesh 'pop shield' to eliminate plosives.





CLASSIC II *1˝ Tube Microphone*

From its custom designed Jensen transformer to the dual pressure gradient transducer this valve microphone is a classic. Machined from solid brass, hand polished, glass bead blasted and coated in satin nickel, this is a microphone with Class A valve circuitry that gives unmatchable warmth, clarity and sensitivity when critical vocal and instrument recording is required.
Weighing in at just over two pounds, the Classic II is a heavy, magnificently crafted instrument. The microphone's body is machine-tooled from solid brass and encapsulates two grille meshes-a finely woven inner headscreen and a heavier gauge, outer screen. A gold dot indicates the front of the mic. The Classic II features a 1˝ dual-pressure gradient capsule with an edge-terminated, gold-sputtered diaphragm. Further, the microphone incorporates a low-noise, 6072 twin-triode tube preamplifier that feeds into a custom Jensen output transformer. There are no controls on the microphone, yet the Classic II provides a wealth of options for securing your desired sound.

- With nine polar patterns (a variety of intermediate positions between the three main patterns—cardioid, omni and figure-8), two pad settings (-10 and -20dB) and a two-position highpass filter (-15 or -21dB of roll-off at 20 Hz), the Classic II can easily be tailored to best suits your needs.
- Frequency response is 20Hz to 20kHz, with a sensitivity rating of 13mV/Pa. Maximum SPL handling is 130 dB, noise is less than 22dB and impedance is 250 ohms.
- Overall frequency response is generally flat, but for vocal recording, the cardioid pattern exhibits a mild boost in the mid to upper midrange. In omni mode, there is a 6dB bump at around 10 kHz.
- A double-shielded, oxygen-free multicore 12-pin cable connects the mic to the power supply. Connectors are custom-tooled military-grade, gold-plated to ensure optimum signal quality. Audio output is routed via a standard XLR from power supply.
- Solidly built external power supply can be set for 220-240V operation. A ground lift switch is provided should you encounter a ground loop hum. Pads and highpass filter are remotely switched from the microphone's power supply.
- Includes and industrial-strength, foam-lined aluminum flight case, dedicated power supply, 30´ multicore cable, stand-mount adapter, and a shockmount suspension.

	NT1-A	NT2	NT3	NT1000	Broadcaster	Classic II	NTV	NTK
Polar Pattern	◡	○ / ◡	◡	◡	◡	○ / ◡	◡	◡
Frequency Response	20 - 20k Hz	20 - 20k Hz	20-20k Hz	20 - 20k Hz	20 - 20k Hz	20 - 20k Hz	20 - 20k Hz	20 - 20k Hz
Output Impedance	100 Ω	40	200	100 Ω	40	200	200	200 Ω
Sensitivity	-31.9 dB	-36 dB	-39 dB	-36 dB	-34 dB	-37 dB	-31 dB	-38 dB
Equivalent Noise	5 dB	< 14 dBa	< 17 dBa	6 dBa	< 14 dBa	< 18 dBa	< 19 dBa	12 dB
Maximum Output	+13dBu	+2dBu	+9.5dBu	+13dBu	+2dBu	+2dBu	2dBu	> +29dBu
Dynamic Range	> 132 dB	> 116 dB	> 123 dB	> 134 dB	> 114 dB	> 113 dB	> 106 dB	> 147 dB
Maximum SPL	> 137 dB	> 130 dB	> 140 dB	> 140 dB	128 dB	131 dB	125 dB	> 158 dB
Signal/Noise Ratio	88dB	> 80 dB	> 77 dB	> 88 dB	> 80 dB	> 76 dB	> 75 dB	> 82 dB

B&H PHOTO - VIDEO - PRO AUDIO *EQUIPMENT LEASING AVAILABLE*

DYNAMIC MICROPHONES

R11 Hypercardioid Vocal Mic

The R11 is a very affordable dynamic mic designed for use in live performance applications such as public speaking, DJ and Karaoke. Hypercardioid pattern minimizes feedback, transformerless design assures reduced noise and low distortion, and Neodymium magnet provides 6dB of noiseless gain. A handy on/off switch is available as well for on-stage control.

Lightweight and compact, the R11 and R21 can be mounted on any standard mic stand (using the supplied mic clip) or handheld for long periods of time. They are also offered in a cost-effective "three-pack" with foam-lined, impact resistant carrying case for convenience when transporting the microphones from venue to venue.



R21 Cardioid Vocal/Recording Mic

Bring a high level of accuracy and audio performance to both live and recording applications. The R21 is a hand held dynamic microphone equipped with a linear frequency response for superior reproduction, a tight cardioid pattern to reduce feedback, and a special shock-mounted element to minimize handling noise. It is also designed to withstand high sound pressure levels. The R21 features gold-plated XLR pins to assure positive connectivity, and a built-in wind screen to protect against most wind and breathing noise. Road-proof design includes rugged zinc alloy die-cast case, heavy gauge mesh and anti-dent ring to ensure reliable performance in the most demanding environments.

PRICING

R11 Microphone **24.95**

R11 Microphone 3-Pack.....**49.95**

R21 Microphone **24.95**

R21 Microphone 3-Pack.....**49.95**

VP1 Mic Kit:
Offering the complete microphone solution the VP1 includes the R21 Dynamic Mic, a heavy duty tripod mic stand, a 25´ XLR microphone cable and a mic clip (with Euro adapter) all in one ready-to-go box **39.95**

S11/S12 Cardioid and Hypercardioid Microphones



The S11 (cardioid) and S12 (hypercardioid) are designed for use in live performance and public address environments. Dynamic design enables them to tolerate high sound pressure levels (SPLs) such as those generated by drums and amplified instruments, and their unidirectional patterns ensure the delivery of a strong, crisp signal from the audio source directly in front of the microphone capsule. The S11 has a linear frequency response that produces a smooth balanced sound, the S12 has a flat, full-range frequency response for exceptional sound quality.
The S12 also incorporates a high energy Neodymium mic element for high output and up to 6dB of additional gain. They are both particularly well-suited for use with Samson wireless systems.

- Lightweight (just 8 ounces) and compact (less than 9˝ long, including head), they can be mounted on any standard microphone stand (using the supplied mic clip) or hand-held for long periods of time
- Rugged zinc-casting, silicon anti-dent ring (protects windscreen) and gold-plated XLR connector ensure reliable performance in the most demanding environments.
- Low impedance 600 ohm output enables strong, full bandwidth signal to be sent over long cable runs.
- Mic clip and carry case included

QMIC

Hypercardioid Vocal Microphone



Fast, highly precise and designed for today's live sound reinforcement and PA situations, the Qmic has a special midrange "peak" at 2 kHz to reproduce vocals with absolute clarity and definition. Manufactured with extreme care and the highest quality components, the Qmic also features a unique humbucking voice coil to eliminate magnetic field interference and a multi-axis shock-mounted mic element that greatly reduces handling noise. A real standout on stage and in the studio, the Qmic is also particularly well-suited for use with Samson wireless systems.

- Full range frequency response with a midrange "presence" peak at 2 kHz for optimum reproduction of vocals and exceptionally clear, crisp sound.
- Unique triple-plated multi-stage windscreen enables "up-close" usage of the Qmic and greatly reduces pops, sibilance and onstage noise.
- Special shock-mounting allows 360° X-Y axis movement of the mic element in order to greatly reduce handling noise.
- Lightweight aluminum humbucking voice-coil eliminates magnetic field interference providing true hum rejection at the source.
- High energy Neodymium element delivers high output and exceptional sound quality.
- Rugged zinc-casting, silicon anti-dent ring (protects windscreen) and gold-plated XLR connector ensure reliable performance in the most demanding environments.
- Includes foam-lined carrying case and mic-clip

High output, linear response, full-frequency reproduction, low feedback and quiet handling. That's what all Samson microphones deliver. From the versatility and exceptional audio performance of their Qmic, Q1, Q2 and Q3 vocal and instrumental microphones to the accessibly-priced and rock-solid R and S series mics. In addition, the QE and QV headset mics break new ground with a clear, balanced vocal response that stands out in vocal performance and fitness situations.

Q1 *Condenser Vocal Microphone*



The Q1 is a hand held condenser vocal microphone that excels in both live and project studio applications. Equipped with a linear frequency response for superior reproduction, it is also designed to withstand high sound pressure levels. Extremely sensitive, it employs a tight cardioid pattern to reduce feedback. A special shock-mounted element minimizes handling noise and provides additional protection. Ideal for live performance vocals, public speaking, project studio vocal.

- Tight cardioid polar pattern minimizes feedback problems and effectively rejects signals not originating directly in front of the mic capsule.
- Special shock-mounting allows multi axis movement of the mic element to greatly reduce handling noise.
- Accepts 11 to 52v DC phantom power
- Withstands high SPLs lending itself to a wide range of miking situations
- Full range frequency response for optimum reproduction and exceptionally clear sound.
- Rugged zinc-casting and silicon anti-dent ring ensures reliable performance in even the most demanding environments.
- Can be mounted on any standard mic stand (using the included mic clip) or handheld for long periods without inducing fatigue.
- Includes foam-lined carrying case for convenience
- Built-in wind screen protects against wind and breath noise. Optional foam windscreen is available for adverse conditions.

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Q2/Q3

Cardioid and Hypercardioid Dynamic Mics

Designed for use in live performance, the compact Q2 (cardioid) is equally effective as a vocal or instrument mic, while the Q3 (hypercardioid) is optimized for drum and amplified instrument miking situations. They incorporate a transformerless design for extended low frequency response without distortion, aluminum humbucking voice coil, and a multi-stage windscreen and filter to reduce pops and sibilance. High energy Neodymium mic element delivers high output with unmatched sound quality. Both include a foam-lined carry case, gold-plated XLR connector and a special Euro-metric adapter. They are also particularly well-suited for use with Samson wireless systems.



- Unique triple-plated multi-stage windscreen enables "up-close" usage and greatly reduces pops, sibilance and onstage noise.
- Switchable 10 dB pad enables use with high sound pressure level (SPL) sources.
- Switchable 12dB per octave high pass filter eliminates unwanted low frequency noise
- Q2 mounts on any standard mic stand (using the included mic clip) or can be handheld for long periods of time.
- The Q3 mounts on a mic stand (no clip required) and rotates mechanically 90° for optimum placement—ideal when miking large drum kits on small stages.
- Rugged zinc-casting and silicon anti-dent ring ensures reliability in the toughest environments.

Q7

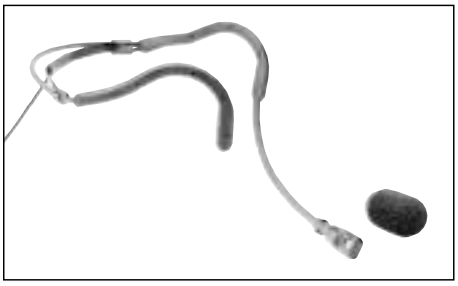
Neodymium Dynamic Cardioid Microphone



Maybe the ultimate all-purpose mic. The Q7 is outstanding for vocals in the studio or on stage. It has a high output, low impedance design, and a shock-mounted Neodymium dynamic mic element providing a cardioid pickup pattern. Includes a carry case and mic clip.

- Outstanding vocal mic for stage and studio
- High output, low impedance design
- Neodymium dynamic mic element
- Cardioid pickup pattern
- Multi-axis shock-mount element
- Includes case and mic clip

QE/QV *Headset Microphones*



Designed for dancers, aerobics and fitness trainers in high-humidity environments, the Qe headset mic is light, comfortable and features a water-resistant mic element to protect against perspiration. This mic can be individually adjusted for a perfect fit. The Qv headset mic allows drummers, keyboardists and other musicians to perform "hands free" vocals without affecting their playing. It produces a full-range sound with a enough definition to ensure that every word is heard clearly, even at higher volume levels. Both headsets offer a lightweight, low visibility design for unobtrusive use and use a high-quality electret condenser element for high output and exceptional sound quality.

- Full range (50Hz-18 kHz) frequency response for clear, crisp audio. The QV is flat to 10kHz, while the QE utilizes a midrange "presence" peak for optimum reproduction of speech.
- The QV is moisture protected, the QE water resistant, making them the choice for aerobics instruction or use in other high-humidity environments.
- Internal RF suppression makes the QE suitable for use in high-RF areas.
- Hypercardioid (QV) and bi-directional noise cancellation design (QE) reduces background noise and feedback.
- Compliant molding reduces mic handling noise, and the booms bend smoothly, hold their shape and are rated to withstand over a thousand 90° bends without degrading.
- The mics are powered via the supplied PM4 phantom power adapter which also transforms the impedance from high to low to compensate for long cable runs.

SAMSON

CONDENSER MICS

C01
Large Diaphragm Mic



Great for recording vocals, acoustic instruments and for use as overhead drum mic, the C01 is accurate, detailed and smooth with a warm bass and extended top end. This sturdy microphone has a large 19mm diaphragm that produces a smooth flat frequency response. It features a heavy gauge mesh grill screen, a gold plated XLR connecter and an LED indicating 48v phantom power.

- Cardioid pickup pattern
- Smooth, flat frequency response (40Hz -18kHz)
- Includes a swivel stand mount
- Optional SP01 shock-mount available

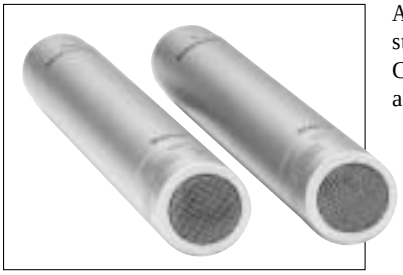
C03 *Multi-Pattern Condenser Mic*



An incredibly flexible and great sounding studio condenser microphone, the C03 has two large (19mm) diaphragm capsules and switchable pickup patterns: Super cardioid, omni and figure-8. Ideal for recording multiple vocals at once, acoustic instruments, room miking and more. It has a switchable high-pass filter, selectable 10dB pad, gold-plated XLR connector and 48v phantom power LED indicator.

- Switchable high-pass filter and 10dB pad
- Gold-plated XLR connectors
- Carry Case & swivel mount included
- Optional SP01 shock-mount available

C02 *Pencil Condenser Mic*



A beautifully crafted pencil-style condenser mic, the CO2 is designed for use with acoustic instruments, cymbals, percussion, piano and in overhead miking situations. Pencil condensers are particularly useful in miking acoustic guitars. The C02 has a highly-focused cardioid condenser pickup pattern, gold-plated XLR connectors comes as a stereo pair in a rugged carry case. Also includes two shock mount mic clips.

- Cardioid Pickup Pattern
- Gold-plated XLR connectors
- Carry case & shock-mounted mic clips included



C05 *Handheld Condenser Mic*

The C05 offers Samson condenser technology in a comfortable to use hand-held design. It has cardioid pickup pattern, smooth, flat frequency response, a multi-axis shock-mounted element, and gold-plated XLR connectors. Includes a convenient carry case and mic clip.

- Cardioid pickup pattern
- 19mm shock-mounted capsule
- Smooth, flat frequency response
- Multi-axis shock-mounted element
- Gold-plated XLR connectors
- Carry case and mic clip included

SAMSON MICROPHONE ACCESSORIES

LS2 Lightweight Speaker Stand
Perfect for small to medium sized pole-mount PA speakers. Adjustable, up to 6′ in height. 1⅜″ diameter pole fits virtually all speakers. 55 lbs. max weight. Sold as a pair.

TS100 Lightweight Speaker Stand
For larger PA speakers. Strong lightweight aluminum construction. 1⅜″ diameter pole fits virtually all speakers. 110 lb. maximum recommended weight.

BL3 Ultra-light Boom Mic Stand
A lightweight collapsible tripod boom mic stand. Super-strong nylon hardware, mic clip included.

BT4 Telescopic Boom Mic Stand
Heavy duty stand, with adjustable boom. All die-cast and steel construction, mic clip included.

MK5 Boom Mic Stand Kit
Includes an adjustable boom mic stand, 18′ Samson XLR mic cable, mic clip and windscreen.

MB1 Mini Boom Stand
Perfect for miking kick drums and speaker cabinets. Heavy duty steel and die-cast construction. Mic clip included.

DMC100 Drum Mic Clip Set
Set includes three adjustable rim-clip drum mic clips with cable management

PS01 Microphone Pop Filter
Features universal mic stand mounting bracket and flexible metal goose neck.

WS1 Windscreen 5-pack
Five black foam windscreens

MC18 Mic Cable 3-Pack
High quality 18″ XLR mic cables

INSTRUMENT MICS

Q Kick
Dynamic Drum Mic



An incredibly well-made kick drum mic. Rigid, and featuring a large diaphragm element, the Q Kick sounds better than kick drum mics costing hundreds more. Mic stand swivel mount and carry case included.

- Large diaphragm dynamic element
- Cardioid polar pattern reduces feedback
- Gold-plated XLR connector
- Mic stand swivel mount included
- Carry case included

Q Snare
Dynamic Drum Mic



Designed for high SPL and durability, the Q Snare is the right choice for stage and studio. The capsule has been specially designed to accurately reproduce the frequencies and SPL of snare drums. Rigid rim-mount mic clip with cable management and carry case included.

- High SPL dynamic mic in a rugged casing
- Cardioid polar pattern reduces feedback
- Smooth, flat frequency response
- Voiced particularly for snare drum
- Gold-plated XLR connector
- Rigid rim-mount DMC-100 mic clip with cable management included
- Carry case included

HM40
Horn/Woodwind Condenser Mic



This mini condenser is designed for horns, rich and clear, focused, with the ability to handle serious SPL. High output miniature condenser mic element with a sturdy clip and flexible goose neck. Excellent for live performance and recording, the HM40 has a P-3 three-pin mini XLR connector for use with Samson wireless systems, and also includes PM5 phantom power adapter (P-3 to standard XLR) for use with standard mic cables. Rugged road case included.

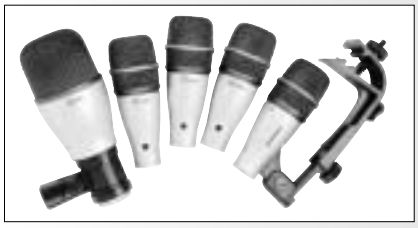


3KIT
3-piece Dynamic Drum Mic featuring Q Tom

Sold in a three pack, the Q Tom is the perfect complement to Samson's Q Snare. It was designed especially for the tone and SPL of tom-toms. Rigid rim-mount mic clips with cable management and carry case included.

5KIT
5-piece Drum Mic Set

A comprehensive drum mic set up in a sturdy road case. Features one Q Kick mic, one Q Snare, three Q Tom mics and a road case.





7KIT
7-piece Drum Mic Set

Everything you could want for miking drums. A complete set of our dynamic drum mics and a pair of C02 pencil condensers for overheads or close-miking cymbals. Features one Q Kick mic, one Q Snare, three Q Tom mics, two C02 pencil condensers and a road case.

SENNHEISER

EVOLUTION SERIES

The New Standard in Live Performance



Offering a complete range of dynamic microphones covering vocals, instrument and backline, Sennheiser's revolutionary Evolution Series answers every conceivable performance requirement. Combining over 50 years of microphone design experience with state of the art production and testing techniques, Evolution delivers the very highest possible standards of performance and construction at remarkably affordable prices.

In developing the Evolution series, Sennheiser listened to musicians and professional sound engineers to find out what kind of performance and pricing they were looking for. What they heard was that the cost of equipping a band with a full set of good mics was too high, that getting the right mic for the right job was more or less pot-luck and that there was little assurance as to quality at affordable price levels. Sennheiser's response was Evolution. A single range of quality microphones for every application, every style and type of voice and every instrument on stage or off.

- Rugged construction and component assembly-designed for the rigors of touring- includes fully die cast zinc castings for all vocal microphones.
- Stunning new capsule design, effected through computer aided acoustic design and laser interferometry techniques.
- Advanced suspension and shock-mount techniques and acoustically engineered grills protect each mic from handling noise and impact, further improving performance.
- High feedback rejection pick-up patterns enable greater signal levels to be delivered before feedback occurs.
- Each capable of extremely high SPL handling (150db+) without distortion)
- Neodymium transducer magnets push output levels even higher, making it possible to balance signals against higher levels of on-stage amplification allowing higher levels of stage monitoring.

E 602 *Cardioid Instrument Dynamic Microphone*



Optimized for high sound pressure levels (160dB) and extended low frequency response, the E 602 is designed for use with bass drums, bass guitar cabs, tubas and other low frequency instruments. The E602 combines the latest lightweight high performance voice coil construction with a traditional large diaphragm capsule design to deliver very low bass response (down to 20Hz) and fast transient signals. Low frequency extension and mid band attenuation produces an ultra smooth pre-shaped signal enabling the mic to be used without corrective EQing, even on the most problematic bass signal sources. Frequency independent cardioid pick-up pattern ensures very high signal output before feedback and isolation from other on-stage signals—making it ideal for stage or studio use.

Kick-Pack



An off-the-shelf, promotional pack, the Kick-Pack is the ideal microphone kit for drummers. It includes an E602 microphone, weighted bass, kickstand, microphone pouch, MC 21N mic cable



SET604B

Drum-Packs

Get everything you need to hit the stage, and save money too! Three cost-saving and popular Drum Pack configurations are available, and all come with a carrying case

SET604A: three e604 mics

SET604B: two e604 and one e602 mics

SET604C: three e604 and one e602 mics

EVOLUTION SERIES

E 604 *Cardioid Dynamic Drum Microphone*



The space-saving compact E604 is ideal for low visibility, close miking techniques. Frequency response and cardioid pick-up pattern are optimized for drum sets and other percussion instrument miking—it also produces exceptional results on all forms of brass and woodwind. An integral stand mount screws directly onto mic stands and included clip attaches mic to rim of drum. Very high sound pressure level handling, in excess of 160dB, is combined with a balanced, clear, low distortion signal similar to a high-end studio condenser microphone. Lightweight voice coil provides extended high frequency and rapid transient response. With its tough reinforced glass fiber body, the E604 can take direct hits from drumsticks without flinching and still provide that ATTACK you've been looking for.

Quick Mount Drum Clamp for E 604 *(MZH504)*.....**28.95** **Swivel Joint for E 604** *(MZJ504)* **29.95**



E 608 *Clip-On Gooseneck Dynamic Instrument Mic*

The world's smallest dynamic microphone, with a full size sound quality. Ideal for almost invisible clip miking of woodwind, brass and drums. Newly developed clamp and goose-neck mounting enables the capsule to be positioned close to the signal source for high realism, high level output. Extensive damping and shock mounting isolates the capsule and signal from extraneous vibration, noise and impact. Humbucking coil protects against induced electrical interference. A high strength, reinforced glass-fiber casing protects the capsule from impact damage.

E 609 SILVER *Supercardioid Dynamic Microphone*



The e609 Silver is based on the legendary MD 409 microphone. Able to withstand high SPLs without distorting, the e609 Silver's flat-profile capsule facilitates extremely close miking of guitar cabinets and is also suitable for drum miking, particularly toms.
The e609 Silver's super-cardioid design improves isolation in live sound reinforcement and recording applications while its increased output and wider frequency response improves performance. Neodymium magnet with boron keeps mic stable regardless of climate. Falcon ring produces consistent sound over time. Its sound inlet basket, made of refined steel, is distinguished by a unique silver address side.

	E602	E604	E608	E609S
	Pickup Pattern	Cardioid	Cardioid	Supercardioid
Transducer Principle	Pressure gradient receiver	Pressure gradient receiver	Pressure gradient receiver	Pressure gradient receiver
Frequency Response	20Hz - 16kHz	40Hz - 18kHz	40Hz - 16kHz	40Hz - 18kHz
Sensitivity (1kHz)	1mV/Pa = - 60 dB	1mV/Pa = - 55 dB	10.8mV/Pa = - 62 dB	1.5mV/Pa =/- 56.5 dB
Nominal Impedance at 1 kHz	250Ω	350Ω	250Ω	350Ω
Min Terminating Impedance	1000Ω	1000Ω	1000Ω	1000Ω
Dimensions	2.4 x 6.1″	1.3 x 2.3″	0.7 x 7.3″	2.2 x 1.3 x 5.3″
Weight	19.4 oz.	2.1 oz.	0.7 oz.	6.3 oz.

EVOLUTION HANDHELD MICROPHONES

E 815s *Cardioid Dynamic Mic*

The E815s is a rugged, great-sounding cardioid mic with a noiseless on/off switch, and it comes complete with an XLR-to-1/8″ cable and 1/4″ adapter. This is the most affordable way to get a mic that is perfect for multi-tracking, for use with digital recorders, and for use with karaoke systems and DJ mixers.

E 825s *Cardioid Dynamic Mic*

An entry level vocal and general purpose dynamic mic, the E825's smooth sound, startling clarity and extended high frequency response give vocals a whole new 'attitude'. A well-balanced frequency response delivers a warm, clear and transparent signal, while its cardioid pick-up pattern provides excellent feedback and spillage rejection. A hum-compensating cable reduces electrical interference, while a neodynum ferrous magnet with boron keeps the mic stable regardless of the climate. Ideal for general vocal and instrument miking and club PA. Rugged metal construction and internal damping isolates handling noise. Complete with silent on-off switch.

E 835/E835s *Cardioid Dynamic Mics*

Quality vocal mic with performance with uniform frequency pick-up pattern that maintains signal quality when moving on and off axis during performance. Gentle presence boost to even tonal response ensures clarity and projection. Minimal proximity effect provides consistently clear bass end performance when singing closer to or further from capsule. Cardioid pick-up pattern provides good signal isolation and feedback rejection, enabling higher sound levels to be obtained. Metal construction and internal shock-mount system minimizes handling noise. Lead vocal stage mic, designed to perform under pressure. Also available with On/Off switch (835-S).





E 845/E 845s Super-Cardioid Dynamic Mics

High quality, powerful performance, vocal mic. Extended high frequency response and super-cardioid pick-up pattern for higher signal output, cuts through high on-stage sound levels. Smooth warm tonal response and full dynamic range, with presence lift for vocal clarity and projection. Consistent on/off axis and proximity response maintains sound quality with moving, providing greater on-stage freedom. Full metal construction and extensive suspension and shock-mounting minimizes handling noise and signal interference. High output performance microphone, offering power, clarity and projection. Also available with On/Off switch (845-S).

E 855 Super-Cardioid Handheld Dynamic Mic

Top-of-the-range, fully professional vocal mic utilizing the very best in dynamic mic technology to achieve new standards of quality and performance. Searing vocal reproduction is matched with bullet-proof construction. Extended dynamic range and assertive presence are achieved with a super-smooth frequency response, delivering the highest quality signal with maximum vocal projection and the lowest possible distortion. Extended high frequency response, combined with the super-cardioid pick-up pattern, enables the mic to handle the highest vocal levels without feedback. Full metal construction and sophisticated shock-mount suspension shields and protects the capsule in the most rigorous conditions.

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EVOLUTION HANDHELD MICROPHONES

e-packs



For added convenience, Sennheiser offers a variety of "e-packs" which consist of one evolution mic, MC21N mic cable, and PM200 boom stand for slightly more than the cost of the mic alone.

e-pack 825S **99.95**
e-pack 835 .....**109.95**
e-pack 845 .....**149.95**
e-pack 855 .....**199.95**

3-packs

Evolution 3-packs include three evolution mics, three MZQ800 mic stand clips, and three mic pouches— all for the cost of two microphones!



three-pack 604 .....**324.95**
three-pack 825S **169.95**
three-pack 835 .....**199.95**
three-pack 835S **199.95**
three-pack 845 .....**279.95**
three-pack 845S **279.95**
three-pack 855 .....**359.95**

E 865

Condenser Vocal Mic



The E865 represents the first condenser microphone in the acclaimed Evolution Series. A live performance, lead vocal microphone, the E 865 easily cuts through high on-stage levels by virtue of its detailed and natural 40 Hz-20 kHz reproduction and superb transient response. Its smooth super-cardioid polar pattern rejects feedback with low, off-axis coloration. The E 865 withstands sound pressure levels of up to 150 dB, and deftly suppresses pop noise. While it picks up the finest sonic nuances, the E 865 is at the same time rugged, benefiting from high- quality workmanship and a robust all-metal housing. Includes pouch and mic clip.

EVOLUTION SERIES SPECIFICATIONS

	E815S	E825S	E835	E845	E855	E865
Transducer Principal	Pressure Gradient	Pressure Gradient	Pressure Gradient	Pressure Gradient	Pressure Gradient	Electret Condenser
Polar Pattern(s)	Cardioid	Cardioid	Cardioid	Super-Cardioid	Super-Cardioid	Super-Cardioid
Frequency Response	20Hz to 20kHz	80Hz to 15k Hz	40Hz to 16kHz	40Hz to 16kHz	40Hz to 18kHz	40Hz to 20kHz
Sensitivity, 1kHz	1.5 mV/Pa	1.5 mV/Pa	2.7 mV/Pa	1.8 mV/Pa	1.8 mV/Pa	3 mV/Pa
Nominal Impedance	350 Ω	350 Ohms	350 Ω	350 Ω	600 Ω	200 Ω
Min Terminating Impedance	1000Ω	1000Ω	1000Ω	1000Ω	1000 Ω	1000 Ω
Dimensions	1.9 x 7″	1.89 x 7.1″	1.89 x 7.1″	1.8 x 7.3″	1.8 x 7.3″	1.9 x 7.72″
Weight	11.6 oz.	11.64 oz.	11.64 oz.	11.64 oz.	11.64 oz.	11 oz.
Supplied Accessories	1 protective pouch and a microphone clip					
Optional Accessories	21′ XLR mic cable, windscreen, floor stand with boom arm					

CLASSIC MICROPHONES

MD-421 II Dynamic Microphone



The MD 421 II continues the tradition of the MD 421 which has been one of Sennheiser's most popular dynamic mics for over 35 years. The large diaphragm, dynamic element handles high sound pressure levels, making it a natural for recording guitars and drums. The MD 421 II's full-bodied cardioid pattern, and five-position bass control make it an excellent choice for most instruments, as well as group vocals, or radio broadcast announcers. One listen and you'll know why it's a classic.

- Glass composite housing and hardened stainless steel basket—rugged and durable
- Five position bass roll-off switch compensates for proximity effect
- Effective feedback rejection
- Easy handling due to pronounced directivity
- Acoustic components are protected from dust and humidity
- Internal bass tubes—accurate low end response
- Handles exceptionally high sound pressure



MD-431 II Speech and Broadcasting Mic

Sennheiser's best dynamic mic for live vocals just got better. The supercardioid MD 431 II has a custom-tailored frequency response to achieve added transparency. A nearly frequency-independent, polar pattern offers incredibly high gain before feedback and provides uncolored off-axis response. Its new rugged housing has a steel-mesh grille, a shock-suspended capsule, and a hum-bucking coil. It features a noiseless, magnetic reed on/off switch which is recessed in its handle, and internal electronics for refining proximity effect.

- Exceptionally good feedback rejection
- Shock mounted capsule—low sensitivity to handling noise
- Excellent for voice recognition
- On/off switch (reed switch) can be locked in the 'on' position if required
- Consistent polar pattern and proximity compensating electronics—excellent tonal consistency; integrated pop filter

MD-441 Versatile Dynamic Mic



The MD 441 is acknowledged as the most accurate and versatile dynamic mic available. It has a nearly textbook perfect supercardioid pattern, a five-position low frequency contour switch, a two-position high frequency switch, a critically dampened internal shock suspension, and a hum bucking coil. Perfect for vocals and virtually every instrument.

- Excellent feedback rejection
- Spring capsule mounting—low sensitivity to handling noise
- Five position bass roll-off switch—compensates for proximity effect
- Integral pop filter
- Brilliance (treble boost) switch—added speech intelligibility

CLASSIC MICROPHONE ACCESSORIES

Grey Foam Windscreen for MD421 II *(MZW421)* **29.95**
Grey Foam Windscreen for MD441 *(MZW441)* **34.95**
Grey Foam Windscreen for MD431 II *(MZW4032)* **36.50**
Lock-on Stand Adapter for MD421 *(MZA421)* **29.95**
Shockmount Stand Adapter for MD421 *(MZS421)* **49.95**
Quick Release Stand Adapter for MD431 *(MZA4031)* **29.95**
Clear Plastic Locking Stand Adapter for MD441 *(MZA441)* **59.95**
Flexible Stand Adapter for MD441 *(MZQ441)* **29.95**

	MD-421 II	MD-431 II	MD-441
Pickup Pattern	Cardioid	Supercardioid	Supercardioid
Frequency Response	30 - 17 kHz	40 - 16 kHz	30 - 20 kHz
Sensitivity	2 mV/Pa ± 3 dB	2.2 mV/Pa ± 2.5 dB	1.8 mV/Pa ± 2 dB
Nominal Impedance	200 Ω	250 Ω	200 Ω
Min Term. Impedance	200 Ω	1000 Ω	1000 Ω
Dimensions	8.4 x 1.9 x 2″	8 x 2 x 1.3″	10.7 x 1.3 x 1.5″
Weight	14 oz.	9 oz.	16 oz.

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SPECIALTY MICROPHONES

MD46 Handheld Dynamic ENG/EFP Microphone



Developed for NBC and the 2000 Sydney Olympics, the MD46 is a cardioid interview mic with studio quality. Unlike omni microphones that pick up everything equally, including background noise, the MD46 focuses on the subject, increasing clarity. However, the tendency for a cardioid mic to emphasize plosives makes them undesirable in the field. To eliminate this problem, Sennheiser put a double layer in the sound inlet basket to reduce the tendency for the microphone to emphasize plosives, enabling the MD 46 to highlight the desired sound (optional windscreen is still available though, if needed).

- Features include a traditional long handle, wind attenuation and off-axis rejection
- Possesses an extended high-frequency response, warm lows, and a clean midrange.
- With low handling and wind noise, and the ability to eliminate vocal popping, it is the ideal interview tool for ENG work.
- Frequency response of 40Hz-18 kHz allows the MD46 to be used in a pinch as an announcer or guest mic in remotes, auxiliary studios or at interview turrets.



MKH-418S Shotgun Microphone

The MKH 418S is a compact MS stereo shotgun microphone utilizing Sennheiser's exclusive RF condenser microphone technology. The MKH418S incorporates a mid capsule based on that of the legendary MKH416 which is combined with a newly designed figure-of-eight side capsule. Its high constant articulation, excellent feedback rejection and functional versatility make the MKH418S a superb all-round cost-effective choice for film, television, and multi-media applications.

MKE-300 Camcorder Microphone



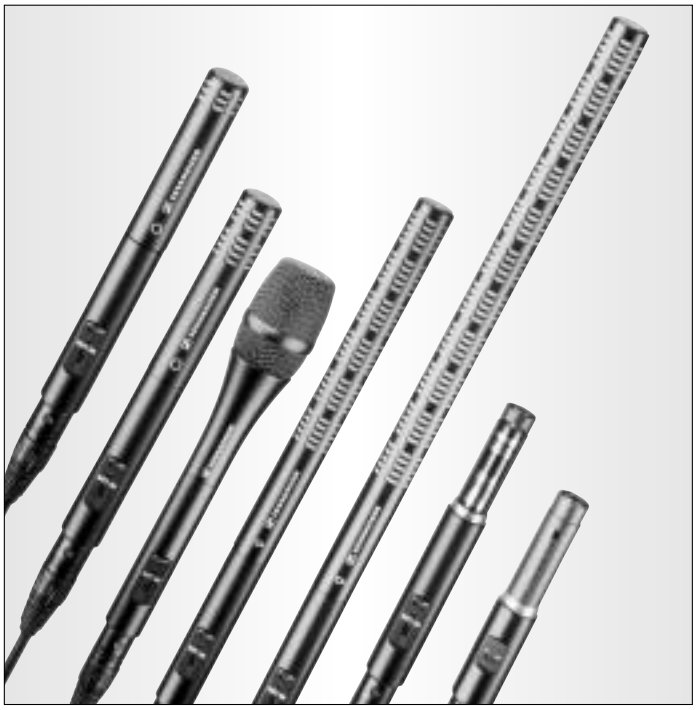
The MKE-300 complements the excellent video capabilities of DV camcorders with the superior audio they deserve. The electret condenser element delivers sonic clarity and high output. Ideally suited for mounting on camcorders due to its integrated shoe assembly, the MKE-300 is compact and extremely lightweight (2.1 oz.). Given its super-cardioid polar pattern, the MKE-300 has the ability to pick up only those sounds that correspond to the scene being filmed and rejects disturbing ambient noise. This high directivity also attenuates the camcorder's autofocus and zoom motors. The mic's suspension mount assembly and integrated windscreen virtually eliminates handling and wind noise. Includes a mini cable.

	MD46	MKH-418S	MKE-300
Pickup Pattern	Cardioid	M: super cardioid/lobar; S: ∞	Supercardioid/lobar
Frequency Response	40Hz - 18kHz	40Hz - 20kHz	150Hz - 17kHz
Sensitivity (1kHz)	2.0 mV/Pa ±2.5dB	M: 25 mV/Pa (-32 dBV); S: 10 mV/Pa (-40 dBV)	16 mV/Pa ±2.5 dB
Nominal Impedance at 1 kHz	350Ω		200Ω
Min Terminating Impedance	1000Ω	1000Ω	2700Ω
Dimensions; Weight	1.9 x 9.8″; 12.7oz.	.75 x 11″; 7.7 oz.	9.1 x 2.6 x 1.96″; 2.1 oz.

K6 SERIES

Modular Electret Condenser Microphone System

The K6 Series is a modular condenser microphone system of exceptionally high quality designed to bring uncompromised studio quality sound and ruggedness to the broadcast and field recording markets. Available with a wide variety of interchangeable modules for different powering arrangements, polar patterns, and capsule sizes, the K6 System provides a flexible tool for all recording and sound reinforcement work. At the heart of the K6 series is the K6 power supply. The K6 is the backend of the multimike system and accommodates a balanced, low impedance 3-pin XLR output connector. The modular design of the K6 power supply allows you to quickly convert from any one of six microphone capsules simply by unscrewing the current capsule from the K6 and replacing it with another. Each capsule has a different polar pattern making the K6 series flexible enough for any situation. Choose from omni-directional to highly directional shotgun mics suitable for boom pole and camera mounting as well as a lavalier capsule ideal for on air talent and live theater applications. A K6 Power Module plus a few mic capsules gives you the flexibility that would otherwise require investing in a number of individual mics.



K6/K6P Powering Modules



The heart of the system, the K6 Module supplies power to any capsule in the series. It can run for 150 hours on a single AA battery, or can be used with 12 to 48v phantom power (AA battery doesn't drain while phantom-powering). The K6 features an on/off switch with LED battery status indicator; and a bass roll-off switch which will compensate for proximity effect in close miking, or for wind rumble in field recording.

The K6P phantom powering module provides an alternative to the K6 that is 42mm shorter and 15 grams lighter, while offering the same integrated bass roll-off switch. Ideal for fixed installations and applications on camera or on stage where size is a factor and phantom power availability is not a concern.

Omni-Directional Lavalier Capsules



MKE2-60

The industry-standard for tv and theatrical production, this sub-miniature omni lavalier is unobtrusive and easy to conceal. Its accurate frequency response reproduces natural, uncolored sound. An extremely thin low mass diaphragm exhibits extraordinary transient response. The stranded, stainless-steel reinforced cable provides maximum tensile strength and low contact (clothing) noise. May also be used as a clip-on instrument mic.



MKE102-60

Similar in size to the MKE2 with several significant differences. The capsule head can easily be detached for use with straight or right angle cables. Highly immune to moisture, it is a natural for theater or outdoor productions. Frequency response is optimized for speech pickup with its 4 to 6 dB rise at 8kHz. Extremely low self-noise makes it ideal for broadcast as well.

MKE104-60

The MKE104 is the same as the MKE102 except it is a cardioid mic, which makes it ideal for applications that require isolation of the source. Use in live musical performances and for reporting when there are several announcers covering a scene with different audio feeds.

K6 SERIES

accessories for lavalier capsules

Windscreens for MKE2
Black Windscreen *(MZW2-A)* **9.95**
Foam Windscreen *(MZW2-F)* **9.95**
Nickel Windscreen *(MZW2-G)* **9.95**

Pin Clips for MKE102/104
Black Pin Clip *(ZH100-ANT)* **5.95**
Grey Pin Clip *(ZH100-GR)* **5.95**
Black Pin Clip *(ZH100-ANT)* **5.95**
Grey Pin Clip *(ZH100-GR)* **5.95**

Clips for Right Angle KA Cables
Black Clip *(MZQ100-ANT)* **3.95**
Beige Clip *(MZQ100-BEI)* **3.95**
Grey Clip *(MZQ100-GR)* **3.95**

Nickel Dual Post Tie Clip *(MZQ222-NI)*
For MKE2, 102, and 104 ..... **11.95**

Stand Off Clip *(S44435)*
Available as add on accessory (black) **2.95**

Gray Magnetic Microphone Holder *(MZM 2/10)*
For MKE2, 102, and 104 ..... **46.95**

Black Steel Mesh Grille
For MKE102 *(MZW102-ANT)* **9.95**
For MKE104 *(MZW104-ANT)* **25.95**

Nickel Steel Mesh Grille
For MKE102 *(MZW102-NI)* **9.95**
For MKE104 *(MZW104-NI)* **25.95**

NB2
The NB2 is a unique device that allows you to adapt any Sennheiser lavalier for use as a headworn microphone. Comfortable and rugged, the NB2 fits any head **54.95**

SEMZ-2
Accessory kit for MKE-2. (It includes the MZQ222-NI and MZQ222 dual post tie clips, MZW2-A and MZW2-G windscreens, ZH100-ANT, and ZH100-BEI **37.50**

MICROPHONE CAPSULES

The K6 Series ME capsules exhibit exceptionally high sensitivity, great "reach" characteristics, and very low self-noise. The result is a versatile line, from omnidirectional to long shotgun models, that offers extremely natural sound reproduction for digital recording and other critical applications.

ME-62 Omni-Directional Capsule



Very broad and smooth frequency response, without proximity effect. The ME62's low handling noise and integrated pop screen make it ideal for interviews and live recording. Great for the recording of ambient sounds. Frequency response: 20Hz-20kHz ±3 dB **139.95**

ME-64 Cardioid Capsule



Feedback resistant due to the ME-64's well defined directional polar pattern as well as its extended frequency response, make this microphone capsule ideal for use in sound reinforcement or recording in noisy environments. Frequency response: 50Hz - 20kHz ± 2.5 dB **159.95**

ME-65 Super Cardioid Capsule



The ME-65 is a super-cardioid handheld microphone especially suited to vocal and speech applications. It has a matte black, anodized, scratch-resistant finish that will look great on camera. The linear directivity across it's entire frequency range will ensure excellent feedback rejection. Can handle extremely high sound pressure levels, has high sensitivity, a wide frequency range and integrated pop filter. Frequency response: 40Hz - 20kHz ± 2.5 dB **179.95**

ME-66 Short Shotgun Capsule



The ME66 offers classic Sennheiser shotgun mic sound, extremely high sensitivity, and very low-self-noise, resulting in unprecedented "reach" characteristics for a mic that can be powered by AA batteries. Extremely smooth frequency response provides for neutral, uncolored sound in the low- and mid-range frequencies and a gentle boost in high frequencies, for increased intelligibility. Great for interviews in crowded situations, as a camera microphone for ENG, for unobtrusive theater sound reinforcement and as a podium mic. Frequency response is 50Hz - 20kHz ± 2.5 dB **209.95**

ME-67 Long Shotgun Capsule



The ME-67 is a long shotgun mic capsule designed primarily for capturing low-level sounds. Its extremely directional design brings distant sound sources into close range, with the extremely tight focus of the on-axis signal. As with the ME66, its frequency response has been optimized for a natural presentation, with good articulation of high frequencies. With a precise combination of high sensitivity and low self-noise, the ME67 is outstanding in difficult acoustic environments. Frequency response: 40Hz - 20kHz ± 2.5 dB **239.95**

MKH SERIES

RF Condenser Microphones

MKH condenser mics represent the perfect fusion between exceptional sound and precision engineering, and have long been the choice of the most demanding professionals in the broadcast, theatrical, and motion picture industries. They are specifically well-suited for use with digital recording devices, which reveal the sonic limitations of less accurate microphones. Their extremely transparent sound, high sensitivity, and excellent signal-to-noise ratio are perfectly complemented by their exceptionally low inherent self-noise, extremely low distortion, and unusually flat, linear frequency response.
Their lightweight, durable, and weather-resistant construction makes them suitable for outdoor applications. The MKH Series consists of eight microphones with different pickup patterns, and are tonally consistent across the entire range because of their lack of coloration. In every MKH mic, the capsule is part of a tuned circuit in a radio frequency system. This makes all of them highly immune to humidity, and is responsible for their exceptionally low self-noise. In addition, their symmetrical push-pull capsule design incorporates optimum resistive loading, which virtually eliminates intermodulation distortion.



MKH-20 *Omni-Directional*

Low distortion push-pull element, transformerless RF condenser, flat frequency response, diffuse/near-field response switch (6 dB boost at 10 KHz), switchable 10dB pad to prevent overmodulation. Handles 142dB SPL with a high output level. Ideal for concert, Mid-Side (M-S), acoustic strings, brass and wind instrument recording.

MKH-30 *Figure-Eight*

The symmetrical transducer design of the MKH-30 produces an unusually wide and smooth frequency response. Particularly suited to soloists and small instrumental groups, it can also serve as an 'S' or side microphone in MS stereo applications. As a support microphone it is particularly useful in the effective suppression of sounds from adjacent sound sources.

MKH-40 *Cardioid*

Highly versatile, low distortion push-pull element, transformerless RF condenser, high output level, transparent response, switchable proximity equalization (-4dB at 50Hz) and pre-attenuation of 10dB to prevent overmodulation. Excellent results have been achieved in vocal applications when used with a pop screen. Recommended for most situations, including digital recording, overdubbing vocals, percussion, acoustic guitars, piano, brass and string instruments, mid-side (M-S) stereo, and conventional X-Y stereo.

MKH-50 *Super-Cardioid*

The super-cardioid MKH-50 offers a higher attenuation of side and rear sounds than the cardioid MKH-40. It is principally designed for use as a soloist's or spot microphone for applications requiring a high degree of side-borne sound muting and feedback rejection while maintaining the integrity of on- and off-axis sound fields.

MKH-60 *Super-Cardioid/Lobe*

Short interference tube RF condenser, lightweight metal alloy, transformerless, low noise, symmetrical capsule design, smooth off-axis frequency response, switchable low cut filter (-5dB at 100 Hz), high frequency boost (+5dB at 10kHz) and 10dB attenuation. Handles very high SPL (135 dB). Widely used in film, theater and TV productions. Ideal for broadcasting, sports recording and interviews in crowded or noisy environments. Excellent for studio voice-overs.

MKH-70 *Super-Cardioid/Lobe*

Extremely lightweight, rugged, long shotgun. Low distortion push-pull element, transformerless, low noise, switchable presence (+5dB at 10kHz), low cut filter (-5dB at 50Hz), and 10dB pad. Handles 133dB SPL with excellent sensitivity and high output level. Ideal for video/film studios, theater, sporting events, and nature recordings.

MKH-800 *Variable Pattern Condenser Mic*

Superior microphone for any recording application, the MKH-800 offers high clarity and the absolute minimum coloration. Frequency response on the high-end extends to 50kHz—far beyond the audible threshold—resulting in superior transient response. In addition to five switchable pick-up patterns (omni, wide cardioid, cardioid, supercardioid and figure-8), it features include a built-in LED for on-axis alignment, switchable treble emphasis, roll-off filter and pre-attenuation pad.

MKH-416P48-U3 *Super-Cardioid/Lobe*

Transformerless, RF condenser designed as a combination of pressure gradient and interference tube microphones. Very good feedback rejection, low proximity effect, 128dB SPL. Rugged and resistant to changing climate conditions. Ideal for boom, fishpole, and camera mounting as a long-distance microphone for video, film, and studio recording. An excellent ENG mic for reporters or as a podium or lectern microphone.

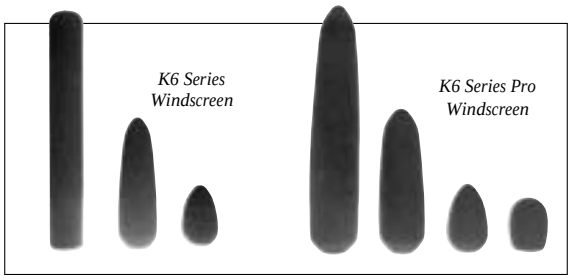
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MKH SERIES ACCESSORIES

K6 Series Windscreen

Standard foam windscreens reduce wind and popping noise.

MZW-64 for ME-62, ME-64, Grey **22.95**
MZW-66 for ME-66, Grey.....**36.95**
MZW-67 for ME-67, Grey.....**42.95**



K6 Series Pro Windscreen

Black velour coated foam windscreens offer even greater protection for reducing wind and popping noise.

MZW-64PRO for ME-62/64 .....**29.95**
MZW-65PRO for ME-65.....**64.95**
MZW-66PRO for ME-66.....**64.95**
MZW-67PRO for ME-67.....**96.50**

MKH Series Windscreen

Open cell foam material represents the best compromise between minimal effects on the frequency response and maximum protection against wind and pop noise.

MZW-41 for MKH-20/40, grey **8.95**
MZW-61 for MKH-60, black.....**82.95**
MZW-71 for MKH-70, black.....**129.95**
MZW-415 for MKH-416, grey **42.95**

K6 Series Mic Adapters



MZQ6 Camera Mount **38.95**
MZQ200 Stand Mount.....**28.50**
MZS6 Shock Mount **36.95**

Elastic Suspension Shock Mount *(MZSCAM)*
With shoe mount for camera use.....**39.95**



MZA14-P48
Phantom Power Supply

Battery operated, 48v phantom power supply with XLR balanced input/output as well as battery test, -10/-20dB pad and 80/140Hz high pass filter controls **383.95**

MZS20-1 Pistol Grip Shock Mount

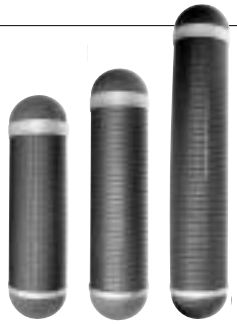
MZS20-1 Combination shock mount, pistol grip and stand adapter for ME-66 and ME-67 as well as MKH Series Microphones



$259.95

- Required for mounting in MZW "blimp" windscreen
- May be boom or stand mounted. Requires MZA-216 adapter (included).
- Includes interchangeable mike clips for MKH-416/ MKH-816, or MKH-20 through MKH-70

Blimp Windshields
(Require MZS20-1)



MZW20-1: Rigid "blimp" windscreen for MKH-20/30/40/50.....**229.95**

MZW60-1: Rigid "blimp" windscreen for ME66/MKH-416/MKH-60.....**229.95**

MZW70-1: Rigid "blimp" windscreen for ME-67/MKH-70.....**269.95**

Long Hair Wind Muffs

Long hair wind muffs greatly reduce high velocity wind-noise when used with MZW series blimp windshields and MZS-20-1 pistol grip/shock mount.



MZH20-1: Requires MZW 20-1 "blimp" windscreen (for MKH-20/30/40/50) **159.95**

MZH60-1: Requires MZW 60-1 "blimp" windscreen (for ME-66, MKH-416/MKH-60).....**149.95**

MZH70-1: Requires MZW70-1 "blimp" windscreen (for ME-66/ME-67 and MKH-70 .....**169.95**

Adapters and Clips

Piggyback Mic Holder *(MZD30)*
For any combination of two MKH20, 30, 40 or 50 .....**54.95**

Swivel-Mount Yoke *(MZG415-3)*
For MZS shock mounts **48.95**

Double Joint Flexible Stand Adapter *(MZQ31)*
Fits MKH20, 30, 40, and 50 .....**69.95**

Flexible "Quick-Release" Stand Adapter *(MZQ40)*
For MKH20, 40, 50 and 60 .....**33.95**

Flexible "Quick-Release" Stand Adapter *(MZQ100)*
For 19-23 mm (approx. 3/4″ to 7/8″) diameter microphones such as the MKH416 .....**25.95**

Shock Mounts

Elastic Suspension Shock Mount *(MZS40)*
For MKH20, 40, and 50.....**169.95**

Shock Mount *(MZS80)*
For MKH80 and MKH30.....**188.95**

SHURE

'PG' PERFORMANCE GEAR SERIES

Affordable, application-specific microphones. Ideal for everyday use and practice. The perfect introduction to Shure engineering excellence for performers ready to plug in and play out. Performance Gear Microphones provide a complete vocal, instrument and drum selection and feature many of the same durability and performance attributes of other Shure microphones at an attractive price. Real gear without compromise. As serious as you want to be.





PG48 and PG58 Cardioid Dynamic Speech and Vocal Microphones

High performance dynamic cardioid microphones tuned specifically for vocal applications, the PG48 is designed for spoken word applications, while the rugged PG58 is tuned to accentuate the clarity of lead and back-up vocals. Both feature a tailored frequency response that is smooth and extended. Their cardioid polar pattern picks up the most sound from in front of the microphone and some sound from the sides. Less susceptible to feedback in high volume settings.

- Dynamic cartridge has a simple, rugged coil. Handles extreme volume levels without distortion.
- Cartridge includes a neodymium magnet for high output level
- Internal shock mount for reduced handling noise (PG58 only)
- Durable metal construction
- On/off switch for onstage control
- Steel mesh ball grille with integral "pop" filter reduces explosive breath sounds and wind noise
- Includes 15´ cable, break-resistant mic clip and storage bag

PG57 and PG81 Cardioid Dynamic and Cardioid Instrument Microphones

A versatile microphone for amplified or acoustic instruments, the PG57 is a cardioid dynamic mic with a tailored frequency response that is smooth and extended. Tuned specifically for instrument applications. A sensitive, flat response microphone ideal for acoustic instruments, the PG81 is a cardioid condenser mic with a flat frequency response that is wide and uniform. Provides consistent sensitivity through the frequency range.

- Cardioid polar pattern picks up the most sound from in front of the microphone and some sound from the sides. Less susceptible to feedback in high volume settings.
- Internal shock mount for reduced handling noise
- Durable metal construction
- Dynamic cartridge has a simple, rugged coil. Handles extreme volume levels without distortion. Cartridge includes a neodymium magnet for high output level. (PG57)
- Hardened steel mesh tapered grille that resists wear and abuse (PG57)
- On/off switch for onstage control
- Condenser cartridge has a lightweight, sensitive diaphragm. Precisely and smoothly captures sound nuances. Powered by AA battery or phantom power supply. (PG81)
- They include 15´ cable, break-resistant mic clip and storage bag



'PG' PERFORMANCE GEAR SERIES

When you buy a Shure microphone you are buying uncompromised precision, performance, ruggedness and reliability in "real world" usage. To back it up, Shure has the most stringent quality assurance testing program in the business. Before it can bear the Shure name, a mic must be able to withstand extended exposure to extreme temperatures (-20° to 165°F) and humidity (100% at room temperature), ultraviolet rays, salt spray, alcohol, sand and water. Also, violent vibrations, "drop tests" (repeated drops from six feet) and ten days of repeated cycling between simulated arctic and tropical environments. If the mic doesn't work after all this, it's not a Shure. This dedication has made Shure "The Sound of the Professionals Worldwide" for decades. And that's why you'll find Shure microphones virtually everywhere—on stages, lecterns, remote broadcast locations and production sets all over the world.



PG52 and PG56 Drum Mics

Cardioid dynamic mics with a tailored response, the PG52 is a high-performance microphone tuned to capture low-end punch (kick drum, bass amp) while the PG56 is a compact drum microphone for close miking of snare drums, rack/floor toms and percussion instruments.

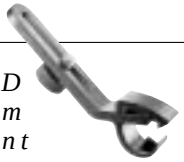
- Cardioid polar pattern picks up the most sound from in front of the mics and some sound from the sides. Less susceptible to feedback in high volume settings
- Tailored frequency response is smooth and extended
- Dynamic cartridge has a simple, rugged coil. Handles extreme volume levels without distortion
- Cartridge includes a neodymium magnet for high output level
- Durable metal construction
- Integral stand mount
- Internal shock mount for reduced handling noise
- Hardened steel mesh tapered grille that resists wear and abuse
- They include 15´ cable and bag

PGDMK4 Drum Mic Kit



An essential drum mic package. The PGDMK4 drum mic kit includes three PG56 snare/tom microphones, one PG52 kick drum mic, three A50D drum mounts, four 15´ XLR-XLR cables and a durable carrying case.

A50D Drum Mount



Can be used on most standard drum rims. The flexible, yet sturdy rubber construction provides stability and ease of installation.

PGDMK6 Drum Mic Kit



A complete drum mic package. The PGDMK6 drum mic kit includes three PG56 snare/tom microphones, one PG52 kick drum microphone, two PG81 instrument mics, three A50D drum mounts, six 15´ XLR-XLR cables and a durable carrying case.

	PG48	PG58	PG57	PG81	PG52	PG56
Microphone Element Type	Dynamic	Dynamic	Dynamic	Condenser	Dynamic	Dynamic
Polar Pattern	◠	◠	◠	◠	◠	◠
Frequency Response (Hz)	70 to 15k	60 to 15k	50 to 15k	40 to 18k	30 to 13k	50 to 15k
Output Impedance @1000Hz	600 Ω	300 Ω	200 Ω	600 Ω	300 Ω	200 Ω
Sensitivity @1000Hz*	-52 dBv/Pa	-53 dBv/Pa	-53 dBv/Pa	-48 dBV/Pa	-55 dBv/Pa	-56 dBV/Pa
Electromagnetic Hum Sensitivity**	22 dB	26 dB	26 dB	-7 dB	25 dB	26 dB
Weight Net	11.1 oz.	11.3 oz.	11.5 oz.	8.8 oz.	16.8 oz.	11.4 oz.

**1 Pascal = 94 dB SPL **equivalent SPL in a 1 millioersted field (60 Hz)*

SM SERIES PERFORMANCE MICROPHONES

SM48/SM48s Dynamic Cardioid Vocal Microphones



The SM48 is a unidirectional dynamic microphone designed for professional sound reinforcement, studio recording, and broadcasting applications. It maintains a true cardioid pattern throughout the frequency range, ensuring high gain before feedback and rejection of off-axis sound. Perfect for singers on the way up, the tailored frequency response is ideal for vocals. A presence rise brightens mid-range performance, and a low frequency rolloff controls proximity effect. Includes a shock mounted cartridge, a steel mesh grille, and an integral "pop" filter. The SM48S adds a lockable On/Off switch.

- Cardioid pickup pattern rejects off–axis sound and provides superior gain before feedback
- Shock-mounted cartridge for exceptional ruggedness and reduced handling noise
- Frequency response tailored for vocals with brightened mid-range. Low-frequency bass rolloff controls proximity effect
- 55Hz to 15kHz frequency response
- Built-in "pop" filter reduces explosive breath sounds and wind noise
- Lockable On/Off switch (SM48S)
- Adjustable stand adapter & carrying case included

SM48 Dynamic Microphone

SM48s Dynamic Microphone with on/off switch

SM48-LC Dynamic Mic without a cable

SM48S-LC Dynamic Mic with on/off switch, no cable



SM-57 Dynamic Cardioid Vocal/Instrument Mic

A true classic, the SM-57's outstanding performance and diversity make it the "workhorse" of stages and studios worldwide. Ideal for musical instrument pickup or for vocals, the SM-57's carefully contoured frequency response means clean, well-defined instrument reproduction and rich, warm vocal pickup on lecterns or stages. Extremely effective cardioid pickup pattern isolates the main sound source while minimizing background noise. In the studio use it to record drums, guitar, and woodwinds. It is also used on the podium of the U.S. President for over 30 years.

- Contoured frequency response for clean, instrumental reproduction and rich vocal pickup
- Professional-quality reproduction for drum, percussion, and instrument amplifier miking
- Ideal for narration/dialogue, vocal music and rhythm pickups
- Unusually effective cardioid pattern isolates the main sound while minimizing reflections caused by poor studio or location acoustics and background noise
- Shock-mounted cartridge reduces mic handling noise and mic stand rumble
- Extremely durable under the heaviest use
- Frequency response: 40 to 15k Hz
- Designed for both field and studio use —hand-held or on a stand. Includes a break-resistant swivel adapter that rotates 180 °

	SM48	SM57	SM58	SM87A
Microphone Element Type	Dynamic	Dynamic	Dynamic	Condenser
Frequency Response (Hz)	55 to 14k	40 to 15k	50 to 15k	50 to 20k
Polar Pattern	◡	◡	◡	◡
Rated Impedance	150Ω	150Ω	150Ω	150Ω
Self Noise	—	—	—	24 dB
Maximum SPL	—	—	—	140.5 dB
Dynamic Range	—	—	—	115.5 dB
Recommended Windscreen	A58WS	A2WS	A58WS	A85WS

SM SERIES PERFORMANCE MICROPHONES

SM-58 Dynamic Cardioid Vocal Microphone



The SM58 is a unidirectional (cardioid) dynamic vocal microphone designed for professional vocal use in live performance, sound reinforcement, and studio recording. Cardioid pickup pattern isolates the main sound source while minimizing unwanted background noise. In addition, it has a highly effective, built-in spherical filter that minimizes wind and breath "pop" noise. The SM58 has a tailored vocal response for a sound which is a world standard. Rugged construction, a proven shock mount system, and a steel mesh grille ensure that even with rough handling, the SM58 will perform consistently. Outdoors or indoors, singing or speech – the SM58 is the overwhelming choice of professionals worldwide.

- Unusually effective cardioid pickup pattern minimizes background noise and undesirable effects of studio and location acoustics. Rear and side rejection uniform to very low frequencies and completely symmetrical about microphone axis
- Shock-mounted cartridge reduces mic handling noise and mic stand rumble
- Designed for both field and studio use — hand-held or on a stand. Includes slip-in swivel adapter
- Self-contained spherical filter controls explosive breath sounds ("pop") and wind noise in outdoor locations

SM58
SM58LC without a cable
SM58CN with a 25´ XLR cable
SM58S with on-off switch, no cable



SM-86/SM-87A Cardioid/Supercardioid Condenser Vocal Microphones

Ideal for professional use in live performance with stage or personal in-ear monitors, the SM-86 is an extremely rugged microphone designed to withstand the rigors of touring while delivering studio-quality sound. The tailored frequency response of the SM-86 reproduces vocals with clarity. A cardioid pickup pattern isolates the main sound source while minimizing unwanted background noise. The built-in three-point shock mount minimizes handling noise, and a two-stage pop filter reduces wind and breath "pop" noise.

The choice of the world's top entertainers, the SM87A features a smooth frequency response that is tailored for warm, rich vocal reproduction. Supercardioid polar pattern permits high gain without feedback and rejects unwanted sounds. A three-stage wind/pop filter minimizes breath noise and an excellent internal shock mount minimizes mic stand rumble and handling noise.

SM-86 Features

- Combines the ruggedness of Shure's SM line with the studio quality audio of a Shure condenser—the first model to join the SM line in over 10 years.
- Cardioid polar pattern and a wide frequency response of 50 Hz to 18 kHz. The mic additionally delivers high gain-before-feedback and a tailored frequency response for a clear reproduction of vocals.
- Equipped with an integral two-stage windscreen and pop filter
- Built-in, three-point shock mount virtually eliminates stand and handling noise
- Housed in a rugged, silver-colored, enamel-painted enclosure incorporating a steel-mesh grille.
- Requires phantom power for operation

SM-87A Features

- Flat frequency response for greater gain before feedback
- Low-frequency roll-off reduces low-frequency rumble and handling noise and compensates for proximity effect
- Low RF and magnetic hum susceptibility
- Low distortion output and wide dynamic range for a variety of load impedances
- Operates on 11 to 52v DC phantom power

BETA SERIES MICROPHONES



Shure's BETA series includes ten premium microphones—some of which have proven legendary performance. The BETA series' outstanding responsiveness to vocals and instruments, specifically their maximum isolation from other sound sources and minimum off-axis coloration, have made them the most respected and widely used microphones in the world. For example, the Beta 58A is unequivocally the best dynamic mic for demanding live vocal applications while the Beta 57A excels in instrument miking applications. They both employ a super-efficient neodymium magnetic circuit design that provides maximum output voltage for improved S/N ratio. All Beta Series mics offer Shure reliability for years of trouble-free performance.

Beta 57A • Beta 58A
Instrument and Vocal Dynamic Supercardioid Microphones



Beta 57A *Beta 58A*

Designed for professional sound reinforcement and project studio recording, the high output Beta 57A provides optimal warmth and presence making it excellent for acoustic and electric instruments including drums, guitar amplifiers, brass and woodwinds, as well as for vocals. Grille design lets you take better advantage of proximity effect.

The Beta 58A is a high-output dynamic vocal mic designed for professional sound reinforcement and project studio recording. The Beta 58A has a shaped frequency response that is ideal for close-up vocals. And the superb performance of this mic is not affected by rough handling because of its rugged construction, pneumatic shock mount system, and hardened steel mesh grille. Typical applications include lead vocals, backup vocals and speech.

Both the Beta 57A and Beta 58A maintain a true supercardioid pattern throughout their frequency range. This ensures high gain before feedback, maximum isolation from other sound sources, and minimum off–axis tone coloration. Both also incorporate a Neodymium magnet for high signal-to-noise ratio output, and an advanced electro-pneumatic shock mount design to effectively minimize transmission of handling or stand noise.

Beta 57A

- Tailored frequency response captures drums, guitars, vocals, and horns with studio quality sound
- Hardened steel mesh grille facilitates use of proximity effect and resists wear and abuse

Beta 58A

- Wide and smooth frequency response adds extra clarity
- Humbucking coil permits use in strong hum fields produced by lighting
- Plug element pins are gold plated for durable and low-noise transfer

	Beta 57A	Beta 58A	Beta 87A	Beta 87C	Beta 53
Microphone Element Type	Dynamic	Dynamic	Condenser	Condenser	Condenser
Polar Pattern	◌	◌	◌	◌	○
Frequency Response (Hz)	50 to 16k	50 to 16k	50 to 20k	50 to 20k	20 to 20k
Rated Impedance	150Ω	150Ω	150Ω	150Ω	150Ω
Output Level Sensitivity[1]	-51 dBV/Pa	-51 dBV/Pa	-54 dBv/Pa	-51 dBV/Pa	-55 dBV/Pa
Maximum SPL[2]	—	—	142 dB	139 dB	143 dB
Self Noise[3]	—	—	25 dB	22 dB	—
Signal-to-Noise Ratio[4]	—	—	69 dB	72 dB	54 dB
Dynamic Range	—	—	117 dB	117 dB	103 dB
Recommended Windscreen	A57AWS	A58WS Series	A58WS Series	A58WS Series	—

[1] 1 Pa = 94 dB SPL *[2] At 1kHz (0.25% THD, 1000 ohm load)* *[3] Typical, A-weighted* *[4] At 94 dB SPL*

BETA SERIES MICROPHONES

Beta 87A • Beta 87C

Supercardioid & Cardioid Condenser Vocal Mics



The choice of performers and audio technicians around the world, the Beta 87A and Beta 87C are premium quality vocal microphones with superb accuracy, detail, and rich vocal production. Used in professional sound reinforcement, broadcasting, and studio recording applications, they combine superb performance with the ruggedness required for touring and field production.

*The **Beta 87A (**supercardioid) and **Beta 87C** (cardioid) provide maximum isolation while offering high gain before feedback. Both feature a controlled low–frequency roll-off that compensates for proximity effect and prevents the "boomy" sound often associated with close-up use. Ideal for live sound and music production, they maintain performance characteristics even at very high sound pressure levels (SPL). Additional features include improved shock mounting and a hardened, dent-resistant steel mesh grille with a built-in pop filter.*

FEATURES

Beta 87A Only

The Beta 87A's supercardioid pattern provides maximum isolation from other vocalists or instruments while offering minimal off-axis coloration and high gain before feedback. It has an exceptionally smooth frequency response with gradual presence rise that brighten the upper midrange.

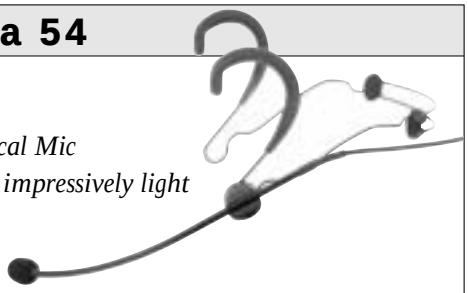
Beta 87C Only

The Beta 87C maintains its cardioid pattern throughout its frequency range, ensuring maximum isolation from other sound sources and high gain before feedback. Vocalists who use in-ear monitor systems will appreciate how effectively the Beta 87C rejects ambient sound from the rear. Exceptionally smooth, wide frequency response includes a slight presence rise for a warm, natural sound.

They Both Feature

- Superior gain before feedback
- Low-frequency roll-off compensates for proximity effect
- Advanced cartridge shock-mount system reduces handling noise and absorbs mechanical shocks
- Very low susceptibility to RFI and electromagnetic hum
- Phantom powered (11-52v DC)
- Built-in three-stage pop filter minimizes breath pop and wind noise
- High S/N ratio of 69dB (Beta 87A) and 72dB (Beta 87C)
- Maximum SPL of 142dB (Beta 87A) and 139dB (Beta 87C)
- Wide dynamic range (117dB) and low distortion characteristics.
- Thick handle for a comfortable feel when hand-held. They include a break-resistant swivel adapter
- Shure ruggedness and reliability

Beta 54



Designed for active professional sound applications, the Beta 54 Headworn Vocal Mic provides incredible sound quality in an impressively light and inconspicuous design. The Beta 54 features an extended frequency response, omnidirection polar pattern, and optimized sensitivity perfect for close microphone placements.

- Adjustable headband, detachable boom mount, and flexible wire frame for maximum comfort and optimum mic placement
- Comfortable earpieces made with soft, latex-free "Dynaflex"
- Corrosion resistant and easy-to-clean
- Interchangeable frequency response cap (two types included) can be replaced if ever clogged by sweat or make-up
- Ideal for use in theatre, event production, trade shows, and seminars

WBH54B Black, with detachable boom mount and flexible wire frame

WBH54BX Black, for wireless use

WBH54T Tan, with detachable boom mount and flexible wire frame

WBH54TX Tan, for wireless use

BETA SERIES MICROPHONES

Beta 98 Miniature Supercardioid Condenser Mic



For toms, snares and percussion instruments, no other mic compares to the Beta 98. A versatile, high performance mini-condenser microphone, the Beta 98 excels at professional sound reinforcement or recording. High maximum sound pressure level (SPL) of 160 dB makes it ideal for a variety of acoustic instruments, including drums, percussion, piano, reed, wind, and string instruments. It can also be used for amplified guitars. The Beta 98 has an ultra-uniform supercardioid pickup pattern that provides high gain before feedback and excellent rejection of unwanted noise. It also includes an in-line preamplifier that accepts 48v DC phantom power. Finally, An optional cardioid cartridge is also available, allowing you to easily adapt the mic to different applications.



The A98D drum mount has a specially designed gooseneck that allows the microphone to be positioned freely, yet holds it securely in place once the ideal placement has been found.

The hardware has been redesigned for quicker, more secure, rim-mounting without the need for a drum key.

Flexible gooseneck lets you position the mic in practically any configuration, so you get a virtually unlimited choice of sound options. Its unobtrusive design and compact size help the Beta 98D/S reduce stage clutter to an absolute minimum.

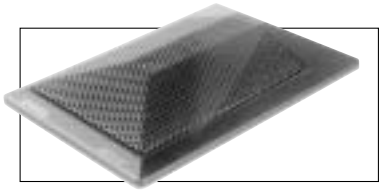
The Beta 98 is available in two variations:
BETA 98/S ***and*** BETA 98 D/S

- **Beta 98/S:** Includes microphone, 25´ lightweight cable, in-line preamplifier, storage bag and shockmount swivel adapter
- **Beta 98D/S:** Includes microphone, detachable 15´ heavy-duty Triple-Flex cable, in-line preamplifier, storage bag and A98D drum mount

Optional Accessories

RPM108 Cardioid cartridge.....**42.95**
A412MWS Metal Locking Windscreen.....**19.95**

Beta 91 Cardioid Condenser Kick Drum Mic



A high performance condenser microphone with a tailored frequency response designed specifically for kick drums and other bass instruments, the Beta 91 combines superior "attack" and "punch" for studio quality sound – even at the extremely high sound pressure levels encountered inside a kick drum. Ideal for live concert applications.

- Boundary effect design produces a strong, solid low-end response specifically tailored for heavy bass applications.



- Wide dynamic range for use in high SPL environments – up to 161dB
- Half-cardioid pattern throughout the frequency range insures high gain before feedback and excellent rejection of unwanted sound
- Low profile design eliminates the need for external mounting hardware
- Includes 15´ heavy-duty Triple-Flex cable and case

Beta 98H/C Mini Condenser Cardioid Mic



Designed specifically for brass and wind instruments, the Beta 98H/C is a high SPL mini-condenser microphone that's ideal for live performances. With its integrated gooseneck and clip-on clamp design, you will experience excellent sound quality with extreme flexibility.





- Cardioid pattern allows maximum sound before feedback
- Low-profile design, requires no external mounting hardware
- Gooseneck with clip-on clamp provides secure fit and reliable hold to virtually any wind, brass, or percussion instrument
- Small, low-profile design provides low visibility and minimizes stage clutter.

BETA SERIES MICROPHONES

Beta 52A & Beta 56A
Supercardioid Dynamic Instrument Mics



*The **Beta 52A** is a high output dynamic mic with a tailored frequency response designed specifically for kick drums and other bass instruments. It provides superb attack and "punch", and delivers studio quality sound even at extremely high sound pressure levels.*

*Specifically engineered for drum and instrument applications, the **Beta 56A's** compact, pivoting design makes it the ideal choice for close miking of tom-toms and other percussion instruments, as well as guitar amplifiers, brass instruments, and woodwinds.*

Designed for professional sound reinforcement and project studio recording, the Beta 52A's modified supercardioid and the Beta 56A's extremely uniform supercardioid pickup pattern provides high gain before feedback and excellent rejection of unwanted noise. Both feature a built-in dynamic locking stand adapter with an integral XLR connector to simplify installation and keep them in place even when subjected to sharp blows and strong vibrations. A hardened steel mesh grille protects them from the wear and abuse associated with touring.

Beta 52A Features

- Frequency response shaped specifically for kick drums and bass instruments
- Built–in dynamic locking stand adapter with integral XLR connector simplifies setup, especially inside a kick drum
- Studio quality performance, even at extremely high sound pressure levels
- Supercardioid pattern for high gain before feedback and superior rejection of unwanted noise
- Hardened steel mesh grille that resists wear and abuse
- Advanced pneumatic shock mount system that minimizes transmission of mechanical noise and vibration
- Neodymium magnet for high signal–to–noise ratio output
- Low sensitivity to varying load impedance

Beta 56A Features

- Tailored frequency response provides drums, amplified instruments, and horns with studio quality sound
- Built–in stand adapter with dynamic locking system and XLR connector simplifies setup and provides greater flexibility
- Uniform supercardioid pattern for high gain before feedback and superior rejection of off–axis sound
- Compact design reduces stage clutter
- Hardened steel mesh grille resists wear and abuse
- Neodymium magnet for high signal–to–noise ratio output
- Minimally affected by varying load impedance
- Advanced pneumatic shock mount system that minimizes transmission of mechanical noise and vibration

	Beta 98	Beta 98H/C	Beta 91	Beta 52A	Beta 56A
Microphone Element Type	Condenser	Condenser	Condenser (Electret Bias)	Dynamic	Dynamic
Polar Pattern	◌	◌	◌	◌	◌
Frequency Response (Hz)	20 to 20k	20 to 20k	20 to 20k	20 to 10k	50 to 16k
Output Impedance	150Ω	150Ω	150Ω	150Ω	150Ω
Output Level Sensitivity[1]	-59 dBV/Pa	-56 dBv/Pa	-59 dBV/Pa	-64 dBV/Pa	-51 dBV/Pa
Maximum SPL, 1kΩ[2]	160 dB	155 dB	156 dB	174 dB	—
Output Noise[3]	35 dB SPL	31 dB SPL	35 dB SPL	—	—
Signal-to-Noise Ratio[4]	59 dB	63 dB	59 dB	—	—
Dynamic Range	125 dB	112.5 dB	125 dB	—	—
Recommended Windscreen	RK183WS	RK183WS	—	—	A1WS

[1] 1 Pa = 94 dB SPL [2] at 1000Hz (0.25% THD, 1000 ohm load) [3] typical, A-weighted/weighted per DIN 45 405 [4] at 94 dB SPL

LIVE PERFORMANCE MICROPHONES

WH20/WH30 Headworn Cardioid Mics



The WH20 (dynamic) and WH30 (condenser) are rugged, lightweight headset mics with high-quality voice pickup. They fit securely for active users, such as aerobics instructors and musicians, with low visibility. Also suitable for any voice communications application where comfort and reliability are required.

- Cardioid pattern provides greater gain-before-feedback and isolation from ambient noise or unwanted sound like air-conditioning
- Carefully shaped frequency response includes a voice-frequency presence peak. Low-frequency rolloff compensates for the proximity effect inherent in cardioid microphones during close-talk operation
- Integrated elastic CROAKIES headband comfortably secures the headset. The headband, wireframe, and gooseneck are adjustable.
- Extra-strong microphone cable resists breakage. Collapsible mic boom for easy storage and transportation. Supplied with foam windscreen and clothing clip

WH20TQG with Tini Q-G connector for wireless applications.....**79.50**
WH20XLR with XLR-M connector and detachable belt clip **74.95**

WH30TQG with Tini Q-G connector for wireless applications...**101.95**
WH30XLR with XLR-M connector and detachable belt clip **163.95**

Classic Microphones

520DX Harmonica Microphone



The definitive blues harp voice, the 520DX "Green Bullet" is a cardioid, dynamic harmonica microphone with a signature green and chrome die-cast case, contoured to fit perfectly between harmonica and hands. Comes with an attached cable

55SH Series II Vocal and Speech Mic



Featuring a signature chrome-plated die-cast case with a vintage '50s and '60s look and feel, the 55SH Series II is a cardioid dynamic vocal and speech mic with a tailored response (50 Hz to 15 kHz) that is deal for PA or theater-stage sound systems, as well as for broadcast and recording. Low-impedance balanced output designed for connection to mic inputs rated at 75 to 300 ohms. Cartridge shock mount reduces stand noise. Has an attached, self-tensioning, 5/8″-27 thread swivel mount. Doesn't come with a cable.

512 Cardioid Dynamic Headworn Mic



Designed for sports and news announcing, communication systems, and remote broadcasting, the 512 consists of the SM10A mic element mounted on an adjustable lightweight headband with a single earphone. By combining an efficient "open-air" headphone with a close-talking dynamic microphone, the 512 offers convenient hands-free operation, and a lightweight frame to reduce fatigue.

- 50 to 15,000 Hz frequency range
- Cardioid pattern reduces unwanted background noise, permits use in noisy environments
- Mic characteristics are tightly controlled resulting in consistent response and output. Headphone response is tailored to enhance voice intelligibility
- Includes an attached 7′ cable, foam windscreen to reduce explosive "popping" sounds, and a cable clip to secure the cable to clothing

SM10A/SM12A Cardioid Dynamic Headworn Mics



Like the 512, the SM10A and SM12A are headworn cardioid dynamic microphones designed for close-talk applications such as remote-site sports broadcasting and corporate intercom systems. Also ideal for singing drummers, keyboard artists, and other performers who need a head-worn vocal mic with superior noise rejection and a smooth, natural-sounding frequency response. The SM12A has the same features and performance as the SM10A, but with one in-the-ear monitor.

- They mount on the supplied steel headband assembly. An adjustable pivot on the headband assembly allows the microphone boom to be moved 20° in any direction. The boom also slides out so the microphone can be positioned up to 3½″ from the pivot housing
- Close-talk design provides excellent isolation and minimizes gain-before-feedback. Cardioid pattern offers superior rejection of unwanted sound
- Extremely rugged with fully adjustable headband and mic boom
- Smooth, natural-sounding frequency response (50 to 15,000 Hz)
- Supplied windscreen reduces wind noise and explosive breath sounds. A belt clip that fits over the XLR connector and attaches to the user's belt or waistband provides cable strain relief

SM10A-CN with cable.....**119.50** **SM12A-CN** with cable **189.95**

LIVE PERFORMANCE MICROPHONES

SM2

Dynamic Headworn Mic with Dual Receivers



The SM2 is a comfortable, self-adjusting headset with dual enclosed ear receivers and an attached broadcast-quality boom-mounted microphone. Designed for convenient hands-free news and sports broadcasting, remote announcing, interviews, and intercom applications. The SM2 has a close-talking cardioid mic that discriminates against distant sounds in favor of close sounds. This allows use under noisy conditions without loss or masking of the voice signal.

Dual earphones have a frequency response of 100Hz to 8kHz with excellent voice intelligibility for cue, program, sidetone, or intercom signals. High sensitivity and enclosed ear block out room noise and ensure clarity of the incoming signal.

- Smooth, natural microphone voice frequency response
- Supplied windscreen protects against wind noise and explosive breath sounds
- Mic boom pivots through 155° so that the mic can be correctly positioned at the corner of the mouth from the left or right side. A locking thumbscrew keeps the mic from slipping or twisting out of place
- Self-adjusting, free-swiveling cushioned earpieces offer user comfort, lower fatigue during extended periods of use

WL50/WL51

Professional Subminiature Lavalier Mics

Only 5.8mm in diameter, just .34 grams in weight, the WL50 and the WL51 set a new standard for miniature lavalier microphones. Visually unobtrusive they virtually disappear when in use making them ideal for theater, television broadcast, video/film, and event production. And despite their small size, the microphones' condenser element provides full, clear and natural reproduction of speech. Available individually, as well as with the Shure UHF, UC, SC and LX Wireless Systems.

The omnidirectional WL50 features an extended frequency response and is available in three colors (black, beige and white) and two sensitivity options; Standard (-45dBv/Pa) and Reduced (-55dBv/Pa). It also includes four frequency response caps: two silver mesh (4dB mild boost) and two gold mesh (10dB high boost). Interchange them as needed to achieve optimal sound and clarity for various microphone placements and mounting configurations.

The cardioid WL51 (available in black and white) features a smooth frequency response and low sensitivity of -49dBv/Pa to provide optimal gain with very low noise output. Perfectly suited for those situations where loud ambient environments limit the use of an omni mic. Includes a watertight protective carrying case.

SM11

Lavalier Omnidirectional Dynamic Mic



The Shure Model SM11 is a miniature dynamic lavalier microphone designed for use in television broadcasting (on camera), for sound reinforcement, in certain radio and motion picture applications, and similar uses where a small, professional-quality, microphone is required. The SM11 has an omnidirectional pickup pattern, and its frequency response is optimized for lavalier use. The microphone is supplied with a tie clasp assembly, tie tack assembly, connector belt clip, and zippered carrying/storage bag. An optional kit for mounting the SM11 on various musical instruments is available as Shure Model RK279.

- Frequency response optimized for lavalier use. Acoustically matches most stand- or desk-mounted microphones
- Small and lightweight for unobtrusive, virtually unseen operation
- Smooth exterior and recessed grille screen minimize clothing noise
- Rugged aluminum case construction
- Extremely flexible long-life cable
- Two mounting options–tie clasp or tie tack
- Convenient, secure belt clip attaches connector to belt or clothing

SM93

Miniature Omnidirectional Lavalier Mic



The SM93 is an economical subminiature lavalier condenser microphone designed for use in speech applications. Despite its small size, the SM93 mic element provides full, clear sound comparable to much larger mics. Its smooth frequency response is specially tailored for chest-worn microphone operation, while controlled low-frequency rolloff reduces clothing and room noise. Runs on 11-52v DC phantom power.

- Subminiature size ideal size for film/video/TV, theater, broadcasting and sound reinforcement
- Designed for wired applications, the SM93 includes a preamplifier assembly with XLR connectors. The preamp can be pocketed, strapped to the body, or clipped to belt or waistband
- Supplied with small, inconspicuous mounting hardware. Includes a sew-on mounting bracket, a mounting block with attached tie bar, and an acoustic windscreen to minimize wind noise

BROADCAST FIELD & STUDIO PRODUCTION



SM7B *Cardioid Dynamic Studio Vocal Mic*

Dynamic mic with a smooth, flat, wide-range frequency response, the SM7B is ideal for music and speech in pro audio applications including recording studio—instrumental and vocal, location recording, motion picture and television scoring, talk shows, radio announcing and production. Offers excellent shielding against electromagnetic hum from computer monitors, neon lights, and other electrical devices. Yoke mounting with captive stand nut for easy mounting and dismounting offers precise control of mic positioning and greater stability.

- Flat, wide-range frequency response for exceptionally clean and natural reproduction of both music and speech.
- Cardioid pattern uniform with frequency and symmetrical about axis, provides maximum rejection and minimum coloration of off-axis sound
- Switchable bass rolloff and mid-range emphasis (presence boost) controls with graphic display of response setting
- Internal "air suspension" shock isolation virtually eliminates mechanical noise transmission
- Pop filter eliminates need for any add-on protection against explosive breath sounds, even for close-up vocals or narration. However, for high-wind conditions and "problem talkers", it includes A7WS detachable windscreen (also gives a warmer tone for close-talk vocals)
- Swiveling bracket with integrated stand adapter for easy mounting and precise microphone positioning

VP88 M-S Single-Point Stereo Condenser Microphone

The VP88 is a single-point, stereo condenser microphone for use in professional studio recording, field production, electronic news gathering (ENG), and studio broadcasting applications. It combines two condenser cartridges in a single housing to create a stereo audio image of the sound source. It is unique in its ability to capture the realism of a live event and yet able to withstand the rigors of field production.

The VP88 uses a mid-side (M-S) configuration. One microphone cartridge (Mid) faces forward to capture on-axis sound using a cardioid pickup pattern. The other mic cartridge (Side) is bi-directional, capturing sound from either side. The VP88 contains an internal M-S matrix with three settings for different degrees of stereo image separation. The internal matrix can be bypassed if an external matrix is used or if stereo imaging is done in post-production. Switches on the top of the microphone control output mode, stereo imaging, low-frequency rolloff settings, and battery on/off.



- Time-coherent, Mid-Side (M-S) design for optimal stereo imaging
- Built-in stereo matrix for direct control of the stereo spread; direct M-S outputs available
- Mono compatibility ideal for broadcast applications
- 40 to 20kHz frequency response
- Low noise and distortion
- Wide dynamic range
- Selectable, low-frequency rolloff (12 dB/octave below 80 Hz) for reduced pickup of ambient noise and stand vibration
- Internal shock mount for reduced stand vibration and handling noise
- Built-in "pop" filter provides protection against moderate breath or wind noise.
- Operates on phantom power or a 6V camera battery
- LED power indicator
- Supplied with a multi-connector "Y-splitter" cable, foam windscreen, swivel stand adapter, 6-volt battery, and zippered carrying/storage bag.
- Optional accessories include an isolation mount, a 25´ microphone extension cable, a phantom power supply, and a 14´ microphone stand

OPTIONAL ACCESSORIES

A88SM Isolation Mount: The optional A88SM Shock Mount provides optimum isolation from stand-transmitted shock and vibration, making it ideal for critical studio applications. It also contains a standard mounting foot for use with most ENG video cameras.....**158.50**

C110 25´ Extension Cable (4 conductor, shielded).....**66.95**

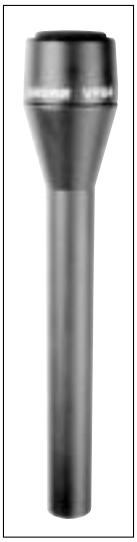
PS1A Phantom Power Supply.....**125.95**

S15 14´ Microphone Stand.....**157.95**

BROADCAST FIELD PRODUCTION

VP64A/VP64AL

Omnidirectional Dynamic Mic



High-output omnidirectional mics designed for professional audio and video production, the VP64A and VP64AL combine exceptional performance and comfortable feel with handsome on-camera appearance. Identical except for their handle lengths, the 7⅞″ long VP64A is ideal for close-up use and can be used outside as well as indoors. The longer VP64AL is well-suited to location interviews, sports broadcasting, and other situations where the 9⅝″ length is an advantage. Frequency response for both is 50 to 12,000 Hz

- The omnidirectional pickup pattern provides uniform pickup on or off axis, eliminating the need for critical mic positioning
- Tailored frequency response with mid-range presence rise adds crispness and clarity to speech
- Neodymium magnet for increased output and maximized S/N ratio
- Internal rubber isolation mount protects the microphone cartridge and minimizes handling noise
- Water-resistant mesh grille allows use in adverse weather conditions
- Supplied windscreen further reduces "pop" and wind noise

VP64A **77.95**
VP64AL **89.95**

SM63

Omnidirectional Dynamic Mic



Elegant, yet rugged, the high output SM63 is designed for applications where performance and appearance are critical. Smooth, wide frequency response is tailored for optimum speech intelligibility. Controlled low-frequency rolloff reduces pickup of stand and wind noise. Virtually immune to strong hum fields such as those produced by studio lighting.

The champagne-finished SM63 is designed for hand-held use on stage and in broadcast, recording, and TV studios, while the champagne SM63L and black-finished SM63LB are for interviews, sports broadcasts, and other situations where longer microphones are desirable.

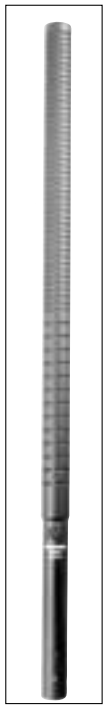
- Exceptionally rugged VERA-FLEX grille makes the mic impervious to dents, rust, corrosion, moisture and protects against damage from a drop of up to 6 feet
- Patented shock mount system minimizes handling noise and stand vibration
- Pop filter effectively reduces breath noise
- Includes snap-in stand adapter and case

The SM63L and SM63LB include a two-stage windscreen designed for outdoor use under adverse wind conditions.

SM63.....**118.95**
SM63L Champagne finish **132.95**
SM63LB Black finish.....**132.95**

SM89

Shotgun Condenser Mic



The SM89 is a highly directional condenser shotgun mic with distant pickup characteristics suitable for on-location film, television production, theater sound reinforcement, spot news coverage and wildlife recording.

- Discriminates at a distance against ambient noise – even in noisy surroundings
- On-axis frequency response is very smooth and extended. For clarity and speech intelligibility, a slight presence rise optimizes the high-frequency response to compensate for high-frequency losses
- Low-frequency roll-off pickup of wind, mechanical vibration, ambient noise, and rumble without affecting voice frequencies
- Off-axis response is virtually free of comb-filter effects. This is important when it isn't possible to aim the mic precisely or when you want to pick up dialogue or sound from moving sources
- Operates on 11 to 52v DC phantom supply
- Rugged and reliable, operates over wide range of temperature and humidity
- Extremely lightweight, can be suspended on fishpole for long periods of time without operator fatigue
- Supplied with foam windscreen and strong luggage-grade carrying case

SM89 .....**672.50**

	SM7B	VP88	VP64	SM63	SM89
Microphone Element Type	Dynamic	Stereo Condenser	Dynamic	Dynamic	Pressure Gradient Condenser
Polar Pattern	∩	Mid: ∩ Side: 8	○	○	∩
Frequency Response	50 to 20k Hz	40 to 20k Hz	50 to 12k Hz	80 to 20k Hz	60 to 20k Hz
Impedance	150 Ω	150 Ω	300 Ω	150 Ω	150 Ω
Output Level at 1kHz	-59 dB	—	–51.5 dBV/Pa	-56.5 dBV/Pa	–53 dB (2.2 mV)
Dynamic Range	—	105 dB	—	—	111 dB
Signal to Noise Ratio	—	70 dB @ 94dB SPL	—	—	79 dB @ 94 dB SPL
Weight	1 lb. 11 oz.	14.7 oz.	VP64: 5.6 oz. VP64L: 6.3 oz.	SM63: 3.5 oz. SM63L/LB: 1 lb. 5 oz.	6.9 oz.

INSTRUMENT RECORDING MICROPHONES



SM81 Studio Condenser Microphone

One of the world's great condenser microphones, the SM81 provides precise, detailed sound reproduction for high quality studio recording or broadcasting, and is rugged enough for live sound reinforcement. Its wide frequency response, low noise characteristics, and low RF susceptibility have made it a standard for applications involving acoustic instruments, especially guitar, piano, and cymbals. Operates on phantom power and performs over a wide range of temperatures and humidity conditions.

- Flat response curve for accurate reproduction of sound sources
- Low noise and high output clipping level
- Cardioid pattern is uniform with frequency and symmetrical about axis for maximum rejection of unwanted sound and minimum of-axis coloration
- Low distortion over a wide range of load impedances
- Selectable low-frequency response: flat, 6 or 18 dB/octave rolloff
- 0 dB/10 dB lockable attenuator switch
- Phantom powered (12 to 48v DC)
- Supplied with a swivel adapter, attenuator-switch lock, foam windscreen, and case for carrying and storage

SM81-LC with foam windscreen, without cable

849 Cardioid Handheld Dynamic Microphone



The 849 is a high-quality electret condenser microphone with a smooth, flat frequency response and cardioid pickup pattern. It provides accurate, uncolored reproduction of the sound source. The 849 is designed for a wide variety of sound reinforcement and recording applications, and is a good choice for use with choirs, amplified sound, or instruments, including strings, brass, woodwinds, and percussion.

- Wide, flat frequency response
- Internal shock mount for reduced stand and handling noise
- Long-life on/off switch for on-stage control
- Low distortion output and wide dynamic range characteristics for a variety of load impedances
- Cardioid pattern is uniform over a wide range of frequencies
- Very low RF and magnetic hum susceptibility
- Can use either phantom power or an internal AA battery
- Rugged construction for outstanding reliability, usable over wide range of temperature and humidity conditions
- Supplied with swivel adapter and a zippered vinyl storage bag



SM94 Unidirectional Condenser Microphone

The SM94 is a unidirectional condenser mic tailored to the instrument miking needs of professional musicians and sound engineers in live performance and sound recording. Its smooth, flat frequency response is free of either a "presence peak" or a low-frequency rolloff, making it a perfect choice for all types of musical instruments. The SM94 is equally at home in sound reinforcement systems or in sound studios and motion picture/ TV scoring stages. When used with the optional wind-screen, the SM94 can be used by vocalists and speech-makers who desire a wide, flat response with minimum coloration.

- Powered by AA battery (up to 5,000 hours) or 12 to 48v DC phantom power (also has a built–in battery power backup)
- Supplied with a swivel adapter for use on a mic stand, boom or gooseneck, and a zippered vinyl storage bag.
- Wide, flat response for optimum musical instrument sound reproduction
- Full spectrum sound—no presence peak or low-end rolloff
- Low distortion output and wide dynamic range for variety of load impedances
- Cardioid pattern is uniform with frequency and symmetrical about axis for maximum rejection of unwanted sound and minimum of-axis coloration
- Very low susceptibility to RF and electromagnetic hum
- Shock–mounted cartridge for reduced handling noise

SM94-LC without cable **174.95**

KSM STUDIO INSTRUMENT MICROPHONES

The KSM141, KSM137, and KSM109 are instrument condenser microphones designed for studio use. As the latest additions to the KSM line of professional studio products, the group is designed to mic instruments including: acoustic guitar, drum overheads, high-hat cymbals, guitar cabinets, pianos, percussion, stringed instruments, and more. Additionally, with a range of features and prices, the trio offers a solution for high-end recording, project, and home studio users alike.

KSM141 Switchable Omnidirectional Cardioid Condenser



As the premier entry within the group, the KSM141 is a dual-pattern mic offering a selection of either cardioid or omnidirectional polar patterns with the turn of a rotating mechanical switch. Other features include Shure's Class A transformerless preamp circuitry, an ultra-thin, highly sensitive diaphragm, and a switchable, 3-position low frequency filter to limit unwanted background noise and proximity effect. The mic can reproduce sounds between 20 and 20,000 Hz. Further complementing the feature set is a switchable pad (15 or 25 dB) for managing extremely high SPL sources such as electric guitar cabinets and snare drums. The KSM141 includes a break-resistant mic clip, carrying case, and Popper Stopper windscreen, and is available in stereo pairs.

- A mechanical polar pattern switch for highly consistent cardioid and true omnidirectional polar patterns. Provides flexibility in a wide variety of recording applications
- Ultra-thin, 2.5 micron, 24 karat gold-layered, low mass Mylar diaphragm for superior transient response
- Class A, discrete, transformerless preamplifier for transparency, extremely fast transient response, no crossover distortion, and minimal harmonic and intermodulation distortion
- Premium electronic components, including gold-plated internal and external connectors
- Subsonic filter eliminates low frequency rumble (less than 17 Hz) caused by mechanical vibration
- Three-position switchable pad (0 dB, 15 dB, and 25 dB) for handling extremely high sound pressure levels (SPLs)
- Three-position switchable low-frequency filter to reduce background noise and counteract proximity effect

KSM137 Cardioid Condenser

A single pattern (cardioid) version of the KSM141, the KSM137 performs with the same detailed sonic precision, and excels on acoustic guitar, drum overheads, and hi-hats. Also like the dual-pattern KSM141, the KSM137 is available in a stereo pair, sports a champagne finish, and comes with accessories including a break-resistant mic clip, carrying case, and Popper Stopper windscreen. The Shure KSM137 is an end-address condenser microphone with a cardioid polar pattern. Designed for studio use, yet rugged enough for live applications, the KSM137 can withstand extremely high sound pressure levels (SPL). Its low self-noise and extended frequency response make it ideal for recording musical instruments.



▲ *KSM137*

KSM109 ▶

KSM109 Cardioid Condenser

Last among the group is the affordable, cardioid KSM109. A value-engineered cardioid condenser unit, it too offers an extended frequency response (20 Hz - 20,000 Hz) for natural reproduction of sound with smooth high-end delivery and tight low-end. Equipped with a switchable pad (15 dB) to accommodate high SPL sources, as well as circuitry using the same architecture employed by its higher-priced counterparts within the KSM product group, the champagne-finished mic comes with a zippered carrying pouch, microphone clip, and Popper Stopper windscreen. The Shure KSM109 offers incredible performance for its price. It combines an extended frequency response for natural sound reproduction with a smooth high end and a tight, controlled low end. It's all the mic you need for studio instrument recording.

	SM81	849	SM94	KSM141	KSM137	KSM109
Microphone Element Type	Condenser	Cardioid Condenser	Cardioid Condenser	Condenser	Condenser	Condenser
Polar Pattern	◠	◠	◠	◠ ○	◠	◠
Frequency Response (Hz)	20 to 20k	40 to 16k	40 to 16k	20 to 20k	20 to 20k	20 to 20k
Rated Impedance	150 Ω	600 Ω	150 Ω	150 Ω	150 Ω	150 Ω
Sensitivity (Phantom/Battery)[1]	-45 dBV/Pa	-51 dBV/Pa	-49 dBv/Pa / -50 dBv/Pa	–37 dBV/Pa	–37 dBV/Pa	–41 dBV/Pa
Maximum SPL[2] **(Phantom/Battery)**	136 dB	131 dB / 128 dB	141 dB	134 dB	139 dB	139 dB
Self Noise[3]	16 dB	22 dB	22 dB	14 dB	14 dB	19 dB
Signal-to-Noise Ratio[4]	78 dB	71 dB	69 dB	80 dB	75 dB	75dB

[1] *1 Pa = 94 dB SPL* [2] *at 1kHz, 800Ω* [3] *typical, A-weighted* [4] *at 94 dB SPL*

SHURE

KSM STUDIO MICROPHONES

KSM27 Studio Condenser Mic



Designed to be the workhorse for project and home recording studio, the KSM27 is a large-diaphragm, side-address cardioid microphone optimized for vocals, but also capable of handling high-input SPLs to accommodate sound sources such as drums and guitar amplifiers. And although used primarily in the studio, it is rugged enough for live applications as well.

The KSM27 features a 1-inch ultra-thin Mylar diaphragm, an internal shock mount to keep self-noise low, and utilizes Class A transformerless preamp circuitry to virtually eliminate crossover distortion and bring improved linearity across its entire operating range. Like other KSM microphones, the KSM27's durable zinc die-cast housing comes finished in an attractive champagne color. To protect the microphone from damage, a hardened low-carbon steel grille shields its capsule. Accessories provided with each mic include a Velveteen pouch and a rubber-isolated, external locking shock mount.

- 1″, externally biased, ultra-thin, 24k gold-layered, low mass, Mylar diaphragm provides superior transient response
- Class A, discrete, transformerless preamplifier circuitry for transparency, extremely fast transient response and no crossover distortion—improved linearity across the entire frequency range
- Premium electronic components and gold-plated internal and external connectors
- Subsonic filter eliminates rumble from mechanical vibration below 17 Hz
- Switchable 15 dB pad for handling extremely high sound pressure levels (SPLs)
- Internal shock mount reduces handling and stand noise (keeps self-noise low at 14 dB).
- Switchable three-position (flat, –6 dB/octave below 115 Hz, –18 dB/octave below 80 Hz) low-frequency filter lets you to adjust the the mic's low-frequency response to reduce unwanted background noise.
- Integrated three-stage "pop" protection grille reduces plosives and protects against wind and breath noise



KSM32 Cardioid Studio Condenser Mic

The KSM32 has a classic, elegant appearance and even more impressive performance provided by its extended frequency response, low self-noise, high output level, and increased dynamic range. The KSM32 is also flexible enough to handle a variety of demanding sound sources including vocals, acoustic and wind instruments, ensembles, and overhead miking of drums and percussion. In addition, it has the warmth and sensitivity necessary for superb sound reproduction in professional studio production and live sound recording.

- For studios, the KSM32/SL has a light champagne finish and includes an aluminum carrying case, shock and swivel mounts and a velvet pouch. For live applications, the KSM32/CG has a charcoal grey finish and includes a swivel mount and padded zipper bag.
- Frequency response 20Hz - 20kHz



THE KSM32 AND KSM44 BOTH INCLUDE:

- *ShureLock swivel mount*
- *Protective velveteen pouch*
- *ShureLock elastic-suspension shock mount*
- *Locking aluminum carrying case*

B&H PHOTO - VIDEO - PRO AUDIO
ONE HOUR FREE PARKING
AT 349 W. 34th STREET (with purchase of $100 or more)

KSM STUDIO MICROPHONES

KSM44 Multi Pattern Studio Condenser Mic

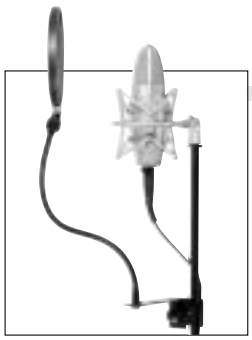
The KSM44 is a multiple pattern (cardioid, omnidirectional, bi-directional), externally biased, dual large diaphragm condenser microphone with extremely low self-noise (7dB). Developed to meet the needs of studio recording musicians, the warm, rich sound of the KSM44 delivers exceptional performance, tailored to deliver vivid reproduction of voice for the most critical recordings. The KSM44 also offers great flexibility in a variety of recording applications, working well on acoustic and amplified instruments, as its capacity for managing high SPLs accommodates loud sound sources such as drums and guitar amplifiers.



- Dual 1-inch, externally biased, ultra-thin 24-karat gold-layered, Mylar diaphragms provide superior transient response
- Class A, discrete, transformerless preamplifier for transparency and extremely fast transient response
- Subsonic filter eliminates rumble from mechanical vibration below 17 Hz.
- Switchable 15 dB pad for handling high SPLs
- 3-position switchable low-frequency filter helps reduce unwanted background noise or counteract proximity effect.
- Integrated three-stage "pop" protection grille reduces plosives and other breath noise; internal shockmount reduces handling/stand noise.

	KSM-27	KSM-32	KSM-44		
Cartridge Type	Externally Biased Condenser	Permanently Biased Condenser	Externally Biased Condenser		
Polar Pattern	ᔓ	ᔓ	ᔓ	○	∞
Frequency Response (Hz)	20 to 20k	20 to 20k	20 to 20k	20 to 20k	20 to 20k
Output Impedance	150Ω	150Ω	150Ω	150Ω	150Ω
Sensitivity[1]	-37 dBV/Pa	-36 dBV/Pa	-31 dBV/Pa	-37 dBV/Pa	-36 dBV/Pa
Self Noise[2]	14 dB	13 dB	7 dB	10 dB	10 dB
Maximum SPL, 1k Ω Load[3]	133 dB	133 dB	127 dB	132 dB	131 dB
Output Clipping Level, 1k Ω Load[3]	1dBv	3dBv	1dBv	1dBv	1dBv
Dynamic Range, 2500 Ω Load	124 dB	126 dB	125 dB	128 dB	127 dB
Signal-to-Noise Ratio[4]	81 dB	81 dB	87 dB	84 dB	84 dB
Attenuation Switch	0 or -15 dB	0 or -15 dB	0 or -15 dB	0 or -15 dB	0 or -15 dB
Low Frequency Response Switch	Flat, -6 dB/octave below 115 Hz, -18 dB/octave below 80 Hz				
Phantom Power	48 Vdc ± 4 Vdc (IEC-268-15/DIN 45 596) positive pins 2 and 3				
Current Drain	5.4 mA typical at 48 Vdc	4.65 mA typical at 48 Vdc	5.4 mA typical at 48 Vdc		
Common Mode Rejection Ratio	≥ 50 dB, 20 Hz to 20 Hz	≥ 50 dB, 20 Hz to 20 Hz	≥ 50 dB, 20 Hz to 20 Hz		
Polarity	Positive pressure on diaphragm produces positive voltage on pin 2 relative to pin 3				
Dimensions	2.20″ max. body diameter, 6.15″ long	2.20″ max. body diameter, 7.37″ long	2.20″ max. body diameter, 7.37″ long		

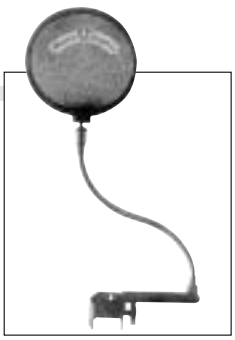
[1]Typical, at 1kHz; 1 Pa = 94 dB SPL [2]Typical, equivalent SPL; A-weighted (IEC 651) [3]20Hz-20kHz, THD <1%
[4]Difference between 94 dB SPL and equivalent SPL of self-noise A-weighted



PS-6 Popper Stopper

The Popper Stopper is widely recognized throughout the recording and post-production industry as the leading tool in pop elimination technology.

- 6″ diameter highly durable four-layer screen readily softens harsh "P" and "B" consonants
- Fully adjustable 14″ gooseneck with "position memory" (stays put until you change it)
- Attaches to virtually any floor or boom stand via a mounting clamp equipped with a non-scratching nylon insert



INSTALLED SOUND—MICROFLEX SERIES

The entire Microflex family of miniature condenser microphones have been designed with performance, consistency and ease of installation in mind. Their cardioid, super-cardioid and omni-directional condenser cartridges are completely interchangeable and fit every Microflex Gooseneck, Overhead, Boundary, and Lavalier model for the ultimate in convenience and uniform sound. Plus, each Microflex model includes all the basic accessories needed for complete installation.

MX-391/392/393
Surface-Mount Mics



The low-profile Microflex Series boundary mics are designed to be unobtrusively placed flat on a stage, table top or wall while offering high quality audio for sound reinforcement, boardrooms, broadcast or recording. Available with interchangeable cardioid, super-cardioid and omni-directional polar patterns in battery power, phantom power or pigtail versions, giving you the flexibility for any application.

- 50Hz to 17kHz frequency response
- Easy to use paint mask for your custom-color installations
- Programmable silent membrane switches with LED indicator can be set for Push to Talk, Push to Mute or Push On/Push Off modes (MX392 and MX393 only)

Surface-Mount Microphone Models Include:

MX391/C (cardioid) includes attached 12′ cable and separate in-line preamp, phantom power only **158.95**

MX391/S super-cardioid version of above **169.50**

MX391/O omni-directional version of above.....**149.95**

MX391W/C, S *or* **O** White versions of above available for the same price

MX391BP/C (cardioid) Battery/phantom power of above includes MX1BP battery/phantom power module.....**174.95**

MX391BP/S super-cardioid version of above.....**179.95**

MX391BP/O omni-directional version of above.....**162.50**

MX391LP/C (cardioid) without preamplifier for wireless systems, uses Tini Q-G connector.....**102.50**

MX391LP/S super-cardioid version of above.....**108.50**

MX391LP/O omni-directional version of above.....**93.50**

MX391WLP/C, S *or* **O** White versions of above available for the same price

MX392/C (cardioid) includes programmable membrane switch with LED indicator, Built-in preamplifier, logic input/output control capability, unterminated, phantom power only.....**202.50**

MX392/S super-cardioid version of above.....**208.50**

MX392/O omni-directional version of above.....**193.95**

MX393/C (cardioid) includes programmable membrane switch with LED indicator, Built-in preamplifier, detachable cable terminates to a male XLR connector, phantom power only.....**202.50**

MX393/S super-cardioid version of above.....**208.50**

MX393/O omni-directional version of above.....**193.95**

MX-183/184/185 Lavalier Microphones

Three wired lavalier microphones from the Shure Microflex series no larger than a paper clip. Optional interchangeable mic cartridges allows you to tailor the microphones to any audio environment. The mics attach to the supplied power supply with the Shure Tini Q-G connector. When the power supply is removed, the microphones can be used with any Shure wireless transmitters. Well suited for all levels of location video work, recording and live applications.



- Condenser lavalier microphones
- Interchangeable cartridges for omni-directional, cardioid and super-cardioid polar patterns
- Frequency Response 60 - 20 kHz
- Includes both a pivoting single and dual microphone tie clip
- Black, snap-fit foam windscreen
- Compatible with all Shure wireless transmitters
- Available with belt-clip phantom power only preamp and MX1BP battery/phantom power preamp
- Attached 4′ cable with Tini Q-G connector

LAVALIER MICROPHONE MODELS INCLUDE

MX183 Omni-directional condenser lavalier, belt-clip preamp, rotatable tie clip, dual tie clip, snap-fit windscreen.....**162.95**

MX184 Super-cardioid version of above **179.95**

MX185 Cardioid version of above **174.95**

MX183BP Omni-directional condenser lavalier microphone with 4′ attached cable, MX1BP preamplifier.....**174.95**

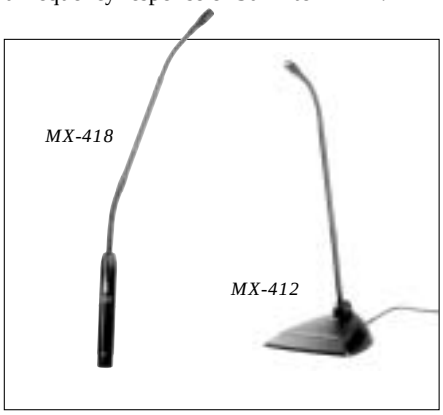
MX184BP Super-cardioid version of above **193.95**

MX185BP Cardioid version of above **187.95**

INSTALLED SOUND—MICROFLEX SERIES

Gooseneck Microphones

Designed with premium audio quality and ease of installation in mind, these miniature condenser gooseneck microphones feature swappable cardioid, super-cardioid and omni-directional cartridge capability and come complete with all the necessary accessories. A variety of mounting options make these mics suitable for any installation. All models run on phantom power and have a frequency response of 50Hz to 17kHz.



SWAPPABLE MICROFLEX SERIES CARTRIDGES



R183B Omni-directional **38.95**
R184B Super-cardioid.....**5395**
R185B Cardioid.....**47.50**

Gooseneck Microphone Models Include:

MX412/C (cardioid) 12″ gooseneck, terminates to XLR connector **179.95**

MX412/S (super-cardioid version of above) **187.95**

MX412/O (omni-directional version of above) **169.95**

MX418/C (cardioid) 18″ version of above.....**179.95**

MX418/S (super-cardioid version of above).....**187.95**

MX418/O (omni-directional version of above) **169.95**

MX412D/C (cardioid) 12″ gooseneck with desktop base, with programmable switch, logic input and output terminals and attached 10′ XLR cable.....**237.50**

MX412D/S (super-cardioid version of above).....**243.50**

MX412D/O (omni-directional version of above) **229.95**

MX418D/C (cardioid) 18″ version of above.....**237.50**

MX418D/S (super-cardioid version of above) **243.50**

MX412S/S (super-cardioid version of above) **202.50**

MX412S/O (omni-directional version of above) **189.95**

MX418S/C (cardioid) 18″ version of above.....**196.50**

MX418S/S (super-cardioid version of above) **202.50**

MX418S/O (omni-directional version of above) **189.95**

MX412SE/C (cardioid) 12″ gooseneck with attached 10' XLR cable and inline preamplifier, threaded flange mount and shock mount adapter.....**193.95**

MX412SE/S (super-cardioid version of above) **196.50**

MX412SE/O (omni-directional version of above) **186.95**

MX418SE/C (cardioid) 18″ version of above.....**193.95**

MX418SE/S (super-cardioid version of above) **196.50**

MX418SE/O (omni-directional version of above) **186.95**

MX418D/O (omni-directional version of above) **219.95**

MX412S/C (cardioid) 12″ gooseneck with push-button mute switch and LED status indicator, terminates to male XLR connector **196.50**

MX-202 Hanging Microphones

These phantom power only miniature hanging condenser mics feature the same interchangeable capsule design as the rest of the Microflex series. Available with either plate-mount or stand adapters ideal for use as overheads for live theatrical and musical productions. The flexible 4″ gooseneck allows you to unobtrusively aim the mics exactly the way you want them.



HANGING MICROPHONE MODELS INCLUDE:

MX202B/C (cardioid) 4″ gooseneck with in-line preamplifier and stand adapter, black finish **162.95**

MX202B/S super-cardioid version of above **169.50**

MX202B/O omni-directional version of above **154.95**

MX202W/C, S *or* **O**
White versions of above available for same price

MX202BP/C (cardioid) 4″ gooseneck with plate-mount preamplifier and 30' cable, black finish **162.95**

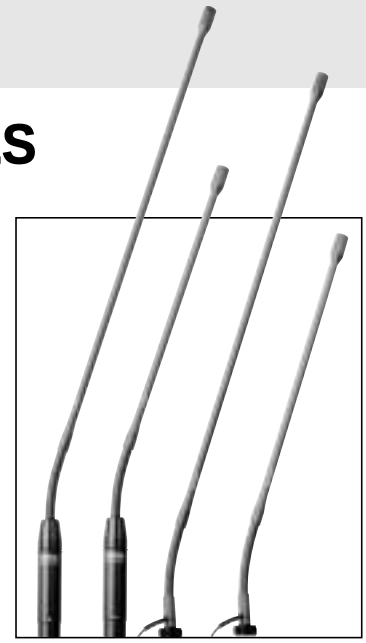
MX202BP super-cardioid version of above **169.95**

MX202BP/O omni-directional version of above.....**156.95**

MX202WP/C, S *or* **O**
White versions of above available for same price

INSTALLED SOUND—EASYFLEX SERIES

EZG Miniature Cardioid Condenser Gooseneck Microphones



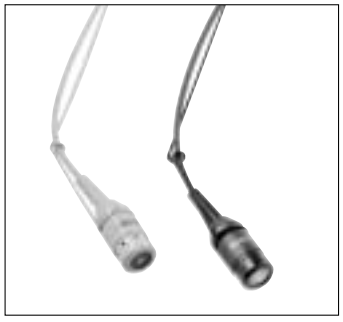
These miniature gooseneck condenser microphones are the perfect choice to capture speech and vocals in environments where quality, value and budget are primary considerations. Designed for podium or desk applications, they offer a cardioid polar pattern, and are available in four different models, dependent on height and mounting application. Available in either 12˝ or 18˝ lengths, each charcoal gray microphone comes standard with a single flexible gooseneck section for ease in accurate positioning, as well as a matching slip-on foam windscreen. Choose a locking flange mount with an integrated preamp and XLR connector for easy permanent install. Or, choose side-exit mounting with 10´ cables, in-line preamps and threaded mounting flanges - easily adaptable to a standard mic stand.

EZG/12 (12˝) and EZG/18 (18˝) Mics

- Cardioid polar pattern
- Integrated preamplifier with XLR connector
- Locking flange mount

EZG/12SE (12˝) and EZG/18SE (18˝) Mics

- Cardioid polar pattern
- Attached 10´ side exit cable with in-line preamplifier
- Threaded mounting flange



EZO Overhead Microphones

The perfect choice to reinforce choirs and performance groups in recording or sound reinforcement in environments where quality, value and budget are primary considerations. The overhead condenser microphones feature a cardioid polar pattern, and are available in charcoal gray or white for unobtrusive placement in most surroundings.

Each Easyflex overhead microphone ships with a flexible wire form for accurate and stable positioning when suspension mounting to a ceiling, and ships with 30´ cable with in-line preamplification, and a matching slip-on windscreen.

EZO/G Charcoal Gray and EZO/W White Overhead Microphones

- Cardioid polar pattern
- Attached 30´ cable with in-line preamplifier
- Adjustable wire frame for easy positioning

EZB/C Cardioid and EZB/O Omnidirectional Boundary Microphones

- Charcoal gray finish
- Attached 12´ cable with in-line preamplifier
- Recessed mounting holes for quick installation

EZB Miniature Cardioid Condenser Boundary Mics



The perfect choice for all types of low profile applications, where quality, value and budget are primary considerations. The boundary condenser microphones feature either a cardioid or omnidirectional polar pattern, and are available with a charcoal gray finish for unobtrusive placement in most surroundings. Each Easyflex boundary microphone ships with an attached 12´ cable with in line preamplifier, and features recessed mounting holes for ease of installation.

PAGING, DISPATCH & TALKBACK MICROPHONES

515BSLX

The 515BSLX is designed for mounting on a gooseneck or fixed pipe with a 5/8 in.–27 thread. Ideal for use in communications, paging, and talkback systems, the mic is supplied with a slide switch controlling the microphone audio and a relay circuit, permitting hands-free operation. Mic comes with a 16-ft. four-conductor, two conductors shielded, cable. The 515BSLX is wired with the microphone circuit normally open for use in parallel-wired microphone installations.

515BSLX.....**57.50**

515SBGX, 515SBG-18X and 515SBG-18XF

The 515SBGX, 515SBG-18X, and 515SBG-18XF are designed to be mounted on flexible goosenecks or fixed pipes. Use in paging applications, and base station communications, or as talkback or cuing mics in TV, film, and recording studios. Because of their low impedance, these mics retain their level and response characteristics even when long cable lengths are used. Each is equipped with a push-to-talk button and a four-conductor cable (two conductors shielded).

515SBGX: Attached cable .....**Call**

515SBG-18X: 18˝ gooseneck with A13HD mounting flange .....**Call**

515SBG-18XF: 18˝ gooseneck with A12 mounting flange .....**Call**

515BSM

The 515BSM is designed for hand-held or fixed use. With a push-to-talk switch and a four-conductor retractable cable (2 shielded conductor) it can be used in public speaking, stage performance, and paging or public address systems. The mic is wired with the microphone circuit normally open for use in parallel microphone installations.

515BSM: Attached cable **89.50**

515BSLG24F & 515BSLG27F

The 515BSLG24F and 515BSLG27F are each threaded on a black gooseneck with a black mounting flange. Especially suited for use in communications, paging, and talkback systems., they have a slide switch that sets their audio and relay circuit on or off, while the integral gooseneck permits hands–free operation. The attached four conductor cable (two conductors shielded) extends 1.2m (4 ft.) from the mounting flange and terminates in a female 4–pin XLR connector.

515BSLG24F: includes a 24-in. black gooseneck and 3-inch black mounting flange. Cable has 4 pin XLR female connector **68.95**

515BSLG27F: includes a 27-in. black gooseneck and 3-in. black mounting flange. Cable has 4 pin XLR female connector **68.95**



515SDX

Suitable for speech pickup in schools, churches, and meeting rooms, the 515SDX has a locking ON/OFF switch and an internal impedance selection socket mounted on the XLR connector for switching between high and low impedance. A slip-in swivel adapter is supplied for stand-mounted use.

515SDX: Attached cable **54.95**

588SDX

The 588SDX is a ball-type, dual-impedance, cardioid dynamic microphone that provides excellent voice reproduction. Its spherical grille is a very effective wind and pop filter which is especially good when used for closeup vocal miking. This microphone is highly suited for speech or music pickup in schools, churches, and meeting rooms. Its unidirectional polar pattern greatly reduces feedback problems, permitting operation closer than usual to loudspeakers without the annoying squeal or howl caused by feedback. The microphone is equipped with a lockable ON/OFF switch and an internal impedance selection socket located on the XLR connector for switching between high and low impedance. A slip-in swivel adapter is supplied for the convenience of stand-mounted use.

588SDX: Attached cable **57.95**

	515BSM	515BSLX	515BGX	515SDX	515BSLG24F	588SDX
Microphone Element Type	Dynamic	Dynamic	Dynamic	Dynamic	Dynamic	Dynamic
Polar Pattern	◠	◠	◠	◠	◠	◠
Frequency Response (Hz)	80 to 15k	80 to 15k	80 to 15k	20 to 10k	50 to 16k	80 to 15k
Rated Impedance	150Ω	150Ω	150Ω	150Ω	150Ω	150Ω
Output Level Sensitivity[1]	-57 dBV/Pa	-57 dBV/Pa	-56.5 dBv/Pa	-64 dBV/Pa	-51 dBV/Pa	-55.5 dBV/Pa
Recommended Windscreen	included	A1WS			A1WS	A58WS

[1] *1 Pa = 94 dB SPL* [2] *at 1kHz (0.25% THD, 1000 ohm load)*

PAGING, DISPATCH & TALKBACK MICROPHONES

503BG Unidirectional Dynamic Microphone



Developed for computer voice recognition applications, the 503BG offers excellent voice fidelity while reducing background noise. Equally well-suited for close-talking applications such as paging and talkback. The 503BG features a low-impedance, moving coil type transducer with an integral humbucking coil and an attached two-conductor shielded cable with a built-in strain relief. It produces a frequency response with an essentially constant output voltage over the voice spectrum when used close to the sound source. The 503BG's case is aluminum with a black enamel finish. It mounts to a gooseneck, a fixed pipe or a mic stand using a standard 5/8˝-27 thread. Includes #4 Allen (hex-head) wrench.

- Optimized for voice processing and computer voice recognition applications
- Smooth frequency response with excellent fidelity in the voice range
- Cardioid pickup pattern and close-talking operation keep unwanted signals and background noise at low levels
- Constant output over voice spectrum when close talked
- Humbucking coil minimizes undesirable hum effects such as CRT interference
- Low-impedance design permits long cable runs in extreme hum fields
- Small unobtrusive design does not interfere with CRT visibility
- Rugged and reliable for years of consistent operation

561 Omnidirectional Dynamic Mic



A low-impedance dynamic mic, the omnidirectional 561 is designed for use on a stand or flexible gooseneck. Its smooth frequency response is tailored for optimum speech intelligibility, making it ideal for language labs, paging applications, and base station communications. The 561 is also suitable for use as a talk-back or cuing microphone in TV, film, and recording studios.

- Frequency response tailored for speech applications, with bass rolloff and mid-range "presence" rise
- Shock-mounted cartridge for reduced stand and handling noise
- Rugged die–cast aluminum construction
- Lockable stainless steel mesh grille
- Optional gooseneck and mounting flange

562 Noise-Canceling Omnidirectional Dynamic Mic



The 562 is a noise-canceling mic designed for voice applications in fixed-station paging and radio communications. It provides highly intelligible speech communication where high volume background noise is present. The 562 has a smooth response and is ruggedly constructed to withstand a wide range of humidities and temperatures. With its standard 5/8" – 27 thread mount, the 562 attaches directly to boom or a gooseneck with the cable concealed inside. Ideal for language labs, paging applications in noisy environments and base station communications, as well as a talkback or cuing microphone in TV, film, and recording studios.

- Excellent response for voice reproduction
- Noise-canceling design
- Low impedance
- Rugged construction
- Withstands wide range of temperatures and humidities
- Mounts on gooseneck and lockable with Allen set-screw (Allen wrench supplied)

545 Unidyne III



The 545SD Unidyne III is a dual-impedance, cardioid dynamic microphone. Shipped connected for low-impedance operation.

The 545SD is a favorite for instrument pickup and re-cording, and can be used on lecterns for public address systems such as those used in legislatures, convention halls, hotels, public auditoriums, stadiums, schools, and churches.

- Selectable dual–impedance operation
- Silent magnetic reed on/off switch with lock option
- Exceptionally uniform cardioid pattern minimizes feedback in PA applications
- Frequency response: 50 to 15,000 Hz

545SD-LC: Without cable

565 SD



The 565 SD is a dual-impedance, cardioid dynamic microphone with a strong, wire-mesh, spherical front grille that contains a very effective wind and "pop" filter.

Shipped connected for low-impedance operation, the mics provide excellent vocal or speech reproduction on lecterns and stages, and are a popular choice for use in public auditoriums, churches, convention halls, and schools.

- Selectable dual impedance operation
- Silent magnetic reed on/off switch with lock-on option
- Frequency response: 50 to 15,000 Hz

565SD-CN: With 20´ XLR cable **125.95**

565SD-LC: Without cable **107.50**

B&H PHOTO - VIDEO - PRO AUDIO

PAGING, DISPATCH & TALKBACK MICROPHONES

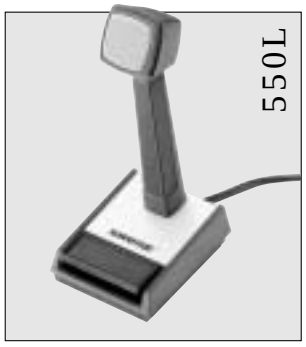


The 450 Series II is a high output dynamic microphone designed for paging and dispatching applications. Extremely rugged, it features an omnidirectional pickup pattern and a frequency response tailored for optimum speech intelligibility. It also includes an impedance selection switch and a locking press-to-talk switch. The press-to-talk switch can be converted to a monitor/transmit switch with the Shure RK199S Split-Bar Conversion Kit.



The 522 is a dynamic base station microphone designed for radio communications, paging, and dispatching systems. Cardioid pickup pattern suppresses unwanted background noise (other dispatchers working nearby, ventilating equipment, office machines). It also eliminates or reduces feedback in PA paging applications. Convert to Monitor/ Transmit switching with the RK199S Split-Bar Conversion Kit.

The 550L is an omni directional dynamic base station microphone with a frequency response specially tailored for voice intelligibility. It is equally useful for radio communications, paging and dispatching systems, and its low-impedance connection makes it adaptable to most PA amplifiers.



Equipped with a transistorized preamplifier, the 526T Series II can be used to replace ceramic or dynamic, high- or low-impedance microphones supplied as original equipment. Designed for maximum versatility it has a momentary or locking press-to-talk transmit/receive switch, a modulation level volume control for undistorted output with high- or- low impedance inputs, and six-wire cable arranged for universal mic-transceiver connection.



514B

Omnidirectional Dyamic Handheld Microphone

The 514B is a low impedance, dynamic, hand-held microphone designed for paging and public address systems. It provides clear, natural voice response and high intelligibility. The 514B connects to microphone inputs rated at 19 to 300 W. It can be used in applications that require long cable lengths, or where severe hum conditions are present. For connection to high impedance inputs, use Shure A95 Series Line Matching Transformers. Designed to fit naturally and comfortably in the hand unaffected by heat or humidity, the mic includes a four-conductor (two shielded) coiled cable and a "Million-Cycle" leaf-type switch designed for constant use under the most rigorous conditions. The exclusive ARMO-DUR case resists mechanical shocks and vibration, and is impervious to oil, fumes, salt spray, ultraviolet radiation, and corrosion..... **CALL**



527 Series

Dynamic Communications Mic

The 527 Series are extremely reliable, high impedance (527A) and low impedance (527B) dynamic hand-held microphones with an extended low- and high-frequency response particularly suited to ham radio use, mobile FM transmission, and top-quality paging systems.

The mics have a clear, crisp, and natural voice response. They feature professional design, small size and light weight. The modular construction of the 527 Series account for their tremendous ease of field serviceability. The cartridge, switch, and cable are easily replaceable, if necessary. The mics are wired so that the wires will not crimp or break. They fit naturally and comfortably in the hand and are unaffected by heat or humidity. The exclusive ARMO-DUR case is immune to oil, grease, fumes, salt spray, sun, rust, and corrosion and is outstanding in its ability to withstand mechanical shocks and vibrations. The "Million-Cycle" leaf-type switch is double-pole, single-throw, designed to withstand severe operating conditions and constant usage.

527A Microphone
200Hz to 5kHz frequency response, high impedance **48.95**

527B Microphone
300Hz to 5kHz frequency response, low impedance..... **54.50**

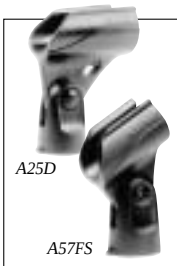
527C Microphone Built-in transistor amplifier **59.50**

MICROPHONE ACCESSORIES

SWIVEL ADAPTERS, WINDSCREENS AND SHOCK MOUNTS for SHURE MICROPHONES

MICROPHONE	SWIVEL ADAPTER	SHOCK MOUNT	WINDSCREEN
515 (all)	A25D*	A55M/A55HM	A1WS
545L	A25D*	A55M/A55HM	A2WS
565 (all), 588SDX	A25D*	A55M/A55HM	A58WS
849-LC	A25D*	A53M	A3WS
BETA 53	—	—	RPM304, RPM306
BETA 56	—	—	A1WS
BETA 57A	A25D*	A55M/A55HM	A57AWS
BETA 58A	A25D*	A55M/A55HM	A58WS
BETA87 (all)	A25D*	A53M	A85WS
BETA 98 (all)	—	—	RK183WS
EZG/12	—	A27SM	RPM310
EZG/18	—	—	RPM310
EZO	—	—	RPM312, RPM314
KSM27/SL	—	A27SM	A32WS, PS6
KSM32	—	A32SM*	A32WS, PS-6
KSM44	—	A44SM*	A32WS, PS-6
MX183, MX184, MX185	—	—	RK183WS
MX202B(P)/C, S, N	—	—	RK183WS
MX412, MX418	—	A400SM	RK412WS
PG48, PG58	A25D*	A55HM, A55M	A58WS
PG56	—	—	A85WS
PG57	A25D*	A55HM, A55M	A85WS
PG81	A25D*	A53M	A3WS
SM2	—	—	RK184WS
SM7B	—	—	A2WS, RK345
SM10A, 12A	—	—	RK184WS
SM48, SM58	A25D*	A55HM, A55M	A58WS
SM57	A25D*	A55HM, A55M	A2WS
SM63	A57F*	A53M	RK229WS
SM81	A57F*	A53M	A81WS
SM-86	—	A55HM, A55M	A85WS
SM87A	A25D*	A53M	A85WS
SM89	A57F*	A89SM	RK257WS
SM93	—	—	RK303BK
SM94	A25D*	A53M	A3WS
VP64A	A57F*	—	49A130
VP88	—	A88SM	—
WH20XLR	—	—	RK318WS
WH30XLR	—	—	90WP1371
WL50, WL51	—	—	RPM304, 306, 308*

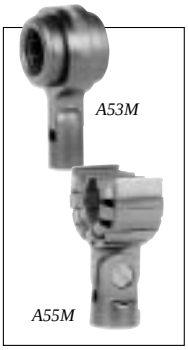
** included accessory*



A25D Stand Adapter
Break resistant adapter for the SM57, SM58, SM87A, Beta 87A and all other mics with 3/4″ and larger handle ...**6.50**

A57FS Stand Adapter
Swivel stand adapter for the SM62, SM63, SM63L, SM81, VP64 .....**6.50**

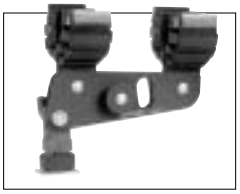
Shock Mounts



A53M Isolation Mount
The A53M isolation mount/swivel adapter reduces transmission of mechanical noise and vibration by more than 20dB. Compatible with the 849, SM81 and SM94, or any Shure mic with a tapered handle **37.50**

A55M Isolation Mount
An isolation mount, the A55M Shock Stopper reduces transmission of mechanical noise and vibration by more than 20dB. Compatible with the SM57, SM58, SM87/87A/87C, Beta 58A and Beta 87A, or any mic with 3/4″ and larger handle.....**44.95**

A55HM Isolation Mount
Designed for microphones with a tapered handle, the A55HM (Half Mount) Shock Stopper features a patented design that allows microphones to be snapped in and out without removing the cable **39.95**



A88SM Isolation Mount
Designed for the VP88 or any mic with a handle diameter of 1⅛″, the A88SM provides exceptional freedom from mechanically transmitted noise. Through its use of the highly effective Shock Stopper half-mount isolators and secure cable locking design, the A88SM eliminates most noise associated with stands, booms, and cable movement **158.50**



A89SM Shock Mount
The A89SM reduces mechanical noise transmission by more than 20 dB. Designed for the SM89 shotgun mic, it also fits any 3/4″ diameter microphone. Three integrated thread sizes allow you to attach the mount to stands or booms with 3/8–16, 5/8–27, or 5/16–18 threads. A cable securing mechanism prevents the transmission of mechanical noise through the cable and connector. Locking, pivoting design provides maximum aiming capabilities.....**91.95**

Quick-Release Shock Mounts for Gooseneck Mics

These accessories allow permanent installation of a quick-release shock mount for use with a wide variety of gooseneck mics.



A400SM
Offering Shock-Stopper construction, the A400SM decouples the mic from the mounting surface and reduces impact or surface noise by up to 20 dB **24.95**

A400XLR
The A400XLR assembly consists of a 3-pin XLR-type (female)connector and insert. Existing A400SM Shock Mounts can be retrofitted for quick release performance with the A400XLR Connector/Insert.....**22.50**

A400SMXLR
An A400SM Shock Mount with the A400XLR Connector/Insert **39.95**

MICROPHONE ACCESSORIES

MIC-SPECIFIC FOAM WINDSCREENS

For 10A and 515 Series *(A1WS)* Gray **5.95**

For 16A and 16L *(A4WS)* **5.95**

For the 545 Series and SM57 *(A2WS-BLK)*
Locking black foam **6.95**

For the 545 Series and SM57 *(A2WS-GRA)*
Locking gray foam **6.95**

For 838, 839, SM83 *(RK242WS)*
Contains four **12.50**

For Beta 57 and Beta 57A *(A57AWS)*
Locking foam **7.50**

For KSM27, KSM32, KSM44 *(A32WS)*....**12.50**

For MX183, MX184, MX185, Beta 98 and WH30 *(RK183WS)*
Contains four black snap-fit screens **18.95**

For MX412 and 418 *(RK412WS)*
Contains four snap-fit screens **18.95**

For SM1, SM2, SM10, SM12, and SM14 *(RK184WS)* **15.95**

For SM7A, SM7B, and SM7 *(A7WS)* **23.95**

For SM61 and SM62 Gray. *(A61WS)*.....**5.95**

For SM63 *(RK229WS)* **8.95**

For SM81 and SM57 *(A81WS)* **29.95**

For SM81 *(A81G)*
Locking pop filter grille and windscreen..**39.95**

For SM84, MX183, MX184 and MX185 *(RK261BWS)*
Contains four. Small, black **17.50**

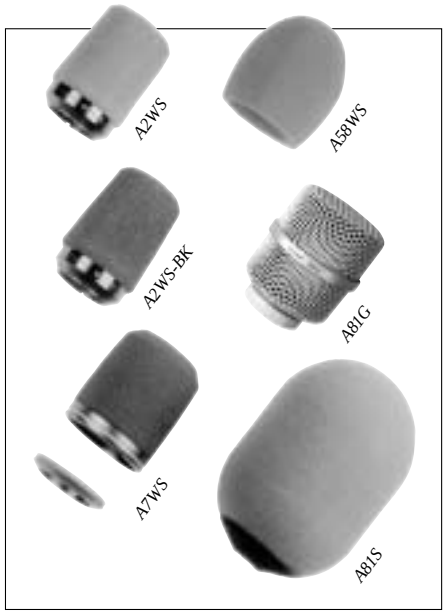
For SM85, SM87A, SM87A, and SM87C *(A85WS)*.....**12.50**

For SM89 *(RK257WS)* **39.95**

For SM81-LC *(RK311)*.....**6.50**

For SM93 *(RK355WS)*
Contains four. Black. **17.95**

For SM94, 849 and BG4.1 Gray. *(A3WS)*....**8.50**



Foam Windscreens for all Shure Ball Type Microphones

Black Foam *(A58WS-BLK)*.....**6.25**
Blue Foam *(A58WS-BLU)*.....**6.25**
Gray Foam *(A58WS-GRA)* **6.25**
Green Foam *(A58WS-GRN)*.....**6.25**
Red Foam *(A58WS-RED)* **6.25**
Yellow Foam *(A58WS-YEL)*.....**6.25**

Gooseneck Windscreens

For Easyflex Gooseneck Mics *(RPM310)*
Charcoal Gray, 4 Pieces **6.50**

For Easyflex Overhead Mics *(RPM312)*
Charcoal Gray, 4 Pieces **6.50**

For Easyflex Overhead Mics *(RPM314)*
White, 4 Pieces **6.50**

High Performance Ball Foam Windscreen *(A99WS)*
For Microflex Gooseneck Mics **5.95**

Locking Metal Windscreen *(A412MWS)*
For Microflex Gooseneck Mics **19.95**

Windscreen and Clip Packages

Tan Foam Windscreens and Tie Clips *(RK304T)*
For SM93 (Contains two of each).....**24.95**

Foam Windscreens & Clothing Clip *(RK318WS)*
For all WH20 headsets. Contains 2 windscreens and 1 clothing clip **5.95**

Stands

3˝ Extension Tube for Desk Stands *(A26X)*
Adapts Beta 56 to various mounting devices.....**8.95**

Small Black Desk/Table Stand *(A202BB)*
For MX202 microphones.....**28.50**

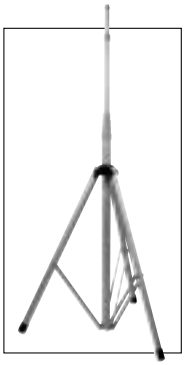
Accessory Base *(A412B)*
For Microflex Gooseneck microphones, 10´ attached XLR cable.....**59.50**



Chrome Floor Stand *(MS-10C)*
A chrome floor stand with gray metallic base, the MS-10C quickly and easily adjusts from 35˝ to 64˝ high.....**39.95**

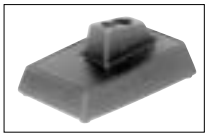
Tripod Desk Stand (S40B).....19.95

Baby Boom Arm for MS-10C *(BB77)*
A 31˝ chrome-plated, adjustable boom that fits the MS-10C floor stand.....**29.95**



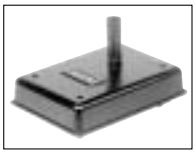
S15A Telescoping Mic Floor Stand
The S15A is a rugged and stable, yet portable and light-weight tripod floor stand. Features five telescoping sections, adjustable height between 3.5´ and 14´. Includes a carrying case.....**157.50**

S37A Desk Stand



Provides a low profile, stable base for microphones with swivel connector assemblies, or microphones using swivel adapters and shock mounts. Has a non-reflective, textured, gray finish **26.95**

S39A Vibration-Isolation Stand



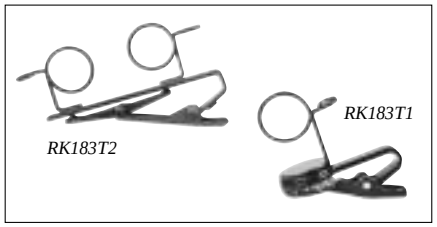
Provides excellent shock isolation, preventing floor or table vibrations from reaching a mounted mic. Use the S39A for sportscasting, conference table or front-of-stage installations, or any application where severe vibrations may occur **62.95**

MICROPHONE ACCESSORIES

Tie Clips

Single Mount Tie-Clips *(RK183T1)*
For MX183/184/185 (2 pc.) **32.95**

Dual Mount Tie-Clip *(RK183T2)*
For MX183, MX184 and MX185.....**14.95**



Tie Clasp for SM11 *(RK203TC)*.....**12.95**

Single Mount Tie Bars *(RK240SB)*
For 839, SM83, SM84 (4 pc.) **22.95**

Dual Mount Tie Bars *(RK241DB)*
For 839, SM83, SM84 (4 pc.) **23.95**

Double Tie Clips *(RK307DB)*
For SM93 (2 pc.).....**17.95**

Black Single Bar Tie Clips *(RK354SB)*
For SM93 (2 pc.).....**12.95**

Tan Windscreens & Single Tie Clip *(RK304T)*
For WL93T (2 pc.).....**24.95**

Black Swiveling Lapel and Dual Tie Clips
For MC50B (two of each) *RPM500*.....**37.95**

Tan Swiveling Lapel and Dual Tie Clips
For WL50 (two of each) *RPM502* **37.95**

White Swiveling Lapel and Dual Tie Clips
For WL50 (two of each) *RPM504* **37.95**

Black Swiveling Lapel Clips *(RPM510)*
For MC50B, WL50 and Beta 53 (Contains 5 clips).....**27.50**

Tan Swiveling Lapel Clips *(RPM512)*
For WL50 and Beta 53 (5 clips).....**26.50**

White Swiveling Lapel Clips *(RPM514)*
For WL50 (5 clips) **27.50**

Accessory Cables

15´ Triple-Flex Cable (TA4F to TA3F)
For BETA91, BETA 98/S, BETA 98D/S...**24.95**

25´ Extension Cable for VP88 (C110)...**66.95**

15´ Accessory Extension Cable (TA4F to TA4M).....**24.95**



Mounts and Adapters

Dual Microphone Mount *(A26M)*
For 545 Series, SM57 and Beta 57A.....**39.95**

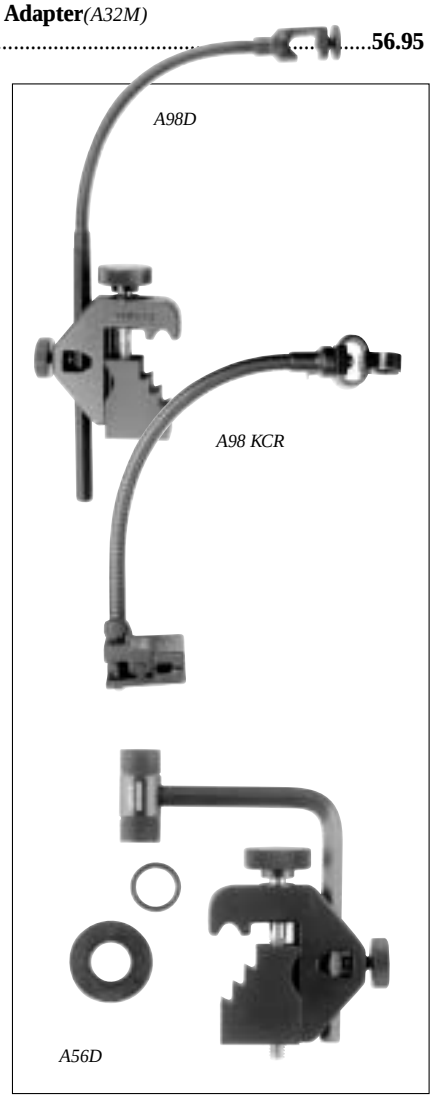
ShureLock Black Swivel Mount Adapter*(A32M)*
For KSM27 microphone.....**56.95**

Universal Microphone Drum Mount *(A56D)*
Mounts a mic onto a drum. A screw on the mount secures the mic, and an adaptive brass washer (european insert) is available to accommodate an even wider range of microphone models. Flexible, sturdy rubber construction. Accommodates 5/8˝ Swivel Adapters **37.95**

Drum Mount *(A98D)*
Mounts the Beta 98/S or SM98A mics on a drum. The gooseneck-adapter allows for optimum mic placement, and can be inserted into the horizontal or vertical slots of the clamp for positioning flexibility. The clamp is designed to fit drum rims measuring from ½˝ to 2¼˝ in height, or on cymbal stands measuring from ⅜˝ to 1⅛˝ in diameter.....**38.50**

Universal Horn Mount *(A98KCS)*
Attaches the Beta 98/S or SM98A to a horn, wind, or reed instrument in the exact position desired while protecting the instrument and its finish. The cushioned grip is manually tightened and securely clamps the rim of the horn. Non-marking neoprene rubber on the grip prevents marring the bell or its rim. A short 6½˝ gooseneck attached to the clamp terminates in a shock-mount adapter for the microphone. The gooseneck length can be adjusted to facilitate locating the microphone in precisely the desired position in the bell of a horn. A side screw locks the gooseneck in the chosen position.....**49.50**

Swivel Adapter *(RK282)*
Shock-mounted swivel adapter for SM98A and Beta 98S microphones.....**18.50**





A27M Stereo Microphone Adapter

The A27M is a stereo microphone adapter designed to hold two mics in selectable positions for stereo broadcasting, recording or sound reinforcement. It allows two microphones to share one stand without losing independent positioning flexibility.

- Permits coincident or closely spaced mounting with a wide range of directional angles, and vertical microphone separations of 1¼˝ (31.8 mm), 2⅝˝ (66.7 mm) or 4˝ (102 mm).
- The A27M consists of two vertically stacked sections that rotate on their centers. Each contains a 5/8˝–27 threaded stud and locking ring to accept a variety of microphones. The bottom section contains a 5/8˝–27 adapter for mounting on a mic stand. Includes cable clamp for holding mic cables.....**59.95**

MICROPHONE ACCESSORIES

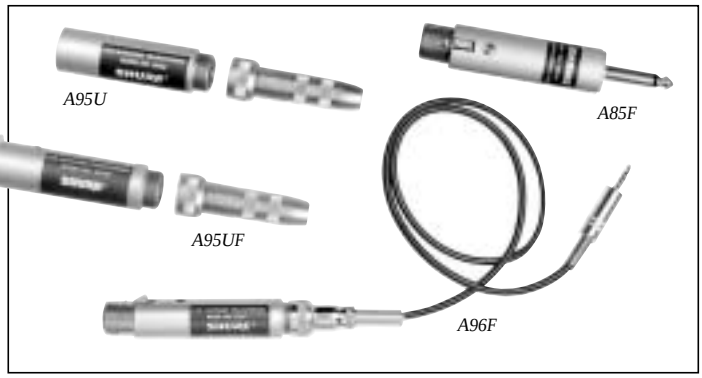
A96F
Low- to Medium- Impedance Matching Transformer

Low impedance, female XLR to medium impedance 3.5mm mini-plug output connector with 18″ cable. Adapts a balanced, low-impedance microphone to camcorder/ recorder input. Also has a DC Bias Filter **36.50**

A95UF/A95U
Low- to High-Impedance Mic Transformer

The A95UF and A95U provide 24 dB of voltage gain and may be used in reverse (high- to low- impedance). The A95UF comes with a balanced low impedance 3-pin XLR female input and the A95U comes with a balanced 3-pin XLR male low-impedance connector. An Amphenol MC1M-type output connector with mating 1/4″ male phone plug and female phone jack adapt to a variety of high-impedance signals.

A95UF.....**34.95** **A95U**.....**32.50**



A85F
Economy Low- to High-Impedance Mic Matching Transformer

Provides 24 dB of voltage gain and may be used in reverse. Balanced 3-pin XLR female input and 1/4″ male output.....**17.50**

Problem Solvers

No matter how well you plan ahead, sooner or later you will probably run into an audio-related problem. For example, dealing with incompatible signals of a low impedance microphone and a high impedance mic input of a camcorder or eliminating the low end rumble picked up by a sensitive mic due to handling noise or even automobile traffic. Shure has developed a number of problem solving solutions to help you out in these types of situations and many others.

A15AS Switchable Mic Attenuator



Prevents preamplifier overload when strong signals are applied.
- Provides 15, 20, or 25 dB of microphone attenuation
- Requires use of balanced signals; XLR female in and XLR male out
- Passes phantom power **38.50**

A15PRS Switchable Phase Reverser



Provides instant switch selection of normal or reversed phase of a balanced line without modification of equipment.
- Reverses polarity of pins 2 and 3 of a balanced line
- Slender, in-line XLR (F) to XR (M) configuration; may be used in reverse
- Requires use of balanced signals.....**38.50**



A15HP High-Pass Filter

Reduces unwanted low end rumble in microphone signals.
- Provides low-end roll-off of 12dB per octave below 100 Hz
- Slender, in-line XLR (F) to XLR (M) configuration
- Requires use of balanced signals.....**38.50**



A15LA Line Adapter

Provides 50 dB attenuation, allowing a balanced line-level source to be connected to a balanced microphone input.
- Slender, in-line XLR (F) to XLR (M) configuration
- Requires use of balanced signals.....**34.95**

A120S Accessory In-Line Switch



Provides on/off, push-to-talk, cough button, and transmitter relay keying functions for wired mics.
- Convenient handheld size with belt clip
- For hardwired, in-line use with various cable diameters **24.95**

A15BT Bridging Transformer



Matches balanced or unbalanced devices of different impedances.
- 33 kOhm primary: 600 or 7500 ohm secondary; may be used in reverse
- Slender in-line XLR (F) to XLR (M) configuration...**39.95**

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ECM-44/55/66/77/88

Lavalier Microphones

Sony's lavalier microphones are manufactured to meet the most demanding professional broadcast and recording requirements. They are designed to deliver the highest level of performance and reliability. There are five primary lavalier mics in the series, offering different sizes and applications. Additionally, they offer three big advantages over other systems.

High Performance

Sony's lavalier microphones are distinguished by their superior performance, as represented by the wide (20Hz~20kHz) frequency response of the top-of-the-line ECM-88. They will more than measure up to your expectations.



Wide Selection

Five models in the series, each finished in black plating. The ECM-88 is an extremely miniature, omni-directional electret-condenser mic designed for quality-critical applications in broadcasting, theater, and field productions. While slightly bigger, the ECM-77B is also incredibly small and light. The uni-directional ECM-66B is designed for close-miking of musical instruments. The popular ECM-55B incorporates a 10.6mm omni-directional capsule for versatility, while the ECM-44B offers high performance at a lower cost. There is also a full line of lavalier accessories available.

Reliability and Convenience

Increased reliability is another important plus for these mics. Careful attention is paid to make the capsule immune to external noise and adverse ambient conditions. The cable is strengthened to achieve a high resistance to damage caused by over flexing.



ECM-44B

Omni-directional Electret Condenser Mic

The ECM-44B makes the exceptional quality of Sony's electret condenser mics available for budget conscious productions, and a variety of sound reinforcement requirements, such as lectures and demonstrations. Measuring 8.5 x 14.5mm, the ECM-44B delivers clean, bright, yet natural reproduction. Complete with in-line battery for 2-way powering (battery or 48v phantom power). Operates on a single AA battery for over 5000 hours. Supplied with urethane windscreen and clip.

ECM-55B

Omni-directional Electret Condenser Mic



The ECM-55B is suitable for applications ranging from news and interviews to stage dramas and instrument pick-up. Frequency response is a wide 30Hz to 18 kHz. Above 2KHz, there is a gentle rise in the response to give a touch of presence to any instrument, while offering crisp, clean voice quality without any masking effect when used as a lapel mic. Directivity is optimized to ensure uniform, dependable output, regardless of the direction of the sound source. The metal mesh windscreen effectively eliminates both wind noise outdoors and pop noise in close-up situations. May well prove to be your most versatile and useful lavalier microphone. Complete with in-line battery for 2-way powering (battery or 48v phantom power). Supplied with metal windscreen and metal holder clips

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ECM-44/55/66/77/88

ECM-66B *Uni-Directional Electret Condenser Mic*



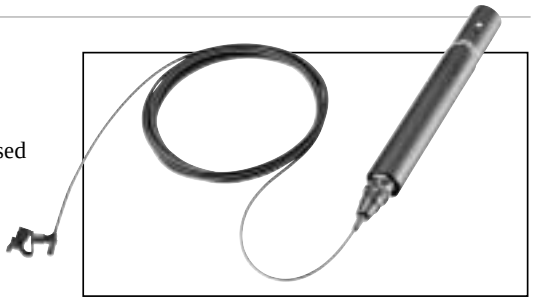
A uni-directional miniature microphone developed specifically for recording musical instruments and voice when superb rejection of the ambient sound is desirable. The design of the ECM-66B yields clean, natural reproduction of musical instruments while maintaining excellent isolation from the acoustic environment. Unlike most miniature microphones, this mic has a uni-directional polar pattern, carefully contoured for better control of feedback. The higher voltage supplied to the microphone by the built-in DC-to-DC converter increases the maximum input sound pressure level to an amazing 130dB SPL. This special construction also complements the electret condenser design, resulting in excellent transient response and thus better tonal quality. Complete with in-line battery for 2-way powering (battery or 48v phantom power). Has a battery on/off switch to save power. An LED illuminates to show battery condition when the power is turned on.



ECM-77B *Omni-directional Condenser Microphone*

At only 1⁄4″ x 1⁄2″, and a mere 0.04 oz. the ECM-77B is microscopic. Increased mechanical impedance using a metal vaporized, fine ceramic backplate provides the ECM-77B's exceptional 40Hz to 20 kHz frequency response. And, by design, the ceramic backplate assures resistance to humidity and temperature extremes for increased reliability. Combined with advanced techniques for sealing the capsule and advanced design for the acoustic housing, the ECM-77B delivers sound that is audibly superior and more pleasing than any other lavalier mic. Ideal for TV broadcasting and stage applications where the microphone must be inconspicuous, the ECM-77B enables unobtrusive miking plus also minimizes glare. Even when hidden inside a costume, its frequency response is wide enough to pick up the voice with no perceptible change in quality, while minimizing the pick-up of rustling noise.

ECM-88 *Omni-directional Condenser Microphone*



The ECM-88 is an extremely miniature, omni-directional electret-condenser microphone designed for quality-critical applications in broadcasting, theater, and field productions. It used a dual-diaphragm mechanism to achieve high-sensitivity, flat-and-wide frequency response and low noise characteristics, while offering great enhancements over the acclaimed ECM-77B. The diaphragm and back-plates are fixed vertically to the microphone capsule, which effectively reduces the mechanical noise caused by cable vibration and friction. Water-resistant architecture reduces the risk of water or perspiration entering the mic capsule. The mic capsule measures just 5/32 x 5/32 x 11/16″ allowing for easy, natural-looking concealment in a stage costume.

The ECM-88 includes two types of mic holders (double-pin and tie-clip) and a urethane windscreen. It is supplied with a Sony 4-pin connector (SMC9-4P) for use with the optional DC-78 power-supply or the WRT-8B/822B wireless bodypack transmitters. The optional DC-78 power supply has a 4-pin connector (SMC9-4P) input and XLR output. It can operate via 12- 48v external DC power or up to 5000 hours on a single AA battery.

SONY MICROPHONE SPECIFICATIONS

	ECM-44B	ECM-55B	ECM-66B	ECM-77B	ECM-88
Capsule type	Electret Condenser	Electret Condenser	Electret Condenser	Electret Condenser	Electret Condenser
Frequency response	40 - 15k Hz	30 - 18k Hz	70 - 14k Hz	40 - 20k Hz	20 - 20k Hz
Dynamic range	≥90dB	98dB	≥101dB	≥90dB	≥99dB
Signal-to-noise ratio*	≥62dB	66dB	≥62dB	≥64dB	≥68dB
Max input SPL	122dB SPL	126dB SPL	130dB SPL	≥130dB SPL	125dB SPL
Battery life	Approx. 5000 hrs.	Approx. 5000 hrs.	Approx. 300 hrs.	Approx. 5000 hrs.	
Sensitivity (0dB=1V/Pa,at 1kHz)	-53.0dB ± 3dB	-52.0dB ± 2dB	-50.0dB ± 2dB	-52.0dB ±2dB	52 dB +/- 2 dB
Output impedance at 1kHz	250Ω	100Ω	100Ω	150Ω	100Ω

**(A weighted, 1kHz, 1 Pa.)*

STUDIO MICROPHONES

Holder Clips and Windscreens for Lavalier Mics



Tie Clips
For the ECM-44 *(SOSC44)* .**14.95**
For the ECM-77 *(SOSC77)* .**14.95**

Horizontal Single-Clip (10-Pack)
For the ECM-44 *(SOSADH44B)*.....**73.95**
For the ECM-55 *(SOSADH55B)*.....**73.95**
For the ECM-77 *(SOSADH77B)*.....**73.95**

Horizontal Double-Clip Pack (Pack of 6)
For ECM-55 *(SADW55B)*.....**Call**
For ECM-77 *(SADW77B)*.....**Call**

Black Windscreens
Urethane. For ECM-44 *(SOWS44)*.....**17.95**
Urethane. For ECM-77 *(SOWSU77S)*.....**19.95**
Metal. For ECM-77 *(SOWSM77S)*.....**19.95**



Urethane Windscreen (6 colors x 2 pc)
For ECM-77 *(SOADC77)*.....**91.95**

Metal Windscreen (Black, 6-Pack)
For ECM-66 (*SOADR66B)* **94.95**
For ECM-77 *(SOADR77B)*.....**94.95**

C-48
Studio Condenser Mic



A multi-pattern, large diaphragm condenser microphone, the C-48 is powered via battery or external power supply. Vibration-proof structure eliminates noise transmitted through cable or stand. Use for vocal and instrumental recording

- Selectable patterns include: Uni-, Omni- or Bi-Directional
- Powered via internal battery for up to 50 hours or external power supply
- l0dB attenuation switch
- 2 position low cut switch

C-800G Pack
Tube Studio Microphone



Excellent for vocal recording in studios and film post production houses, Sony's C-800G studio tube condenser mic is designed for the highest possible sound reproduction quality. It features a high-quality dual large diaphragm for true reproduction of vocal qualities, vacuum tube warm sound quality through a 6AU6 vacuum tube, a unique Peltier-based cooling system to achieve optimum tube operating temperature, and an electronically selectable pick-up pattern (Omni/Uni) for various recording applications. The package includes a power supply (AC-MC800G) and all necessary accessories.

- Suitable for vocal recording in studios and film post production houses
- High sensitivity of -33dB/Pa
- Large diaphragm capsule
- Electronically selectable directivity, either omni- directional or cardioid
- Low noise and low distortion due to built-in cooling system

ECM-670 / ECM-672
Shotgun Microphones



Compact and lightweight, the 9˝ ECM-670 and 12˝ ECM-672 are high-performance, electret condenser super-cardioid shotgun-type microphones designed for use on portable video cameras such as Sony's Betacam and DXC series. Tailor-made for ENG applications, they also offer versatility in applications such as interviews, sound reinforcement and speech. Superb controlled directivity assures rejection of ambient and camera noise, while the supplied windscreen reduces wind noise and provides pop protection during extra close miking.

ECM-670

- Low-cut switch provides a low frequency roll-off for optimum voice pick-up reducing pop and wind noises.
- Powered by external power supply (12-48v DC) via an XLR connector.
- Includes windscreen, mic holder, mic spacer and 2 stand adapters.

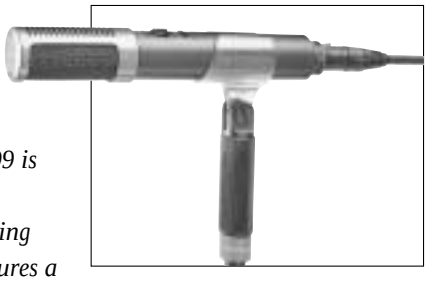
ECM-672

- A 2-position low-cut switch further contributes to the mic's low noise level and emphasizes desired sound sources by eliminating boominess in the low range.
- Powered by 12-48v DC power supply or via AA battery (up to 3000 hours of operation).
- Includes a windscreen

STEREO MICROPHONES

ECM-999

Stereo Electret Condenser Mic



With a selectable stereo image, AA battery operation, and a built-in low cut switch, the ECM-999 is ideal for a multitude of recording and broadcasting applications. It also features a single-point stereo pickup and an MS design with 3 electret condenser capsules for stereo recording with a single microphone.

- Electret condenser microphone with back-electret condenser capsules permits good sound pick-up.
- Mid/Side system: The sum of signals of the mid microphone unit (uni-directional) and side microphone unit (bi-directional) and the difference between them are used for R and L channels respectively. The Mid/Side system is employed to give excellent sound image and faithful stereo sound reproduction with less "hole in the middle".
- The directive angle between the left and right channels can be changed progressively from 0° (monaural) to 150° according to the sound source.

ECM-MS957

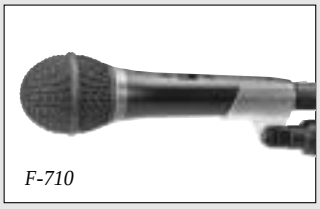
One-Point Stereo Mic



Incorporating a sensitive electret condenser mic, a rotating mid-capsule and Mid/Side switch, the ECM-MS957 is designed for high-quality instrument recording. Ideal for use with Minidisc and DAT recorders, it also features a wide frequency response, superb dynamic range, and an XLR connector.

- Mid/Side (MS) capsules for natural stereo panorama; the Mid capsule picks up monophonic sound while the Side capsule picks up left/right difference sound; subtracting and adding the two capsule signals yield separate.
- The mid-capsule rotates to adjust the stereo pick-up pattern for the widest range of recording situations.
- Mid/Side (MS) switch selects pickup angle between left and right channel; choose 90° for a single voice or instrument or 120° to pick up many voices and instruments, arranged across the stage
- XLR connector for secure and reliable, low-noise connection
- Supplied accessories include detachable cable, stand, holder, windscreen and carrying case

F-Series Dynamic Microphones



F-710



F-720



F-740/9X



F-780/9X

F-710

Low cost, multi-purpose dynamic handheld mic with high sonic quality and high sensitivity. Suitable for voice and instrument pickup, the mic's Neodymium magnet offers high output. On/off switch.

F-720

Rugged dynamic hand-held cardioid mic with a built-in LC switch. Replaceable capsule. Efficient one-piece shock mount to protect the capsule from external noise and vibration. Integral double windscreen to prevent pop/wind noise. Reasonably priced to suit both professional and institutional applications. Black finish.

F-740/9X

Designed for critical live vocal and instrumental reproduction, the F7409X features an improved capsule design for enhanced audio performance, as well as a AlNiCo magnet assembly and CCAW (Copper Clad Aluminum Wiring) voice coil for very crisp, clean, transparent sound.

- Low handling noise
- Superb sound quality and high sensitivity
- Extremely durable design

F-780/9X

A perfect choice for critical live vocals or where high-quality, rugged reliability, and resistance to feedback are required. Improved sonic and mechanical characteristics make the F-780/9X a must have for professionals in the music and recording business.

- Rigid structure allows easy movement of capsule for higher sensitivity and reduced mechanical noise
- Urethane coating for extremely low handling noise

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The Professional Audio SourceBook

Section 1a

Microphone Accessories

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MICROPHONE CASES

Gator Tough - Musician Friendly

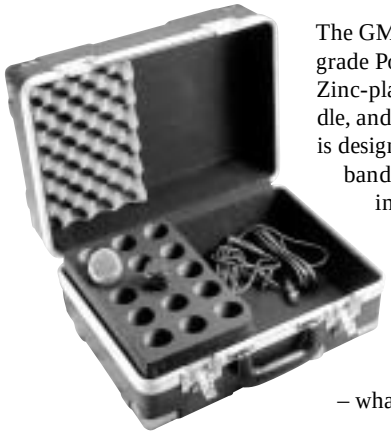
Gator Cases was founded to meet the needs of professional and hobby musicians who need to safely protect their gear. All of their cases are designed by musicians to insure that they meet the most demanding needs.
Gator Cases offers innovation, quality and price. They use only the highest quality latches and components to insure the best performance. To back up this dedication, all Gator Cases have an unlimited lifetime warranty.

GM-4
Padded Bag for 4 Microphones and Cables



The GM-4 is the perfect carrying case featuring 4 form fitted foam microphone drops and a slot for a belt pack. The exterior has a large dual pocket for cables and accessories. The case comes with a comfortable top carrying strap as well as a shoulder strap. Made from 600 Denier Ballistic Nylon, this case is made for heavy usage at an affordable price.

GM-15
Professional 15 Drop Microphone Case



The GM-15 is constructed from military grade Polyethylene and features recessed Zinc-plated twist latches, padded handle, and triple-plated Chrome corners. It is designed to take the beatings of a band-on-the-road hitting 70 venues in 18 countries. The interior has 15 microphone drops that are formed with thick foam to insure maximum protection plus a generous area for cables, belt packs or accessories. Lockable and affordable – what a concept!

GX-33
Microphone/Stand Bag

The GX-33 has 3 separate compartments for tripod-type mic stands as well as 5 form-fitted mic drops. Both bags are constructed of 600 Denier Nylon, have 2 webbed handles and a carrying strap.



GM-6
Molded Plastic Case with Six Microphone Drops

Gator's GM-6 microphone case is molded from ABS plastic and is lined with thick die-cut foam to fit six microphones. The durable latches are lockable and triple chrome plated.



MIC CASE DIMENSIONS

	Interior				Exterior			Price
	Length	**Width**	**Depth**	**Comments**	**Length**	**Width**	**Depth**	
GM-4	12″	4½″	6″		13″	8¼″	5″	**16.95**
GM-6	15″	9¾″	2½″		16″	11″	3½″	**29.95**
GM-15	9 1/2″	12⅛″	8″	15 Mic Drops	19″	13½″	9″	**79.95**
GX-33	43″	7½″	57½″	5 Mic Drops	45″	8″ Diameter		**26.99**

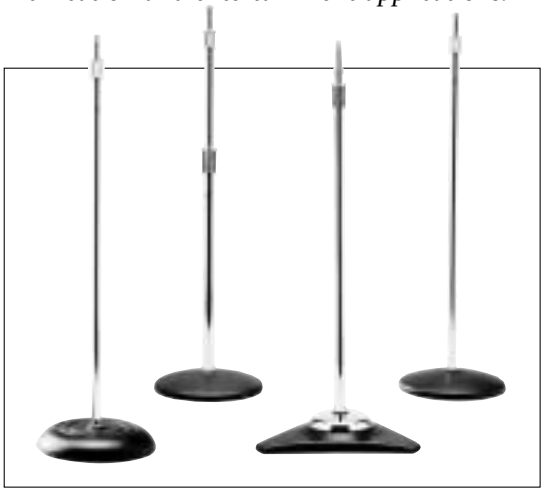
MIC STANDS

Microphone and Equipment Stands, Booms, & Accessories

Synonymous with durability and quality, Atlas is the world recognized leader for cast-base and tripod style floor stands, desk, and studio boom stands; boom attachments; loudspeaker/equipment stands; and over 40 adapters and fittings. The unique selection, developed over the past 60 years, offers the professional musician, performer, educator, and lecturer great flexibility in choosing models that satisfy unique application, performance, and budget requirements. Classic styling in mirrored chrome or non-reflective ebony epoxy (perfect for video application) combine with the famous Atlas Sound wear-proof clutches to provide attractive, durable, and reliable service for all types of communication and entertainment applications.

MS Series Microphone Floor Stands

Your microphone is a valuable investment, so why trust it to a cheap stand. Manufactured with fine U.S. craftsmanship, Atlas stands are simply built better. Each clutch is made of a wearproof, three piece construction. The bases and housings are high-stability cast metal, and the tubings are finished in ebony or mirrored chrome. With more than 60 years of performance, both on-stage and off-stage, these stands are proven performers. And with 12 models to choose from, Atlas offers the widest selection of floor stands available.



MS-10C / MS-10CE
All-Purpose Floor Stands

The cost-efficient MS-10C (chrome finished tube) and MS-10CE (ebony finished tube) assemblies are the most popular stands in the world for dependable performance in music, entertainment, business, education, and social functions. Stability is provided by the circular cast iron ebony base.

MS-10C.....**28.50** **MS-10CE**.....**28.50**



MS-12C / MS-12CE
Professional, Full-Height Stands

These all purpose stands feature the same wearproof clutch and high-quality craftsmanship as the MS-10 series with the beauty of a lower profile base section.

MS-12C.....**28.50** **MS-12CE**.....**28.50**

MS-4 / MS-4E / MS-43 / MS-43E
Fully Adjustable Three-Section Stands

These versatile stands adjust from 25″ to 65″ to meet the needs of seated and standing instrumentalists and performers. Model includes three-section chrome (MS-4/MS-43) or ebony tubed (MS-4E/MS-43E) assembly with two wearproof clutches for optimum vertical height adjustment and high stability. The MS-4 Series has a low-profile cast iron base finished in ebony (diameter of 10″, weighs 6 lbs), while the the MS-43 Series base is 12″ in diameter and weighs 8 lbs. (also finished in ebony).

MS-4.....**69.95** **MS-4E**.....**69.95**
MS-43.....**69.95** **MS-43E**.....**69.95**

MS-20/ MS-20E
Extra-Stable Stands with Oversized Base

Heavy-duty stands for studio, recording, church and school applications, the MS-20 (chromed finish tube) and MS-20E (ebony finished tube) include an extra-height 1⅛″ diameter tube assembly with wearproof clutch, top adapter, and low contour high-stability cast iron ebony base.

MS-20.....**59.95**
MS-20E.....**57.95**

MS-25 / MS-25E
Studio Stands with Air Suspension System

The MS-25 (chromed finish tube) and MS-25E (ebony finished tube) are heavy-duty studio recording stands featuring a 1⅛″ diameter tube assembly with integral air suspension system to protect sensitive microphones. Triangular cast base includes sculptured sides, ebony finish, and decorative chrome cover.

MS-25.....**89.95** **MS-125E**.....**89.95**

MIC STANDS

TRIPOD STANDS

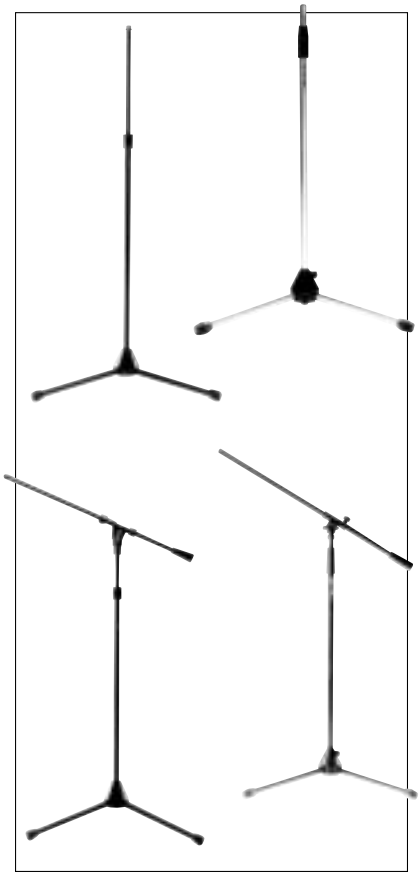
TL34E Performer Series Tripod Stand
The TL34E features a durable, all-metal housing with a unique double-cam locking device and Glide'n Lock legs for secure positioning and convenient transportation/storage. The wearproof Atlas clutch withstands years of stage and studio handling, while the high-density rubber feet will reduce mechanical noise to the microphone. 2-piece adjustable tubing (34″–63″H) terminates in the standard 5/8″-27 thread pattern. Non-reflective ebony epoxy finish **46.50**

TL34-BBE/ TL34-15E/ TL34-21XE
Performer Series Tripod/Boom Combinations
Tripod/boom stands offer maximum ease and portability. Packages include the patented TL34E Performer Series tripod with either a 34″ economy boom, or a choice of Performer Series booms: 34″ or 25¼ - 38½″. Packages are finished in ebony epoxy. The Performer Series booms feature the patented all-metal swivel clamp with Glide'n Position tubing, nylon brake bushings and wool washers. There is also a 3/4 lb. counterweight for balance, and steel T-bar knob for no-fail locking.

TL34-BBE TL34E tripod with 34″ BB-77E economy boom **73.50**

TL34-15E TL34E tripod with 34″ PB15E Performer Series boom **89.95**

TL34-21XE TL34E tripod with 25¼-38½″ PB21XE Performer Series boom **99.95**



TE-E Musician Series Tripod Stand
Lightweight, professionally styled stand with the fit and feel of tripod costing much more. Features full-height extension (37 to 65″) to meet versatile miking needs, a three-piece clutch with protective insert & metal clutch barrel for secure and scratch-free positioning, and easy-locking legs that tighten with a single motion of the metal leg attachment "cup" for fast set-up and teardown. Durable tripod legs attach to a metal retainer assembly that includes a thumbscrew/ brake system to provide secure positioning and prevent tube marring. Tube assemblies are durable CRS construction terminated in 5⁄8″-27 thread pattern. Satin ebony finish.

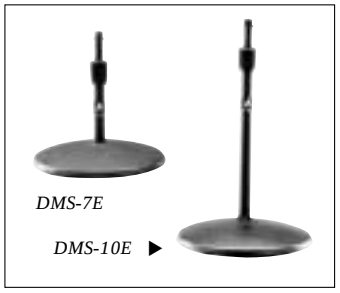
TE-E Tripod Stand (5-1⁄2 lbs) **29.95**

TEB-E Tripod/Boom
Features the TE-E stand with the removable BE-E telescopic boom to provide microphone extension from 24 to 39″. The boom includes a molded plastic handle, metal swivel clamp with protective rubber inserts, and an easy-grip knob for vertical alignment. Adjustment for the main and extension boom include a thumbscrew/brake system to provide secure positioning and prevent tube marring Tube assemblies are manufactured of CRS and terminate in 5⁄8″-27 thread pattern. Satin Ebony finish.

TEB-E Tripod Stand/Boom (7½ lbs) **49.95**

INSTRUMENT & DESKTOP MIC STANDS

DMS-7/ DMS-10 Low-Height Adjustable Stands



Low-profile stands with high-stability cast base, the DMS-7 and DMS-10 are ideal for professional drum miking applications. They offer adjustable-height tubing and Atlas Sound clutch for precise orientation and a 10″ diameter cast iron base for dependable positioning and isolation from drum vibrations. Threaded tip assembly for compatibility with mic clamps, goosenecks, and booms. (PB11X Mini-Adjustable Booms is a great companion).

DMS-7: Floor-stand base with desk-stand height (8-13″), ideal for kick-drum miking. Chrome or Ebony finish (DMS-7E).....**23.50**

DMS-10: Taller version (14½-26¼″), perfect for miking toms and larger guitar amps. Chrome or Ebony finish (DMS-10E)**.....25.95**

DS-5 / DS-5E General Purpose Stands
These fixed-height stands include traditional, circular cast iron base and choice of chrome or ebony tubing finish. Stands are suitable for application with any standard microphone.



DS-5 Chrome finish .....**9.95** **DS-5E** Ebony finish.....**9.95**



DS-2 Vibration Isolating Stand
A low-silhouette stand, the DS-2 features an integral tension mount to reduce conductivity of external mechanical vibrations. High-stability metal base with protective pads includes notched area for convenient placement of cards or pens. Fixed-height tubing is chrome finished. Base is ebony **24.95**

DS-7 / DS-7E Adjustable Stands
These versatile stands have a vertical height adjustment from 8″ to 13″ and include wearproof clutch and ebony finish cast iron base.
DS-7 Chrome finish .....**16.50** **DS-7E** Ebony finish.....**16.50**

MIC STANDS AND BOOM ARMS

STUDIO BOOM STANDS

SB-11WE Economical Studio Boom
An economical studio boom, the SB-11WE is especially suited for small studio/broadcast applications and for stage miking of drums and percussion instruments. Unit features a 60″ long, two-piece horizontal boom assembly. Boom includes a Performer Series swivel clamp and an adjustable 2 lb. counterweight for effortless boom arm orientation. Tubing incorporates a 90° angled end for optimum microphone positioning and terminates in the standard 5/8″-27 thread pattern. Stand features an Atlas Sound wearproof clutch for positive locking control. Triangular die-cast base offers extra stability and is equipped with hard rubber swivel casters for effortless and silent mobility. Finish is non-reflective ebony epoxy **149.95**



SB-36W Professional Studio Boom S
A heavy-duty stand for stage and studio applications, the SB-36W includes an integral piston-type air suspension system for effortless height adjustment and microphone protection.

- Equipped with a two-piece horizontal chromed-tube assembly, adjustable 6 lb. boom counterweight, heavy-duty gyromatic swivel, Atlas wearproof clutch, and triangular cast iron base
- Includes silent-motion, ball-bearing swivel casters of hard rubber for friction-free and vibration-absorbing mobility
- Tubing terminates in the standard 5/8″ - 27 thread pattern, and supplied with guide clips for microphone cable attachment. Tubing is finished in chrome; base is ebony epoxy with a chrome cover

SB-36W Includes wheels for a height span of 49″-73″ **199.95**

BOOM ATTACHMENTS

Performer Series booms are manufactured in the U.S. to the highest quality standards, and they combine exceptional styling with a unique, all-metal swivel clamp for greater flexibility. Clamp features Glide'n Position tubing with nylon brake bushings for smooth horizontal adjustment. Resilient washers are made from compressed virgin wool fiber (in lieu of rubber) to offer quiet and secure positioning even after years of use. Extra-strength, steel T-bar knob assures no-fail locking with simple one-hand adjustment in the vertical plane. All versions include counterweights for increased balance and stability. All clamps are finished in non-reflective ebony epoxy. Tubing is manufactured of CRS and terminates in the standard 5⁄8″-27 thread pattern.

PB11X / PB11XE Mini Adjustable Boom
Compact booms extend from 16¼″ to 24½″ with a single motion—perfect for drum/piano/guitar amp and short distance miking. Units include a threaded end cap and a fixed 3⁄4 lb. counterweight for stability.
PB11X Chrome.....**39.50**
PB11XE Ebony epoxy.....**39.50**

PB15 / PB15E Fixed Length Booms
All-purpose, high-quality booms are 34″ long and include fixed 3⁄4 lb. counterweights for stability.
PB15 Chrome.....**32.95**
PB15E Ebony epoxy **32.95**

PB21X / PB21XE Extendable-Length Booms
Booms extend from 25¼″ to 38½″ with a single motion and are suitable for a wide variety of miking applications. Units contain a fixed 3⁄4 lb. counterweight and threaded end cap.
PB21X Chrome.....**39.95**
PB21XE Ebony epoxy.....**39.95**



BE-E
Musician Series Extendable Length Boom
Versatile, low-cost BE-E boom has adjustable-length positioning for extended microphone range applications. Two-piece boom extends 24-39″. The boom includes a molded plastic handle, metal swivel clamp with protective rubber insert, and an easygrip knob for vertical alignment. Adjustment for the main and extension boom include a thumbscrew/brake system to provide secure positioning and prevent tube marring.
BB-E Satin Ebony (2-1/2 lbs.) **24.95**

BB-77 / BB-77E
General Purpose Fixed-Length Booms
Basic, multi-purpose 34″ booms, they feature a fixed 3⁄4 lb. counterweight, thread-on die cast swivel, single-action positive locking, and new easy-grip knob for positioning control. Ideal for application with floor stands or accessory components.
BB77 Chrome finish **19.95**
BB77E Ebony epoxy.....**19.95**

B&H PHOTO - VIDEO - PRO AUDIO EQUIPMENT LEASING AVAILABLE

MIC STAND ACCESSORIES

Atlas offers the industry's largest selection of problem-solving microphone adaptors, extensions, fittings and flanges for use with floor, desk, loudspeaker, and studio boom stands.

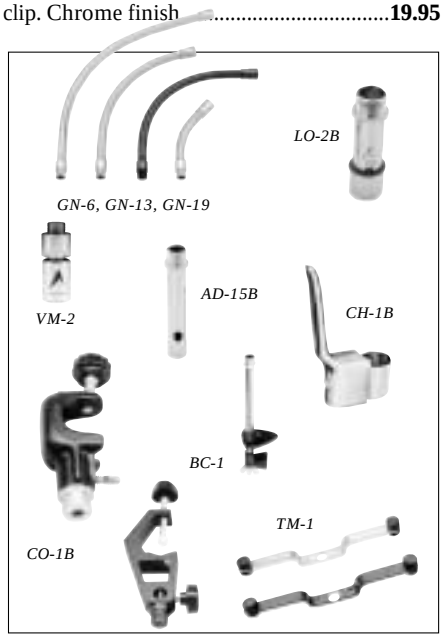
Gooseneck Flexible Extensions
6, 13, and 19″ flexible goosenecks attach to any mic stand or adapter with 5⁄8″-27 male and female threads. Inside diameter is .338″.
GN-6 Chrome finish. 6″ gooseneck.....**4.95**
GN-6E Ebony finish. 6″ gooseneck.....**4.95**
GN-13 Chrome finish. 13″ gooseneck.....**6.95**
GN-13E Ebony finish. 13″ gooseneck.....**6.95**
GN-19 Chrome finish. 19″ gooseneck.....**7.95**
GN-19E Ebony finish. 19″ gooseneck.....**7.95**

LO-2B Clip-Mounting Microphone Accessory
Lock-On accessory, respectively, allow fast connection and disconnection of plastic mic clamp holders and boom attachments with 5⁄8″-27 threads for safe transportation.
LO-2B Chrome...**13.50** **LO-2BE** Ebony...**13.50**

TM-1 Versatile mic mount for live sound and stereo recording. Mounts two or three mics onto any stand. Unit is 8¾″ wide and fits mic clamps with 5⁄8″ receptacles. Available in chrome or ebony.
TM-1 Chrome....**14.95** **TM-1E** Ebony.....**14.95**

US-2 Flange-mounting, multi-purpose mic support includes 12¾″ L boom on a spring-loaded swivel with 180° vertical cable feed-thru. Mounts mics to lecterns, consoles, tables, and desks. Chrome finish.....**49.95**

VM-2 Durable shock-mount accessory reduces external mechanical vibration pick-up. Die-cast unit can be left on a stand during transit unlike similar plastic accessories. Fits 5⁄8″-27 threaded mics and mounts between the top tube of a 5⁄8 ”-27 threaded stand and the mic clip. Chrome finish.....**19.95**



BC-1 Multi-Purpose Clamps Bracket-style surface clamp with 6″ high chrome tube attaches to tables, desk, keyboard stands, and countertops for mounting mic and accessory items with standard 5⁄8″ threads.....**19.95**

CO-1B Connect-on swivel adapter/clamp provides for the addition of a second mic with standard 5⁄8″-27 threads to any floor or desk stand with 5⁄8″ or 7⁄8″ diameter tubing. Ebony finish.....**14.95**

MAC-1 Multi-use adapter/clamp holds mics, booms, or accessories having 5⁄8″-27 threads in vertical or horizontal orientation. Versatile unit also clamp mounts to 5⁄8″, 7⁄8″, and 1⅛″ diameter tubing or to freestanding instruments, keyboards, and drums. Made of high-strength extruded aluminum. Ebony finish. Weighs 1.5 lbs.....**34.95**

CH-1B Cable/Tambourine Hangers and Cable Feed-Thru Adapter Versatile accessory can hang cabling or a tambourine. Single screw-mounting assembly fits all tube assembly sizes.
CH-1B Chrome.....**14.95**
CH-1BE Ebony.....**14.95**

AD-15B Chrome 4″ cable feed-thru adapter with standard 5⁄8″ male and female threads.....**9.95**

CLAMP-ON *Mic-Eze*

Made of nylon with glass, each Mic-Eze model features lockable clamp design, while the clamp opens to 2 inches. They weigh 2 oz. and fit in the palm of your hand. Telescopic arm extends 1/4″ or 3″, and their arm swings toward clamp 180°. Clamps mounts in horizontal and/or vertical plane. The M-4 holder is a perfect match as the mic holder.

M-1: Patented mic placement design enables drummers to mic their drum set using their personal choice of mic without the need of mic stands. Simply clamp and lock to rims and stands.....**17.95**

M1-E3: Modified version of the M-1. A male threaded 3″ extension arm is perpendicular to the clamp.....**24.95**

M1-E5: Same as above with 5″ extension arm.....**24.95**

M3-E3: The versatility of this model sets the 3″ extension arm vertically with the clamps in either a vertical or horizontal plane. The flexible arm can bend in any direction.....**24.95**

M3-E5: Same as above with 5″ extension arm.....**24.95**

M1A-E3: A 3″ male extension arm runs in line with the clamp and can swing and bend in all directions.....**24.95**

M1A-E5: Same as above with 5″ extension arm.....**24.95**

M2: Heavy duty mic holder that threads to the base models or any mic stand.....**17.95**

M4: Heavy duty spring action microphone holder that holds any size microphone.....**17.95**

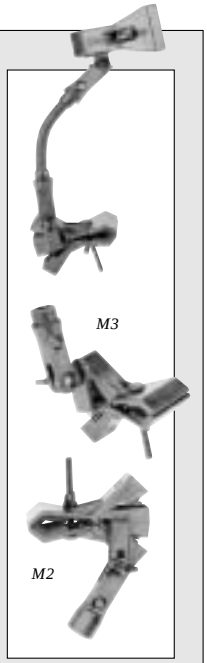
FE3: Dual perpendicular clamp set with a 3″ flexible center that can bend in any direction.....**34.95**

FE5: Same as above with a 5″ flexible center.....**34.95**

Min-Eze: Dual perpendicular clamp set with a trapeze style center attachment. Very popular for the congo drum.....**34.95**

Mac-Eze: Dual vertical clamp arrangement with a short center attachment.....**34.95**

PB (Piggy Back): Mounts mics to mics for dual feed and adjusts for phase cancellation. Dual vertical in-line clamp set with a swing type center attachment.....**34.95**



MICROPHONE STANDS

MC-05B Round Base

- Proven clutch design for easy adjustment and reliable locking
- Easy and comfortable adjustments
- Traditional, sturdy, cast-iron base

ULMC05B **19.95**

MC-07B Liberty

Same as MC-05B plus:

- Cut-out design allows six stands in the footprint of one - less than 1 square foot of floor space
- Heavy base is tip resistant

ULMC07B **27.95**

STACKING STANDS

The cut-out of the cast iron base allows six **MC-07B** or **MC-77B**'s to stack in the footprint of one. Each stand interlocks with the one underneath for convenient transport and storage.



MC-77B Freedom

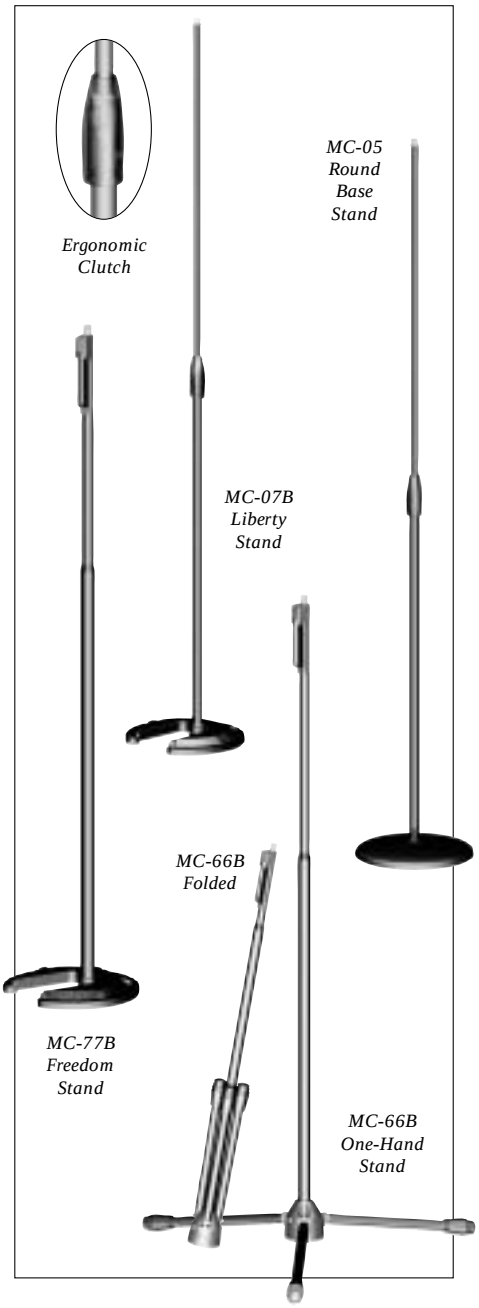
- Unique one-hand clutch lets you adjust height with a quick squeeze to easily raise or lower the mic
- Cut-out design allows six stands in the footprint of one - less than 1 square foot of floor space
- Heavy base is tip resistant

ULMC77B **54.50**

MC-66B One-Hand

- Unique one-hand clutch lets you adjust height with a quick squeeze to easily raise or lower the mic
- Durable lightweight aluminum construction makes a very portable stand
- Tripod legs fold up and lock for compact transport

ULMC66B **72.50**



MMB-962B Mini Mic Boom



The 9⅝″ mini boom is the perfect accessory to any Ultimate mic stand.

MMB-962B Standard threads
MMB-962B Metric threads

MC-11B Low-Level Stand



Telescoping stand with lightweight and portable glass-fiber reinforced resin base and telescoping boom. Black finish. It's height is 19″- 46″, it weighs 3 lbs. 7 oz., and has a boom length of 16″- 29″

ONE-HAND CLUTCH



The one-hand clutch on **MC-66B** and **MC-77B** make these mic stands one of the most innovative, yet easy stands on the market. Simply squeeze the clutch, releasing the internal friction mechanism, and set your height. Let go of the clutch and the internal lock engages for a secure and reliable adjustment.

MICROPHONE STANDS

	MC-05B	MC-07B	MC-77B	MC-66B
Height	34¾ - 64″	34¾ - 64″	39 - 64″	40½ - 66″
Width	8 lbs. 13 oz.	9 lbs.	8 lbs.	3 lbs.
Base	Round	Round Stacking	Round Stacking	Tripod
One-Hand Clutch	—	—	✓	✓
Finish	Black	Black	Black	Black
Warranty	3-year	3-year	Lifetime	Lifetime

MIC STANDS/STAND & BOOM PACKAGES

Mic Stands

Quality tripod base mic stands allow you to mix and match with the boom of your choice.

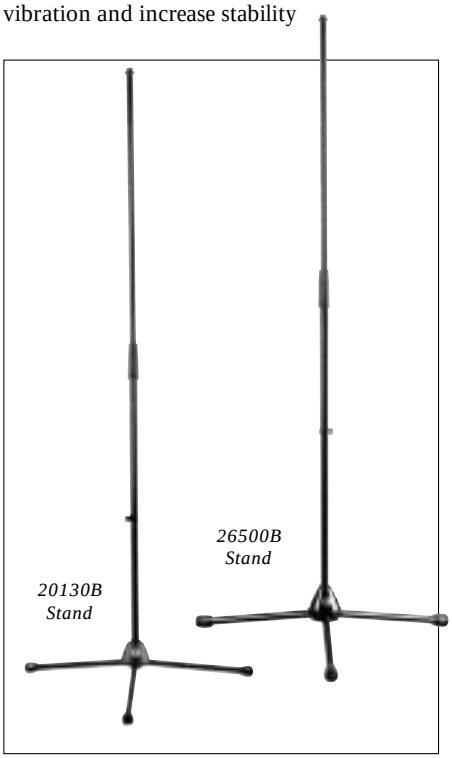
Height Adjustment
Proven clutch design for easy adjustment and reliable locking

Heavy Duty Construction
Heavy gauge steel stand is very stable

Durable Finish
Plastic inserts prevent metal-to-metal contact and protect the finish

Folds Easily and Compactly

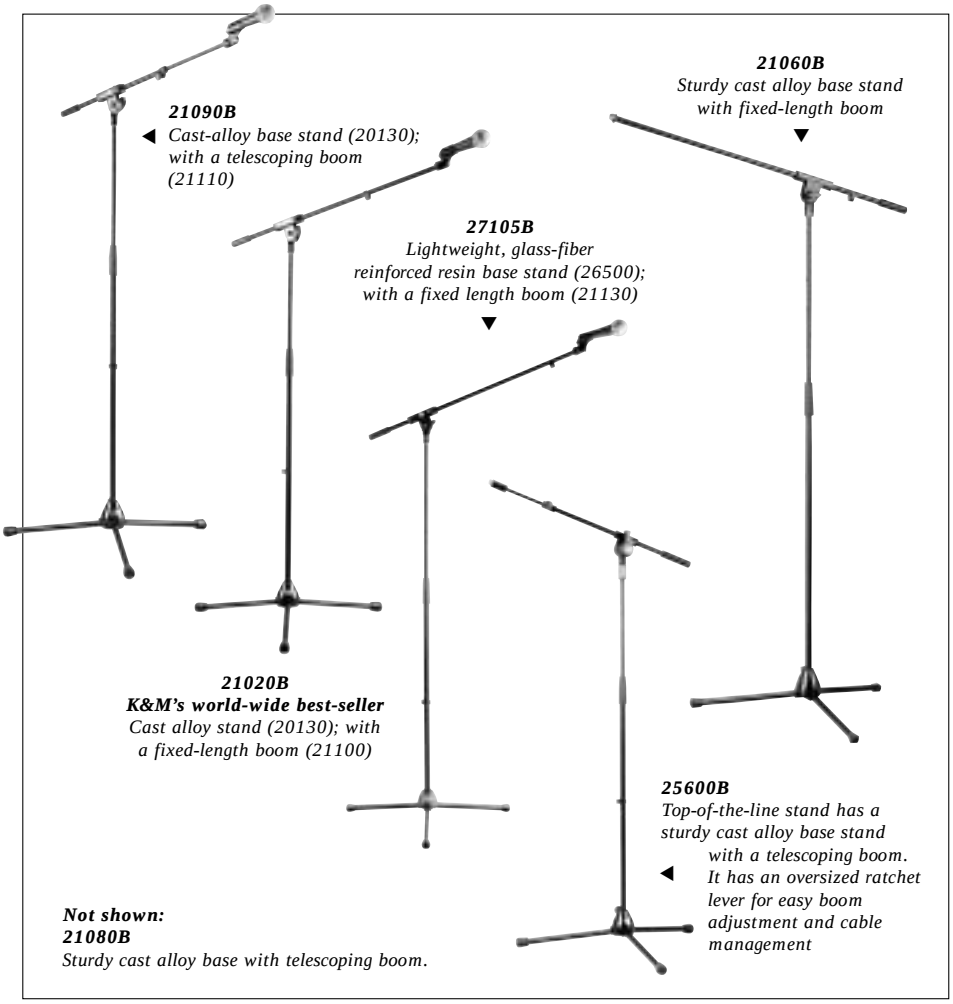
Sturdy Base
Stands with cast alloy sockets have stress-absorbing inserts to protect legs, reduce vibration and increase stability



	20130B	26500B
Stand Height	35½″ - 65″	35½″ - 63½″
Weight	6 lb.	4 lb. 4 oz.
Base	Cast Alloy	Cast Resin
Base Diameter	28″	27″
Price	**42.95**	**33.95**

Mic Stand and Boom Packages

The heavy-duty K&M Mic and Boom packages have a durable finish and feature rubber washers and easy-to-grip knobs that fasten booms securely. The stands with cast-alloy sockets have stress-absorbing inserts to protect legs, reduce vibration and increase stability.



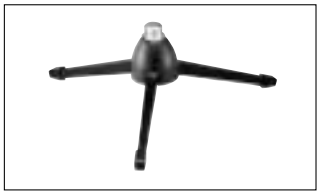
Stand	Finish	Height	Weight	Boom Length	Base Diameter	Price
KM21090B	Black	35 - 65″	7 lb. 10 oz.	18 - 30″	28″	**64.95**
KM21090C	Chrome	35 - 65″	7 lb. 10 oz.	18 - 30″	28″	**72.50**
KM21020B	Black	35½ - 65″	7 lb. 10 oz.	32″	28″	**59.95**
KM21060B	Black	35½ - 64½″	7 lb. 4 oz.	32″	27″	**57.95**
KM21080B	Black	35½ - 64½″	7 lb. 4 oz.	18 - 32″	27″	**57.95**
KM27105B	Black	35½ - 63½″	5 lb. 2 oz.	32″	27″	**49.95**
KM27105N	Nickel	35½ - 63½″	5 lb. 2 oz.	32″	27″	**49.95**
KM25600B	Black	37 - 65″	8 lb. 5 oz.	23 - 37″	28″	**106.50**

KONIG & MEYER

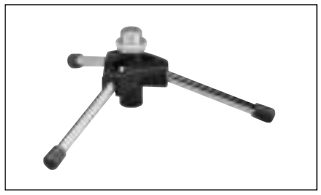
MICROPHONE STANDS

TABLE TOP MIC STANDS

TRIPOD BASE FLOOR STANDS WITH EXCEPTIONAL STABILITY



KM23105B - Small black stand with tripod base for table or desktop. Sturdy and foldable. 9.5″ base diameter (8oz.) **17.50**



KM23110N - Sturdy and foldable small nickel stand with tripod base for table or desktop. 8.5″ base diameter (6 oz.) **12.95**



KM23200 - Anti-vibration ring on round cast-iron base. Black color. 7″ height, 5″ base diameter. Weighs 3 lbs..... **29.95**



KM 23400 - Black telescoping, tiltable table or floor mic stand with a cast-iron base. Weighs 5 lbs. Height 15-25″. .....**Call**

SPECIALTY MIC STANDS

HARD-TO-FIND MIC STANDS FOR SPECIALTY APPLICATIONS

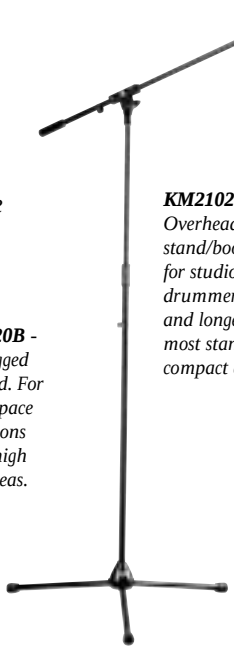
As with all K&M Mic Stands, these stands feature a proven clutch design for easy adjustment, a heavy gauge steel stand, plastic inserts to prevent metal-to-metal contact and stress-absorbing inserts to protect legs, reduce vibration and increase stability.



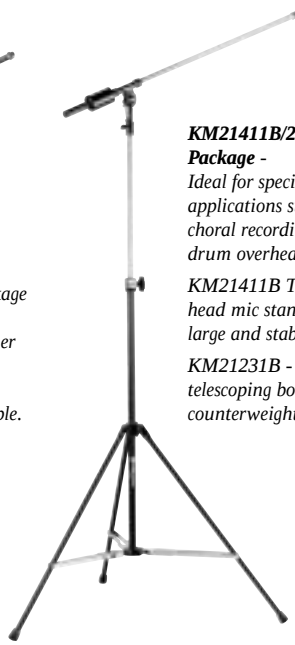
KM19900B - *Lightweight 3-section mic stand. Extra compact, this stand folds to 22″.*



KM20120B - *Short-legged mic stand. For limited space applications such as high traffic areas.*



KM21021B - *Overhead mic stand/boom package for studios and drummers - higher and longer than most stands yet compact and stable.*



KM21411B/21231B Package - *Ideal for special applications such as choral recording and drum overheads*

KM21411B Tall overhead mic stand with large and stable tripod

KM21231B - Extra long telescoping boom with counterweight.

Available separately. Order the KM21411B and the KM21231B for the complete package



KM25200B - *3-section mic stand/boom package. Cast alloy base with folding legs. Easy height adjustment. Complete with boom arm.*

	Stand Height	Boom Length	Weight	Base Diameter	Price
KM19900B	25½ - 59″	–	4 lb. 2 oz.	26″	**54.50**
KM21021B	43 - 80½″	42″	13 lb. 4 oz.	32″	**84.95**
KM20120B	35½″ - 64″	–	4.5 lbs	22″	**49.95**
KM25200B	24½ - 59″	18 - 30″	7 lbs.	28″	**84.95**
KM21411B/21231B	48 - 90″	45 - 81″	22 lb.	46″	**127.95/92.50**

B&H PHOTO - VIDEO - PRO AUDIO

STANDS, BOOMS & ACCESSORIES

Low Level Floor Stand Packages



KM25500B - Fixed height stand complete with extra-long telescoping boom. Can be used as a conventional upright stand.

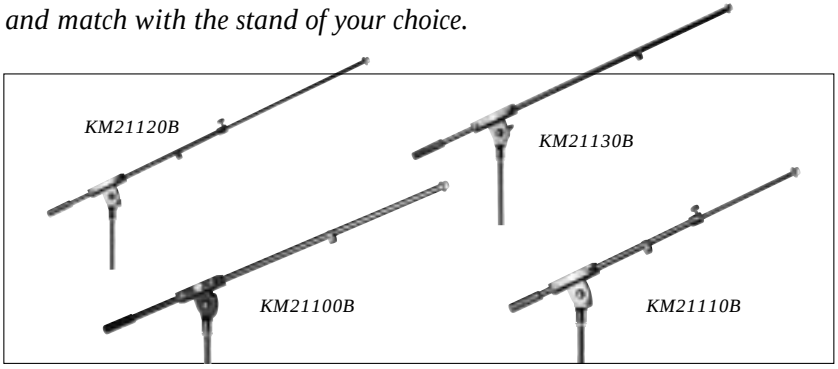
KM25910B - For bass drums or special use. Cast alloy base with folding legs. Comes complete with fixed length boom.

25900B - Telescoping stand and telescoping boom for bass drums or special use.

Stand	Height	Weight	Boom Length	Base Diameter	Price
KM25500B	16″	9 lbs	34 - 61″	26″	**68.95**
KM25900B	14 - 25″	5 lbs.	18 - 30″	22″	**64.95**
KM25910B	10½″	4 lb. 8oz.	21″	22″	**54.50**

Mic Booms

The quality you expect from K&M microphone booms so you can mix and match with the stand of your choice.

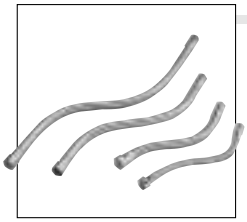


KM21100B (Black)
33″ Fixed length boom with counter weight (1 lb. 10 oz.) **26.50**

KM21120B (Black)
22″ - 38″ extra long telescoping boom with counter weight (1 lb. 14 oz.)....**29.95**

KM21110B/C (Black/Chrome)
18″ - 30″ telescoping boom with counter weight (1 lb. 10 oz.) .....**Call**

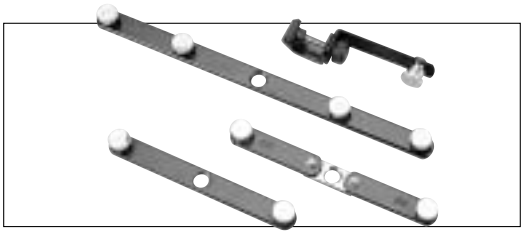
KM21130B (Black)
31″ Fixed length boom with counter-weight (1 lb. 14 oz.) **22.50**



Goosenecks

	22300N	22400B	22400N	22600B	22600N
Length	16″	12″	12″	9″	9″
Diameter	5/8″	5/8″	5/8″	1/2″	1/2″
Finish	Nickel	Black	Nickel	Black	Nickel
Price	18.50	16.95	16.95	14.95	14.95

Mic Bars and Holders

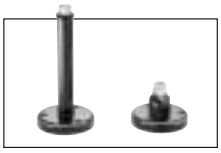


8″ Mic Bars
Available straight or hinged. Connect to the top of mic stands. Holds 3 mics or booms.
KM23500B - 8″ straight bar **14.95**
KM23510B - 8″ hinged bar **18.95**

16″ Mic Bar *(23600B)*
Similar to the 23500B, but holds 5 mics.....**26.50**

Mic Holder *(23800B)*
Attaches microphones to sheet music stands or microphone stands up to 1″ diameter **14.95**

Metal Flanges



Mounts to any flat surface.

KM22110B - 5″ tube with a 3″ diameter.....**18.95**

KM22130B - 1½″ tube with a 3″ diameter. Lateral mount for gooseneck and hole for cable passage**18.95**

KM22140B - 1½″ tube with a 3″ diameter.....**8.95**



Mic/Accessory Mounting Arms

KM24050B - 10″ arm attaches to round or flat surfaces up to 2¼″. Angle adjustable. Weighs 1 lb .**29.95**

KM24010B - 12″ arm attaches to stands with a prism clamp locking assembly. Complete with 22″ boom arm that adjusts and swivels. Weighs 2 lb.....**33.95**

Threaded Adapters



KM21500N - 1/2″ & 3/8″ female to 5/8″- 27 male.....**2.95**
KM21600N - 5/8″ -27 female to 3/8″ male **3.95**
KM21700N - 3/8″ female to 5/8″ -27 male **2.95**
KM21800N - 1/2″ female to 3/8″ male.....**2.95**
KM21900N - 3/8″ female to 1/2″ male.....**2.95**

MIC ARMS

ProBoom Elite Mic Arms

Offering the ultimate in ease-of-installation and aesthetics, the ProBoom Elite Series supports most modern studio mics. Mic cables are almost completely hidden in the provided wire channel by a zip top cap, yet are easily installed – with connectors attached. Music wire springs offer maximum holding power and minimum noise. ProBoom Elite Series come in overall extended lengths of 29″ or 45″, and are available as a set, or with arm and riser separately.

61900-BG: 29″ black mic arm with riser, gold springs **199.00**

61900: 29″ beige mic arm with riser, chrome springs.....**199.00**

14295-BG: 29″ black mic arm without riser, gold springs **129.00**

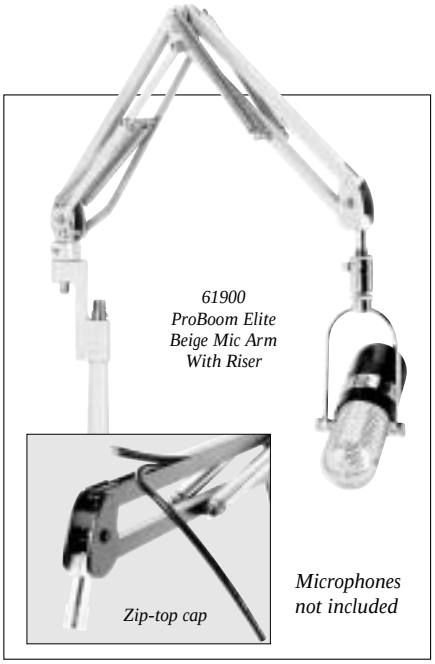
14295: 29″ beige mic arm without riser, chrome springs.....**129.00**

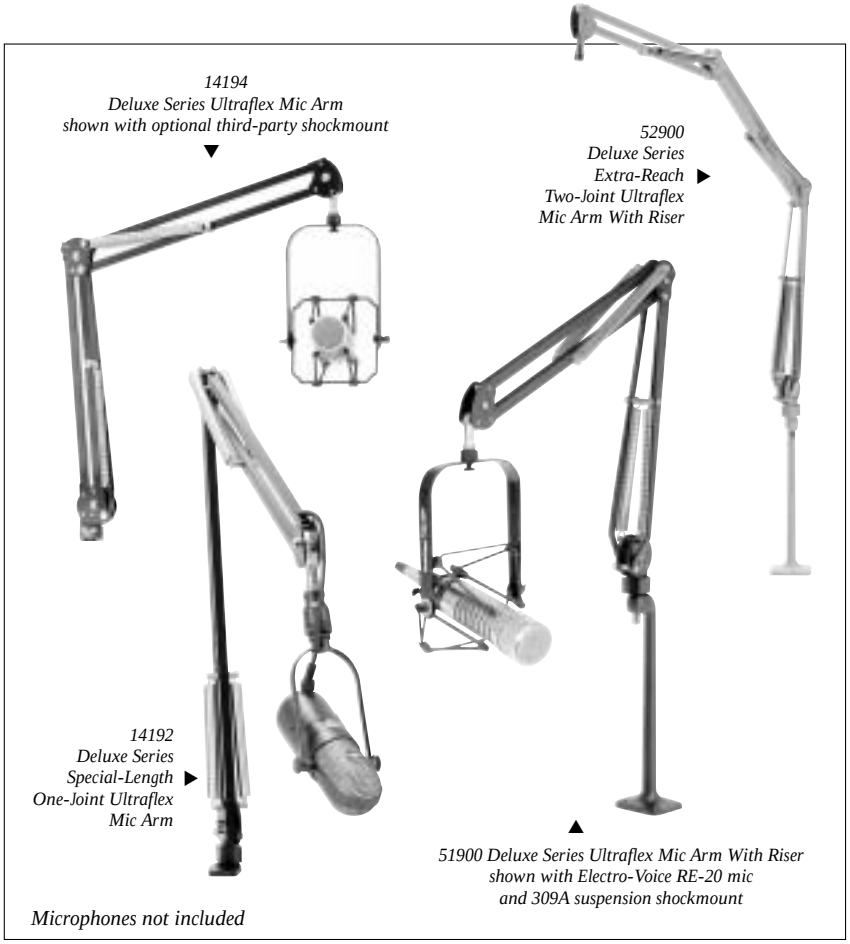
14199-BG: 45″ black mic arm without riser, gold springs **149.00**

14199: 45″ beige mic arm without riser, chrome springs.....**149.00**

14250-B: Black riser with 15″ overall height and built-in XLR connector that is prewired to a 5′ blunt.....**75.00**

14250: Same as above in beige color **75.00**





Deluxe Series Ultraflex Mic Arm

The industry-standard spring-counter-weighted Mic Arm is at home in production and control rooms, announcer's booths, and for voiceover situations. Constructed for years of trouble-free use, it offers a silent operation. It has an overall extended arm reach of 29″ and weighs 4 lbs. Supports most microphones without irritating spring changes. Accepts optional heavy-duty springs. (mount sold separately)

14194 .....**99.00**

Deluxe Series Ultraflex Mic Arm With Riser

This package adds 12″ of vertical height to a 14194 Mic Arm by combining it with a 14100 Riser Accessory. Permits placement of studio equipment (up to 10.5″ high) in close proximity to the arm, without obstructing movement. Weighs 5 lbs.

51900 .....**149.00**

Deluxe Series Extra-Reach Two-Joint Ultraflex Mic Arm With Riser

A double-jointed mic arm with a reach of a full 46″. Includes the 14100 Riser Accessory. Arm weighs 9 lbs.

52900 .....**179.00**

Deluxe Series Special-Length One-Joint Ultraflex Mic Arm

Popular in sophisticated studio integrations, it offers a totally-extended reach of up to 45″. Weighs 5 lbs.

14192 .....**129.00**

MIC ARMS

All O. C. White microphone stand products come with a standard 5/8˝ 27-thread mic connector and internal, through-the-arm wireways that allow cables to be kept hidden.

Available in either black or beige. Gray and other colors are available upon request.



Deluxe Series Ultraflex Talk-Show Roundtable Assembly

The easy way to accommodate multiple microphones in talk and roundtable broadcasts and webcasts. This combination includes three 14194 Deluxe Series Ultraflex Mic Arms and a TMA triple arm cast-iron mount. Weighs 17 lbs.
51900-3.....**367.00**

Wall Mount

Securely mount this heavy-construction bracket to the wall and you have the perfect match for mounting the O.C. White range of Mic Arms. Weighs 1 lb.
11426.....**22.00**

Bench, Table and Counter-Top Mount

Screws into place on counter top for permanent Mic Arm placement. Weighs 2 lbs.
11427.....**22.00**

C-Clamp Mount

This surface-saver permits Mic Arm anchoring to a table edge with the option for permanent or temporary mounting. Jaw capacity 2˝. Weighs 1 lb.
14005.....**25.00**

3 Arm Riser/Mount

This triple-mount accessory is perfect in interview and roundtable situations. Plug in 3 arms for multiple guests. Screw-down base. Overall height 11⅞˝. Weighs 5 lbs.
14111.....**75.00**

Diagonal Extension Arm

Extend arms a full 12.5˝ laterally, point-to-point. Upward 15° angle. Weighs 2 lbs.
EB-1.....**75.00**

Riser

Allows you to pull studio equipment up close by placing the mic arm a full 12˝ above the mounting surface, before any positioning commences. Studio equipment may be up to 10.5˝ high. Screw-down base. Weighs 1 lb.
14100.....**60.00**

Heavy Duty Springs

Helps to support vintage microphones such as the RCA 44BX and early BBC models on the Deluxe Series mic arms. Adding heavy-duty springs allows any of the Deluxe Series mic arms to support up to 8 lbs. Weighs 1 lb. Requires two.
12402.....**3.00 each**
12402-G Same as above, gold color **12.00**

Regular Duty Springs

Supplied with Deluxe Series arms. Accommodates a wide variety of popular modern-day mics. Weighs 1 lb. Requires two.
12404.....**2.00 each**
12404-G Same as above, gold color **10.00**



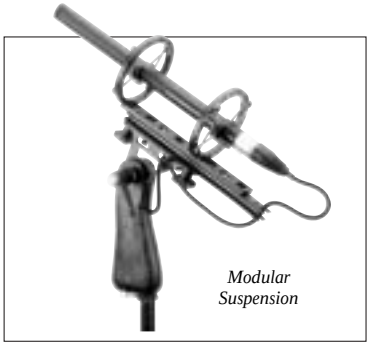
RYCOTE

WINDSHIELDS & SUSPENSION MOUNTS

Rycote's industry-standard windshield system components are available in multiple sizes to accommodate most of the mono and single-point stereo microphones of leading manufacturers. Each system consists of a Windshield mounted to a Modular (shockmount) Suspension, with the option of adding a Windjammer or a Hi-Wind Cover for additional wind protection.

Modular Suspensions

- Lightweight 4-point suspension with a high precision modular frame
- Available in 3 sizes to accommodate a wide range of microphones
- Includes both pistol grip and boom adapter as well as a range of clips and O-rings ensuring a custom fit



Modular Suspension

Windshields

- Up to 25dB exterior wind attenuation without unnecessary high frequency roll-off – an 8 to 10 dB improvement over foam windscreens
- Lightweight and durable – gray plastic netting tube with a specially attached screening material
- Twist and turn locking end caps are rattle free and secure
- Unique slide-on / slide-off system for easy removal of mic and suspension



Windshield

Windjammers

- Provides an extra 8 - 10 dB of protection over a standard windshield
- Specially designed materials with virtually no adverse effects on sound level or quality
- Available in sizes to fit all windshields



Windjammer

Hi-Wind Covers

- The Hi-Wind cover is a soft, dark gray, custom fitted stretch fleece fabric cover that fits snugly over the Windshield providing an extra 3 - 5 dB of attenuation during windy conditions



Hi-Wind Cover

Microphone	Windshield, Windjammer and Hi-Wind Cover Order Codes (A-Z)	Modular Suspension Size
CK8 451 / 460	P	M2
C568EB	I	M2
CK98 SE300B	P	M2
CK1, 61, 63 451 / 460	C	M1
CK91, 92, 93 SE300B	C	M1
CK68, 69 460 / 480 (Short)	N	M2
CK68, 69 460 / 480 (Long)	U	M2
Audio Technica		
AT 4071a	Q	M2
AT 4073a	I	M2
AT 822	F	M1
AT 825 / AT 835ST	H	M1
AT 815	W	M3
AT 835 / AT 815ST	Q	M2
Beyerdynamic		
716 / 726 / 736 / 836	L	M2
MCE86 / MCE87	M	M2
M160	B	M1
M88	F	M1
717 / 727 / 737 / 837	Y	M3
Neumann		
KM 84 / 184	B	M1
KM 100 Series	A	M1
KM 100F	C	M1
KMR 81	I	M1
KMR 82	S	M2
RSM 191 (100mm Windshield)	Y	M1
Schoeps		
CMC Series	B	M1
CMC Series CUT 1	E	M1
Sennheiser		
MKH 20, 40, 50	D	M1
MKH 30	E	M1
MKH 60	K	M2
MKH 70	R	M2
MKH 816	X	M3
ME64-K6 / MKE 44P	F	M1
ME66-K6 / K3U-ME80	O	M2
ME67-K6	U	M2
MKH 416	I	M2
Shure		
SM81	D	M1
SM89	X	M3
Sony		
C74	T	M3
C76	Z	M3
ECM 670	I	M2
ECM 672	L	M2
ECM MS5	E	M1

Ball Gags & Stereo Windshields

Rycote has been manufacturing windshield systems for more than thirty years, and, over that period, has built a reputation of being the industry standard. They were honored with an Academy Award for the design and development of their modular system. Rycote designs, develops and manufactures all of their own products, and continues to provide the industry with the most effective microphone windshield and shockmount systems available.



Baby Ball Gag windshields offer the wind attenuation properties of a full size windshield but in a much smaller size.
Available in three sizes – **20mm** for Schoeps CMC Series, **22mm** for Neumann KM 100 series and **25mm** for Sennheiser MKH 20, MKH 40 and MKH 50.
Baby Ball Gag Windjammers are available to provide additional wind protection.



140mm Stereo Windshields are available for a wide range of single point source stereo microphones and M/S stereo arrays. Please specify the make and model you wish to house and we will do the rest.

140mm Stereo Windshield.....**CALL**

Stereo Modular Suspension
Comes standard with Boom Adapter and Pistol Grip .....**CALL**

Stereo Hi-Wind Cover .....**CALL**

Stereo Windjammer.....**CALL**

Extended Ball Gags

Extended Ball Gags are windshield systems for remote capsule use of AKG Blue Line, Neumann KM 100 Series and Schoeps CMC and CCM microphones. Available with a 100mm diameter for single capsules and a 140mm diameter for Stereo M/S. Extended Ball Gag Suspensions come standard with a boom adapter.

100mm Mono Extended Ball Gag Components

- Mono Extended Ball Gag
- Mono Extended Ball Gag Suspension
- Mono Extended Ball Gag Suspension M/S Configuration
- Mono Extended Ball Gag Hi-Wind Cover
- Mono Extended Ball Gag Windjammer

140mm Stereo Extended Ball Gag Components

- Stereo Extended Ball Gag
- Stereo Extended Ball Gag Suspension M/S Configuration
- Stereo Extended Ball Gag Hi-Wind Cover
- Stereo Extended Ball Gag Windjammer



Stereo Extended Ball Gag XY Mount Assembly
Optional suspension assembly mounts inside of a 140mm Extended Ball Gag Suspension and permits variable XY (120°, 105°, 90°, 70°) or M/S capsule use.

Connbox

The Connbox is fitted to the Modular Suspension to isolate the mic from cable-borne noise. The mic tail uses an anti-rattle XLR at the mic and a snap-fit Binder at the Connbox. The output lead is a three pin or five pin XLR.



Order Code	Description	Microphone
CB1	Mono Connbox, Standard	Up to a short shotgun length
CB2	Mono Connbox, Long	Long shotgun
CB3	Stereo Connbox	Two mics in an M/S array
CB4	Stereo Connbox, Five Pin	Single shank stereo mic
CB4 Ext.	Stereo Connbox, Five Pin	Shure VP88
CB5	Mono Lemo ConnBox	(1) Schoeps CCM Mic
CB6	Stereo Lemo Connbox - M (2)	Schoeps CCM's in 100mm Ball Gag
CB7	Stereo Lemo Connbox - S	(2) Schoeps CCM's in 140mm Ball Gag

RYCOTE

SOFTIES



The Rycote Softie is a 70mm slip-on cover with an integral synthetic fur exterior designed to provide a flexible and extremely effective solution for location audio on the go. Softies are available in a number of sizes and configurations and can be purchased with a shockmount and choice of pistol grip/stand mount or boom adapter. The Softie windshield by itself, provides a fast and cost effective solution for camera-mounted ENG mics.

- The (70mm diameter) Softie windshield is constructed of an acoustic foam base covered by a specially designed synthetic fur fabric
- Available with or without a separate shockmount with its own bracket that includes a choice of a pistol grip handle or boom adapter
- The combined Softie windshield and shockmount provides wind attenuation comparable to a standard Rycote windshield
- The Softie windshield alone provides a fast, flexible and cost effective solution for camera-mounted ENG microphones
- Available in a range of sizes to fit most popular microphones from 5cm to 32cm and a choice of three hole diameters:
  - Small (SH) for 19/20mm mics
  - Medium (MH) for 21/22mm mics
  - Large (LH) for 24/25mm mics

Softies with Mount*	Fits	Price
5cm SH Softie *(RYSM5S)*	Sennheiser 405, 406, ME20, ME40, Schoeps CMC Series, AKG CK91, CK92, CK93 and SE300B	**275.00**
5cm MH Softie *(RYSM5M)*	AKG CK1, CK63, 460, Azden SG-1X (omni), Neumann KM100, KM184, Sennheiser ME62 and 64	**275.00**
5cm LH Softie *(RYSM5L)*	Sennheiser MKH20, 40 and 60	**275.00**
	Specify Pistol Grip or Boom Adapter Mount	
12cm MH Softie *(RYSM12M)*	Audio-Technica AT 877, 4073a	**285.00**
	Specify Pistol Grip or Boom Adapter Mount	
18cm SH Softie *(RYSM18S)*	Beyer 716, 726, Sennheiser MKH 416, Sanken CS 3	**285.00**
18cm MH Softie *(RYSM18M)*	AKG C568EB, CK68, 69 Short, Beyer 836, Neumann KMR 81, Sennheiser ME66K6, ME80, MKE 300, and Sony ECM 670	**285.00**
18cm LH Softie *(RYSM18L)*	Beyer MCE86, MCE87, Sennheiser MKH 60, Sony ECM 672	**285.00**
	Specify Pistol Grip or Boom Adapter Mount	
24cm SH Softie *(RYSM24S)*	AKG CK98 SE300B	**295.00**
24cm MH Softie *(RYSM24M)*	Audio-Technica AT 835b	**295.00**
	Specify Pistol Grip or Boom Adapter Mount	
29cm MH Softie *(RYSM29M)*	Audio-Technica AT 4071a, AT 815ST, Neumann KMR 82, and the Sennheiser ME67K6	**295.00**
29cm LH Softie *(RYSM29L)*	Sennheiser MKH 70	**295.00**
	Specify Pistol Grip or Boom Adapter Mount	
32cm MH Softie *(RYSM32M)*	AKG CK69 Long, Audio-Technica AT 815b	**295.00**
	Specify Pistol Grip or Boom Adapter Mount	

**Softie also available without mount*

Softie Windjammer	Already own a Softie but are faced with adverse wind conditions? The Softie Windjammer fits right over your existing Softie to provide you with extra wind protection. (Please specify length.)



Short Hair Softie mounted on a Sony Camera

Camera Mounted Softies

Provides a fast, flexible and cost-effective solution for ENG on-camera mics.

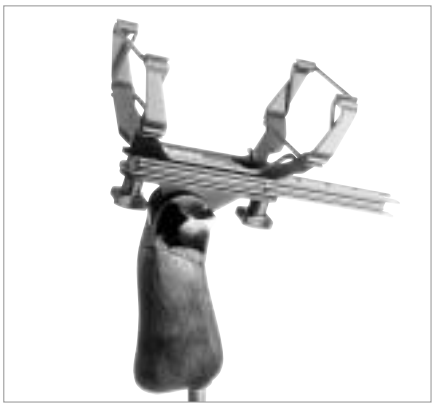
Camera Mounted Softie Models	Fits the:
10cm LH Softie Front	Sony DVW 600, 700, DNW 7, DVW 790, DNW-9WSP, DNW 90 and Canon XL-1
12cm SH Softie Front	Ikegami HL45, HLV55/MC10, HLV59 and Sony UVW 100, BVP550
12cm MH Softie Front	Audio-Technica AT 877, AT 4073a, Azden 660, Ikegami MC230, HLV79, DV7W/MC11, Panasonic AJD 610, 700, 800, Sony DSR-130WSP, DSR 300, 500, BVP 570, BVP 950, DXC-D30P, PD-150
12cm LH Softie Front	Sony BVW, BVP5/BVP 7, HD700, DVW700A
14cm MH Softie Front	AT 835ST, Sony DXC 637, and DSR-PD58

Short Hair Softies

- Available for the same range of microphones as the standard Camera Mounted Softies
- Constructed of the same acoustic, reticulated foam and snug fitting end cap as the Sofites
- Designed to avoid casting unnecessary shadows when working in camera mounted conditions
- Closely cropped dark gray synthetic fur, just 10mm high, offers a lower profile and a more compact design without severely altering the efficiency of the Softie

MOUNTS & WINDJAMMERS

Utility Mount



A universal shockmount that accepts microphones from 6″ in length and up to 1.5″ in diameter. Rugged, light-weight aluminum extrusion with two universal bands. Use alone for interiors or slide it into a Windshield for out-door use. Includes Pistol Grip

Softie Mount Accessories

Softie Mount
Boom Adapter or Pistol Grip
(Specify SH, MH, LH).....**150.00**

Pistol Grip Handle
For Softie Mount **45.00**

Boom Adapter
For Softie Mount **45.00**

Rubber
for Softie Mount
(Specify SH, MH, LH).....**48.00**



Softie Boom with Boom



Softie Mount with Grip



Softie Mount Pistol



Softie Mount only



Softie Boom Adapter

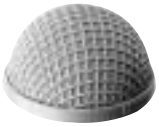
Multi Mount



A modified Softie Mount with a female 3/8″threaded brass insert suitable for stand and pole mounting. An integral accessory shoe allows for camera hot-shoe mounting.

REPLACEMENT ITEMS

Mono Windshield End Cap $40.00



Stereo Windshield End Cap.....**45.00**

Microphone Tail
To reduce handling noise. A 20″ cloth jacketed cable with three pin XLRM/XLRF connectors.....**65.00**

Mini Windjammers

Mini Windjammers are designed to improve the performance of consumer camcorders with on board microphones and are to be positioned over a foam windscreen.

Small Mini Windjammer
Up to 1.25″diameter by 1.25″ length.....**36.00**

Medium Mini Windjammer
Up to 1.50″ diameter by 2.50″ length.....**40.00**

Large Mini Windjammer
Up to 1.50″ diameter by 4.0″ length.....**44.00**

Special 60
Up to 2.0″ diameter by 2.75″ length including the Sony VX 1000/2000 cameras **50.00**

Special 90
Up to 2.0″ diameter by 3.5″ length including the EV RE50 .....**50.00**

Special 130
Up to 2.0″ diameter by 5.0″ length including the Sony PD-150.....**55.00**

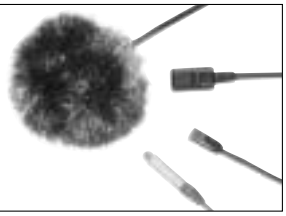
Special 155
Up to 2.0″ diameter by 6.25″ length including the Cannon XL1.....**55.00**

Special 190
Up to 2.0″ diameter by 7.50″ length including the Sennheiser MKE300 .....**60.00**

Shure VP88
For use on Shure VP88 over the foam windscreen **60.00**

Lavalier Windjammer

The Furry is designed to be used on lavaliers worn on the outside of clothing. To ensure a proper fit, it includes an internal removeable foam insert that holds many popular sized lavs and provides the air space required to make the Windjammer effective. (Sold in pairs)

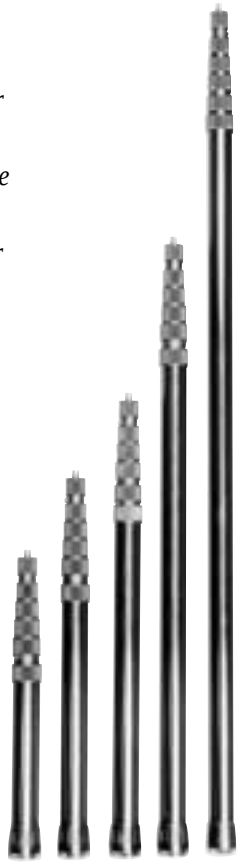


VAN DEN BERGH

CARBON FIBER BOOM POLES



VDB has been the foremost manufacturer of carbon fiber boom poles for more than a decade. Building upon the worldwide experience gained during this time, VDB set out to design the world's finest boom pole. The results speak for themselves.
VDB offers these boom poles with their unique "quick" locking collars or with "classical" one-way locking collars. The choice is yours. Both styles of locking mechanisms features VDBs creak-proof designs, making VDB the pole to own.



- The foremost manufacturer of carbon fiber boom poles for more than a decade
- Individually designed and filament wound carbon fiber tubes
- Able to pass a full size XLR connector through the pole without tools or unsoldering
- Precision machined double concentric locking collars
- Unique six section design for reduced collapsed length
- A special coating of the primary tube reduces handling noise
- Available without cable, straight cabled, or with an internal coil cable

Model	Length Closed	Length Open	# of Elements	Weight	Price
Baby	16″	5′ 4″	6	9 oz.	**Call**
Small	20″	7′ 3″	6	10 oz.	**Call**
Medium	24″	9′	6	13 oz.	**Call**
Large	31″	12′ 10″	6	16 oz.	**Call**
X-Large	44″	18′ 4″	6	20 oz.	**Call**

LIGHTWAVE AUDIO SYSTEMS

GT-10 Boom Pole

5-Section High Grade Aluminum Boom Pole Designed for Broadcast and ENG

The GT-10 is a strong 5-section aluminum boom pole designed for broadcast and all other users. The pole is robust enough for the ENG market and priced to be attractive to all users. The GT-10 provides the quality of a far more expensive pole.

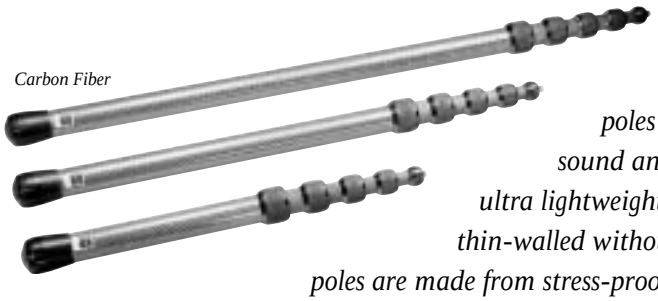


Large, strong aluminum collars allow easy tightening and loosening of pole section

- Substantial easy to grip locking mechanism with a 1.5″ long aluminum collar
- Minimum and Maximum Length: 30″ - 9′ 8″
- Weighs only 30 oz
- Industry standard 3/8″ threaded tip
- Powder coated soft-feel handle, and anodized tubing
- Poles can be supplied with or without cables – poles without still have the base attachment fitted to allow users to add their own cable without having to remove the connectors on the cable. The coil cable is high quality, with two conductors (26 GA - 19 strand tinned copper) with matte black urethane insulation and is terminated by Neutrik gold pinned connectors

GT10 without mic cable .....**155.00** **GT10C** with coiled mic cable.....**245.00**

GITZO



Microphone Fishpoles

Three carbon fiber and four aluminum microphone poles available in various lengths, designed for the location sound and broadcast audio recordist. The carbon poles use ultra lightweight, high modulus carbon fiber tubing that is very thin-walled without compromising their load capacity. The aluminum poles are made from stress-proof, high tempered aluminum alloys with black anodized finish. Telescopic extensions lock into place with a convenient twist-lock system, allowing simple and efficient setting of the required boom length. Complete with unique 1/4″ - 3/8″ reversible mounting screw, port hole for microphone cabling at both ends and soft rubber hand grip. Gitzo's commitment to fine European craftsmanship ensures smooth gliding movement with minimum bow when poles are fully extended.



Model Number	Sections	Min Length	Max length	Weight	Capacity	Price
	Carbon Fiber					
G1551 Carbon Mic Boom 142	5	17.75″	57″	1 lb	5 lbs.	**176.95**
G1552 Carbon Mic Boom 243	5	25⅝″	79″	1.25 lbs.	5 lbs.	**194.95**
G1553 Carbon Mic Boom 344	5	33.5″	135⅞″	1.6 lbs.	5 lbs.	**249.95**
	Aluminum					
G555 Loisir Fisher	2	31.25″	55″	.75 lbs.	5 lbs.	**76.95**
G556 Weekend Fisher	3	30.75″	79″	1 lb	1.0	**84.95**
G557 Sport Fisher	4	30.75″	103.75″	1.5 lbs.	1.5	**108.95**
G7680 Studex Fishpole	6	31″	150″	2.25 lbs.	5 lbs.	**169.50**

LTM – Mic Poles and Accessories

All LTM Microphone Poles Feature:

- Undentable carbon and fiber composite material
- Telescopic extension system
- Knurled friction locks
- International standard threaded stud
- Scratch-resistant black finish
- Robust and super lightweight
- Minimum bow when fully extended
- Available in four sizes from 20″ to 16′ 7″
- Size, weight and balance for the most comfortable feel



Model Number	Length	Weight	Price
MP 90A 613 *(LTMP2063)*	20″ - 63″	9.5 oz.	**348.95**
MP 90A 614 *(LTMP319)*	31″ - 9′	13.7 oz.	**369.95**
MP 90A 615 *(LTMP413)*	49″ - 13′ 8″	17.6 oz.	**469.95**
MP 90A 616 *(LTMP516)*	62″ - 16′ 7″	23 oz.	**479.95**



Shock Mount *(LTSM)* **77.95**
Microphone Suspension *(LTMS)*.....**79.95**
Windscreen *(LTWS)*.....**513.50**
Windscreen Muff *(LTWSM)* **217.50**



BoomMate
$65

Boompole Holder
$30

The BoomMate and Boompole Holder allow you to mount a boom/fish pole onto a standard mic stand, C stand or any stand with a 5/8″ thread. Ideal for hands-free control of your boom pole in the field or in the studio. The longer and more rugged BoomMate is well suited for longer boom poles where greater balance and control is required. Both the BoomHolder and BoomMate require the Matthews or Avenger Grip Head for mounting. Available in black.

WINDSCREENS



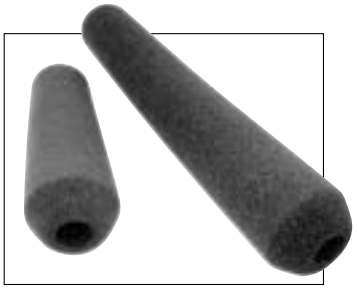
For the past 25 years WindTech has manufactured high quality microphone windscreens for the broadcast, live sound and studio markets. Available in as many as 25 brilliant colors, in a wide variety of sizes and shapes to fit most popular microphones.

- Constructed from a specially developed hypo-allergenic, open-cell acoustical foam called SonicFoam using a proprietary non-heat manufacturing process
- The acoustically transparent SonicFoam reduces wind noise, plosives and breath noise while at the same time permits unobstructed passage of sound without altering the mic's frequency response
- Adding a windscreen will also protect your microphone from dust, dampness and physical damage

STANDARD WINDSCREENS

	Inside Diameter	Inside Depth	Overall Height	Overall Diameter	Colors Available
900 Series	2.125″	1.875″	2.75″	2.375″	1 - 22
300 Series	1.375″	1.875″	2.625″	2.125″	1 - 25
600 Series	1.000″	1.750″	2.625″	2.000″	1 - 22
5700 Series	1.000″	1.750″	2.500″	1.750″	1 - 22
1200 Series	0.750″	1.625″	2.250″	1.500″	1 - 22
1300 Series	0.625″	1.250″	2.125″	1.375″	1 - 22
1400 Series	0.375″	1.250″	1.500″	0.875″	1 - 22
500 Series	0.437″	1.250″	1.500″	1.125″	1 - 22
8801 / LHTD	0.437″	0.750″	1.125″	0.900″	Black
1500 Series	0.375″	0.750″	1.125″	1.125″	1 - 22
1100 Series	0.250″	0.750″	1.100″	0.750″	1 - 22
2100 Series	0.250″	0.750″	1.187″	0.875″	1 - 22
2200 Series	0.75″	0.750″	0.75″	0.75″	1 - 22
74169 Military	0.220″	1.000″	1.430″	0.800″	Charcoal Gray
10376 High Density	0.187″	1.250″	1.500″	1.125″	Black
2300 Series (for lavs)	0.187″	0.350″	0.625″	1.625″	Blk, Off-Wht

Shotgun Windscreens



Model	SG-200	SG-60	SG-70	SG-1	SG-2	SG-3
Inside Diameter	.9867″	.9375″	.9375″	.7500″	.7500″	.7500″
Inside Depth	4.75″	6.00″	11.25″	7.250″	11.50″	20.00″
Overall Length	5.25″	7.00″	12.25″	8.375″	12.50″	21.00″
Overall Diameter	1.718″	2.375″	2.375″	2.125″	2.125″	2.125″
Price	**14.95**	**19.95**	**24.95**	**19.95**	**24.95**	**29.95**

- Prevents unwanted wind, breath and pop noises
- Made from acoustical hypo-alergenic SonicFoam
- Washable long lasting and keeps original shape
- Protects you microphone from dirt moisture and impact damage

EXTRA LARGE WINDSCREENS

Ultra Series Windscreens

Fits most 'straight' or small ball end mics.

US-1: Up to 1¼″ shaft diameter.....**29.95**
US-2: Up to 7/8″ shaft diameter **29.95**
US-3: Up to 3/4″ shaft diameter **29.95**

20 / 421 Series Windscreen

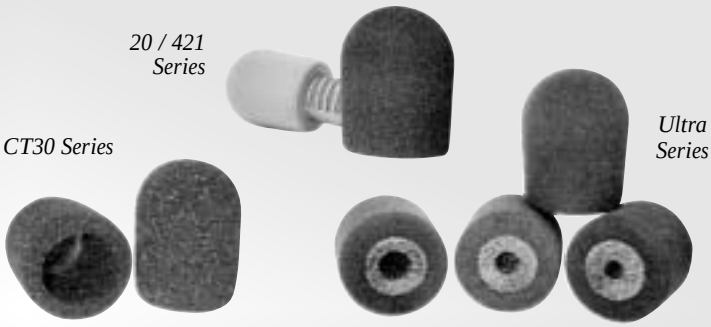
Fits most large diaphragm microphones with an outside diameter up to 2.125″ **29.95**

CT30 Series Foam Pop Filter

Allows you to use an affixed foam "Pop" filter over large studio style mics. Helps reduce diaphragm moisture as well as P-Pop sounds. Fits most large diaphragm mics with an outside diameter up to 2.5″ **29.95**

CT80 Series Windscreen

Fits most large diaphragm microphones with an outside diameter up to 2.5″ **29.95**



SPECIFICATIONS

	Inside Diameter	Inside Depth	Total Length	Overall Diameter	Colors Available
20/421 Series	1.875″	3.250″	4.375″	3.375″	1 - 19
CT-30 Series	2.375″	3.500″	4.375″	3.500″	Black
CT-80 Series	2.375″	3.500″	4.375″	3.500″	Black
Ultra Series US-1	1″	1.750″	4.375″	3.375″	Black
Ultra Series US-2	.75″	1.750″	4.375″	3.375″	Black
Ultra Series US-3	.625″	1.750″	4.375″	3.375″	Black

MICROPHONE ACCESSORIES

Rapido – Mic Stand Quick Release

A real world, and truly affordable solution to the age old problem of changing microphones from stand to stand.

- Allows you to quickly and easily install and remove mics and clips from a mic stand with a simple twist of the wrist
- Speed up and simplify the set-up and tear down between and during recording sessions and live gigs
- Eliminates maintenance issues caused by constant mounting and unmounting of mics and clips from stands– no more broken mic clips or stripped threading



- Only 1.52″ tall and weighs less than 0.5 oz.
- High quality aluminum with a durable anodized black or silver finish
- Standard 5/8″ - 27 thread
- Tops and bottoms can also be purchased separately, which allows you to install Rapido cost-effectively for all of your mic clips and mic stands – ideal if you often exchange mics with stands

SHOCK-PROOF MICROPHONE HOLDERS



- Vibration free microphone shockmount for virtually any application
- Reduces low frequency rumble, stage vibrations and handling noise
- Heavy rubber isolation ring 3/4″ x 1½″
- Snug fit with 35 lbs. of mic holding pressure
- Threaded brass insert with U.S. standard 5/8″-27 thread

SP-20 for mics 3/4″ and larger (smaller condenser type).....**9.95**
SP-25 for mics 1″ and larger (standard hand-held size).....**9.95**
SP-30 for mics 1⅛″ and larger (larger body wireless) **9.95**

MICROPHONE CLIPS

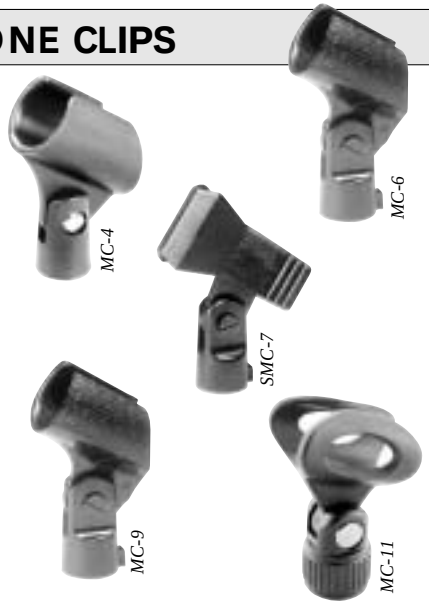
MC-4:
Extra-large, heavy duty clip for wireless microphones up to 1.75″ diameter.....**5.95**

MC-6:
An industry standard slip-in microphone clip. Fits almost all microphones.....**3.95**

SMC-7:
Spring clamp type mic clip. Holds virtually any regular or wireless mic.....**4.50**

MC-9:
High quality mic clip – uses virtually unbreakable ABS plastic.....**4.95**

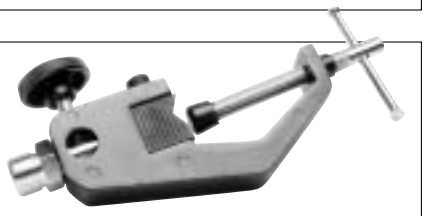
MC-11:
The same ABS material as MC-9 but for smaller mics from .80″ to 1″ diameter **4.95**



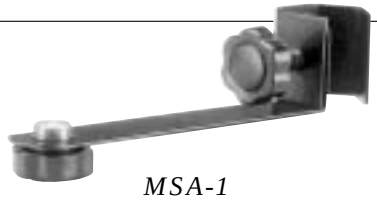
CLAMPS



MA-358
Adapts to any microphone stand to hold up to 3 separate microphones. Female brass insert with 5/8″ - 27 thread and male 5/8″ - 27 thread **11.95**



MPC-10
“C” type, multi-position clamp, fits 1.25″ diameter tube or flat surface. Male 5/8″ - 27 thread. The tread rotates 90° in three positions. Soft rubber insert prevents scratching.....**13.95**

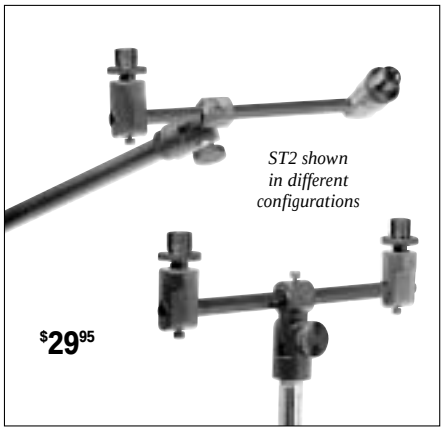


MSA-1
Add-on clamp for a second mic on the same stand. Clamps on to any standard microphone or drum stand. Male 5/8″ - 27 thread. New nylon tip.....**10.95**

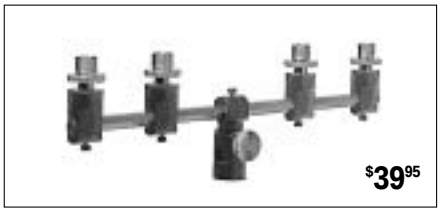
SABRA SOM

UNIVERSAL MIC PROTECTION TOOLS

ST2
Double Mic Support



- Two towers with 3/8″ and 5/8″ threaded mounts can be adjusted into any position along a 20 cm. long hexagonal black anodized aluminum bar
- Allows you to mount two mics on the same stand with variable distance and regulation between them
- A strong serrated articulation 3/8″ and 5/8″ female thread support allows 180 degrees rotation and adaptable to the majority of commercially available mic stands
- Also available with a 10cm long hexagonal black anodized aluminum bar (ST2-S)



ST4
Multiple Mic Support

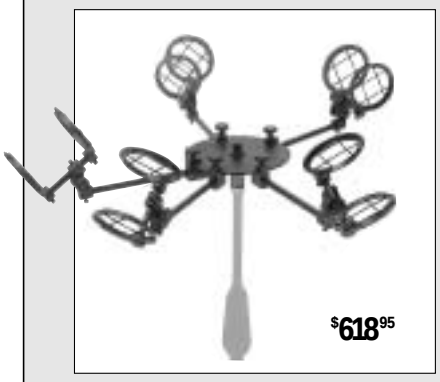
- Same as above with four mic mounting towers that can be adjusted along a 30 cm. long hexagonal, anodized black, aluminum bar
- Ideal for podium mounting – allows you to mount up to four mics on one stand, ensuring the view in front of the speaker is clean and organized

SPK
Protector Kit

- SPK is a set of tools consisting of: 1 ST2, 1 SSM-1 and 1 SPF, mounted together
- When combined with two SSM-1's, as in the SPK kit, the ST2, permits the versatile use of simultaneously forming, for example, a ZEPPELIN support or X/Y for stereo and over all configuration.



SMS-5.1
Surround Capturing Mount



- The SMS-5.1 is a revolutionary conception of natural 5.1 surround capturing
- Allows simultaneous mounting of five microphones with independent positioning over a full 360 degrees spectrum and full angular adjustment
- It consists of five hexagonal 20 cm. long arms attached to a disc that allows radial adjustments
- A SSM-1 is mounted on each of the five arms each of which can be vertically adjusted, allowing the placement of any mic

SSM-1
Universal Shockmount

- Two suspension rings allow you to isolate the mic from the mic stand thereby minimizing noise transmitted from the mic stand to the mic up to 15dB
- The distance between the two suspension rings can be adjusted along a 10 cm long hexagonal rod to accommodate microphones of various lengths
- Suitable for any tubular or conical mic shape up to 2″ diameter

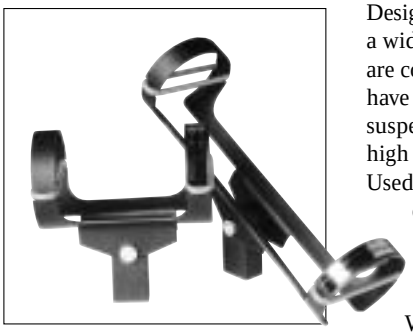




SPF
Pop Filter

- Attenuates sibilance and plosives using two parallel ortophonic rings, mounted onto a brace and fixed on an articulated arm
- Screens are easily replaceable with model PF A pair of spare screens for the SPF

Universal Shock Mounts

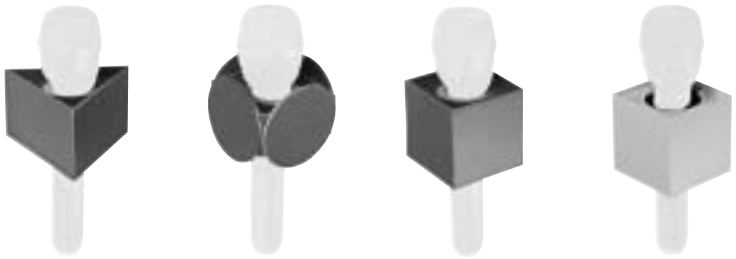


Designed to provide stable and noise-free handling for a wide variety of microphones, these shock mounts are constructed of lightweight aircraft aluminum and have a non-reflective black anodize finish. The two suspension bands are custom-made and provide a high degree of isolation from handling noise. Used by ENG, film/video and TV sound recordists, change mics simply by pulling one out and pressing in another without fear of damaging the stock suspension assembly. (Optional adapters allow installation inside Rycote Windshields). Available in 3″ and 6″ sizes, they accept mics up to 1.25″ in diameter, while 3/8″ x 16 threads allow use with any boompole.

3″ USCM .....**54.95** **6″ USCM** **59.95**

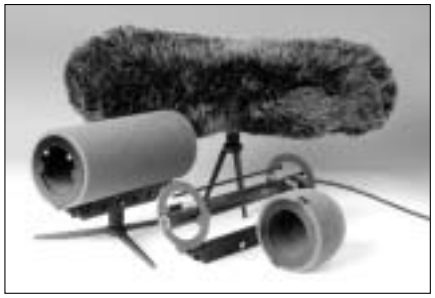
MICROPHONE FLAGS

These blank microphone flags accept your graphics. They feature high-impact news grade construction and high density foam inserts.



2″ Black Square *(GBMFSSB)* **29.95**
2.25″ Black Square *(GBMFMSB)*.....**29.95**
2.5″ Black Square *(GBMFLSB)*.....**29.95**
2x3″ Black Triangle *(GBMFSTB)* **29.95**
2.25 x 3.75″ Blk. Triangle *(GBMFMTB)* ...**29.95**
2.25″ Black Round *(GBMFMRB)*.....**29.95**
2.5″ Black Round *(GBMFLRB)* **29.95**
2″ White Square *(GBMF2W)* **29.95**
2.25″ White Square (*GBMFMSW)* **29.95**
2.5″ White Square *(GBMFLSW)* **29.95**
2.25 x 3.75″ W. Triangle *(GBMFMTW)* **29.95**
2.25″ White Round *(GBMFMRW)* **29.95**
2.5″ White Round *(GBMFLRW)* **29.95**
2.5″ Blue Square *(GBMFLSBL)*.....**29.95**
2.25 x 3.75″ Blue Triangle*(GBMFMTBL)* .**29.95**
2.5″ Blue Round *(GBMFLRBL)* **29.95**

WindBoss Windscreens



The WindBoss series offers a simple and flexible way to improve wind rumble rejection for your condenser mics. They provide the benefits of traditional, hard-shelled windshields without the associated expense and single application restrictions. The universal microphone mounting suspensions allow the WindBoss to be used with virtually any condenser mic. It uses simple silicone rubber o-rings to provide an excellent shockmount. These bands are unaffected by changes in temperature and humidity and they offer an easy and quick way of changing mics without the parts breakage associated with traditional mounts. The design of the WindBoss outer foam "shell" is made of three pieces: a center section and two end caps. These two end caps are interchangeable and allow access to both ends of the windshield. The ends caps are easily removed and replaced. The center section and its associated framework are available in three sizes providing wind protection for a multitude of microphones.

Small WindBoss
13 x 4″ (LxD), weighs 6.4 oz. **294.95**

Medium WindBoss
16 x 4″ (LxD), weighs 7.8 oz. **309.95**

CLAMP ON *Rubber-Neck*



The Rubber-Neck series is a flexible extension that brings microphones closer to their subject and enables hands-free miking. These unique smooth black goosenecks boast of a no-creak technology and super stable positioning.

R10S: 10″ Rubber-Neck with 1/2″ outside diameter, solid center. Male to female 5/8″-27 threads **14.95**

R16S (16″).....**16.95** **R20S** (20″) **17.95**

R10H (10″ with hollow center) **16.50**

R16H (16″ with hollow center) **17.95**

R12X: 12″ Rubber-Neck with 1/2″ outside diameter. Center is wired from XLR end to XLR end. XLRM to XLRF ends **46.50**

R18X (18″ version of above) **48.95**

R12XT (12″ with 3/8″ outside diameter)....**46.50**

R18XT (18″ with 3/8″ outside diameter)....**48.95**

R10HT: 10″ Thin-Line Rubber-Neck with 3/8″ outside diameter, hollow center. Male to female 5/8″-27 threads **16.95**

R16HT (16″ version of above).....**18.95**

R10ST (10″ with solid center) **16.50**

R16ST (16″) .....17.95 **R20ST** (20″).....**18.95**

DXA-4 Series

Dual XLR Adapter for DV Camcorders

The DXA-4 offers an easy, reliable way to hook up professional audio gear to your DV camcorder — and with it, all the benefits of using balanced audio for superb sound every time. Record from two sources at the same time. Each input is recorded on a separate audio track, making it possible for you to mix both channels during editing. You can even feed a time code signal into one input and audio into the other.



- Two channels let you record two sources on separate tracks for post audio mixing. Both channels have their own mic/line level switch and trim potentiometers for flexibility and complete control
- Securely mounts to underside of camcorder, eliminating cable strain
- Also has a threaded hole for mounting on a tripod plate
- Rugged all metal case is built to last and provides complete shielding from noise. Made with Neutrik XLR connectors and transformers for dependability you can count on

DXA-4 (for VX-2000) .....**169.95** **DXA-4S** (for TRV-Series) ..**169.95** **DXA-4C** (for XL-1).....**169.95** **DXA-4P** (for AG-DVC10) .**169.95**



$249.95

DXA-6

Audio Adapter with Phantom Power

All the great features of the DXA-4 adapters with the addition of 48 volt phantom power.

- Two balanced XLR mic/line switchable inputs for stereo recording with independent drift free trim controls that ensure reliable operation
- Switchable dual 48 Volt Phantom Power
- An auxiliary unbalanced 1/8″ mini jack input is available on the right channel to accommodate wireless mics
- MONO/STEREO switch allows recording both inputs to a single channel
- Low Noise Transformers for Superb Sound
- Runs on one standard 9 Volt alkaline battery
- LED Battery Power Indicator
- Stereo right angle unbalanced 1/8″ mini-plug output connector attached to a 10″ cable
- Housed in a rugged aluminum case with a flat black finish
- Measures 5.75 x 3.25 x 1.25″ and weighs 12 oz.

SVU-2

Stereo VU Meter for Camcorders



$134.95

- Multi-purpose stereo VU meter for camcorders and other recording devices
- Bright ten-segment display to easily see under virtually any lighting condition
- Calibrated display eliminates guesswork to set ideal recording levels
- Built-in high powered headphone amplifier with volume control
- Mounts to camera tripod socket or can be carried via belt clip
- Operates on one standard 9 volt battery
- LED battery power indicator

XLR-PRO

Passive Dual XLR Adapter / Mini Mixer

The XLR-Pro is a Dual XLR adapter that lets you feed professional audio to any camcorder, DAT or MiniDisc recorder that provides an audio 1/8″ input jack. This allows you to use professional mics, tap into PA systems and mixers, and use long run, balanced audio cables – all with complete confidence. Special circuitry ensures that the XLR-Pro will work with any camera.





CONVERT YOUR CAMCORDER OR PROSUMER DIGITAL AUDIO RECORDER TO ACCEPT PROFESSIONAL XLR MICS

- Each of the two inputs input offers a choice of XLR and 1/8″ mic/line switchable inputs with volume controls – Hook up two XLR mics, or an xlr mic plus wireless mic, or any other combination of pro and prosumer mics
- Connects to your camera/recorder via an 1/8″ stereo mini plug with a 13″ cable
- A stereo/mono output switch allows you to record from one mic to both channels
- Securely mounts under your camera, for either hand held or tripod operation, or can be easily clipped onto your belt
- Enclosed in a black solid aluminum housing that protects the entire electrical circuit from soundtrack destroying interference
- High performance multi-shielded transformers ensure extremely quiet operation
- State-of-the-art latching Neutrik XLR connectors
- A unique ground switch (not to be confused with a "ground lift" switch) lets you select the quietest ground for any camera
- Voltage blocking circuit permits operation when connecting to plug-in power mic jacks found on some camcorders (Sony VX1000, VX2000, VX700, TRV900 & others)
- 20 - 20k Hz frequency response (±1dB max)
- Measures 5.25 x 2.75 x 1.25″ and weighs only 13.5 oz.



XLR-JR

Single Channel XLR Adapter

Same as XLR-Pro but with only one XLR input, one 1/8″ input and one level control.



VU150

Audio Level Meter

The VU150 audio meter helps you maintain proper audio levels being recorded to your video camcorder.



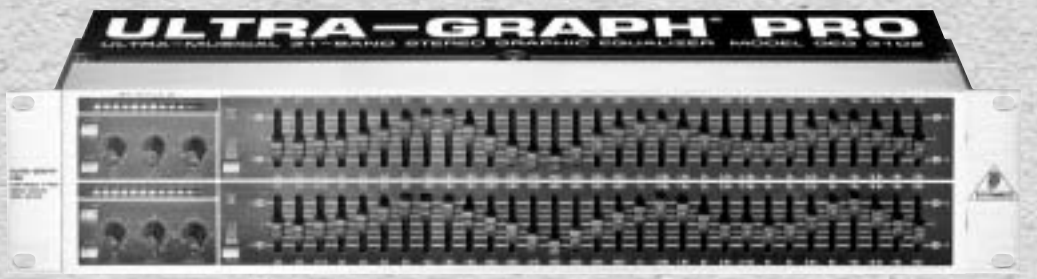
- 1.5″ wide, 10-segment left and right LED meters measure audio levels between -20dB to +3dB
- The large display is switchable between two modes:
  - **Bar mode** displays intensity by a continuous line of LEDs
  - **Dot mode** displays intensity using a single LED at the given intensity level. Dot mode requires less power than bar mode enabling longer battery life
- Works on any camera with a standard 1/8″ stereo headphone output
- Mounts easily and securely on your camera's accessory shoe
- Adjustable brightness for full visibility in daylight or turn down to conserve battery indoors
- Left and Right audio calibration adjustments
- Built in headphone output with a volume control adjustable from -40dB to +20dB
- Runs continuously for approximately 100 hours using a 9V alkaline battery
- Low battery indicator
- Measures 3.5 x 2.5 x 1.5″ and weighs 5.5oz. with battery
- 1-year warranty


FAT MAN


JOEMEEK




PreSonus


ULTRA-GRAPH PRO

B&H

PHOTO - VIDEO - PRO AUDIO

The ProfessionalAudio SourceBook

Section 2

Processors

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INEKO



Stereo Effects Processor

Ineko is a unique table-top, stereo effects processor that doesn't require an engineering degree to operate, or a fat recording contract to afford. A powerful, professional-quality tool designed for music recording, DJ and dance applications, it is exceptionally user friendly with a very intuitive, self-explanatory interface. No complicated menus or difficult to use functions. Just three simple knobs to control 48 dramatic, user-adjustable effects, including reverbs, filters and delays. So simple, even novices can master it right out of the box. With its metallic gun-gray, industrial design and over-the-top sounds, Ineko redefines effects processing by providing top-quality, user-friendly, real-time effects at an unbeatable price.

- Designed for music recording, DJ and dance applications, as well as recording and live performance, Ineko gives the user the power to control real-time effects via a simple intuitive user interface.
- The uncomplicated faceplate sports a 6 x 8 LED grid/program display which shows the 48 high-quality reverbs and delays, traditional effects such as chorus, vibrato, flanger, and phasers.
- Also incorporates an arsenal of unique effects, from sub-harmonic synthesizer, fuzz, vibrowobbel, and formant-filter, to frequency-shifter, band-limiter, decimator and vocoder.
- Three large dedicated knobs provide direct real-time control over effects parameters – the three available parameters for each effect are written underneath the effect name on the front panel.
- Effects programs are grouped by category and are easily selected along the matrix using the simple down and across buttons.
- Bypass button to engage or disengage signal effects during use.
- 24-bit converters
- 1/4″ stereo inputs and outputs
- Input trim control with a signal present LED indicator

AKIRA

Programmable Stereo Multi-Effects Processor



The Akira is a multi-effects processor that is Ideal for studio, live performance, and sound reinforcement applications. Affordably priced for entry- and mid-level recording enthusiast and musician, the Akira has sounds that set it apart from anything else on the market. In addition, its X, Y, and Z parameter knobs allow real-time, playable control over program parameters, making Akira much more than a "set-and-forget" rack mount signal processor.
Akira features 24-bit A-D and D-A conversion, stereo analog I/O on balanced TRS 1/4″ jacks, and an internal switching power supply that works on 90-230v AC. There's also MIDI I/O for program changes and continuous controller transmit and receive, for additional real-time control over the effects. Housed in a 19″ rack-mount enclosure (1RU) high, Akira has an easy-to-understand interface, making it ideal for its intended audience.

- Two channel, 24-bit A/D-D/A, 28-bit internal processing, 48kHz sample rate
- 100 programs - reverbs, delays, pitch mod, filters, and special effects
- MIDI I/O, balanced 1/4″ TRS analog I/O, +4dBu/-10dBV

AIRFX



Motion-Controlled Effects Processor

Connect the airFX to any line-level sound source and speaker system. Then move your hand, or any other body part through the invisible 3D sphere formed by Alesis' patented Axyz technology. You can move any direction; up or down, right or left, or in or out, to twist any sound in an infinite number of ways. The slightest movement of your hand tweaks the tones of of the AirFX and turns your tunes inside out.

Modify the sound of any audio from CDs, LPs, tape, computer - even live instruments and vocals with 50 high-quality, stunning filters and effects from vinyl to vocoder, flanging, and virtually anything and everything in-between.

airFX and airSynth both feature:

- Incorporated into the airFX and airSynth space-age domes are four infrared sensors that generate a halo of light over the instrument. Known as the Axyz (pronounced ax-is) system, this revolutionary technology gives you three planes of control over internal effects:
  X (left to right), Y (front to back) and Z (up and down). Other effect controllers provide only X/Y coverage.
- High-quality sounds and effects:

—airFX: 50 filters and effects, including percussive sounds

— airSynth: 50 synthesized sounds and sound effects, including percussive and legato noises.

- They feature a frequency range of 20Hz-20kHz, 24-bit converters and ship with a 9-volt AC power supply.
- A threaded socket on their base allows you to mount them on a mic stand for ease-of-use in club and studio applications.
- Incorporating the same futuristic design airSynth can be integrated with airFX in a singular system via stereo RCA inputs and outputs without using an external mixer.
- Their simple user interface consists of a single knob, which is used to select, engage, bypass, freeze and release a given sound, and an LED that indicates the current program.

AIRSYNTH

Motion-Controlled Effects Processor and Sound Generator



airSynth produces dramatic synthesized sounds and sound effects when you pass your hand over the invisible 3D sphere above it. Alesis' patented Axyz technology. lets you control up to five sound variables simultaneously by moving your hand left to right, forward and backward, or up and down over the product, making airSynth easy and intuitive to use. The dramatic sound set and unique user interface make airSynth ideal for use in live performances, DJ applications, music postproduction, and on-air broadcast applications.

And while it employs the same technology and simple user interface, unlike the effects processing airFX, the airSynth is a true sound-generating product that features a broad range of sound effects and synthesized noises providing an altogether different and complimentary range of options. The unique airSynth sound-set includes 50 staccato, percussive, legato continuous pads, drum sounds and sounds that emulate things in nature.

MODFX LINE

14 Unique Performance Effect Boxes

The ModFX line is a family of performance effects boxes that includes 14 individual units, each one providing a different set of sound effects and signal processing. Of particular interest to musicians, DJs, and recordists is the extremely versatile modulation section on several of the boxes, providing unprecedented control over the sound of the effects, including the ability to sync modulation to incoming music tempo. Every box in the ModFX line digitally processes analog stereo signals and maintains a uniform, friendly, uncomplicated user interface. Keyboardists, guitarists, and any other studio or live performance artists will find the selection and variety of the ModFX line well suited to a wide variety of musical applications.

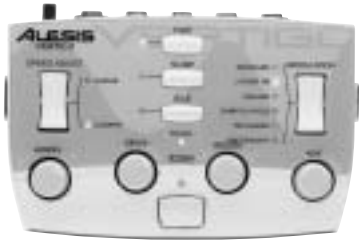
Each box- designed with internal 28-bit digital processing - allows you to connect to multiple units within the family via the link ports (in and out) located on either side of each box. These links allow the units to pass digital audio, word clock and power between ModFX line family members. Analog I/O is provided via four separate 1/4″ unbalanced connectors, and each unit comes with an AC power supply. A fifth 1/4″ connector allows an external footswitch to control the Bypass function. Each unit is sized to fit a 1/3 rack space; an optional rack mount adapter is available.

Fidelity X



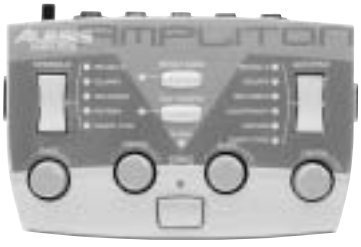
A "communications simulator" that simulates tape, vinyl, radio, megaphone, telephone, and digital. Features include the 'Year' parameter that defines the "age" or "vintage"; a 'Mono' button that combines the input stereo signal into a mono signal and feeds it through the processor; a 'Quality' knob that controls the fidelity; a 'Drive' knob that adjusts the amount of distortion; the 'Tone' knob which filters the output providing a bass or treble boost; and the 'Mix' knob that adjusts the amount of "wet" (affected) signal compared to the "dry" (unaffected) signal.

Vertigo



This world-class Leslie simulator re-creates the vintage sound of the original and can produce great new variations on the classic sound. Vertigo sports Fast, Slow, and Idle buttons to set the speed of the rotating horn simulation. The 'Mix' knob will adjust the amount of "wet"signal compared to the "dry" signal.

Ampliton



This unit combines a tremolo with an autopan. The low frequency oscillator (LFO) of both the tremolo and the autopan can be independently set according to your sound desires. The LFO can be set to triangle, square, sample and hold, triggered, and "uncertainty" waveforms. Each LFO can be beat synchronized to the music, and a 'Beat Tap' button enables the user to adjust the tempo.

Bitrman



Here's an extraordinary effects box comprised of compression, distortion, dual phasing and a fourth selected effect that can either be a comb, decimator, bit reducer, frequency modulator, ring modulator, or frequency shift. The four effects can be configured multiple ways by adjusting the 'Configure' parameter.

Phlngr



Here's a flanger box with an amazing modulation section. All of the modulation waveforms can be beat synchronized to the input music. Some of the features include the 'Type' switch that describes the type of the flanging effect; the 'Beat Tap' switch that enables the user to control the tempo used in the beat synchronizer; the 'Center' knob that controls the range of the frequencies affected; the 'Rate' knob that controls the speed of the oscillations; and the 'Depth' knob, controlling the amount of modulation.

Nastify



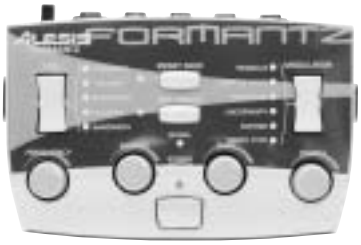
Like its name implies, this is a distortion effect box with attitude and edge. Six different styles of distortion can be selected: Warmth, Fuzz, Distortion, Thick, Chunk, and Stellar. Knobs control drive, tone, and the amount of output gain.

B&H PHOTO - VIDEO - PRO AUDIO

ONE HOUR FREE PARKING
AT 349 W. 34th STREET (with purchase of $100 or more)

ModFX

Formantz



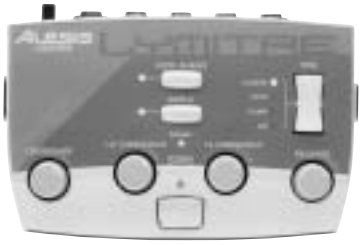
Formants are filters that emulate the sounds of speech; Formantz is an advanced filter modeling box that will leave you speechless. The filter can be modulated by either triangle, sample and hold, triggered, "uncertainty", or "pattern" waveforms, and all of the modulation waveforms can be beat synchronized to the input music. The 'Frequency' knob controls the center frequency of the filter; The 'Modify' knob varies the selected 'Type'.

Faze



This is a phasor effect box loaded with features and controls including the 'Type' of phasor that is selected from options including Deep Mono, Dual Mono, Contrary Stereo, Asynchronous Stereo, and Liquid Metal. Five different modulation waveforms can be selected, each of which can be beat synchronized with the aid of the 'Beat Tap' button.

Lymitre



A remarkable two-band limiter with great controls including the 'Look Ahead' switch, enabling a look-ahead peak limiter; the 'sizzle' switch that adds post-limiting brightness; and the 'Crossover' knob that enables the user to set the frequency that separates the two bands.

Metavox



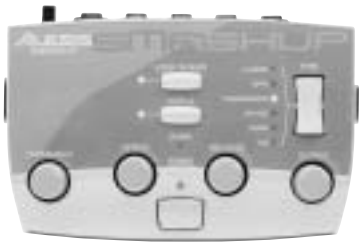
This unit is a vocoder where the user can choose four different types of internal synthesizers (Saw, Rectified Saw, Square, And Noise). The synthesizer is modulated by one of five sources (triangle, sample and hold, triggered, "uncertainty", and tracking) that can all be beat synchronized with the aid of the 'Beat Tap' button. There's also a control that allows for setting the frequency of the internal synthesizer. Another great feature of Metavox is that it also allows an external synthesis source to be input and voice-modulated.

Philtre



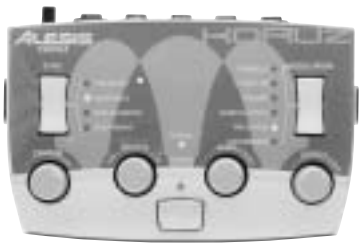
Meet the ModFX line's resonant filter. You can select either Low Pass, High Pass, Band Pass, Low Pass Alias or any combination as filter types. Five different modulation waveforms can be selected, each of which can be beat synchronized. The 'Reset Mod' button will reset the phase of the LFO; the 'Steep' button switches between 4- and 6-pole filters.

Smashup



Yes, it's a compressor. The user can select from five different type of compression: Classic, Opto, De-Ess, Pump, and Fat. The 'Threshold' controls the threshold level; The 'Attack' and 'Release' control the attack and release values for the compressor. The 'Output' parameter controls the output gain.

Koruz



What's a family of performance effect products without a chorus effect box? The chorus effect can be modulated by triangle, hyper-triangle, square, sample and hold, "uncertainty", or triggered LFO shapes. The 'Depth' knob controls the amount of modulation.

Spectron



This unit adds bass and treble harmonics to a signal. The user can define the frequency and volume of the harmonic generator for the bass and treble synthesizers. Controls can also eliminate the original signal from the output and generate only the synthesized harmonics. The 'Bass Limiter' provides the user with a means to limit the strength of the bass synthesizer.

MICROVERB 4

Multi-Effects Processor



An easy-to-use, programmable reverb and effects unit, the MicroVerb4 provides a wide variety of great-sounding, easy-to-use effects for small studios, engineers and performing musicians. The key to MicroVerb 4's power is in its simplicity. It offers 200 programs that include everything from high-quality reverb, delay, chorus and flange to exciting multi-effects and more. To customize your programs, two front panel knobs allow for quick, easy, user-storable edits, or use its MIDI inputs for control over program changes and modulation. You can then save your edited programs in the 100-space user program bank.

FEATURES

- 200 programs (100 preset, 100 user-storable) of Reverbs (Hall, Room, Plate), Chorus/Flange, Delay (Mono, Stereo, Ping Pong and Multi-Tap) and Pitch Shift as well as multi-effect and dual mono send algorithms.
- Each Program has two parameters which can be adjusted via the dedicated Edit A and Edit B controls. Depending on the type of Program selected, these might alter Reverb Decay, Chorus Depth, etc.

PROGRAMS	EFFECT	EDIT A	EDIT B
01 - 10	HALL	DECAY	ANSW DLY
11 - 19	ROOM	DECAY	ANSW DLY
21 - 29	PLATE	DECAY	ANSW DLY
31 - 40	CHORUS/FLANGE	RATE	DEPTH
41 - 50	DELAY	TIME	FEEDBACK
61 - 69	PITCH SHIFT	SEMI-SHIFT L	FINE-SHIFT R
70 - 80	MULTI-EFFECT	DECAY	RATE/TIME
81 - 90	DUAL SEND	DECAY	RATE/TIME

The Parameter Chart shows you what effect type is assigned to various groups of Programs – it also shows the function of the Edit A and Edit B controls

Front Panel

- Dedicated input and output level controls
- Mix control adjusts the balance between the dry and wet (effected) signals
- Dual 4-segment LED input level meters
- The multi-purpose LED display typically shows the currently selected Program but will also display the MIDI channel or parameter values
- The rotary Value Encoder is used to select Programs and to change the MIDI channel
- Bank/MIDI Button lets you quickly change between the Factory and User program banks as well as select a MIDI channel

Inputs and Outputs

- Left and right unbalanced 1/4″ inputs and outputs accommodate four different configurations: Mono In/Out, Mono In/Stereo Out, Stereo In/Out and Dual Mono In/Out
- MIDI In and Out/Thru connectors allow you to perform several MIDI functions including Program Changes, send/receive Sys-Ex data and realtime control of up to two parameters simultaneously
- 1/4″ stereo phone jack lets you connect one or two momentary footswitches – a single footswitch functions as an effect bypass. Add a second footswitch via insert cable to use as a control switch for tap tempo, or to control Leslie speed (slow/fast) – if the control footswitch is held down and audio is played into the inputs, these impulses will be used to set the delay time.

NANOVERB

Preset Digital Reverb and Effects Processor



The NanoVerb offers 16 of Alesis' best preset effects algorithms in a compact, easy-to-use and incredibly affordable package. You'll enjoy its lush hall, plate and room reverbs, rich true stereo chorus, flange, delay, rotary speaker emulation and useful multi-effects.

- 18-bit A/D and D/A converters with a 20-bit internal processor ensure the sonic quality that allows you to use it for any effects application from guitar rigs to studio recording.
- Power/input and dual-color signal input/clip LEDs
- Front panel adjust knob allows you to tweak NanoVerb's great-sounding programs until they're just right for your music, while the input.output levels and Mix controls ensure proper effect levels for a wide variety of applications.
- Stereo 1/4″ inputs and outputs
- Ultra-compact 1/3 rack-space package

MIDIVERB 4

Fully Programmable Multi-effects for Live and Studio



The MidiVerb 4 is the perfect effects processor for professional project studios and musicians who require an affordable solution for fully-programmable, high-fidelity effects. Its excellent effects algorithms produce dense, natural reverb, rich chorus, flange, delay, pitch effects and up to three simultaneous multi-effects. MidiVerb 4's thirty-two effects algorithms give you a complete arsenal of fully programmable effects, many using true stereo parallel processing. You can take advantage of the 256 powerful programs with 128 spaces for user-created effects. With its easy-to-use graphic interface, programming your own effects is a snap. And by using MidiVerb 4's MIDI controller routing and assignable footswitch input, you'll have complete creative control over every aspect of your sound.

FEATURES

Inputs & Outputs

- Left and right unbalanced 1/4″ inputs/outputs with 18-bit A/D and D/A converters and 24-bit internal processing for a 20Hz-20kHz bandwidth and 90dB dynamic range, making it ideal for digital recording.

Effects

- 256 programs (128 Preset/128 User-storable) include single stereo, multi-effect and dual mono effects configurations based on 32 different algorithms
- Reverb: Concert Hall, Real Room, Ambience, Plate, Nonlinear
- Delay (1299 ms of total delay memory): Mono Delay, Stereo Delay, Ping Pong Delay, Multi Tap Delay and a BPM Mono Delay which responds to MIDI clock
- Pitch: Stereo Chorus, Quad Chorus, Stereo Flange, Stereo Pitch Shifter
- Special: Auto Pan
- Double: Real Room+Delay, Real Room+Chorus, Real Room+Flange
- Dual Mono: Delay/Delay, Chorus/Chorus, Flange/Flange, Pitch/Pitch, Chorus/Delay, Flange/Delay, Pitch/Delay
- Multi Chain: Delay->Real Room, Chorus->Real Room, Flange->Real Room, Lezlie->Room, Real Room->Flange, Chorus->Delay->Room, Flange->Delay->Room
- Display is divided into 5 sections: Program Number, Bank (Preset, User), Program Name (up to 8 characters), Configuration (Stereo or Dual effect), Level Meters (Peak-style inputs meters).

Editing

- Access up to four edit pages per preset – the currently selected page will be indicated in the display by a box around its number
- Each edit page contains up to four parameters accessed via the A, B, C and D buttons. Currently selected parameter can be adjusted by the value knob.

External Control Functions

- MIDI controller routing and an assignable 1/4″ footswitch input for flexible control on stage
- Up to two effect parameters, pre-configured in each preset, can also be modulated via MIDI – the modulation range for each parameter can be set

Conveniences

- Automatic Input Level Sensing– simply press and hold the Input and Output buttons simultaneously for five seconds or longer and the Auto Input function will automatically optimize the input level
- Once you have accessed Edit mode, if you hold any of the parameter buttons (A, B, C or D) for more than one second, the upper display will provide a detailed description of the selected parameter

PICOVERB

Ultra-affordable Digital Multi-Effects Unit



The PicoVerb is great sounding reverb and effects unit designed for the budget-conscious musician or recording enthusiast. The 16 multi-effects presets include chorus, flange, delay, chorus/reverb, rotary speaker simulator, and more. Super-compact, expertly designed and superbly priced.

- Perfect for studio and sound reinforcement applications
- Stereo, 24-bit A/D-D/A, 28-bit internal processing, 48kHz sample rate
- 16 Presets including Halls, Rooms, Plates, Chorus, Flange, Delay, Lezlie
- Unbalanced 1/4″ analog I/O, -10dBV
- 1/4″ rack wide

CLX-440

Compressor/Limiter/Expander



The CLX-440 is a full-featured compressor/limiter/expander that combines separate compression and expansion sections for each stereo channel, with "look-ahead" capability, side-chain and "key" functions, continuously-variable soft/hard knee and peak/RMS controls. Two sets of front-panel meters provide confirmation of input, output, compression and expansion activity. Special algorithms provide a broad range of dynamics options—from gentle to aggressive—to meet the requirements of live sound and studio production. CLX-440 also features a side chain I/O that allows you to insert an external EQ such as the PEQ-450, into the dynamic side chain. A special "look-ahead" function allows dynamic processing to start before the signal is present, by delaying the audio path by approximately one millisecond.

- Dual stereo architecture allows processing of 2 mono or 2 stereo signals, up to 4 channels of signal processing
- Sidechain and key inputs and outputs
- Separate compression and expansion sections for each stereo channel
- Continuously-variable soft/hard knee and peak/RMS controls
- Two sets of front-panel meters provide confirmation of input, output, compression and expansion activity

400 SERIES PROCESSORS

Designed for use in studio, live-sound, installation and broadcast applications, the 400 Series signal processing line features a striking industrial design and unique "dual-stereo" processing to provide highly flexible and very powerful sound shaping capabilities for the audio professional. Rather than relying on complex menus and complicated interfaces, each 400 series product is designed to perform a dedicated function extremely well, using the highest quality components and familiar, analog-style controls. Additionally, they feature four-channel "dual-stereo" processing to give users more channels of processing at very affordable prices.

Hidden beneath their aluminum-extruded chassis is a tremendous amount of processing horsepower and a unique routing scheme. Unlike standard 2-channel processing products which traditionally "link" both channels for a simulated stereo operation, the CLX-440 allows two true stereo programs to be processed simultaneously, and the PEQ-450 allows dual-stereo 5-band or "chained" 10-band dual-stereo operation. Dual-stereo capability is very useful in studio and live sound applications where multiple channels of audio need to be processed simultaneously.

PEQ-450 *Dual Stereo Parametric Equalizer*



The PEQ-450 is a dual-stereo 5-band, fully parametric equalizer with 20Hz to 20kHz EQ control, +/- 18dB boost/cut controls, continuously-variable Q, high and low shelving, high- and low-pass filters and input/output metering. The PEQ-450 also includes qualities offered in more expensive parametric equalizers such as high quality pots and switches, separate frequency gain and Q controls for each band, plus high-pass and low-pass filters. The unique "chain" mode allows all 10 bands of EQ to be used in tandem to process two stereo signals simultaneously. An internal power supply and balanced 1/4˝ TRS input and output jacks complement the impressive features of both two-rack products.

- Dual 5-band parametric equalizer with frequency, gain, and Q for each band plus high- and low-pass filters and high and low shelving.
- Up to four channel processing: dual mono, dual stereo or"CHAIN" mode allows 10 bands of EQ to be applied to a stereo signal
- Two sets of dedicated controls for all EQ functions
- Front-panel meters indicated input and output activity

3630

RMS/Peak Dual Channel Compressor/Limiter with Gate



The 3630 is an affordable single rack space dynamics processor that provides two independent full-featured channels of compression, limiting and gating, suitable for any application from subtle gain control to in-your-face punch. The compression section features variable Ratio, Threshold, Attack and Release controls as well as switchable RMS and Peak detection circuitry and Hard and Soft knee compression curves. The two fully independent gates feature variable threshold and rate controls to easily silence ambient noise. You can use the 3630 as an independent dual-channel compressor, with separate controls for each channel or push the Link button to strap the left and right channels together for stereo processing. Also features 1/4″ inputs and outputs, precise signal metering and a sidechain insert for keying or ducking effects.

FEATURES

- Two discrete channels with identical controls and I/O provide stereo or dual mono operation – you can use each of the two channels independently, or you can strap both channels together, using the Link button, for stereo operation with both channels being controlled by channel A

Compression

- Each channel features a Threshold control variable from -40dBu to +20dBu and a Ratio control variable from 1:1 to ∞:1 (limiting)
- Selectable Peak or RMS response lets you decide whether the sidechain circuit reacts to signal peaks and potential clipping or average signal levels
- The selectable Knee button provides a choice of the more severe “Hard knee” curve in which compression only kicks in when the input level reaches the threshold or a gentler “Soft knee” curve in which the compression ratio gradually increases up until the input level reaches the threshold
- Bypass switches make it easy to compare the processed and unprocessed signals
- When using the Peak response setting, you are able to adjust the Attack Time from 0.1ms - 200ms and the Release Time from 50ms - 3S – when the RMS response setting is used, attack and release times are program dependent

Noise Gate

- The Threshold control (variable from continuously open to 10dBu) determines the level that a signal needs to exceed before the gate will open
- The Rate control, variable from 20ms - 2S, determines how long it takes for the gate to fade smoothly from the open to closed setting
- A pair of LEDs indicate whether the gate is open or closed

Output and Metering

- An Output level control variable ±20dB is available for making up gain that is lost due to compression
- 12-segment LEDs are provided for gain reduction metering (-1 to -30 dB range)
- 12-segment LED meters are switchable to show either input or output levels (-30 to +6 dB range)

Rear Panel

- 1/4″ unbalanced inputs and outputs are switchable for +4 dBu or -10dBV operation
- 1/4″ TRS side chain I/O enable you to:

  Insert EQs for frequency-selective dynamics control such as de-essing or,

  Key (trigger) dynamics control from an external audio source i.e., using a voice-over track to lower the volume of background music (known as ducking)

NANOCOMPRESSOR

Stereo Compression Unit In A One-third Rackspace Package

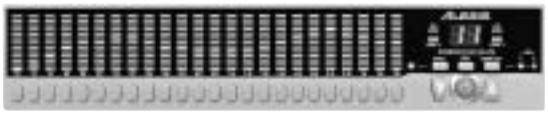


- A single set of controls provides full-featured dynamics processing of left and right channels simultaneously
- Front-panel controls include Ratio, Threshold, Attack, Release and Output Level as well as a Bypass button
- Selectable Peak and RMS compression input detection as well as hard and soft knee compression curves
- Indicators include a 6-segment switchable input/output meter and a 6-segment gain reduction meter
- Rear panel connections include unbalanced 1/4″ stereo inputs and outputs as well as a 1/4″ TRS sidechain insert that allows you to interface with other devices for ducking and de-essing

DEQ-224/DEQ-230

Programmable Dual 1/3-Octave Graphic Digital EQs





Unlike so many EQs on the market today, the DEQ-230 has a friendly, intuitive nature, making it one of the easiest to use, manipulate and control. With a sleek industrial design including over 800 individual LEDs to give a richly detailed view of EQ curves, the DEQ-230 is a clear standout in the world of graphic equalization. Plus its feature set is well suited for professional studio applications, while its very affordable price also makes it attractive to the recording hobbyist or performing musician. The DEQ-224 delivers the same digital performance and quality as the DEQ-230 in a highly compact package. The DEQ224's half-rack form factor and powerful EQ capabilities make it ideal for professional and home studios with limited rack space, as well as the perfect addition to any gig bag.

DEQ-230

Taking advantage of its internal digital signal processing and unique user interface, the DEQ-230 has the ability to store and recall 25 preset and 25 user-defined curves, as well as a single bypass button that affects both channels simultaneously. Sized to fit a single rack mount chassis, it includes 1⁄4″ unbalanced analog I/O and an external AC power supply.

DEQ224

Easy-to-use and understand interface, with 30 preset EQ and 30 user-recallable EQ curves. Two independent channels of EQ: 24-bands, ±12dB resolution. 13-LED resolution of cut/boost per frequency band. Individual Signal and Clip indicators per channel. Analog I/O using 1/4″ balanced TRS connectors. Bypass and Channel A-B link controls.

DEQ-230D/DEQ830

Programmable Dual 1/3-Octave Graphic Digital EQs





Based on the extraordinary technology of the popular DEQ-230, these two programmable rackmount EQs give recording engineers and live sound professionals an expanded feature set, enhanced 24-bit sound quality, and more flexible routing options than any other EQ in their class—at breakthrough prices.

DEQ830

The DEQ-830 boasts eight independent programmable 1/3-octave graphic equalizers in an amazingly compact 1U rackmount design. Connections include eight balanced 1/4″ TRS analog I/O and eight channels of ADAT Optical digital I/O, making it perfect for a range of modern recording applications. It also features 100 user-customizable EQ programs, 50 template "curves" for rapid setup, and +4dBu/-10dBV switchable operation—allowing a high degree of user customization. Add an ultra-intuitive user interface and MIDI I/O, and the DEQ-830 is everything a high-end programmable EQ should be, without the price tag.

DEQ230D

For those who don't need eight channels of processing, the DEQ-230D is a dual-channel 1/3-octave programmable EQ with expanded I/O and more features than the acclaimed DEQ230, including two channels of fully balanced 1/4″ TRS analog I/O, switchable +4dBu/-10dBV operation, S/PDIF input and MIDI I/O for use in a range of stage and studio applications. Additionally, 30 Preset and 30 User programs and an extremely intuitive user interface provide for fast sonic tweaks and various customized settings

B&H PHOTO - VIDEO - PRO AUDIO EQUIPMENT LEASING AVAILABLE

ATR-1a

Auto-Tune Intonation Processor



Back in 1998, Antares introduced the ATR-1 and made the unlikely claim of "perfect pitch in a box" a solid reality. It corrected the pitch of vocals or solo instruments, in real time, without distortion or artifacts, while preserving all of the expressive nuance of the original performance. Since then, thousands of ATR-1s have found their way into touring racks, live performance rigs, and recording studios of artists and producers like Cher, Everclear, Al Schmitt and many, many more. Preserving the great sound quality and ease of use of the ATR-1, the ATR-1a adds a Bass Mode (for correcting the pitch of low bass range instruments) and the innovative Make Scale From MIDI function which allows the ATR-1a to automatically create a custom target scale for any melody.

FEATURES

The Right Note

A key requirement for effective pitch correction is the ability to accurately specify the right notes to correct to. The ATR-1a gives you a variety of choices to ensure that its output is always what you want. The basic tool for target pitch identification is the Scale:

- The scale is a list of the notes that the input notes are corrected to
- Factory Programs are provided for chromatic and standard diatonic scales as well as the ability to use the Scale screen to program custom scales
- Individual notes can be bypassed. When the input pitch is near a bypassed note, the input is passed through with no correction.
- For more flexibility, the MIDI Note Mode allows you to specify scales in real time via a MIDI keyboard or sequencer – you can even specify the exact melody via MIDI

Bass Mode

- In normal mode ,pitch is reliably detected down to A0 (55Hz). Turning Bass Mode on lowers the lowest detectable frequency by about one octave to 25Hz (the lowest E string on a bass guitar is approx 41Hz).

Expression and Emotion

Great performance is a lot more than just hitting the right notes. Depending on the style of music, expressive gestures like scoops, bends, vibrato and more can all contribute to the emotional effect.

- The speed control ensures that the rate of pitch correction (i.e., the speed at which the input pitch is slewed to the target pitch) can be matched to virtually any style of performance.

Program Mode

- Call up one of 50 individual Programs to control the correction algorithm
- Each Program consists of a Scale, a Speed setting and Vibrato settings.
- Programs can be accessed remotely via a foot switch or via MIDI
- Use Program Mode when a single scale (or maybe two) is all that's required for a particular correction.
- If you are working in the studio to correct an already recorded track, you might use Program Mode to deal with the track one section at a time, stoping to change Programs between sections.

How it Works

Uses proprietary digital signal processing algorithms to continuously detect the pitch of a periodic input signal, compare it to a desired target pitch and, if necessary, instantly (no greater than 4 milliseconds) and seamlessly correct it to that target pitch. And with pristine 20-bit data path, 56-bit internal processing and balanced I/O, the only difference between what goes in and what comes out is the intonation. (Dynamic range is 103dB).

Use in Live Situations

- Song Mode lets you follow the most complex harmonic song structures by programming the appropriate series of scales and then stepping through them in performance with a footswitch or MIDI control.
- Most performance parameters can be controlled with MIDI continuous controllers. Everything can be controlled via SysEx.
- 20 different songs are provided each of which can be programmed with a series of up to 15 Song Steps. Each Step can contain a Program (with associated scale) or one of a number of navigation controls
- Each Song also contains its own Speed and Vibrato settings that override those settings in the individual Programs. In this way, an individual Program's scale can be used in any number of songs, each with different Speed and Vibrato settings.

Front & Rear Panel Controls

- Easy-to-read 20x2 LCD display
- Six-Segment LED level meters indicate the input signal level in dBs
- Pitch Change Meter indicates the amount of pitch correction being applied to change the input pitch to the target pitch in real time.
- Balanced XLR and 1/4″ TRS (bal/unbal) inputs and balanced XLR and unbalanced 1/4″ TS outputs
- MIDI In port
- 1/4″ TS (tip-sleeve) footswitch input

ANTARES

AMM-1

Microphone Modeler



Inside this unassuming but stylish 1U rack is contained the spirit of many of the world's most revered microphones. Based on the award-winning technology that made the Antares Microphone Modeler the most talked-about software plug-in of 2000, the AMM-1 lets almost any reasonable quality microphone sound like any of a wide variety of high-end studio mics. From historical classics to modern exotics to a selection of industry-standard workhorses, simply tell the AMM-1 what microphone you are actually using and what microphone you'd like it to sound like. It's really as simple as that. You can even mix and match the bass and treble characteristics of different microphones or add the warmth of classic tube saturation.

FEATURES

- Proprietary DSP-based acoustic modeling allows any reasonable quality microphone to sound like any of a wide variety of high-end studio mics Models reproduce the effects of windscreens, low-cut filters, pattern-dependent frequency response and proximity effects.
- Over 100 mic models are already built in, including an extensive collection of digital models of historical classics, modern exotics, and industry-standard workhorses. You can also add new models by downloading from the Antares web site
- Create hybrid mics that combine the bass response of one mic with the treble response of another
- Add a model of classic tube saturation distortion
- Use during mixdown to change the mic on an already recorded track
- Incredibly easy to use — simply select the mic you're using and the mic you want it to sound like.

Using patented Spectral Shaping Tool technology, Antares created digital models of a wide variety of microphones, from historical classics to modern exotics, as well as a selection of industry-standard workhorses. Simply tell the AMM-1 what microphone you are actually using and what microphone you'd like it to sound like. It's as simple as that.

With the AMM-1, you can afford to record each track through a model of the specific mic that will best produce that ideal sound you're looking for. Or use it in live performance to get the sound of mics you'd never consider using on stage. You can even use it during mixdown to effectively change the mic on an already recorded track.

Not only do the models reproduce the sonic characteristics that make each microphone unique, but they also give you control of each mic's specific options. Does the mic have a low cut filter? If so, it's in the model. Wind screen on or off? Close or far placement? Each option results in the same sonic effect that it would have with the actual modeled mic. And for that final touch of perfection, you can even add some tasty tube saturation.

Signal Path

Input – For setting the input level of the audio to be processed

Source Mic – For indicating the mic (and the state of its various parameters) that was actually used to record the audio.

Modeled Mic – For selecting the mic (and the state of its various parameters) whose sound you would like to model.

Tube Saturation – For adding a model of analog tube saturation distortion.

Output – For setting the output level of the processed audio. The use of each of the individual controls is covered below.

Rear Panel

- Balanced XLR and 1/4″ TRS line inputs
- Balanced XLR and unbalanced 1/4″ TS line outputs
- AES/EBU digital I/O is also provided – the output is always active, regardless of the input source (analog or digital)
- MIDI In connector lets you control the AMM-1 remotely from a MIDI sequencer
- A 1/4″ TS bypass footswitch input is compatible to two types of footswitches; those that are shorted by default and those that are open by default (the ATR-1a will automatically detect which kind of foot switch you have when the unit is powered up).

AVP-1

The following is the list of most of the mics that are included in the AMM-1 ROM. There are many more in the pipeline which will be added as their modeling is completed.

AKG: C12A; C12VR; C414; C414EB; C414B/ULS (Limited Edition Gold, modified by Audio Upgrades and modified by Jim Williams); C535EB; 460B/CK61-ULS; D1; D112; C1000S; D790; D100S; C3000; C4000B; The Tube; Solid Tube

Alesis: AM61

Audio Engineering Associates: R44C

Audio-Technica: ATM-11; ATM-31; AT853Rx; AT3525; AT4047/SV; AT4033a/SM; AT4050; AT4051; AT4055; AT4060

Audix: D4; OM2; OM3-xb; OM5; OM6; CX111; SCX1

B&K: 4007

Behringer: Ultravoice XM8500

Beyerdynamic: M500; M-500 Limited Edition; MC-834

Brauner: VM1; Valvet

CAD: Equitek E100; Equitek E200; Equitek E350; C400S; VSM1; 95Ni

Coles: 4038

Earthworks: Z30x; TC30K; QTC1

ElectroVoice: PL20; N/D357; N-D868; RE55; RE15; RE16

Groove Tubes: MD-1

Lawson: L47MP

Manley Labs: Reference Gold

Marshall: MXL 2001P; MXL 2003

MicroTech: Gefell UMT800

Neumann: U47; U87; U87 70th Anniversary Gold Edition; M147; M149; KM 84; KM184; TLM 103; TLM193; TLM103

Oktava: MC-012; MK-219; MK-319

RCA: BK5A

Rode: NT1; NT2; NTV

Royer: R-121

Sennheiser: MD421; MD441; E609; E835S

Shure: Beta 52; Beta 57A; Beta 58; Beta 87A; Beta 98D/S; SM7A; SM57; SM58; SM81; SM98A: SM98A; KSM32; VP88

Sony: C37P; C48; C800G; C800G(w)

Soundelux: U95S

Tannoy: Large and Small Vintage Ribbons

Telefunken: U-47



The heart of any great song is a great vocal sound. The Vocal Producer combines world-renowned Auto-Tune Pitch Correction with award-winning Microphone Modeler technologies and state-of-the-art vocal processing modules to give you everything you need to create stunning vocal tracks in any musical style. Live or in the studio, the AVP-1 lets you instantly select from a large library of sounds. From gorgeously mellow to seriously twisted, they've included factory presets for a wide variety of vocal styles as well as an interface that makes it easy to create your own signature sounds. (And given the power and flexibility of the AVP-1's processing modules, Antares even included a selection of presets for instrumental and percussion tracks.)

- **Auto-Tune Real-time Pitch Correction** - Antares"s world-renowned Auto-Tune technology lets you correct the pitch of vocals (or solo instruments), in real time, without distortion or artifacts, while preserving all of the expressive nuance of the original performance.
- **Antares Microphone Modeling** - The TEC Awarding-winning Microphone Modeler technology lets you give your vocal tracks the characteristics of a variety of high-end studio mics as well as adjust the proximity effect associated with mic distance.
- **Analog Tube Modeling** - Gives your vocals the warmth of a classic tube preamp.
- **Variable Knee Compressor** - A state-of-the-art dynamics processor with threshold, ratio, attack and decay controls as well as a continuously variable knee characteristic.
- **Downward Expanding Gate** - The AVP's gate, with threshold and ratio controls, works independently of the compressor to eliminate noise and breath sounds.
- **Variable Frequency De-Esser** - The AVP's de-esser tames vocal sibilance with threshold, ratio, attack and decay controls as well as a variable highpass frequency to match any vocal performance.
- **Flexible Parametric EQ** - You can fine-tune your vocal sound with two independent bands of equalization that let you select from 6dB or 12 dB high or low cut, high or low shelving with variable slope, bandpass, notch and fully parametric peaking.
- **Automatic Mono or Stereo Double Tracking** - You can automatically mix a doubled track into the AVP's main output or route it to a separate output for post-processing and mixing.
- **Fully Programmable** - Once you've created the perfect vocal sound for a particular track, every parameter can be saved as a preset for instant recall.
- **Factory Presets for a Wide Variety of Vocal Styles** - The AVP comes out-of-the-box with an extensive collection of factory presets for a variety of vocal styles. (There is even an included selection of presets for instrumental and percussion tracks.)
- **MIDI Automation** - Every variable module parameter can be controlled via MIDI continuous controllers for realtime automation.
- **Really Easy To Use** - No scrolling though endless menus to find the parameter you want. Virtually every major function is only a single button press away.

APHEX

204

Aural Exciter and Optical Big Bottom



Incorporating Aphex's patented Aural Exciter and Optical Big Bottom, the 204 enhances detail, clarity and imaging, and adds low-end punch with deeper, more powerful bass to dramatically enhance live, recorded, broadcast and webcast sound.

By extending the harmonics of the audio path, the 204 is able to restore the natural brightness, clarity, and presence that is often lost during analog/digital conversions, processing, or playback through 'less-than-perfect' audio equipment, bringing sound back to life. This makes the 204 ideal for sound recording and mixing, radio and TV, streaming audio, live sound reinforcement, home theater, nightclubs, discos, theme parks and other fixed installations. The Big Bottom with adjustable tune control provides deeper, more powerful bass performance without adding more power amps or loudspeakers and with little or no increase in peak output. There might be many other "enhancers" on the market, but none can match the effectiveness and musicality of the patented processes in the 204— regardless of price.

- Aural Exciter is scientifically proven to increase intelligibility. It pulls a vocal or an instrument up out of a mix. It punches up an entire mix, even in a noisy or reverberant environment.
- Two independent, low noise channels
- Adjustable Tune, Harmonics and Mix Controls. Tunable harmonics enhances the detail and presence of any instrument, voice or even an entire mix.
- Adjustable Drive, Tune and Mix Controls on Big Bottom
- Switchable -10dBV/+4dBu operating level
- Balanced XLR and 1/4″ input/output

207 *2-Channel Tube Mic/Instrument PreAmp with MicLim*

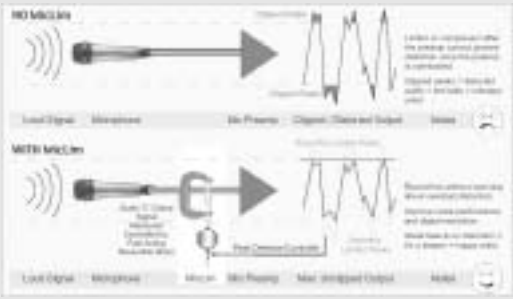


The 207 is an audiophile quality tube mic preamp without the high cost. In fact, it exceeds the performance of preamps costing many times more. The 207 will reveal the subtlety and power of vintage condensers, ribbons, electrets and popular dynamic mics. Even the world's most respected microphones will sound better. The instrument inputs are also quieter and have more gain than conventional instrument preamps. The MicLim circuitry, previously only available in the flagship 1100 and 1788 mic preamps, makes the 207 virtually crashproof -making it a perfect front end for digital recorders. By combining the best traits of tube and solid state circuitry, the 207 captures a sound that is warmer, fuller and more true to life—whether you are recording, mixing live, broadcasting or webcasting.

MicLim vs. No MicLim

The MicLim is custom designed optical attenuator, located directly on the mic input line that smoothly limits the microphone's output signal according to the MicLim peak detector's control current. Since attenuation comes prior to preamplification, the patented MicLim can increase the Model 207's effective headroom by up to 20dB.

MicLim detects the preamp's output signal and instructs the input attenuator to proportionately reduce the microphone's output level just enough to prevent clipping. MicLim has no effect at all when the preamp's output level is not approaching clipping.



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EQUIPMENT LEASING AVAILABLE

320A COMPELLOR

Dual Channel Compressor/Leveler/Limiter



The 320A delivers intelligent compressor action, leveling and peak limiting simultaneously. This intelligent, versatile and highly affordable processor can be used to solve audio level problems and improve audio signals in the broadcast studio, recording studio, tape duplication house, film dubbing studio and in live sound applications. Patented control circuits include analog computers that continuously analyze the input signal and vary the control characteristics to provide for virtually undetectable operation, regardless of the dynamics of the program. Extremely easy to use, you only need to set the Drive level to generate the desired amount of processing, set the Process Balance control between leveling and compression and adjust the output level for unity gain. The 320A is then ready to provide complete dynamic control - smooth, inaudible compression, increased system gain, desired program density and the freedom from constant "gain riding" - fully automatically! Its unique circuit design actually enhances transient qualities, thus making even heavy processing undetectable.

FEATURES

- ◆ Dual mono or stereo operation.

– Mono mode offers two completely independent channels of processing to accommodate independent monaural signal feeds.

– In stereo mode, you can link only the leveling control signals or you can link both the compression and leveling signals.

- ◆ "Invisible" compression characteristics assure tighter dynamics and virtually transparent performance
- ◆ Intuitive front panel metering system displays input/output or gain reduction levels.
- ◆ Intelligent Automated Gain Control (AGC) for consistent program levels
- ◆ All potentiometers are detented for accurate resetting of controls.
- ◆ Operating reference levels are selectable from -10dBV, +4dBu or +8dBu.
- ◆ Leveling Speed (fast/slow) switch is located on the front panel, as is the defeat switch for the peak limiter.
- ◆ An RJ-11 connector on the rear panel facilitates remote relay bypass of the unit.
- ◆ Instantaneous peak limiting for effective system protection (user defeatable)
- ◆ Adaptive control circuits make for fast, simple set-up and no readjustment for varying program dynamics

Dynamic level control is easy to attain. Just sit in front of your mixer, with your fingers on all the faders for all of your tracks. Then, when the signal on each track gets louder, immediately pull down that fader. When the music gets softer, instantaneously turn it up. Like we said, easy! Especially if you only have a 24 or 32 channel mix going...

Okay, maybe NOT so easy - unless you own an Aphex Compellor! Not a compressor or limiter, the Compellor is a an incredibly intelligent leveler. It automatically gives you dynamic control over your mix, letting you maintain optimum average levels at all times. Your recordings will sound louder, fatter, fuller and punchier - without sounding squished, squashed, flat or lifeless (the usual compressor artifacts) because the transients still get through, and your mix still has room to breathe.

If you want a more polished, mastered, and overall louder "pro" sound for your recordings, the Compellor is the best answer!

Frequency Discriminate Leveler (FDL) Circuit

Lab tests and exhaustive research led to the discovery that, under conditions of program leveling, the human ear perceives the onset of low frequency (bass & percussion) transients differently from transients at higher frequencies. This perception, as it turns out, is a direct function of the relative attack time of the leveler. Without FDL, there is a much greater chance that low frequency transients can create an audible "bass pull back" effect. In addition to a potential loss of bass and/or low end "punch", mid and high frequency processing can be negatively impacted. To the listener, the effect can be heard as a perceived loss of bass or even "pumping" at the mid and treble frequency ranges.

FDL eliminates this problem by allowing low frequency transients to trigger a slower attack time on the initial transient. High frequency leveling is still controlled within the attack time determined by the onboard computer. For the listener the benefits are:

- ➢ No more bass pullback effect
- ➢ More bass punch for better music mixes
- ➢ Fast leveling can be used in more applications
- ➢ Reduced audio distortion in the leveling mode

APHEX

622

Logic Assisted Expander/Gate



The choice of professionals in recording, broadcast and sound reinforcement, the 622 incorporates proprietary detection circuitry to provide the most reliable, accurate and stable operation of any gate. And with the ability to increase your dynamics along with lowering the noise floor of your gear, comes the highest quality VCA's to assure that you will never hear a click in your material again! The 622 also offers the highest quality audio path, flexible high "Q" key filters, key headphone outputs on the front panel and remote controlled bypass relays. Whether using it to keep your vocals clean and your bass sounds tight and fat or to control feedback, the 622 is a phenomenal addition to your processing rack!

- Logic Assisted Gate Circuitry combines sophisticated level detection with logic generated control voltage assuring positive, stable and accurate triggering regardless of attack time
- VCA 1001 voltage controlled attenuator assures the highest audio quality, speed and freedom from DC control feed through (no clicks regardless of attack time)
- Switchable parametric key input filters (-24 dB/octave filters with frequency and bandwidth controls)
- Lowest distortion (0.005% typical) and widest frequency response (5Hz-100kHz)
- Front headphone jack to monitor key input without disrupting audio output
- Dedicated downward expander mode allows high resolution of ratio adjustment from 1.2:1 to 10:1.
- Switchable ducking mode with fully adjustable control of trigger threshold, attack time, hold time, release time, expander ratio, and attenuation range
- 119 dB dynamic range—suitable for digital and analog recording applications
- Can be used in all recording and live performance applications
- Simple and accurate metering system
- Servo-balanced input and output circuits
- +4/-10 operation, rear panel selectable
- Hardwire relay bypass: remote controllable
- 5-year limited warranty

661

Tube Compressor/Limiter Expressor / Easyrider with Tubessence



This single channel compressor/limiter incorporates three patented Aphex circuits in one rack space: Tubessence, Easyrider and High Frequency Expander. Use the 661 to control dynamics for mixdowns, protect your sound system's loudspeakers from transients and distortion or shape the sound of individual instruments and vocals. Compress dynamic levels without fear of reducing frequency response. What ever type of dynamics control you're looking for, it can be done with the 661.

- Easyrider is an Auto mode that takes the guess work out of compressing individual instruments and vocals especially in live mixes. WDC (Wave Dependent Compressor) circuit automatically adjusts attack and release times according to the complexity of the program material. It's No-Knee compression curve makes the transition into compression virtually invisible.
- Tubessence provides that "vintage" tube sound without high heat, bulk, short life, fragility and sonic variability.
- In the manual mode, the 661 provides the full compliment of compressor adjustments, such as ratio, attack and release, for maximum flexibility. Express yourself and get those "punch", "slap", "fat", "pump", and "squeeze" sounds fast and clean.
- High Frequency Expander (HFX) decompresses high frequencies, with user-adjustable ratio and corner frequency. Add "life" and "air" to your compressed signals without fear of background noise pumping.
- Spectral Phase Refractor (SPR) restores clarity and punch to tracks. Bass frequencies are more defined with more apparent power. Higher frequencies gain detail and presence.

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ONE HOUR FREE PARKING
AT 349 W. 34th STREET (with purchase of $100 or more)

108

Two-Channel Auto Compressor



The 108 is a two channel auto compressor with Easyrider technology for the best sounding, most transparent compression available. Easyrider thinks for you, continuously monitoring the complexity of the program material and instantly adjusting the compression ratio and time constants accordingly—just set and leave it alone! Think of it as having an extra set of hands while you play. Easy to use, simple to set up, the 108 sounds completely transparent on individual voices, instruments and submixes as well as full program material.

- ◆ Rotary speed control varies the range of the WDC release time
- ◆ Dial in the desired amount of compression, up to 20dB available
- ◆ Gain reduction meter with 10 LED steps
- ◆ Channels linkable for stereo operation
- ◆ Switchable -10dBV / +4dBu operation
- ◆ In addition to the world-class Aphex 1001 Class A VCA, Aphex's Easyrider Technology is a combination of two circuits:

—Wave Dependent Compressor (WDC), a patented circuit which automatically adjusts compressor time constants according to program complexity:

—No-Knee compression curve circuitry which automatically adjusts the compression ratio. One of the most annoying aspects of conventional compressors is the audible "pumping" or "breathing" side effects heard when the signal exceeds the threshold and the compression "kicks-in". No-Knee ensures there is no abrupt transition to a high compression ratio.

109

Two-Channel Tubessence Parametric Equalizer



In a class by itself, the 109 has the flexibility to switch between Mono 4-band (1x4) operation when you need to address complex equalization issues or Dual 2-band (2x2) operation when more general equalization is required. For even more control, a switch is provided for selection of either peak/dip or shelving on all four bands. Each band provides variable boost/cut (±15dB), variable frequency (20Hz – 2kHz or 200Hz – 20kHz) and variable bandwidth (1/5 octave to 2 octaves) control. The 109 can be used for a combination of applications including gentle frequency response shaping and notching of specific frequencies, like 60Hz hum.

The 109 also includes the patented Tubessence circuit which provides true vacuum tube circuitry and sound without transformers, high heat, short life, fragility, and sonic variability. Due to the added "sweetness" of Tubessence, you will never hear an equalizer sound like the 109. Operating the unit in the EQ flat mode allows the signal to pass through the Tubessence vacuum tube stage without any gain change, providing you with a great Tube Processor. This feature proves helpful for digital users who wish to "warm-up" their digital signals.

720/722 DOMINATOR II

Precision MultiBand Peak Limiters



The Dominator II is a stereo MultiBand peak limiter designed to fit a wide range of applications. Through the use of MultiBand techniques along with proprietary circuits, the audibility of limiting action has been greatly reduced, especially when compared to conventional limiters. This means that greater limiting depth is possible, resulting in higher loudness with maintained audio quality. At virtually any limiting depth, the Dominator II is free of "hole punching", "dullness", and most other effects normally associated with limiters. As a peak overshoot protection limiter, the Dominator II is undetectable in line while it absolutely prevents peak levels from exceeding a user settable output level. In addition, the desired limiting effects of greater audio density and increased punch are readily available with the Dominator II.

Why is The Dominator So Special?

A significant problem with wideband processing is "spectral gain inter-modulation" which occurs when one part of the spectrum controls the level of another part. A typical situation is a vocalist being "sucked down" every time the kick drum hits. Since most energy is contained in the lower frequencies, they tend to control the level of the entire spectrum. When the lower frequencies are above the limit threshold the higher frequencies are attenuated thus causing the output to be dull.

Multiband processing solves this by splitting the audio into two or more frequency bands and processing each band separately. However, more bands can result in more parameters to control including a method of summing the bands together again. While giving you flexibility, it also requires different settings for almost every different source. The 720 and 722 use program dependent, intelligent circuits to reduce the number of controls. You therefore have flexibility to shape the sound while quickly and easily achieving the goal of consistent, effective limiting.

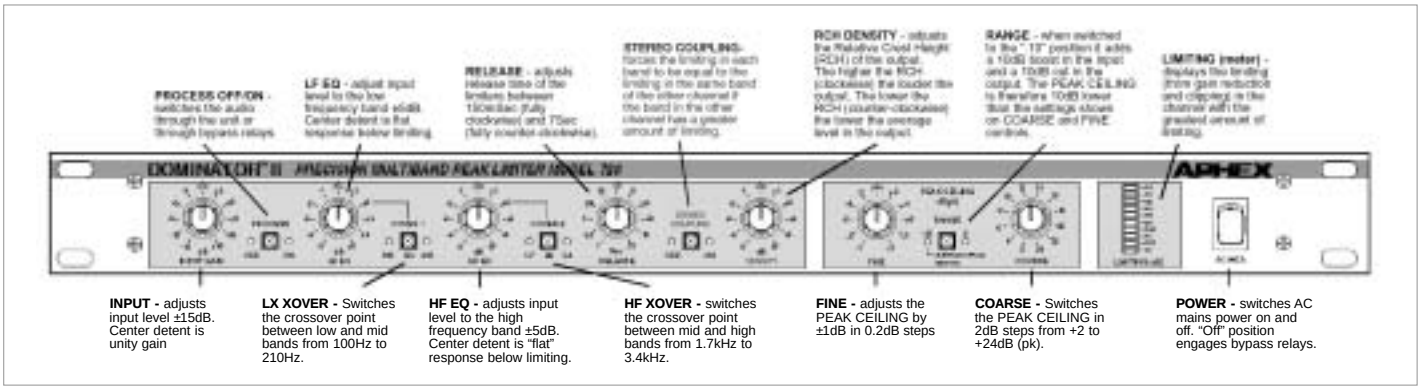
The Secret Ingredient: ALT

A multiband processor splits the audio into separate bands, individually limits each band, and then sums the bands together again. Even though each band's peak output is predictable, summing the bands together produces an unpredictable peak output.

Using a wideband limiter after summing introduces the drawbacks of wideband limiting. Using a clipper on the summed output can cause too much clipping distortion if the summed output is too high. To avoid this, the limiters' thresholds are set far below the clipper threshold resulting in a loss of loudness, and a great amount of processing.

The Dominator II produces a predictable peak output while maintaining maximum loudness without audible distortion using the Automatic Limit Threshold (ALT).

The outputs of the three bands are summed and sent to the ALT detector circuit. If the sum exceeds a reference value, the ALT reduces the thresholds of the individual limiters. When the summed output falls below the value the limit thresholds return to their original setting.



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720/722 DOMINATOR II

They Both Feature

- Peak Ceiling Trimmable in 0.2dB steps over a 34dB range
- 3-bands of limiting with switchable crossover frequencies
- Patented Automatic Limit Threshold (ALT) circuitry
- Freedom from pumping
- Freedom from Spectral Gain Intermodulation
- Adjustable density (relative crest height)
- Calibrated detented potentiometers
- 104dB dynamic range
- LF and HF EQ provides shaping equalization below peak ceiling
- Relay bypass is remote controllable
- Servo-balanced transformerless inputs and outputs

722

'Broadcast Dominator' Step-up Features

- Pre-emphasis is an equalization curve expressed as a time value based on the ratio of a resistor and capacitor. The higher the value, the greater the equalization. It has been employed as a noise reduction technique for broadcast and transmission links. There are primarily two world standards- 50 and 75 microseconds.
- The 722 Dominator II, has separately switchable pre and de-emphasis curves. When pre-emphasis is switched in, either 50 or 75 microseconds, the equalization curve is added after the input stage and before the limiters. When de-emphasis is switched in, the complementary de-emphasis curve is inserted after the final clipper and before the output stage.
- When both pre and de-emphasis are switched in, the frequency response of the output is flat. When the input is below threshold and as the input increases above threshold the output takes the shape of the de-emphasis curve.
- When both pre and de-emphasis are switched out, the 722 works exactly like the 720.

120 - 1 x 4 Servo-Balanced Distribution Amp



If you are in the market for a high-quality, performance driven Distribution Amp, then the Aphex 120 is just what you are looking for! With specifications better than digital recordings (better than 90db and 5Hz - 100kHz specs), the 120 is designed from the ground up to be rock-solid in any situation. So, for any situation that you can throw at it, Aphex's 120 Distribution Amp offers 4 outputs that perform exceptionally well!

The Model 120 is a high performance audio distribution amplifier with a single high impedance input and four low impedance outputs, all electronically servo-balanced. The 120's transformerless circuits are designed for wide, flat frequency response free from ringing or overshoot, making it ideal for distribution of SMPTE time code as well as audio. Each output has its own amplifier and level control for maximum versatility and isolation. The sturdy steel chassis may be used stand-alone or rackmounted, singly or in pairs.

Maximum gain without distortion. Whether working in analog or digital, its what we all strive for. The easiest and best sounding way to get it? The incredible Dominator II multi-band limiter.

Aphex's goal with the Dominator was first and foremost incredible SOUND QUALITY. Other manufacturers fool around with their digital this and that, look ahead schemes, and other voodoo tricks to attempt to attain what Aphex has been providing all along - effective level control— with undeniably transparent, punchy audio quality.

Not settling for just stopping peaks, the Dominator uses controlled, instantaneous musical clipping to control peaks, while still letting you increase the average level of your mix. This adds punch and power, but establishes a multi-band brickwall limit. In short, the Dominator lets you push your levels to their absolute maximum, with NO DISTORTION or negative artifacts. Your mix is simply as loud as it can be, with no worries about overload!

The Dominator is also ideal for establishing optimum input levels to your digital gear with no chance of peaks overshooting or causing harsh digital clipping. Putting one in front of your digital recorder is the best thing you can do for your recordings! And, are you using "in-ear" monitors on stage? The Dominator is the only limiter to consider for protecting your hearing, nothing else comes close - just ask virtually ever touring pro out there.

124 - Bi-directional Audio Level Matching Interface



So it's time to mix and match your studio gear's inputs and outputs... but everything is starting to sound awful??!! A CD player there, an ADAT recorder there... Throw in a few keyboards and a mixer. Throw in trying to match levels with Pro Tools and you've just created yourself a nightmare! Why? Simple... Start mixing different line levels (-10dBV for your keyboards and +4 for your ADAT or Mixer) and you're mixing levels are going to be all over the place. One line is going to be really hot while another is barely audible. But don't forget the noise or loss of dynamics that you are going to have with your gear when you start trying to match levels! That's where the Aphex 124a comes into the picture! The Aphex Audio Level Interface is designed to allow use of -10dBV consumer hi-fi equipment with +4 or +8dBm professional and industrial audio systems. It provides an extremely clean, reliable two-way buffer so both systems can operate at maximum performance levels, matching impedances and operating levels. We are not exaggerating when we tell you that the Aphex 124 is going to make your recording life so much easier!

APHEX

1100

2-Channel Tube Mic Preamp with 24-Bit 96KHz A/D Converter



With the 1100 your microphones come alive with unprecedented focus, clarity, space and detail. Recorded voices demand attention because they are uncannily present, powerful, and appealing. Most tracks will need much less technical fixup to stand on their own. Offering the lowest noise possible, 24-bit 96kHz digital outputs and Aphex's exclusive MicLim protection circuit, the dual channel 1100 is designed for digital or analog recording, broadcast and production. The breadth of features, superlative specifications and warm sound of tube processing make it the mic preamp of choice for state of the art audio production. Breakthrough technology allows the 1100 to deliver an EIN of -135dB. The MicLim circuit adds an additional 20dB of head-room, making it virtually impossible to overload the preamp. Since the A/D converter is scaled to clip at the same point as the mic preamp, the digital conversion will be at the highest possible resolution without overload. Aphex's patented tube circuitry provides the open, present, real and warm sound.

Each channel has a tunable low cut filter (30-195 Hz), tunable gain from 21 to 65 dB and multi-turn output trim. Both channels have individual switches for 20dB pad, polarity, test tone, phantom power on/off, mute, MicLim, Clock Source (Internal at 44.1, 48 or 96kHz and External). The back panel includes (for each channel) includes XLR In/Out, 1/4˝ balanced out, AES/EBU out (XLR), Word Clock In/Out (BNC), remote mute control jack.

MicLim Commands Enormous Headroom

Wide dynamic range means more than low noise. It also means generous headroom to handle extremely loud sounds. Normal mic preamps can't be protected from clipping, because any compression or limiting device comes after the fact. Unexpected loud sounds will jam the preamp into clipping, forcing you to ride gain or suffer distortion. Aphex's exclusive MicLim peak limiter works on the mic's own output signal, before any amplification takes place. It automatically and transparently limits the peaks before they can get clipped! MicLim is based upon custom engineered optocoupler technology so it is sonically pleasing while in action. Typically 16dB of useful limiting is available, effectively adding that much more to the already huge dynamic range of the 1100.

Class A Circuitry

All discrete Class A PNP first stage and the patented Reflected Plate Amplifier (RPA) tube circuitry in the second stage and in the output stage. These circuits provide the 1100 with unique sonic characteristics while maintaining low noise and high common mode rejection.

What's Thermionics?

The art and practice of using and developing electron tubes (which are also known as "valves" or "vacuum tubes") is frequently referred to as "Thermionics" in scientific and engineering literature. Inside of electron tubes, thermionic emission is obtained by heating the cathode to incandescence by the use of a filament. This causes electrons to escape from the cathode's surface. The cathodic emission is ultimately gathered by the tube's plate circuit and converted into an output signal. The plate current is controlled by a comparatively smaller input signal applied to an intervening electrode grid, yielding signal amplification. Any device utilizing this tube circuit technique may therefore be called a "Thermionic Amplifier".

Superior Dynamic Range

Besides the rich, involving sonics of the 1100, imagine what its incomparable dynamic range can bring? -135dBu EIN (Equivalent Input Noise) commands awesome quietude. Most professional mic preamps have an EIN of -124dB or so. Some may even reach as low as -127dBu under optimum conditions. With an unprecedented low of -135dBu at 65dB of gain — the only source of noise will be the mic itself. The increase of the perception of size and depth during recording can be awesome.

LoCaF Low Cut Filter

LoCaF (Low Cancellation Filter) allows up to 20dB of low frequency cut without using any headroom or triggering MicLim. This sweepable second order low cut process is realized through a servo cancellation method that is nodally meshed at the intersection of the first and second stages. Tunable in 11 steps from 30 to 195 Hz, LoCaF gives you complete freedom from rumble and wind blast overload, even in extreme conditions.

The 96KHz/24-Bit A/D Converter

The 1100 incorporates a full spec A-D converter to offer full-featured AES/EBU digital audio output. Clock synchronization options allow locking to standard "word clock" and to AES/EBU clock received via back panel BNC input. The internal clock may be set at 44.1, 48 or 96kHz sample rates. When set for external clock reference, the BNC input jack directly links to the BNC output for easy daisy-chaining of multiple 1100s. All digital audio settings are controlled and displayed on the front panel.

700 Hz Tone

How often have you wished you could get a calibrated tone from your mic preamp to set the console match gain or the record level? The 1100 provides a selectable test tone at exactly 20dB below the digital and analog clipping points. This tone transfers to both the analog line output and the digital audio output whenever activated.

Line Output Calibrate & Level Normalization

With the 1100 you don't have to compromise noise or headroom to get a level match to your system input. Using the test tone, simply use the Line Output Calibrate precision adjustment on the front panel in conjunction with the back panel +4dBu/-10dBV Normal Level switch to get a perfect signal level on your recorder or mixer meters. You'll be guaranteed 20dB of headroom and the lowest possible noise.

Soft Switch Mute

Wouldn't be nice to have a convenient way to soft mute the microphone? The 1100 includes a clickless, thumpless, mute feature that can be operated from the front panel or from a remotely located switch. A back panel phone jack allows you to attach any kind of switch to mute the channel from any desired location.

Slow Ramping Phantom

Thumps are greatly reduced by the slow changing 48-volt phantom power source of the 1100. Not only does this protect your ears and monitors, but it protects delicate microphones from the shock of a sudden power inrush. You will normally hear the mic fade in quietly rather than loud thumps and flutters when phantom is turned on.

1788—REMOTE CONTROLLED 8-CHANNEL PREAMP



1788 shown with the 1788R Remote Control

The 1788 is an 8-channel remote controlled mic preamp designed for production, broadcast, theater, recording and live sound. For live sound, not only does the 1788 place high quality preamps on stage near the microphones, but it allows a preamplified signal to be split for monitors, recording, and other feeds without a loss of quality due to loading of the microphone. The result is simply higher quality audio wherever the signal is needed. When used with the optional 1788R Remote Controller, up to sixteen 1788s (128 channels) can be remotely operated at the mix position with the actual 1788s located as close as possible to the microphones. The benefits are:

- The ability to run long cables without degrading the signal or audio quality
- The 1788's MicLim limiter circuit allows the signal to run at as high a level as possible without the fear of clipping at the consoles, amplifiers, signal processing or analog and digital recorders.
- Multiple splits can be made from any input without loading the microphone or causing any audio degradation.
- The 1788 can be controlled by either MIDI, RS-232 or RS-422. The control protocol is MIDI so that anything producing MIDI can act as the controller, such as a light controller board, a show control system, a sequencer, a DAW, a MIDI controller, a keyboard or a computer using 1788 control software.
- The 1788 has an RS-422 output which can be used to daisy chain units. Of course, all mic preamp functions can be controlled on the units as well.Up to 16 units (128 channels) can be controlled by one control line.
- The control software has a screen which displays all parameters and metering of one unit at a time and up to 16 units may be called up. All channel status information and metering are displayed in real time. A channel can be selected and settings modified. Scenes can be saved, modified and recalled in the control software.
- When using the optional 1788-1 digital module, digital audio can be simultaneously routed using three digital formats, AES/EBU, ADAT optical, and TDIF.
- Using the 1788R Remote Control minimizes the amount of hardware at the mix position
- Since there is no splitter, there is no need for cumbersome power supplies for condenser mics - the 1788 also provides phantom power
- You will no longer have the expense of buying and transporting heavy splitter boxes

TUBE MIC PREAMPS

Tube MP Series

ART's Tube MP series is a range of compact, professional tube preamps that provide sound quality and features that far surpass the "on-board" preamps found in today's low-cost mixers and multi-track recorders. Their TEC award nominated hybrid design utilizes the latest solid state and tube technology, based around a hand-selected 12AX7a tube, allowing them to add warmth and fatness to a signal while maintaining exceptionally low-noise and high quality. And though their primary application is microphone pre-amplification, they are also exceptional direct boxes, capable of impedance matching, amplifying and improving the sound of any instrument plugged into it—making them a must-have in any audio toolkit. Whether you're working with four-tracks or twenty four, tape or disk, analog or digital, a Tube MP will yield professional results at a fraction of the cost of comparable equipment.



TUBE MP

There's no better way to get that smooth, warm sound than with the world's #1 mic pre, the Tube MP. The one that started it all, the Tube MP provides superior preamplification for microphones, instruments and line level sources, and uses a 12AX7A tube for over 60dB of gain to warm even the most frigid signal.

Pre Amp

- Variable input and output gain controls
- +20dB Gain/Norm Switch
- Switchable +48V phantom power
- Phase Reverse/Norm Switch –reverses the polarity of the output signal
- Power/Peak LED – Green when power is on, red if 6dB below hard clipping

Superior Circuitry

- Transformerless design throughout ensures exceptional signal integrity and extremely low noise
- XLR mic input utilizes a hybrid multiple paired transistor/opamp design providing extremely low noise and excellent CMRR

Inputs/Outputs

- Active balanced XLR mic input and high impedance 1/4″ instrument input made it excellent as a tube DI for guitar, bass or keyboard as well
- Active balanced XLR and unbalanced 1/4″ outputs

TUBE MP OPL

The Tube MP OPL offers all the features and functionality as the Tube MP plus it adds ART's "OPL" Output Protection Limiter, which precisely and accurately, controls and maintains the output peak signal. The OPL circuitry is crucial in protecting the next link in a signal chain - such as a hard-disk recording system or a sound card.

TUBE MP Studio



The Tube MP Studio steps up from the Tube MP OPL with the inclusion of an output level VU meter, enabling you to keep the signal consistent with desired levels. The meter also reflects the impact of the OPL circuitry on the signal. For example, if the signal is "in the red" on the meter, the meter will reflect the attenuation of the signal when the OPL is activated, and the signal is brought out of the "red".

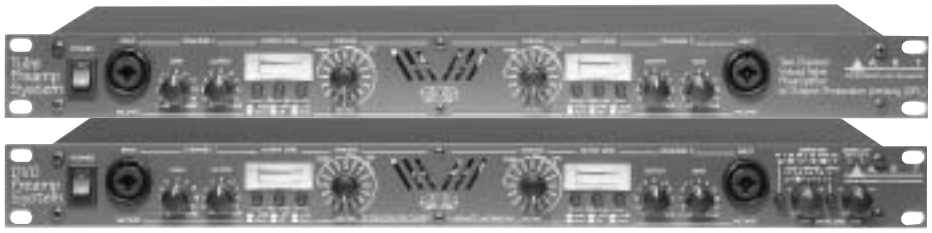
TUBE MP Studio V3

Tube MP Studio V3 takes things to a new level with Art's exclusive V3 (Variable Valve Voicing) technology. Using presets created and fine-tuned by some of the industry's top studio and live-sound engineers, V3 provides optimized reference points to begin the recording process. V3 allows you to select between a multitude of preamp settings designed for guitars (electric and acoustic), keyboards, bass guitars, drums, vocals and more. V3 gives you incredible presets for every instrument you record, so all your recordings will sound professional. You can also tweak the presets to hone your perfect tone.

TUBE MIC PREAMPS

TPS / DPS

2-Channel Tube Mic Preamp Systems



*The **Tube Preamp System (TPS)** is basically a two-channel, rack mountable version of the Tube MP Studio V3. Like it's single-channel counterpart, the Tube Preamp System (TPS) features twin high performance discrete preamplifiers coupled with ART's proprietary Variable Valve Voicing (V3) 12AX7A gain stage. OPL (Output Protection Limiter) is available to control overshoots and normalize levels before overloading or digital clipping occurs to the next link of the signal path. It can accept +20dB peaks while maintaining over 120dB dynamic range and incredibly low distortion. And unique to its class, the TPS provides level monitoring via two analog meters. High-Z instruments or balanced microphone signals are input through front and rear panel jacks.*

*The **DI/O Preamp System (DPS)** has all the features of the TPS plus adds digital outputs. A versatile insert loop on each channel provides access for additional signal processing or direct access to ART's high quality A/D converter. Separate gain controls on analog and digital outputs allow you to optimize the unit for simultaneous applications. Digital outputs include S/PDIF, TOSLINK or ADAT (front panel selectable). The A/D is front panel adjustable from 44.1 to 96kHz or syncs to ADAT or external word clock (32kHz to 100kHz).*

PRO MPA / DIGITAL MPA

Professional Two-Channel Tube Mic Preamps





Multi-purpose tools for audio engineering and recording, the PRO and Digital MPAs are ART's top-of-the-line mic preamp. Developed with studio and live sound engineers using the best components available, they deliver "the sound" while ensuring a lifetime of quiet, reliable performance. Backed by five-year warranty.

They both feature:

- Variable input impedance for flexible mic voicing (150 to 3000 ohms)
- Selectable tube plate voltage
- Large VU meters and Peak-Hold LED meters
- Metering is switchable between output level and tube warmth
- 1/4″ input has a high impedance which makes the PRO MPA perfect for DI or line level applications.
- Each channel features independent input and output level controls and a variable high pass filter.

Digital MPA Step-up features:

- A/D insert jacks, 44.1/48/88.2/96/176.4/192 kHz internal sample rates
- Rotary encoder for quick selection sample rate and output format
- Separate analog and digital level controls; fast and accurate digital level LED meters
- ADAT optical I/O; sync to incoming ADAT data rate
- Selectable optical (S/PDIF or ADAT) and AES/EBU output
- Two word clock jacks allowing loop through
- Attractively styled with a light gold front panel and matching knobs

PRO VLA

Professional Two-Channel Tube Compressor with Vactrol



The rack-mountable (2RU) PRO VLA (Vactrol-based Leveling Amplifier) is a powerful tool for audio engineering and recording. It features two independent channels of analog leveling/compression designed to work seamlessly with any recording, sound-reinforcement, or electronic instrument setup. ART's PRO VLA circuitry utilizes the latest and most advanced analog and tube technology. Using a transformerless design throughout, the PRO VLA maintains exceptional signal integrity and extremely low noise. Its VCA-less design utilizes optical electronics (Vactrol) coupled with a 12AX7 vacuum tube gain stage for superior musical performance.

The PRO VLA is a soft knee leveling amplifier by design. Although it is capable of providing a thoroughly "squashed" signal, it excels in areas where transparent, expressively musical dynamics control is desired.

In addition to making signal levels more manageable, it is common practice to apply compression or limiting to a signal to make it louder or more "in-your-face". By utilizing opto-electronics, the PRO VLA allows you to add more compression without sounding like you're squashing the daylights out of the signal.

Unlike typical compressors which use VCA's to control level detection, the PRO VLA is very musical. The nature of its operation is much like the way your eye adjusts to light. Just as your eye transparently adjusts to changes in light, the PRO VLA adjusts to changes in signal level.

Developed in partnership with studio and live sound engineers, the PRO VLA possesses a "sound" that is not available from any other compressor — at any price! Designed and constructed with the absolute best components, assuring a lifetime of quiet, reliable performance.

TUBE COMP *Tube Compressor*



The Tube Comp features electro-optical (Vactrol) and tube electronics to deliver superior dynamics control, musicality and classic sound without the pumping, breathing and noise associated with most VCA-based compressors. Very fast attack and release response times provide the classic punchy and transparent characteristics of the "LA-" type levelers; finely tuned auto setting behaves like the classic stereo modules, providing a smooth, loud, and fat sound.

FX-1 *Digital Effects Processor*



Housed in a compact, all-steel chassis are 60 stunning digital signal processing effects. Choose from clean, lush reverbs including halls, chambers, plates, gated, reverse and true stereo rooms and plates. Add powerful delays, thick chorus, shimmering flange, vintage tremolo and panning effects, pitch shifting, precise combination multi-effects and you have the easiest to use, best sounding, compact processor on the market.

TUBE CHANNEL STRIPS

TUBE PAC *Tube Mic Preamp/Optical Compressor*



The TUBE PAC combines the world's most popular tube mic/line preamp (Tube MP) and an optical tube compressor into a single convenient table-top design. Easily placed in the studio or on the stage, the Tube Pac offers immediate access to all the controls, features, and metering. The mic/line preamp stage provides over 70dB of gain, +48v phantom power, phase reversal, and is capable of warming up any instrument or microphone. The tube compressor is a VCA-less, optical design–similar to that of the "classic" tube compressors. Its ultra-musical sound is ideal for voice and instrument applications. Threshold, compression/limiting ratios, adjustable release times, bypass, gain reduction metering, and output gain controls are provided.

TUBE PAC

- Hand-selected 12AX7a vacuum tubes – one for the preamp and one for the compressor.
- Precision detented potentiometers
- Bypassable compressor circuitry
- Balanced XLR and unbalanced 1/4″ inputs and outputs
- Input level controls allows you to dial in the desired amount of tube warmth and then independently set the output level from the compressor circuitry.
- LED level indicators let you monitor the output of the tubes for harmonic content.
- Switchable +48v DC phantom power
- The ultimate tube DI box for instruments
- Fat and warm gain with musical dynamics control
- Aircraft-quality extruded aluminum chassis

TUBE CHANNEL *Tube Mic Preamp/Optical Compressor/EQ*

A rackmount version of the TUBE PAC, the TUBE CHANNEL also adds a 4-band, tube-based parametric EQ and analog VU meter.



- Award-winning, tube-based preamp features over 60dB of gain, +48v phantom power, and a 20dB pad.
- The optical compressor features compression and limiting settings, selectable release times, adjustable threshold, and output level controls. Gain reduction is displayed via a four-segment LED array.
- The 4-band parametric EQ section provides selectable high and low shelving filters with two mid-bands which sweep from 20Hz to 20kHz. The mid-bands overlap from 200Hz to 2kHz for the ultimate in flexibility.
- Insert points are available between each of the three processing sections. These inserts provide direct outputs from the preamp and the compressor for signal routing flexibility. Additionally, these inserts allow each processing section to be isolated for independent usage or to change the order of the processing sections (eg. you can place the EQ before the compressor).

PRO CHANNEL

Professional Tube Mic Preamp/Optical Compressor/EQ

Redefining the term "channel strip" the PRO CHANNEL combines three independent world-class modules: tube mic preamp, switchable optical/variable mu compressor, and tube EQ to create the world's best-sounding and most affordable professional recording channel.

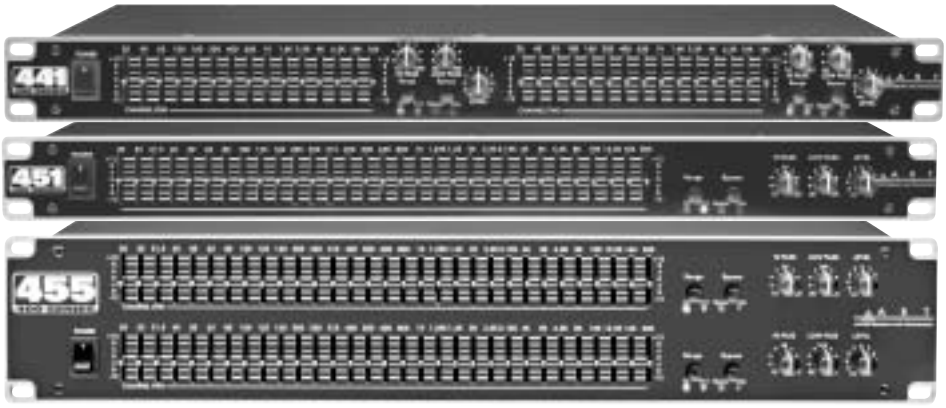


In addition to the compressor module, the PRO CHANNEL features a Variable Mu circuit. Users can select between ART's award winning optical/tube compressor or the coveted Variable Mu circuit (a design made popular by the sound of the old broadcast limiters – now costing thousands of dollars). The Variable Mu circuit is fast, fat and a bit aggressive, and has joined the Pro VLA in becoming a legend among compressors. The 4-band parametric equalizer has selectable Q for the two sweepable mid-bands. As with all design elements of the Pro Channel, these Q options were fine-tuned by some of the industry's top studio and live-sound engineers.

- Three hand-selected vacuum tubes
- Warm, smooth, and silky sound quality
- Selectable VU metering (mic pre out, compressor out, or main out)
- Tube character and gain reduction LED arrays
- Balanced XLR and unbalanced 1/4″ inputs and outputs
- Precision detented potentiometers
- Insert points between preamp and compressor, and compressor and EQ
- Five-Year Warranty

441/451/455

Dual 15-Band/Single and Dual 31-Band Graphic EQs



ART's high performance 441 (Dual Channel 15 band 2/3 octave), 451 (Single Channel 31 band 1/3 octave) and 455 (Dual 31-Band 1/3 octave) graphic EQs feature constant Q circuitry, 20mm oil-damped precision detent sliders with a selectable boost/cut range of ±6 or ±12dB, multiple inputs/outputs, adjustable high pass and low pass filters, 4-segment LED level metering with clip level indicators, variable input level controls, internal power supply with selectable line voltage, and ground lift switch. They also offer automatic relay bypass of audio, an essential function if power is lost. They are ideal for any audio application where precision frequency tailoring, reliable performance, rugged design and extremely silent processing is of the utmost priority.

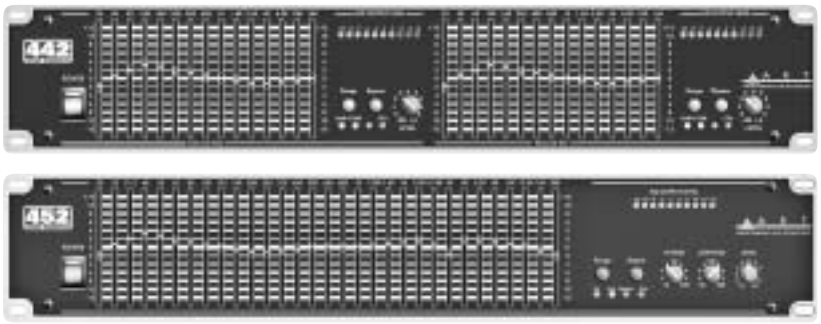
- The high pass filter rolls off lower frequencies to decrease rumble or low frequency hum from a signal. Its range is adjustable from 10Hz to 250Hz. Frequencies below this setting are rolled off, while frequencies above are unaffected.
- The low pass filter rolls off higher frequencies to reduce hiss or sibilance from a signal. Its range is adjustable from 3kHz to 40kHz.
- Multiple input and output configurations are provided for ease of use and incorporate paralleled connections. The XLR and 1/4″ TRS connectors use active-balanced, low-noise circuitry and both connectors may be used in an unbalanced configuration. The RCA jacks are unbalanced. The input level control covers a wide range and easily accommodates -10 and +4dB signal levels.
- Each channel has its own bypass switch with LED indicator.
- The 441, 451 (1RU high) and 455 (2RU) are housed in a rugged, all-steel chassis
- They are ideal for any audio application requiring precise equalization. Use them in live sound systems/PA, permanent or fixed installations, church, DJ, home and project recording, and monitor systems.

ART's EQs are used in a variety of applications such as live sound, recording studios, instrument racks as well as any conventional fixed installation environment. Use them wherever precise modification of the frequency contour of a sound is needed. The graphic EQ is a powerful tool for solving a number of audio problems and creating interesting sound textures.

Patch the EQ between your mixing console and power amps to allow you to alter the overall mix to better match your environment. Use them between your monitor mixer and monitor power amps to aid in the removal of feedback inducing frequencies. Patch into a channel insert to EQ one channel or track individually. Run your instrument directly into the EQ to enhance its sound before reaching a power amplifier or instrument amplifier. Patch them into the effects loop of an instrument amplifier or between a preamp and power amp to have precise control over the complete sound.

442/452

Long Throw, Dual 15-Band and Single 31-Band Graphic EQs

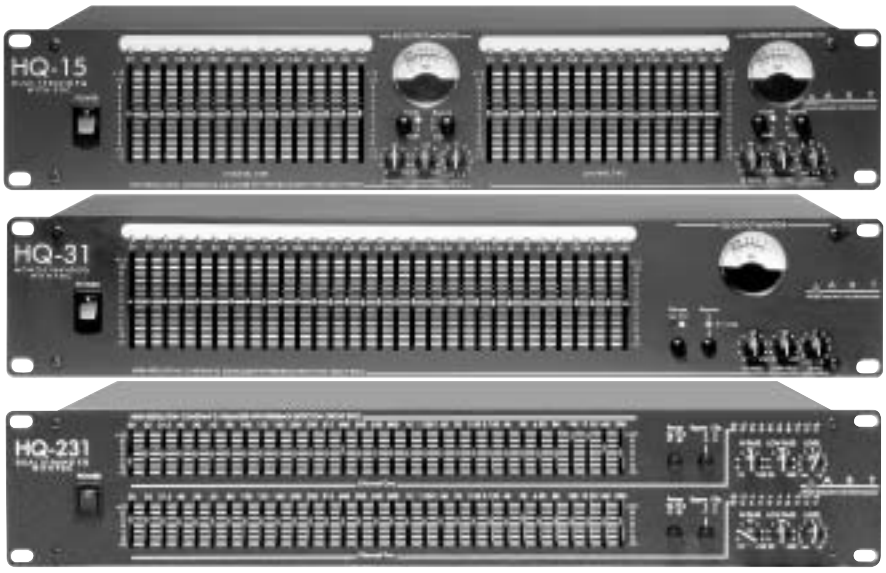


Same as the 441 and 451, the 442 and 452 step-up with 60mm long throw sliders & 10-segment LED meters for output

B&H PHOTO - VIDEO - PRO AUDIO

HQ-15/HQ-31/HQ-231

Dual 15-Band/Single and Dual 31-Band Graphic EQs with Feedback Detection



Stepping up from the 442 and 452, the HQ-15 (Dual Channel 15-Band EQ), HQ-31(Single-Channel 31-Band EQ) and HQ-231 (Dual-Channel 31-Band) are equipped with ART's proprietary FDC (Feedback Detection Circuitry) to help you keep live sound under control. Different from other systems in that it is extremely fast and accurate, FDC utilizes a set of led indicators that show which of the many EQ bands has the greatest energy. And while FDC illuminates the band with the greatest energy with the brightest LED, it doesn't fully discriminate between bands and will not get "stuck" on one band when there is no signal. The LED indicators can immediately show which band corresponds to the feedback frequency, allowing sound engineers to reduce gain in that band to quickly kill the feedback while having minimal impact on the sound of the live program material. The FDC also acts as a simple spectrum viewer while the concert proceeds showing where the "hot" areas are. During system setup and sound check, FDC may be used to help identify room and sound system resonances. By increasing the gain until feedback, it is easy to identify those frequencies that are likely to be problems. Reducing the gain in these bands helps tune the sound system, and, remove problem areas before the show begins. It also helps increase the gain margin before feedback.

	441	442	451	452	455	HQ15	HQ31	HQ-231
Frequency Bands (ISO spacing)	2x15, 2/3 oct.	2x15, 2/3 oct.	31, 1/3 oct.	1x31, 1/3 oct.	2x31, 1/3 oct.	2x15, 2/3 oct.	1x31, 1/3 oct.	2x31, 1/3 oct.
Filter Type		Constant-Q -3% Center Accuracy						
Slider Travel	20mm	60mm	20mm	60mm	20mm	45mm	45mm	20mm
Range (Selectable)	+/-6dB or +/-12dB							
Input/Output Connections	Active balanced XLR and 1/4" TRS {XLR pin 2/ 1/4" tip = Hot (+)} Unbalanced RCA							
Input Impedance (Bal/Unbal)	24kΩ/12kΩ	24KΩ/12KΩ	24kΩ/12kΩ	24KΩ/12KΩ	24kΩ/12kΩ	20KΩ/15KΩ	20KΩ/15KΩ	20KΩ/15KΩ
Maximum Input Level	+22dBm	+22 dBm	+19dBm	+22 dBm	+19dBm	+22 dBm	+22 dBm	+22 dBm
Output Impedance (Typical)	<150Ω	<150Ω	<150Ω	<150Ω	<150Ω	<150Ω	<150Ω	<150Ω
Max. Output Level	+22dBm	+22dBm	+19dBm	+22dBm	+19dBm	+22dBm	+22dBm	+22dBm
Frequency Response (+/-.5 dB)	20Hz - 20kHz	20Hz - 20kHz	20Hz - 20kHz	20Hz - 20kHz	20Hz - 20kHz	20Hz - 20kHz	20Hz - 20kHz	20Hz - 20kHz
THD + Noise	0.01%	0.01%	0.01%	0.01%	0.01%	0.01%	0.01%	0.01%
Signal to Noise Ratio	-94dB	-94dB	-94dB	-94dB	-94dB	-94 dB	-94 dB	-94 dB
Channel Separation	>50dB		>50dB					
Size (HWD)	1.75 x 19 x 7″	7.5 x 19 x 3.5″	1.75 x 19 x 7″	7 x 19 x 3.5″	3.5 x 19 x 7″	8.5 x 19 x 3.5″	8.5 x 19 x 3.5″	7.5 x 19 x 3.5″
Weight	7.5 lbs.	7.5 lbs.	7.5 lbs.	7.5 lbs.	12 lbs.	4.5 lbs.	4.5 lbs.	12 lbs.

AVALON

VT-737SP

Pure Class A, Vacuum Tube Discrete Preamp, Opto-Compressor and 4-Band EQ



The Vt-737sp is a high-quality recording channel that combines a Class A vacuum tube preamplifier, vacuum tube opto-compressor and discrete parametric equalizer — all of the essential elements needed to provide a musical direct recording path to the input of your audio recorder or DAW. Only the best active and passive components available are used in the Vt-737sp including 4 military-grade vacuum triode tubes integrated into a minimal signal path with 100% discrete, high-bias pure Class A audio amplifiers. Twenty-two sealed silver relays are employed providing hard-wire bypass and multiple routing options for total creative control and minimum signal degradation. Balanced XLR inputs are available for both mic and line-level signals as well as an XLR output with a balanced DC coupled output amplifier. A high impedance 1/4″ instrument input jack is also provided on the front panel allowing direct connection of a guitar or bass.

FEATURES

Vacuum Tube Design

- Combination of a tube preamp, opto-compressor, sweep equalizer, output level and VU metering in a 2U space nineteen inch welded steel chassis
- Allows you to amplify, control, and shape a signal from its source and deliver the signal directly to the input of your audio recorder or DAW completely bypassing the mixing console and its sonic limitations
- 4 dual triode vacuum tubes (Sovtek 6922), high-voltage discrete Class A with a 10 Hz to 120kHz frequency response ±0.5dB
- Twenty-two sealed silver relays employed for signal routing and (hard-wire) bypass switches, provide the most direct signal path and allow signal to pass through the unit when power is turned off
- An LCD display shows the number of hours that the tubes have been operating
- 150W toroidal shielded power transformer
- Switchable AC power supply 100-240V
- To ensure that there is no strain on the electronic components when the unit is activated, a “soft-start” feature slowly brings the unit to life when the unit is powered on

Mic Preamp

- The Mic Preamp stage employs a high voltage vacuum tube design that can be used for all types of dynamic and condenser mics, Hi-Z instruments such as guitars and basses, and for line level devices such as keyboards, mixers, recorders or DAWs
- Three different devices can be connected simultaneously and easily selected with the input mode switch on the front panel
  - ➢ A high performance XLR balanced mic input transformer with switchable +48v phantom power
  - ➢ A high impedance (one meg ohm) unbalanced 1/4″ jack located on the front panel allows an instrument such as a guitar or bass to be connected directly to the tube preamp
  - ➢ Discrete high-level Class A balanced line input
- A High gain switch, available to all inputs, boosts overall preamp gain and can be used to increase weak input signals or to achieve a wide range of distortion effects ranging from loose tube overdrive to tight discrete distortion
- A passive variable high pass filter and phase (polarity) reverse are available for all inputs and are, hardwire relay bypass switchable

Opto-Compressor

- The compressor uses a minimum signal path design and features twin Class A vacuum tube triodes for gain matching. The passive optical attenuator, used for gain reduction, serves as a simple yet musical level controller
- Continuously variable threshold level, ratio, attack, and release controls provide a variety of useful dynamic effects, from soft compression to hard-knee limiting
- To provide maximum flexibility, the compressor can be positioned either before or after (Pre/Post) the equalizer
- For even greater sonic possibilities, the two sweepable mid-EQ bands can be side chained to the compressor allowing a broad range of spectral control including de-essing
- A hardwire relay bypass switch engages the compressor into the signal path
- Stereo linking (tracking) between two VT737SPs is possible via an unbalanced 1/4″ link jack on the rear panel

VT-737SP

4-Band Passive-Active EQ

- The four-band parametric equalizer section features a 100% discrete high voltage class A design with both variable-active and switched-passive filter topologies
- The High and Low Mid frequency bands each feature active peak/dip filters each with a boost and cut of ±16 dB as well as a Q switch which allows you to select the appropriate bandwidth (Medium or Wide)
  - ➢ The low-mid frequency is variable between 35 to 450 Hz
  - ➢ The high-mid frequency is variable from 220Hz to 2.8 kHz
- The low frequency passive shelving EQ is selectable between 15, 30, 60 and 150 HZ with a boost and cut of ±24dB
- The high frequency passive shelving EQ is selectable between 10, 15, 20 and 32 kHZ with a boost and cut of ±20dB
- As previously mentioned, the High and Low Mid band EQs can be assigned to the compressor's side chain circuit. When this is the case, the low and high EQ's are still available for tonal adjustment

Metering

- The high quality analog VU METER is switch selectable to indicate line output level or gain reduction for the compressor **(0 VU= +4dB)**
- The VU meter's needle is also speed sensitive for measuring gain reduction which helps in setting the ATTACK and RELEASE of the compressor
- A recessed screwdriver adjustment is provided for calibrating the VU meter to 0dB when set to read the compressor's gain reduction

Output Section

- The final output stage utilizes yet another dual triode vacuum tube driving a 100% Class A, high-current balanced and DC coupled low noise output amplifier
- A continuously variable trim pot lets you control the output level (-40dB to +10dB with center detent at 0dB)

Pre Amplifier Controls	
Preamp GainLine	-20 to +10dB (+18dB with gain boost)
Preamp Gain Instrument	-30 to +10dB (+28dB with gain boost)
Preamp GainMic	0 to +40dB (+58dB with gain boost)
Input Source Select	Line XLR balanced, unbalanced 1/4″ Instrument jack, Mic XLR balanced
Input Mode Switches	
High gain	Adds +8dB to Line level signals and +18dB to Mic or Inst signals
Phase	Inverts the polarity of all 3 input sources
+48v	Phantom power to XLR Mic input
Passive Filter	6dB per octave passive bass cut filter variable from 30Hz to 140Hz
Compressor Controls	
Threshold	-30dB to +20dB
EQ to COMP	Places the EQ pre the compressor
Meter	Selects gain reduction (in) or output (out) metering
Ratio	1:1 to 20:1
Compress Switch	Engages compressor into signal path
Attack	Variable from 2ms to 200ms
Release	Variable from 100ms to 5 sec
4-Band Equalizer Controllers	
Bass (Passive Shelf):	
Boost/Cut	±24dB with center detent at 0dB, Shelf response
Frequency Range	4-position switch (15 / 30 / 60 / 150 Hz)
Low Mid (Active Peak-Dip):	
Boost/Cut	±16dB with center detent at 0dB, Peak-dip response
Frequency Range	Variable from 35 to 450 Hz or 350 to 4.5k Hz w/ x10 switch engaged
Bandwidth (Q)	Switchable for Narrow (Q=0.5) or Broad (Q=0.2)
High Mid (Active Peak-Dip):	
Boost/Cut	±16dB with center detent at 0dB, Peak-dip response
Vari frequency	Sweepable 220 to 2.8k Hz or 2.2k to 28k Hz w/ x10 switch engaged
Bandwidth (Q)	Switchable for Narrow (Q=0.5) or Broad (Q=0.2)
Treble (Passive Shelf):	
Boost/Cut	±20dB with center detent at 0dB, Shelf response
Frequency Range	4-position switch (10 / 15 / 20 / 32 kHz)
EQ to SC:	Routes the mid band EQ sections into the compressor sidechain for de-ess, effects and frequency selective compression
Output Gain:	45dB to20dB
Inputs and Outputs —	
Hi-Z Instrument DI	Unbalanced (1 meg ohm) 1/4″ input on front panel (+30dB max input level)
Mic Input	Active Balanced XLR (850/2500 ohm) with a maximum input level of +30dB
Line Input	Active Balanced XLR (20k ohms) with a maximum input level of +36dB
Line Output	DC-coupled, electronically balanced XLR discrete Class A on rear panel with a maximum output level of +30dB into 600 Ω
Compressor Link	Unbalanced 1/4″ jack for stereo linking the compressor sections of 2 Vt-737SPs

System Performance

Frequency Response (-2.5dB)	10Hz to 120kHz input filter included
Frequency Response -3dB	1Hz to 200kHz line in-out
Noise (20kHz Bandwidth)	-92dB unweighted
Noise Microphone EIN	-116dB, 22Hz to 22kHz unweighted
Distortion THD, IMD	0.5%

AVALON

VT-747SP

Class A, Vacuum Tube-Discrete Twin Signal Path Opto-Compressor / 6-Band Program Equalizer



The Vt-747sp is a pure Class A, stereo opto-compressor and program equalizer that delivers unmatched routing flexibility and creative control for input signal conditioning, stereo mix bus and mastering applications. The Vt-747sp's TSP (Tube Signal Path) design gives you a choice of warm tube processing with three hand selected vacuum tubes in the signal path or you can bypass the tubes and switch to classic Class A discrete transistor sound. The passive six band graphic equalizer offers gentle tone shaping combined with the two-band parametric EQ, assignable to the sidechain in the compressor section, provide complete spectral and musical control. Like all Avalon products, the Vt-747sp is hand built in the USA using only the finest active and passive electronic components available. A "no compromise" approach in every stage of design and production ensures that the Vt-747sp will give many years of dependable high-quality service.

FEATURES

Overview

- Combines a stereo tube/ discrete Class A optical compressor and a musical six-band program equalizer with separate output level and gain reduction metering and an internal regulated power supply — all in a 2U space nineteen inch welded steel chassis with rugged stainless steel metric hardware
- Ideal for high performance DAWs, input signal conditioning, stereo buss processing, stereo keyboard tracking as well as mastering applications
- Features a "no compromise" high voltage, high current, minimum signal path design with 22 sealed silver relays for all signal routing and hard-wire bypass functions
- 150W toroidal shielded power transformer
- The AC power supply is switchable for 50/60Hz, 100-240V operation
- To ensure that there is no strain on the electronic components when activated, a "soft-start" feature slowly brings the unit to life when the unit is powered on

TSP - Twin Signal Path

- The TSP (Tube Signal Path) gives you the choice of either tube tone (utilizing three high-voltage dual triode tubes), or the classic, Class A discrete transistor sound (utilizing discrete transistor amplifiers) in the primary opto buffers and output stages

Input Stage

- The input stage provides +36dB of headroom before overload
- A continuously variable trim knob controls the input level of the signal source. A wide range of sounds and colors can be achieved simply by varying the input level — In tube mode, the harder you drive the input, the more tube tone, can be sent into the compressor. In discrete mode (TSP disengaged) driving the transistors harder will achieve a more classic discrete transistor sound
- A +10dB Gain switch can be used to (overdrive) the input signal, or in a more utilitarian fashion, for matching low level sources such as keyboards and synthesizers



Rear view of the Vt-747sp

Opto-Compressor

- Compression is achieved by twin optical attenuators that act as passive level controllers. The opto-compressor design was chosen because it provides the most musical compression control
- The compressor section is immediately followed by a Class A amplifier with a variable gain control that provides up to +10dB of gain make-up
- Full dynamic control from soft compression to hard-knee limiting can be achieved with variable threshold, compression ratio, attack and release controls as well as a hard-wire bypass

Spectral Control

- Parametric Low and High frequency (LF and HF) spectral contour controls can be routed into the on-board side-chain path on the compressor for enhanced frequency dependent compression, such as de-essing. Variable frequency and threshold level controls are provided independently for the Low and High frequency
- An SC LISTEN switch allows you to monitor the signal going into the Side Chain before it is effected by the compressor

VT-747SP

6-Band Graphic Equalizer

- Extremely smooth and musical — The six frequency bands and corresponding Q settings have each been carefully chosen to provide the most natural harmonic balance and lowest phase change while offering simple and effective tone control
- The EQ's passive filter was to designed to allow you to subtly shape and color the sound source while still keeping the signal musically balanced
- 100% discrete, Class A high voltage transistors are employed to ensure optimum sonic performance
- The EQ section is controlled by six faders with center detents and can be hard-wire bypassed
- The EQ can be routed pre or post the compressor providing greater creative flexibility and sonic capabilities

Inputs & Outputs

- Fully balanced inputs and outputs with XLR connectors are driven by DC coupled, Class A discrete amplifiers that provide +36dB input headroom

Metering

- A high quality analog VU meter indicates accurate gain reduction
- A fast-acting blue LED illuminates when peak gain reduction begins to occur
- Twin LED meters provide a 60dB range with fast L-R output status of all levels

System Performance

Noise 20-20kHz unweighted	-92dBu
Distortion (THD, IMD) @ 1kHz	0.5%
Bandwidth (-3dB)	1 to 200kHz
Frequency Response	10 to 40kHz +/-0.2dB

Input Stage	
Input Gain Trim	-20dB to +8dB detent at 0dB (+18dB with +10dB IN)
+10dB switch	Boosts input amplifier by +10dB for extra tube over-drive
TSP Twin Signal Path Switch	Swaps between tube and discrete Class A amp topology
Opto-Compressor Controls	
Threshold	Variable from -30dB to +20dB
Attack	Variable attack from 2mS to 200mS
Release	Variable from 100mS to 5S
Ratio	Variable from 1:1 to 20:1
EQ to COMP	Inserts the six (6) band EQ pre the compressor
Compressor Switch	Engages compressor into signal path and offers hard-wire bypass
Make-Up Gain	Variable from 0dB to +10dB
Sidechain Controls	
Side-Chain	Inserts spectral EQ into sidechain
Spectral Sweep Frequency Controls	
LF	60Hz to 1kHz
HF	600Hz to 10kHz
Sidechain Threshold Controls	
LF Threshold Range	-15dB to +15dB, detent at 0dB
HF Threshold Range	-15dB to +15dB, detent at 0dB
SC Listen Switch	Inserts Spectral EQ signal into audio path for monitoring
Six Band Passive Class A, Program Equalizer	
Fader 1 (Low-Bass)	±24dB boost/cut @ 15Hz shelf
Fader 2 (Mid-Bass)	±8dB boost/cut @ 125Hz peak-dip
Fader 3 (Low-Mid)	±4dB boost/cut @ 500Hz peak-dip
Fader 4 (Mid-Range)	±dB boost/cut @ 2kHz peak-dip
Fader 5 (Presence)	±10dB boost/cut @ 5kHz shelf
Fader 6 (Treble)	±20dB boost/cut @ 32kHz shelf
EQ Switch	Engages 6 band passive EQ and offers hard-wire bypass
Metering	
Blue LED	Indicates compressor speed and activation of compressor
High quality VU meter	Indicates gain reduction 0dB to -20dB
Recessed VU Adjust	Screwdriver adjust for 0dB when meter is switched for gain reduction
Fast LED Output Meter	Indicates both L-R spectral content and output level -30dB to +30dB
Output Level	Stereo output level control -20dB to +6dB detent at 0dB
Inputs and Outputs	
Left and Right Line Inputs	Balanced XLR for line level signals to +36dB maximum
Left and Right Line Outputs	Balanced XLR DC coupled, capable of +30dB into 600 Ω

Avalon systems utilize 100% discrete, high-bias pure Class A audio amplifiers. These pure Class A amplifiers are incorporated into minimal signal path designs which include sealed silver relays, balanced DC coupled high-current "outside-world" drivers, fully discrete linear DC regulators, low noise toroidal power transformers and selected high quality active and passive components. This powerful combination of design features gives increased musical headroom and greater dynamic transient capability for a truly transparent audio passage.

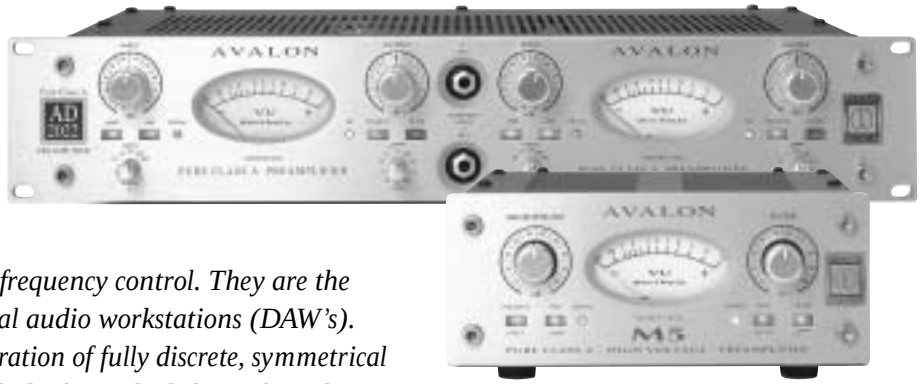
Avalon microphone preamplifiers have been in use around the world since 1986. All preamplifiers share the combined technologies of low ratio transformer balanced inputs coupled with high-voltage cascade FET twin bipolar discrete class A amplifiers.

- Pure Class A Discreet Amplifiers - no op amps
- Bipolar symmetrical design - mirror image components
- Conductive plastic potentiometers
- 100% discrete, Pure Class A signal amplifiers, practical user features and rugged hardware designed to deliver true high performance audio for many years

AVALON

M5 • AD2022

Mono & Dual Mono Pure Class A Preamplifiers



The AD2022 and M5 are world-class discrete Class A preamplifiers designed to realize the full potential of all microphones as well as electric instruments. Both preamps feature advanced high-performance transformers that deliver smooth detailed highs, delicate liquid midrange and extended low frequency control. They are the perfect front-ends for analog tape and digital audio workstations (DAW's). The AD2022 represents Avalon's fourth generation of fully discrete, symmetrical Pure Class A microphone preamplifiers with dual matched channels and several additional features, including selectable microphone source loading for optimized cable/mic matching, ±3dB fine output trim controls, ultra high operating headroom +36dB as well as balanced and unbalanced outputs.

FEATURES

Symmetrical Pure Class A Design

- 100% discrete, high-voltage, symmetrical Pure Class A signal amplifiers are utilized for each channel — The first amplifier operates in a cascaded-cascode FET configuration while the second stage amplifier is fully bipolar. The twin amplifiers share the total available gain requirement. This shared architecture increases the high level headroom capability, improves transient response and doubles the internal bandwidth of the ultra-high performance pure Class A signal amplifiers
- The most advanced high-performance mic transformer available — Split low-ratio primary windings are carefully combined within a custom mu-metal core for extended smooth frequency response and very low distortion

Input Stage

- Input gain is variable from +20dB to +64dB in 4dB steps
- A switchable passive high pass filter, (6dB/oct, variable from 30Hz to 185Hz)

Microphone Input

- -20dB resistive pre transformer input attenuator (pad), +48v regulated phantom power (50mA capability) and a Polarity reverse switch are available on the microphone input
- To ensure minimal signal invasion, sealed silver relays are used for all signal routing including the input signal selector, high pass filter, polarity reverse, +48v power and the 20dB pad

Input Source Load Selector (AD2022 Only)

- A 5-position rotary switch, on each of the channels, enable you to change the preamplifier's input impedance between five different input impedance values: (DI), 50, 150, 600 and 1.5k Ω (MIC)
- This transformer balanced input load selector is provided as a "creative tone shaper" that can enhance the sonic signature of your microphones and cables — As you lower the input impedance you place a load on the microphone and cable the effect of which can create subtle to dramatic changes in frequency and distortion characteristics as well as filtering effects due to cable capacitance

Inputs

- XLR mic inputs are electronically balanced with a maximum input level of +36dB (with -20dB pad)
- 1/4″ Hi-Z instrument DI is provided with a maximum input level of +18dB at 100k Ω

Metering

- Large, high-quality, illuminated analog VU meters provide accurate signal readings
- Bi-color LED's offers peak indication – green at 0dBu and red at +20dBu

Output Section

- High-current DC coupled, Class A discrete amplifiers are provided for the outputs of both the M5 and AD2022
  - ➢ The M5 features an XLR unbalanced output with +30dB capability into 600Ω
  - ➢ The AD2022 features an XLR balanced output with +36dB capability into all known loads
- The AD2022 also features a ±3dB variable fine output gain adjustment that can be used for gain riding during tracking

Power Supply

- The external B2T toroidal AC power supply is switchable for 100V to 240V, 50-60Hz operation, 150 watts maximum

B&H PHOTO - VIDEO - PRO AUDIO EQUIPMENT LEASING AVAILABLE

M5 • AD2022

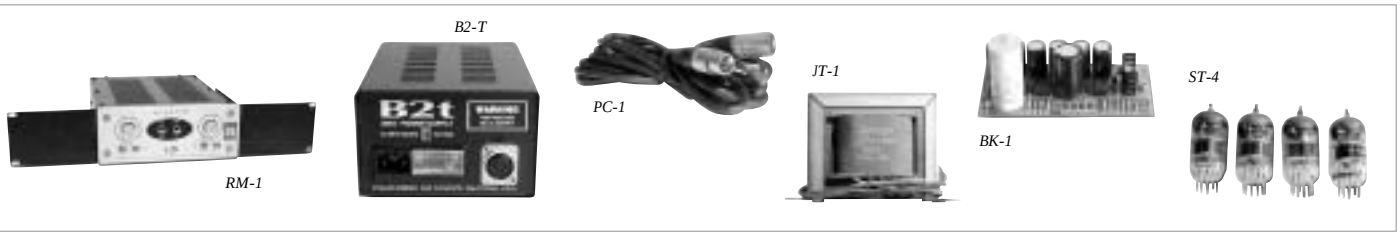
Specifications	M5	AD2022
	Preamp Controls	
Input (Gain Control)	+20dB to +64dB in 2dB steps	Variable from +20dB to +64dB in +4dB steps
Output Level Control	–	Variable ±3dB (Fine Adjustment)
	Passive High Pass Filter	
Filter Frequency Control	6dB per octave continuously variable from 30Hz to 185Hz	
Filter (In)	Engages the high pass filter	
	Microphone Input Only	
Switchable Phantom Power	+130v and +48v regulated 50mA capability	+48v regulated 50mA capability
-20dB Switchable Pad	Resistive pre-transformer	
Input Source (impedance)	Transformer balanced low-ratio 1.5k Ω (fixed)	Transformer balanced low-ratio, 50, 150, 600 and 1.5k Ω (MIC)
Switchable Polarity Reverse	✓	✓
	Metering	
VU meter	High quality, illuminated analog VU meter indicates the output level in dB. Calibrated: +4dBu = 0 VU	
Output Peak LED	Bi-color LED flashes green at 0dBu (peak) and red at +20dBu (clip)	
DC (Power Indicator)	Blue LED indicates the DC power is operational.	
	Inputs and Outputs	
1/4" Hi-Z Instrument DI	Unbalanced, +18dB at 100k Ω	
XLR-Balanced Mic Input	+36dB (with -20dB passive attenuator)	
XLR Unbalanced Line Output	DC coupled, +30dB max. output into 600 Ω	
XLR-Balanced Line Output	–	DC coupled, +36dB maximum into 600 ohms
B2T Power Connection	XLR-4 cable connects to the power connector of the external	
B2T power supply	External B2T toroidal 100V to 240V, 50-60Hz selectable, 150 watts max	

M5 System Performance

Frequency Response -3dB	1 to 120k Hz transformer limit
Noise EIN Unweighted	-126dB 150 ohm.
Noise 20kHz Unweighted	-100dB
Distortion THD, IMD	0.05% nominal
System Bandwidth	DC to 1MHz

AD2022 System Performance

Frequency Response -3dB	1 to 120k Hz transformer limit
Noise EIN Unweighted	-126dB 150 ohm.
Noise 20kHz Unweighted	-102dB
Distortion THD, IMD	0.05% nominal
System Bandwidth	DC to 1MHz



RM-1
Converts a single U5 or M5 to a standard 19″ rack system. Includes two steel rack ears and eight stainless steel screws **69.95**

RM-2
Joins two U5 or M5's together in an standard 19″ rack. The RM-2 contains two stainless steel rack ears, two T-bars, and ten stainless steel screws **99.95**

VP-1
1U 19″ vent/spacer panel, ideal for cooling large racks.....**79.95**

B2-T
External AC power supply 100v-240v selectable **99.95**

PC-1
Four pin, low voltage power supply cable for AD2000 series 8′ length **69.95**

JT-1
High performance Jensen (JT-11-BM) output transformer option (M5 only) **69.95**

BK-1
Special 130v option power supply plug-in card for B&K microphones (M5 only).....**149.95**

ST-4
Spare tube kit for VT-737SP and VT-747SP. Includes 4 matched military 6922 tubes.....**79.95**

AVALON

AD2044

Dual Mono-Stereo, Pure Class A Opto-Compressor



The AD2044 combines Avalon's legendary 100% discrete, pure class A signal amplifier design with custom designed high speed current monitored optical control elements. These "invisible" opto elements enable the AD2044 to deliver totally transparent, low noise gain reduction within the minimalist signal path design. Variable attack and release controls plus side chain access, provide unlimited creative control from soft compression to hard limiting. The perfect solution for two buss music-program compression, mono bass, instrument and mastering applications.

FEATURES

Sonic Excellence

- Combines advanced electronics with unequaled sonic excellence — The advanced true symmetry design offers high-voltage, large headroom, extended bandwidth and very low noise
- State-of-the-art, balanced 100% discrete, Pure Class A signal amplifiers offer the serious music professional unlimited sonic character and a natural harmonic detail that enhances the program material and becomes one with the music itself
- Low noise Pure Class A optical-compressor designed to optimize absolute signal integrity and musical performance
- Practical "real-world" user features and rugged hardware designed to deliver true high performance audio for many years
- The perfect solution for two buss music-program compression, mono bass and instruments and mastering applications

Linear Optical Compression

- Custom manufactured, high speed linear opto elements are used to create a non invasive passive attenuator system that ensures transparent gain reduction
- A unique external control loop and current monitored optical driver is incorporated that provides smooth control plus the benefits of traditional vintage LDR (Light Dependent Resistor) compression

Compression Controls

- Dual mono or stereo operation via Stereo Link switch
- Fully variable threshold, ratio, attack and release controls
- Hard-wire relay bypass for compressor in-out

Inputs & Outputs

- XLR-balanced inputs and outputs
- XLR-balanced sidechain input for spectral dynamics control

Metering

- Fast acting blue peak compression LEDs
- Large professional VU meters

Opto-Compressor Controls	
Threshold	Variable -24dB to +20dB
Ratio	Variable 1:1 to 20:1
Attack	Variable 0.5mS to 150mS
Release	Variable 80mS to 5 seconds for 12dB release
Bypass	Hard-wire relay bypass for compressor in-out
Link Switch	Large illuminated push switch for stereo L-R linking
Make Up Gain	Variable 20dB range (±10dB), center detent 0dB
Side-Chain (In)	Engages side-chain input
Switchable VU Metering	
Gain Reduction Meter	Illuminated gain reduction meter 0dB to -20dB range
Output Meter	Professional moving coil type -20dB to +3dB (0VU=+4dB)
XLR Balanced Inputs and Outputs	
Maximum Input Level	+30dB balanced
Maximum Output Level	+32dB balanced 600 ohms, DC coupled
Side-Chain Input	XLR balanced
Power Supply	B2T External AC supply, 150w toroidal transformer, 4 pin cable 90v isolated, 100-240v selectable 50/60Hz, 150w max

System Performance

Noise 20kHz Unweighted: -94dB (compressor in)
Distortion THD, IMD: 0.5% (typical 0.05% at +6dB 1kHz
Frequency Response: -3dB 1Hz to 450kHz (input band limited)

AD2055

Dual Mono, Pure Class A Parametric Music Equalizer



The Avalon AD2055 is a dual mono, four-band parametric equalizer combines 100% discrete, pure class A signal amplifiers with state-of- the-art passive and active filter topologies to give the serious music professional unlimited sonic character and a natural harmonic detail that is sure to enhance any program material. The AD2055's unique circuitry will breathe life into musical performances offering very high resolution transient detail at the operational extremes with very low noise at all settings. Avalon's true symmetry design offers high-voltage and high headroom (+32dB output) with an extended bandwidth from 1Hz all the way up to 500 kHz.

FEATURES

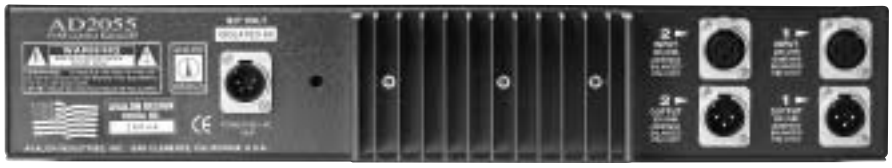
- Extremely musical and easy to use dual four band equalizer
- State-of-the-art, balanced 100% discrete, Pure Class A amplifiers
- High resolution transient detail
- Ultra low noise (-94dB) design optimized to provide absolute signal integrity and musical performance at all settings.
- High headroom +30dB
- Fully balanced DC inputs and outputs using XLR connectors
- Auto bias DC servo loop control eliminates the need for all interstage capacitor coupling
- Conductive plastic potentiometers for low noise
- All signal routing with sealed silver relays
- Perfect for stereo buss music-program equalization, special instrument EQ and FX applications as well as ultra high performance mastering studio applications
- External 150W torroidal BT-2 power supply provides clean, hum-free power (100-240v selectable 50/60Hz,)

System Performance

Noise 20kHz Unweighted -94dB (EQ in)

Distortion THD, IMD
0.5% (typical 0.05% at +6dB 1kHz)

Frequency Response -3dB 1Hz to 450kHz (input band limited)



Transparent Passive-Active Filter Design

- Passive high and low bands plus two fully parametric mid bands deliver truly musical performance with high speed transient detail at the operational extremes
- The passive high and low bands are powerful and full-bodied with up to ±32dB of boost/cut for the low band and ±26dB of boost/cut for the high band and 10 frequency selector switches chosen for their sweet sound and musical performance
- Full bandwidth twin mid bands provide variable frequency selection (X10 frequency multipliers for very wide range), variable Q (width) and amplitude control

Passive High and Low Bands Plus Two Fully Parametric Mid Bands	
Bypass	Hard-wire relay bypass for equalizer in-out
Low band F1	Passive, ±32dB boost/cut, shelf or peak-dip curve
F1 frequency range (Hz)	Switched 10 position 18, 25, 30, 50, 72, 100, 150, 215, 300, 450
Mid band F2	Active, ±16dB boost/cut, peak-dip curve
F2 frequency range	Variable 35Hz to 450Hz (x10) 350Hz to 4.5kHz, Q (width) 0.3 to 3.0
Mid band F3	Active, ±16dB boost/cut, peak-dip curve
F3 frequency range	Variable 160Hz to 2k0Hz (x10) 1k6Hz to 20kHz, Q (width) 0.3 to 3.0
High band F4	Passive, ±26dB boost/cut, shelf or peak-dip curve
F4 frequency range (Hz)	Switched 10 position 1.5k, 2.5k, 3.5k, 5k, 7.2k, 10k, 12.5k, 15k, 20k, 25k
XLR Balanced Inputs and Outputs	
Maximum Input Level	+30dB balanced
Maximum Output Level	+32dB balanced 600 ohms, DC coupled
Power Supply	B2T External AC supply, 150w toroidal transformer, 4 pin cable 90v isolated, 100-240v selectable 50/60Hz, 150w max

AVALON

U5

Mono Instrument DI & Preamplifier

The U5 is a high-voltage DI-preamp that combines a unique passive tone selector with a variable gain preamp and filter. The variable gain Class A preamplifier is designed to optimize low level pick up signals with absolute signal integrity. The high input impedance accepts a wide variety of signal levels and instruments from acoustic guitars to high-output active bass guitars and keyboards. A high level speaker input is also provided for capturing the live sound from an instrument amplifier.



FEATURES

Input Stage

- State-of-the-art, 1/4″ high impedance (3,000,000 Ω) input stage enures zero load effect on sensitive pickups and keyboards
- A 1/4″ high level (400 watt) speaker input on the rear panel, is provided for capturing the live sound of an instruments amplifier
- A variable gain Class A preamplifier can boost low level pickup signals to +30dB
- A high-cut switch eliminates unwanted acoustic pickup and high frequency noise
- A rotary Tone control switch is provided with six passive EQ curves specifically designed to enhance a variety of acoustic and electric instruments including bass, guitar and keyboards
- The active-to-thru switch selects either the instrument input directly or sends the boosted-equalized signal to the amplifier via the front panel jack

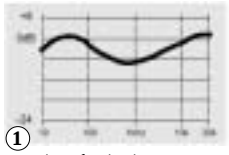
Outputs

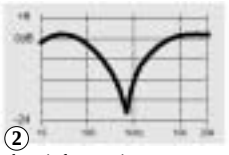
- A headphone monitor amp with a 1/4″ output connector is included for personal listening
- Twin DC coupled Class A output amplifiers drive both low level microphone preamp inputs and high level +4dB inputs for direct to tape recording or processing
- A rear panel ground isolation switch eliminates any possible earth loop or AC buzz problems with complete safety

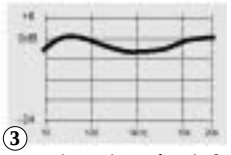
Deep and Controlled Bass

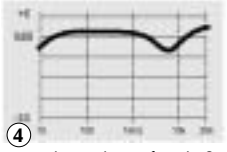
Avalon U5 DI-preamps have been in use around the world since 1995. They have found their way into the world's leading musicians rigs and recording sessions and been featured on thousands of the best selling (and sounding) albums and live performances.

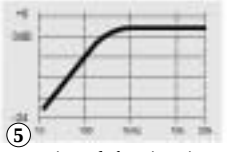
Six Position Tone Selector
This rotary switch lets you choose from 6 passive filters optimized for use with a variety of acoustic and electric instruments.

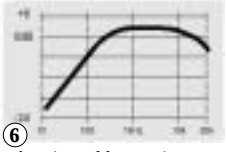

① *acoustic and string instruments, electric bass and keyboards*


② *electric bass guitar*


③ *acoustics, strings, electric & bass guitar and keyboards*


④ *acoustics, strings, electric & bass guitar and keyboards*


⑤ *acoustic and electric guitar*


⑥ *electric and bass guitar*

Gain	Switched 3dB steps, maximum gain +30dB
Switchable High Cut Filter	-3dB at 8kHz minimum phase design
Tone Selector	Six position rotary switch, all passive filters
Inputs and Outputs	
Instrument Input	1/4″ high impedance DI
Maximum Input Level	+24dBu unbalanced, 400w speaker
Mic Output	150Ω Electronically-Balanced XLR
Line Output	600Ω Electronically-Balanced XLR
Maximum Output Level	+30dBu DC coupled, balanced 600 Ω (mic and line out)
Headphone Output	0.5w into 600 ohms
Ground Lift Switch	
AC Power	Internal toroidal AC supply

System Performance

Noise 20kHz Unweighted	-100dBu minimum gain position
Distortion THD, IMD	0.1% at +10dB
Frequency Response	5Hz to 100kHz (±0.5dB)
Frequency Response (-3dB)	1Hz to 500kHz (input band limited)
Rack mount kit (1) RM-1	Single U5 19″ rack mount kit (holds one half rack unit)
Rack mount kit (2) RM-2	Dual U5 19″ rack mount kit (holds two half rack units)

MIC2200 ULTRAGAIN PRO

2-Channel Tube Mic Preamp/Line Driver/DI Box



The MIC2200 is the ideal extension to your console, MIDI setup or hard disk recording system. Integrated EQs give you extra sound-shaping power, while its built-in tube adds warmth and transparency to your signal. Also features direct injection and level conversion functionality.

- Mic input stages are based on high quality, discrete conjugate transistor pair circuitry
- Ultra-wide bandwidth from 2Hz to 200 kHz
- Built-in high-quality vacuum tube for outstanding, ultra-musical tube sound, warms up your music without unwanted noise
- Two parametric EQs with user adjustable center frequency, bandwidth and level
- Independent line driver to convert -10 dBV into +4 dBu pro level
- Completely versatile DI-Box due to servo-balanced inputs and outputs
- Tunable and switchable 12dB high-pass filter
- 'Soft Mute' 48v phantom power avoids power-on "thumps"; Phase Reverse switch
- 12-segment LED metering for output level
- Servo-balanced gold-plated XLR and 1/4″ TRS inputs and outputs

T1953 TUBE ULTRAGAIN *2-Channel Tube Mic/Line Preamp*



The T1953 has everything you need in a high-quality preamp; discrete input stages, input gain adjustment, phantom power, sweepable high-pass filter and a phase inversion switch. Plus you can add exactly the amount of tube warmth you want—without adding noise or sacrificing audio quality.

- Mic input stages are based on high quality, discrete conjugate transistor pair circuitry. Gain is adjustable from +10 to +60 dB
- Ultra Tube circuitry warms music without unwanted noise; "Warmth" control lets you add the amount of tube sound you want.
- High pass filter can be swept from 15Hz to 350Hz to eliminate floor rumble from mics and to tighten up tape tracks in line mode
- High quality detented potentiometers and switches with authentic vintage style knobs; stylish "retro" design with polished front panel
- Ultra-wide bandwidth from 18Hz to 30kHz
- Back-lit, analog VU and "warmth" meters; accurate metering for output level
- Servo-balanced gold-plated XLR and 1/4″ TRS inputs and outputs.
- 'Soft Mute' 48v phantom power; phase reverse

VX2496 ULTRA-VOICE PRO *Mic Preamp and Voice Processor*



The VX2496 is a channel strip designed to meet all your vocal processing needs. With a mic preamp, compressor, expander/gate, EQ, de-esser and tube emulation in a single rack space, it's not only the perfect tool for direct-to-disk recording, but also for demanding live situations. It also offers AES/EBU output with selectable 44.1/48/88.2 and 96kHz sampling rates or external clocking.

- Discrete ultra low-noise mic/line input stage with soft mute 48v phantom power
- Authentic tube emulation circuitry for typical tube and tape saturation sounds
- RMS expander for smooth noise reduction
- Opto compressor for inaudible level control and creative sound processing
- Integrated dynamic enhancer replaces high frequencies lost through compression'
- EQ specially designed for voice enhancement
- Opto de-esser for removal of excessive sibilance from your vocal track
- Accurate LED metering; precision potentiometers; illuminated switches
- Servo-balanced, gold-plated XLR and 1/4″ TRS inputs and outputs

BEHRINGER

MDX1600 • MDX2600 • MDX4600

Autocom Pro-XL • Composer Pro-XL • Multicom Pro-XL



Behringer's Pro-XL Series feature several innovative circuit designs that place it at the forefront of dynamics processing technology—at home in the studio or live. The MDX1600, MDX2600 and MDX4600 feature Behringer's revolutionary IKA (Interactive Knee Adaptation) circuit that successfully combines the "hard knee" compressor circuit with the "soft knee" characteristic. They implement the IGC (Interactive Gain Control) peak limiter that cleverly combines a clipper and program limiter. You'll find the IRC (Interactive Ratio Control) expander that automatically changes its ratio settings with the program material for ultra-musical and inaudible noise suppression. A switchable low contour filter prevents "pumping", while Behringer's IDE (Interactive Dynamic Enhancer) specifically makes up for the compression-induced loss of treble energy—assuring brilliant, lively audio even with heavy compression.

- Low contour filter prevents "pumping" due to low frequency dominated compression
- Stereo couple function with independent output level settings
- Low-noise amplifiers and state-of-the-art THAT VCAs guarantee best audio possible
- High-quality detented ALPS potentiometers and illuminated switches
- Servo-balanced, gold-plated 1/4″ and XLR inputs and outputs (switchable for +4 dBu or -10 dBV operation)
- Separate 12-segment (8-segment on the MDX4600) LED meters for input/output levels and gain reduction.
- Interactive Dynamic Enhancer (adjustable on the MDX1600)
- Automatic or manually adjustable attack and release times (MDX1600/2600 only)
- Switchable side chain input with side chain monitor function (MDX1600/2600 only)
- The MDX1600 has a de-esser while the MDX2600 has a switchable VAD (Voice Adaptive) de-esser that differentiates between male and female voices.
- The MDX2600 incorporates ATS (Authentic Tube Simulation) to simulate tube circuitry for extra warmth and transparency.

T1952 TUBE COMPOSER

2-Channel Tube Compressor/Gate/Peak Limiter



The T1952 features an IKA compressor/limiter with switchable Auto function for automatic attack and release time adjustment, IRC expander/gate and IGC program/peak limiter, selectable compression modes, switchable side chain input, stereo link, selectable operating level – and adds 12AX7 tubes and Ultra Tube circuitry, allowing you to sweeten and personalize your sound by adding the desired amount of harmonics.

- The desired effect of warm, silky enhancement is usually generated near the level of saturation of the tube circuit. That's when the the additional harmonics are generated that give tubes their distinctive sound or personality. Behringer's Ultra Tube circuitry overcomes the level of dependencies traditionally associated with tube circuitry while eliminating virtually all hiss, noise and hum.
- "Warmth" control smoothly blends the additional harmonics into the signal without having to maintain a specific amplitude as you normally would with traditional vacuum tubes. This allows the desired sound to be achieved without running the risks of overheating or breakdown that occur when tubes are driven at maximum levels. Dial in as much tube sound as you want.
- Highly accurate metering for input/output level and gain reduction

DYNAMIC PROCESSORS

	MDX2600	MDX1600	MDX4600
Channels	2	2	4
Stereo couple	Yes	Yes	Yes
Servo-balanced input / output	Yes	Yes	Yes
1/4" TRS + XLR connectors	Yes	Yes	Yes
Hard bypass	relay	switchable	switchable
Low contour filter	Yes	Yes	Yes
Ext. side chain	Yes	Yes	—
In / out meter	12-digit LED	12-digit LED	8-digit LED
Voice-adaptive de-esser	adjustable (Male/female switch)	auto	—
Dynamic Enhancer	auto	adjustable	auto
Tube simulation	Yes	—	—
Compressor / limiter			
Envelope	auto + manual	auto + manual	auto
Characteristic	IKA / “hard knee”	IKA / “hard knee”	IKA / “hard knee”
Gain red. meter	12-digit LED	12-digit LED	8-digit LED
“Traffic light” threshold indicator	Yes	Yes	—
Expander / gate	Yes	Yes	Yes
Fast / slow release time	Yes	Yes	Yes
Switchable exp. / gate function	Yes	Yes	Yes
Threshold LED indicator	Yes	Yes	Yes
Peak limiter	Yes	Yes	Yes
Limit LED indicator	Yes	Yes	Yes

DSP9024 ULTRA-DYNE

Digital 2-Channel Tube Compressor/Limiter/Gate /Peak Limiter



The ultimate 6-way Multiband Dynamics Processor for analog and digital mastering and sound reinforcement systems, the DSP9024 has more functions than you can imagine. Sound spectrum split into six separate frequency bands. Edit effect parameters with each band or globally. Parameter settings can be stored, copied, etc. Smooth compressor, super-clean gate, incredibly transparent Noise Reduction system and rich Multiband Exciter. Tube emulation and an integrated delay for “look-ahead” parameter adjustment. Finally, interactive signal analysis and unique automatic functions enable you to get results fast, whether you need multiband compression, loudness maximization, de-essing or selective gating. Add digital inputs/outputs with the optional AES/EBU interface.

- 24-bit A-D/D-A converters for ultra-high dynamic range and resolution of detail with selectable sampling rate of 44.1 or 48 kHz
- Sophisticated tube emulation with selectable tube types (12AX7 or EL34)
- 6-band compressor/limiter with gate and peak Limiter for “inaudible” compression.
- 6-band Noise Gate with IRC for ultra musical performance.
- Ultramizer adjusts output level and signal density for maximum perceived loudness
- “Look Ahead” automatic parameter adjustment via internal 600-msec. delay
- 3-band Harmonics Exciter with user-definable balance
- Full MIDI parameter and snapshot control allow for real time editing
- Level peak meter with peak hold and selectable levels (+4 dBu, -10 dBV, Dig Max)
- 100 settings can be stored under any alphabetic name giving you instant sound results for numerous standard applications
- Extremely flexible Stereo Link, Band Link & Clone functions
- Large high-resolution LCD graphic display
- Relay-controlled hard-bypass with an auto-bypass function during power failure
- “Virtuoso” function for super-easy, program dependent and self learning program setup
- Gold-plated XLR and 1/4″ RF-suppressed, servo-balanced inputs and outputs
- Optional 24-bit AES/EBU interface for digital inputs/ outputs at 32, 44.1 and 48 kHz

XR2000 INTELLIGATE • XR4400 MULTIGATE PRO



2- and 4-Channel Expander/Gates

The XR2000 was designed to fulfill the highest demands and offers maximum signal control. Voice-over applications, creative envelope shaping, noise suppression in quiet passages... No problem, especially with the transparent user interface and "traffic light" LED displays.

The XR4400 gives you four frequency-selective UTR expanders/noise gates in a single rack space without compromising on functionality. The side chains' parametric filters allow you to fine-tune control signals so that the gate reacts only when you want it to. Separate hold and release controls and the unique FlexLink system offer additional flexibility.

UTR Gate

Slow noise gates can rob your audio's brilliance by gating signals during the attack phase. UTR (Ultra Transient Response) circuitry with ultra-fast signal detectors prevents this from happening. Attack time is very short is very short; < 10 µsec. (XR4400) and <3 µsec. (XR2000). Clicks or pops at the gating point (regardless of attack speed) are also eliminated by the low leakage of the control signal into the VCA's audio signal path.

IRC Expander

IRC (Interactive Ratio Control) expander virtually eliminates all switching noises. Even at ratio values above 1:8, they sound musical without side effects.

XR2000

- Integrated high-pass and low-pass filters for frequency-selective gating
- External side chain input with "key listen" function
- Attack, hold, release and ratio controls
- Maximum attenuation 100 dB
- "Traffic light" display for threshold working point
- Precise gain reduction display with 8 LEDs
- Servo-balanced XLR and 1/4″ TRS inputs and outputs
- Relay-controlled hard bypass switch with auto bypass function in the case of power failures (safety relay)

XR4400

- Four separate expander/gate circuits
- IRC expander/UTR gate circuitries
- Extremely short attack time (<10 µsec.)
- Parametric side chain filter with monitor function
- Maximum attenuation 80 dB
- Independent hold and release controls for flexible envelope shaping
- FlexLink system for flexible master/slave configurations
- Servo-balanced XLR and 1/4″ TRS I/O
- "Traffic light" display for threshold working point and gain reduction display

PEQ2200 ULTRA-Q PRO *5-Band Parametric EQ*

The PEQ2200 is an effective musical frequency correction and sound-shaping tool with center frequency, bandwidth and amplitude adjustment for each filter. Its state-variable, constant-Q filters and parallel filter configuration ensure musical operation, while overlapping frequency bands allow up to 30 dB of level correction.



- For gentle contouring and audio sweetening as well as for room equalization
- Constant-Q principle ensure absolutely stable filter characteristics
- Sweepable High and Low Cut filters remove unwanted frequencies
- Bands are switchable in/out and fully adjustable from narrow notch filter (0.03 octave) to broadband EQ (2 octaves)
- A wide overlap between frequency bands allows for extreme cut or boost
- Cut-in delay to avoid switch-on "thumps"
- Relay-controlled Hard-Bypass with an auto-bypass function during power failure
- 12-segment LED input/output metering
- Servo-balanced, gold-plated XLR and 1/4″ TRS input and output

T1951Q ULTRA-Q PRO

Dual Channel 4-Band Parametric Tube EQ



The T1951Q is a dual-channel, 4-band version of the PEQ2200 with state-variable, constant-Q filters, four fully parametric stereo bands with center frequency, bandwidth and amplitude controls, plus the option to use the low- and high-frequency bands as shelving filters. In addition, you get 12AX7 tubes and Ultra Tube circuitry, enabling you to selectively add tube warmth to your EQ'd signal.

GEQ3102 ULTRA-GRAPH PRO

Dual-Channel 31-Band Graphic EQ

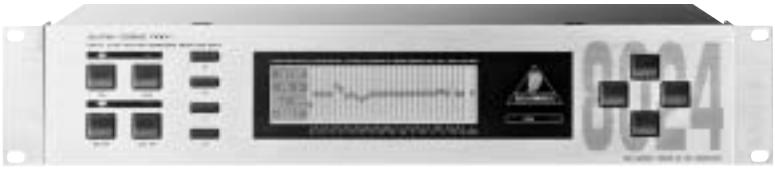


The GEQ3102 puts a powerful graphic EQ tool in your hands, with two channels of 31 standard 1/3 -octave center frequencies plus sweepable high- and low-pass filters for "pre-graphic" trimming. The dual 12-segment LED chains can display either input or output level, and each EQ section can be bypassed for quick comparisons.

- 2 x 31 ISO frequencies controllable via slide controls with boost/cut range switchable from 12 dB to 6 dB
- High-precision illuminated Alps faders
- 12-segment LED input/output metering
- High-quality detented potentiometers and illuminated switches
- Relay-controlled hard-bypass with an auto-bypass function during power failure
- Servo-balanced, gold-plated XLR and 1/4″ inputs and outputs

DSP8024 ULTRA-CURVE PRO

Digital Dual Channel 31-Band Graphic/Parametric EQ



The DSP8024 features a 31-band graphic EQ with a real-time analyzer and an Auto-Q function for automatic room measurement and correction plus three bands of parametric equalization. Additional features include a peak limiter, an adjustable delay of up to 2.5 seconds (selectable in milliseconds, meter and feet), a noise gate and Behringer's renowned Feedback Destroyer. You can also add digital inputs/outputs with the optional AES/EBU interface (AES8024).

- High-end crystal 24-bit A-D/D-A converters for ultra-high dynamic range and resolution
- Ultra-musical 31-band graphic EQs with "True Frequency Response" characteristics
- Low / high / bell shelving tool with variable slope (3 to 30 dB)
- Precise real-time analyzer with peak hold, variable integration, cursor read-out and 10 user-memories for room equalization using mic input and internal noise generator.
- 6 bands of fully parametric equalizer/Notch Filter with up to 1/60th octave bandwidth
- "Feedback Destroyer with with intelligent signal analyzer for fast feedback suppression
- Integral digital Limiter protects against clipping and dangerous sound pressure levels
- Integral digital Noise Gate with Behringer's unique IRC (Interactive Ratio Control)
- Peak level meter with peak hold and selectable reference levels (+4 dBu /-10 dBV /Dig Max)
- Full MIDI parameter and snapshot control
- 100 user-memories
- EQ and analyzer curves may be copied, compared, added or subtracted
- Crossfade between two settings and stereo link facility to synchronize both channels
- Relay-controlled hard-bypass with an auto-bypass function during power failure
- Servo-balanced XLR and 1/4″ inputs/outputs

DSP1400P ULTRAMIZER PRO

Digital 2-Band Loudness Maximizer/Program Enhancer



The digital 2-band concept of the ULTRAMIZER PRO allows separate, program-adaptive compression and limiting for low and high frequencies, ensuring ultimate punch and loudness. Its surround processor and 2-band enhancer give your mixes that extra gloss, while the noise reduction system keeps things clean. Full MIDI capability allows real-time parameter control and program selection.

- Doubles the loudness of recordings or reinforcement systems without any distortion
- Maximizes signal energy with absolutely "inaudible" and transparent compression
- Variable band-split compression eliminates gain modulation effects like 'bass pumping'
- Leveler provides constant average output while maintaining instantaneous dynamics
- Denoiser and Exciter for noise-free and ultra transparent sound
- 3-D stereo sound processor provides spatial enhancement and improved stereo imaging
- 8-segment LED level and gain reduction meters for optimum performance
- Multiband brickwall limiter
- 50 user preset memories
- Servo-balanced, gold-plated XLR and TRS inputs and outputs



T1954 TUBE ULTRAFLEX

2-Channel Tube Multiband Sound Enhancement System

The T1954 features a flexibly adjustable enhancer/exciter circuit for high-frequency processing, a bass processor with a special LC coil/cap filter for extra low-end punch and definition, and a surround processor for stereo image improvement. 12AX7 tubes and Ultra Tube circuitry let you add the silky brilliance and warmth of tubes to your audio.

- LC coil/cap filter produces authentical vintage "soft" and "tight" bass sounds
- High-quality detented potentiometers and switches with authentic vintage style knobs
- Huge back-lit analog VU meters and "Retro" design

EDISON EX1 *Stereo 3-D Processor with Correlation Meter*



The EDISON EX1 is an unbelievably powerful psycho-acoustic processor, which allows you to completely manipulate all spatial parameters of the stereo image – such as width and depth – in a way never before possible. You can widen or narrow the stereo spread, bring instruments to the front or move them to the background, and make corrections to the stereo image – even on a finished master. All with five easy-to-use knobs. Whether PA, studio, Hi-Fi or audio for video: everything sounds more natural and transparent.
The integrated correlation meter guarantees fool-proof control of mono compatibility. Frequency cancellation and undesirable phase deviations in the mono and stereo content of your audio material are immediately visible.

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EX1200 ULTRABASS PRO

2-Channel Subharmonic Synthesizer



The ULTRABASS PRO generates musically consequential subharmonics that can be set to one or two octaves below the lowest frequencies present in the original signal. Based on waveform analysis, the generated tones harmonize perfectly with the original sound, giving you big, fat low end, while the integrated limiter effectively protects your system from overloading.

- Adds unbelievable bass power to sound system, perfect for discos, clubs, theatres, sport and aerobic studios and your HiFi system
- Digital synthesis based on waveform analysis generates ultra-low frequencies for driving subwoofers
- Waveform Processor musically restores low frequencies lost in the recording process
- Releases untapped sonic resources and adds power to instruments, vocals and mixed program material
- Dynamic Punch control adds breathtaking kick bass to your program material
- Switchable x-over splits off dangerous low frequencies from your club system
- Built-in Limiter
- Bass Mode control allows you to fade over from “ultra-low” to “punchy” bass sounds
- Separate subwoofer output
- Output level and gain reduction LEDs
- Servo-balanced gold-plated XLR and 1/4″ TRS inputs and outputs

EX2200 DUALFEX PRO • EX3200 ULTRAFEX PRO

2-Channel Multiband Spatial Sound Enhancement Processors





Geared for use in smaller setups or clubs, video or Hi-Fi systems, the EX2200 and EX3200 are among the most popular psychoacoustics devices in the world. Features include a “Natural Sonic” processor that guarantees aural improvement and high-end transparency, and the integrated bass processor that works wonders for your audio’s low end. In addition, their adjustable surround processor allows flexible widening of the stereo spatial effect.
The EX3200 adds a Noise Reduction system, hard bypass and XLR connectors.

- Gives your music that extra sparkle and makes your instruments or mixes stand out
- Releases untapped resources and details instruments, vocals and mixed program material
- Multiband concept for bass power and high frequency transparency
- “Natural Sonic” processor for ultra-musical sound improvement
- VSP (Variable Sound Processing) circuit for simultaneous Enhancer and Exciter process
- “Dual Mode” ultra-bass enhancer produces “soft” and “tight” bass sounds
- Surround processor provides real spatial enhancement and improved stereo imaging
- High-quality detent potentiometers and illuminated switches
- Servo-balanced 1/4″ TRS and RCA inputs

EX3200 Step-up Features

- Shift function allows you to control the frequency range of added bass punch
- Built-in Noise Reduction system with LED
- Solo facility for effect loop application
- Relay controlled Hard Bypass with an auto bypass function during power failure
- Servo-balanced gold-plated XLR and 1/4″ TRS inputs and outputs

MULTI-EFFECTS PROCESSORS

Modulizer Pro (DSP1224P)



The DSP1224P puts tons of trendy stuff at your fingertips, like 8-voice chorus, spatial flanger, ring modulator and resonator. There are also LFO- and MIDI controllable moduleable filters with resonance, developed using physical modeling with analog filters as archetype as well as amp and speaker simulations, a 3D processor and much more.

- True stereo processing for open-sounding enhancement of the sound sources
- XLR and 1/4″ TRS servo balanced inputs and outputs
- 100 user preset memories to store programs for instant recall
- Accurate eight-segment LED level meters simplifies settings
- Real-time parameter control and program selection via MIDI
- Cutting edge effects such as 3D Processor, Stereo Imager, Lo-Fi, Super Bass, Ring Modulator, Voice Canceler etc.
- 24 breathtaking effects such as Ultra Phaser, Jetstream Flanger, Auto Filter, Ultra Compressor, Guitar Combo, Speaker Emulation Resonator and more
- Hundreds of effect variations plus two individual parameters and separate low and high EQ section

VIRTUAL ROOMS

Have you ever heard an orchestra perform in a concert hall? The sound is all-enveloping, richly textured. However, did you know that the sound you hear in a concert hall or room is 90% reflected? Come into Behringer's "virtual rooms".

Some digital reverbs only give you a crude approximation of real acoustic reverb. Behringer's reverb algorithms are infinitely more complex and detailed. The have accurately analyzed the acoustic properties and taken physical measurements of real rooms, concert halls and theaters. Precise mathematical models are used to calculate the reverb algorithms. The accuracy of hundreds of parameters used in the physical modeling of real buildings which also includes the important determination "early reflections" ensure rich, dense and natural sounding reverberation. When you place an instrument inside one of Behringer's "virtual room" you don't get a muddy sound. You end up with realism, vibrancy, separation and a clarity you never thought possible from a digital device.

Virtualizer Pro (DSP2024P)



The DSP2024 Virtualizer Pro features 71 effect algorithms, giving you powerful modulation, amp simulation, distortion and special effects as well as effective dynamic and psychoacoustics processing and equalization on top of Behringer's renowned wave-adaptive Virtual Room reverb algorithms. There are also 7 adjustable parameters plus high and low EQ per effect, 11 serial or parallel effect combinations, true stereo processing in stereo image with most algorithms and an improved user interface.

DSP2024 STEP-UP FEATURES

- Over 70 algorithms and special effects including Virtual Room reverb algorithms. Also includes dynamic and psychoacoustics processing algorithms, distortions, amp and tube simulations, sampler, vocoder and much more.
- 11 effect combinations using modulation effects like chorus, flanger, pitch or tremolo combined with reverb or delay algorithms.
- 7 parameters per preset plus separate high and low EQ section.

DSP1124P Feedback Destroyer Pro

2-Channel Digital Feedback Suppressor/Parametric EQ



Powered by a 24-bit high-speed DSP the DSP1124P locates feedback frequencies immediately and sets extremely narrow notch filters to "destroy" them, leaving the remainder of the signal virtually untouched. With its various modes you can master just about any live sound situation or use it as a creative sound-shaping tool with 24 bands of fully parametric equalization.

- Automatically and "intelligently" finds and destroys up to 12 frequencies per channel
- "Set-and-forget" default setting enables immediate Feedback Destroyer performance
- Single-Shot mode automatically searches and destroys feedback and locks the filter until manually reset
- Auto mode continuously monitors the mix, resetting programmed filters automatically
- Select single-shot, auto or manual modes for each filter
- Manual mode allows individual settings of up to 2 x 12 fully parametric filters with frequency, bandwidth and gain
- 24-bit A/D and D/A converters
- Two software engines allow independent or coupled functions on left and right channels
- Accurate eight-segment LED meters
- Full MIDI capability
- Servo-balanced XLR and 1/4″ TRS I/O

DSP110 Shark

Digital 24-Bit Multifunction Signal Processor

An all-round problem-solver, the DSP110 combines mic/line preamp (60 dB gain control, +48v phantom power) with a variety of signal processing options, including Feedback Destroyer, variable delay line for speaker time-alignment (up to 2.5 seconds of delay, adjustable in msec, meters or feet), ultra-musical compressor with adjustable density and speed parameters, and an extremely fast automatic noise gate with learn function and optional manual setting. Add 24-bit A/D and D/A converters, servo-balanced XLR and 1/4″ inputs/outputs, mic/line level conversion, external power supply for maximum signal integrity and headroom, and you can use the DSP110 in just about any application.



- Status LEDs for all 12 filters
- 4-digit multi-function display
- 6-segment clip level meter
- Subsonic filter with adjustable cut-off frequency
- 19″ rack-mounting kit included with purchase of 5 DSP110s

SNR2000 Multiband Denoiser



2-Channel Single-Ended Noise Reduction System

With true multiband detection in the side chain, IRC noise gates and TAC filter circuitry, the SNR2000 ensures smooth and effective noise reduction with minimal loss of audio integrity.

- Multiband frequency analysis and processing
- TAC (Transient Attack Control) dynamic filter circuitry responds to extremely fast attack signals
- Auto Filter for quick setup
- IRC (Interactive Ratio Control) downward expander "musically" removes any noise in pauses
- Noise reduction of up to 80dB
- Adjustable cut off frequency and release times meters
- Dual mono or stereo couple function
- Servo-balanced XLR and 1/4″ inputs and outputs
- Relay controlled Hard Bypass
- Cut-in delay to avoid switch-on thumps
- Illuminated switches, 41 detent potentiometers

AUDIO SOLUTIONS

Troubled by feedback despite your best and knowledgeable efforts? Wondering how to tame that unruly, time-consuming studio wiring? Crossovers not cutting it anymore? Tired of accessories big on promises and short on performance? For each of these problems and more, Behringer has an answer.
Any performance can be plagued by feedback without warning. Fortunately, Behringer has the tools required to stop the problem instantly. In the world of professional audio, every second counts. Any time spent in a rat's nest of cabling, for example, figuring out how to get things hooked up and running, is definitely time better spent doing something else. Enter the Ultrapatch Pro patchbay and Ultralink Pro splitter/mixer.

SRC2496 Ultramatch Pro

A/D and D/A, Sample Rate and Format Converter



If you frequently have to deal with various interfaces, digital standards and so on, you've probably discovered that compatibility doesn't always mean that your gear will harmonize perfectly. The SRC2496 Ultramatch Pro enables you to transfer digital audio signals between devices with different sample rates, formats or interfaces, disable SCMS for 1:1 digital copies and remove dropouts or jitter from digital media. Simultaneously operable outputs offer splitter functionality, and the 24-bit/96-kHz compatible SRC2496 offers A/D and D/A conversion as well.

- Format conversion between AES/EBU and S/PDIF (coaxial or optical)
- Converts 30 to 100 kHz sample rates into 32/44.1/48/88.2 or 96 kHz with selectable 16, 20- or 24-bit output resolution.
- Parallel A/D and D/A conversion at identical sample rate
- Universal sample rate synchronization via wordclock or digital input
- Allows direct manipulation of emphasis and copy-protection bits
- XLR, RCA and optical outputs are simultaneously operational, while inputs are separately selectable.
- Removes jitter and corrects incorrect sample rates

DI100 Ultra-DI

Ultra-Rugged Single-Channel DI Box



The DI100 puts an end to hum and impedance problems, and ensures that any imaginable sound source (electric guitar, bass or keyboards etc.) will reach your mixing console balanced and noise-free. You can even plug your guitar amp's speaker output into the DI100—it will deal with ratings of up to 3,000 watts and give you a perfect signal at the other end. In addition, OT-1 output transformers ensure clean, powerful output with ultra-flat frequency response.

- Input-buffering amp ensures clean, pure and powerful sound, even with extremely long cables
- Rugged metal housing with hard rubber corners for protection
- Internal battery gets de-coupled when you use phantom power
- Gold-plated XLR and 1/4″ TRS connectors for input and link output

DI4000 Ultra-DI Pro

Active 4-Channel DI Box



Ideal for stage and studio applications, the multi-purpose DI4000 gives you four first-class DI (Direct Inject) channels in one rack space. Capable of handling almost any kind of signal, the DI4000 is equipped with two inputs (XLR and 1 /4″ TRS), a balanced output (XLR) and a 1/4″ link output per channel, up to 40 dB of switchable input attenuation and up to 20 dB of gain. In addition, OT-1 output transformers ensure clean, powerful output with ultra-flat frequency response.

- Allows direct connection to speaker outputs rated at more than 3000 watts
- Converts unbalanced line inputs to balanced outputs
- +20 dB gain switch for pre-amplification of low level signals
- Switchable high cut filter (8 kHz) for guitar applications
- Phase reverse switch
- Ground Lift switch eliminates typical ground loop problems
- Accurate eight-segment LED meter simplifies level settings
- Gold-plated XLR and 1/4″ TRS inputs and outputs
- Switchable attenuation allows maximum input of +50 dBu

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CX2310 Super-X Pro

2-Way Stereo/3-Way Mono Frequency Crossover with Subwoofer Out



The CX2310 offers absolute precision, state-of-the-art circuitry, professional components, ultra low-noise op amps, fully balanced in and outputs and intuitive operation. In addition to full-fledged 2-way stereo/3-way mono operation, the CX2310 features a separately adjustable subwoofer output, giving you an extra low-frequency band.

- World-class performance 24 dB per octave Linkwitz-Riley filters
- Absolute flat summed amplitude response, zero phase difference
- Separate subwoofer section with independent frequency control
- Individual output level controls for all bands; individual output mutes for easy band adjustment
- Individual phase reverse switches for instant phase correction
- Switchable 25 Hz subsonic filter on each input for low-frequency driver protection
- Servo-balanced, gold-plated XLR inputs and outputs
- Precision Alps potentiometers for accuracy and repeatability
- Illuminated switches for operation in dark stage environments
- Shielded toroidal mains transformer for minimal noise

CX3400 Super-X Pro

3-Way Stereo/4-Way Mono Frequency Crossover with Limiters



The CX3400 is an active frequency crossover enabling 2 or 3-way stereo or even 4-way mono operation. Each band features two 24-dB Linkwitz-Riley filters, as well as individual IGC limiters, while the integrated delay enables flexible time alignment. Polarity switches for each output allow quick and easy runtime compensation for various driver and horn types.

Same features as the CX2310 PLUS—

- Individual limiter on each output for loudspeaker protection
- "Low Sum" function provides low-level mono output for subwoofer operation
- Adjustable time delay for phase alignment between drivers
- Switchable equalization for constant directivity horns

PX2000 Ultrapatch Pro

48-Point Switchable Patchbay with 4 Modes



The PX2000 offers the reliability and flexibility you need to connect your entire studio in a clear, orderly fashion. With its four modes, easily selectable via topside switches, the patchbay's 24 jack pairs can operate in parallel, half normalized, normalized or open mode. High-grade components include metal housing, metal-ring jacks and fiberglass boards.

MX882 Ultralink Pro

8-Channel Splitter/Mixer, Level Conditioner/Converter



This compact signal distribution wizard has helped countless musicians and engineers solve diverse problems, whether as a splitter, line mixer or level conditioner. Regardless of your application, the MX882 keeps your signal paths short and clean. And its multiple matching amplifiers allow you to convert home-recording level to professional level and vice versa. Use as effects mixer, for PA monitoring, live sound systems, theaters, conference rooms, hotels, churches, etc.

- Any channel can be selected either as mixer or splitter
- Use as 8-In/2-Out Line Mixer; as 2-In/8-Out Line Splitter; or as 6-In/6-Out Line Driver or DI-box
- Convert -10 dBV into +4 dBu or vice versa
- Extremely high headroom - offering more dynamic range
- Ultra-wide bandwidth from 2 Hz to 200 kHz for "open" sound
- 6 mono inputs/outputs, 2 main inputs/outputs with 6 input level controls, 6 balance/pan controls, a main input/output control
- Main Link switch allows routing of the main input to the Main Output to link several units
- 4 /8-segment LED metering for each individual gain section
- Servo-balanced gold-plated XLR and 1/4″ TRS inputs and outputs

COMPRESSORS / GATES

266XL Dual Channel Compressor/Gate



The 266XL brings affordable, high-quality compression and gating with intuitive operation so you can easily smooth uneven levels, add sustain to guitars, squash drums or tighten up mixes. Auto Attack and Release emulates the classic dbx compressors by continuously adjusting the attack and release times to optimally match the program material. Manual controls are also available, allowing you tailor the attack and release response specifically for individual tracks, mixed program material, special effects or system protection.

- Stereo or dual mono gating, compression
- Expander/Gate Circuit with variable release time and +15dBu maximum threshold.
- Advanced gate circuitry uses an automatic timing algorithm to produce ultra-smooth release characteristics—even with complex signals, such as voice or reverb decays
- Compressor/Limiter circuit with variable threshold and ratio controls as well as auto (program-dependent) or manual Attack and Release control.
- Selectable OverEasy/Hard Knee compression curves, depending if you're looking for "heavy" almost-special-effects compression or soft, smooth gain leveling.
- Sidechain Inserts enables an outboard processor to control compression or gating.
- RMS Level Detection senses the power in the program in a musical manner, for much better results than peak or average detection.
- 8-segment LED display for gain reduction (up to 30dB).
- In Stereo Couple mode, Channel 1's controls become the master, and Channel 2 follows precisely to ensure a rock solid stereo image, even with high amounts of compression
- Signal does not pass thru any of the parameter controls. Instead a DC voltage controls all functions, eliminating any possibility of potentiometer noise developing over time.
- Electronically-balanced XLR and 1/4″ TRS inputs and impedance-balanced XLR and 1/4″ TRS outputs

166XL Dual Channel Compressor/Gate



The 166XL steps up from the 266XL by adding two channels of PeakStop limiting to provide studios, sound reinforcement companies, musicians, or anyone who needs quality processing quickly and easily—with complete control of signal dynamics. Most compressor/gates provide less than musical compression, coupled with gating that swallows transients—or closes early, cutting off decay and reverb tails. The superb engineering in the 166XL ensures that both its compression and gating provide versatility and excellent sonic performance in situations where other compressor/gates typically produce undesirable processing artifacts.

- Stereo or Dual Mono operation of gating, compression and PeakStop limiting.
- PeakStop Limiting provides control of maximum peak levels at the a channel's output regardless of any other control.
- PeakStop comes after the compression, gating and other circuitry including the output gain, so it sets an absolute limit for peak excursions before they reach the output.
- A Sidechain enable switch on each channel lets you activate or de-activate the Sidechain inserts without repatching
- A Contour button in the Sidechain Path prevents sustained low frequency program material from dominating the compressed signal. This allows faster attack times which preserve transients and allows higher compression ratios with less artifacts.
- Hardwire System Bypass Buttons on both channels allow the audio to pass even if the unit loses power — also useful for A-Bing the processed and unprocessed signals.
- 10-segment LED display for gain reduction (up to 30dB)
- Electronically-balanced XLR and 1/4″ TRS inputs and outputs

160A Compressor/Limiter



The latest generation of the legendary dbx 160 family of compressor/limiters, the 160A offers the same classic sound, easy set-up and no compromise metering plus it has an improved signal path and stronger chassis design. Digital-ready and road tough, the 160A features switchable OverEasy and Hard Knee compression, extremely wide threshold ranges, and controls for ratio and output gain. It also includes true RMS level detection, providing the most natural-sounding dynamics processing available—from subtle compression to "brick wall" peak limiting. And with its unique "INFINITY +" inverse compression mode, the 160A actually decreases the audio output level below unity gain when the input exceeds threshold. Two 160As can be coupled to process a stereo mix without shifting the left/right image.

- Choose low ratios and OverEasy compression to transparently smooth out fluctuating vocal and instrumental levels. Or use the 160A's Hard Knee compression and high ratios (up to and beyond infinity:1 and over 60dB gain reduction) for bombproof protection against overload distortion in digital recording, PA systems and broadcast signals.
- 12-segment LED display for gain reduction up to 40dB
- 19-segment LED display for high resolution monitoring of true RMS input/output levels
- Input/output meter calibration allows the 160A to be used in a variety of situations where the 0 reference is not consistent
- Electronically balanced /unbalanced input and output stages are fully compatible with +4/-10dB. The output stage is an outstanding line driver for long cable runs.
- "INFINITY +" inverse-compression mode decreases the output level below unity gain when the input exceeds threshold — ideal for correcting overbearing vocal choruses or controlling runaway house mix levels.
- Detector input allows special applications including frequency conscious compression in which an external equalizer is used.
- Input Ground Lift Switch eliminates any ground loop hum problems that may arise.

Hard Knee vs OverEasy Compression:

When the compressor is set for Hard knee, the compression ratio applies only to signals above the threshold level. If the compressor is set for Soft knee (OverEasy), the compression ratio gradually increases from 1:1 to the current selected ratio over a range through the threshold area which allows the transition from uncompressed to compressed to be more gradual. This greatly reduces compression artifacts and allowing faster attack times and higher compression ratios while still maintaining the natural characteristics of the signal. Hard knee compression, on the other hand, is ideal for "brick wall" limiting because it stops any transients from slipping through without affecting lower level signals.

	266XL	166XL	160A
	Expander/Gate Controls		
Threshold Control Range	-60 to +10dBu	Off to +15dBu	N/A
Ratio Control Range	1:1 to 4:1	1.2:1 to 8:1	N/A
	Compressor Controls		
OverEasy Button with LED	Yes	Yes	Yes
Threshold Control Range	- 40dB to +20dB	- 40dB to +20dB	- 40dB to +20dB
Ratio Control Range	1:1 to ∞:1	1:1 to ∞:1	1:1 to ∞:1
Attack Time	Switchable; Manual and Automatic	Switchable; Manual and Automatic	Automatic
Release Time	Switchable; Manual and Automatic	Switchable; Manual and Automatic	Automatic
1/4″ TRS Side Chain Insert	Tip = Input, Ring = Output	Tip = Input, Ring = Output	
Bypass Switch w/LED	Yes	Yes	Yes (Hardwire)
PeakStop Limiter Threshold Range	N/A	0dBu to +20dBu	N/A
Gain Make-Up Range	Variable; -20dB to +20dB	Variable; -20dB to +20dB	Variable; -20dB to +20dB
Stereo Couple	True RMS Stereo Summing	True RMS Stereo Summing	True RMS Stereo Summing (with 2nd unit)

COMPRESSORS / GATES

1066 Dual Channel Compressor/Limiter/Gate



Designed to provide simple, flexible operation for recording, broadcast or live sound reinforcement applications, the 1066 steps up from the 166XL with several important features. At the heart of the 1066 is dbx's V2 VCA (voltage-controlled amplifier). The V2 offers superb dynamic range characteristics while maintaining very low distortion and almost immeasurable noise. This combined with it's world-class design, enables the 1066 to perform like compressors selling for hundreds of dollars more. The 1066 also improves on the 166XL's limiter, with a PeakStopPlus limiter. In addition to simply preventing unwanted transients from blowing your speakers or causing digital clipping while minimizing the distortion common to many other "hard" limiters, PeakStopPlus offers a two-stage limiting process for more transparent control. The 1066 also adds a Side Chain Monitor button (SC Mon) that makes setting up and adjusting the Side chain much easier by allowing you to listen to the audio source inserted into the Side Chain. Last, selectable input and output metering helps determine that everything is matched up level-wise, maximizing signal-to-noise ratio.

SAME FEATURES AS THE 166XL *PLUS*—

- RF-filtered, gold-plated electronically-balanced/ unbalanced inputs and outputs
- V2 VCA boasts superb dynamic range characteristics while maintaining very low distortion and immeasurable noise.
- 12-segment LED for gain reduction (up to 30dB)
- Side Chain Monitor (SC) function makes setting up and monitoring the Side Chain much easier
- Precision metering of input/output levels (between -24 and +18 dBu) with switchable 8-segment LED meters

PeakStopPlus Level Control

Applies fail-safe limiting using a unique two-stage process to control the output signal.

Stage 1 –
Instantaneous Transient Clamp

This is a soft logarithmic clamp function that ensures that the signal doesn't exceed the threshold set by the PeakStopPlus level control by more than 2 dB (typically) and will not introduce harsh artifacts.

Stage 2 –
Intelligent Predictive Limiting

This is a unique program limiter which monitors the input signal and intelligently predicts the amount of gain reduction needed to keep the output below the ceiling set by the Instantaneous Transient Clamp.

System Performance (same as 166XL except)

Dynamic Range	>120dB, unweighted
THD + Noise	<0.1% any amount of compression up to 40dB at 1kHz
Intermodulation Distortion	<0.1% SMPTE
Noise	<-95dBu unweighted, 22 kHz measurement bandwidth
Interchannel Crosstalk	< -100 dB, 20 Hz to 20 Hz



Input Specs	**Impedance**	**Max. Input Level**
XLR & 1/4″ TRS	>50k Ω bal/>25k Ω unbal	> +24dBu (bal or unbal)
Side Chain Specs	**Impedance**	**Max. Level**
1/4″ TS Input	>10kΩ	> +24 dBu
1/4″ TRS Output	2k Ω bal/1k Ω unbal	+21 dBu (bal or unbal)
Output Specs	**Impedance**	**Max. Output Level**
XLR & 1/4″ TRS	120 Ω bal / 60 Ω unbal	+21dBu into 600Ω

VCA	dbx V2
Side Chain Input	RF Filtered, unbalanced 1/4″ TS
Side Chain Output	RF Filtered, impedance-balanced 1/4″ TRS
PeakStop Plus Limiter	
PeakStop Plus Limiter	Two-stage limiter
Threshold Range	+4 dBu to >+22 dBu (off)
Stage 1	Instantaneous Transient Clamp
Attack and Release Time	Zero
Stage 2	Intelligent Predictive Limiting
Auto Attack Time	Program-dependent, typically <5 msec
Auto Release Time	Program-dependent, typically 22 dB/sec
Ratio	∞:1 (Fixed)
Output Gain Adjustment	20dB to +20dB
Stereo Couple Switch	Links channels as stereo pair
Bypass Switch	Hardwire bypass of all front panel controls

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COMPRESSORS/GATES

1046 Quad Channel Compressor/Limiter



Equally at home in recording studios and live sound reinforcement venues, the 1046 provides 4 channels of pristine sound and smooth classic dbx OverEasy or hard knee compression for a variety of applications from multitrack recorders to PA systems. Stereo coupling is independently selectable for channels 1 and 2 and channels 3 and 4 letting you configure the 1046 for 4 mono channels, two stereo pairs, or two mono and one stereo pair of processing.
Additionally PeakStopPlus is available to for protect your system from the oppressive peaks that can take out valuable drivers in your sound reinforcement rig or studio monitors. So whether you need to control the level, placement in the mix, or overall characteristics of 4 independent signals or control the gain leveling on a couple of stereo pairs, the dbx 1046 is for you.

The 1046 offers the same features and specifications as the 1066 except that it has no manual Attack & Release and no Side Chain functions



1074 Quad Gate

The 1074 Quad Gate offers four dedicated and independent channels of user selectable Noise gating in any combination including: dual stereo, or two mono and one stereo. Each of the 1074's four channels offers gating, threshold, depth and release controls, gold plated XLR inputs and outputs as well as a 1/4˝ key input that allows you to trigger the gates using an external audio source. And like 1046 and 1066, the 1074 is based on the legendary dbx V2 VCA. An internal variable filter allows frequency-selective control of each gate. The 1074 is ideal for a wide range of applications including gating dry and percussive sounds or sounds that have longer decay times such as cymbals and pianos, gating hum or buzz from live instruments or recorded tracks, or eliminating headphone leakage into microphones.



Input Specs	Impedance	Max. Input Level
XLR	>50k Ω balanced >25k Ω unbalanced	> +22dBu balanced or unbalanced
Key Input Specs	**Impedance**	**Max. Input Level**
1/4˝ TRS	50k Ω balanced 25k Ω unbalanced	–
Output Specs	**Impedance**	**Max. Output Level**
XLR	60 Ω balanced 30 Ω unbalanced	> +22dBu balanced > +20dBm unbalanced

VCA	dbx V2
Rear Panel Inputs & Outputs	
XLR Line	RF Filtered, Electronically-balanced / unbalanced switchable for either +4 dBu or -10 dBV operation
1/4˝ TRS Keyed Input	RF Filtered, Electronically-balanced / unbalanced
Gate Controls	
Filter Control	80Hz to 8kHz
Filter Switch	On/Off
Key Input Switch	On/Off
Stereo Couple Switches	Couples channels in stereo pairs. Channels One and Three become the master channels
Threshold Control	Off to +10dBu
Depth Control	
Release Control	0.01 seconds to 3 seconds

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SILVER SERIES

376 Tube Channel Strip with 24-Bit/96kHz Digital Output



The 376 combines the warmth and tone of a vacuum tube mic preamp with a 3-band EQ, compressor and de-esser along with analog and digital outputs to provide a versatile yet affordable "Front End" for any recording rig. The 376's AES/EBU and S/PDIF digital outputs feature selectable sampling rates up to 96kHz, and 24-bit selectable dithering and noise shaping. dbx's Type IV conversion system ensures that all of the analog warmth and detail is captured to digital tape or disk.

Insert

- An unbalanced 1/4˝ TRS insert (effects loop) permits connecting the 376 to an external processor. Insertion point is located after the tube section but before the signal processing section. The **tip** of the 1/4˝ TRS connector provides a SEND (100 Ω) while the **ring** is used as a RETURN (20k Ω)

Analog Outputs

- Electronically-balanced/unbalanced XLR (Pin 2 Hot) and 1/4˝ TRS analog outputs with level control range of ± 20dB.

Mic Preamps

- Drive control sets the amount of gain at the input of the tube stage and provides a broad range of sonic possibilities from subtle tube warmth to harmonically rich overdrive. Gain is variable from +30 to +60 dB for the mic input and ±15 dB for the line input.
- Mic preamp channels have access to selectable 48 volt Phantom Power, 20 dB Pad and Phase (polarity) reverse switches.
- A switchable 12dB/octave Low Cut Filter @ 75Hz is available for any input source — mic, line or instrument.

Compressor

- OverEasy or Hard Knee compression
- Threshold range control is variable from -40 dBu to +20 dBu. (Compression ratio is variable from 1:1 to infinity).
- Program-dependent Attack and Release
- Selectable Slow switch can enable a slower attack/release time— 15 ms for the Attack Time for 15 dB of gain reduction and 8 dB/sec for the Release Time

De-Esser

- A wideband variable gain reduction (compression) circuit and a selectable frequency control with a range of 800 Hz to 10 kHz High Pass are designed to suppress sibilance (hissing S or SH sounds) without affecting the desired frequency characteristics of the program material.
- Program-dependent release time is typically 12 dB/octave; 1 ms/dB

3-Band Equalizer

- 3-band EQ with high (12kHz) and low (80Hz) frequency shelving filters and sweepable mid-band (100Hz to 8kHz. Each band has a ±15dB boost/cut range

Metering

- LED metering is provided at each processing stage
  - 4-segment input drive signal meter
  - EQ Clip LED indicates that the EQ'd signal is clipping
  - Three stage compressor threshold meter
  - 8-segment Gain Reduction meter displays the amount of signal being attenuated
  - Two Stage De-Esser threshold meter
  - Meter Select Switch toggles between the digital or analog output signal displayed in the 8-segment LightPipe meter– analog levels are scaled to dBu while digital levels are scaled to dBFS (Full scale)

Built-in Digital Output (376/386)

Type IV Conversion System

dbx's Type IV Conversion System lets you transfer the depth and complexity tube processing directly to the digital domain via AES/EBU or coaxial S/PDIF digital outputs without the need for an external A/D converter.

- Proprietary analog and digital processing techniques capture a much wider dynamic range than an A/D converter could, preserving the maximum amount of information from the analog signal.
- TSE (tape saturation emulation) prevents digital distortion by remapping audio signals approaching 0dBFS to a built-in soft clipping and limiting algorithm that emulates the effect of tape saturation.

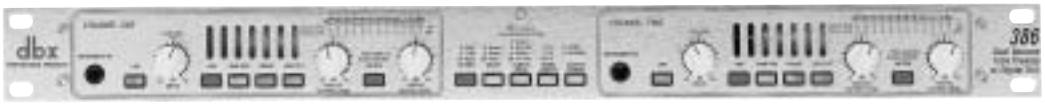
Selectable Dither and Noise-Shaping Algorithms

- Dither is random noise added to the audio signal to eliminate the harmonic distortion created by truncation (word length reduction). Three types of dither are available: TPDF, SNR2, or none
- The Noise-Shaping switch works in conjunction with Dither and allows you to select one of three psycho-acoustic noise-shaping curves: Shape 1, Shape 2 or none

Sync In and Out Connectors

- Internally terminated 75Ω BNC connectors are provided for word clock input and output allowing the processors to be used in a master or slave configuration
- dbx custom VCXO clock chips specifically designed for low-jitter performance
- World Clock signals of 96, 88.2, 48 or 44.1 kHz are supported

386 Dual Channel Tube Mic-Preamp w/Digital Output



The Silver Series 386 is a stripped-down dual channel processor with the same tube mic pre and Type IV conversion system as the 376. The 386 will accept mic, line and Hi-Z instrument inputs and its dual channel capabilities make it an ideal solution for dual mono or stereo tracking of vocals, instruments —miked or direct, as well as stereo keyboards and samples. The hand-selected 12AX7 tube lets you add just the right amount of tube warmth or even overdrive to any signal.

376 AND 386	
Inputs	
Rear Panel	Electronically-bal / unbal XLR Mic and 1/4″ TRS line
Front Panel	High impedance 1/4″ TS unbalanced instrument
Vacuum Tube Mic Preamp	
Input Drive Control Range	+30 to +60 dB (mic) & -15 to +15 dB Line
Insert	1/4″ TRS (unbalanced) tip =SEND ring = RETURN
Analog Outputs	Electronically-bal / unbal XLR and 1/4″ TRS
Output Level Control Range	-20 dB to +20 dB (Analog or Digital)
Digital Outputs	Coaxial S/PDIF (75Ω), XLR AES/EBU (110Ω)
Supported Sample Rates	96, 88.2, 48, or 44.1 kHz
Supported Word Lengths	24-, 20- and 16- bit
Word Clock Sync I/O	75Ω BNC (internally terminated)
Supported W/C Signals	96, 88.2, 48, or 44.1 kHz

376 ONLY	
3-Band EQ	
Low Frequency	80Hz Shelving w/ 15dB boost & cut
High Frequency	12kHz Shelving w/ 15dB boost & cut
Mid Frequency	Sweepable from 100Hz to 8kHz, ±15dB boost/cut
Mid Freq. Bandwidth	Fixed 1.5 octave
Compressor	
Threshold Control Range	Variable -40 dBu to +20 dBu
Ratio Control Range	Variable 1:1 to ∞:1
Program-Dependent Attack and Release Times	
Attack Time	Typically 5 ms for 15 dB of gain reduction
Attack Time (Slow)	Typically 15 ms for 15 dB of gain reduction
Release Time	Typically 50 dB/sec
Release Time (Slow)	Typically 8 dB/sec
De-Esser with variable gain reduction (compression) circuit	
High Pass Filter Range	800 Hz to 10 kHz
Program-dependent Release	Typically 12 dB/oct approx. 1 ms/dB

System Performance

Equivalent Input Noise **Input Noise**	Typically -120 dBu with a 150 Ω source load, 20Hz to 20kHz
Analog Frequency Response	<10Hz to 75kHz
THD+Noise	0.35% typical at +4dBu out, 1kHz, 40 dB gain
Balanced Mic/Line CMRR	>40dB, Typically 55dB
Interchannel Crosstalk (386 only)	Typically -80dB, 20Hz to 20kHz

Input Specs	**Impedance**	**Max. Input Level**
XLR Mic Inputs	330 Ω balanced	-9 dBu or +11 dBu w/20 dB pad
1/4″ TRS Inputs	40k Ω balanced	+21dBu
1/4″ TRS Inputs	20k Ω unbalanced	+21dBu
1/4″ instrument inputs	470k Ω unbalanced	+21dBu
1/4″ TRS Insert Specs	**Impedance**	**Nominal Level**
Send	100Ω	
Return	20Ω	
Output Specs	**Impedance**	**Maximum Output Level**
XLR Outputs	120 Ω (balanced)	>+21 dBu, >+20 dBm*
1/4″ TRS Outputs	120 Ω balanced	>+21 dBu, >+20 dBm*
1/4″ TRS Outputs	60 Ω unbalanced	>+21 dBu, >+20 dBm*

** (into a 600Ω load)*

Mini-Pre Single Channel Vacuum Tube Mic Preamp



The newest member of the Silver Series, the Mini-Pre incorporates the classic dbx tube microphone preamp sound in an easy to use, compact, affordable package. A single channel preamp, the Mini-Pre features the same hand-selected 12AX7 vacuum tube to add warmth and tone to any microphone signal, selectable 48v phantom power, 20 dB Pad and Phase reverse switch, and peak LED.

- ◆ XLR microphone input
- ◆ 1/4″ TS line input
- ◆ Balanced 1/4″ TRS and XLR outputs
- ◆ Selectable +48 volt phantom power
- ◆ 20 dB Pad
- ◆ Phase reverse switch
- ◆ 60 dB gain and -infinity to +10dB of output level
- ◆ Peak LED

BLUE SERIES

The craftsmanship of the Blue Series is as stunning as the engineering is innovative. A striking blue front panel machined from 1/4˝ aircraft aluminum, hand-crafted solid aluminum knobs, LEDs mounted individually in machined stainless steel housings, custom VU meters with peak indicators, and heavy gauge chassis solidify the 160SL and 786 as the benchmark compressor and preamplifier for decades to come.

160SL Stereo Compressor with AutoVelocity Dynamics



Equipped with patented AutoVelocity technology, switchable Over Easy/hard knee compression, variable attack and release controls, Side Chain capability, and switchable PeakStop/PeakStopPlus limiting, the dbx 160SL is the standard by which all compressors are measured. With AutoVelocity it is now easier than ever to dial up the exact attack and release. But the heart of the 160SL is its dual proprietary V8 VCA modules. This state-of-the-art implementation of dbx's original Blackmer decilinear VCA boasts an unheard-of 127dB dynamic range and ultra-low distortion. Encased in a specially designed aluminum-zinc housing for shielding and thermal characteristics, the V8 maintains its superior performance even in harsh environments.

The 160SL and 786 take full advantage of the most technologically superior components available today. Premium active electronics, precision 0.1% and 1% metal film resistors, great sounding temperature stable polypropylene capacitors, high-reliability board-to-board connectors with gold-palladium-nickel contacts, Jensen transformers, gold-plated Neutrik XLRs, rare earth magnet relays with gold contacts in a hermetically sealed nitrogen environment, military grade glass epoxy circuit boards, to mention a few, contribute to the most technologically advanced compressor and preamp in the world.

786 Precision Mic Preamp



The ideal companion to the 160S, the 786 is a two-channel precision preamp designed to provide the truest reproduction of the mic source as is electronically possible. In addition to an absolutely pristine audio path and the best pre-amp features such as +48V phantom power, 20 dB pad, and phase invert functions, the 786 also includes a switchable "super low Z" setting which optimizes the pre-amp performance for microphones with very low source impedance. An 11-position coarse gain switch sets the gain between +10 and +60 dB, while the fine gain control varies between -3 and +3 dB. Levels may be monitored on peak-program VU meters with adjustable reference levels.

The signal path consists of a transformer-coupled, discrete-design premium input stage utilizing the proprietary dbx M8 mic pre-amp module. The minimum signal path design routes the amplified signal directly to a patented high output drive circuit. Each stage throughout the mic pre is small-signal square wave compensated to assure precise phase alignment at all audio frequencies.

The 786 also features a patented EQ circuit called “Spectrum” with frequency cut and boost up to 40kHz. A unique variable high-frequency equalization circuit employed in the mic pre-amp, Spectrum enhances the high frequency spectral content of the signal while maintaining phase integrity. The “Detail” control varies the amount of high frequency shelving boost at the frequency determined by the Spectrum control. The combination of the Spectrum and Detail controls allow pristine high frequency equalization adjustments without the added “hash” of traditional high-frequency EQ topologies.

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BLUE SERIES

The Blue Series Difference

±24V Power Supply Rails:
Most solid-state professional audio equipment has ±15 volt power supply rails. This is fine for most applications, but it isn't good enough for the Blue Series. They use ±24 volt power supply rails that allow higher headroom, which translates to greater dynamic range.

Triple Shielded Power Supply:
The Blue Series power supply was designed to overcome hum related problems typically found in power supplies. dbx started with a low-noise toroidal style transformer with a hum shield around it, then they added a second mu-metal can that encloses the entire transformer, and then mounted that in a separate steel power supply chassis. The result – no radiated hum into the rest of the circuitry.

Burr Brown Op Amps:
dbx uses the Burr-Brown (makers of the best and most expensive op-amps in the world) OPA2604 FET based op amp throughout the audio path. The OPA2604 boasts a distortion spec of 0.0003% THD at 1kHz. A FET based op amp has a more linear transfer function -- it generates less odd order harmonics than a bipolar designed op amp. This is one reason why the Blue Series sound is so transparent.

Patented High Drive and Low Distortion Output Circuitry:
This stage has patented circuitry to reduce the distortions introduced by an output transformer. This beefy output stage is capable of driving thousands of feet of cable to +30 dBu. Conventional 5532 op amp style output stages are good, but they don't come close to the performance of the Blue Series output stage.

Jensen Audio Output Transformers:
Output isolation transformers are essential for eliminating grounding problems in audio systems. Both the 160SL and the 786 have custom oversized Jensen output transformers along with patented circuitry that essentially nulls out the distortions caused by the transformer. What this means is that you get all the benefits of an audio transformer but not the problems of a transformer.

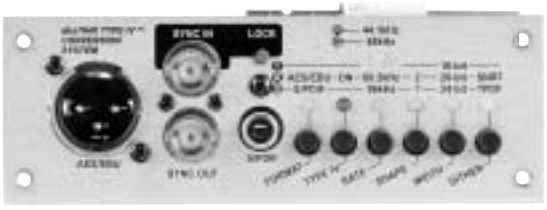
Jensen Audio Input Transformer:
The 786 uses the Jensen JT-16 mic input transformer. It has frequency response of 0.45 Hz to 220 kHz (-3dB down). It is encased in a 30dB magnetic shield to eliminate pickup of any hum. Couple this with a common-mode rejection ratio of 117dB at 60 Hz, and you have the best mic input transformer available.

dbx V8 VCA:
The heart of any compressor is the gain element. dbx took the original Blackmer-designed deci-linear VCA and made state-of-the-art improvements to come up with the V8 VCA. This VCA module boosts an unheard of 127dB dynamic range and ultra low 0.007% THD. Encased in a specially designed, aluminum-zinc housing (for it's shielding and thermal characteristics), the V8 maintains superior performance even in harsh environments.

Precision Potentiometers:
Grab and twist one of the pots on a dbx Blue Series product. You will immediately notice the nice damped response of these controls. We exclusively use expensive precision Alps potentiometers throughout on these products. These parts are extremely reliable and accurate.

Precision 0.1% and 1% Resistors:
To obtain the performance we desired in the Blue Series products, it was mandatory to use precision parts throughout the design. We specifically use all 0.1% and 1% precision metal-film resistors. This ensures that the response from unit to unit is always the same.

704X *Optional 96kHz Digital Output for 786, 160SL and Quantum II*



The 704X is the first 96kHz digital output system that offers a 48-bit internal signal processing path, and up to 24-bit digital output in either AES/EBU or S/PDIF formats. The 704X brings super high-end digital conversion to the top-of-the-line Blue Series processors. Combined with the Blue Series means the 786 now has the ability to offer the industry's most pristine signal path in both analog AND digital, while the 160SL is now the most comprehensive analog compression system with digital output now available. The 704X uses the patented TYPE IV Conversion System to convert your analog signals to digital in a way that preserves the analog warmth, yet gives the clarity and precision offered by the digital domain. Think about it, the warmth and smoothness of your analog signal, coupled with a digital system that offers the best performance in headroom management through custom dither algorithms and noise floor shaping. It really is the best of both worlds.

- 44.1/48/88.2 or 96kHz sample rate
- Dither outputs to 16, 20, or 24 bits using TPDF or SNR 2 algorithms
- 48 bit internal signal path for increased headroom and low level resolution
- Sync input/output using dbx's proprietary clock chips for extremely low jitter sync
- Two user selectable noise shape algorithms for lower perceived noise floor
- AES/EBU (gold-plated XLR) or S/PDIF

DDP

Digital Dynamics Processor



The DDP combines the precision, flexibility and extended parameter control of digital, with the warmth and musicality of analog, to offer superb gating, compression, limiting, de-essing and EQ. The DDP features a choice of true stereo or dual mono operation with ultra-wide range 24 bit A/D and D/A converters that utilize dbx's TYPE IV Conversion System featuring TSE (Tape Saturation Emulation). The LCD display uses a hybrid display technology that combines exceptional graphics, high resolution metering as well as character and icon based displays to show you all the information you need to know. Active effects in a entire setup are intelligently displayed in the form of a graphical curve making it a snap to set up even the most challenging of systems. 50 great factory preset setups are provided to help you get started while the DDP's operating system offers you the ability to refine and save presets to your liking then name and save them as your own presets for later recall. Digital input/output is also available as an option, giving you the ability to go all-digital. Full MIDI SysEx automation functionality makes the DDP one of the most versatile processors on the market today.

FEATURES

Processing Modules

- All of the classic dbx sounds are provided in gating, compression, limiting, de-essing, and sidechain EQ with precision control over every parameter
- Stereo Coupled (True RMS Power Summing) or dual mono operation
- Gate controls – threshold, ratio, attack, hold, release as well as a variable Transient Capture Mode
- Transient Capture Mode (TCM) controls the delay of an incoming audio signal up to 3ms. allowing the very beginnings of fast transient signals to be captured. This results in a smoother sounding signal which requires less aggressive processing in the compressor, limiter, and de-esser functions
- Delay time is variable from 0µs to 3ms in 1000th/sec increments
- Compression controls – threshold, ratio, gain, OverEasy with VariKnee, auto and manual attack, hold, and release modes
- OverEasy with VariKnee algorithm – lets you choose 1 of 10 knee slopes from soft to hard knee. With softer compression knees the compression ratio is gradually increased until the threshold is reached. This makes the transition between uncompressed and compressed signals as seamless as possible
- Limiting controls – threshold, attack and release
- De-esser controls – variable High Pass Filter and de-ess amount
- Sidechain / Inline EQ controls – 3-Band EQ with variable frequency and Q controls, ±12dB boost and cut, and Side Chain Monitor

Preset Management

- 50 factory presets & 50 user programmable presets that can be saved and recalled at any time
- Build your own presets using your favorite building blocks
- Change programs, parameters, and bypass via midi controllers
- "Building Block" style operating system lets you build and save your own presets

Controls

- Data Wheel – changes selected parameters, programs and modules
- Function Buttons – These backlit buttons activate programs, modules, utilities, and parameters

Metering

- 8-step analog input and output metering
- Hi-res graph shows composite output vs. input plot of audio signal

Type IV Conversion

- Ultra wide dynamic range 24-Bit A/D and D/A converters with TSE Tape Saturation Emulation
- TSE (Tape Saturation Emulation) is a process of giving a digital signal an analog-type sound or color, making it more pleasing to the ear. TSE lets you drive the input past the zero mark without risking digital overs. Instead, you get a pleasant saturation modeled after the sound of analog tape being saturated with signal.

Analog Inputs & Outputs

- XLR and 1/4″ balanced ins and outs

Optional Digital I/O Card

- Provides 24-bit digital input and output capabilities in either AES/EBU or S/PDIF formats
- Supports 44.1 and 48 kHz sampling frequencies

MIDI In and Out/Thru

- Provides full MIDI functionality and automation via CC and Sysex
- Out/Thru jack allows you to use the DDP at any point in the MIDI chain

DDP

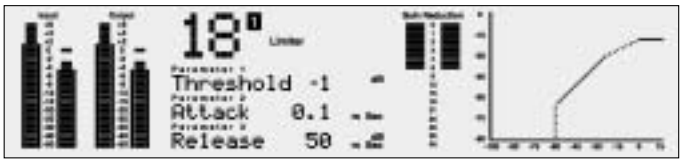
Hierarchy

- The DDP's hierarchy works on a "building-block" philosophy. Every program number consists of a processing "setup" which is built from a chain of processing elements. Chained elements consist of any or all of the following processors: expander/gate, compressor, limiter, parametric EQ, sidechain parametric EQ, and/or de-esser and each has a full complement of parameters that can be precisely manipulated via the Function buttons and the Data Wheel. Chain elements can be used in different combinations to produce a desired effect. Setups can be named and saved to the user library area.
- Several preset mono and linked setups are provided. Processing setups are linked together using True RMS Power Summing for superior stereo operation, or two separate setups may be used in dual mono mode.

Analog Inputs & Outputs	
XLR and 1/4″ TRS line	RF Filtered, Electronically-balanced / unbalanced compatible with +4dBu and -10dBv operation
Input Gain Range	-∞ to +16dB
Output Gain Range	-∞ to +16dB
Gate Controls	
Threshold Range	-75dB to 0dB
Ratio	1:1 to 1:∞
Attack Control	0.1ms to 200ms
Hold Control	0ms to 500ms
Release Control	360dB/sec to 5dB/sec
TCM	Transient Capture Mode Off/On
TCM Time Control	0µs to 3ms in 1000th/sec increments
Compressor Controls	
OverEasy Control	Off / variable knees 1-10
AutoMode Off/On	Enables / disables the Attack, Hold and Release controls
Threshold Control Range	-60dB to +4dB
Ratio Control Range	Variable 1:1 to ∞:1
Make-Up Gain	-20dB to +20dB
Manual Attack Control	0.1ms to 200ms
Manual Hold Control	0ms to 500ms
Manual Release Control	360dB/sec to 5dB/sec
Limiter Controls	
Threshold Control	-60dB to +4dB
Manual Attack Control	0.1ms to 200ms
Manual Release Control	360dB/sec to 5dB/sec
De-Esser	
Frequency Control	800Hz to 8kHz:
Amount	0% to 100%
SideChain EQ / 3-Band Parametric EQ	
SideChain Monitor	Off/On
TSE Mode	Dark, Warm, None, Light, Bright
Frequency Center Range	25Hz to 20kHz
Q (Bandwidth) Control	.25 to 16
Boost / Cut Range	±12dB

LCD Display

- The big and clear LCD screen combines just the right combination of size and luminance and provides a full compliment of metering and processing information, as well as navigational hints that allow you to see what you are doing in a hurry, even under less than ideal lighting
- Digital input and output meters measure internal digital processing levels and show both peak and average levels



- High resolution gain reduction metering
- The screen displays which processing element you are working with along with the amount of gain reduction, as well as input and output levels for the digital processing in both peak and RMS forms, so you can tell exactly what you are doing to the signal with the DDP.
- The Curve Window provides a realtime graphic representation of the combined effects of all compression-related parameters. When working with the sidechain EQ, or the in-line EQ, the curve window changes to show a graphical representation of the 3 parametric bands in a frequency grid.

24-bit A-D System Performance

Dynamic Range	107 dB typical, unweighted, 22 kHz bandwidth
THD+Noise	0.002% typical at +4 dBu, 1 kHz, input gain at 0 dB
Frequency Response	20 Hz to 20 kHz, +0/-0.5 dB
Interchannel Crosstalk	< -85 dB at 1 kHz, input gain at 0 dB
Common Mode Rejection	>40 dB, typically >55 dB at 1 kHz

24-bit D-A System Performance

Dynamic Range	102 dB typical, unweighted, 22 kHz bandwidth
THD+Noise	0.002% typical at +4 dBu, 1 kHz, output gain at 0 dB
Frequency Response	20 Hz to 20 kHz, +0/-0.5 dB
Interchannel Crosstalk	< -85 dB at 1 kHz, output gain at 0 dB



Input Specs	**Impedance**	**Max. Input Level**
XLR & 1/4″ TRS Line	>26 kΩ bal / >13 kΩ unbal	> +24dBu
Output Specs	**Impedance**	**Max. Output Level**
XLR & 1/4″ TRS	120 Ω bal / 60 Ω unbal	>+20dBu into 600Ω

QUANTUM II

Digital Multi-Band Processor



The Quantum II is a digital mastering tool that offers a 4-band compressor, limiter, expander and gate, 5-band parametric EQ, de-esser, sampling rates up to 96kHz with a true 48-bit digital signal path, and the superior dbx dynamics processing. Insert the Quantum II into the master outputs of any console or workstation and turn out a professional master every time. The 4-way crossover splits the signal into 4-bands, and each band may be individually gated, compressed, and limited. dbx's patented TYPE IV Conversion System with the help of TSE (Tape Saturation Emulation) allows you to retain the original warmth and body of the analog signal while giving it the punch and depth needed in today's digital world of music production and mastering. You can also dither your signal down to 24, 20, or 16 bits, using dbx's custom dither algorithms. For more versatility use full MIDI/SysEx control, and int/ext sync capability. An intuitive user interface puts all the mastering processes you need right at your fingertips. The 5-band parametric EQ offers unprecedented control, with variable Q, adaptable or constant Q, and Lo and High shelving.

Performance

- Dual Mono or True Stereo Operation with True RMS Power Summing
- 96 kHz, 24 bit A/D, D/A with a 48 bit internal signal path for increased head room and low level resolution

Dynamics Control

- Multiband (4 Band) and Stereo or dual mono (Broadband) Gating, Compression and Limiting
- Control of each of the bands can be done via all band (adjustments are made to all four bands as a group) or individually
- MS (Middle-Side) Compressor

Parametric SideChain EQ

- 5-Band Parametric Side Chain EQ with either an adaptive Q or a constant Q response and MS (middle-side) EQ

5-Band Parametric EQ

- Each of the five EQ bands can be switched On/Off, and can be positioned pre or post dynamics for maximum versatility.
- All five bands have a sweepable frequency range of 20Hz to 20kHz with ±12dB boost/cut in .5 dB steps
- Bands 1 and 5 are Hi and Low shelves with a 3 to 12 dB/oct slope; bands 2-4 are fully parametric with variable Q (.25 to 16)
- Choice of constant or adaptive Q response
  - ➢ Constant Q operates similar to an analog graphic EQ where the Q (bell) is constant regardless of cut or boost.
  - ➢ Adaptive Q operates similar to the EQ that is found on a mixing console where the width of the band is constant regardless of the cut or boost.
- MS (Middle-Side) EQ is available within all stereo linked setups

4-Band Crossover

- Three crossover points using phase compensated, Butterworth 1st and 3rd order filters, are used to create 4 bands. Tuning them allows you to select which frequencies will be processed with one of the multiband dynamics processors.

Multiband Chain with De-Esser

- The De-Esser is only available for Stereo Broadband or Dual Mono modes
- ➢ Choice of High Pass (HP) or Band Pass (BP) filter modes
- ➢ Width – sets the Q of the BP Filter
- Meter Ballistics
- TYPE IV Conversion System with TSE Tape Saturation Emulation
- AES/EBU (XLR) or S/PDIF (Coaxial) input/output
- Sample rate conversion (full up/down)
- Final output can be dithered to 8, 16, 20, or 24-bit word lengths using SNR2, HPTPDF, or TPDF dither algorithms
- Dither signal can be routed to both the digital and analog outputs or to the digital output only. This allows the full 24-bit signal to be output to the D-A converters
- Two different digital noise shape algorithms (S1, S2) are also available
- Sync input/output uses dbx proprietary clock chips for extremely low jitter sync

Ask the Wizard...

The easiest way to program the QUANTUM II is using the Wizard. In Wizard mode you select the task you need to perform, such as mastering, mixing or tracking, and the Wizard will automatically choose the correct chain type for the task. After selecting a task, you then choose music type, then select the type of EQ, gating, compression, and limiting you are looking for. Once all of the questions have been answered, the Wizard will automatically load a custom setup tailored to your application.

If you are working in a stereo setup, you can optimize output gain by first pushing the EDIT ALL/BAND button and allowing the loudest part of the program material to be recognized. Pressing the button again once the peak has passed, will allow the Wizard to optimize the output level so that the highest peak is set to 0 dBFS.

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EQUIPMENT LEASING AVAILABLE

QUANTUM II

Normalizer/Stereo Adjust

Normalizer is a gain optimizing algorithm located at the end of the signal chain that ensures that the signal being output to tape, disk, etc. is as "hot" as possible without going over 0dBFS

- ➢ ±12 dB level control lets you adjust the final volume sent to the output
- ➢ Ceiling control allows the maximum output level to be set between -3.0 to 0.0 dBFS
- ➢ Stereo adjust – lets you change the width of the stereo mix
- ➢ ±100% balance control adjusts the panning of your signal from left to right
- ➢ M-S controls the width of your stereo image. 0% is untouched, -100% is the narrowest (mono) while +100% is the widest possible stereo image

Ambience Effect

- ◆ Ambience Effect uses a combination of gain and compression to enhance low level information such as reverb tails or acoustic guitar finger picking.
- ➢ Amount 1.0:1 to 5:1 sets the amount of ambience
- ➢ Thresh -60 to 0 dB – sets the threshold of the ambience processor
- ➢ Width 10 to 30 dB controls the portion of the signal that ambience is applied to

Digital I/O

- ◆ TYPE IV Conversion System with TSE Tape Saturation Emulation
- ◆ AES/EBU (XLR) or S/PDIF (Coaxial) input/output
- ◆ Sample rate conversion (full up/down)
- ◆ Final output word length can be dithered to 8, 16, 20, or 24-bit word lengths using SNR2, HPTPDF, or TPDF dither algorithms
- ◆ Dither signal can be routed to both the digital and analog outputs or to the digital output only. This allows the full 24-bit signal to be output to the D-A converters
- ◆ Two different digital noise shape algorithms (S1, S2) are also available
- ◆ Sync input/output uses dbx proprietary clock chips for extremely low jitter sync

Additional Features

- ◆ PC GUI control via the RS232 port
- ◆ Software updateable via Internet and MIDI and RS 232

Analog Inputs & Outputs	
XLR and 1/4″ TRS line	RF filtered, electronically-balanced/unbalanced compatible with +4dBu and -10dBv operation
Input Gain Range	-∞ to +16dB
Output Gain Range	-∞ to +16dB
Gate Controls	
Threshold Range	-75dB to 0dB
Ratio	1:1 to 1:∞
Attack Control	0.1ms to 200ms
Hold Control	0ms to 500ms
Release Control	360dB/sec to 5dB/sec
TCM	Transient Capture Mode Off/On
TCM Time Control	0µs to 3ms in 1000th/sec increments
Compressor Controls	
Band	1 to 4 On/Off
OverEasy Control	Off / variable knees 1-10
AutoMode Off/On	Enables/disables the Attack, Hold and Release controls
Threshold Control Range	-60 to 0 dB
Ratio Control Range	Variable 0.75 to ∞:1
Gain Make-Up	-20dB to +20dB
Manual Attack Control	0.1ms to 200ms
Manual Hold Control	0ms to 500ms
Manual Release Control	360dB/sec to 5dB/sec
Limiter Controls	
Threshold Control	-60dB to +4dB
Manual Attack Control	0.1ms to 200ms
Manual Release Control	360dB/sec to 5dB/sec
De-Esser	
Frequency Control	800Hz to 8kHz:
Amount	0% to 100%
5-Band Parametric EQ	
Frequency Range	20Hz to 20kHz (for all bands)
Boost / Cut Range	±12dB in .5 dB steps(for all bands)
Band 1	Low frequency shelving EQ with a 3 to 12 dB/oct slope
Band 2, 3, 4	Parametric EQ
Q (Bandwidth) Control	.25 to 16 (for bands 2, 3, 4)
Band 5	High frequency shelving EQ with a 3 to 12 dB/oct slope
POS Pre/Post/MS	Sets the position of the EQs in the chain — Pre, Post or MS (Middle-Side) dynamic processors

24-bit A-D/D-A System Performance

Dynamic Range	112 dB typical, unweighted, 22 kHz bandwidth
THD+Noise	0.002% typical at +4 dBu, 1 kHz, input gain at 0 dB
Frequency Response	20 Hz to 20 kHz, +0/-0.5 dB
Interchannel Crosstalk	< -85 dB at 1 kHz, input gain at 0 dB
Common Mode Rejection	>40 dB, typically >55 dB at 1 kHz



Input Specs	Impedance	Max. Input Level
XLR & 1/4″ TRS Line	>18 kΩ bal / >9 kΩ unbal	> +24dBu
Output Specs	**Impedance**	**Max. Output Level**
XLR & 1/4″ TRS	120 Ω bal / 60 Ω unbal	>+20dBu into 600Ω

PROVOCAL

Digital Vocal Strip with Digital Output



Think of it as the being the best sound man you've ever had. The ProVocal takes all of the laborious guesswork out of selecting tools for providing great vocal tracking. Offering unmatched versatility at an unparalleled value, the ProVocal is an all-inclusive toolbox perfect for applications including live, studio and karaoke. The ProVocal features a superb microphone preamp coupled with state-of-the-art microphone and mic-pre amp modeling for a wide variety of sonic characteristics. From there you go into classic dbx dynamic processing such as gating, compression, de-esser, and limiter, then on to chorus, flanger, delay and reverb. Other features include a 24-bit S/PDIF output with selectable 44.1 or 48k sample rate and MIDI control.

- Painlessly helps vocalists create warm and rich vocal tones in the digital domain with crystal clear digital effects, while providing dynamics processing powers that can all be controlled via the smart front panel.
- Takes models of some the most sought after mics and mic-preamps in the industry, and makes them available at your finger tips.
- Front panel instrument input
- Selectable mic/line switch
- 20dB pad, 48-volt phantom power
- Output versatility includes XLR and 1/4″ analog inputs and outputs and 24-bit S/PDIF digital outputs, making it a truly flexible work horse for any vocal application.

286A *Microphone Processor*



Forget about the hassle and cost of patching together multiple processors for use on one track. Now you can have all the tools you'll need in one box, with the shortest signal path to help keep your music sounding clean. The 286A's mic preamp and five processors can be used independently or in any combination.
The 286A's sonically pristine mic preamp features wide-ranging input gain control, switchable 48v phantom power and an 80Hz high-pass filter. Use the patented OverEasy Compressor to transparently smooth out uneven acoustic tracks or deliver that classic "in your face" rock vocal. Take out vocal sibilance and high frequency distortion in cymbals with the frequency tunable De-Esser. Fine-tune the Enhancer's HF Detail control to add sparkle and crispness to tracks. LF Detail control adds fullness and depth to vocals and bass instruments while simultaneously cleaning up muddy low midrange frequencies. And, the Expander/Gate's separate threshold and ratio controls allow you to subtly reduce headphone leakage or radically gate noisy guitar amps.

- A host of metering and status LEDs visually guide you to achieving the right sound.
- The floating balanced XLR mic input accepts balanced or unbalanced inputs.
- An additional 1/4″ TRS phone jack can accept balanced/ unbalanced line signals to process live electronic instruments or pre-recorded tracks at mixdown.
- An insert jack between the mic pre and signal processing sections can be used to "loop out" to external processors or to mix the signal out to an external destination.

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120A

Subharmonic Synthesizer with Modeled Waveform Synthesis



The disco boom of the 70's was fueled by the throbbing bass beat of dbx Subharmonic Synthesizers in discos around the world. During the 80's, patented dbx Subharmonic Synthesis was the secret weapon used by mobile DJ's and film and sound professionals to produce an impact unavailable from any other device—unique because the dbx process actually produces a new, Waveform Modeled bass note, exactly an octave below the bass in the original audio. For 2003, the 120A Subharmonic Synthesizer has been specially optimized by dbx engineers for the needs of audio professionals. Two separate bands of bass synthesis provide the best combination of smoothness and control, while the independent Low Frequency Boost circuit gets the most out of high-performance low frequency speaker systems.

- Patented subharmonic synthesis process builds the synthesized waveform using the waveshape of the original bass material. Unlike other systems, the dbx process produces smooth, musical low frequencies that don't interfere with the mid and high-band, even when maximum synthesis and boost are applied. The result is a low-end punch that you really feel—even at levels that won't destroy equipment or damage hearing.
- 1/4″ (balanced) and RCA inputs, along with main 1/4″ outputs which can be full range (including synthesis) or high frequency only, plus separate subwoofer output with its own level control.
- Individual control for two ranges of subharmonic frequencies
- Front panel LEDs that show crossover status and synthesis activity
- Patented circuitry ensures that mid and high frequencies are not affected
- Built-in crossover with choice of 80Hz or 120Hz crossover point
- Enhance bass audio material for use in a variety of applications including:

• Nightclub and dance mixing • DJ mixing
• Theater and Film sound • Music Recording
• Live Performance • Broadcasting • Aerobics

223/234 • 223XL/234XL

Stereo 2-Way/Mono 3-Way and 2/3/4-Way Crossovers

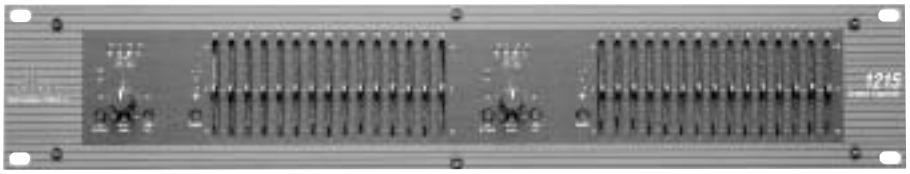


Extremely well built and great looking, the 223 and 234 feature switches on the rear panel for stereo 2-way or mono 3-way (mono 4-way on the 234) operation, LF mono sum (for a subwoofer feed) and a range of individual channel crossover frequencies. The XLR versions (223XL and 234XL) feature TRS differentially-balanced XLR inputs and outputs in place of the standard 1/4″. Perfect for permanent installations and touring PA systems.

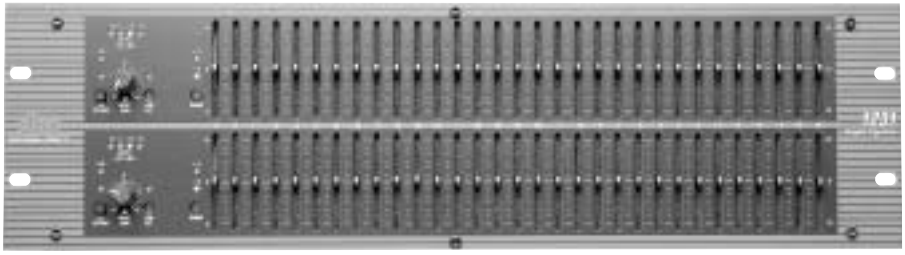
- Mode switch for stereo 2-way or mono 3-way operation (223)
- Mode switches for mono 4-way or stereo 2-way/3-way operation (234)
- x10 range switch on both channels
- LEDs indicate when "x10" switches on the back panel are activated and whether the unit is in stereo or mono mode.
- Linkwitz-Riley 24dB per octave filters
- Recessed 40Hz low cut (HPF) switches on each channel are available to remove unwanted low frequencies.
- Both the low and high outputs on each channel have a gain control ranging from infinity to +6dB to allow muting of individual outputs and for level matching. Phase reverse switch on all outputs
- Controls exude great precision, and the feel is a solid "click".
- Internal 120/240v power supply eliminates the need for annoying wall-warts.
- TRS differentially balanced 1/4″ inputs and outputs are bolted to the chassis, so even if someone stomps on a plugged in cable, it won't tear up the circuit board inside.

12 SERIES EQUALIZERS

Dual Channel 15- Band and 31-Band Graphic Equalizers



The 1215 and 1231 equalizers are designed to meet the needs of the most demanding sound reinforcement environments, while offering simple and straightforward controls. Both units offer dual channel operation. The 1215 provides 15 2/3 octave bands with ISO frequency centers, while the 1231 provides 31 1/3 octave bands with ISO frequency centers. Both feature 45 mm faders, selectable ±6dB or ±15dB boost/cut ranges, ±12 dB input gain range, and switchable 40Hz/18 dB per octave low-cut filters. Precision engineered to provide years of maintenance-free operation in any application, they utilize only the best possible components. RF Filtered, electronically-balanced XLR, barrier strip, and 1/4˝ TRS inputs and outputs provide ease of installation, while magnetically isolated transformers and chassis/signal ground lift capabilities ensure quiet performance.



The Last Word in Affordable, High Quality EQs

With the dbx 12 Series the goal was simple; build the best bang-for-the-buck, no nonsense equalizers on the market. Price/performance champions, the 1215 and 1231 are designed for down-to-earth sound shapers who don't need a feature-laden20 Series EQ, but who absolutely demands a pristine signal path and consistent performance in an equalizer.

System Performance	
Common Mode Rejection	>40dB, typically >55dB at 1kHz
Frequency Response	<10Hz to >50kHz, +0.5/-3dB
Signal to Noise ±15db Range	90dB
Signal to Noise ±6db Range	97dB
THD+ Noise	<0.04%, 0.02% typical at +4dBu, 1kHz
Dynamic Range ±15db Range	109dB
Dynamic Range ±6db Range	115dB
Interchannel Crosstalk	<-80dB, 20Hz to 20kHz

Input Specs XLR, 1/4˝ TRS and Barrier Strip	**Impedance** >40k Ω balanced >20k Ω unbalanced	**Max. Input Level** >+21dBu (2kΩ) balanced or unbalanced
Output Specs XLR, 1/4˝ TRS and Barrier Strip	**Impedance** 200k Ω balanced 100k Ω unbalanced	**Max. Output Level** > +21dBu (2kΩ) balanced or unbalanced

Inputs & Outputs	
Rear Panel	RF Filtered, Electronically-balanced / unbalanced XLR, 1/4˝ TRS and Barrier Strip
Input Gain Range Control	-12dB to +12dB
Function Switches	
EQ Bypass	Hardwire bypasses of the graphic EQ section with 2-second power-up delay
Low Cut (recessed)	Activates the 40Hz 18dB/octave Bessel high-pass filter
Range (recessed)	Selects either ±6dB or ±15dB slider boost/cut range
Metering	
Output Level	4-Segment LED @ -10, 0, +10 and +18dBu
Clip	Red LED
Power Supply	
Operating Voltage:	100V AC 50/60Hz; 120V AC 60Hz

- ◆ Selectable ±6dB or ±15dB boost/cut range for precise gain adjustments
- ◆ Electronically balanced/unbalanced inputs
- ◆ Servo balanced/unbalanced outputs
- ◆ RF filtered inputs and outputs
- ◆ XLR, Barrier Strip, and 1/4" TRS connectors
- ◆ ±12dB input gain range
- ◆ 18dB/octave 40Hz Bessel low-cut filter
- ◆ Chassis/signal ground lift capability
- ◆ Internal power supply transformer
- ◆ Power-off hardwire relay bypass with 2-second power-up delay

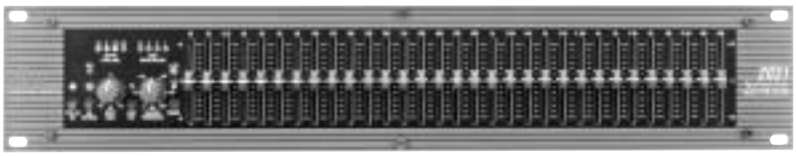
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ONE HOUR FREE PARKING
AT 349 W. 34th STREET (with purchase of $100 or more)

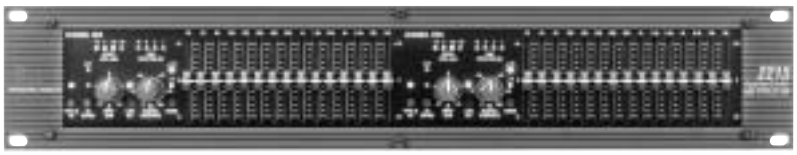
20 SERIES EQUALIZERS

Graphic EQs with Type III Noise Reduction

There are three EQs available in the 20 Series; the 2031 single channel 31 band 1/3 oct, the 2215 dual channel 15 band 2/3 oct, and the 2231 dual channel 31 band 1/3 oct. Each of these offers the same precision engineering, top shelf components, I/O and features plus, they add two powerful tools that let you use your system with confidence — dbx's proprietary Type III Noise Reduction which is capable of increasing signal-to-noise ratios by up to 20dB, as well as the patented PeakPlus limiter which provides fail-safe system protection for your sound system. A four stage LED ladder gives you a great visual indication as to what the limiter and signal levels are doing. Offering great value and an unequalled feature set, dbx's 20 Series equalizers have become crucial links in the sound systems of countless professionals all over the world.



2031 Single Channel 31 band 1/3 octave



2215 Dual Channel 15 band 2/3 octave



2231 Dual Channel 31 band 1/3 octave

SAME FEATURES AS THE 12 SERIES *PLUS*—

Type III Noise Reduction

- Each channel features individually switchable, transparent noise reduction circuitry, capable of restoring up to 20dB S/N ratio
- A virtually instantaneous encoding and decoding algorithm allows radical EQing without substantially affecting noise performance

PeakPlus Limiter

- The PeakPlus Limiter circuitry available on each channel utilizes a variable 0dBu (off) to +24dBu threshold control to provide fail safe system protection

Metering

- Four segment LED bargraph provides metering for BOTH Gain Reduction and Output Level
- Status LEDs offer visual feedback for all settings on the front panel.

System Performance	
Common Mode Rejection	>40dB, typically >55dB at 1kHz
Frequency Response	<10Hz to >50kHz, +0.5/-3dB
Signal to Noise ±15db Range	>90dB (NR out), >100dB (NR in)
Signal to Noise ±6db Range	>94dB (NR out), >100dB (NR in)
THD+ Noise	<0.04%, 0.02% typical at +4dBu, 1kHz
Dynamic Range ±15db Range	108dB (NR out), >118dB (NR in)
Dynamic Range ±6db Range	112dB (NR out), >118dB (NR in)
Interchannel Crosstalk	<-80dB, 20Hz to 20kHz

Input Specs	**Impedance**	**Max. Input Level**
XLR, 1/4″ TRS and Barrier Strip	>40k Ω balanced >20k Ω unbalanced	>+21dBu (2kΩ) balanced or unbalanced
Output Specs	**Impedance**	**Max. Output Level**
XLR, 1/4″ TRS and Barrier Strip	200k Ω balanced 100k Ω unbalanced	> +21dBu (2kΩ) balanced or unbalanced

Inputs and Outputs	
Rear Panel	RF Filtered, Electronically-balanced / unbalanced XLR, 1/4″ TRS and Barrier Strip
Input Gain Range	-12dB to +12dB
Function Switches	
EQ Bypass	Hardwire bypasses of the graphic EQ section with 2-second power-up delay
Type IV NR	Activates dbx Type III[a] Noise Reduction
Low Cut	Activates the 40Hz 18dB/octave Bessel high-pass filter
Range	Selects either ±6dB or ±15dB slider boost/cut range
PeakPlus Limiter	
Threshold Control	0dBu - +24dBu (OFF)
Metering	
Output Level	4-Segment LED @ -10, 0, +10 and +18dBu
Clip	Red LED
Gain Reduction	4-Segment LED display for PeakPlus Limiter
Operating Voltage:	100v AC 50/60Hz; 120v AC 60Hz

ECLIPSE

Harmonizer Effects Processor





The Eclipse unit provides the power, performance and signature Eventide audio effects in a single rack space unit, the company's first in that configuration. The folks at Eventide were not looking to build a "junior" Harmonizer product. In fact, even though it's just half the size of the legendary H3000 series Harmonizers, Eclipse has five times the processing power, and twice as many features. Eclipse features a dual-engine architecture, that can be configured in series, parallel, stereo or dual mono. Setup and operation is intuitive, with an enhanced user interface and a powerful search facility for rapid program/preset selection by category or application. The audio performance of the Eventide Eclipse Harmonizer unit surpasses just about anything in its class, with 24-bit digital conversion, 96kHz sampling and a signal to noise ratio exceeding 104dB. Digital S/PDIF and AES/EBU I/O, analog balanced and unbalanced I/O and ADAT lightpipe are all supported. Eclipse's heavy-gauge, all metal faceplate, solid, positive-action controls and sharp, bright display further demonstrates the "fully professional" claim for the product. Eventide's famous "knob" and full numeric keypad are included, as well as customizable "hot keys" for maximum flexibility and ease of operation. Eclipse answers the call for many musicians and working professionals seeking Eventide-quality pitch change, reverb and special effects programs in a compact box that is affordable and uncompromising.

FEATURES

- Up to five times the processing power, yet half the size of the legendary H3000 series Ultra-Harmonizer brand effects processor.
- Dual-engine architecture (two configurable effects blocks), configurable in series, parallel, stereo or dual mono.
- 24-bit resolution for superb fidelity at up to 96kHz sampling and a signal to noise ratio better than 104dB.
- The famous Eventide "knob", full numeric keypad and customizable "hot keys" for frequently used parameters provide maximum flexibility and ease of use.
- A global tempo feature synchronizes LFOs, delay times, and other time-sensitive parameters to the music being processed.
- A full complement of LFOs, envelope followers, and ADSRs can be patched to any parameter.

500 Programs

- 400 factory and 100 user programs with a full complement of Eventide's signature pitch change, reverb, and special effects presets are provided all with, all with uncompromising quality.
- An additional 500 programs can be stored on a compact flash memory card.
- Besides viewing programs by program number, they are flagged so they can be sorted by application (Guitar, Vocals) and effect type (Pitch, Reverb, Delay).

Algorithms

- Presets are made up of 1 or 2 algorithms.
- Each algorithm is a block of signal processing elements, such as mixers, shifters, compressors, etc, that can be a powerful and sometimes complex effect in itself.

I/O Connectors

- Dual (left/right) balanced XLR, unbalanced 1/4″ combi inputs
- Dual (left/right) balanced XLR and unbalanced 1/4″ outputs
- AES/EBU (XLR) and coaxial S/PDIF digital I/O plus an optical interface that can be switched between ADAT and S/PDIF.
- Word Clock I/O (BNC)
- The MIDI Out and Thru connectors plus a 7-pin MIDI In port for use with pedal boards or standard 5-pin devices.
- A 9 pin RS232 serial connector allows you to transfer information to and from your computer.
- Two Stereo 1/4″ multi-function foot pedal / foot switches inputs can be used for adjusting parameters

PLUGZILLA



Standalone Plug-in Player

Plugzilla is a unique two rack space effects processor capable of running nearly any Windows-based VST plug-in. Since its preview at the Audio Engineering Society's 113th annual convention, Plugzilla's launch has been widely anticipated throughout the industry. Plugzilla employs two independent fully routable machines capable of running 32 channels of reverb, and the ability to simultaneously power as many as eight plug-ins. Plugzilla's brings a new level of versatility, power, and portability to the rapidly expanding world of plug-ins. Plugzilla adds a new dimension to front of house mixing, live stage rigs, recording studios and mastering rooms. It can store thousands of plug-ins and presets in its internal Flash memory and features USB and Compact Flash connections for efficient plug-in transfers. In addition, 32 midi channels plus four footswitch inputs and two sets of eight "hot knobs" and "hot switches" that can easily be assigned and routed to plug-ins add incredible flexibility to the product. Plugzilla is designed by Manfold Labs, a new company comprised of some of the pro audio industry's leading effect processor designers.

FEATURES

World's First Standalone Plug-in Player

- Starting with Steinberg's VST 2.0 standard, and adding a quick, intuitive user interface, stellar audio performance, and wrapping it in a sexy, quiet 2RU package, Plugzilla will forever change your idea of VST plug-ins, rack mounted effects processors, and synthesizers.
- Fully compliant with Steinberg's VST 2.0 specification
- Runs virtually any Windows-based VST plug-in* - including VSTi instruments.
- Two independent, fully routable machines
- Runs up to eight plug-ins simultaneously
- Store thousands of plug-ins and presets in internal Flash memory
- High performance processing - capable of running 16 channels of reverb.
- Eight snapshot memories allow quick comparisons and creative flow

*(*depends mostly on copy protection scheme)*

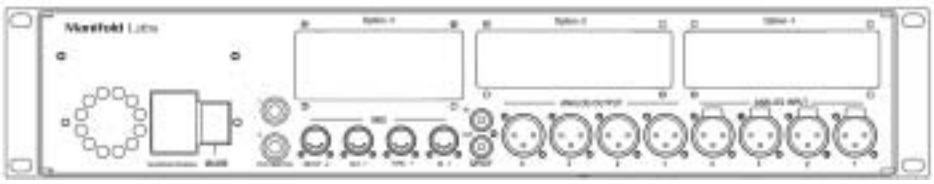
Bring your VST plug-ins to the live gig, recording studio and mastering session. Add DSP to your computer-based Digital Workstation.

Quick, Simple and Easy-to-Use Interface

- Two sets of eight "Hot Knobs" and "Hot Switches" can easily be assigned and routed to plug-ins providing incredible flexibility for accessing realtime control and automation functions.
- Dual bright fluorescent information displays
- Compact FLASH slot and a USB port are available for additional plug-in storage.
- More than 200 plug-ins are loaded when shipped.

Inputs and Outputs

- Dual stereo/four channel 96kHz/24 bit balanced XLR analog inputs and outputs
- Input to output latency at 48kHz sample rate is less than 4 msec.
- Coaxial S/PDIF Digital inputs and outputs
- Two MIDI In and Out ports provide 32 MIDI channels
- Two footswitch inputs with full routing and assignment to plug-ins.
- Universal power supply (90 - 240VAC / 50-60Hz



FOCUSRITE

PLATINUM OCTOPRE

8-Channel Mic Preamp



With eight pristine channels, class A Focusrite processing, eight compressor/limiters and almost all the interface options known to man, the OctoPre makes high quality, multi-channel recording easy. The perfect partner for any digital audio workstation, it is equally suited as an instant location recording solution, or as an additional set of mic preamps for any analog, digital or hard disk recorders. Each channel features a revolutionary compressor/limiter circuit, providing a warm-sounding compressor, which morphs into a brick wall limiter to avoid those critical overs, ensuring total control over all eight channels. The first two 'Super channels' also feature phase reverse, as well as TRS inputs on the front for quick and easy, DI-free plug-in. Finally, the high quality digital converter options cover almost every interfacing eventuality. Up to sixteen digital outputs can run simultaneously with any mix of analog outputs, and all the settings are adjustable, directly from the front. With this entire package fitting into a single 1U chassis, OctoPre sets the standard for multi-channel recording solutions.

FEATURES

Dynamics Control

- OctoPre features eight independent compressor/limiters, whose revolutionary circuit design allows dynamics to be handled via a single control. Starting with a brick-wall limiter the rotary control then allows you to add increasing amounts of compression.
- Two independent side-chain control circuits are used for each channel of dynamics, one for limiting and one for compression. The limiter and compressor side-chains each generate two separate control signals, one fast and one slow, that are then fed to the gain change elements in the audio path.
- The gain control elements consist of a pair of custom Optos, which follow a crossover network that splits the audio into two bands, "high" and "low". The fast side-chain drives the high frequency opto, while the slow side-chain drives the low frequency opto. This ensures less distortion when the limiter is reacting to very fast transients. By splitting the response, only the high frequency section of the waveform is drastically affected, reducing distortion, and therefore non-related harmonics, providing a far more musical effect.
- In addition, both the side-chains operate as feedback systems. Thus, the amount of compression and limiting is calculated using the level after the split opto stage. The advantage of this feedback system is that the limiter senses the level after compression has been applied and therefore the limiter only works when it absolutely has to.
- The dynamics control also introduces an additional make-up gain, balancing the level change caused by compression. This auto make-up gain ensures the output level to the A/D remains constant while further compression is applied. Consequently, any input levels already set will not require any further adjustment during dynamics set-up. Only one dial, but all the control, providing the quickest route to a warm, controlled, yet entirely musical signal.

Class A Pre Amp

- Mic preamps are the same as those in the award-winning, VoiceMaster PRO. The custom preamp ensures low noise and distortion, delivering clarity and warmth without unwanted artifacts.
- Also found in this section are phantom power, a Line level switch and a High Pass filter, enabling OctoPre to handle a multitude of audio sources simultaneously, with style and grace.

Super Channels

- The first two channels also feature high quality instrument inputs and phase reverse, both available direct from the front, avoiding the need for a separate two channel DI box.
- Both channels allow all three sources to be patched simultaneously, with inputs selectable directly from the front panel allowing the unit to be permanently racked and patched to your own specifications.

Outputs

- Eight line level balanced (+4 dBu) analog inputs and outputs are provided as standard via 25-pin connectors. These connectors are routed to 8 XLRs via a breakout cable (sold separately). In addition to the analog outputs, there are two optional digital outputs available:
- Two high-quality, 24-bit A/D converters handle rates up to 96kHz, but can dither down to 20 or 16 bit. Frequency settings and bit rates can be altered with an ADC lock LED to show when word clock is synchronized. An external wordclock input is also provided (BNC).

• ADAT outputs only - 8 channels of ADAT lightpipe

• 8 AES/EBU or S/PDIF outputs plus 8 channels of ADAT all on one board. The AES/EBU or S/PDIF outputs can run simultaneously with ADAT outputs, giving you 16 channels of digital output.

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PLATINUM MIXMASTER DIGITAL

Stereo Dynamics, EQ and Image Processor



The MixMaster is an analog stereo audio processor designed primarily for project studio mastering. However, with so many useful features in one box, anyone involved in the business of making music will quickly find it indispensable at other stages of the recording process too. This piece of outboard genius refines the final balance and dynamic energy of the mix, introducing stereo enhancement while restoring essential elements which may have become overwhelmed in the mixing process. Simple yet effective processing sections cover every aspect of the mix, giving you the quickest route to that seemingly 'out of reach' sound that makes a hit record. Within minutes you can give your mix punch, width and sheen with the signature transparency and sonic integrity that made Focusrite famous.

FEATURES

High Quality A/D Converter

- A 16/24-bit and 128x over-sampled digital interface enhancement with a sampling rate of up to 96kHz is built-in.

Stereo Expander

- Essentially a soft noise gate, the stereo expander decreases unwanted background noise without sounding harsh or brutal. With only two controls (threshold and release) and an LED monitoring display, controlling the noise floor becomes a breeze.

Stereo EQ

- The Stereo EQ on the MixMaster is tri-band, with stepped frequency shelving high (HF) and low (LF) bands, and a parametric mid band. A serious Focusrite EQ with the ability to add musicality and subtlety to your mix.
- The "tilt" feature allows gentle emphasis or de-emphasis at either end of the audio spectrum and is a vital mastering function. The MixMaster EQ is a very powerful tool that can gently enhance the tonal quality of the mix, according to the user's objective.

Stereo Spectral Compressor

- The Stereo Spectral Compressor is a multi-band compressor with three bands of compression, Low, Mid and High, each being visually displayed – vital when you need to smooth out different frequency bands and have true control over the mix.
- When using a single band compressor, a loud kick drum can, for example, cause the compressor to pump on every kick. In dance music this effect might be desirable and consequently you have the option to lock the compressor across the bands. The MixMaster also allows you to individually compress each of these three bands, so taming the kick alone is as easy as turning down the LF trim. Taming an overly loud vocal or hi-hat is just as easy; Using the MID band for vocals and the HF band for hi-hat and snare, you have complete control over these specific areas.
- The MixMaster simplifies this process to 5 knobs, and 2 buttons, all of which are easy to understand, and allow you not only to hear but also see what effect they are having on your mix.
- A track put through this section will, when adjusted, sound tighter, bigger and smoother. Finally, a device that delivers that elusive finished, professional sound. The process is not difficult to employ. Use your ears and you will soon be grinning from one to the other as your mix comes to life! Also included in this section is the "LF slope adjust" switch. Switching the circuit in gives a fuller bass sound, an instant enhancement for dance mixes at the flick of a switch.

Analog Spatial Enhancer

- For widening your stereo image. A slight adjustment of the image controller allows you to create an ambient, spacious mix that extends beyond the normal stereo extremities, breathing new life into your projects. A tweak in the opposite direction produces a narrowed, focussed image.
- Many width enhancers can cause phase coherence problems for centrally-panned mix elements, but the MixMaster's additional direct inputs allow critical central elements (like lead vocals or a kick drum) to be added to the mix after stereo enhancement. This allows you to enjoy the benefit of wide, airy mixes, but retain critical tight focus on, say, the lead vocal.

Output Control

- The output section gives fine control of stereo balance and trim is available before the A/D converter or analog outputs. For output protection, switch in the stereo frequency adaptive limiter, with its own LED meter, to stop those critical digital 'overs'.

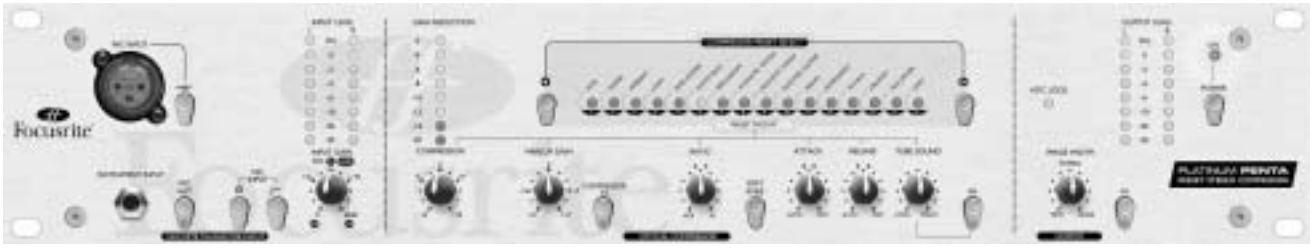


Rear view of the Platinum MixMaster Digital

PLATINUM PENTA

Stereo Analog Preset Compressor

A stereo compressor with 16 preset compression settings set among a host of Class A analog circuitry, the Penta offers that sought-after professional sound at the touch of a button. The presets, configured by a prestigious group of music industry professionals, can be freely adjusted, enabling users to tailor compression to their individual needs.



Packed with analog processing power including a Focusrite Class A mic pre, TubeTran technology and spacial enhancer, the Penta offers a huge number of recording benefits within a matter of seconds while still exhibiting the Focusrite signature sound. Use as a front end when tracking quality signals, as a mix-down processor, or as a final mastering processor. Every dynamics processor you'll ever need, squeezed into one 2U rack-mountable unit, the Penta will be the best investment you ever made.

Preset Stereo Compressor

The Penta offers you instant compression settings, as formulated by the Focusrite design team. Immediately available are 16 presets specifically designed to give you perfect compression at the touch of a button. But to make sure they don't limit your potential, these presets are entirely editable, giving you a quick, yet effective route to perfect compression.

Kick	Creates a full and deep effect. Use the attack to control the amount of punch desired for your track.
Snare	Set up for perfect control of stick (attack) and ringing overtones (release).
Ambient	Use with live drums or instruments and lift out natural room effects or get the best from digital reverbs.
Loop	Tight and fast compression, especially suited to sampled loops and electronic sounds.
Bass Guitar	Plug straight into the front panel and call up deep and rounded sounds or punchy slap.
Synth	Bass Fat sounding especially when used with the warmth control to give a true vintage valve sound.
Percussion	Acoustic or sampled drums.
Acoustic picked	High threshold and ratio give a controlled yet highly dynamic response.
Acoustic strum	Clean and crisp with a touch of warmth. Gives a clear but warm and naturally compressed effect.
Electric Guitar	Clean or distorted, plugged straight in or mic'd up, the opto compressor is perfect for guitar.
Piano	Achieve clarity and low distortion. Wide dynamic range and long sustaining mid range make the piano the most difficult of instruments to compress well, ...until now.
Keyboards	Great for pads, this compressed and warm setting is especially suited for use with the image control to maximize stereo width.
Vocals	Smooth, controlled and clean for male or female vocal recording.
Crunch	An essential setting for big beat or drum loops.
Mix (pump)	The classic loud and full on mastering compression setting.
Vocals (limit)	High threshold and plenty of warmth give a very dynamic upfront vocal performance.

PLATINUM COMPOUNDER

The Compounder is a high performance stereo dynamics processor designed for the quality conscious professional and project studio owner. This highly featured unit is an essential tool for any style of music. The combination of high quality compression with the powerful Bass Expander, make this unit a must have for any dance music engineer or musician.



Discrete Transistor Mic Pre

- The Penta offers the same Platinum mic pre featured in the award-winning VoiceMaster, ToneFactory, and the industry transforming Control/24 from Focusrite. The Penta ensures professional performance within the audible frequency range and beyond, producing cleaner recordings and delivering clarity without coloration.

Direct Input

- The Penta features a direct input on the front panel for quick and easy plug-in. Use the Penta as a front end for getting all your instruments down to track in seconds.
- Use the Penta as a mono compressor when recording a vocal or instrument, AND as a stereo compressor when mixing down. (When necessary, the Penta may also be used as a stereo compressor when recording, if it is inserted across two channels of a mixing console, or when compressing e.g. a stereo keyboard signal.)

TubeTran Technology

- Focusrite's Exclusive TubeTranTechnology gives you controllable tube style warmth. This unique effect is achieved using a dedicated discrete mosfet circuit, specifically designed to create this indistinguishable audio illusion. Instead of limiting your studio to one sound, now you can choose – less tube, more tube, no tube, you decide.

Analog Spatial Enhancer

- The Penta features an image width enhancer for altering the apparent image width of a stereo source. With the Image Width control you can widen the stereo image of your mix to create an ambient, spacious mix that extends beyond the normal stereo extremities, breathing new life into your projects. A tweak in the other direction produces a narrow, focused image.

Output

- Featured in the final section is a comprehensive visual meter to help you ensure optimum levels at all times. The Penta also uses the same optional high quality A/D converter as the VoiceMaster and Trak Master, making digital interfacing a breeze.

Gate/Expander, Compressor, Limiter

- The unique opto circuit featured in the Gate/Expander section ensures quiet operation with no pops or clicks. You have accurate control over gating and the expander switch ensures professional noise reduction on difficult audio tracks such as vocals.
- Compounder's compressor features Class A amplification and a VCA circuit design derived from the legendary Red 3, resulting in superb sound and very low distortion.
- Soft and Hard Knee compression curves and a wide ratio control that takes you beyond infinity, means you have substantial control over compression.
- With the unique Bass Expander section you can make kick drums and bass riffs far more powerful and effective. The expander circuit generates extra bass harmonics to fatten up any low frequency signal.
- A matched pair of high quality Optos offers improved performance over VCA limiter designs which tend to add high levels of distortion, even when they are not limiting. The Limiter features a Class A low distortion design and accurate threshold control to ensure quality precision limiting. The limiter enables you to prevent overload when recording to an A/D.

True Stereo Linking

The link feature enables the Compounder to act as a true stereo compressor, enabling complete stereo control of every parameter. When switched to stereo the left channel becomes the master section and controls both left and right compressor and limiter sections simultaneously. You can also choose to operate the unit as a dual mono device allowing you to process two separate channels independently.

COMPOUNDER SPECS:

Compressor	
Threshold Range:	-24dB to 12dB
Ratio:	1.3:1 through infinity to over-compression
Slope:	select between hard and soft knee.
Attack and Release:	Variable or auto (program dependent) Attack: 100µS to 100mS Release 100mS to 4S

Limiter	
Threshold Range:	12dBu to 26dBu
Ratio:	infinity
Attack:	Fast

Gate	
Threshold Range:	-40dB to +10dB
Gate Range:	-20dB or -80dB(full)
Attack:	switched fast or slow
Release/ Hold:	variable or fixed 100mS to 4S
Expander Ratio:	2:1

FOCUSRITE

PLATINUM VOICEMASTER PRO

Dedicated Engineering for Vocals



An end to the search for that elusive perfect vocal—VoiceMaster PRO represents a new generation of Channel strip design. The award-winning Class A Pre-amp is capable of capturing every nuance from any source, while latency-free monitoring ensures direct and delay-free mix control. Tools such as the Voice optimized EQ, 'Vintage Harmonics' and 'Tube Sound' allow you to get creative with a touch of class, putting your own stamp on every recording. Combined with the 24-bit, 96kHz A/D option, the VoiceMaster PRO is the perfect interface between your mic and DAW or other recording media.

FEATURES

Discrete Transistor Mic Pre

Audiophile-quality, Class A transistor mic pre features Rupert Neve's wide bandwidth design to ensure low noise and distortion, delivering clarity without coloration. Provides the signature transparency that Focusrite is famous for

Also found in this section are Phantom Power, Phase reverse and a variable High Pass filter for controlling rumble, proximity effect and pop reduction, enabling VoiceMaster PRO to handle any audio source with style and grace.

Direct Input

Also included on the front panel are mic and instrument inputs, allowing quick and easy connection without the need for a separate DI box. Although optimized for vocal use, VoiceMaster PRO can equally turn its hand to any instrument with outstanding results.

Tube Sound

Equipped with cutting-edge valve emulation technology, the Tube Sound section provides all the warmth and control, without the unreliability/unrepeatability that valves often entail. Tube Sound circuit produces 2nd, 3rd and 5th harmonics relative to the source audio, the frequencies being controlled by the Tone control knob ('Mellow' to 'Bright'). The level of effect applied is adjusted via the 'DRIVE' control ('Cool' to 'Warm'). This section ensures you get exactly the sound you're looking for.

Optical Expander

The optical expander is the perfect tool for removing ambient noise and headphone bleed, while avoiding signal pumping. The release control is variable, allowing for a very accurate set-up. Can also be tailored for a more aggressive approach on less sensitive signals such as electric guitar or drums.

Optical Compressor

Designed specifically for voice with easy-to-set-up dynamic control of performance changes. Servo-controlled optical attenuators give more punch and fat sounding compression, especially when driven hard. Offers soft (vocals) or hard (drums or electric guitars) compression. The additional enhancer (treble) enables you to add brightening harmonics to a vocal which lacks character. A 'post EQ' switch enables frequency-conscious compression.

Voice Optimized EQ

Designed to give immediate results to live or pre-recorded vocals. Controls include:

- Breath -shelving EQ, switchable from 16K to 10K adds air to both female and male vocals
- Mid -A bell-shaped curve at 1K3 to give vocals bite in a busy mix
- Absence -A notch filter with a narrow Q centered at 3K9 to remove harshness
- Warmth -bell curve, tuneable from 120Hz to 600Hz, adding depth and volume

Vintage Harmonics

Vintage Harmonics ensure your vocal sits prominently in the mix, while adding character to an otherwise sterile digital environment. Emulates a classic tape-based vocal enhancement technique employed during the 1970's.

Optical De-esser

The same optical de-esser as the ISA 430, the circuit allows you to transparently remove excessive sibilance from your vocal. A 'Listen' switch allows you to isolate and monitor the excess sibilance which is triggering the De-esser, rather than trying to pick out the effect within a complex full-bandwidth signal.

Latency-Free Monitoring

You can monitor your source signal within the mix directly from the unit, avoiding having to monitor via your DAW or digital desk, both of which may be prone to latency issues. A headphone mix can be created for either the artist or the engineer, with immediate and tactile control over levels, as well as the opportunity to feed in and control an external effect (e.g. reverb) for the source signal.

Output Features and Controls

A custom peak-reading meter, when combined with the fader control and an overload LED, provide superb visual indication and instant control of output levels. A 'Process Bypass' switch provides instant A/B comparisons.

PLATINUM TRAK MASTER



Never before has there been a more affordable tracking device, which still manages to encompass the design philosophy and integrity that have ensured that Focusrite be held in such high esteem over so many years. The Trak Master boasts the same high quality mic pre and compression circuits as the Platinum Penta, with performance specifications never before seen in this price range, setting a new standard for project studio signal processing. A high quality mic pre, intuitive compression, a 3-band flexible EQ, and 'tube sound' control come together to ensure you have all you require to get a quality signal tracked. You even have the option to fit a Focusrite A/D card taking you directly from the Trak Master into your digital workstation, while retaining the Focusrite signature sound of 'clarity without coloration', as you flow from analog to digital.

FEATURES

Discrete Pre Amp

Audiophile quality Class A transistor mic pre design features the wide bandwidth design philosophy featured in original Focusrites from years past. This design ensures low noise and distortion, delivering clarity without coloration, giving the signature transparency for which Focusrite have become famous. This section also features phantom power and a high pass filter, enabling it to handle any microphone with grace, while removing proximity effect and wind noise.

Direct Inputs

Features both a mic input and an instrument input on the front panel for quick and easy plug-in, without the use of a separate DI box. Use the Trak Master as a front end to your system and enjoy quality vocal and instrument recordings, tracked in seconds.

Tube Sound

Focusrite patented 'Tube Sound' technology gives you controllable tube style warmth. This unique effect is achieved using a custom, dedicated discrete MOSFET circuit, specifically designed to emulate tube harmonics. Instead of limiting your studio to one sound, now you can choose – more tube, less tube, no tube.

Output Level

Comprehensive output metering and control ensures that levels accurately matches the following processor's requirements.

Optical Compressor

Comprised of servo-controlled optical attenuators that deliver better results than equivalent VCA designs, the optical compressor gives more punch and fatter sounding compression. A single control for more or less compression is complemented by a 6 LED meter for clear visual indication of gain reduction.

'Tight' and 'Punch' switches allow easy selection of compression settings, and extra control over the dynamic range. Compression can be applied either Pre or Post EQ allowing more control over extreme dynamic effects and EQ.

Threshold: -22dB (MORE) to +12dB (LESS)

Ratio (TIGHT switch out): 3:1, soft knee

Ratio (TIGHT switch in): 6:1, hard knee

Attack (PUNCH switch out): 3ms

Attack (PUNCH switch in): 45mS

Release: 100mS (FAST) to program-dependent slow (SLOW)

Makeup gain: +20 dB

3-Band EQ

Intuitive, versatile and extremely easy to use, the EQ section boasts two modes of operation to guide you to the best sounding EQ curves. The bass control features specifically selected frequency ranges and filter configurations, stretching down to 25Hz with earth shattering results. Equally as versatile, the treble turnover frequency can be switched between 10K and 3.3K enabling you to add shine to any sound. At the flick of a switch the EQ can be optimized for either vocals or instruments.

Bass – Boost/Cut: +12/-14 dB

VOCAL mode (Bell curve)
Frequency range: 50 Hz to 800 Hz

INSTrument mode Shelving EQ
Frequency range: 25 Hz to 400 Hz

Presence – Boost/Cut: +12/-15 dB

Bell curve Frequency: 1.5 kHz

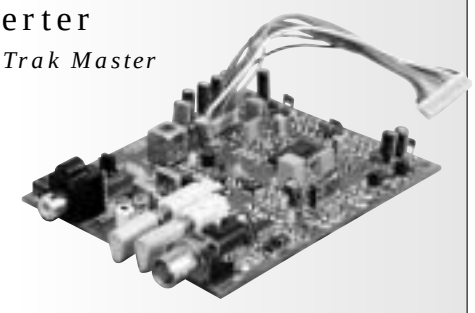
Treble Boost/Cut: ±14dB

VOCAL mode Shelving EQ frequency: 10kHz
INST mode (Bell curve) frequency: 3.3kHz

Optional A/D Converter

For Penta, VoiceMaster PRO and Trak Master

The optional high quality, 24-bit/96kHz and 128x over-sampled A-D converter card handles sampling rates of up to 96kHz with an S/PDIF output. Available for the Platinum Penta, VoiceMaster PRO and Trak Master, the board is easily retrofitted without soldering etc. Just a few screws to undo and one clip to connect.



PLATINUM QUICK REFERENCE GUIDE

	OctoPre	Mix Master Digital	Penta	Compounder	Voice Master Pro	Trak Master
Mono/Stereo	8 mono stereo	Stereo	Mono/Stereo	Dual Mono/Stereo	Mono	Mono
Mic Preamp	Yes, 8	No	Yes	No	Yes	Yes
Instrument Input	Yes, via 25-pin d-sub	No	Yes	No	Yes	Yes
Line	Yes, via 25-pin d-sub	+4db/-10db	+4db/-10db	+4db/-10db	+4db/-10db	+4db/-10db
Compressor Type	Opto w/Limiter	MultiBand Opto	Opto	VCA	Opto	Opto
Gate	No	Expander	None	Gate	Expander	none
Saturation	No	None	TubeTran Valve Warmth	None	Voice Optimized	TubeSound
24 bit/96 kHz Digital A-D Option	Yes. 2 available	Included	Yes	No	No	Yes
Filters	No	No	Hi Pass	No	Hi-Pass	No
Parametric EQ	No	1x Full Range	No	No	Voice Optimized 3-band	Yes
Passive EQ	No	No	No	No	No	No
Bass Enhancement	No	Yes	No	Huge	No	No
Presets	No	No	Yes	No	No	No
Stereo Image Widening	No	Yes	Yes	No	No	No
De-esser	No	No	No	No	Yes	No
Gate Key Input	No	No	No	Yes	No	No
Comp SideChain Input	No	No	Yes	No	No	No
Limiter	Yes, in Compressor circuit	Yes	No	No	No	No

RED 2 *Dual-Channel EQ*



The Red 2 brings two full channels of the classic Focusrite EQ to any mixing and mastering system requiring world-class equalization. Both channels feature transformer balanced inputs and outputs, ensuring excellent electrical isolation and sonic clarity and buffering between EQ bands.
The Red 2 features EQ circuits identical to those found on the classic ISA 215 dual channel mic-pre/EQ processor. A frequency response from 5 Hz to 200 kHz ensures linear audio response resulting in the classic Focusrite smooth, open high-end with no phase problems. The high and low shelving bands employ switched capacitor circuits rather than conventional variable resistor capacitor circuits, resulting in a filter curve shape that provides smooth and predictable results.

- Transformer-balanced inputs and outputs provide superb isolation while contributing to the warm sonic signature of the Focusrite EQ. (Many run their signal paths through the Red 2 to obtain this warmth whether or not they are applying EQ).

Input Gain: ±12dBu continuously variable
Input Impedance: 10kΩ ±15%, 20 - 20k Hz
Balance: >60dB
Frequency Response: 5Hz to 200kHz
Noise: Better than -98dB below +4dBu
Output: +26dBm with 600Ω output load

- Derived from the classic ISA 110, the Red 2 offers switched-frequency high and low-pass filters, switched frequency shelving bands for low and high frequencies, and two fully parametric mid bands with frequency sweep and Q controls.

– The shelving high and low bands offer up to 18dB cut and boost from 3.3 -18kHz and 33 - 460Hz respectively, with a constant filter curve shape —unaltered by frequency selection —for smooth, predictable results.

– The two fully-parametric, mid bands have considerable overlap in available frequency selection. The low-mid band is sweepable from 40-400Hz or (x3)120Hz –1.2kHz.

– The high mid band is sweepable from 600Hz-6kHz or (x3)1.8kHz-18kHz. The bandwidth (or Q) is continuously variable from 0.3 to 1, allowing a broad or very fine "peaking "or "dipping "curve to be obtained with up to 18db of gain or attenuation.

– The high and low-pass filters provide 12db per octave attenuation and are switchable in steps between 36 -330Hz and 5.6- 2200 Hz.

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ONE HOUR FREE PARKING
AT 349 W. 34th STREET (with purchase of $100 or more)

RED 1 • RED 8

Quad and Dual Mic Preamps





High resolution digital audio workstations such as Pro Tools offer very accurate and detailed audio quality. When recording at increased resolutions, it is important to use high-end mic/line preamplification to ensure that the most precise signal possible is being fed into your audio interface. The Focusrite Red 1 and Red 8 respectively provide four or two perfectly matched channels of perhaps the most sought after mic preamplification in the audio industry. Using the same circuit topology as found in the ISA 215 mic preamps, the ultra-high quality Red 1 and Red 8 provide a transparent link between the microphone and the recording destination. Their transformer-based inputs and outputs provide superb isolation and ultra-wide bandwidth. Used with high quality ribbon, valve and condenser mics, the Red 1 and Red 8 obtain outstanding results with any sound source, but especially voice, piano and string instruments.
With their distinctive deep burgundy, anoplated aluminum chassis and solid design, the Red 1 or Red 8 are at home in any critical audio recording application when the ultimate sonic integrity is in demand.

FEATURES

- With two or four mic pre-amps in a single unit, the Red 8 or Red 1 is ideal for those looking for improved audio performance, as a compact 'way in' to digital recording systems, or for location multi-mic recordings.
- Each channel offers custom-wound Focusrite input transformers, switchable phantom power, phase reverse, and easily-read illuminated VU meter, and a handy scribble disc for denoting channels.
- Mic gain is switched in 6dB steps over a 66dB range, for accurate, precise channel matching and recall.
- Benefits of the unique Focusrite mic amp topology include superb common-mode rejection, a good overload margin and, with its shared gain structure, (20dB from transformer and up to 40dB from the amplifier) a very low noise floor with the signature wide bandwidth (10Hz to 200KHz).
- High performance levels are maintained with a very wide range of impedance across the inputs.
- Output stages with their custom transformers can drive long cable runs –several kilometers – without significant loss of quality. Makes them ideal for remote recordings.
- The Red 8, with a perfectly matched pair of mic amps, offers identical channel controls to the Red 1, and is especially suited to demanding mono or stereo recording work, such as location classical recording with digital recording media.

What makes the Red Range so special?

- The Red Range is handmade, crafted to the highest standard. Red processors are made in England using machine-tooled, half-inch thick bordeaux red anodized aluminum.
- All potentiometers are made from conductive plastic which gives more sensitivity and no mechanical resistance (easy to make very small adjustments).
- The chassis features recessed section grooves and styling motifs, porthole windows, printed silver control text, and firm-response illuminated switches.
- They feature sealed relays (gold-plated silver in an inert gas), situated in the middle of the circuits in which they switch. The precious metals mean that all contacts are low resistance, and the gas prevents any kind of corrosion of the surface of the contacts, meaning the most perfect audio switching device ever designed—utterly passive, zero distortion caused and no FET's required.

FOCUSRITE

RED 3

Dual-Channel Compressor/Limiter



Employing an unusual and original single-VCA design, the Red 3 provides two perfectly matched channels of exceptional-quality and truly independent compression and limiting. Incorporating a TEC award-winning Class A VCA design results in a short signal path and extremely low noise, allowing the Red 3 to achieve truly transparent compression/limiting.

True stereo compression is provided from a single set of controls, ensuring perfect phase coherence between both channels when working with stereo audio signals. A frequency response of 5 Hz to 200 kHz provides excellent linear audio performance. With its distinctive deep burgundy, anoplated aluminum chassis and solid design, the Red 3 is at home in any critical audio recording application when the ultimate sonic integrity is in demand.

FEATURES

- The Red 3's VCA is a proprietary Focusrite design, fully discrete and balanced, offering superb low noise and distortion, and excellent common-mode rejection.
- In order to separate compression and limiting, the side-chain electronics contain three VCAs in series to generate compression and limiting control voltages which drive the main VCA. The result is true compression followed by limiting, rather than the more common characteristic of compression that turns into limiting.
- The side-chain electronics are Class A, ensuring superb transient response. Both compression and limiting therefore, offer true peak response, not averaging RMS response. The use of such a high quality audio VCA as the only element between input and output means a very short signal path, and an ability to retain a natural, unobtrusive sound even when significant compression is being applied.
- Each channel has clear and identical controls, for compressor ratio, threshold, make-up gain, attack and release, and limiter threshold.
- A program-dependent auto-release mode is available, and the VU meters can be switched to show levels in two ranges, or gain change. In stereo mode the lower set takes control of both channels.

BLUE 230 *Broadcast Dual-Channel Compressor/Limiter*



The Focusrite Blue 230 is a special version of the Red 3, optimized for broadcast applications. It features the same award-winning signal path, but with switched control of critical parameters for improved recall.

- Key to the broadcasting facility is the ability to recall/notate precise settings. A higher level of control markings across the front panel provide the user with preferred level of information, while both Threshold and Ratio potentiometers are stepped for precision control and optimum resetability.
- MU metal cans around all transformers reduce all electromagnetic radiation
- Another key requirement for broadcast is electronic isolation. The Blue 230 features transformers at both the input and output stage, ensuring total isolation and an absolute minimum level of ground-related hum.
- Machine-tooled in half-inch thick, deep blue anoplated aluminum, which is scratch-resistant and can be marked and unmarked forever.
- Chassis features recessed section grooves and styling motifs, porthole windows and firm-response illuminated switches.
- Relays are gold-plated silver in an inert gas, which means that the contact is low resistance, and the gas prevents any corrosion of the surface of the contacts – the most perfect audio switching device ever designed – utterly passive with zero distortion.

RED 7

Mic Preamp and Channel Strip



The Red 7 combines a single channel of classic Focusrite microphone preamplification (the same superb mic preamp as the ones on the Red 1 and Red 8) with a full dynamics section, optimized for vocals. In addition to the mic preamp, the Red 7 features a single channel compressor from the Red 3 and adds a de-esser/exciter. So now you can get the warmth of classic Focusrite mic preamps and the transparent, smooth characteristics of their compressors in one affordable package. With its superb signal path, outstanding ease of use, and integrated all-in-one design, the Red 7 is a powerful tool for voice recording in all situations from music studios to post- production.

- Features the classic Focusrite transformer-based inputs and outputs for outstanding isolation and superb sonic performance.
- The Red 7 employs the same superb mic pre-amp as the Red 1 and 8, with a dual-range mic gain pot that allows precise control across the full range of the device. Phantom power and phase reverse are also provided.
- The Line input is electronically balanced, with continuously variable gain. A high quality output fader offers +6dB gain to infinite attenuation, essential for direct recordings, or for accurate level matching after EQ and dynamics.
- The compressor design is taken straight from the Red 3, and as on that unit, is characterized by low noise and distortion even with heavy processing. Relatively heavy compression can be applied while retaining a transparent natural sound. Ratio, gain make-up, threshold, attack and release are all continuously variable, and a program dependent auto-release mode can be switched in. A swept high-pass filter allows effective treatment of problems such as rumble, bass lift and proximity effect.
- Illuminated push-button switches for compressor in/out, de-esser/exciter in/out and high-pass filter in/out.
- The de-esser/exciter stage can be switched to provide frequency-selective compression (de-essing) or excitation. The amount and center frequency of the processor are continuously variable, and as all the unit's dynamics still use the Focusrite single-VCA technique, unwanted coloration of the sound is kept to an absolute minimum.
- A clear VU meter shows level after the pre-amp gain control, or display compressor gain change.

	Red 1	Red 2	Red 3	Red 7	Red 8
Mic Input Gain	-6dB to +60dB in 6dB steps	Not Applicable	Not Applicable	-6dB to +60dB in 2 variable ranges	-6dB to +60dB in 6dB steps
Mic Input Impedance	1200Ω ±15%, balanced and floating	Not Applicable	Not Applicable	1200Ω ±15%, balanced and floating	1200Ω ±15%, balanced and floating
Frequency Response	10Hz to 140kHz (-3dB points), ±0.1dB within passband	5Hz to 200kHz (-3dB points) ±0.1dB within passband	5Hz to 200kHz (-3dB points)	10Hz to 140kHz (-3dB points), ±0.1dB within passband	10Hz to 140kHz (-3dB points), ±0.1dB within passband
Noise (EIN)	-127dBu (input loaded 200Ω) @ 60dB gain	Better than -98dB below +4dBu	Better than -80dB below +4dBu output level	Better than -80dB below +4dBu output level dynamics level	-127dBu (input loaded 200Ω) @ 60dB gain
Distortion	0.15% (-20dBu @ 20Hz) 0.06% (-20dBu @ 40Hz) 0.003% (-20dBu @ 1kHz) 0.005% (-20dBu @ 10kHz)	0.016% (-20dBu @20Hz) 0.02% (-10dBu @20Hz) 0.03% (0dBu @20Hz) 0.03% (0dBu @20Hz)	0.02% with input at 0dBu at 1kHz 0.006% (+10dBu @ 1kHz) 0.004% (+20dBu @ 1kHz)	0.006% (+10dBu @ 1kHz)	0.15% (-20dBu @ 20Hz) 0.06% (-20dBu @ 40Hz) 0.003% (-20dBu @ 1kHz) 0.005% (-20dBu @ 10kHz)
Output	+24dBm into 600Ω +26dBm into 10kΩ, balanced and floating	+26dBm with 600Ω output load balanced and floating	+24dBm with output loaded 600Ω, balanced and floating	+24dBm into 600Ω +26dBm into 10kΩ, balanced and floating	+24dBm into 600Ω +26dBm into 10kΩ, balanced and floating

ISA 220: SESSION PACK

Channel Strip



The ISA 220 Session Pack provides all the audio precision tools required to infuse your session with Focusrite's renowned sonic performance. It features many of the original circuits of the flagship ISA 430 Producer Pack, and also includes some new facilities of its own. It also features the same digital output option as the ISA 430, providing you with a high quality digital route direct from the ISA 220 into your digital recording system.

With a topology based on the flagship ISA 430, the ISA 220 is designed for professionals in music and post. A channel strip style processor, the 2RU high ISA 220 includes a transformer-coupled mic/line/instrument preamp, 4-band EQ with filters, compressor, optical de-esser and frequency adaptive limiter.

FEATURES

Input Stage

- Input stage features a precise VU meter, switchable to allow monitoring of either input level or compressor gain reduction
- Above the VU meter is a 32 LED digital meter for monitoring both the 'internal' input and the rear panel 'external' input (the latter routes directly to the second channel of the optional A/D card).
- A global bypass switch next to the digital meter allows for instant A/B comparisons
- Mic, line and instrument inputs are all transformer-based, the cornerstone of the Focusrite signature sound, clear yet warm, with incredibly low noise figures and a THD below 0.001% on the mic input!
- Switching between these input options is available on the front panel with LED's indicating the active input.
- The Mic Pre is the classic Focusrite transformer-based design with illuminated switches for phantom power and phase reverse, all available on the front panel
- From the front panel you also have control over the digital output option and in/out selection for each of the processors. An instrument jack is also provided on the front panel for quick 'plug and play' access.

De-esser

- Same De-esser as the ISA 430, based on a low distortion optical technology design, letting you transparently remove excessive sibilance from a vocal performance.

EQ Section

- The EQ featured on the ISA 220 is similar to the ISA 430 and ISA 110, with the exception of the Shelving EQ's having four frequency selections rather than six. Whether fattening-up bass, fore grounding or trimming middle, or adding airy top, this EQ can handle anything with grace and ease.

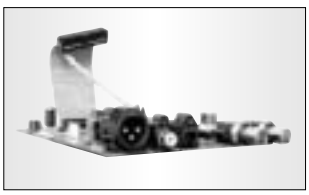
Compressor

- Next to the compressor section (the same compressor as the ISA 430) is a 'Compressor Pre EQ' switch. This allows you to place the compressor either pre or post-EQ.
- Alongside the auto release is the Blend control, a unique function in the ISA 220 that allows smoother compression at more extreme settings. When switched in, 'Blend' lets you mix the uncompressed signal with the compressed,thus retaining the power of the dynamics from the original source.

Output

- The output section of the ISA 220 features the same frequency adaptive limiter as the ISA 430, a design which uses three separate stages of optical-based circuits, each of which has different limiting properties to give true distortion-free limiting.
- Fast limiters tend to have problems dealing with complex signals that contain sustained low and mid frequency information and thus "chop holes" in the audio when HF transients trigger the limiting. To overcome this, the ISA 220's frequency adaptive limiter has three frequency bands with different attack times as follows: LF slow, MF quick and HF ultra-quick, the latter designed to catch fast transients. An upper threshold is fixed at +20dBu to prevent overload of the internal (or an external) A/D converter. Finally, a variable control adjusts the module output level between -60dB and +6dB ensuring optimum output levels.

OPTIONAL ISA 220 A/D CARD



An optional 24-bit/ 96 kHz digital output (AES/EBU, S/PDIF and TOSLINK) offers 44.1 - 96 kHz operation, 16, 20 and 24 bit resolution selection and can lock to external Word Clock, including Pro Tools Super Clock. It can be used as a high-quality mono/stereo tracking converter or at mix down as a stereo converter for final mastering.

B&H PHOTO - VIDEO - PRO AUDIO EQUIPMENT LEASING AVAILABLE

ISA 430: PRODUCER PACK

Super Channel Strip



Stepping up from the ISA 220, the ISA 430 adds proprietary Focusrite discrete Class A VCA Gate and Expander designs, plus a host of very thorough routing and monitoring options. The discrete insert send and return may be placed before EQ or Dynamics, between the two or after both. Each band of the EQ, including the filters, can be routed to the sidechain of the compressor or gate for accurate drum gating or frequency selective compression. The classic, large analog VU meter can be switched to display input, insert return or gain-reduction level. "Listen" mode allows monitoring of the sidechains of the Compressor, Gate and De-esser so that accurate frequency adjustment can be made when tailoring the various dynamic effects. Another very unique feature of the Producer Pack's routing capability is "split mode". The ISA 430 can be run as a single channel "Producer Pack" or split, to act as two independent processors running discrete audio paths.

SAME FEATURES AS THE ISA 220 *PLUS*—

- The Gate section uses the Focusrite Class A VCA as the control element to remove the effect of unwanted interference and high levels of wide-band system noise build-up. The Gate acts as a straight-forward on/off control, switching off the audio path when the signal level drops below the threshold set, killing noise in the quiet passages of a performance.
- Switching this section to expand mode causes the Gate to function as an Expander, which gives a more gentle gating effect; instead of cutting any signal below the threshold, an expander proportionately decreases it. This will give a more natural sound when reducing noise from non-percussive sources (especially vocals). The adjacent LED meter indicates in dB the amount of gain reduction caused by the Gate/Expander. As with the compressor, this section offers both the 'Listen' and 'Ext key' features.
- 'Hysterisis' increases the level-difference between the gate switching on and off, and prevents the gate oscillating ('chattering') with particular combinations of input signal and threshold settings. This function is particularly useful when gating a signal with a very long decay time and large amounts of level modulation (for example a Grand Piano).

Insert, Pre EQ and Split Dynamics Switches

The ISA 430's real benefit as a "do-all" processor for demanding engineers are the insert, pre EQ and split dynamics switches for controlling the position of the sections within the signal flow.

Insert Position

The Insert Send and Return can be positioned in three places within the signal flow for maximum versatility.

Pre—Insert is after the input trim, but before EQ or Dynamics

Mid—After EQ but before Dynamics

Post—After EQ and Dynamics but before the main output

Dynamics Pre EQ

Normally the EQ section precedes compression. Pressing Pre EQ reverses the position of the EQ and Dynamics sections, placing Dynamics first and EQ afterwards. This function gives greater flexibility in the order of the processing blocks and the relative position of the insert point when combined with the Pre, Mid and Post selector, allowing EQ changes to be independent of compressor section

Single/Split Mode Selection

The ISA 430 can be run as a single channel "Producer Pack", or split to act as two independent processors running discrete audio paths. Split mode allows the Insert Return and Insert Send to act as independent inputs and outputs to the dynamics section only, creating two devices in one, with separate EQ and Dynamics channels—perfect for mix down problem solving.

ISA 220 AND ISA 430 SPECIFICATIONS

Inst. Hi Z input

Noise:	-96dBu
THD:	0.003% with 0dBu

Mic

Noise:	123dB EIN with 150Ω input resistance at 60dB of gain
THD:	0.0008%

Compressor

Threshold Range:	-28dB to +12dB
Ratio:	1.5:1 to 10:1
Slope:	Soft knee
Attack:	500µS to 25mS
Release:	100mS to 4S, variable or auto (program dependent)

De-Esser

Threshold Range:	22dB
Frequency Range:	2K2 to 9K2
Ratio at Center Frequency:	2:1

Limiter

Threshold Range:	20dBu
Ratio:	infinite (Brick Wall)
Attack:	Frequency dependent

ISA 428: PRE PACK

4-Channel Mic Preamp



The ISA 428 Pre Pack features four Rupert Neve-designed, transformer-based microphone preamps with selectable impedance, direct instrument and line inputs, and an optional 8-channel 192 kHz analog-digital converter. An all-new Soft Limiter circuit precedes the A-D, providing the perfect path into any DAW, digital console or stand-alone hard disk recorder. The ISA428 also functions as the perfect input expander for mixing consoles, featuring the four mic pres, four instrument inputs, and 8 line inputs. Any combination of 8 of these inputs can be routed discretely to the optional 8-channel A-D converter. Equally suited for traditional stand-alone preamp roles in broadcast, post-production, live sound and installations.

FEATURES

Four Classic Neve-Designed Mic Preamps

- Incorporates four of the original Rupert Neve-designed transformer-based preamps found in the classic ISA110's. The benefits of this pre-amp topology include superb common-mode rejection, an excellent overload margin, and, courtesy of the shared gain structure (20dB from the hand-wound transformer and up to 40dB from the amp), an extremely low noise floor and super-wide bandwidth. This pre-amp design is the cornerstone of Focusrite's signature 'warm-yet-transparent' sound and is a standard point of reference for many of the industry's most discerning audio professionals.
- Four front panel instrument inputs—no separate DI box required.

Switchable Impedance

- The input stage also provides enhanced control and creativity, by allowing you to switch between four carefully selected input impedance settings—the original ISA110 setting plus three more impedance settings, to either perfectly match the pre-amp with any microphone (and so maximize level,) or to use different settings creatively to interactively shape the sound of your chosen classic mic.
- Switchable insert points are also featured on every channel, allowing routing, (between pre amp and output,) to additional external processing if desired.
- Full output metering for all 8 channels is provided by 6-segment LEDs on the ISA428's front panel, while input levels can be tracked using 'moving coil' peak reading VU-style meters, designed to catch even the fastest and most elusive of transients.
- Two ISA428's can be used with a single A-D converter utilizing the four extra line input channels on the rear of the unit containing the optional A-D converter. This allows expansion from a 4-pre system to an 8-pre system – hence the name "Four t(w)o Eight!"

Custom Global Soft Limiting

- Protecting the A-D circuit is Focusrite's "Soft Limiter," a custom optical design, which both protects the A-D converter by preventing 'digital overload', and also eliminates the unpleasant distortion that standard limiter circuits often generate. Alternatively, it provides worry-free, musical A/D conversion, handling even the hottest analog signals with style and grace.
- Rather than simply protecting the A/D in a brick-wall fashion, the Soft Limiter tailors the last 6dB dynamic range of the A/D converter to cater for the last 12dB of analog headroom. This ensures the A/D converter never overloads at any point during the analog performance. Unlike conventional limiters, it provides an absolute limit without destroying the audio integrity of the source signal.

Optional 8-channel 192kHz A-D Converter

- Embodying cutting-edge conversion technology, encompassed within pristine Focusrite circuitry, the optional A-D converter provides eight channels of the highest quality conversion at sampling rates of 44.1, 48, 96 and 192 kHz. (Running at 96kHz provides 16 simultaneous digital outputs alongside the four main analog outputs.) Digital output formats include 8 channel single/dual wire AES/EBU, S/PDIF and ADAT lightpipe, all available on a single card.

ISA 428 INPUTS:

	GAIN RANGE
Line	±18dB
Mic	0 to 60dB (both in 6dB steps)
Inst	+10 to +40 dB variable

	INPUT IMPEDANCE
Line	10K Ω
Mic	Variable -600 Ω,2K4 Ω,6K8 Ω and original ISA110 settings
Inst	>1M Ω

NOISE			
Line	-96dB	**Mic**	-128dB (EIN with 150 Ω input resistance at 60dB of gain)

RADIUS 3 FAT MAN

Stereo Tube Compressor



The Radius 3 Fat Man provides instant access to 15 high quality, stereo tube compression presets, making it easy to add fat, professional sounding tube compression when tracking, mixing or playing live. Created in consultation with leading audio engineers, including Pink Floyd's Andy Jackson, the programs ensure a polished, professional sound to your mix. The pre-sets include optimum compression settings for vocals, keyboards, bass, acoustic and electric guitars, snare, kick and whole drum kits and pop, rock and dance mixes, making the Fat Man a very user-friendly tube compressor.
For those who wish to create their own compression settings, a manual mode provides fully adjustable control over threshold, ratio, knee, attack and release settings. Input / output gains and gain make-up controls remain adjustable in both preset and manual modes to allow gain balancing for different systems. A bypass switch is provided to compare compressed and direct signals and a vintage-style VU meter can be switched to meter output level or the amount of gain reduction. In addition to line inputs and outputs, the Fat Man also features front panel instrument inputs for the direct injection of guitars, basses and keyboards.

- Balanced 1/4″ analog I/O, unbalanced compatible
- -10 / +4dBu switchable
- 1/4″ jack instrument inputs
- 15 preset settings and a manual mode
- Compressor with adjustable threshold, ratio, attack, release and knee
- Adjustable input and output gain
- Gain make up
- Compressor bypass
- VU metering of output level or gain reduction
- Optional 19″ rack-mount kit holds 2 units

RADIUS 5 FAT MAN 2

Tube Preamp and Compressor

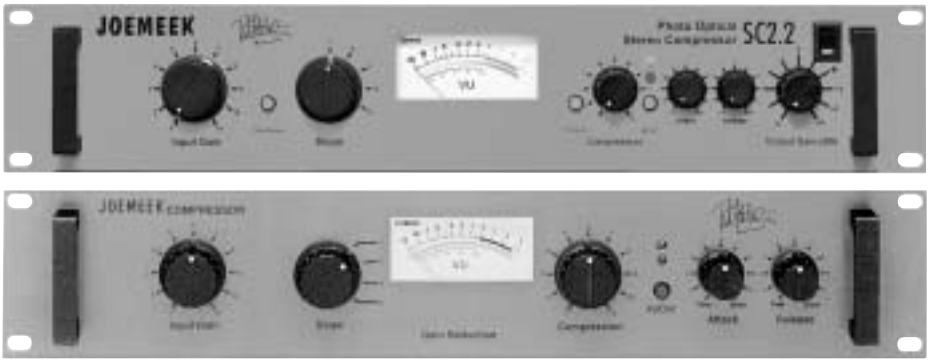


The Fat Man 2 is an affordable, high quality tube preamp and compressor, offering an ideal way of adding tube warmth and punch to recordings and live performances. It can also be used as an instrument preamp, a vocal processor or as a 'warm' front end for a digital recording system. The Fat Man 2 has 15 high quality presets, including five vocal compression settings, with other presets for guitars, keyboards, basses and drums. A fully manual mode controls individual compression parameters, including the threshold, compression ratio, knee, attack and release settings. Input and output gains are adjustable in both manual and preset modes, while a gain make-up control enables quick and simple balancing of the original and the processed signals.
The first stage of the Fat Man 2's dual-stage 12AX7A triode tube is housed in the preamp, with the second stage in the compressor section. The tube preamp accepts mic, line and instrument inputs. A source switch selects either the mic or line input and also adjusts the sensitivity for the instrument input between high and low. Phantom power is switchable and there is a 90Hz high-pass filter to remove unwanted rumble or hum.

JOEMEEK

SC2 • SC2.2

Stereo Optical Compressors



The JOEMEEK SC2 and SC2.2 are two channel stereo photo-optical "effects" compressors designed, not to perform protective functions like modern VCA compressor/limiters, but to be used as artistic tools that produce the psychoacoustic effect of power and dynamics while at the same time, retaining sparkle and adding warmth to your audio tracks. Due to the very nature of photo compression along with the use of modern servo control techniques and electronically-balanced operation, the SC2 and SC2.2 are virtually free from noise and distortion. Both units are easy to setup and operate with variable input and make-up gain, attack and release controls as well as JoeMeek's unique interactive Compression Drive and Slope (ratio) controls. The SC2.2 also features a 'Dark Mode' switch that allows you to choose between the classic JoeMeek sound and a modern 'bright' punch mode that allows more precise level control. Both units offer the distinctive JOEMEEK sound (and bright green faceplate) that cannot be duplicated by any of today's digital technology. The SC2 is considered the "Classic" JoeMeek compressor which delivers uncompromising audio quality sought after by top producers, engineers and musicians throughout the world. The SC2.2 uses the same cell compressor design as the SC2 but is made more affordable by only offering XLR outputs (as opposed to XLR and 1/4˝ TRS) and uses a smaller chassis.

FEATURES

The Stereo Compressor

- Recreates the warmth and power of the 60's style analog tape and compressor combination without the side effects — Photo-electrical compression is achieved using modern servo-control techniques that ensure the accuracy and speed of response of the light source
- No distortion producing voltage controlled amplifiers (VCAs) are used. Although the compression is non-linear, the effect is true volume compression with no "limit" action
- Before the left and right channels are introduced to the compression circuit, they are converted to 'sum and difference' mode. This ensures that the centre image stays perfectly in the centre even under heavy compression. The sum and difference coded signal is compressed, and then converted back into 'left/right' format

Controls

- Since the two compressor channels are linked as a stereo pair, a single control is provided for each parameter that controls both the left and right channels
- Separate rotary controls are provided for adjusting input gain (approx 20dB max) and output (make-up) gain (+26dB max)
- The rotary Attack control is adjustable between 1.5 and 10 ms
- The rotary Release control adjusts the recovery time of the compression. The wide range gives the option of high speed "pumping" down to very slow gain riding
- An In/Out 'hard wired' bypass switch allows instant comparison between the original uncompressed, and the compressed signals

Interactive Compression Control

Instead of the standard 'threshold' and 'ratio' controls, found on conventional compressors, JoeMeek compressors use interactive Compression and Slope controls that work together in a musical, non-linear (non-predictive) manner.

- The rotary Compression control adds drive (gain) to the optical system. It not only controls the 'Threshold' level, it helps define the 'knee' characteristic of the compressor causing the ratio to increase along with program content and amplitude. In simple terms, winding up the compression control increases the amount of compression
- The 4 position rotary Slope switch is similar to a ratio control but is not linear as it reacts with the Compression control — At the lowest rate (1), the maximum compression ratio is about 2:1 with a maximum possible compression of about 6dB. At the highest rate (4), the compression ratio usable maximum is about 8:1

SC2 • SC2.2

Inputs & Outputs

- Dual XLR and 1/4″ TRS balanced line level inputs and outputs are provided on the rear panel but can be operated unbalanced without loss of quality (the SC2.2 has only XLR ins and outs)
- Inputs are precision-electronic balanced using the "Superbal" principle, invented by Ted Fletcher in 1976, that delivers excellent noise rejection and sonic performance
- Outputs are electronically balanced and floating at a source Impedance of 100 Ω

Metering

- A large illuminated VU/COMP meter is switchable between reading audio input level and gain reduction for both channels

System Performance (same for both units)

System Gain	Max. 20dB
System Noise (20Hz to 20kHz)	Generally better than -94dBu
System Distortion	Within 0.004% 100Hz to 10kHz at all levels
Frequency Response	Substantially flat 5Hz to 30kHz with minimal phase shift at extremes
Crosstalk	Better than -60db at 10kHz
Push button Hardwire Bypass with LED indication	Blue (in operation) Red (bypass)
METER VU/compression meter	Switchable between input level or gain reduction
Power	115/230VAC (switchable) 10 watts
Housing	2RU Steel case enclosure

Additional

- The units can be used as single mono processors by just driving one side only
- A mains voltage selector lets you change the input voltage to operate between 95 and 125V and between 210 and 240V AC

The SC2.2 Adds

- The Dark Mode switch gives the option of using a 'conventional' sounding compression effect, or the more dramatic and original 'JOEMEEK' sound — In 'Dark Mode', the time curve for the release of the compression behaves in a much less predictable way
- A fifth slope position has been added for achieving extreme compression effects — The attack time is also faster adding to the fifth slope positions more whacky effect sound
- Detented pots make it easier to return to preferred settings

The JoeMeek Compressor:
What It Does

Anyone who has tried to record a human voice in the simplest possible way has found that the dynamics of real world speech and music are impossible to handle with a 'linear' recorder: Even if the signals do not overload, the final result is a recording that seems to be thin, quiet and too wide in dynamic range.

Early analogue tape recorders had a built-in answer to the problem, Slightly overloading the record channel produced (predominantly) 2nd order harmonic distortion and some volume compression. This made recordings sound 'warm' and reduced the problems of dynamics. Unfortunately (?), modern recording mediums don't react that way; they produce an accurate reflection of the input with all its built in problems.

Most thinking engineers know of these problems and correct with the use of equalisers (to change the psychoacoustic distance from the microphone), limiters (to reduce dynamic range at the louder end) and enhancers (to sparkle it up). And then find that it is extremely difficult and time consuming to get a warm and cosy sound**.

Joe Meek, in common with all engineers, experimented with the limited facilities of the time (1960 - 1965) and came up with a way of enhancing the 'tape bend' effect with compression. The compressor he used was primitive in the extreme, but coupled with the compression distortion provided by the valve tape machines he had, the result was voice and music sounds that were unique and sold records by the million.

I had to make some significant improvements in the way the early compressors worked to make the JOEMEEK acceptable to today's engineers, but now that they are in general use in top studios all over the world, the consensus is that the sound is right, and really does recreate the warmth and power of the old equipment without the side effects.

** The classic way is to use a U47 or similar large diaphragm capacitor microphone; which in itself 'distorts' the sound with complex high frequency phasing across the diaphragm and some 2nd order distortion from the amplifier tube.

There are probably hundreds of types of compressor/limiters on sale in the world, all struggling for a place in the market with extra knobs and facilities and all missing the fundamental point;- that a compressor is there to produce a psychoacoustic effect of power and dynamics. not to perform any protective function electronically; it's a creative tool.

The JOEMEEK recreates the dynamics of the old analogue tape and compressor combination. And yes; it is entirely possible that one day this algorithm could be written for a digital processor —but would it be as easy to use?

VC1Qcs

Studio Channel with Current Sense Mic Preamp



The VC-1QCS "Studio Channel" offers six separate pieces of equipment all in one 2U 19˝ package: a CurrentSense Microphone Preamp, a JOEMEEK "photo-optical" Compressor, Instrument Pre-Amp, a JOEMEEK Enhancer / De-esser and a full channel of the 3-band JOEMEEK Meequalizer "EQ". Other features include balanced XLR and 1/4˝ TRS line-level outputs, comprehensive input and gain reduction metering as well as LED status indicators for each processing stage. In addition, an optional (VC1QD) card, that plugs right into the rear of the VC1QCS, provides a 24 bit 96K digital SPDIF output so you can connect the Studio Channel directly to your DAW.

FEATURES

CurrentSense Preamp Design

- Optimizes microphone performance by automatically loading the correct impedance to required by the microphone
- A floating balanced input easily accepts microphone levels from -70dB up to more than 0dB and provides (microphone) input headroom of more than 30dB
- Other benefits include a more even and extended frequency response as well, any mic can be used regardless of impedance requirements and cable resistance/quality is no longer relevant
- Individual switches with LED indicators are provided for mic/line selection, phantom power, a hi-pass filter, and phase reverse
  - ➢ The 12dB per octave (3dB down at 75Hz) High Pass Filter removes extreme rumble frequencies
  - ➢ A Phase Reverse switch allows you to reverse the signal polarity

Compressor

- The Compression control adds gain to the compression sidechain thereby increasing the compression effect
- A five position Slope control sets the ratio from gentle compression to pumping effects
- The Attack control sets the time that the compressor takes to act
- The Release control sets the amount of time it takes gain to return to normal after compression

Enhancer / De-Esser

- The enhancer uses a drive and resonance control that, when used judiciously, allow you to add a unique sparkle and brightness to sounds that seem otherwise "flat". The effect is particularly well suited for vocals, guitars and acoustic instruments
- The enhancer works by picking off the higher frequency part of the sound, compressing and dynamically altering it, filtering off the original sound and remixing the resulting harmonics back with the signal.
- De-essing is made possible by turning the enhancer pot counter-clockwise

The EQ

- One complete channel from the 3-band VC5 "Meequalizer" delivers a musical sound, unlike any software plug-in or analog project studio mixer EQ
- The Treble control is a shelving filter with a boost/cut of approx ±18dB at a fixed frequency of 8kHz
- The Mid band is a sweepable EQ filter with a boost/cut of approx ±16dB and a frequency control variable from 600Hz to 3.5KHz sweep variable. The Q (bandwidth) value of the mid frequencies varies (increases) with frequency.
- The Bass control is a shelving filter with a boost/cut of approx ±18dB at a fixed frequency of 100Hz
- An In/out switch (bypass) with indicator is provided

Metering

- A large illuminated VU meter is switchable between reading audio input, and gain reduction (compression)

Gain Make-Up

- A high quality make-up gain amplifier with a level control, located post-VU meter, allows you to compensate for gain changes due to compression or EQ

Inputs & Outputs

- The Studio Channel is optimized for the three main types of inputs found in recording studios —
  - ➢ XLR microphone inputs are available on the front and rear of the unit
  - ➢ A balanced 1/4˝ TRS line input is provided on the rear
  - ➢ An unbalanced 'instrument' input is also provided on the front panel
- A 1/4˝ TRS and an XLR connector provide high level electronically balanced line outputs
- A 1/4˝ TRS insert point allows you to add an external processor into the signal chain

24-bit / 96kHz Digital Output

- The user-installable VC1QD digital option card allows you to add a 24-bit S/PDIF or AES/EBU output with a 115 dB dynamic range
- One switch allows you to select 44.1, 48 kHz sample rates while a x2 switch doubles the sample rates to 88.2 or 96kHz

TwinQcs

Dual Studio Channel with Current Sense Mic Preamp



The TwinQcs offers JOEMEEK's Studio Channel in a compact, dual channel format. Each channel features the same 'CurrentSense' mic pre as in the VC1Qcs, with the ability to optimize its impedance with any microphone or instrument source, resulting in a beautifully rich and full sound from all sonic sources, i.e. from condenser to ribbon mics, to passive guitars. Each mic preamp also features switchable phantom, High Pass Filter and polarity reverse. The optical compressor is configurable as a dual mono or linked stereo processor — perfect for tracking stereo keyboards or as an insert on the mix-buss. The classic three band Meequalizer is also on hand to provide superb tonal control. In addition to balanced mic and line inputs on the rear panel, there is also a specially optimized instrument input on the front of each channel for direct connection of an electric guitar or bass. Add the optional VC1QD digital output card and you can capture every nuance of the JOEMEEK sound to your digital workstation.



VC6Qcs

British Channel

Hand made in England, the VC6Qcs is a single rack space channel strip designed to give you the best possible performance from any audio source whether tracking from a microphone or direct instrument input or during final mixdown. It combines the warmth and character of three legendary JOEMEEK components – the CurrentSense impedance matching mic pre amp, JOEMEEK photo optical compression, and 3-band Meequalizer. Individual bypass switches for the compressor and Meek-Q help you to lower the noise of recordings. The Dual 1/4˝ Superbal outputs provide separate recording and monitor outputs – ideal for latency-free computer based recording.

5-Stage Preamp

- Transformerless input stage and a low-noise input amplifier delivers high headroom
- 5-stage amplifier design, controlled by a single volume control, gives (microphone) input headroom of more than 30dB; the input is floating balanced and will easily accept mic levels from -80dB up to more than 0dB (100dB with the pad engaged)
- Current Sense mic preamp design significantly improves the performance of dynamic microphones
- Input Gain control provides headroom greater than 60dB line (30dB mic)
- Switchable +48v phantom, 20dB pad and phase reverse as well as a peak LED

JOEMEEK Compressor

- Legendary photo optical compressor provides classic warmth and punch
- Controls are provided for 'compression', threshold, fully variable slope (ratio).
- Compression ratio varies from 1.5 to 1 up to 7 to 1 depending on musical content and the setting of the 'slope' control
- Attack is variable from 0.5 to 5 ms. and the release from 250ms to 5 seconds
- A 5-segment gain reduction LED meter and In/Out switch with blue status LED

3-Band Meequalizer

- Meequalizer provides the smooth tones of vintage EQs – ideal for tracking or mixdown
- LF and HF shelf, plus parametric mid sweep each with ±15db boost/cut
- In/Out bypass switch and status indicator

Output Stage

- 9-segment LED smooth tracking LED input meter to helps you control signal levels
- Dual 1/4˝ TRS 'Superbal' outputs will also accept unbalanced outputs without any gain loss – one output can be used for recording while the other is available for monitoring

Additional Features

- Pre-compressor/EQ insert point allows you to add additional effects or processing in the signal chain
- Front panel 1/4˝ instrument input
- XLR mic and 1/4˝ line inputs on the rear panel
- Extended 10Hz to 25KHz frequency response (+0, -0.5dB) delivers clearly defined bass

JOEMEEK

VC8 FATHEAD

Instrument Head



The FATHEAD VC8 is a single channel front end for musicians and vocalists that combines JOEMEEK's acclaimed CurrentSense preamp design, opto-compressor and Meequalizer with a unique opto-distortion designed specifically for use with amplified instruments such as guitar and bass. Inputs are provided for a microphone or direct insertion of an instrument. Two "Superbal" outputs as well as a DI output with a gain control allows you to use the Fathead with a guitar amp, mixer and recorder (simultaneously if need be). The included Fathead Controller allows you to bypass the EQ, distortion and compressor as well as mute the outputs.

FEATURES

- 1/4″ instrument input on the front panel as well as 1/4″ line, XLR mic inputs on the rear panel.

CurrentSense Instrument/Mic Preamplifier

- The latest generation CurrentSense pre amplifier circuit gives ultimate flexibility and quality whether using line, mic, or instrument sources – equally at home as an on stage head unit, or in the studio as a pre amplifier for a mic or instrument
- Switchable +48V Phantom, 20dB pad and Phase reverse

Opto-Distortion

- Provides vintage analog distortion ranging from valve warmth through to relentless screaming fuzz
- Unique Optical Distortion feature, with in/out and optical drive
- In/Out bypass switch

Classic JoeMeek Optical Compressor

- 5 position preset switch with settings ranging from 'Warm Meek' for subtle warmth, to 'Super Joe'- for absolutely massive crunching compression
- Bypass control
- 4-segment LED gain reduction meter
- Threshold control
- Automatic gain compensation
- Signal sensitive ratio



3-Band 'Meequalizer'

- LF and Mid sweep, and high shelf controls optimized for instruments
- Highly interactive, overlapping LF and Mid sweep controls allow phasing and other classic effects, while always maintaining musicality – even at extreme settings

LF Range	150 to 1k Hz
LF Boost/Cut Range	±16dB
MF Range	1k to 5k Hz
MF Boost/Cut Range	±16dB
HF (fixed)	Centered at 8kHz
HF Boost/Cut Range	±14dB

Output Stage

- A 5 segment input LED meter shows the level of the signal before the output gain control
- Output gain knob controls the level of the main and optional digital outputs
- A silent operating Mute function, operated from the front panel and the foot switch, aids in instrument changes, and silent studio operation by cutting all sound from the main outputs, and DI output, while allowing audio to be passed the tuner output
- A 1/4″ tuner output jack
- An 1/4″ TRS insert jack allows you to add external effects processors

Outputs

- Dual floating balanced 1/4″ TRS main amplifier outputs
- 1/4″ DI output with pre/post switch and independent gain control
- Optional 24bit/96kHz digital output via the VC1QD module
- Ground lift switch
- Includes the Fathead Controller Footswitch with EQ, distortion, compressor and mute controls.

VC1QD – 24-bit / 96kHz Digital Output Option



- User installable 24bit Coaxial S/PDIF output switchable between 44.1/48, 88.2/96KHz sample rates
- Compatible with the VC1Qcs, TwinQcs and FatHead preamplifiers
- 115dB dynamic range

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EQUIPMENT LEASING AVAILABLE

MicroMeek Series

MC2 Stereo MicroMeek Compressor



The MicroMeek MC2 is a professional quality stereo compressor that gives you the full sound of the JOEMEEK photo optical compression circuit in an affordable package designed for the project studio. With only a few simple controls, the MC2 will allow you to add vintage warmth, sparkle and power to vocal tracks, instruments, subgroups – even entire mixes.

- Unparalleled build, design and sound quality gives project studios, and musicians access to the true sound of real photo optical compression at an affordable price
- Auto-variable slope (ratio) control allows a wide range of compression and limiting sounds to be achieved with effective ratios from 1.5:1 up to 14:1
- Variable input and output gain controls as well as variable attack, release controls
- DIC (Dynamic Image Control) utilizes M&S sum and difference encoding and decoding to achieve perfect stereo imaging and balance across it's stereo optical compressor
- TTT (Transparent to Transients) optical circuit design allows transients through unharmed. This allows more aggressive compression without dulling the sound
- 9-segment LED gain meter and 5-segment LED compression metering
- Dual 'superbal' input and output stages provide and optimized signal path that minimizes common mode interference
- Although designed as a stereo compressor, the MC2 can also be used as a single channel mono compressor
- 1/4″ TRS inputs and outputs accept both balanced and unbalanced signals

MQ3 MicroMeek with Current Sense Mic Preamp



The MQ3 Pro Channel is a 1U half-rack channel strip that packs the sonic punch and features of its big brothers including the "Current Sense" microphone pre amplifier, the JOEMEEK compressor, the stunningly musical JOEMEEK Meequalizer "EQ" and gain make-up output volume control. Don't be fooled by its small box — the MQ3 offers almost identical sounds and quality as JOEMEEK's more expensive preamps, with the ability to add warmth and thickness into the pure and clinical sounds of modern digital equipment, but at a fraction of the price. The Pro Channel truly delivers world class sound at an amazing price, and is a must for the user whose quality requirement is greater than their budget.

Mic Pre

- Current Sense auto-impedance matching mic preamp with a variable input level from-70dB to 0dB and switchable phantom power

The Compressor

- The compressor channel provides the same sonic capabilities as the VC1Qcs but is a little easier to use — a single rotary pot is used to control that compression amount as well as (auto ratio) compression ratio which varies from 1.5 to 1 up to 7 to 1 depending on musical content
- Attack and release controls are also available

The EQ

- The same 3-band VC5 "Meequalizer" channel as the VC1Qcs except for the Mid band which has a fixed frequency of 1.8kHz

Inputs

- XLR balanced microphone input, as well as a 1/4″ TS jack suitable for any line or instrument level audio signal
- A 1/4″ mix input (pre compressor) mixes with the normal mic or line inputs. It can be used to add a second signal post EQ and pre compression
- A 1/4″ TRS insert point located after the mic amp and pre compressor allows you to add an external processor

Outputs

- Dual balanced low impedance 1/4″ TRS line level outputs, connected in parallel, can be used simultaneously as a recording output while the other is used for monitoring/PA.

Metering

- A five stage LED meter indicates power on and output levels from -12dB to +8dB
- A red overload LED (post EQ) indicates that a signal is within 6dB of overload

Additional Features

- External power supply
- 1RU 1/2 rack aluminum case

LEXICON

MPX-110

24-Bit Dual Channel Processor



The MPX 110 is a true stereo dual-channel processor with 24-bit internal processing and S/PDIF digital output. Utilizing Lexicon's proprietary Lexichip, the MPX-110 offers 240 factory presets, 16 user-definable presets, plus Lexicon classic true stereo reverb programs such as Ambience, Plate, Chamber and Inverse as well as Tremolo, Rotary, Chorus, Flange, Pitch, Detune, Delay and Echo. Dual channel processing allows control of completely independent effects on the left and right channels. Powerful and affordable, the MPX-110 offers the critically acclaimed Lexicon sound to a wide range of users from performing and recording musicians to professional studios and desktop developers

FEATURES

General

- Proprietary Lexichip engine provides independent dual channel effects including true stereo reverb programs such as Ambience, Plate, Chamber and Inverse as well as Tremolo, Rotary, Chorus, Flange, Pitch, Detune and 5.7 seconds of Delay and Echo
- 24-bit A-to-D and D-to-A conversion as well as 24-bit internal processing
- 20Hz – 20kHz ±1dB frequency response
- 19″ rackmountable (1RU high)

Presets

- 240 factory presets and 16 user-definable presets are easily accessible using the Program and Variation Knobs—the Program knob selects among Single, Dual, and User programs while the Variation knob selects one of 16 variations for the selected program
- The 240 carefully-crafted presets feature legendary Lexicon reverb, as well as dual programs that combine two independent effects in four routing configurations
- Dual Programs – Combine two independent effects in four routing configurations:

—Dual-channel processing allows you to combine Delay and Reverb algorithms, or either algorithm with a Flange, Pitch, or Chorus algorithm

—Dual programs are available in four routing configurations:
Dual Stereo (Parallel), Cascade, Mono Split, and Dual Mono

Easy Front Panel Operation

- Dedicated Input, wet/dry Mix and Output level controls
- The Adjust knob controls the most critical parameters for each preset – for some presets, the Adjust knob is patched to multiple parameters to provide simultaneous control of related effects
- The Effects Lvl/Bal knob allows you to control the effect level of Single programs or the balance of effects in Dual programs
- Dual, 2-stage LED headroom indicators

Tap Tempo

- Tap Tempo simplifies the process of matching the delay times and modulation rates of tempo-based presets with your music – Whenever a tempo-based preset is loaded, the Tap button LED flashes, press the Tap button twice in time with the music and the appropriate delay time will be automatically calculated
- Tempo can also be set using audio input (a must for live performances), a dual footswitch, or an external MIDI device that uses Continuous Controller or Program Change messages
- Tempo-controlled delays lock to Tap or MIDI Clock
- Tap tempos can be controlled by audio input, the front panel Tap button, dual footswitch, external MIDI controller or MIDI Program Change

MIDI Control

- Full MIDI control is provided via a powerful editing tool called Learn Mode that allows patching of five front panel controls to a MIDI Continuous Controller value.
- You can use MIDI CC or Program Change messages to manipulate the Adjust, Effects Lvl/Bal, and Mix knobs, as well as the Bypass and Tap buttons as well as use Program Change messages to load programs
- Non-learnable MIDI patches can also be recognized providing access to audio parameters that are not available from the front panel
- Controls can be automated and recorded into a sequencer allowing complete preset automation

Inputs and Outputs

- Left (mono) and Right unbalanced 1/4″ analog inputs accept direct instrument input
- Left and Right unbalanced 1/4″ analog outputs – the left output can be used as a mono out while the Right output can be used to feed a pair of stereo headphones
- The coaxial S/PDIF output has a 44.1kHz sample rate – it is always active so it can be used simultaneously with the analog outs and can be set to wet or dry for use as a high-quality stand-alone A-D converter
- A 1/4″ T.R.S. footswitch connector is provided for remote bypass and tap operation

MPX-200

24-Bit Dual Channel Processor



The MPX-200 combines the best of the MPX 110 with an expanded set of control capabilities to provide deeper editing possibilities. The MPX-200 features the same independent 24-bit dual channel processing, 240 presets, Tap Tempo and MIDI Learn functions as the MPX-110, but also offers a digital compressor algorithm that is available in all 240 programs, including the dual programs. This allows you to have two effects plus a compressor running simultaneously. It also includes 64 User locations for storing variations of presets, coaxial S/PDIF digital I/O, three stage input level metering, and a universal power supply. The combination of Lexicon reverb and effects, ease of use, and flexible routing capabilities, makes the MPX-200 ideal for a broad range of users including project studios, desktop audio/sound designers, performing musicians and professional recording studios.

Legendary Effects

- Plate, Gate, Hall, Chamber, and Room – Legendary Lexicon programs recreate the acoustics of reverberant spaces
- Ambience – world-class programs available for room sounds without reverberation
- Tremolo – offers classic shapes such as square, sawtooth, triangle, sine, and rectified sine. The Adjust knob changes the synchronization of the left and right sweeps to produce mono or stereo effects, and the Tap button matches the tremolo rate of the program with the tempo of the music.
- Rotary – programs simulate a Leslie-style cabinet. Like the physical model, the high (horn) and low (drum) frequencies are separated and "spun" in opposite directions. Horn and drum speeds are independent, designed with the acceleration and deceleration characteristics that simulate the inertia of the original mechanical elements.
- Chorus – inherited from the award-winning PCM Series, pans six independently-randomized delay voices across the stereo field to create a rich, airy effect that multiplies a single source into several
- Flange – programs feature two 2-tap delays, one per channel. Mixing the two tap delays together creates characteristic flange effects such as swishing, tunneling, and fading
- Pitch – programs shift monophonic sources within a range of one octave up to two octaves down
- Detune – programs with one pair of voices per channel, one sharp and one flat. These voices add a delayed or pitch-shifted version of the source to thicken the sound, creating a particularly effective simulation of double-tracking
- Delay and Echo – programs include mono (5.5 sec.), stereo (2.7 sec.), and 6-voice multi-tap effects, each of which can be used to create tape echo or digital delay effects.

Lexicon introduced the first digital audio processor in 1971 and followed with the first commercially successful digital reverberation unit for professional audio studios in 1979. Presently, the "Lexicon sound" is heard on more than 80% of the world's most successful music albums and theatrical soundtracks.

SAME FEATURES AS THE MPX110 *PLUS*—

General

- 240 carefully crafted factory programs plus a bigger User bank with 64 locations to store individual variations of these programs
- Coaxial S/PDIF digital inputs and outputs

Front Panel

- Front panel controls include input level, program load, program edit, compressor, multi-function adjust, bypass, tempo tap.
- The Adjust knob not only allows instant manipulation of the critical parameters of each preset, it is also used for selecting, editing, and cueing programs as well.
- Up to eight adjustable parameters are available in each program
- Adjustable parameters for each program include: Mix, Adjust, EQ and Level/Balance

Display

- Easy to read LED displays indicate current Effects and Routing, Edit Parameters, program Load, MIDI, Store and tempo Tap
- 3-stage headroom indicators are provided for the left and right channels
- A 4-stage compressor gain reduction indicator is also provided

Digital Compressor

- Independent control over Ratio, Threshold, Attack and Release are provided
- The compressor can be used to subtly reduce volume changes and to increase the volume of low level signals — at more extreme settings the compressor acts as a limiter that prevent the volume from exceeding a certain level
- The compressor is available in any program and is situated in front of any other effects in the wet component signal path – it has no effect on the dry signal coming out of the Mix control or the Bypass signal when Bypass Mode is set to dry.

Conveniences

- Assignable bypass mode allows push button or footswitch selection of dry or muted audio output
- Cue Program Mode allows you to jump from one program to another – simply set the adjust knob to the desired program and hit load at the appropriate time– ideal for live performance or mixing in the studio
- Built in power supply, switchable between 120/240 volts AC, 50 - 60Hz.

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MPX-550

24-Bit Dual Channel Processor



The MPX-550 offers the same true stereo, dual channel processing found in the MPX-110 and MPX-200 plus adds a host of professional features including balanced analog I/O, sixteen adjustable parameters per preset and four realtime control knobs not to mention the unmistakable Lexicon sound - all at an affordable price. The unit's deeper editing capabilities are aided visually by a 150 x 32 backlit LCD display with adjustable contrast. The MPX-550 also features an additional bank of 25 presets dedicated to compression effects. For users craving that authentic Lexicon sound along with the need to have detailed control over a sounds color and shape, the MPX-550 delivers.

SAME FEATURES AS THE MPX200 *PLUS*—

General

- 255 presets and 64 user presets including a bank named Compressor which features dedicated presets for compressor—only as and combination compressor/reverb presets with parallel (dual stereo) routing
- 24-bit 44.1kHz and 48kHz operation

Enhanced Digital Compressor

- Dedicated stereo compressor acts on both L and R channels simultaneously and uses the sum of the two channels as its trigger
- Use as a single-channel compressor with side-chain trigger by feeding one channel a level that is at least 6 dB hotter than the other, thus making that channel's signal dominate the compressor's performance
- Gain reduction is indicated by a descending bar between the two input level meters
- The Compressor/Reverb presets are constructed to give the user both a reverb component and a zero-delay "dry" component, with the compressor acting on both

Enhanced Tap Tempo

- The Tap button responds to consecutive presses instead of every other press – this allows end users to more accurately tap in tempos for delay, echo, etc…

Graphics Display

- 150 x 32 backlit LCD display gives you program and system status at a glance, or easy access to the edit pages in each program

Easy Front Panel Operation

- The Program Select knob allows you to scroll through stored programs, and jump between banks for fast and easy selection
- Each program has up to 16 adjustable parameters, organized into "edit pages" consisting of as many as four parameters each – the Edit button cycles through all available pages for the current program
- Four dedicated edit knobs corresponding to displayed parameters make editing easy

Conveniences

- Auto-switching power supply, 90-250v AC, 50/60Hz
- Switchable Locked Mode is available to load only specialized User programs—ideal for live sound and fixed installations

Inputs And Outputs

- Balanced XLR and 1/4″ analog inputs/outputs as well as coaxial S/PDIF digital I/O

MPX-R1

MIDI Remote Controller for the MPX-550 and MPX-1



With the MPX-550 or MPX-1 in the effects loop of your stage rig and an MPX-R1 on the floor, you're ready to take your favorite studio effects on the road. A single cable provides power and two-way communication with the processors.

- Dedicated stomp-box style buttons and LEDs give you complete access to the entire arsenal of MPX-550/MPX-1 effects
- Programmable expression pedal with toe switch
- MIDI program select and control
- Dedicated footswitches for TAP tempo and A/B switching
- All-metal chassis, pedal and switches

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MPX-1

Multi-Effects Processor





The MPX-1 represents the benchmark of high-quality audio effects, interactive control, intuitive operation and affordability. The MPX-1 combines Lexicon's proprietary LexiChip dedicated to uncompromised stereo reverb algorithms, with a second 32-bit DSP that provides up to 4 of additional effects – each with a true stereo audio path. A large library of factory programs, based on 56 algorithms, provide a versatile array of sounds designed for a wide variety of performance, sound design and production applications. The MPX-1 also features in-depth editing of every parameter and complete flexibility of routing and effect order in every program allowing you to take your sound as far as you want to go quickly and creatively.

FEATURES

General

- 200 preset programs designed for a wide variety of audio sources and applications plus 50 user-definable presets.
- A built-in DataBase function makes it easy to find the program that best suits your needs – and because the database is completely user-definable, you can reorganize all 250 programs any way you want.

Two Separate DSP Processors

- Combines Lexicon's proprietary Lexichip dedicated to delivering uncompromised Lexicon stereo reverb or ambience with a separate, fast-math DSP processor that provides up to 4 additional stereo effects.

Instant Access

- Six effect types, based around 56 effects algorithms, are accessible via back-lit buttons on the front panel including 32-bit parametric EQ, Pitch, Chorus, EQ, Modulation, Delay (including looping and ducking) and world-class reverb effects.
- Up to 5 independent stereo effects are available simultaneously and are routable in any configuration, with individual mix and gain for each effect.
- Effects can be placed in any order by dragging and dropping them on a visual map.
- Dedicated Mix and Patch buttons give you instant access to mix and level settings of any or all effects, and the patch system.

Interactive Front Panel

- A numeric display makes program and patch numbers highly visible while an alphanumeric display shows program, parameter and routing values.
- Bypass button lets you toggle master bypass
- An A/B glide button lets you morph between effects or parameter variations.

Tempo Control

- A Tempo LED flashes at the current tempo in any program that utilizes delay times or modulation rates – a Tap button lets you change tempo on-the-fly.
- Tempo parameters will synchronize to incoming MIDI clock and MIDI clock can also be transmit based on Tap tempo.

Editing

- "Soft row" mode offers direct access to the most useful parameters of any program – so you can fine-tune any preset without having to enter a separate Edit mode.
- When you want to completely restructure a program, or create a new one from scratch, the Edit mode lets you dive as deeply as you want into an extensive array of effect and program parameters.

MIDI Functionality

- In addition to separate OUT and THRU connectors, the MPX 1 supports remote power with a 7-pin DIN connector.

Built-in MIDI Arpeggiator

- MIDI arpeggiator processes held notes received from an external sound module to the MIDI in and transmits them as arpeggios through MIDI out. There are two presets, as well as a full range of parameters for building an arpeggio from scratch.

Patch System

- The Patch System provides more than 150 internal and standard MIDI controllers that can be assigned to modulate any effect parameter – up to 5 patches are available per program.
- Internal controllers include: two LFOs, two ADRs, two Envelope Followers, Random Generator, Arpeggiator, A/B Glide (morph), Tempo and Sample and Hold Generator.
- MIDI Learn feature lets the MPX 1 recognize a MIDI control as a patch source.

Audio Connections

- Balanced XLR and 1/4" TRS analog inputs and outputs
- -10dB to +4dB switchable levels
- 24-bit coaxial S/PDIF digital I/O

Footswitch/Footpedal

- A 1/4″ TRS input connector will accept three simultaneous footswitches
- Another 1/4″ TRS connector is provided for a footpedal

PCM-81

Digital Effects Processor



The PCM 81 is Lexicon's top of the line, single rack space Digital Effects Processor. It has everything that top recording studios require in a multi-effects processor and more. Lexicon's unique Dual-DSP Platform enables the PCM 81 to create the most flexible effects combined with superb reverberation based on the legendary Lexicon 224, PCM60 and PCM 70 processors. The PCM 81 exhibits extraordinary sonic clarity with 24-bit internal processing, 24-bit A-to-D and D-to-A converters, balanced XLR and 1/4˝ inputs and outputs as well as 24-bit AES/EBU and S/PDIF digital I/O. An enormous selection of sounds are 300 factory presets are on-hand based on 17 algorithms that combine an uncompromising stereo reverb or ambience effect with a multi-voice effect. Two modes of editing gives you a choice of quickly grabbing a programs most pertinent parameters or diving way down deep where you can build your own presets from the ground up. Dynamic Patching makes it possible to route any of 150 sources to any parameter delivering a truly unique set of capabilities, from realtime and tempo-based modulating of sounds, to producing unusual and ethereal spaces, to altering the attack and decay characteristics of the sounds.

FEATURES

Presets

- A library of 300 carefully crafted factory presets (plus 50 user-definable presets) range from lush and beautiful to completely over the top and cover a wide variety of applications from performance, recording and remixing, to effects designed specifically for sound design, foley and video post
- Each preset gives you instant access to Pitch, Reverb, Ambience, sophisticated Modulators, over 21 seconds of Delay time as well as Dynamic Spatialization effects for 2-channel or surround applications
- There are a total of 17 algorithms separated into three general classes: 4-Voice, 6-Voice and Pitch – each includes an uncompromised stereo reverb effect, along with several voices of additional stereo effects.
- 4-Voice algorithms combine a specific type of reverberation with a 4-voice, general purpose, stereo effect "toolbox", as well as additional post-processing for the reverb.
- 6-Voice algorithms combine a specific type of reverb with a specialized 6-voice stereo effect.
- Pitch algorithms offer a unique type of pitch shifting, combined with an uncompromised Lexicon reverb.

Pitch Correct

- Pitch Correct provides a simple yet powerful set of tools for correcting off-pitch melody tracks within a range of up to three octaves (up or down). The Pitch Detect display shows the pitch of the input source in real time in chromatic note and +/- cents. The Correction parameter can be patched to a switch or the ADJUST knob, or even to a MIDI keyboard.

3-D Dynamic Spatialization

- These effects utilize two independent spatial processors that allow you to place effects virtually anywhere between, or even beyond, your loudspeakers
- By dynamically controlling spatial effects, you can create different spaces that change along with the music – For example, when playing sustained chords through Steered Rear, the PCM 81 automatically steers the reverb around you (into full rear in surround sound) when the input audio decays past the threshold.

The Interface

- The large, 2-line fluorescent display is easy to see from any angle whether the surroundings are bright or dark
- The ubiquitous Lexicon Adjust knob is on-hand for controlling a programs most critical parameters
- A special Info mode allows you to press and hold any button to find out what its function is, or to get status information such as the name of the running effect, current tempo rate, etc.

Pro and Go Edit Modes

- In Go mode, up to 10 of the most useful parameters within an effect are grouped for instant access via the front panel Edit button – Each preset has a selected set of Go mode parameters which let you make value changes to the effect without losing the original identity of the sound.
- Pro mode gives you access to the full parameter editing matrix (as many as 100) for the algorithm of any loaded effect. In this mode, you can access a complete set of Modulation and Patching parameters, create your own ADJUST knob patch and assign your own Go mode parameters.

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PCM-81

Dynamic Patching Matrix

- Dynamic Patching allows you to map data from any of 143 possible control sources to any effect parameter (destination)
- Up to 10 patches can be created per effect, each with as many as 8 pivot points allowing very complex and mind-altering modulation paths
- You can patch multiple parameters to a single controller, or patch multiple sources to a single destination
- There are three types of Sources available:
  - Internal controllers include Tempo (both internal Tap and external MIDI clock), LFO (Sine, Cosine, Square, Triangle, Pulse, Sawtooth), Time Switches, Latch, AR Generator, and Left & Right Envelope Followers
  - MIDI modulators such as Pitch Bend, Velocity and After Touch
  - Or any of 126 MIDI Controllers as well as external sources such as footswitches and footpedals

Tempo Control

- The Tap tempo control offers the ability to create independent rhythmic values per parameter within the same program based on note values or absolute time.
- Tempo can also be 'dialed-in' so you can set the tempo in exact beats-per-minute.
- You can generate MIDI clock from your tap, as well as receive MIDI tempo from an external sequencer or drum machine.
- Tempo control LFO speeds and Time Switch controls. Thus all of your modulations can be synchronized with your music.

Conveniences

- The PCMCIA card slot accepts RAM cards that can be used to store as many as 2350 of your own programs on a 1 Meg card.
- The Compare button allows you to compare the active version of the current effect with the most recently stored version.
- The Bypass switch can be used to mute the audio output or bypass an effect depending on the setting of each program's bypass parameter.
- Standard 3-pin IEC power connector with 100-240V, 50-60Hz automatic switching power supply to correct voltage range.

Inputs & Outputs

- Balanced XLR–1/4″ combo connector inputs and separate balanced XLR and 1/4″ outputs
- Full AES/EBU (XLR) and coaxial S/PDIF digital I/O – the AES/EBU and S/PDIF outputs can be used simultaneously
- Analog and digital inputs can be mixed together
- MIDI In/Out/Thru connectors – all parameters, programs and registers can be accessed by MIDI.
- The 1/4″ footswitch input can accept two momentary switches
- The 1/4″ footpedal input can be used for realtime control of selected parameters.

Effects

- Fans of classic Lexicon sounds will appreciate updated versions of effects from the legendary PCM 70, like Tiled Room and Concert Hall.
- A wide variety of phone and microphone presets for Sound Design, Surround delays for film, Orchestral Reverbs, Complex EQ, Special FX and Cyber FX
- Presets set up for delays that automatically lock to incoming MIDI clock
- Presets optimized for Keyboards, Guitar, Drums, Vocals and Orchestral instruments.
- Pitch Shifting algorithms include doubling, quadruple-tracking, chorus, as well as pitch correction
- Pitch Correct provides a simple yet powerful set of tools for correcting out of tune melody tracks within a range of up to three octaves (up or down). The Pitch Detect display shows the pitch of the input source in real time in chromatic note and ± cents. The Correction parameter can be patched to a switch or the ADJUST knob, or even to a MIDI keyboard.

OPTIONAL PCMCIA FX CARDS

Dual FX Algorithm Card



- 25 additional algorithms and 250 additional presets.
- Built-in Submixer for completely separate control of inputs, outputs, and routing of reverbs and effects.Allows seamless, continuous and dynamic transition between Serial and Parallel effects routing (controllable by the Adjust Knob, footpedal or MIDI)
- Processed Reverb- extensive tailoring of the effects with presets designed for several applications: Drums, Vocals, Guitars, and other music related needs.
- Analog Sound Modeling Presets: two-track tape echo & rolloff, tape flanging, room mic placement and other “Classic Effects”.

Post FX Preset Card

- Offers 100 presets by Scott Martin Gershin (Hollywood’s top Sound Effect Designer) for post-production and sound design for film and video applications.
- Featuring spectacular presets designed to create special effects from simple impulse inputs.
- Wide variety of phone and microphone futzes and presets for sound design, surround delays for film, orchestral reverbs, complex EQ, special FX and cyber FX.

Music FX Preset Card

- 100 powerful Presets for a wide variety of musical applications designed by David Rosenthal (keyboardist for Billy Joel’s River of Dreams Tour).
- Includes delays that automatically lock to incoming MIDI clock, presets optimized for keyboards, guitar, drums vocals and orchestral instruments as well as performance and stage applications utilizing MIDI and Dynamic controllers for expressiveness.

PCM-91

Digital Reverberator



The PCM 91 Digital Reverberator offers Lexicon's highest quality reverbs in a compact, affordable package with a powerful interface that allows easy access to the plethora of powerful programming capabilities. It combines the same user-interface, editing, dynamic patching and I/O capabilities of the PCM 81 with a unique dual-processor architecture that features two of Lexicon's proprietary reverb DSP ICs, the Lexichip, designed to deliver exceptional reverb quality along with unprecedented flexibility and control. The PCM 91 offers 450 reverberation programs, each of which has been designed specifically with real-world music production, live performance, sound design and video post-production applications in mind. All of the features of the critically acclaimed PCM 90 are included, plus onboard Dual Reverb algorithms and presets with dynamic spatialization effects for 2-channel or surround sound applications. Unique to the PCM 91 is the implementation of four custom controllers that act as four additional Adjust knobs capable of accessing a programs most critical parameters. The onboard Keyword search allows you to find appropriate programs by application.

FEATURES

Reverb Realism

- Built-in library of 450 factory presets and 100 User preset locations containing Lexicon's highest quality reverb effects that simulate realistic halls, rooms and plates, and let you create completely natural, or other-worldly spaces
- Create incredibly realistic small spaces for music or film work - from the ambience of a phone booth to a very small drum room

15 Algorithms

- 5 stereo algorithms to create different types of reverb effects and 10 Dual Reverb algorithms which offer superb dual reverb and cascade configured stereo effects.
- Single algorithm include an uncompromised stereo reverb effect with selected "tools" for ambience, post-processing, compression/expansion, as well as modulation and patching parameters which are common to each algorithm
- Dual algorithms contain two independent reverb blocks, as well as the full set of modulation and patch features in the single effects

Four Custom Controllers

- These controllers are placed on the Soft Row and are a combination of one or more parameters patched together, each with their own individual scaling values. It's like having four additional Adjust Knob controls on the Soft Row.

Intelligent Interface

- The KeyWord Search function allows you to find a group of programs designed for a given application – for example if you choose the Live PA KeyWord, the PCM 91 will automatically locate all the presets that have been optimized for that application. There are 50 keywords in all, including four user-definable groups of effects.
- The History Of Effects Loaded feature allows you to review the last ten effects loaded – useful for when you want to return to an effect you were using earlier, but can't remember its name or location.
- If you want to know more about the function of a particular button (without actually executing any action) press and hold the button down and message with an explanation will appear on the display.

Same As The PCM81

- The Adjust knob allows realtime control of a presets most important parameters without ever going into the Edit mode.
- Pro and Go editing modes offer you the choice between easy access to a programs most important parameters or full access to all of a programs parameters and build an effect from its basic algorithm up
- Dynamic Patching provides an expressive means of modulating sounds as well as the ability to alter the attack and decay characteristics of the sound.
- Tap Tempo with independent rhythmic variations or dial in tempos in BPM.
- Dynamic Spatialization with two independent spatial processors that allow you to place effects virtually anywhere between your loudspeakers – or even beyond them.
- PCMCIA Card Slot for user-edited preset and system set up storage.
- Balanced XLR-1/4″ combo connector inputs and balanced XLR and 1/4″ outs.
- Full AES/EBU (XLR) and coaxial S/PDIF digital I/O
- 1/4″ footswitch and foot controller inputs

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MPX G2

Guitar Effects Processor



The MPX G2 is a hybrid Guitar Effects Processor that combines the highest quality 32-bit digital effects with analog distortion and overdrive as well as dedicated analog tone controls and an analog Speaker Simulator. The MPX G2 will work with any amp, allowing the guitarist to place authentic effects anywhere in the signal chain, without altering the amp's basic tone. For direct recording or PA applications, the MPX G2 can be used without an amplifier as a stand-alone programmable analog preamp with effects. Two separate signal paths allows effects such as compression, wah and analog overdrive to be placed in front of the amp, while other effects like delay, chorus and reverb can be placed in the amp's effects loop. The MPX G2's effects include several authentic recreations of vintage stomp boxes such as Tube Screamer, Mu-tron III, Cry Baby, Dyna Comp and Space Echo, to name a few, and studio effects like JamMan, Intelligent Pitch Shifting, Tap Delay, Chorus, Flange, Rotary Speaker, Parametric EQ and Lexicon Reverb and Ambience. Lexicon's optional MPX R1 MIDI Remote Control pedal board provides enhanced hands-off control of all the MPX G2 features, and creates a powerful, versatile system with two programmable relays to switch up to four amplifier channels and control up to seven effects at once.

FEATURES

- Two separate audio paths let you place effects in front of your amp or in the amp's effects loop.
- Use without an amp as a stand-alone pre-amp with effects.
- All of the hardware of a custom guitar rig is built-in: a loop switcher with analog relay bypass, an effects router, analog and digital noise gates, chromatic tuner and an analog speaker simulator.
- Dynamic Gain, Lexicon's analog distortion technology, provides screaming overdrive and warm distortion tones which can be used as an analog stomp box in front of your amp, or as a standalone preamp for direct recording or live performance.
- Effects can be synchronized with the music by assigning tempo control to modulation rates, delay times or any effect parameter. Tempos can be tapped with the Tap button (or an assigned controller) or dialed in. MIDI clock can be generated from your tempo, or you can receive MIDI clock from an external sequencer or drum machine.

Inputs and Outputs

- 1/4″ guitar inputs are available on both the front and rear panels.
- 1/4″ and XLR left and right outputs
- 1/4″ effects send and 1/4″ left and right effects returns.
- 1/4″ stereo headphone output

Effects

- There are 76 effects organized into seven types. Any active effect can be accessed or bypassed just by pressing the front panel button.
- Effects can be placed in any order, simply by "dragging and dropping" them on a simple visual map.
- Gain and amp effects include Tone controls, Overdrive, Screamer and Distortion effects, along with a fully featured, programmable analog guitar preamp.
- Effect 1 and 2 contain Pitch Shift, Detune and Tremolo effects, Panners, Phasers, Compressors, and an assortment of wah and volume controls including recreations of classics like the Uni-Vibe, Dyna-Comp, Phase 90, Vox and Cry Baby wah.
- Chorus effects include classic chorus and flanger effects along with rotary speakers, and spatial panning effects.
- Delay effects include delay, echo, and ducking effects as well as a JamMan style looper with 20-second full-bandwidth delays.
- Reverb effects include a full complement of Lexicon's classic true stereo reverberation and ambience effects. Dedicated processing resources are allocated to the reverb effects to ensure that you are always able to load an uncompromised reverb into any program, regardless of what other effects are loaded.

Presets

- The large 300 preset library is organized with a database that allows you to quickly search for programs based on their styles like Rock or Blues, by the type of effects such as OverDrive, or to find only programs suitable for stand alone operation.

Editing

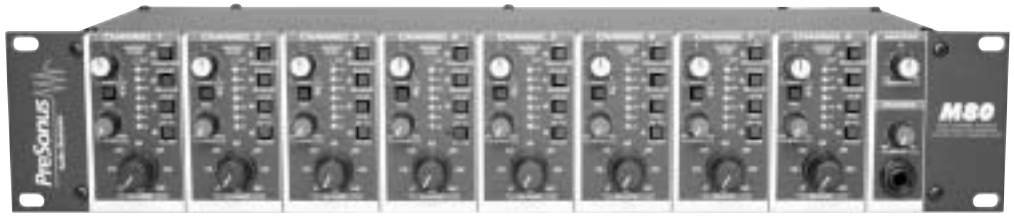
- A front panel Soft Row button takes you directly to the most useful parameters of each program.
- Front panel System and Edit buttons give you access to all of the controls you need to completely reconfigure your system or to create completely new programs.
- The A/B glide function allows you to glide from one program to another with a single button push.

Patching System

- The MPX G2 patch system provides an assortment of controllers that can be assigned to any effect parameter (up to five per program). These controllers include LFOs, A/B Glide and Tempo.
- MIDI In, Out and Thru connectors allow you to control of A/B and Tap tempo, MIDI clock, as well as all effect parameters.
- Connecting an MPX R1 gives you stomp box control of all of the MPX G2 effects.

M80

Eight Channel Mic Preamp with Mix Bus



The M80 provides eight channels of discrete Class A microphone/ instrument preamplification with low noise, balanced input transformers along with several unique features. An ultra low impedance mix bus is available for assigning and panning any of the 8 channels to a pair of stereo outputs. A high gain headphone output for monitoring the main bus. Another unique feature is the vintage warmth control provided on each channel that emulates the effect of analog tape saturation and vacuum tubes. Each channel is equipped with balanced XLR / 1/4˝ TRS mic/line inputs via Neutrik combo connectors and XLR balanced outputs. Separate 1/4˝ TRS send and return jacks are also on hand for inserting outboard dynamics processors.

Each Channel Features



- A discrete Class A input buffer followed by a twin servo gain stage provides ultra low noise and wide gain control allowing you to boost signal without increasing unwanted background noise
- Gain control provides 60dB of gain — the amplifier has inherent gain of 12dB thus delivering a total possible gain of 72dB
- +28 dBu of headroom provides a very wide dynamic range and excellent transient response
- Switchable 48v phantom power, phase (polarity) reverse, 20dB pad and a 6dB/oct low-cut filter at 80Hz
- Phantom power is supplied at a constant rate whether on one or all eight channels, ensuring condenser mics will be free of distortion associated with insufficient power
- Full Scale 7-segment LED metering (-36 dBu to +18dBu) plus a clip indicator

Inputs and Outputs

- XLR / 1/4˝ TRS Neutrik combo connectors accept inputs signals from mics or instruments
- XLR-balanced direct outputs will feed directly into the inputs of digital workstations as well as tape and disk-based multitrack recorders

Vintage Warmth Control

- The IDSS knob on each channel lets you add "warmth", much like a vacuum tube or analog tape saturation, to the audio source by boosting the even harmonic series of the input signal

Stereo Mix Bus (8X2 Mixer)

- A stereo mix bus allows you to combine the input signals of each channel for monitoring or live remote/straight to 2 track recording
- Each channel features a L/R Mix Bus assign switch and a pan knob for placing each signal within the stereo spectrum
- A high output stereo headphone output with level control on the front panel enables you to monitor the Mix bus in loud or ambient environments such as concert halls, clubs or in control rooms monitoring at high volume levels
- XLR mix outputs connectors accommodate both balanced and unbalanced signals
- The output level control has a range of -72dBu to +10dBu

Metering

- Servo balanced send/return jacks for inserting your favorite outboard gear

Additional Features

- An Auxiliary Input lets you daisy chain several M-80s together allowing them to share the last stereo output in the chain
- Housed in a 2RU, steel chassis with a high quality blue anodized front panel
- An external power supply ensure a noise free preamp environment.

System Performance

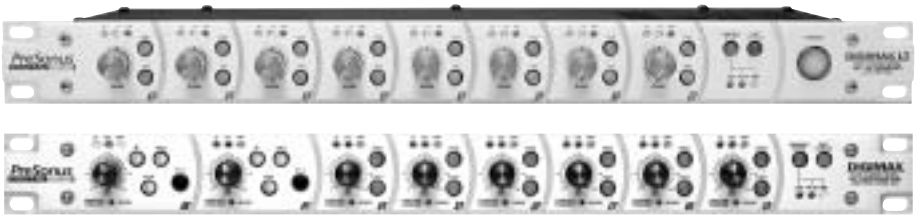
Dynamic Range	>120dB
Headroom	+28dBu
Noise Floor @ Bus	-90.2dBu
Noise Floor @ Main Output	-96.4dBu
Noise Floor @ Channel Output (+24dB Gain)	-88.2dBu
THD + Noise (no idss)	0.0024%
THD + Noise (max idss)	0.035%
THD + Noise	<0.02%
Frequency Response	10Hz to 60kHz ±0.5dB
Crosstalk	>82dB @ 10kHz

Input Specs	**Impedance**
XLR MIC / 1/4˝ TRS Line	10k Ω
Insert Specs	**Impedance**
1/4˝ TRS Send	51 Ω
14˝ TRS Return	10k Ω
Output Specs	**Impedance**
XLR	51 Ω

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DigiMAX & DigiMAX LT

Eight Channel Mic Preamps with 24-Bit Digital Output



The DigiMAX LT and the DigiMAX each combine eight channels of the same pristine, discrete Class A dual servo mic preamps as the M-80, MP-20 and VXP along with eight channels of 24-bit ADAT LightPipe digital outputs, to deliver the ideal front-end solutions for DAW's or for adding mic pre's to digital mixers and sound cards. The DigiMAX LT features balanced XLR, 1/4˝ mic/line inputs with inserts on every channel while the DigiMAX provides eight XLR balanced mic inputs with direct 1/4˝ TRS analog outputs as well as Hi-Z instrument inputs on channels one and two that allow direct connection of an electric guitar or bass. The DigiMAX also features an EQ Enhance and a transparent dual domain limiter on each channel as well as a choice of four stereo pairs of coaxial S/PDIF or AES/EBU outputs via a 9-pin d-sub connector. Both units are housed in rugged 1RU chassis with a sturdy front panel milled from 1/4˝ solid aluminum and blue anodized aluminum knobs.

THEY BOTH FEATURE

8 Dual Servo Mic Preamps

- Each channel contains a Class A discrete input buffer followed by a dual servo gain stage to provide ultra low noise, wide gain control and the ability to boost signal without increasing unwanted noise
- Switchable phantom power and a 20dB pad are provided for each channel

Inputs

- The DigiMAX LT features eight balanced mic/ line inputs using Neutrik XLR / 1/4˝ TRS combo connectors
- The DigiMAX features eight XLR-balanced mic inputs and Hi-Z 1/4˝ instrument inputs on channels 1 and 2

8-Channel 24-bit ADAT Lightpipe Outputs

- A front panel switch lets you choose sample rates of 32, 44.1 and 48 kHz
- An external clock switch, also on the front panel, works in conjunction with the Word Clock I/O, ensuring proper synchronization, and allowing either unit to slave to incoming timecode or be the master clock source within a digital system

Power Supplies

- Linear internal power supply (DigiMAX LT); External, 1/3 rack space power supply (DigiMAX)

DigiMAX LT Only

- 1/4˝ TRS insert points, provided on each channel, allow you to access external dynamics processors and EQs. Or, you can use the insert as an analog output by using a 1/4˝ TS connector

DigiMAX Only

- A dual domain limiter, on each channel, detects RMS and peak levels to achieve instantaneous transparent limiting — the RMS stage acts like a high ratio compressor while the peak stage is used to prevent the input of the A/D converter from clipping
- A dual concentric rotary control on each channel adjusts the channel's gain (inner) as well as the threshold parameter of the dual domain limiter (outer)
- A phase (polarity) reverse switch is also provided on channels 1 and 2
- An EQ enhancement switch on each channel cuts 3dB of signal between 250Hz and 5kHz has a smoothing effect on mid-range heavy signals providing a flatter response characteristic
- 1/4˝ TRS analog outputs on each channel duplicate the signals sent to the digital outs allowing you to monitor channels with an analog mixer and avoid the latency issues often encountered with digital workstations
- A 9 pin connector is internally configurable to provide 4 stereo pairs of either AES/EBU or S/PDIF digital outputs

Preamp Controls	
Input Gain Control	+12dB to +38dB
Switches	Phantom power, 20dB pad
24-Bit Digital Outputs	
ADAT Lightpipe	8 channel optical
DIGIMAX LT ONLY	
Inputs	
Mic / Line	8 Balanced XLR / 1/4˝ TRS
Insert Points	(8) 1/4˝ TRS
Metering	
2 Stage Signal Present LED	-20 & -10 dBu
Clip LED	+22dBu
DIGIMAX ONLY	
Inputs	
Mic	8 Balanced XLR
1/4˝ Instrument	Channels 1 & 2
Polarity Reverse	Ch 1 & 2 only
EQ Enhance Switch	-3dB (250 to 5k Hz)
Dual Domain Limiter	
Threshold Range	0dB to off
Metering	
Signal Present LED	-20dBu
Clip LED	+22dBu
Limiter LED	Limiter Active
Analog Outputs	(8) 1/4˝ TRS
S/PDIF (RCA) Outputs	4 stereo pairs*
AES/EBU (XLR) Outputs	4 stereo pairs**
**requires DM006 optional breakout cable* ***requires DM007 optional breakout cable*	

System Performance

Headroom	+22dB
Analog Dynamic Range	>120dB
Noise Floor	-94dBu
THD + Noise (unweighted)	<0.009%
Frequency Response	20Hz-50kHz
Power Supply Rejection	>98dB

VXP

Dual Servo Voice Processor/Channel Strip Mic Preamplifier



The VXP is a class A channel strip/voice processor featuring a single channel of the M80/MP20's discrete class A microphone preamplifier with IDSS control, plus a preset compressor, expander, tunable de-esser, four-band semi parametric equalizer and separate brick wall peak limiter.

- Discrete class A input buffer followed by a dual servo gain stage
- +24dBu headroom provides a very wide dynamic range and excellent transient response
- Switchable 48V phantom power, phase (polarity) reverse 20dB pad
- XLR balanced inputs and outputs as well as 1/4″ TRS bal / unbal outputs
- 1/4″ TRS bal / unbal send and return jacks allow you to insert an external processor (post preamp /pre processor)
- Comprehensive LED metering is provided for each processing stage

Smart Compressor

- 16 pre-set compression curves offer varying degrees of compression ranging from light to heavy, designed especially for processing voice
- Easy-to-use preset parameters derived from pro audio engineers provide a wide variety of "studio tested and proven" settings uniquely suited for virtually any singing style or vocal application

Expander

- The downward expander uses a dual concentric control with an adjustable threshold and ratio to provide seamless noise reduction ridding the signal of unwanted background noise

De-Esser

- A variable de-esser control with a range of 800Hz to 8kHz removes unwanted sibilance while allowing the signal to remain completely natural
- The threshold control makes setting the de-esser extremely precise and fine-tuning a simple process

4 Band Semi-Parametric EQ

- Fixed low and high shelving bands as well as sweepable low and high mid controls to provide total tonal control of the audio
- Each of the four bands feature a boost/cut range of ±12dB
- The low frequency control is fixed at 100 HZ while the high frequency is tuned to 12kHz with a shelving contour that accentuates the naturalness vocal signals.
- The low mid-range band is sweepable from 90Hz to 700Hz and the high mid-range is sweepable from 450 to 5.8k Hz
- A switchable Q function for each of the mid band frequencies is selectable from .5 (wide) or by a factor of 2 (narrow)
- An 80Hz filter is also provided for eliminating low frequency noise

Master Section

- The peak limit circuit is a sonically transparent brick wall limiter with an adjustable threshold to control the point at which the output signal is not allowed to cross — ideal for recording to digital devices where signal overload is unacceptable
- An output level control, variable from –70 dB to +10 dB, allows gain make-up due to compression or decreasing the output signal after boosting frequencies in the EQ

Optional Digital Output Card

- Provides AES/EBU and SPDIF outputs as well as an auxiliary 1/4″ TRS analog line input that allows two VXP's to share both sides of the A to D converter from one digital output card
- Selectable 96, 48 and 44.1 kHz sample rates
- Superior Crystal Semiconductor A to Ds
- Psycho-acoustic dithering is provided to improve BIT resolution characteristics

Preamp Controls	
Input Gain Control	+12dB to +38dB
Switches	Phantom power, phase reverse, 20dB pad
Smart Compressor	
Input Attenuation/Gain	-12dB to +18dB
Output Attenuation/Gain	-20dB to +20dB
Presets	16 Light to Heavy
Gain Reduction Meter	8-Segment LED
Process	In/Out (Bypass)
Expander	
Threshold Range	-70dBu to +20dBu
Ratio Range	1:1 to ∞:1
Gain Reduction Meter	4-Segment LED
De-Esser	
Threshold Range	-40dBu to +20dBu
Frequency Range	800Hz to 8kHz
Gain Reduction Meter	4-Segment LED
4-Band Semi-Parametric Equalizer	
High Pass Filter	6dB/oct @ 80Hz
Low Shelving (±12dB)	100Hz
Low Mid-Range (±12dB)	90Hz to 700Hz
Switchable Q	0.5/2.0
High Mid-Range	450Hz to 5.8kHz
Switchable Q	0.5 /2.0
High Shelving (±12dB)	12kHz
Process	In/Out Switch
Master (Output) Section	
Peak-Limit Threshold Range	0dBu to +24dBu
Output Gain	-70dB to +10dB
Full Scale Output Metering	8-Segment LED (-28 to +18 dBU)

System Performance

Dynamic Range	>115dB
Noise Floor	-97.2dBu @12dB gain
THD + Noise	<0.003% (0% IDSS)
Freq. Response	10Hz to 50kHz
Input Specs	**Impedance**
XLR mic, 1/4″ TRS line	1.3k Ω
Insert Specs	**Impedance**
1/4″ TRS Send	1 Ω
1/4″ TRS Return	10k Ω
Output Specs	**Impedance**
XLR	51 Ω

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EQUIPMENT LEASING AVAILABLE

MP20

Stereo Microphone Preamp with Mix Bus



The MP20 is essentially a single rack space, two channel version of the M-80 with the same crystal clear microphone / instrument preamplification capabilities, stereo mix bus and IDSS "vintage warmth" control.

- Identical technical specifications as the M80
- 1/4″ high Z instrument input on the front panel
- 12 Segment level meters for each channel
- Housed in a single rack space steel chassis with a thick, high quality, blue anodized front panel
- XLR balanced input/output connectors as well as separate send and return jacks for inserting dynamics processors

Input Specs	Impedance
XLR MIC / 1/4″ TRS Line	1.3k Ω
1/4″ Instrument	1Meg Ω
Insert Specs	**Impedance**
1/4″ TRS Send	51 Ω
14″ TRS Return	10k Ω
Output Specs	**Impedance**
XLR	51 Ω

BlueTube – Stereo Tube Mic/Instrument Preamp



The BlueTube is an affordable dual channel tube mic preamp that utilizes a true hybrid design that combines a 12AX7 tube stage with the same ultra low noise dual servo gain stage as it's big brothers. The Blue Tube brings world class quality to a price that fits any budget.

- Front panel mic/instrument inputs use the Neutrik Combo connector, which lets you use either 1/4" TRS or XLR connectors within the same female input
- Input gain is variable from 0 to 40dB
- A Drive potentiometer controls the amount of signal routed to the 12AX7 vacuum tube from 0 to 30dB — Higher levels of tube saturation give the signal greater warmth and a richer sound
- Switchable phase (polarity) reverse and a 20dB pad are provided for each channel
- A single +48 volt phantom power switch is used for both channels
- 8-segment Level meter for each channel
- Servo balanced XLR and unbalanced 1/4″ outputs operate at +4dBu and -10dBv respectively
- Housed in a half rack space steel chassis with a high quality, anodized brushed aluminum front panel
- Fits perfectly in the PreSonus BMRA rack adapter with the PreSonus BlueMax

DigiTube – Single Channel Tube Mic/Instrument Preamp



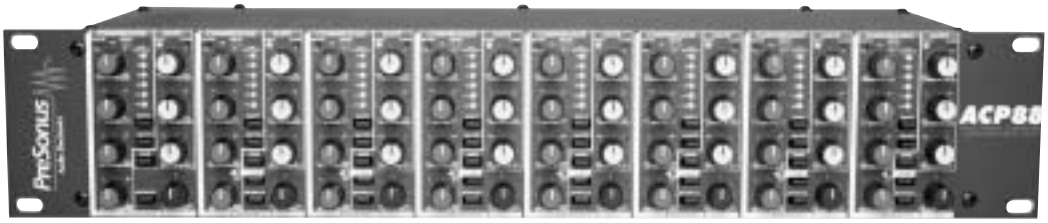
The DigiTUBE combines a single preamp channel of the BlueTube with a three band sweepable EQ and 24bit digital output. The mic preamp features 70dB of gain with 22dB of Headroom. The three band EQ which is fully sweepable with overlapping bands for maximum tone shaping. A 24Bit Digital output via S/PDIF makes the Digitube the perfect front end for soundcards and digital recorders. The DigiTUBE offers XLR analog output and TRS Insert point for patching in a compressor or other outboard effect unit.

- Tube preamp with 70dB of Gain and 22dB of Headroom
- Fully sweepable 3 Band EQ with bypass switch
- Insert point after Mic Pre for patching a comp/limiter or other outboard device
- 24Bit Digital Output via S/PDIF
- Word clock input and output via BNC Connectors
- Rackmountable via optional BMRA Rack Adaptor

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ACP88

Multi Channel Dynamics Processor



The ACP88 is an eight channel dynamics processor that provides crystal clear compression, brick wall limiting and noise gating for a variety of applications including multitrack recording, live sound reinforcement, broadcast and permanent sound installations. Full featured musical compression on each channel is equipped with variable threshold and ration controls, a choice of variable or "Auto" attack and release modes, switchable soft/hard knee compression curves as well as a multi-channel link function. The noise gates' feature set includes threshold, variable attack & release times and gate range controls. Both the Compressor and Gate sections of each channel, offers it's own Sidechain inserts that allow external triggering and spectral control of each processor.

FEATURES

◆ At the heart of each channel is the much respected THAT 4301S VCA which offers unsurpassed dynamic range with extremely low distortion characteristics and virtually no noise

Eight Compressors/Limiters

◆ Each channel offers broad control over threshold, ratio, attack and release for each processor

◆ You can select between auto or manual attack and release curves and hard or soft knee compression types

Eight Dynamic Noise Gates

◆ A separate dynamic noise gate on each channel is provided with control over attack, threshold and release as well as a gate range switch

◆ The gate range switch allows you to select either 15dB or 60dB of level reduction. When the range switch is set at 15 dB, there will only be a slight change in the signal as it crosses the threshold which is useful in creating a more natural sounding blend or mix when gating many instruments at once

Channel Linking

◆ A link switch on each channel, allows you to link channels together in a subgroup to be controlled by a master processor (the channel furthest to the left in the subgroup)

Separate Bypass and Gain for Every Channel

◆ Each Channel has a Bypass for auditioning a signal ' before and after' processing with the compressor, limiter, or the Gate

◆ A Gain control is provided to make up any loss in signal level resulting from the amount of compression being applied

Compressor Sidechain

◆ The 1/4″ TRS sidechain insert on each channel provides a send and return circuit that interrupts the signal that the compressor is using to determine the amount of gain reduction to apply

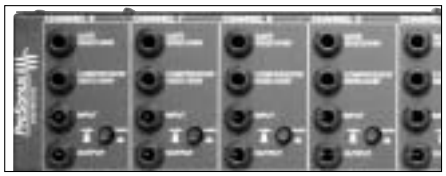
➢ An EQ inserted into the sidechain using 1/4″ TRS connector, accommodates frequency selective processing that allows you to, among other things, use the compressor as a de-esser or for controlling the bass frequencies while leaving the higher frequencies unprocessed.

➢ An audio control signal of a narrator or vocalist can be input to the sidechain. In this application, the audio that's passing through the compressor will automatically 'duck' when the narrator speaks or vocalist sings.

◆ When multiple channels are Linked together, the Send of the Sidechain jack of the Master channel contains a mix of all the channels in the link — this allows you to control all of the linked channels with one sidechain insert.

Gate Sidechain/Keying

◆ A separate 1/4″ TRS Gate sidechain/Keying jack on each channel allows you to use an external sound to trigger the gate. An EQ or filter inserted at the gate key enables frequency-selective gating such as the lower frequency of a kick drum



Inputs and Outputs

◆ Each channel features 1/4″ TRS inputs and outputs that will accept either balanced or unbalanced signals

◆ Each channel also features a switch on the rear panel for selecting the internal operating level between +4dBu to –10dBV

Additional Features

◆ Housed in a two RU, all-steel chassis

◆ Switchable power transformer for domestic and international use

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ACP22

Dual Channel Compressor, Limiter and Gate



The ACP22 is an award winning dual channel version of the esteemed ACP88 and like its big brother is the most comprehensive stereo compressor/gate in its price range today. The ACP22 offers a full array of inputs and outputs including balanced XLR, unbalanced 1/4˝ as well as independent send and return Sidechains for the compressor and gate of each channel. Unique to the ACP22 over its big brother, is a switchable Lo Pass filter which blocks high frequency content such as cymbals from triggering the gate.

FEATURES

- Balanced XLR and 1/4˝ unbalanced inputs and outputs
- The Lo Pass Sidechain filter is designed to make the noise gate less sensitive to high frequency instruments — extremely useful for gating drums as the filter will stop cymbals from opening up the tom or bass drum gates
- The filter is set for -6dB at 2.5kHz rolling off the high frequencies at -12dB/oct
- Stereo Channel Linking allows both processors to follow the setting of channel one which becomes the master
- 8-Segment LED meters can be switched to display either input or output levels

ACP88 Input Specs	**Impedance**
1/4˝ TRS	51k Ω balanced/unbalanced
ACP88 1/4˝ Compressor Sidechain	**Impedance**
Send	10k Ω unbalanced
Return	51 Ω unbalanced
ACP88 1/4˝ Gate Sidechain	**Impedance**
Send	10k Ω unbalanced
Return	51 Ω unbalanced
ACP88 Output Specs	**Impedance**
1/4˝ TRS	51 Ω balanced/unbalanced
ACP22 Input Specs	**Impedance**
XLR or 1/4˝ TS	10k Ω balanced/unbalanced
ACP22 1/4˝ Compressor Sidechain	**Impedance**
Send	10k Ω unbalanced
Return	51 Ω unbalanced
ACP22 1/4˝ Gate Sidechain	**Impedance**
Send	10k Ω unbalanced
Return	51 Ω unbalanced
ACP22 Output Specs	**Impedance**
1/4˝ TRS	51 Ω balanced/unbalanced

System Performance

	ACP88	ACP22
Dynamic Range	>115dB	>115dB
Signal to Noise Ration	>95dB	>95dB
Noise Floor	-94dBu	—
THD + Noise (unweighted)	<0.009%	<0.02%
Headroom	—	+24dBu, unbalanced; +22dBu balanced
Frequency Response	10Hz to 50kHz	10Hz to 50kHz
Crosstalk	<82dB @10kHz	<82dB @10kHz

	ACP88	ACP22
	Gate Controls	
Threshold Range	-70dBu to +20dBu	Off, -65dBu to +20dBu
Attack Time	0.01ms to 100ms	01ms to 100mS
Release Time	0.02ms to 2 sec	0.02sec to 2sec
Attenuation Range	-15dB or –60dB	0dB to -60dB
Sidechain Filter	—	Switchable Lo Pass -6dB @ 2.5kHz (12dB/oct)
	Compressor/Limiter Controls	
Threshold Range	-40dBu to +20dBu	-40dBu to +20dBu
Ratio	1:1 to 20:1	1:1 to 20:1
Attack Time	0.02ms to 200 ms	0.1 ms to 200ms
Release Time	0.5ms to 500ms	0.5ms to 500ms
Auto Attack and Release	Program Dependent	Program Dependent
Curve Type	Hard and Soft Knee	Hard and Soft Knee
Output Gain	-20dB to +20dB	-20dB to +20dB
	Compression Metering	
Threshold	Above/Below LEDs	Above/Below LEDs
Gain Reduction	6-Segment LED	8-Segment LED
Gate Metering	Open and Closed LED	Open and Closed LED
Power Supply	Internal, Linear Supply — Switchable for 100VAC to 120VAC or 200VAC to 240VAC	Switchable for 100VAC to 120VAC or 200VAC to 240VAC

BLUEMAX

Smart Compressor / Limiter With Presets



The Blue Max is a stereo dynamics processor with 15 presets designed to offer you time-tested, compression and limiting settings used by experienced professional engineers. This dynamics processor was built with state of the art components to deliver crystal clear compression that will add punch and presence and can increase the apparent loudness of solo instruments or entire mixes. 1/4˝ inputs and outputs accommodate both stereo and mono operation. A high gain mono input is also provided allowing the BlueMAX to be used as a preamp for Hi-Z instruments.

FEATURES

- Designed with a single set of controls for both the left and right channels to provide true stereo compression. Can also be used as a single mono compressor and act as a preamp for high impedance instruments such as guitar and bass

15 Presets and Manual Override

- Any one of the 15 presets are accessed using a sixteen position rotary encoder which digitally switches the attack, release, ratio and threshold settings of simultaneously for each preset.
- The 15 studio proven settings include:
  3 vocal presets, 8 instrument settings, 2 classic stereo settings, 2 special effects "squeeze guitar" and "pump"
- Manual mode is also provided to give you full control over compression parameters, including: Input and Output gain, Attack/release times and ratio from 1:1 to 20:1 — In manual mode the threshold is fixed at -10dB

Inputs and Outputs

- 1/4˝ unbalanced inputs and outputs

Sidechain

- Sidechain insert allows the unit to be used for special processing applications such as de-essing and ducking

Metering

- Full LED metering for input and output levels are available as well as separate meters for gain reduction

Additional Features

- Internal operating levels are switchable from +4dBu to -10dbV
- Internal power supply is internally switchable from 120 and 240 volt operation



Inputs and Outputs	Unbalanced 1/4˝ TS
Operating Level	+4/-10dB Selectable
Compressor/Limiter Controls	
Input Gain	-20dB to +40dB
Presets	15, Plus manual Override
Ratio	1:1 to 20:1
Attack	0.01ms to 100ms
Release	10ms to 500ms
Output Gain	-20dB to +20dB
Metering	
Threshold	Above/Below LEDs
8-Segment LED Metering	
Input/Output	-24dB to +24dB
Gain Reduction	-1dB to +28dB
Power Supply	Switchable for 100VAC to 120VAC or 200VAC to 240VAC

System Performance

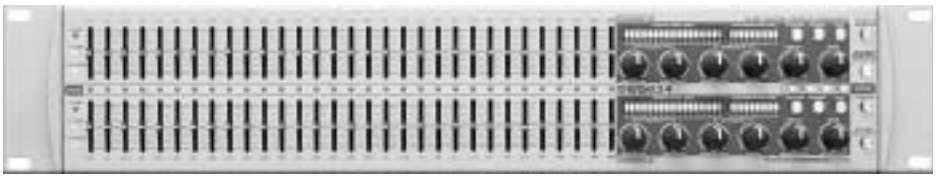
Dynamic Range	>115dB
Signal to Noise Ratio	>95dB
Headroom	+24dBu
Frequency Response	10Hz to 50kHz
Crosstalk	>82db @ 10kHz
THD + Noise	<0.03%

Input Specs	**Impedance**
1/4˝ TS Left/Mono	100k Ω unbalanced
1/4˝ TS Right	10k Ω unbalanced
1/4˝ TRS Sidechain	**Impedance**
Send	10k Ω unbalanced
Return	51 Ω unbalanced
Analog Output Specs	**Impedance**
1/4˝ TS	51 Ω balanced / unbalanced

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DEQ624

Digital Stereo Graphic Equalizer Plus Dynamics Processing



The DEQ624 is a 24-bit digital two channel/stereo 31 band graphic equalizer with a number of features that extends its capabilities way beyond those of a standard graphic EQ including variable high and low pass filters, brick wall limiting and expansion (noise reduction) for each channel. Another feature, unique to the DEQ624 is a proprietary Adaptive Hum Cancellation algorithm that allows you to virtually eliminate 60-cycle ground hum at the flick of a switch. Although the DEQ624 is a digital processing unit, under digital control it has a familiar analog feel with 20mm digital encoding faders, knobs and switches — no menus or sub-menus to scroll through that will slow you down on the gig. The rear panel of the DEQ624 provides balanced XLR and 1/4″ TRS inputs and outputs as well as barrier-strip terminals for permanent installation.

Graphic EQ Plus

- 24-Bit digital signal processing provides accurate processing of analog signals
- Each channel of the graphic EQ section is comprised of 31 digital EQ encoders (faders) centered in 1/3 octave increments from 20 Hz to 20 kHz
- A range switch for each channel lets you choose one of three boost/cut ranges for the digital encoders – ±6dB, ±12dB or +12/-24dB. The selected range is easily identified by LED's located at the left side of the EQ section
- Separate high and low pass filters are provided for each channel —
  - ➢ The low pass filter is variable from 10 kHz to 20 kHz
  - ➢ The high pass filter is variable from 10 Hz to 410 Hz
- To help minimize overall system noise, both channels have separate expanders with variable ratio and threshold controls
- A "brick-wall" limiter with a variable threshold control provides fail-safe system protection from extreme transients
- A HI Q switch let's you narrow the bandwidth of the frequency bands from Normal Q with a 1/3 octave bandwidth to Hi Q which has a 1/8 octave bandwidth. With Hi Q engaged, adjacent frequencies will be less effected by boost/cut frequencies
- AHC-Adaptive Hum Cancellation is a proprietary feature of the DEQ 624 that allows you to greatly reduce 50 Hz /60 Hz system hum caused by grounding problems with the flick of a switch
- A hardwire bypass switch for each channel ensures that audio will still pass from input to output in the event of lost power
- To facilitate ease of set-up, a Channel A master switch, when engaged, will slave the controls of Channel B to Channel A

System Performance

Dynamic Range	>105dB
Noise Floor	-95dBu
Signal to Noise	-90dB
Power Supply Rejection	-98dB
Headroom	+22dBu
Frequency Response	10Hz to 22kHz ±0.5dB
THD + Noise (Un-weighted)	0.005%

Input Specs	**Impedance**
XLR, 1/4″ TRS & Barrier strip	12k Ω
Output Specs	**Impedance**
XLR, 1/4″ TRS & Barrier strip	51 Ω

Inputs and Outputs

- XLR and 1/4″ TRS balanced inputs and outputs are provided as well as barrier-strip terminals for permanent installations

Metering

- Each channel offers a 20-segment LED meter for output gain as well as 10-segment LED for gain reduction metering

Additional Features

- Internal power supply
- Custom molded extruded aluminum and steel chassis

Preset Management

- A preset switch lets you toggle through four user-defined setups that include a "snapshot" of the settings of the front panel controls
- A security lock-out feature, allows you to defeat the front panel controls of the unit which can be restored with user-created three-switch code combination

Inputs and Outputs	Electronically balanced XLR, 1/4″ TRS and barrier strip
Output Gain Control	-20dB to +20dB
Graphic EQ	
Digital EQ Controllers	20mm faders — adjustable ±5 steps
Selectable Boost/ Cut Range	±6dB, ±12dB or +12/-24dB
Switchable Hi Q Control	1/3 oct (Normal Q) to 1/8 oct (Hi Q)
Filters	
Hi Pass Filter	12dB/oct variable from 10 Hz to 410 Hz
Low Pass Filter	12dB/oct variable from 10 kHz to 20 kHz
Limiter	
Threshold Control	off to 0dBu
Expander	
Ratio	1:1 to ∞:1
Threshold Control	off to -10dBu
Metering	
Output Gain	20-segment LED (– 42dBu to +22dBu)
Gain Reduction	10-Segment LED -12dBu to -0.5dBu)
Power Supply	Factory configured for 115V

CL44 and GTX44

Quad Series Dynamics Processors



Presonus' Quad Series dynamics processors consist of the CL44 Four Channel Professional Compressor/Limiter and the GTX44 Four Channel Frequency Dependent Gate/Expander. Both units combine a digitally controlled, high quality analog signal path with the added flexibility of digital sidechain processing which provides vintage compressor emulations, ducking and spectral processing. The GTX44 features high and lo pass filters, variable attack, release and hold parameters and key listen control. The CL44 has a variable Hi Pass Filter with an independent limiter and full featured compressor. Both units also feature stereo linking of adjacent channels as well as sidechain inserts, balanced XLR inputs and outputs and a single unbalanced 1/4˝ TRS I/O connector for each channel.

FEATURES

- Hybrid VCA, analog dynamics processor under digital control with DSP flexibility
- Digital sidechain processing allows multiple functions such as filtering, vintage compressor emulation and other special functions, to be applied simultaneously

Inputs and Outputs

- Balanced XLR inputs and outputs
- Unbalanced inputs and outputs are also available via a single 1/4˝ TRS jack – A 'flip' button reverses the 'tip and the 'ring' on the 1/4˝ jack to match the wiring of your mixer
- 1/4˝ TRS compressor sidechain insert with +4/-10dBv selector

Additional Features

- Adjacent channels are stereo linkable
- Internal Power Supply
- Backlit function buttons
- All steel chassis, one rack space



Partial rear view of GTX44

CL44 Features

- Fully adjustable attack, release, ratio, threshold control
- Variable High Pass Filter Allows you to pinpoint the frequency range in which the compressor operates
- Independent peak stop limiter
- Program dependent attack and release is also available using the Auto function that monitors audio over three separate time constants
- Compander (upward expander) circuit raises an audio signal when it falls below a desired threshold – when used in conjunction with the compressor provides Automatic Gain Control
- Special 'optical' mode emulates the optical transfer curve found in vintage compressors
- 10 segment LED gain reduction meter

CL44 Panel Controls:

Low Cut Filter	20Hz-8kHz
Threshold	+15dBu to –40dBu
Ratio	1:1 to 20:1
Gain	-20dB to +20dB
Compand	0% to 100%
Attack	01mS to 500mS
Release	01S to 3S
Limit	-10dBu to 24dBu (Off)
Optical Mode Switch	
Link 1-2 and 3-4 Switch	
Auto Switch	
Bypass Switch	
Gain Reduction Meters	LED -0.5dB to-24dB

GTX44 Features

- Variable threshold, ratio, attack, release hold and range controls can be applied to achieve noise-gating or downward expansion
- Adjustable High- and Low- Pass filters allow you to control the specific frequency range over which the gate will open – a sidechain monitor listen button lets you hear what frequency range the gate is operating within
- Duck mode allows for external input of microphone signal via Gate Key Input

Zero-Crossing Circuit

- Zero-Crossing function provides truly 'clickless' operation and chattering on gate settings that use quick attack-times

GTX44 Panel Controls

Low Cut Filter	20Hz-8kHz
Hi Cut Filter	250Hz-20kHz
Threshold	+15dBu to –40dBu
Ratio	1:1 to Gate
Range	0dB to -80dB
Attack	01mS to 500mS
Hold	001S to 1S
Release	01S to 2S
Sidechain Monitor Switch	
Link 1-2 and 3-4 Switch	
Duck Switch	
Bypass Switch	
Performance	
Gain Reduction Meters	LED -3dB to-30dB

ACOUSTI-Q

Tube Blender/Preamp



The ACOUSTI-Q is an affordable tube-based preamp/blender for acoustic instruments featuring Presonus' award winning dual servo gain stage design. The ACOUSTI-Q is the perfect solution for acoustic musicians helping to bring their personal style and sound to new heights! The ACOUSTI-Q gives you the choice to use a single (piezo pick-up) or dual (piezo pick-up and mini- internal microphones) source. With two sources, you can blend the signals to achieve the perfect mix. Guitars, woodwinds, basses, brass instruments virtually any acoustic instrument can benefit from the warm tube sound of the ACOUSTI-Q. The ACOUSTI-Q employs two dual-servo microphone preamplifiers with separate gain controls for input, a phase switch, a 12-volt phantom power supply to accommodate miniature, internal condenser microphones, a variable notch filter and sweepable mid- band EQ, as well as low and high shelving EQ controls. The master output section gives you the ability to set two separate volumes, one for normal volume, and the other, a boost, for solo playing.

Two Preamps

- The front panel instrument input is designed to handle 1⁄4″ TRS inputs from instruments equipped with both mini-condenser microphones and piezo pick-ups or 1⁄4″ TS from instruments equipped with pick-ups only
- A dual concentric volume control allows you to blend the incoming signals of two inputs
- A 12AX7 vacuum tube, after the master output, harmonically enhances the final signal
- +12VDC Phantom power is available to the Ring of the 1⁄4" TRS input for use by mini-condenser microphones installed in some acoustic instruments
- A phase (polarity) reverse switch compensates for different connector hook-ups and can also be used to achieve an intentional 'out-of-phase' sound
- A tunable Notch Filter variable from 92 - 494 Hz ±12dB is available via a dual concentric knob for isolating and controlling lower frequencies that may be prone to feedback
- A 3-band EQ is provided via a pair of dual concentric potentiometers
  - ➢ The dual mid-range control features a sweepable frequency range from 250Hz to 5kHz and a boost and cut of ±12dB
  - ➢ A brilliance control is provided with a fixed shelving frequency of 8kHz and a boost and cut of ±12dB
  - ➢ A bass control is also provided with a fixed shelving frequency of 100Hz and a boost and cut range of ±12dB
- The Master section also uses a dual concentric knob to control
  - ➢ The inner knob controls the Main output level from -80dB to +20dB
  - ➢ The outer knob, labeled Cut/Boost Footswitch, sets the amount, ±12db, that the output signal will be raised or lowered when used with an optional footswitch
- A Mute switch defeats the Main output while allowing the signal to remain present at the -20dB AMP/TUNER output
- A 16 segment LED meter, with a range of 42dBu to +24dBu (clip), is provided for monitoring the unit's Main output level

Rear Panel Connections

- Balanced XLR main output
- A 1/4″ TRS Effects Loop jack allows you to insert outboard effects (compressors, delay units, or reverbs). The Effects Loop can also be used as an additional output when used with a 1/4″ TS cable
- An unbalanced 1/4″ jack, operating at -20dB is provided as a signal source for Tuners and guitar amplifier inputs.
- A 1/4″ footswitch input is also provided for muting the main output as well as facilitating the previously mentioned instantaneous boost or cut capabilities

1/4″ TRS Inputs	
Tip	For standard pickups
Ring	For mini condenser mics
Input Controls	
Tip Volume	0dB to 40dB
Ring Volume	0dB to 40dB
Phase Invert	Switchable Polarity
Phantom Power (Ring)	Switchable +12VDC
3-Band Equalizer	
Mid Range	Sweepable 250Hz - 5kHz
Bass	Fixed Shelving 400Hz
Brilliance	Fixed Shelving 8kHz
Boost/Cut	±12dB for all bands
Notch Filter	92 - 494 Hz (±12dB)
Master Section	
Mute	Main Output Only
Master Level	-80dB - +20dB
Cut/Boost	±12dB (Footswitch Only)
Metering	
Threshold	Above/Below LEDs
8-Segment LED Metering	
Input/Output	-24dB to +24dB
Gain Reduction	-1dB to +28dB

System Performance

THD + Noise (Unweighted)	0.005% (0dBu) 0.05% (+10dBu output); 0.3% (+20dBu output)
Signal to Noise	>90dB
1/4″ TRS Input Specs	**Impedance**
Tip	10M Ω unbalanced
Ring	10M Ω unbalanced
Ring w/ Phantom Power	3.2k Ω unbalanced
1/4″ TRS Effects Loop	**Impedance**
Send (Tip)	51 Ω unbalanced
Return (Rings)	50k Ω unbalanced
Output Specs	**Impedance**
Main Out (XLR)	51 Ω balanced
1/4″ TS Tuner/Amp -20dB	51 Ω unbalanced

PREAMPLIFIERS

AP 13 Acoustic Instrument Processor



The AP13 is a dedicated preamp/processor specifically designed to blend the signals from acoustic instruments with two pickups. A single 1/4˝ TRS jack on the front panel, will accept both a very high input impedance piezo pickup and a microphone pickup simultaneously. Each of the two pickup signal paths feature powerful mixing and processing capabilities optimized for preserving the timbre of acoustic instruments while minimizing feedback, including individual Pan and Level controls, a variable Low Cut Filter and 7-band Graphic Equalizer. The AP13's wide ranging I/O capabilities include a stereo effects loop and Insert for each channel, balanced 1/4˝ line-level and stereo XLR mic-level outputs, a stereo headphone output with level control as well as dedicated 1/4˝ piezo pickup and tuner output connectors.

FEATURES

- A stereo 1/4˝ TRS input jack combines the individual signals from any acoustic instrument with two pickups but will also accept instruments with only one pickup
- A front panel switch lets you choose which pickup is wired to the tip or the sleeve of the Input jack
- Two signal processing paths one optimized for piezo pickups and one for mic pickups
  - The piezo path features very high input impedance for extended low bass response
  - The mic channel offers a choice of two phantom power voltages, +6Vdc and +15Vdc, satisfying the requirements of most electret condenser mic pickups
  - Either channel accepts dynamic pickups or line-level outputs

Flexible Outboard Processing

- Separate stereo send/return and insert loops for each channel allow seamless integration of effects and/or outboard dynamics processors into the signal chain
  - Individual send A/B controls adjust the amount of signal sent to 1/4˝ TS send jacks while a single stereo return control adjusts the signal entering the 1/4˝ TS return jacks
  - The insert loops for channels A and B are accessed via 1/4˝ TRS jacks

Each Channel Features

- Bump-proof input trim controls feature slotted openings to allow quick adjustment using picks, fingernails or coins
- Red overload indicators for each type of pickup make gain set-up easy
- An invert switch solves the annoying problem of out-of-phase pickups
- 12dB/oct Low Cut Filters, variable from 15 to 250 Hz help eliminate unwanted bass frequencies
- The 7-band graphic EQs give you separate timbre control over each pickup with ±12dB boost/cut and a 2 octave bandwidth per band
- Pan and level controls allow you to mix both pickups into one perfectly blended stereo signal
- The Low Cut filter, 7-band graphic EQ, pan, level, send and return Level controls are all logically arranged along the front panel using smooth-acting oil-damped sliders to provide ease-of-use and graphical feedback

Outputs

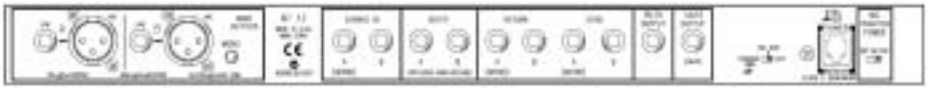
- Two pair of simultaneous, high-current cross-coupled balanced (mono switchable) stereo line drivers are provided
  - The stereo 1/4˝ TRS line-level outputs are provided for driving amplifiers
  - XLR mic-level outputs accommodate direct console patching
- A powerful stereo headphone amp with level control and Master Mute switch allow private auditioning and practice
- A 1/4˝ piezo output with a dedicated level control is available as an auxiliary output
- Separate 1/4˝ tuner output

Additional Features

- 1/4˝ TS expander inputs accommodate an external audio source such as a CD or tape for practicing
- UL/CSA/CE and 100/120/230 VAC remote power supplies

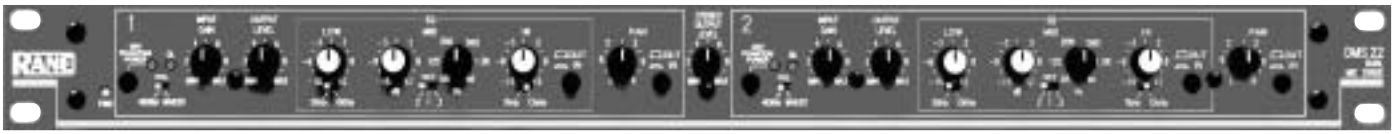
System Performance

Frequency Response	15-40 kHz, +0/-3 dB
Signal-to-Noise Ratio re +4 dBu, 20 kHz BW	81dB
THD+Noise (+4 dBu, 20-20 kHz)	0.03
Equivalent Input Noise (EIN) 60dB gain	120dB Inst Mic In to Send 115dB Inst PZO In to Send



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DMS 22 2-Channel Microphone Preamplifier

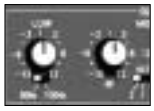


The DMS 22 is a dual channel studio-grade mic preamp with 3-bands of equalization and a stereo mixer. Perfect for studio recording in direct-to-tape or hard disk applications with audio quality surpasses that found in most professional mixing consoles. The two balanced XLR microphone inputs features switchable 48V phantom power, polarity invert switch, and variable input gain. Balanced 1/4˝ TRS line outputs with level control is provided for each channel. Each channel also features a defeatable pan control sent to a separate stereo pair of XLR mix outputs, allowing a stereo mix of the two microphones when hard left/right is not appropriate. The 3-band EQ (High and Low shelving plus a fully parametric Mid-band) on each Channel offers a broad palette of tonal control. A selectable Low Frequency Filter is available on each output channel to remove mic bumps and rumble that might interfere with a recording. The DMS 22 goes beyond the concept of a dual mic preamp in two ways—the built-in Mixer section, and the comprehensive Equalizer circuitry. This built-in mixer for the two Channels eliminates the need for an external console when a portable setup is desired.

FEATURES

- Each XLR microphone input features switchable 48v phantom power, polarity invert switch, and input gain control

3-Band Equalizer



- Each channel features low and high shelving bands and a parametric mid band, each with a boost/cut range of +12/-15 dB
- Patented Accelerated-Slope tone controls with steep phase-corrected slopes eliminate most of the traditional interaction between shelving and midband controls
- The low and high bands provide switch selectable shelving frequencies
  - ➢ Low – 50 Hz and 100 Hz
  - ➢ High – 7 kHz and 12 kHz
- Parametric bandpass Mid band
  - ➢ Sweepable frequency, variable from 95 to 4k Hz
  - ➢ 3-position Q control switchable to 0.33 / 1.0 / 2.0 oct bandwidths
- An EQ engage switch serves as a bypass control of the EQ section
- A selectable 15/50/100 Hz low frequency filter is provided on each channel

Stereo Mixer and Outputs

- Independent (defeatable) pan controls for each channel allow you send a stereo mix of the two mic channels into a pair of XLR mix outputs, adjustable by a stereo output level control — this can be useful when recording direct-to-tape or disk
- Balanced, line-level 1/4˝ TRS outputs with Level controls are provided for each channel
- The direct outputs and stereo mixer outputs are available simultaneously

System Performance

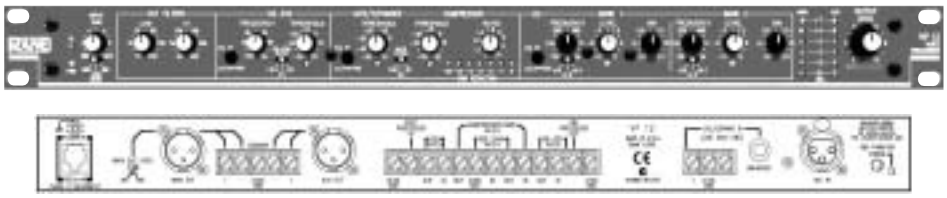
Equivalent Input Noise	-130 dBu (full bandwidth, 60 dB gain, 150 Ω)
Frequency Response	20-20 kHz +0/-.5 dB
Signal to Noise Ratio	97dB (re +4 dBu, +15 dB gain, full bandwidth)
THD+Noise	0.02% (+4 dBu, full bandwidth, max gain)
IM Distortion	0.03% (+4 dBu)
Mic Input	Impedance 1k Ω
Output Impedance	100 Ω

Input Gain Range	+15 to +60 dB
Phantom Power	+48 Vdc
3-Band Accelerated Slop EQ Section	
Boost/Cut Range	+12/-15 dB
Low Frequencies	Switchable 50 or 100 Hz
Hi Frequencies	Switchable 7k or 12k Hz
Mid Frequency Sweep	Variable from 95 to 4k Hz
Mid Bandwidth Range	3-position switch 0.33 / 1.0 / 2.0 oct
Low Frequency Filters	Switchable 15/50/100 Hz 15 dB/octave, Butterworth
Metering	
Overload LED	Threshold 16 dB
Inputs and Outputs	
Mic Inputs	RF Filtered, XLR Balanced, Max Input Level 6 dBu Min gain
Outputs	Active balanced XLR and 1/4˝ TRS



PREAMPLIFIERS

VP 12
Voice Processor



The VP 12 is a mic or line level voice processor incorporating a studio-grade microphone stage, low and hi cut filters, de-esser, gate/compressor, and two channels of parametric equalization.

- XLR Input with switchable 48V phantom power and input gain control
- Balanced screw terminal and 1/4″ TRS line level inputs
- The line level input is designed to easily accept the output of wireless mic systems
- A front panel switch selects whether the mic, line, or the sum of both inputs will be processed
- The sum feature allows the line input to function as an aux input for additional outboard mic preamps
- Variable Low and Hi-Cut Filters tailor the overall frequency range to the specific application.
- A de-esser features variable frequency and threshold controls to help remove sibilance

Gate and Compressor Controls

- The gate features a variable threshold control and a 3-position ratio control that insures a smooth transition from active output to off without pumping artifacts
- The compressor features variable threshold and ratio controls
- Seven-segment gain reduction meter lets you know just how much the compressor is compressing

Two band Parametric EQ

- Each band has a multiplier switch that covers a frequency range of 10 to 20k Hz with +12 dB of Boost and -15 dB of Cut as well as a variable wide to narrow bandwidth control
- Because the filters are in series, setting the filters to the same frequency can double the amount of available boost or cut

Outputs

- Dual concentric front panel level controls for both the main and aux outputs
- Fully balanced XLR and screw terminal main outputs switchable to either line or mic level as well as fully balanced XLR and screw terminal aux outputs – main and aux outputs can both be used at the same time
- Independent six-segment LED meters for the main and aux outputs are accurately calibrated in peak dBu

Additional Features

- Independent bypass switches are provided for each processing section
- Each section of processing can be re-patched in any order via screw terminal jumpers on the rear of the unit

System Performance

Frequency Response	20 - 20k Hz +0/-3 dB EQ out, no cut filtering
Signal to Noise Ratio	97 (re +4 dBu) dB +15 dB gain, 20 kHz BW
THD+Noise	0.2 % +4 dBu, 20-20 kHz BW max gain
IM Distortion	0.03% (+4 dBu, 60 Hz/7 kHz, 4:1)
Equivalent Input Noise	-130 dBu 20 kHz BW, 60 dB gain, 150 ohms

Input Specs	**Impedance**	**Maximum Input Level**
XLR Mic	1k Ω Balanced	+3.5 dBu
1/4″ TRS and screw terminal line	20k Ω	+20 dBu
Output Specs	**Impedance**	**Maximum Output Level**
	100 Ω	

RF Filtered Inputs	
Mic	XLR Balanced
Line	Active Balanced/Unbalanced 1/4″ TRS and Screw Terminal
Preamp Controls	
Gain Range	+15 to +60 dB
Phantom Power	+48 VDC
12dB/Octave Low & Hi Cut Filters	
Low Cut Filter	Variable from 10 Hz to 250 Hz
Hi Cut Filter	Variable from 3 kHz to 40 kHz
De-esser	
Frequency Range	700 Hz - 9 kHz
Threshold	-50 to +20 dB
Ratio Settings	Switchable Mild, Norm, Max %
Gate Controls	
Threshold Range	-50 to +10 dB
Ratio Settings	1.5:1, 2:1, 3:1 (Switchable)
Compressor Controls	
Threshold Range	-50 to +20 dB
Ratio Range	1:1 to 10:1
Two-Band Fully Parametric EQ Section	
Frequency Range	10-20 kHz
3-position range switch	
x0.1	10 Hz to 200 Hz
x1.0	100 Hz to 2 kHz
x10	1 kHz to 20 kHz
Boost/Cut Range	+12/-15 dB
Bandwidth	.03 to 2 Octaves
Outputs	Active balanced 1/4″ TRS and screw terminal
Metering	
Gain Reduction	7-Segment LED
Main	6-Segment LED
Aux	6-Segment LED

All Rane products are certified to meet all local and federal agency regulations Specific agency listings are available on applicable data sheets

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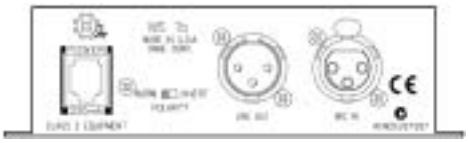
PREAMPLIFIERS



MS 1b Microphone Preamplifier

The Rane MS 1b is a compact mic preamplifier designed for direct recording or for adding a microphone channel to line-level mixers. Go from either a dynamic, condenser or electret microphone to a line-level input with a minimum of noise, distortion, cost and hassle. The MS 1b utilizes one of the finest ultra low noise amplifier designs available. Featuring a true differential input with high common-mode rejection, use of the MS 1b guarantees performance usually found only in mixing consoles costing thousands of times as much.

- Ultra low-noise, low distortion design utilizing the Burr-Brown mic preamp chip improves the dynamic range and boosts performance over the original MS1
- RF filtered XLR balanced input
- Continuously variable rotary gain trim between 18 dB and 66 dB
- Switchable 48 volt phantom power with indicator LED
- Polarity switch
- Signal/overload LED
- Can mount close to the mic to provide a local volume control, with a stronger signal that minimizes RFI and hum
- True high current cross-coupled XLR balanced output line driver designed to emulate a transformer output
- U.L. listed, C.S.A. certified, and CE certified



System Performance

Signal to Noise Ratio (18 dB Gain)	96 dB 20Hz -20kHz, 150Ω (+4 dBu)
Equivalent Input Noise (60 dB Gain)	-128dBu (typical) 20Hz -20kHz, 150Ω
Dynamic Range (18 dB Gain)	120 dB typical
CMRR (60 dB Gain)	80 dB (typical) 150Ω, 120 Hz
Frequency Response (18 dB Gain)	30 to 200k Hz (+0, -3db)
THD+Noise (18 dB Gain)	.001% (Output=+20 dBu) 50 to 20k Hz, 20 kHz BW, 10k Ω

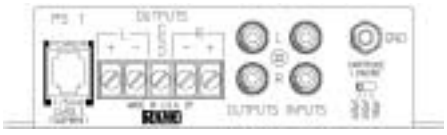
Input Specs	**Impedance**	**Max. Input Level**
XLR	10k Ω Balanced	+10 dBu (Gain 18dB)
Output Specs	**Impedance**	**Max. Output Level**
XLR	50 Ω	+22 dBu (unbal)/+27 dBu (bal) 2k Ω load

PS 1 High Fidelity Phono Preamplifier



The PS 1 Phono Stage is a complete professional RIAA phonograph stereo preamplifier designed for studio and remote turntable applications as well as permanent archival installations such as libraries, schools and museums. The PS 1's features include balanced screw terminal outputs capable of driving extremely long lines, the correct loading capacitors for all popular phono cartridges, and properly designed rumble filters that remove infrasonic signals without interfering with the audio signals. Ideal for expanding existing systems or for mixers which have no available phono inputs.

- Converts RIAA (phono) to line level with 0.1 dB RIAA accuracy
- Selectable cartridge loading capacitors
- 18 dB/oct rumble filter @ 15 Hz
- Cross-coupled high-current line drivers with balanced screw terminal & unbalanced RCA outputs
- Pristine audio path using audio grade components
- Linear tech and analog devices IC'S
- UL/CSA/CE remote power supply

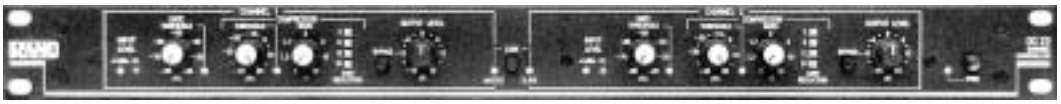


System Performance

Max Output Level	+22 dBu Cross-coupled balanced into 600 Ω
Output Impedance	50 / 100 Ω Balanced / unbalanced
Channel Separation	80 dB 1 kHz (50 dB @ 20 kHz)
THD+Noise	0.02% (+4 dBu, 20-20 kHz)
Signal-To-Noise Ratio	84 dB (re +4 dBu, 20 kHz Noise BW)
Gain	+42 / +36 dB Balanced / unbalanced out
RIAA Equalization Accuracy	0.1 dB NAB (RIAA) standard curve
Infrasonic (Rumble) Filter	15 Hz, 18 dB per Octave Butterworth high-pass
Input Capacitance	Selectable 100 / 250 / 425 pF

COMPRESSOR/LIMITERS

DC 22 2-Channel Dynamic Controller



The DC 22 is an easy to use, dual channel, dynamic controller that combines high quality VCA compression and gating with downward expansion. The Compressor section features a variable threshold ratio control capable of subtle dynamics control to peak limiting. The Expander/Gate musically attenuates signals below the set threshold level. This in contrast to a Gate only which shuts off any signals below the threshold. The link switch allows the DC22 to be used for true stereo or independent dual channel processing. Compatible with balanced and unbalanced equipment. Designed for the working musician or DJ, The DC 22 provides reduced complexity without compromise in audio quality or dependability.

FEATURES

- Dual channel compressor/gate using high quality log RMS (averaging) detectors and VCAs
- A link switch allows the DC22 to be used as a true stereo processor with channel 1 controlling both channels

Inputs & Outputs

- Both XLR balanced and 1/4″ TRS balanced/unbalanced inputs are provided
- An input gain switch, on the rear panel, allows you to connect professional (+4dBu) and consumer (-10dBv) equipment to the 1/4″ TRS inputs
- XLR balanced and unbalanced 1/4″ TS outputs connectors are also provided

Expander/Gate

- The adjustable gate is a downward expander (a compressor working in reverse) with a variable threshold control and fixed ratio of 2:1. When the signal drops below the set threshold, the gate threshold indicator lights and the output level is reduced by 2 dB for every 1 dB the input signal level drop

Compressor

- The compressor features variable threshold and a ratio controls
- The ratio control is variable from 1:1 up to ∞:1 to allow the compressor to be used for limiting applications including system protection and digital recording
- The ±15dB output level controls provide make-up gain lost due to compression

	DC22	DC24
	Expander / Gate	
Threshold Range	-20 to +80 dB	-50 to +10 dB
Ratio Range	2:1 (fixed)	1:1 to 20:1
Threshold Activity Meter		LED
	Compressor	
Threshold Range	-40 to +10 dB	-50 to +20 dB
Ratio Range	1:1 to ∞:1	1:1 to 10:1
Threshold Activity	LED	LED
Gain Reduction	4-Segment LED (-3 to -24dBu)	–
	Limiter	
Threshold Range	–	-20 to +20 dB
Gain Reduction	–	5-Segment LED (0 to -24dBu)
	Crossover	
Linkwitz-Riley	–	4th-Order 24 dB per octave slopes
Frequency Range	–	70 Hz to 7 kHz (41-detent pot)
	Output Section	
Gain Range	±15dB	±12 dB, center detent @ unity gain
Bypass Switch	✓	Hardwire
Signal Present LED	+4 dBu	-40 dBu
Overload LED	✓	+22 dBu
Inputs	20k Ω / Max Input +20dBu	20k Ω / Max level +20dBu
XLR Inputs	Balanced	Balanced
1/4″ TRS Inputs	Bal/Unbal (switchable +4dBu/-10dBv)	
Outputs	Max Output Level +20dBu (2kΩ)	Max Input level +20dBu (600Ω)
XLR Outputs	100 Ω Balanced	100 Ω Balanced
1/4″ Outputs	300 Ω TS Unbalanced	100 Ω TRS Balanced
1/4″ TRS SideChain	–	Tip = Send, Ring = Return

System Performance

	DC22	DC24
Frequency Response	20 Hz - 40 kHz +0/-.5 dB R load > 2 kHz	20 Hz-20 kHz (+0/-.5 dB)
THD+Noise	0.02% type.(+4 dBu, 1 kHz)	0.05% type. (+4 dBu, 1 kHz)
IM Distortion (SMPTE)	–	0.1% (60 Hz / 7 kHz, 4:1, +4 dBu)
Signal-to-Noise Ratio	100dB (Unity gain, +4 dBu, 20kHz BW)	92dB (Unity gain +4 dBu, 20kHz BW)
Common Mode Rejection	40dB min.	

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COMPRESSOR/LIMITERS

DC 24 2-Channel Dynamic Controller with Built-in Crossover



The DC 24 is a two channel dynamics processor with independent expander/gate and compressor functions as well as dedicated limiter circuits that provide overall system protection on top of the other dynamics controls. Unique to the DC 24 is its built-in 24 dB/octave Linkwitz-Riley crossover which allows the unit to operate as a two-way speaker dividing network or for multiband dynamics control of a single audio source. Both balanced XLR and balanced/unbalanced 1/4˝ TRS inputs and outputs are provided as well as a 1/4˝ TRS side-chain insert on each channel.

SAME FEATURES AS THE DC22 *PLUS*—

- The expander/gate section features variable threshold as well as ratio controls allowing you to select the amount of downward expansion — higher ratios will provide greater attenuation to signals below the set threshold.
- The attack and release times of the compressor and expander/gate circuits use dedicated RMS-sensing ICs that automatically adjust to the program material –
  - ➢ If the input is predominantly low frequency, the times are made more gradual and slowed
  - ➢ For more transient signals, the times are tightened

Limiter

- Limiter circuits, located after the compressors, feature their own threshold control, providing dedicated peak-stop limiting before the units output

Crossover

- 24 dB/octave Linkwitz-Riley crossover with two operating modes —
  - ➢ Low-High crossover mode (1 In/2 Out) allows the DC24 to be used as a two-way speaker dividing network along with a fully featured expander/gate, compressor and limiter on each output
  - ➢ Bandsplit Combine mode (1 In/1 Out) Provides multi-band dynamics processing by dividing a single channel of audio in two separate frequency ranges that can independently processed and then recombined into one channel

Inputs and Outputs

- RF filtered, active balanced XLR and bal/unbal 1/4˝ TRS inputs and outputs
- -10 dBV / +4 dBu gain switch
- 1/4˝ TRS side-chain inserts, on each channel, allow spectral dynamics and ducking control from external sources

Additional Features

- Passive hard-wire bypass switches ensure total bypass of the unit's active circuitry in the event of power failure
- All steel construction
- UL/CSA remote power supply (120 VAC)



*The following is taken from "**Squeeze Me, Stretch Me: The DC 24 Users Guide**" to expound the virtues of the DC-24's split-band dynamics processing capabilities.*

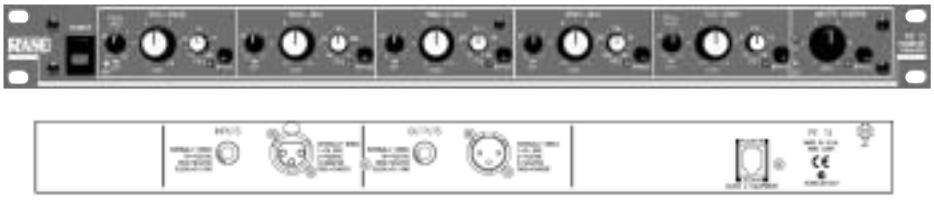
The DC 24 has two great advantages over other compressors – the crossover and the dual channels. It gives you complete control of the signal and processing of it. This is something that wasn't available before in a single unit. One stereo or two mono comp/limiters and one crossover would be required to do what the DC 24 does in a single rack space.

Use it on bass guitar, piano, drums, vocals — anywhere you've used a compressor/limiter before. The DC 24 gives you more control and a less tortured sound. In fact, split-band processing works so well that a DC 24 sounds good compressing an entire mix (two required for stereo in split-band mode).

Of special interest are instruments which have large level differences in their different tonal ranges. String pops on a bass are one, but flute is another. The higher tones require more breath and are much louder than the lower. Another good application would be a drum mix or submix. A split-band compressor does a better job of smoothing the performance out.

PARAMETRIC EQUALIZERS

PE 15 5-Band Parametric Equalizer



The PE 15 is a single channel parametric equalizer with five filter sections. Each of the five bands features independent control over the center frequency, bandwidth, as well as boost and cut. In addition, bands one and five can be switched to allow peaking or shelving characteristics. Special attention to the design of the state-variable filters allows any control to be operated in any order without affecting the others. Ideal for any sound shaping application from razor sharp notch filtering to broad-band program contouring.

FEATURES

5-Band EQ Section

- Five overlapping filters cover a center frequency range of four octaves
- Bandwidth for each band is variable from .03 to 1.5 octaves
- The boost/cut range of each band is variable from +15 dB to -20 dB
- Bands 1 and 5 are switchable to allow peak or shelving characteristics
- Individual bypass switches and LED indicators allow separate auditioning of each band

Flexibility

- The extra-deep 20 dB cut range and narrow (1/30th octave) bandwidth capabilities available on all 5-bands, are effective against troublesome feedback or removing any undesired slice of audio resonance without audibly affecting sonic quality
- The four octave range allows multiple band correction to the same frequencies when required — at least two bands will always overlap; most frequencies can be reached by three bands; and 1 kHz is covered by four bands.

Low & High Cut Filters

- A steep 18 dB/octave Low Cut Filter at 15 Hz, and a 12 dB/octave High Cut Filter at 33 kHz allow bandlimiting without using any of the five parametric bands

Output Section

- An overall Master Control Level covers a range of completely off to +20 dB
- A “hard-wire” bypass switch defeats all effects in the unit, ensuring that, in the event of a power failure, audio will continue to pass through the unit
- A system overload LED indicates a level of 18dBu or higher

Inputs and Outputs

- RF filtered, active balanced/ unbalanced XLR and 1/4″ TRS inputs
- Active balanced XLR and 1/4″ TRS outputs

Additional Features

- UL/CSA/CE and 100/120/230 VAC remote power supplies
- All steel construction

Five Band Equalizer	
Type	Fully Parametric Bands 1 and 5 switchable to shelving
Boost/Cut Range	+15 to -20 dB Minimum phase design
Bandwidth Range	.03 to 1.5 Octaves Continuous sweep
Frequency Range Per Band	4 Octaves
Band 1	20 Hz to 300 Hz
Band 2	60 Hz to 1 kHz
Band 3	150 Hz to 2.5 kHz
Band 4	450 Hz to 8 kHz
Band 5	1 kHz to 20 kHz
Bypass Switches w/ LED	One per band
Filters	
Infrasonic Filter	15 Hz, 18 dB/Oct., Butterworth
Ultrasonic Filter	33 kHz, 12 dB/Octave
Output Section	
Output Gain Range	Off to +20 dB
Bypass Switch	Passive “hardwire”
Metering	
Overload LED	+18 dBu
Inputs	
XLR & 1/4″ TRS	RF Filtered, Active Balanced/Unbalanced
Outputs	
XLR & 1/4″ TRS	Active Balanced

System Performance

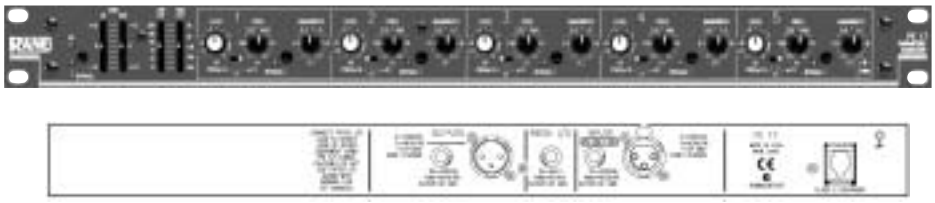
Frequency Response	15-33 kHz +0/-3dB
THD+Noise	0.02 % +4 dBu, 20-20 kHz
IM Distortion (SMPTE)	0.009 % 60 Hz/7 kHz, 4:1, +4 dBu
Signal-to-Noise Ratio	73 dB Boost/Cuts maximum, max gain 92 dB Boost/Cuts centered, unity gain (re +4 dBu, 20 - 20k Hz)

Input Specs	**Impedance**	**Max Input Level**
XLR & 1/4″ TRS	20k Ω	+20 dBu
Output Specs	**Impedance**	**Max Input Level**
XLR & 1/4″ TRS	200 Ω	+24 dBu (> 600 ohms)

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PARAMETRIC EQUALIZERS

PE 17 5-Band Parametric Equalizer



The PE 17 is a 5-band parametric equalizer/notch filter with advanced features and performance capabilities over the PE 15's already well thought out design. Some of these enhancements include separate in and out gain controls, 100% overlapping bands for all 5 EQ sections as well as improved dynamic range and noise performance. Special design techniques and components used in the design of the PE 17 guarantee residual noise levels below digital recording equipment. This makes the PE 17 ideally suited for any critical digital audio studio, sound reinforcement and broadcast applications.

FEATURES

5-Band EQ Section

- All five parametric bands are identical and offer 100% overlap with a frequency sweep range of 10 Hz to 20 kHz. This provides the unit with the flexibility to allow:
  - ➢ All bands to work together within a specific frequency range with summing of boost and cut
  - ➢ For each band to cover five separate parts of the audio spectrum *or*
  - ➢ Anywhere in between
- Each filter has a boost/cut range of +12 dB to -15 dB as well as continuously variable bandwidth control from as narrow as 1/30 of an octave to as wide as two octaves

Low and High Cut Filters

- Variable 12 dB/octave Low and High Cut band limiting Filters offer great flexibility for tailoring a sound systems overall frequency response to its exact application
  - ➢ The Low Cut Filter has a range of 10 Hz to 250 Hz
  - ➢ The High Cut Filter has a range of 3 kHz to 40 kHz

Inputs and Outputs

- Balanced XLR and 1/4″ TRS ins and outs
  - ➢ The input stage uses 0.1% matched resistors, to provide superior common-mode rejection of unwanted signals
  - ➢ The output stage uses "cross-coupled" high current line drivers which accommodate extremely long cable runs
- A 1/4″ TRS patch I/O jack allows you to patch the unit into a console's insert points using a single 1/4″ TRS cable

Additional Features

- Separate ±12 dB input and output gain slide controls —
  - ➢ The In control is calibrated top-to-bottom ±12 dB, while the Out is calibrated just opposite. This allows you to move both sliders together, ensuring that you always maintain unity gain through the EQ section, while varying the gain/attenuation occurring at the unit's input or output
- A system overload LED illuminates when signals come within 3-4dB of clipping

Five Band Equalizer	
Type	Fully Parametric
Boost/Cut Range	+12 to -15 dB Minimum phase design
Bandwidth Range	.03 to 2 Octaves Continuous sweep
Frequency Range Per Band	100% Overlap via three frequency range switches
Freq Sw: x.1	10 Hz to 200 Hz
Freq Sw: x1	100 Hz to 2 kHz
Freq Sw: x10	1 kHz to 20 kHz
Bypass Switches w/ LED	One per band
Variable 12 dB/Oct Butterworth Filters	
Low Cut Filter	10 Hz-250 Hz
High Cut Filter	3 kHz-40 kHz
Input / Output Section	
Input Gain Range	±12 dB
Output Gain Range	±12 dB
Bypass Switch	Passive "hard-wire"
Metering	
Overload LED	+20 dBu
Inputs	
XLR and 1/4″ TRS	RF Filtered, Active Balanced/Unbalanced
Outputs	
XLR and 1/4″ TRS	Active Balanced

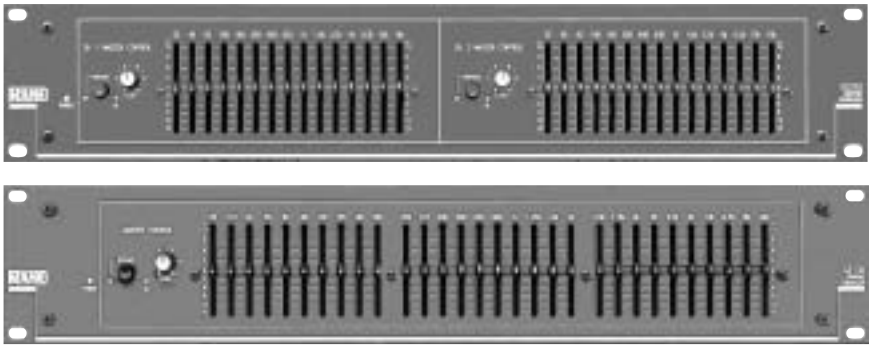
System Performance

Frequency Response	20 Hz-20 kHz (±0.5 dB)
CMMR	65 dB (20-1 kHz, rising to 50 dB @ 20 kHz)
Dynamic Range	120 dB (Noise floor-to-clipping into >2k Ω)
THD+Noise	0.005 % (+4 dBu, 20-20 kHz, 80 kHz Bandwidth)
Signal-to-Noise Ratio	98 dB (+4 dBu, 20 to 20k Hz Boost/Cut centered, unity gain)

Input Specs	**Impedance**	**Max Input Level**
XLR & 1/4″ TRS	6.6k Ω	+23 dBu
Output Specs	**Impedance**	**Max Input Level**
XLR & 1/4″ TRS	200 Ω	+22 dBu (> 600 ohms)

GE 215 / GE 130

Dual Channel 2/3 Octave and Single Channel 1/3 Octave Graphic Equalizers



The 2RU GE 215 dual 15-band and GE130 single 30-band graphic equalizers each feature Rane's innovative constant-Q (constant bandwidth) design which guarantees that the bandwidth of each filter is narrow enough to prevent unwanted interaction between filters, yet wide enough to produce exactly the type of correction curve required to control even the most unusual acoustical environment. Each active filter band features a boost/cut range of +12/-15dB on rugged 45 mm filter slider controls. Additional front panel controls and indicators, include an overall input level control, overload LED indicators as well as fail-safe hardwire bypass switches for each channel. The rear of the unit provides both active balanced XLR as well as Euroblock screw terminal inputs and outputs.

EQ Section

- Dual channel, 15-band and single channel 30-band graphic EQs — frequency bands located on standard ISO center frequencies
- Constant-Q filter design minimizes interactions between adjacent bands
- +12/-15 dB boost/cut range
- 45 mm filter slider controls with grounded center detents and dust dams

Inputs and Outputs

- Fully active balanced XLR and Euroblock terminal inputs and outputs

Additional Features

- Input level control and overload indicators for each channel
- Built-in infrasonic and ultrasonic filters eliminate low-end rumble and RFI
- Hard-wire bypass switches are provided for each channel allowing direct comparison between the equalized and non-equalized signal – these switches require no power to operate, and provide automatic bypass in case of power failure.
- Output relays provide a brief turn-on delay, eliminating power-up thumps
- 2 RU high and only 5″ deep
- UL/CSA/CE and 100/120/230 VAC remote power supplies
- Constructed from cold-rolled steel

Equalizer Section	
Frequency Range	25 Hz to 16 kHz
Accuracy	3% from center frequency
Type	Constant Q
Boost/Cut Range	+12 dB / -15 dB Cut
Faders	45mm sliders, grounded center detent
Output Section	
Overall Gain Range	Off to 0dB (Unbalanced) Off to +6dB (Balanced)
Bypass Switches	Hard-wire fail-safe design
Built-In Filters	
Infrasonic Filter	20 Hz, 18 dB/Oct., Butterworth
Ultrasonic Filter	RF Interference
Metering	
Overload LED	3 dB below clipping
XLR & Euroblock Inputs and Outputs	
Inputs	RF Filtered, active balanced / unbalanced
Outputs	Active balanced / unbalanced

System Performance

Frequency Response	20-140 kHz (+0/-3 dB)
THD+Noise	0.009 % (+4 dBu, 20-20 kHz)
IM Distortion (SMPTE)	0.005 % (60 Hz/7 kHz, 4:1, +4 dBu)
Signal-to-Noise Ratio	(re +20 dBu/+4 dBu) 110/94 dB Sliders centered, unity gain, bal. 93/77 dB Full boost, unity gain, bal.
Channel Separation	90 dB (typical @ 1 kHz)

Note: 0 dBu = 0.775 Vrms

Inputs	**Impedance**	**Max Input Level**
XLR & Euroblock	20k Ω bal.	22 dBu
XLR & Euroblock	10k Ω unbal.	22 dBu

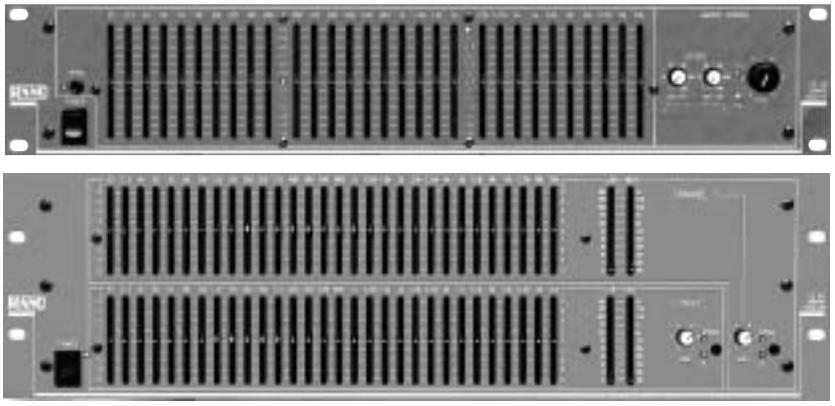
Outputs	**Impedance**	**Max Output Level**
XLR & Euroblock	200 Ω bal.	+19 dBu
XLR & Euroblock	+21 dBu (2k Ω)	(600Ω)
XLR & Euroblock	100 Ω unbal.	+13 dBu
XLR & Euroblock	+15 dBu (2k Ω)	(600Ω)

Constant-Q Graphic EQ Design

Constant-Q graphic equalizers arose from the sound professional's need for greater control with less interaction than previously possible with proportional-Q equalizers. Truth in slider position became a requirement. The curve traced out by the slider positions on constant-Q designs indeed represents the actual changes to the frequency response. On proportional-Q designs they do not. Use a constant-Q graphic the same way as a proportional-Q graphic. Desired results are achieved quicker, with far less after-adjustment to the adjacent sliders thus, eliminating a phenomena Rane calls "equalizing the equalizer."

GE 30/GE 60

Single and Dual Channel 1/3-Octave Graphic Equalizers



The GE 30 single-channel and GE 60 dual channel, 1/3 octave graphic equalizers represent Rane's top of the line EQs recommended for studios and broadcast as well as the most demanding live sound reinforcement applications. They take the already professional capabilities and robustness of the GE 215 and 130 to the next level with added flexibility and advanced features. Both units feature long throw, 60 mm high resolution slide controls on each filter band, sweepable high and low band-limiting filters as well as switch-selectable active balanced or transformer balanced XLR and screw terminal outputs that will ensure that any stubborn system ground problems can be eliminated. Another important feature is the use of Interpolating Constant-Q filters, which allow you to equally boost or cut two adjacent frequency bands and achieve a frequency peaks (or dips) that are in between the center frequencies of the filters. The GE 30 also adds the ability to choose between of boost/cut or cut only operating modes.

SAME FEATURES AND SPECIFICATIONS AS THE GE 215 AND GE 130 *PLUS*—

EQ Section

- 60 mm long throw, high resolution slide controls on each filter band with grounded center detents
- Dual operating modes (GE 30 only) — boost/cut and cut-only (switchable on the rear panel) allows your EQ choice to be made at the job site, instead of having to anticipate the requirements beforehand

Sweepable Band-Limiting Low Cut and High Cut Filters

- 12dB/oct sweepable filters from 10-250 Hz on the low end and from 3.1-40 kHz on the high end — restricting these signals can greatly improve system intelligibility

Inputs and Outputs

- RF balanced XLR and screw terminal inputs and outputs
- The GE 60 also provides balanced 1/4˝ TRS inputs and outputs
- A push button, on the rear panel, allows you to select active balanced or transformer coupled balanced outputs
- Signal ground and chassis ground terminals are also provided to ensure complete interconnect flexibility

Additional Features

- Overload and signal present LEDs
- Steel security panel included

Equalizer Section	
Frequency Range	25 Hz to 20 kHz
Type	Interpolating Constant Q
GE 60 Boost/Cut	±12dB
GE 30 Modes	Boost/Cut or Cut-Only
Boost/Cut Range	+12 dB / -15 dB Cut
Cut Only	-20dB
Faders	60mm sliders, grounded center detent
Output Section	
Overall Gain Range	Off to +6dB (Balanced)
Sweepable Filters	
Low Cut Filter	10 Hz-250 Hz, 12 dB/octave
High Cut Filter	3.1 kHz-40 kHz, 12 dB/octave
LED Metering	
Overload	20 dBu (Output or any internal level)
Signal Present	-20 dBu (Input level)

System Performance

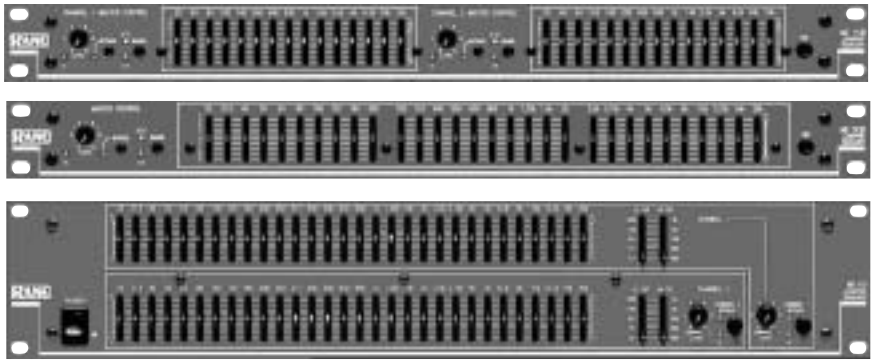
	GE 30	GE 60
Frequency Response +0/-3 dB	10 Hz - 40 kHz	10 Hz - 40 kHz
THD+Noise (+4 dBu, 20-20 kHz)	0.015 %	0.009 %
IM Distortion (SMPTE) (60 Hz/7 kHz, 4:1, +4 dBu)	0.008 %	0.005 %
S/N Ratio (re +20 dBu/+4 dBu) Sliders @ 0 dB, max gain	111/95 dB (Boost/Cut Mode) 104/88dB (Cut-Only Mode)	115/95 dB
Channel Separation	—	75 dB (1 kHz)
Common Mode Rejection	—	40 dB 1 kHz

Note: 0 dBu = 0.775 Vrms

GE30 Inputs	**Impedance**	**Max Input Level**
XLR & Screw Term.	10k Ω bal.	22 dBu
GE30 Outputs	**Impedance**	**Max Output Level**
XLR & Screw Term.	200 Ω bal.	+24 dBu (600Ω)
GE60 Inputs	**Impedance**	**Max Input Level**
XLR, S.T, 1/4˝ TRS	20k Ω bal.	21 dBu
GE60 Outputs	**Impedance**	**Max Output Level**
XLR, S.T, 1/4˝ TRS	200 Ω bal.	+22 dBu (600Ω)

ME 15B / ME 30B / ME 60

microGraphic Equalizers



The Rane ME Series microGraphic Equalizers exist for people requiring the best equalizer value possible in the smallest space allowable. They offer the same high quality constant-Q performance as their long throw relatives, the GE's. No compromises or trade-offs exist in selecting the microGraphics —Only the slider throw is different. All circuitry, components and specifications are essentially identical. The microGraphic series consists of three models — the ME 15B dual channel, 15-band 2/3-octave and the ME 30B, single channel 30-band 1/3-octave are housed in single space rack mount enclosures and feature a range switch that allows you to choose between ±12dB and ±6dB boot/cut. The ME 60 is a dual channel 30-band 1/3-octave model house in a two space rack mount chassis that inherits the Interpolating Constant-Q design of its big brothers the GE 30 and the GE 60.

They All Feature

- Constant-Q design minimizes interactions between adjacent bands eliminating the need to "equalize the equalizer"
- Compact 20 mm sliders with dust dams detented and positively grounded at 0 dB
- The boost/cut range is switchable between ±6dB and ±12dB — The ±6 dB mode offers the highest slider resolution while the ±12 dB mode provides a wider range of control over system audio (ME 15B & 30B only)

Filters

- Built-in infrasonic and ultrasonic filters eliminate low-end rumble and RFI

Inputs and Outputs

- Active balanced XLR input and outputs are provided for working with balanced only signals
- Active balanced 1/4˝ TRS inputs and outputs are also provided for working with balanced or unbalanced signals

ME 60 Adds

- The same Interpolating Constant-Q design as the GE 30 and GE 60
- Sweepable High and Low Cut band limiting Filters
- Unbalanced RCA inputs and outputs

System Performance

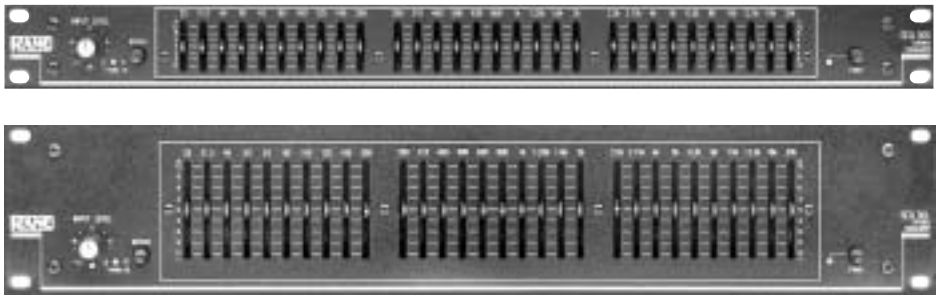
Note: 0 dBu = 0.775 Vrms	ME 15B / ME 30B	ME 60
Frequency Response (+0/-3dB)	20 Hz -100 kHz	20 Hz -40 kH
THD+Noise (+4 dBu, 20-20 kHz)	0.015 %	0.008 %
IM Distortion (SMPTE) [60 Hz/7 kHz, 4:1, +4 dBu]	0.01 %	0.005 %
S/N Ratio (re +20 dBu/+4 dBu) [Sliders centered, unity gain, bal.]	107/91 dB	112 / 96 dB
Channel Separation (1kHz)	85 dB (ME 15B)	75 dB
Common Mode Rejection (1kHz)	40dB	46dB

		ME 15B / ME 30B	ME 60
Inputs XLR & 1/4˝ TRS	**Impedance** 20k Ω bal./10k Ω unbal.	**Max Input Level** 20 dBu/20 dBu	**Max Input Level** 21 dBu
Outputs XLR & 1/4˝ TRS XLR & 1/4˝ TRS	**Impedance** 200 Ω bal. 100 Ω unbal.	**Max Output Level** +19 dBu (600Ω) +13 dBu (600Ω)	**Max Output Level** +19 dBu (600Ω) +13 dBu (600Ω

Equalizer Section	
Bands	
ME 15B	Dual (15) 2/3-octave ISO spacing
ME 30B	(30) 1/3-octave ISO spacing
ME 60	Dual (30) 1/3-octave ISO spacing
Frequency Range	From 25 Hz to 16 kHz
Accuracy	3% from center frequency
Type	Constant Q
Boost/Cut Range	±12 or ±6dB switch selectable (ME 60 ±6dB only)
Fader Travel	20mm sliders, grounded center detent
Output Section	
Overall Gain Range	Off to +0dB (Unbalanced) Off to +6dB (Balanced)
Bypass Switch	Hard-wire fail-safe design Auto-Bypass with power loss
Filters	
Infrasonic	20 Hz, 18 dB/octave, Butterworth
LED Metering	
Overload	4 dB Below clipping
XLR & 1/4˝ TRS Inputs & Outputs	
Inputs	RF Filtered, active balanced / unbalanced
Outputs	Active balanced / unbalanced
ME 60 ADDS	
Sweepable Filters	
Low Cut Filter	10 Hz-250 Hz, 12 dB/octave
High Cut Filter	3 kHz-40 kHz, 12 dB/octave
XLR, 1/4˝ TRS & RCA Inputs & Outputs	
Inputs	RF Filtered, active balanced / unbalanced
Outputs	Active balanced / unbalanced

SEQ 30S • SEQ 30L

Stereo Graphic Equalizers



The SEQ 30S and SEQ 30L are professional 30-band stereo graphic equalizers that use a single set of sliders to control the left and right channels simultaneously offering the ultimate in ease of use and stereo accuracy. Both units have exactly the same features and specs except that the SEQ 30S is a single space rack unit with 20 mm short-throw sliders, and the SEQ 30L is a dual space rack unit and employs 45mm long-throw sliders. Basic features include ±12dB boost/cut, input level controls, signal present and overload metering as well as XLR and 1/4˝ inputs and outputs.

30-band 1/3 octave Stereo Equalizers

- SEQ 30S uses short space-saving 20 mm sliders while the SEQ 30L uses long and more precise 45 mm sliders — all other features are identical
- A single set of faders controls the left and right channels simultaneously, maintaining true stereo tracking
- Fills the void for true stereo applications while delivering precise equalization down to a fraction of a dB
- A ±12dB input level control is provided along with a bypass switch, enabling easy gain comparisons between equalized and unequalized signal

Inputs and Outputs

- Balanced XLR inputs and outputs are provided as well as balanced 1/4˝ TRS inputs and unbalanced 1/4˝ TS outputs
- The XLR and 1/4˝ TS outputs can be used simultaneously to drive two sources such as an amplifier and a recorder

Additional Features

- Both units use low noise toroid output transformers that provides a wide 115 dB dynamic range
- +4 dBu and overload indicators detect both input and post-EQ levels
- Internal power supply

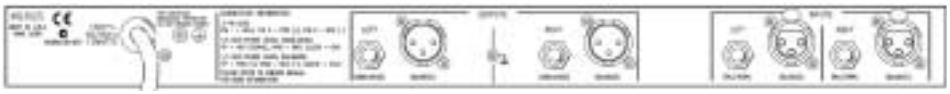
System Performance

Frequency Response (+0/-3dB)	20 Hz -20 kHz
THD+Noise (+4 dBu, 20-20 kHz)	0.009 %
IM Distortion (SMPTE)	0.005 % (60 Hz/7 kHz, 4:1, +4 dBu)
Signal-to-Noise Ratio (re +4 dBu) (Sliders centered, unity gain, balanced)	96dB
Channel Separation (1kHz)	80dB
Common Mode Rejection (1kHz)	40dB

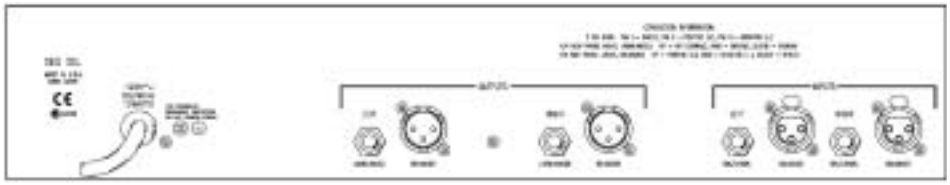
Note: 0 dBu = 0.775 Vrms

Inputs	**Impedance**	**Maximum Input Level**
XLR & 1/4˝ TRS	20k Ω bal.	21 dBu
XLR & 1/4˝ TRS	10k Ω unbal.	20 dBu
Outputs	**Impedance**	**Maximum Output Level**
XLR	100 Ω bal.	+20 dBu (2kΩ)
1/4˝ TRS	100 Ω unbal.	+20 dBu (2kΩ)

Equalizer Section	
Bands	Dual (30) 1/3-octave ISO spacing
Frequency Range	From 25 Hz to 20 kHz
Accuracy	3% from center frequency
Type	Constant Q
Boost/Cut Range	±12
SEQ 30S Fader Travel	20mm sliders
SEQ 30L Fader Travel	45mm sliders
Output Section	
Overall Gain Range	±12dB (sliders centered)
Bypass Switch	Hard-wire fail-safe design Auto-Bypass with power loss
LED Metering	
Signal Present	+4 dBu
Overload	Clipping indicator
XLR & 1/4˝ TRS Inputs and Outputs	
Inputs	RF Filtered, active balanced / unbalanced
Outputs	Active balanced / unbalanced



SEQ 30S rear illustration



SEQ 30L rear illustration

AC 22 • AC 22B

Active Crossovers



The Rane AC 22 and AC 22B are both active crossovers configurable for stereo 2-way or mono 3-way operation using state-variable 4th-order Linkwitz-Riley filter alignments designed to minimize phase difficulties in the critical crossover regions. The model AC 22 utilizes 1/4˝ TRS connectors with balanced/unbalanced inputs and unbalanced outputs while the AC 22B uses XLR connectors with active balanced inputs and outputs. They both feature, variable input and output level controls, band mute switches as well as user-adjustable time delay circuits to ensure proper mechanical phase alignment of adjacent drivers.

They Both feature

- Two independent active crossover channels configurable as stereo 2-way or mono 3-way
- State-variable 4th-order Linkwitz-Riley filter alignments with 24 dB per octave slopes are employed to minimize phase difficulties in the critical crossover region
- Crossover frequencies are selected using a continuously variable control with 41 detents that provide mechanical reference of crossover settings

Automatic 2-Way/3-Way Configuration

- Both units are smart enough to know whether you want to run a 2-way stereo, or a single channel 3-way system by the way the plugs are inserted and how your system is to be configured — plugging a signal into the channel 1 input and nothing into the channel 2 input tells the unit that you are running a single channel system in mono 3-way mode, and therefore sets the unit up to be a single channel device

Time Delay (Correction) Circuits

- To ensure the mechanical and electrical phase alignment of adjacent drivers will be acoustically correct, adjustable time delay circuits are provided on the low (and mid when used in 3-way mode) outputs of each channel to compensate for any physical misalignment of the drivers
- The low delay circuit can be internally "transplanted" to the high output when necessary

Stereo 2-Way mode

- A mono sub mode switch on the rear panel, allows you to disconnect the channel 2 low output jack and sum it with the channel 1 low output jack

Band Limiting Filters

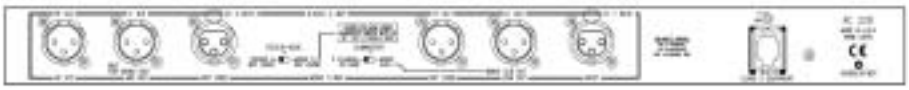
- Built-in 18 dB/octave low cut filters at 20Hz virtually eliminates infrasonic rumble while provided overall system protection

Inputs and Outputs

- Both units features two inputs, to allow stereo operation and four outputs configured as dual low and high outputs in 2-way stereo mode or low, mid and high in 3-way mode
- The model AC 22 utilizes 1/4˝ TRS connectors with balanced/unbalanced inputs and unbalanced outputs
- The AC 22B uses XLR connectors with active balanced inputs and outputs

Additional Features

- An internal CD horn equalization modification is possible allowing a constant directivity horn to cover the same area as a long throw horn. This modification should be made by an experienced technician



AC 22		
Inputs	**Impedance**	**Max Input Level**
1/4˝ TRS	20k Ω bal.	21 dBu
Outputs	**Impedance**	**Max Output Level**
1/4˝ TS	100 Ω unbal.	+20 dBu (≥ 600Ω)

AC 22B		
Inputs	**Impedance**	**Max Input Level**
XLR	20k Ω bal.	21 dBu
Outputs	**Impedance**	**Max Output Level**
1/4˝ TS	200 Ω bal.	+20 dBu (≥ 600Ω)

System Performance

Frequency Response	20 Hz -40 kHz +0/-3 dB
THD+Noise	0.02 % (+4 dBu, 20-20 kHz)
IM Distortion (SMPTE)	0.005 % (60 Hz/7 kHz, 4:1, +4 dBu)
Signal-to-Noise Ratio	92 dB (re +4 dBu, 20 kHz noise bandwidth)
Channel Separation	75 dB (1 kHz)
Common Mode Rejection	46dB (1 kHz)

Note: 0 dBu = 0.775 Vrms

Crossover	
Alignment	Linkwitz-Riley Proprietary 4th-order state-variable
Slopes	24 dB/octave
Range	70Hz - 3.6 kHz Low to High 41-detent continuously variable pot
Time Delay Adjust Range	Off to +2ms Low & mid outputs
Band Muting Switches	Low & mid outputs
Band Limiting Filters	
Infrasonic Filter	20 Hz, 18 dB/oct., Butterworth
Ultrasonic (AC22)	40 kHz, 6 dB/oct., Bessel
Gain Controls	
Input Gain Range	Off to +6 dB
Output Gain Range	Off to +6 dB (AC 22) Off to +12 dB (AC 22B)
AC 22 Inputs & Outputs	
1/4˝ TRS inputs	RF Filtered, Active Bal/Unbal
1/4˝ TS outputs	Active Unbalanced
AC 22B Inputs & Outputs	
XLR	RF Filtered, Active Balanced
XLR	Active Balanced

AC 23 • AC 23B

Active Crossovers



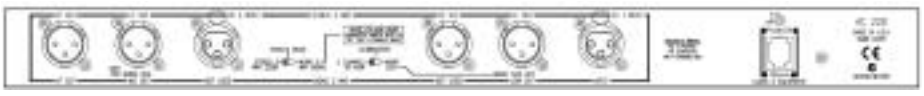
The Rane AC 23 and AC 23B active crossovers with identical features and specs as the AC 22 and AC 22B but with the ability to be configured for stereo 2- or 3-way, or mono 4- or 5-way operation. The AC23, like the AC 22 is available with 1/4˝ TRS balanced/unbalanced inputs and 1/4˝ TS unbalanced outputs while the AC 23B is provided with XLR connectors with active balanced inputs and outputs.

System Performance	
Frequency Response	20 Hz -40 kHz +0/-3 dB
THD+Noise	0.02 % (+4 dBu, 20-20 kHz)
IM Distortion (SMPTE)	0.002 % (60 Hz/7 kHz, 4:1, +4 dBu)
Signal-to-Noise Ratio	92 dB (re +4 dBu, 20 kHz noise bandwidth)
Channel Separation	75 dB (1 kHz)
Common Mode Rejection	46dB (1 kHz)

Note: 0 dBu = 0.775 Vrms

AC 23		
Inputs	**Impedance**	**Max Input Level**
1/4˝ TRS	20k Ω bal.	21 dBu
Outputs	**Impedance**	**Max Output Level**
1/4˝ TS	100 Ω unbal.	+20 dBu (≥ 600Ω)
AC 23B		
Inputs	**Impedance**	**Max Input Level**
XLR	20k Ω bal.	21 dBu
Outputs	**Impedance**	**Max Output Level**
1/4˝ TS	200 Ω bal.	+20 dBu (≥ 600Ω)

Crossover	
Alignment	Linkwitz-Riley Proprietary 4th-order state-variable
Slopes	24 dB/octave
Ranges	41-detent continuously variable pot
Range 3-Way	70-1 kHz Low to Mid
Range 3-Way	190-7 kHz Mid to High
Range 2-Way	190-7 kHz Low to High
Time Delay Adjust Range	Off to +2ms Low & Mid Outputs
Band Muting Switches	Low & Mid Outputs
Band Limiting Filters	
Infrasonic Filter	20 Hz, 18 dB/oct., Butterworth
Ultrasonic	40 kHz, 18 dB/Oct, Bessel
Gain Controls	
Input Gain Range	Off to +6 dB
Output Gain Range	Off to +6 dB (AC 22) Off to +12 dB (AC 22B)
AC 23 Inputs and Outputs	
1/4˝ TRS inputs	RF Filtered, Active Bal/Unbal
1/4˝ TS Outputs	Floating Active Unbalanced
AC 23B Inputs and Outputs	
XLR	RF Filtered, Active Balanced
XLR	Active Balanced



Linkwitz-Riley Alignment

Simply put, a Linkwitz-Riley alignment is two cascaded 2nd-order Butterworth filters exhibiting identical phase characteristics on their Low pass and High pass Outputs. This guarantees in-phase outputs at all frequencies, mandatory for proper acoustic summing of common signals from adjacent drivers in the crossover region. An added benefit of this topology is steep 24 dB per octave rolloff slopes. A slope of this magnitude guarantees drivers designed to produce a specific range of frequencies, and no more, will not be driven past their limits, thereby minimizing distortion and driver fatigue.

Choosing the Right Configuration:

Mono, Two-Channel Mono or Stereo?

Very few systems indeed will utilize a two channel crossover for the purpose of true stereo imaging. Discrete stereo channels which are run from the mixing board are usually used for panning effects and/or for separate equalization of left and right speaker stacks. Different sides of the room often require significantly different equalization due to varying room acoustics, dimensions, positioning of speaker stacks near walls, curtains and the like.

Even though you may not plan to use stereo equalization or panning effects, it is recommended that your system utilize discrete crossover channels for each stack of speakers to ensure flexibility and control for consistent, optimum sound quality. For example, if you plan to run a multi-stack system mono three-way, use the AC 23 rather than the AC 22 for separate control over each set of speakers—especially since phase alignment may differ with each stack requiring separate time delay adjustments. Even with only a single system equalizer, the AC 23 can deliver the extra independent control which can make a difference in sound throughout the listening area. If all drivers are built into a single cabinet, or you are running bi-amped monitors, then the AC 22 is the one for you.

RANE

SAC 22 • SAC 23

Stereo 2-Way and Stereo 3-Way Crossovers



The SAC 22 and SAC 23 employ the same state-variable 4th-order Linkwitz-Riley filters with steep 24 dB per octave rolloff slopes, as the AC22 and AC23 ensuring identical phase characteristics on its low pass and high pass outputs. The SAC 22 utilizes a single 31-position precision DC control voltage potentiometer to simultaneously select the low and high frequency points while the SAC 23 uses two 31-position controls divided between low/mid and mid/high crossover regions. This crossover circuit design provides a smart and easy way to biamp or triamp your audio system while at the same time ensuring consistent accuracy from channel-to-channel and unit-to-unit that provides a distinct advantage over continuously variable designs using ganged potentiometers which can yield large variations in channel-to-channel matching. Both units also feature active balanced XLR inputs and outputs.

They Both Feature

- 31-position detented stereo crossover controls provide the simplest way to biamp (SAC 22) or triamp (SAC 23) your system
- Same 4th order Linkwitz-Riley crossover design as the AC22/23
- An input level control allows decreasing the overall sensitivity of the entire sound system, including the mono subwoofer if one is used
- Separate low, high and mono subwoofer output level controls allow compensation for sensitivity variations in amplifiers and drivers

Inputs and Outputs

- Separate balanced XLR connectors are provided for the left and right inputs as well as the stereo low and high outputs. Additional left and right XLRs are provided for the mid outputs of the SAC 23

Subwoofer Output

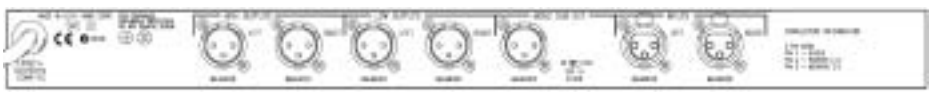
- The mono subwoofer output, with a switchable 100 Hz Low Pass Filter, provides a separate mono sum of the left and right low outputs
- The subwoofer output may be used along with the left and right low outputs

System Performance

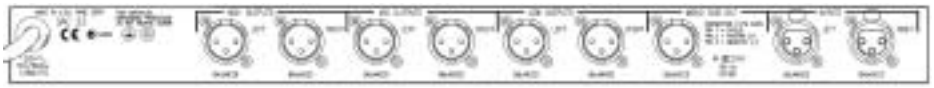
Frequency Response	15 Hz-40 kHz +0/-3 dB
THD+Noise	0.1 % typical +4 dBu, pass band
IM Distortion (SMPTE)	0.02 % (60 Hz/7 kHz, 4:1, +4 dBu)
S/N Ratio	84 dBr minimum (Max. gain re +4 dBu, 20 kHz BW)

Note: 0 dBu = 0.775 Vrms

Inputs	**Impedance**	**Max Input Level**
XLR	20k Ω bal.	+20 dBu
Outputs	**Impedance**	**Max Output Level**
XLR	100 Ω unbal.	+20 dBu (2k Ω)



SAC 22 rear illustration



SAC 23 rear illustration

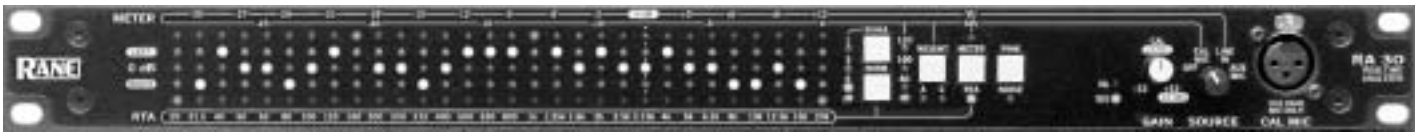
Crossover	
Alignment	Linkwitz-Riley Proprietary 4th-order state-variable
Slope	24 dB/octave
Range Control	Stereo 31-detent continuously variable DC
SAC 22 Range	100 Hz-3.2 kHz
SAC 23 Low/Mid Range	80 Hz-1 kHz
SAC 23 Mid/High Range	350 Hz-7 kHz
Tracking	2 % Channel-to-channel tracking error
Frequency Selector Accuracy	10 % max.
Gain Controls	
Input Gain Range	Off to +6 dB
Output Gain Range	Off to 0 dB
Mono Sub	
Low Pass Filter	100 Hz, 12 dB/octave 3rd-order Butterworth
Gain Trim	Off to +0 dB
Band Limiting Filters	
Infrasonic Filter	15 Hz, 18 dB/Octave Butterworth
Ultrasonic	40 kHz, 18 dB/Oct, Bessel
Indicators	
Green Signal Present LED	+4 dBu
Red OverLoad LED	3 dB Before Clipping
Inputs & Outputs	
XLR Inputs	RF Filtered, Active Balanced
XLR Outputs	Active Balanced

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RA30

30-Band Multi-Function Realtime Analyzer



Real Time Analyzer Mode



Sound Pressure Level Mode



Stereo VU Mode

The Rane RA 30 Realtime Analyzer is a single rack space unit providing 30 bands of realtime frequency/amplitude information in a 5-LED per band format. The RA 30 serves three functions: Realtime Analyzer with Pink Noise generator, SPL Meter and Stereo VU Meter with Peak Hold.

RTA (Real Time Analyzer) Mode

- RTA Mode allows you to visually judge the character of the signal being generated by the built-in Pink Noise generator (or any sound) using the included measurement microphone. You are then able to adjust an external 1/3-octave equalizer for the optimum response of a sound system based on the readings presented by the LED display.
- RTA mode divides broadband signals applied to either the mic or line inputs into 1/3-octave increments using IEC 61260 Class 2 filters. The outputs of the filters are displayed on the front panel to visually indicate the amplitude of each band, in user-selectable 1, 3 or 6 dB per LED scales.
- A Normalize button sets the 0 dB reference to the maximum band level.

SPL (Sound Pressure Level) Meter Mode

- Accurately displays SPL readings from a minimum of 51 dBA to a maximum of 120 dBA
- In SPL Meter Mode, the display is read horizontally and calibrated to a 1 dB resolution using the included calibration mic which auto ranges for maximized operation.
- A- or C-weighting filters are user selectable.

Stereo VU Meter Mode

- Stereo VU mode is automatically enabled when the Line input source is selected
- Levels are displayed as two horizontal rows of yellow LEDs with a 1.5dB resolution: The top row corresponds to the Left Input, and the bottom row corresponds to the Right Input. Levels above +3 dB on the VU scale switch to the red LED rows.
- The VU can be calibrated to accommodate +4 dBu or -10 dBV signals.
- An instantaneous Peak Hold function with a 2-second hold time is also provided using a single LED display for each channel
- Peak Hold is enabled or disabled by pressing the NORM button.
- If the held peak value is greater than +12 dB on the VU scale, the +12 dB LEDs stay illuminated. In this case, the Input gain can be decreased until the peak value is not "stuck" at +12 dB. For a calibrated peak measurement, the input gain can be set to minimum, which sets the 0 dB mark on the VU scale to +16 dBu.

MIC 2 Condenser Microphone

A professional quality back-electret condenser microphone with an omnidirectional pick-up pattern. This microphone is supplied with each RA 30, and connects to the front panel Mic Input jack only. The tapered aluminum mic housing fits most mic stand clips (one is included), and comes complete with a 25 foot (7.6 meter) cable to facilitate distant placement of the mic from the analyzer. These factory tested mics are flat to within 1 dB from 20 Hz to 16 kHz. An Aux Mic Input is provided on the rear panel, facilitating the use of other microphones.



Mic 2 Condenser Microphone

Rear Panel

- Pink Noise Output jack with a recessed Pink Noise Level adjustment
- Aux Mic Input (XLR)
- Stereo Line Inputs (1⁄4″ balanced, can operate in mono by using the left input only).

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SAMSON

S•COM / S•COM PLUS

Stereo Compressor / Limiters



The S•com is a dual-channel, single rack space stereo compressor designed to control signal levels in a variety of applications. It's logical design and layout make it simple to understand and easy to use. Sonic integrity offers clean, distortion free audio, and a predictable, usable range of control from undetectable to clearly identifiable. Each channel combines an Expander/Gate, a Compressor/Limiter and a fully adjustable Enhancer. This unique circuit reduces noise, increases loudness and recovers high frequencies that might otherwise be compromised when audio is heavily compressed. The Spectra switch adds a mid-band boost to the detector circuit to help smooth out harsh vocals.
The S•com PLUS has basically the same features, but also offers a variable de-esser to improve audio quality by removing "S" from vocals and sibilance from overly bright sources such as cymbals. Both include a three-year extended warranty.

FEATURES

Expander/Gate

- The Expander/Gate features a variable Threshold control and switchable (fast/slow) Release time.
- Twin (open and closed) LEDs provide visual confirmation of the gate's operation

Compressor / Limiter

- Compressor/Limiter controls include variable threshold, ratio, attack, release and output level.
- Switchable AEG (Auto Envelope Generator) function continuously analyzes the audio input, and automatically adjusts the attack and release times based on the changing level.
- EFR (Enhanced Frequency Recovery) circuit helps restore high frequency content that can be lost when high gain reduction is applied.
- The SKD (Smart Knee Detector) automatically chooses between hard or soft knee compression curve—based on the program material. A hard knee curve activates the compression circuit only when the audio signal exceeds the threshold vs. a soft knee curve which gradually increases the compression ratio until the threshold is met.

Metering

- 12-segment LED meter is provided for Gain Reduction levels
- Another 12-segment LED meter can be switched to display input or output levels

Inputs and Outputs



- Servo balanced inputs and outputs on XLR and 1/4″ connectors with switchable +4 and -10 operating levels.
- 1/4″ key inputs and outputs

Additional Features

- A Key input switch, allows you to use an external audio source to control processing. For example, the signal from a kick drum trigger the amount of compression applied to a bass or you can use an EQ to specify which frequencies should be used to control dynamics.
- A link switch allows true stereo operation with channel one controlling the processing of both channels.

S•COM Only

- A Key Listen feature allows you to audition the Key input signal.
- An Enhancer level control allows you to adjust the amount of Enhanced Frequency Recovery being applied to the signal
- In lieu of a dedicated de-esser, a switchable Spectra circuit allows you to add a mid-band boost to the detector circuit to tame sibilant vocals

S•COM PLUS Only

- A switchable de-esser circuit with variable level control, helps you tame harsh and sibilant vocals. A five segment LED meter shows you the amount of high frequency gain reduction being applied to the signal
- A dedicated Peak Limiter ensures that output levels do not exceed the desired threshold level. Ideal for speaker protection in your PA rig and for recording and mixdown, especially to digital devices. An LED lets you know when the threshold level has been exceeded and the Limiter circuit is activated.

S•COM 4 / S•GATE 4

Four Channel Dynamics Processors



Compact and versatile, the single-rack space S•Com 4 and S•Gate 4 each four channels of high quality dynamics processing at an affordable price. The S•Com 4 features an Expander/Gate and Compressor/Limiter on each channel while the S•Gate 4 features an Expander/Gate/Ducker on each channel. Both units feature balanced XLR and 1/4˝ inputs and outputs and have a link function that allows you to configure each unit to work as dual stereo processors.

S•Com 4

- 4 Channel Compressor/Limiter, Expander/Gate with Enhancer
- Individual Threshold and Ratio controls on each channel
- SKD (Smart Knee Detector) circuit automatically switches from soft to hard knee based on the level of input signal applied
- EFR (Enhanced Frequency Recovery) circuit helps restore high frequency content that can be lost when high gain reduction is applied.
- Expander/Gate circuit with variable Trigger control and switchable Fast/Slow Release
- 5-Segment LED meters provided for input/output levels and Gain Reduction on each channel

S•Gate 4

- Four channel Expander/Gate/Ducker
- Each of the gate channels has variable Threshold, Attack, Release, and Range controls
- Variable High and Low Pass Filters, on each channel, allow frequency selective triggering
- Ducker feature allows you to automatically attenuate (as opposed to gate) the signal of a channel or stereo pair.
- Ducking can also be triggered by an external audio source using the 1/4˝ Key Input. A Key Listen switch lets you monitor the key input signal.
- 8-segment LED meter lets you monitor the amount of Gain Reduction on each channel

Samson's S•Class processors represent a new perspective on professional-level audio processing. Elegant and powerful, they combine proven circuitry, innovative features and ease-of-use in beautifully crafted cabinets.

Both Feature

- Linkable in two stereo pairs. This allows the settings on channels 1 and 3 to control the processing of channels 2 and 4.
- Advanced circuit design utilizing low noise operational amps and high quality VCA's
- Servo balanced inputs/outputs on XLR and 1/4˝ connectors with switchable +4/ -10 operating levels.
- Three-year extended warranty

S•PATCH PLUS

48-Point Patch Bay



Fully balanced and extremely durable, the S•Patch PLUS is a 48-point patch bay that makes all your connections easy. Offers normal, half-normal and Through Mode operation via the 3-way front panel mode switch. Balanced 1/4˝ TRS connectors. 19˝ rackmountable chassis.

SAMSON

S•VOX

Stereo Voice Channel/Input Mixer



The S•Vox is a sophisticated stereo voice channel including a four input mixer with digital outputs. An ideal front-end device for hard disk recorders, the S•Vox features two high quality microphone preamps with 3-band EQ, an optical compressor and convenient insert points. Also available on the mic channels are line inputs specially matched for guitar and bass guitar outputs. The S•Vox also features two independent line inputs, which are ideal for keyboards or for a sub-mix input. The extensive master section offers control room monitoring, as well as analog and S/PDIF digital outputs.

FEATURES

Channels 1 and 2

- Channels 1 & 2 feature high-quality microphone preamps with switchable 48v phantom power and Phase Reversal switch is also provided for advanced two channel miking techniques.
- A 3-band EQ with sweepable mids and an In/Out switch that provides clean and silent toggling between equalized and non-equalized modes.
- Channels 1 and 2 also feature an optical compressor with variable threshold which provides wide-ranging high quality control of signal dynamics.
- EFR (Enhanced Frequency Recovery) circuit helps restore high frequency content lost when high gain reduction is applied.
- Convenient insert points are also provided
- Two 5-segment LED meters show accurate information of input or output signals so all signal changes can be monitored.

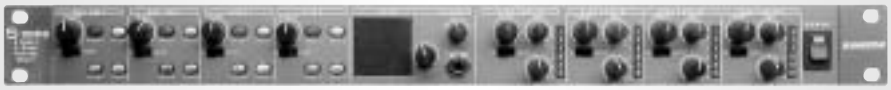
Channels 3 & 4

- Channels 3 and four provide line inputs, with volume and pan controls, that can be used for keyboards and other processors or they can be used to return a stereo sub mix.

Inputs and Outputs

- XLR mic, 1/4″ line combo connector inputs are provided on the front panel for channels one and two
- S/PDIF inputs and outputs allow you to connect the S•Vox digitally to your DAW
- 1/4″ TRS balanced mixer outputs are also provided on the rear panel for analog recording
- Analog control room/headphone outputs can also be used to monitor the signal before or after it enters the DAW

S•ZONE *Stereo Multi-zone Mixer*

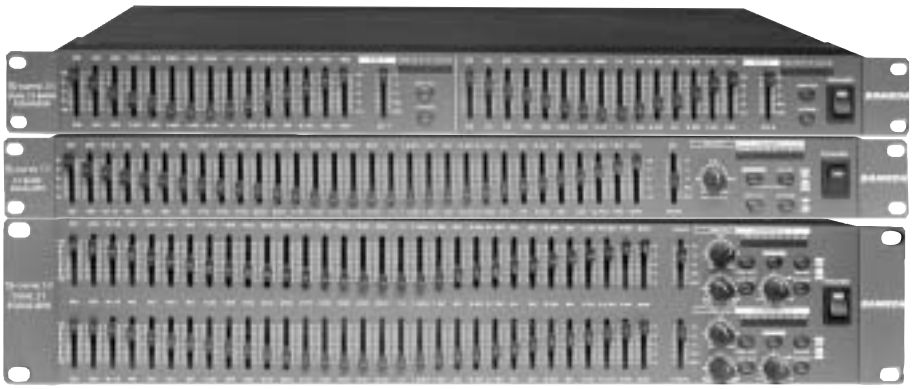


The perfect solution for multi-zone audio installations, the S•Zone is a 4-channel / 4-zone stereo mixer featuring two mic/stereo line inputs with ducking, two stereo CD/line inputs, on-board speaker and headphone monitoring. It also offers Euroblock output connectors and remote volume control. Great for controlling and monitoring audio in restaurants, bars, clubs, houses of worship and office or boardroom situations.

- 4 channel stereo zone splitter/mixer
- 4 balanced stereo zone outputs
- Unique front panel monitor section with selectable zone monitor switch, headphone output and built-in super-speaker
- 2 Stereo CD/Line inputs with backlit zone assignment switches
- 2 Mic/Stereo Line inputs with backlit zone assignment switches
- Phantom power available on mic inputs
- Independent, adjustable Zone ducking on microphone inputs 1 and 2
- Independent 2-band equalizer on each zone output
- All stereo inputs and outputs switchable to mono
- 6 segment LED output VU meters for each zone
- Euroblock screw-terminal input and output connectors
- Remote VCA volume control capability for each zone output
- Internal voltage-selectable power supply
- Standard EIA single rack space chassis

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S•CURVE 215 / S•CURVE 131 / S•CURVE 231



Designed and built for professionals, S•Curve EQs feature easy to read LED displays and the extra features professionals need. S•Curve EQs are also exceptionally quiet, in fact up to 15dB lower than comparable EQs. Three models available, the S•Curve 215 is a great, multi-purpose EQ packed with features and easy-to-read lighted display. The S•Curve 131 is for pros looking for a full 31-band EQ, while the S•Curve 231i s a totally comprehensive, feature-rich dual 31-band EQ for the most demanding applications. They all feature Constant Q circuitry, balanced XLR and 1/4˝ TRS inputs and outputs, LED faders and LED bar VU meters, boost/cut and cut only modes, and a bypass switch

S•Curve 215
Dual 15 Band Graphic EQ

- 2/3 octave 15 band stereo graphic equalizer
- High pass filter
- Single rack space chassis with aluminum extrusion face plate

S•Curve 131
Dual 15 Band Graphic EQ

- 1/3 octave 31 band graphic equalizer
- Variable Low cut filter
- Single rack space chassis with aluminum extrusion face plate

S•Curve 231
Dual 31 Band Graphic EQ

- 1/3 octave 31 band stereo graphic equalizer
- Switchable ±6dB or ±12dB filters
- Variable Low and High cut filters
- Double rack space chassis with aluminum extrusion face plate
- Subwoofer output

S•3-WAY *Stereo/Mono Crossover*



The S•3-way is a versatile 2-way, 3-way and 4-way crossover with a difference. First, it's a perfect 2-way or 3-way stereo crossover. But its mode is also switchable for use as a 4-way mono crossover. Both feature 2ms of delay for time alignment, Mute and Phases switches for each output and XLR balanced outputs.

- Full-featured, Stereo 3-way, Mono 4-way electronic crossover
- The input gain features ±12dB range with LED metering and Peak indicators.
- The first crossover point is sweepable in three ranges from 35Hz to 800Hz, from 350Hz to 8kHz with the 10X-Multiplier switch engaged, or from 16Hz to 400Hz in 4 way low-mode.
- The second crossover point is sweepable from 175Hz to 4kHz or from 350Hz to 8kHz depending on the setting of the mode switch.
- A Delay section with up to 2ms of delay to time-align low frequency outputs for improved phase response of any PA system
- Low, Mid and High Frequency outputs with ±6dB of gain control.
- Each output incorporates a Mute switch for monitoring the individual frequency bands and a phase switch to invert the polarity of the output.
- High-quality 41-position detent pots and backlit switches
- The global section features a variable threshold Limiter, a CD function (for constant directivity horns) and High Pass Filter.

C•CLASS SERIES

C•Class is audio gear designed to work seamlessly with your digital setup— but it works outside your computer. Each C•Class component is optimized for today's digital studio. When combined, C•Class components become an integrated suite of studio tools and effects. The "C" stands for compact. C•Class products bring the feel of analog equipment back to what has become a digital world. C•Class System integration features include ins and outs that are optimized for each other, new interlocking C•Class bumper-legs and cohesive industrial design.

Five processors in the C•Class series, the C•Valve is a tube preamp that adds the warmth of analog tubes to digital recordings; the C•Com 16 is a compact, full-featured compressor for the desktop; the C•Com Opti adds the magic of optical compression to recordings: the C•Control provides a control room matrix previously available only on consoles costing thousands; and the C-Que 8 is the perfect compact headphone amp.



FEATURES

C•Valve:
C•Valve is a tube mic/instrument preamp that will give you pure signal and silky sound up-front, where it counts. It features variable gain and variable tube process saturation for adding harmonics that give warmth to vocals and instruments. Regardless of which mic you use, it will maintain detail and transparency. 48-volt phantom power, phase and peak limiter round out the control set. A large analog VU output meter and a 6-segment LED input meter makes it easy to adjust the gain. Use its insert points along with a C•Class compressor for a direct path to a gorgeous front end sound. The C•Valve provides a digital output as well.

C•COM 16:
A Stereo Compressor/Limiter with 16 presets, the C•Com 16 gives you the fastest way to achieve useable dynamic control and the cleanest path for your audio. There are fifteen presets for the most standard situations and a manual mode for more critical adjustments when required. The C•Com 16's controls include variable threshold, ratio, attack, decay, release and output level. An enhancer provides the added character that is lost on heavy compression. Key output/input for outboard triggering or external filtering of the key signal. A 6-segment LED gain reduction meter and a 6-segment LED input/output meter provides visual monitoring of the circuit's affect.

C•Control:
The C•Control is a control room matrix selector and a unique solution for studios with multiple monitors, headphone mixes and tape or digital sources. Provides easy, heretofore impossible to achieve, instant monitor mixes or dubs. Select from three pairs of speakers with dim and mute. Use the Talkback mic to give instructions to the musicians in the headphone mix. And push-to-talk for slates to tape or disk. Includes a headphone amp with level control for private listening. The C•Control provides three sets of stereo ins and outs with dubbing capability to three discrete outputs.

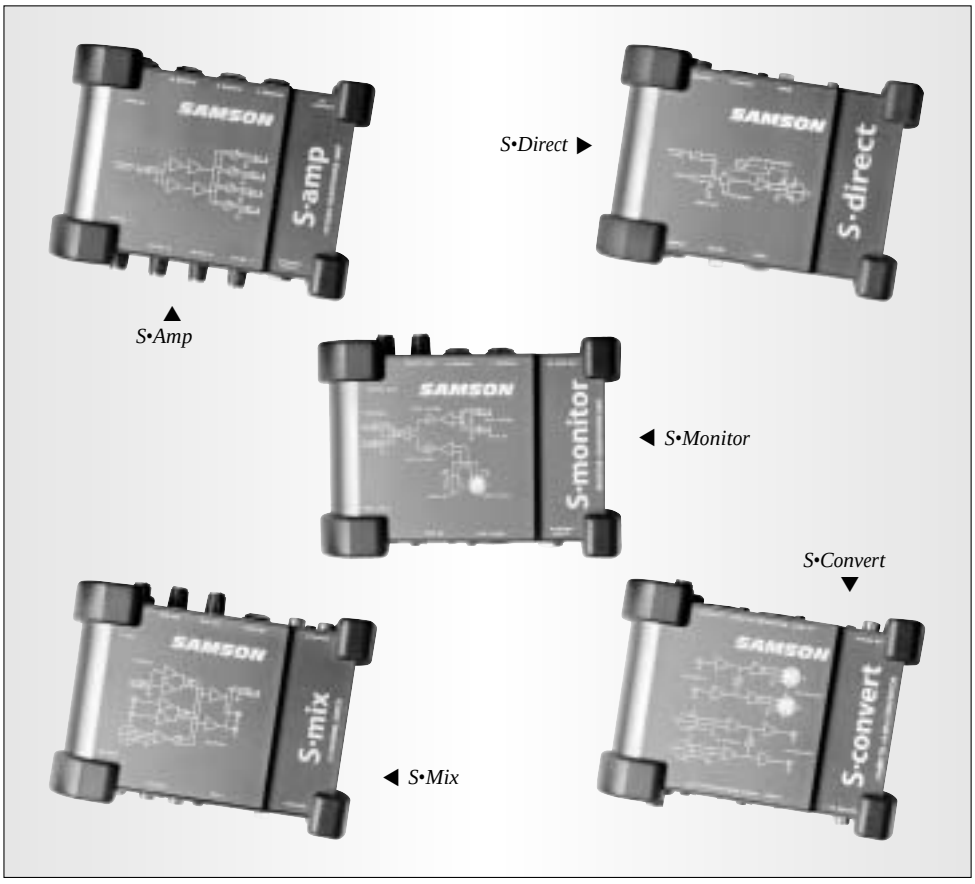
C•QUE 8:
Further enhancing your control in the studio is the C•Que 8, a 4-Channel Headphone Amplifier with eight outputs. Independent volume on each channel makes this a truly versatile headphone amplifier. A 6-segment LED indictor shows the output for each channel. There's a Main volume for overall level adjustment. An effective EQ shape circuit offers tone control of each channel and listeners can manage their own personalized mixes. Link outputs allow more than one C-Que 8 to be used when needed.

C•COM Opti: Optical compressors are sought after for the distinctive musicality they give to vocals, guitars and other instruments. A perfect complement to the computer's hygienically clean makeup, the C•Com Opti's adjustable threshold, ratio, attack, release and output controls are calibrated to generate a distinctive soft, rich character while providing magnificent control over dynamics. An enhancer provides the added character that is lost on heavy compression. Key output/input for outboard triggering or external filtering of the key signal. Large analog VU meter for output level or gain reduction.

B&H PHOTO - VIDEO - PRO AUDIO EQUIPMENT LEASING AVAILABLE

S•CLASS MINI PROCESSORS

The S•Class Mini processors offer advanced signal processing in incredibly durable, portable enclosures. Very affordable, the S•Class Minis are perfect for home, on the road, or as a quick fix to a house sound system.
Eight processors in the series, they include the S•Direct DI box, S•Monitor for personal monitoring on stage or in the studio, S•Mix miniature 5-channel mixer, S•Amp four channel headphone amplifier, S•Convert for interfacing and level matching consumer and professional audio equipment, S•Combine a 2 to 1 mic combiner, and S•Split a 1-in 3-out mic splitter. Last, the S•Phantom is a 2-channel 48v phantom power supply with two balanced XLR mic inputs and two balanced XLR mic outputs and an AC adapter included.



S•Combine

2 to 1 mic combiner, this tiny box has two balanced-XLR mic inputs, one balanced-XLR mic output and includes an AC adapter.

S•Monitor

Elegant solution for personal monitoring on stage or in the studio

- XLR microphone input
- XLR microphone Thru, mic output to send to main mixer
- 1/4″ stereo mix input
- Mic and mix volume control
- 2 headphone outputs
- 1/4″ mix Thru for daisy chaining additional units
- Includes 12v AC adapter

S•Direct

- Rugged and sturdy– ideal for stage and studio.
- Switchable input level handles instrument and speaker levels.
- 48v phantom power or 9v operation with auto battery shut off
- Switchable ground lift switch
- Gold-plated balanced XLR output
- 1/4″ link output for stereo using 2 boxes

S•Split

1-in 3-out mic splitter with one balanced XLR mic input and three balanced XLR mic outputs. Transformer isolated output for clean signal. AC adapter included.

S•Mix

- Miniature 5-channel mixer
- XLR balanced mic input with volume control
- Stereo 1/4-inch and RCA inputs with volume control
- RCA + 1/4-inch stereo outputs
- Includes 12v AC adapter

S•Amp

Four channel headphone amp ideal for home recording

- Four 1/4″ stereo headphone outputs
- Stereo 1/4″ input
- Individual volume control for each output
- Includes 12v AC adapter

S•Phantom

The S•Phantom is a 2-channel 48-volt phantom power supply with two balanced XLR mic inputs, two balanced XLR mic outputs and includes an AC adapter.

S•Convert

A “bump-box”, S•Convert is a problem solver for interfacing and level matching consumer and professional audio equipment

- +4 / -10 audio level converters
- +4 XLR/ -10 RCA inputs with level control
- +4 XLR/ -10 RCA outputs
- Includes 12v AC adapter

421M

AGC-Leveler with Mic/Line Input



The 421M is a sophisticated audio gain controller—it makes quiet sounds louder and loud sounds quieter, just like a skilled audio engineer. Set the desired, "target" output level and the 421M gently boosts signals that drop below your target, and smoothly pulls back those that rise above it. Operation is automatic, precise, and completely transparent - no pumping or breathing.

The 421M is designed for installed sound systems, recording studios, broadcast facilities or any audio application where clarity and intelligibility are important. Since everybody speaks at different levels and works at varying distances from the microphone, intelligibility can vary from person to person or moment to moment. The 421M puts everyone on the same level. It is equally well suited to processing program material (for stereo applications two 421M's may be linked). Program levels from soundtracks, CD jukeboxes, or broadcast audio go up and down unpredictably. The 421M gently and unobtrusively raises the low level audio and compresses the high level audio without side effects. It's flexible input configuration will handle just about any audio source, from studio microphones to telephone-based paging systems.

How It Works

The 421M's Automatic Gain Control (AGC) section incorporates a smooth-acting leveling amp working with a make-up gain stage coupled to the ratio control. Increasing the ratio makes your program denser, but output level stays constant no matter what happens at the input. This means you could talk two feet from a mic and have the same volume output that you had at 6″ (and vice versa).

To deal with program noise, the 421M has a full-featured downward expander section that effectively quiets the output when input signal is absent. A separate "brick wall" peak limiter provides speaker and overload prevention. Last but not least, switchable speech curve filters are incorporated to optimize the 421M for voice range performance.

The 421M is set apart by it's "smart" circuitry. Other AGC designs confuse noise and feedback with the program signal, boosting noise or cutting off soft-spoken phrases. Not on the 421M. Proprietary Auto Release Monitor circuit instantly distinguishes between "real" signals (music/speech), noise and feedback.

Metering system makes it easy to setup and operate. Parallel LED displays show input compared to output so it is obvious if the 421M is adding gain or subtracting. Because you can hear and see the net results of the leveling action there is no more guess work.

421M Specifications:

Connectors

Line Inputs: 1/4″ TRS jack and screw terminals, 20k Balanced Bridging

Mic Input: XLR, 10k Balanced Bridging

Outputs: XLR, screw terminals, 200 ohm source, Impedance, Differentially Balanced, +23 dBm maximum level TS phone, Unbalanced, +18 dBm maximum level

Bypass: Relay controlled, hard-wire bypass in power off and bypass conditions

Sidechain: TRS phone, unbalanced

Downward Expander

Ratio: 1:2

Threshold: -50VU (bypass) to -20VU (Auto)

Attack Time: 1 ms

Release Time: Program dependent, 0.3-3.5 seconds depending on amount and duration

AGC-Leveler

Ratio: 1:1 to 4:1

Auto Release Threshold: -70 dBu to -30 dBu

Attack Time: approximately 1 ms

Release Time: Program dependent, 500 ms - 5 sec. depending on amount and duration

Target Output

Range: +20 dB **Peak Limiter Ratio:** 10:1

Threshold: -15VU to +23VU (bypass)

Attack Time: 1 ms for 90% gain reduction

Release Time: 0.8 seconds

Mic Preamp

Gain Range: +15 dB to +45 dB

Impedance: 10k Ohms

Max Input Level: +8 dBu

THD+Noise: -85 dBu (gain control fully clockwise)

Phantom Power: 48-volt (±2)

Speech Curve

Type: Switch selected cutoff filters allow tailoring LF and/or HF response for speech applications

Frequencies: LF=125 Hz, 12 dB/octave; HF=6 kHz, 24 dB/octave

Input/Output Metering

Type: LED Bar Graph, 12 steps + clip

Range: (min to max) 66 dB

422

Stereo AGC-Leveler



Audio comes in all shapes and sizes. There's loud audio. There's quiet audio. There's pretty audio. There's ugly audio. There's music. There's speech. There are CD's mastered at drastically different levels. There are movie sound tracks where the effects are too loud and the dialog so soft you can't understand the words. Have you ever been on an airplane trying to watch the movie and found yourself repeatedly turning the volume up and down, up and down. Well, if you need to free yourself from unpredictable audio program levels then you need the 422 Stereo AGC/Leveler. With simple and intuitive controls, the 422 converts "all over the map" signal levels into smooth, intelligible, constant level audio. Unlike a compressor/limiter which only does half a job— it pushes down from the top, preventing overload and distortion in subsequent stages, but does nothing from the bottom (the low level signals that contribute so much to the intelligibility of speech and the enjoyment of music)—the 422 Stereo AGC-Leveler does it all. It makes the loud sounds quieter and the quiet sounds louder. And it does it with finesse. The 422 works without the side effects of compressors and limiters—no noise, pumping, and modulation.

FEATURES

Easy To Use, There Are Basically Only Four Controls

The 422 is a remarkably sophisticated volume controller that is amazingly easy to use:

– The first and most important is the target level control. This sets the volume where you want it. The 422's input over output parallel VU meters simultaneously show you the unmodified input signal on top and the result of your target level setting just below it.

– The detector control increases the sensitivity of the AGC. As you turn it counterclockwise the 422 gently reaches down for the lower volume audio and brings it up.

– Set the target level and detector, then use the ratio control to increase or decrease the amount of leveling. At high ratios the program density increase results in a more "present" or "up front" sound. At low ratios the 422 performs subtle, yet effective, automatic gain riding.

– Last, adjust the peak limit control to create an absolute ceiling level. This is very handy for protecting amps and speakers in discos where DJ's often succumb to a disease known as "volume creep" as the night wears on.

◆ Remote bypass port

Why Not Just Use A Compressor/Limiter?

When it comes to maintaining constant output levels, a compressor/ limiter can only do half the job, at best. Sure, when things get too loud the comp/limiter kicks in, but what about when things get too soft? A comp/limiter is a top down device –pushes down from the top, preventing overload and distortion in subsequent stages.But what about the bottom up part of the deal? What about the low level signals that contribute so much to the intelligibility of speech and the enjoyment of music?

The 422 brings the volume to where you want it and keeps it there. It can be used in virtually any type of sound system for processing just about any kind of audio. Insert the 422 at a convenient patch point where you have line level audio. There are no annoying -10, +4dBu level matching switches – just give the 422 a basic line (not mic level) signal and you're ready to go.

Input/Output	
Inputs:	1/4″ TRS and XLR Stereo, Balanced Bridging or Unbalanced
Outputs:	1/4″ TRS and XLR Stereo, Balanced or Unbalanced
Polarity Input:	tip is high, ring is low, sleeve is ground
Output:	tip is high, ring is low, sleeve is ground
Max. Input Level:	+24 dBu
Max. Output Level:	+22 dBu into 600 Ohms

Performance	
Frequency Response:	20 Hz - 20 kHz
THD+Noise:	05%
Dynamic Range:	>110 dB
Crosstalk:	-60 dB, +20 dBu in, 20Hz - 20kHz
AGC Detector Range:	-40 dBu to +24 dBu
Ratio:	1:1 to 5:1
Target Level Range:	30 dB
Limiter Threshold:	-15 dBu to +25 dBu
Limiter Ratio:	>15:1

528E

Voice Processor (Channel Strip)



The 528E is a complete, self-contained voice processor with six separate functions: mic preamp, de-essing, compression/limiting, downward expansion, parametric EQ, and voice symmetry alignment. All six processors may be used simultaneously. Although the 528E is called a "Voice Processor", it is perfectly suitable for any signal, vocal or not. Each function features a full complement of controls in an easy-to-use layout. Separate LED meters monitor mic gain and dynamics gain reduction functions thus facilitating quick and accurate adjustment of controls.

Works with any mic (or line input), enhancing vocal intelligibility, increases perceived loudness and "presence". Great for voices as well as instruments and effects. Ideal for broadcast mics, voice-overs and music recording, and sound reinforcement for PA/Paging systems.

Mic Preamp

Variable gain up to 60dB and 48v phantom power. Switchable 15dB pad prevents distortion when close miking. Mic and line inputs are balanced-transformerless XLR and equipped with filters to prevent RFI.

Voice Symmetry

Corrects for excessive positive or negative signal peaks of the human voice. A simple in/out switch controls voice symmetry.

De-esser

Program controlled high-cut filter, 12 dB per octave senses and regulates selectable high frequencies to reduce or eliminate annoying sibilance and "lip smacking". Controls are frequency and range (800 Hz to 8000 Hz).

Compressor and Expander

The compressor/limiter maintains uniform levels while the downward expander eliminates "pumping", "breathing", and noise build up. And because it's program controlled, the 528E's dynamic range processor responds quickly to transients, and gently to smaller level changes. Controls are expand and compress threshold (-30 to 0 dBu) and compression/limiter ratio (1:1 to 10:1).

3-Band EQ

3-band— 16 to 500 Hz (Low), 160 to 6300 Hz (Mid) 680 Hz to 22 kHz (High) parametric EQ has a variable bandwidth from .3 octave to 4 octaves, 15 dB boost/cut, and overlapping frequency ranges. "Leap frog" topology minimizes the number of amps in the signal path while ensuring that each band interacts with its neighbor in a desirable and musical fashion.

Metering

- Type: Multi-segment LED bar graph
- Output Level: -20 to +3 VU (0 VU = +4 dBu), VU calibrated, peak responding
- Gain Reduction: separate displays for de-esser, downward expander, compressor
- 0 to 20 dB per display.

628 *Digital Voice Processor*



A digital version of the 528E, the 628 rolls a premium quality mic preamp, 20-bit A/D and D/A converters, de-esser, expander/gate, compressor and parametric EQ into a single rack space unit. Perfect for any voice and many other instruments or sounds the 628 combines proven digital signal processing and an easy to use analog-like interface with the power of 8 finely- tuned factory and 119 user-programmable presets. An optional RC-1 remote controller allows selection of the first eleven presets, plus a bypass function, eliminating frantic dashes to the equipment rack.

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Mic Preamp

The 628's first stage is a proprietary transformerless mic preamp incorporating filters to destroy radio frequency interference. A switchable 15 dB pad prevents overloading by hot condenser mics. A front panel LED indicates when phantom power is on. If you're into high end tube preamps then you can select line level input from the 628's front panel and bypass the 628's preamp.

De-esser

Tuning the 628's de-esser to the offending frequency minimizes overly bright sibilance without resorting to brute force equalization to solve the problem. The Threshold control lets you precisely apply this frequency selective gain reduction. LED metering displays the degree of de-esser action.

Expander/Gate

Independent Ratio and Release controls allow tuning of the expander/gate to fit any studio noise or performer isolation requirements. An LED meter shows the amount of downward expansion or gating taking place. Most compressors use a “one size fits all” approach to compression; the design of the 628 reflects the realization that voice and mixed music require different design philosophies. While the control complement is simple (threshold, ratio and release), each has been highly optimized for voice work. The result is a compressor that can tightly control gain (from hard compression to gentle level control) with no side effects.

3-Band Parametric EQ

Three overlapping bands of digital parametric equalization round out the processing power of the 628. The EQ can notch out interference, boost low frequency energy, cut mid-range grunge and brighten a muddy, dull voice simultaneously. The 628's parametric approach to equalization allows cuts and boosts exactly where needed to make every voice sound outstanding. Three 7-segment LEDS display all parameter values and preset numbers. An output level meter continuously monitors output. Selectable digital sample rates of 48, 44.1 and 32kHz. Balanced and unbalanced analog output as well as AES/EBU and S/PDIF digital outputs are provided.

300 SERIES—Half Rack Audio Tools

302 *Dual Mic Preamp*



Designed for broadcast, sound reinforcement, and digital and analog recording. Offering substantial sonic improvements over stock mixers or console preamps, the 302 has solid stereo imaging, excellent transient response, very low noise, and almost undetectable distortion.

- Polarity reversal switches and 15dB pads on both channels
- 48 volt phantom power
- Gain variable from 20-60dB, works with virtually any professional microphone.

303 *Interface Amplifier (Bi-Directional)*



A stereo level matching amplifier that accepts balanced or unbalanced inputs and converts them to balanced or unbalanced outputs. Use the 303 to interface a consumer DAT, cassette recorder or computer sound card to a professional recording console or to other +4dBu nominal level equipment.

- Stereo XLR and RCA inputs and outputs
- External power supply ensures low noise

304 *Headphone Amplifier*



Stereo in, four output amp drives headphones of any impedance. Used in broadcast, installed sound, and recording applications. High voltage technology provides 6dB volume increase over most competing units. Compact half rack chassis permits 8 channels in one rack space.

- Individual volume controls for each output, master input volume
- Stereo or mono operation, balanced or unbalanced input
- Pass Thru jacks permit daisy-chaining of multiple units.

305 *1x4 Distribution Amplifier*



A one input/four output mono distribution amplifier. Use the 305 to route your most critical audio to four separate outputs. Sound contractors will find the 305 essential for feeding multiple power amps. Broadcast engineers can easily and inexpensively distribute satellite audio around their station.

- Euroblock connectors for quick installation
- Four-segment LED input level meter
- Precision circuitry yields less than .009% THD+Noise
- Individual trim pots for input and output volume attenuation

TC ELECTRONIC

GOLD CHANNEL

Dual Channel Digitally Enhanced Mic Preamp



The Gold Channel is a two channel digitally enhanced mic preamp/processor that uses premium 24-bit A-D converters, a high resolution 24-bit 96kHz signal path and a full range of powerful refinement tools to capture the highest quality audio to any analog or digital recorder. Features include an expander/gate, Softlimiter, compressor/limiter, EQ and other specialized tools, all accessible through an easy to use channel strip interface with high resolution metering. Standard Mic/Line inputs are provided as well as AES/EBU, Toslink, S/PDIF and ADAT digital I/Os. 200 User Preset memory locations allow you to customize settings and store them for instant recall – a real time saver when working with specific musicians over several sessions. In addition to delivering a pristine preamplification, the Gold Channel can also be inserted into a console or DAW where it can be used to process recorded material in 96kHz resolution.

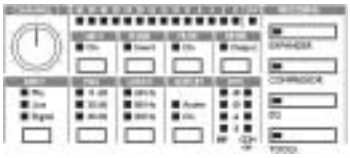
FEATURES

Auto Gain

- Plug in a mic, select Auto Gain and press Enter. Let the musician or vocalist perform and press Enter when finished. Auto Gain has now set the optimal gain level

Channel Overview

- Functions for each channel are on the front panel with a user friendly display:



  – Input Gain knob
  – Input Select – Mic, Line, Digital
  – Switchable +48V phantom, Phase Invert, Mute, Soft Clipper and Meter select
  – PAD – 0dB, 20dB, 40dB
  – Lo-Cut – 60Hz, 80Hz, 120Hz
- Separate bypass buttons are also provided for each processing block
- 14 segment LED meter is switchable between showing input and output right after the digital Master Output. Two 4 segment LED meters show gain reduction for the Expander and Compressor/Limiter

Inputs and Outputs

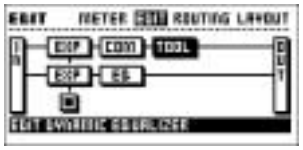
- Wordclock input (RCA)
- External control input
- Dual balanced XLR Mic/Line inputs and Line outputs with 24 bit A/D-D/A converters as well as AES/EBU, S/PDIF, Toslink and ADAT digital I/O – simultaneous multiple digital outputs is supported and can be dithered between 8 and 22 bits

Total and Partial Recall

- 100 factory, 200 user presets – you can also store up to 999 presets on a PCMCIA card
- Recall an entire preset or perform a Partial recall which enables you to load specific parts of a preset into the currently preset
- Instantly recall customized settings that match a specific instrument or vocalist

Four Processing Blocks

A variety of full featured dynamics, EQ and other specialized algorithms can be placed freely within any of four processing blocks available per channel. When using the 96kHz option, you can either use 1 channel and gain access to 3 blocks or use both channels with 1 block available per channel



- Use either an over-easy Noise-Gate or a more comprehensive Expander with additional parameters
- Two types of EQ each with proprietary SoftSat limiter technology on each band:

Advanced EQ— 5-band Parametric with a 20Hz-20kHz frequency range, ±18dB boost/cut range, variable bandwidths.

Easy EQ—emulates vintage analog EQs with -12 dB/oct Hi and Lo cut filters, ±18dB Hi and Lo shelving fixed-frequency bands and a ±18dB Mid band with 5 frequencies

- Two Compressor algorithms – The Soft Compressor provides smooth and transparent dynamic control while the Vintage Compressor emulates the warm musical sound of classic analog compressors
- The Tools block provides a choice of specialized algorithms including a De-esser, Dynamic EQ (a De-esser with additional frequency range and control), Digital Radiance Generator (emulates tube saturation) and R.I.A.A. equalization which lets you connect a turntable to the mic-inputs.

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FINALIZER EXPRESS

Studio Mastering Processor



The Finalizer Express is an affordable studio mastering processor that provides a fast and efficient way of optimizing the overall perceived level of your mixes with the tools that will deliver the finishing touches of clarity, warmth and punch that your music deserves. The Finalizer Express is based around a Multiband Dynamics algorithm, inherited from the TC Finalizer 96K, and an LED matrix that enables you to choose between 25 different settings organized by Compression Rate and Compression Style ranging from smooth and subtle to aggressive and hard compression. Three Spectral Balance controls let you change the level relationship between low, mid and high, making it possible to independently control the level of the three compression bands. Post-processor fades can be performed within the digital domain using the fade knob or by using the optional TC Master Fader, and even controlled via MIDI. All I/O and internal processing is done at true 24-bit – 16 and 20 bit dither is also provided for delivering your material to a 16, 20 or even 24 bit digital media.

FEATURES

General

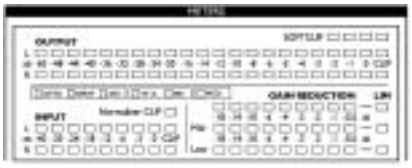
- Single rack space 24-bit mastering processor equalizes tracks with frequency selective compression – adds energy and clarity into mixes without creating digital "overs"
- Selectable 44.1kHz and 48kHz sample rates
- The multi-band compressor works over three frequency bands with fixed crossover points at 315Hz and 3.15kHz using 6dB slope linear phase digital filters.
- Auto Make-up Gain function compensates for the loss of gain through the compressor
- Balanced XLR analog inputs/outputs with 24-bit A-D/D-A converters and digital I/O including AES/EBU (XLR), coaxial S/PDIF and Optical Toslink

Spectral Balance

- Three-band EQ section with separate Low, Mid and High level boost/cut controls let you adjust the spectral balance of the signal
- Three Emphasis keys for adding compression and gain to the respective EQ bands

Fade In/Out

- Fades can be controlled using the front panel Fader knob, a remote Master Fader, or automated using a MIDI sequencer
- Fader values can also be transmitted via MIDI and recorded into a sequencer



Normalize

- Switchable Softclip algorithm emulates the input stage of analog preamps providing the desirable effect of the analog soft saturation that adds warmth to your signal
- The Gain control sets the amount of gain for the normalizer and determines how hard the compressor section will be driven.

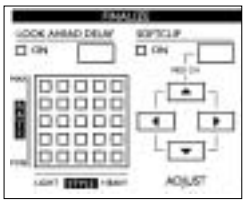
Metering

High resolution LED metering section gives you a clear readout of I/O levels, soft clipping, normalizer clipping and gain reduction for each of the three compression bands. LEDs also indicate digital input source, sample rate as well as MIDI activity.

Output Section

- Disengages all functions except Dithering letting you compare the original signal with the enhancements added by the Finalizer
- Control analog output gain (-26 to +6dB)
- Sync button lets you set the digital clock
- Toggle the digital Status bits used on all digital outputs

Finalize Matrix – The Heart Of The Finalizer Express

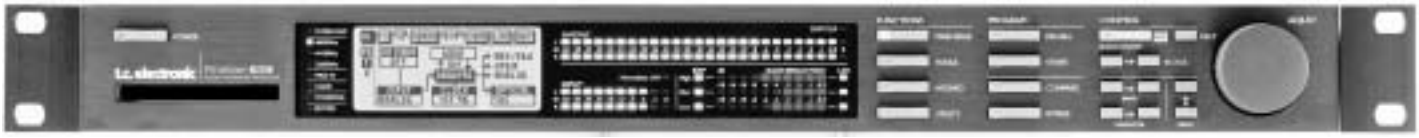


The 5x5 LED Finalize Matrix delivers 25 compressor presets that are easily selected using four cursor keys – each preset contains independent Ratio, Threshold, Attack and Release settings for each of the three compression bands.

- The vertical axis of the matrix represents Compression Rate, the horizontal axis shows Compression Style
- Switchable 3ms Look Ahead Delay switch allows unwanted peaks and transients to be tracked and controlled at the exact moment they occur
- Softclipper smoothly eliminates any overshoot that might occur after the three bands are joined together – starts at -3dBFS and gradually increases with the level
- A multiband brick wall Limiter with a fixed ratio of ∞:1 prevents unintentional compressor overshoots from causing overloads

FINALIZER 96K

Studio Mastering Processor



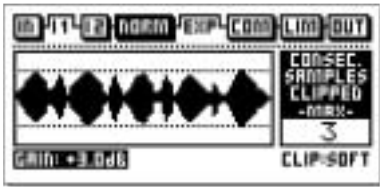
The Finalizer 96K takes the all-in-one stereo mastering processor concept to its highest level with true 24-bit / 96kHz processing and absolute control over every parameter over its advanced feature set including independent multi-band expansion, compression and limiting with definable crossover points, two insert effects blocks that allows you to choose from a wide range of appropriate shaping and enhancing tools. The Wizard function immediately calculates an optimal setting for your by asking you four simple questions concerning the source material and the type of processing you require. A group of analysis tools provide a visual reference and comparison between source material and the processed signal. The Finalizer 96K truly delivers unprecedented levels of clarity, warmth and punch to your mix putting the world of professional mastering within reach of every studio.

FEATURES

- Balanced XLR analog inputs and outputs and true 24 bit resolution A-to-D and D-to-A converters
- 100MHz processor makes it possible to process the entire signal using a 96kHz sample rate and achieve a maximum frequency response of 48kHz
- Sample Rate Converter translates AES/EBU, S/PDIF and Tos Input to the Finalizer's internal or external rate.
- The ADAT lightpipe interface enables you to freely choose 2 ADAT channels and direct them to other channels on the ADAT enabling you to bounce tracks while processing the sound with the Finalizer

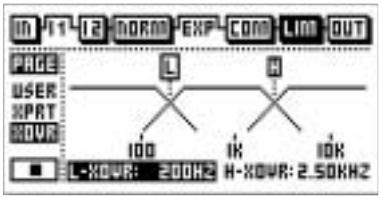
Analysis Tools

- A wide range of extremely useful analyzer functions are available providing visual feedback of source material and processed signals including: Phase Correlation Meter, Level Flow Meter, Peak Hold Meter, Digital I/O Status as well as a Calibration Tone Generator
- The Digital Status Tool recognized incoming status bits from an external device (DAT, CD) and lets you decide how the status bits such as SCMS settings are handled and for converting from S/PDIF to AES/EBU or vice versa



Normalizer

- The Normalizer block lets you optimize the signal's gain ensuring maximum utilization of headroom and benefit from the dynamics and effects blocks that follow
- A logical graphic display makes the adjustment a breeze
- You can choose between hard or soft limiting to avoid clipping

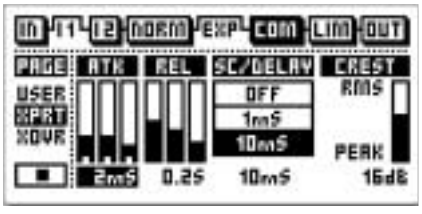


Crossover

- The Compressor, Limiter and Expander share three user-definable frequency bands allowing you to treat the low, mid and high frequency ranges independently without the 'pumping' or 'breathing' associated with conventional full bandwidth compressors

Expander/Compressor/Limiter

- These tools allow you to optimize the dynamics of your mix independently in three frequency bands
- The Expander, Compressor and Limiter each feature variable Threshold, Attack and Release controls for each of the three bands – Ratio is also variable for the Expander and Compressor but is fixed at ∞:1 for the Limiter. Each of the three Limiter bands can be individually bypassed
- Look Ahead Delay is individually adjustable for each of the dynamics processors - from off to 1ms to 10ms - this gives each processor more time to react to the present signal
- A Range control allows you to set the maximum amount of gain reduction in the Expander as opposed to cutting off signals below the threshold
- The adjustable Crest parameter determines whether the Compressor should react to peaks or RMS (average) or anywhere in between



FINALIZER 96K

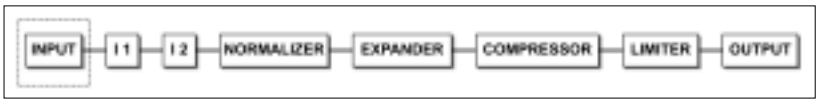
Recall

- Recall page features a selected list of 30 presets in ROM suitable for any type of source material including: Commercials, Jazz, Techno, Classical, Rock, Live and more. An available RAM bank holds 128 user-definable presets

TPDF Dithering

- Signals can be dithered from 8-22 bits
- Three types of dither are available: Uncorrelated Dither (for low level stereo signals), Correlated Dither (the most unobtrusive for mono signals), Inverse Dither (for unfocused low level signals).

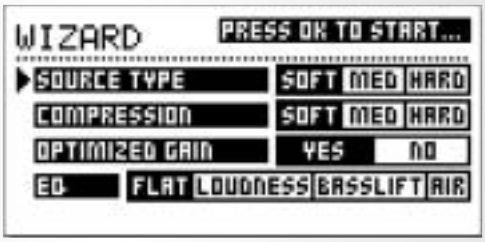
Insert Blocks



Two insert blocks located after the input section offer many useful effects including:

- Stereo Adjust allows you to increase the stereo width of a signal or collapse the image towards mono. You can also move the center balance from left to right
- Digital Radiance Generator (DRG) uses a drive control that adds analog-like second harmonic distortion to the signal – this process can provide a desirable warmth to digitally processed material. A curve control alters the character of the harmonics to better suit the specific source material
- Dynamic EQ is a compressor that works within a specific frequency range. It's similar to a de-esser but with more parameters (threshold, ratio, attack and release) and a broader frequency range with a choice of a bell curve or a shelving filter
- 5-Band Digital EQ – Lo and Hi shelving plus three full range bell-curved frequency bands, each with variable Q and ±12dB boost and cut – adjust the spectral content while preserving or enhancing the transparency and energy of your mix
- MS Encoding/Decoding includes balance and fine balance controls that allow you to change the relationships between the Mid and Side signals
- Spectral Stereo Image controls the stereo content in three bands independent of the Dynamics crossover. Use this to enhance the Stereo Image in a specific frequency range, e.g. widening the vocal area without removing the punch of the bassdrum
- External Inserts – The Insert block give you the ability to insert an external processor using the analog Inputs or digitally via AES/EBU, S/PDIF or Toslink

Wizard



With the Wizard function you are asked four basic questions about your source material and the type of processing you require and the Wizard will immediately calculates an optimized setting for your material. The more experienced user can use the optimized Wizard presets as a starting point and tweak the signal path extensively with more than 90 parameters to choose from.

Up- and Down Sampling

- Input sample rates of 44.1. 48, 88.2 and 96kHz are accepted and can be up/down sampled to 44.1. 48, 88.2 and 96kHz
- When working with old or already finished material, you have the ability to Up Sample to 96kHz re-master the material
- Up/down sample option also makes you well prepared for the DVD-Audio or super-CD audio format

Finalizer Model:	**96K**	**Express**
24 bit converters (A/D, D/A)	Yes	Yes
Multi-band Compression (3-Band)	Yes	Yes
TC Softlimiter	Yes	Yes
High Resolution Metering	Yes	Yes
Automation capability	Yes	Yes
Midi In/Out/Thru	Yes	Yes
External Control Input	Yes	Yes
Auto Sensing Power Supply	Yes	Yes
Automatic "make-up" Gain	Yes	Yes
RMS & Peak Detection Modes	Both	RMS
"Look Ahead" Delay for 3 bands	Global	Global
S/PDIF I/O's	Yes	Yes
ADAT I/O's	Yes	No
Tos Link I/O's	Yes	Yes
Normalizer	Yes	Yes
Simultaneous Digital Outputs	Yes	Yes
Dithering	Yes	Yes
Optional Master Fader	Yes	Yes
Adjustable Crossover Points	Yes	No
Storable Presets	Yes	No
Digital Radiance Generator	Yes	No
External Sidechain triggering	No	No
Inserts-External or Internal	Yes	No
5-Band EQ	Yes	No
De-Esser	Yes	No
Dynamic EQ	Yes	No
Stereo Width Adjust	Yes	No
MS-Encode/Decode	Yes	No
Expander/Gate	Yes	No
BNC Word Clock	Yes	No
Adjustable Limiting (3 Band)	Yes	No
Sample Rate Conversion	Yes	No
PCMCIA Card Slot for Preset Storage	Yes	No
Individual Crest Factor on 3 bands	Global	No
96 kHz/88.2 throughput	Yes	No
Full up and Down-Sampling	Yes	No
3-Year Parts/Labor Warranty	Yes	Yes

TC ELECTRONIC

M300

Dual Engine Processor



The M300 is a 24-bit effects processor with a dual engine structure that provides separate Multi-effect and Reverb effect engines that can be used independently or combined in a number of configurations. The no-nonsense user interface features dedicated, easy to use controls that provide immediate access to effects parameters without the need to scroll through layers of menus. High quality 24-bit A-D/D-A converters with balanced 1/4˝ TRS connectors, 24-bit coaxial digital I/O and 24-bit internal processing ensures pristine sound quality with a dynamic range of 100dB or more. Other features include tap tempo, 355 presets, MIDI I/O and more. Incorporating TC's legendary reverb and effects capabilities at an unprecedented price, the M300 is ideal for any PA, live or studio requiring high quality effects with immediate results.

FEATURES

General

- Dual engine structure – a Multieffect Engine with 15 different delay, modulation and dynamics effects and a Reverb Engine also with 15 effects
- Serial and dual mono routing allows the two independent effects engines to be used simultaneously
- High quality 24-bit A-D/D-A converters and 24-bit internal processing
- A total of 355 Presets are available — 256 factory presets comprised of combinations of 16 Multieffects and 16 Reverb effects and 99 user-defined presets

Input Section

- An input level control lets you set the optimum level for the analog or digital inputs
- A dedicated mix control sets the ratio between the dry and wet signals. In Dual Send/Return Routing, Mix controls the overall wet/dry mix of both effects sections; In Serial Routing, the Mix control works only on the selected effect engine
- The Bypass switch operates as a Mute function for each effect with Dual Send/Return routing and simply passes the dry signal unprocessed to the Output in Serial mode
- Effects Balance control allows you to set the ratio between the two effects engines

The Multi-effect Engine

- The EFFECT selector control lets you select between one of 15 effects and Off
  - Dynamic Delay
  - Studio Delay
  - Tape Delay
  - Delay
  - Ping Pong
  - SlapBack
  - Vintage Phaser
  - Phaser
  - De-Esser
  - Compressor
  - Chorus
  - Flanger 1
  - Flanger 2
  - Soft Tremolo
  - Hard Tremolo
  - Off
- Two multifunction knobs control different parameters depending on the selected effects algorithm — For delay effects, the knobs control a Timing multiplier and feedback, For modulation effects the knobs control amount and depth and for dynamics effects the knobs control drive amount and ratio (or frequency with the de-esser)
- -12dB, -6dB, -3dB Gain Reduction LEDs indicate the applied Gain reduction in the Compressor and De-Esser algorithms
- The tempo of Delay-based effects are set by tapping the Tap key to set a global tempo and then rotating the Timing knob (operates as a multiplier) to control the note value of the delay. The range of the timing knob is 0.5 to 2 (1 = a quarter note
- Up to 5 seconds of delay is possible

The Reverb Engine

- 15 True Stereo Reverb algorithms deliver the best from TC in an extremely user friendly fashion:
  - Concert Hall
  - Living Room
  - Club
  - Plate I
  - Plate II
  - Spring
  - "Live" Reverb
  - Ambience
  - TC Classic Hall
  - Vocal Studio
  - Vocal Room
  - Vocal Hall
  - Drum Box
  - Drum Room
  - Large Cathedral
  - Off
- Dedicated PreDelay, Decay and Color parameter controls help you achieve your desired sound in seconds

Rear Panel

- 1/4˝ TRS analog inputs and outputs as well as auto-switching (44.1 & 48 kHz) digital coaxial S/PDIF I/O
- The 1/4˝ TRS Pedal Input gives you the ability to Bypass and Tap the global tempo via momentary switches
- MIDI In/Out recognizes MIDI Clock Tempo Sync and transmits/receives program change, CC data for realtime parameter control and SysEx data
- Internal switchable 100- 240v power supply

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D•TWO

Multi-tap Rhythm Delay



Based on the history of TC's classic 2290 Delay, the D-Two offers a very musically oriented Rhythm Tap feature which can easily be applied to live sound and recording applications -as well as post production, broadcast, installed sound and a variety of other situations where a creative and intuitive delay can be used. Featuring up to 10 seconds of delay, the D-Two provides six direct-access add on features, including Spatial, Ping Pong, Reverse, Dynamic Delay, Chorus and Filter. 50 ROM presets and 128 user RAM preset locations are available. The D-Two's hardware compliment consists of full resolution 24-bit A/D and D/A converters. 1/4˝ analog I/O connectors are provided for fast and simple connectivity, additionally a S/PDIF digital I/O is included for an all-digital, 24-bit signal path supporting both 44.1K and 48K sample rates.

FEATURES

True 24-Bit

- State-of-the-art hardware with true 24 bit resolution A-to-D and D-to-A converters and true 24 bit RAM memory ensure the best possible audio quality
- Left and Right balanced analog 1/4˝ TRS inputs and outputs are provided as well as coaxial S/PDIF digital I/O – the digital output can be dithered from 16 to 24-bit
- Sample rates of 44.1 and 48kHz are supported

Three Delay Modes

- Traditional mode is a standard delay setup with feedback control
- In Straight mode, repeats are caused by a number of taps that are not fed back into the delay line providing you with ultimate control over the exact number of repeats
- Rhythm mode allows you to tap a specific rhythm using the Rhythm/Tap key or quantized according to a specific tempo and subdivision – after the last tap, the signal can be fed back into the delay line, generating a complete rhythm sequence
- The delays and rhythm patterns can be up to 10 seconds each
- There are 50 factory presets and 100 user-definable presets

Six Unique Add-On Effects

- Access to all parameters of the six add-on effects is available on the front panel – double click the key of the effect you wish to work with, and to instantly access the effects parameters
- TC's world acclaimed Chorus is provided using pitch-modulated delay
- Hi and Low cut filters with a frequency range of 19.95Hz to 20kHz can be applied to the delayed signal or to the incoming source signal allowing you to get a warmer and more natural sound or create delays that are less obtrusive on a mixes
- Spatial effect allows you to add more width to the sound of your material by offsetting the left channel up to ±200ms or by reversing the phase on any channel or both
- Reverse Delay effect processes the signal, and adds a backward delay
- Dynamic Delay allows the delay output level to be dynamically controlled by the input level – this leaves the source material undisturbed while played and delicately accompanied by the Delay between phrases
- The Ping Pong effect syncs panning to the delay time – three styles are available: L-R, L-C-R and right and Dynamic which will fit the number of Delay repeats with the number of panning positions



User Interface

- The intuitive user interface is aided by the detailed Multi Spectral LCD Display which gives you a complete overview of what is going on in the processor including Preset numbers, Input level and Dynamic (gain reduction) meters, Delay time, Tempo and Sub-Division and more.

MIDI

- MIDI clock is supported for tempo sync – incoming MIDI clock can be subdivided
- MIDI continuous controller capabilities allow you to control parameters externally in realtime or via a sequencer – all effects parameters can be changed via an external MIDI device through Sys-Ex

Conveniences

- Autoswitching internal power supply will power-up anywhere in the world
- The 1/4˝ pedal input can recognize two pedals simultaneously: The Ring pedal is fixed to Tap tempo while the Tip pedal can be set to Bypass, Tap tempo or Rhythm tap

TC ELECTRONIC

M•ONE XL

Dual Effects Signal Processor



The M•One XL is a 24-bit multi-effects processor with a dual engine design that provides you with high quality TC Reverbs along with a wide variety of other dynamics, pitch, modulation and delay algorithms. Flexible routing capabilities allows you to access the effects engines in a number of combined and independent configurations. The LCD display gives you a complete overview of input levels, selected input, clock source, routings and current algorithms. The dedicated Tap key and MIDI clock compatibility ensure that tempo-based effects including delay and modulation effects such as tremolo, will groove with the music. Positive-locking balanced XLR inputs and outputs with 24-bit A-D and D-A converters as well as coaxial S/PDIF digital I/O provide just the right connectivity for any live sound or recording studio application.

FEATURES

Dual Engine Effects

- 200 Factory presets are provided covering almost any imaginable application – 100 presets can also be stores on the User bank
- Dual Engine structure allows you to run two high quality effects, derived from 25 algorithms – run two of the best sounding reverbs or other quality effects simultaneously without compromising sound
- A wide range of high quality reverbs algorithms from classic Halls and Rooms to grainy snare reverbs such as Live and Plate each create sound reflections in various environments allowing you to easily add different levels of depth and color to your source material
- Other high quality effects algorithms include Compressor and Limiter, Delays, Chorus, Parametric Equalizers, Flanger, Gate, Expander, De-esser, Tremolo and Phaser

Inputs and Outputs

- 24 bit A-to-D and D-to-A converters and 24-bit internal processing at internal sampling rates of 44.1and 48kHz or DI which locks to the incoming digital clock
- Balanced left and right XLR inputs/outputs
- Coaxial S/PDIF Digital I/O is also provided the output can be dithered to 16-, 20- or 24-bit (off)

Tap Tempo

- TAP key lets you Tap in the global tempo for time-based effects like delay and chorus. Tempo is displayed in BPM or milliseconds
- The tap key also lets you enter the Tap menu where you can set up the subdivision of the tapped tempo

User Interface

- Dedicated controls are provided for controlling input level, wet/dry mix and effect balanced between both effect engines
- Easy Parameter Level helps control important parameters and create cool effects
- Each algorithm has its own Edit button – up/down cursor keys and a large control wheel as well as dedicated enter/exit buttons allows you to intuitively navigate through each of the effects comprehensive range of sound shaping possibilities
- For maximum flexibility, separate bypass keys, each with its own assignable Bypass Mode, are provided for each effect engine –

  0% Mix – passes the input directly to the output

  FX Input – shuts off the Engine Input in order to let the effect "ring out", while allowing the dry signal to pass through

  FX Output – shuts off the engine output killing the FX instantaneously while allowing the dry signal to pass through

Routing Flexibility

- A dedicated Routing key gives you the flexibility of choosing from 6 different routing configurations for the two effects engines
- The Routing Lock function allows you to stick with a single routing configuration when switching presets
- The Dual Send/Return setups the system up as two independent (mono in, stereo out) effects processors – The left input is sent to engine 1 and the right input is sent to engine 2. The four internal FX Outputs are then summed to the stereo outputs
- Parallel routing sums the left/right inputs, and both engines are fed with the exact same signal but not into each other – ideal for adding two effects to the same source
- Serial routing sends the input signal through engine 1 then engine 2 – ideal for combining two effects
- The Parallel/Serial option uses separate inputs on the two engines while an internal send control lets you decide how much of Engine 1 is fed into Engine 2
- Stereo Linked Routing allows true stereo processing with synchronized parameter settings for both engines (engine 1 is the master) – The left I/O feeds engine 1 while the right I/O feeds engine 2
- Dual Mono routing offers two totally independent engines with mono in/mono out of each Engine

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M•ONE XL

Multi-Spectral LCD Display



- Peak meters shows the Input level of the left and right channels within a range of 0 to -40 dB – Overload LEDs indicate whether the Input level is too hot or if there is an internal DSP overflow
- The Analog/Digital LED indicates the currently selected input source
- The Sample Rate indicator shows the clock source and the incoming master clock
- The Routing indicator shows what Routing mode(s) are currently in use
- The ALGO Indicator shows the current algorithms in each of the effects engines
- Two Dynamic meters show the amount of gain reduction when an Engine is using any Dynamic algorithms
- The preset number and the preset type (Factory or User) are also displayed
- The "Edited" icon is lit as soon as the current recalled preset has been modified
- The MIDI IN icon shows any incoming MIDI activity



MIDI Section

- All effect parameters can be accessed via standard MIDI controllers allowing real-time control and automation with virtually any sequencer or other MIDI devices
- You can also use MIDI to dump & save the entire user bank to a MIDI sequencer
- The TAP Tempo function of the can be locked to the incoming MIDI clock

Additional Features

- A 1/4″ pedal input accepts a momentary switch for controlling bypass or tap
- Internal auto-switching power supply – 100 to 240 VAC, 50 to 60 Hz
- 1 year parts and labor warranty

Effects Algorithms

- Hall
- Room
- Small Room
- Plates 1&2
- Spring
- Live
- Ambience
- Delay One Tap
- Delay Two Tap
- PingPong Delay
- Chorus Classic & 4-voice
- Flange: Classic & 4-voice
- Pitch: Detune & Pitch Shift
- Parametric EQ
- Compressor/Limiter
- Gate/Expander
- De-esser
- Tremolo
- Phaser Vintage & Smooth

Also Available: The Original M•ONE

The original M•ONE is still available and may be a viable option for users with unbalanced equipment and for guitarists. It has the same features as the M•ONE XL less the XLR connectors (the M•ONE has 1/4″ I/O) as well as the specific algorithms that have been optimized for live applications. Other features of the M•ONE XL that are not available with the M•ONE include Ping-Pong Delay, additional reverb size tunability, MIDI mapping and Tremolo Pitch.

TC 2290 *Dynamic Digital Delay*

Acclaimed as the industry standard Digital Delay, the TC 2290 can be found in recording studios, PA rigs and guitarists' FX racks the world over. The TC 2290 combines unique operational features and superior sound quality, allowing you a greater degree of musical creativity in the use of effects.



- 1/4″ and XLR inputs and outputs
- Programmable Chorus, Ducking, Gating & Panning
- 4 seconds memory (8 seconds in Double Sample / Double Delay time Mode)
- 20Hz - 20kHz bandwidth and a 100dB dynamic range
- The most functions are accessible from the front panel including a numeric keypad for data entry, "Learn" button for easy entry of delay time and tap tempo control.
- Feedback control with phase invert as well as Low Pass (8, 4, 2 kHz) and Hi Pass (100, 200, 400 Hz) filters.

Dynamic Control

- The input signal dynamically controls 3 independent LFOs providing unique chorus, flanger, ducking, gating and pan effects
- The Dynamic Function enables you to suppress the effects when you are playing and let the effects reappear when you stop playing or the opposite.

Modulation

- Individual control is provided for Delay, Pan and Dynamics – waveforms include Sine, Random, Envelope, & Auto Trig.

MIDI

- MIDI enables the TC 2290 to operate as a "main brain" of an FX rack. External FX units or foot pedals can be incorporated into the chain of five programmable effects loops which can be stored within the patch memories.

TC ELECTRONIC

M3000

24-Bit Studio Reverb Processor



The M3000 is a single rack space, dual engine effects processor that features an advanced set of reverb algorithms along with a full compliment of other TC-quality signal processing algorithms including delays, pitch,modulation, dynamics, EQ and more. At the heart of the M3000 is the VSS3 (Virtual Space Simulator) reverb algorithms—representing years of research in reverberation technologies in pursuit of optimal realism, smoothness and clarity of both real and imagined spaces.
A fast and intuitive user interface provides easy recall and editing of 100s of factory presets which can be stored in RAM or PCMCIA card. The back panel features gold plated connectors and includes AES/EBU, S/PDIF, Toslink, ADAT-optical, MIDI and balanced XLR analog I/Os as well as a pedal input. Multiple simultaneous digital outputs can be used with all of the digital connectors. Additional features include LCD display, LED metering, and output dithering.

FEATURES

Dual Engine

- Dual Engine processing fuses TC's proprietary DARC chip technology with a powerful Motorola 80 Mips DSP chip to deliver the intensive processing power necessary to calculate the huge number of sonic details made available within the VSS 3 algorithms
- The Dual Engine structure allows any combination of full-blown algorithms to be run on each engine simultaneously – the two engines can be linked providing true stereo processing
- Choose from 6 different routing setups including serial, parallel and dual input, dual mono, stereo linked as well as Preset Glide which allows you to crossfade between two effects

VSS 3 Virtual Space Simulator

- Provides the highest degree of smoothness and clarity to the reverb tail with the ability to keep the signal 100% free of sound deteriorating modulation – modulation is still available as an option
- A large number of directional taps (between 40 and 100) are employed, creating accurate early reflections that aid in simulating the sonics of actual rooms
- Gentle signal processing ensures that the signal is always 100% correct pitch, even with extensive effects processing

More Algorithms

- Apart from the VSS3 and VSS3 Gate Reverb algorithms, you can choose between C.O.R.E. Reverb (based on TC's M5000 processor) , REV-3 Reverb, Delay, Pitch, EQ, Chorus, Flanger, Tremolo, Phaser, Expander/Gate, Compressor and De -Esser

Presets

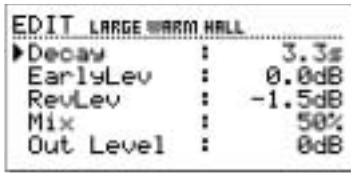
- 600 high-grade factory presets – 500 single engine presets plus 100 presets that combine engines 1 and 2, cover most any application imaginable
- An internal RAM bank allows you to store store 250 single and 50 combined presets
- The PCMCIA card slot on the front panel can be used to store an additional 250 single and 50 combined presets

Tap Tempo

- Tap Tempo key not only lets you match the Delay Time to the rhythm of a track, you can also use tap tempo to set the parameter for Reverb Decay, Chorus Speed, Flanger Speed, Phaser Speed or Tremolo Speed

Dynamic Morphing

- Dynamic Morphing allows you to morph from the preset on one engine to the preset on the other – controls are provided for the threshold level and the transition speed



Editing

- Each of the two Engines has it's own Edit key giving you the option of editing presets and changing any parameter to create the exact ambience or effect you want
- Every parameter is on one scrolling page with the most used parameters at the top – No "nested" menus with their hard to find parameters
- VSS3 and VSS3 Gate reverbs can be edited in either Easy Mode or Expert Mode
  - In Easy Mode the Hi Color and Lo Color tools let you edit the complex spectral balance of the reverb in a second
  - In Expert mode you have access to a wide number of well-arranged parameters

Wizard

- Wizard function is a fast and intuitive way to get the sound needed for your work. Simply select the application, source and size you are looking for and the M3000 suggests a number of presets matching your request.

FIREWORX

Studio Effects Processor



The FireworX is a radical sounding, extremely flexible multi-effects processor designed to challenge your imagination. A wide variety of internal effects, based on over 35 different algorithms can be inserted on an easy to use 8x8 routing grid, and controlled in realtime via a powerful yet intuitive modulation matrix. An advanced digital signal processor along with TC's proprietary DARC 3 chip technology allows the FireworX to run multiple effects simultaneously. A dynamic processing power allocation structure ensures that you always get the advantage of the full power of the FireworX allowing you to keep adding algorithms to the signal chain until all the processing power is used. Hundreds of factory and user presets plus a PCMCIA card slot for additional storage. Balanced analog inputs and outputs using 24-bit converters as well as a full compliment of digital I/O and word clock in is provided as is MIDI In/Out/Thru and a 1/4˝ external control input.

FEATURES

Effects

- 200 Factory Presets and room for up to 200 User Presets (depending on the size of the presets) – up to 999 additional presets can be stored on an optional PCM CIA card
- A word search filter allows you to scroll through presets that contain a specific algorithm such as Reverb

Effects Routing

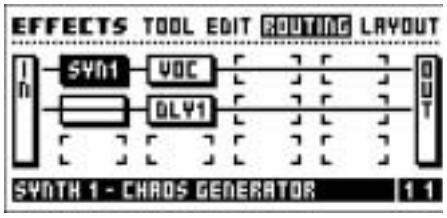
- 8x8 position routing grid lets you place any of the over 35 algorithms anywhere on the grid where they will be automatically "wired" together – Left/Right can be set up separately for Input/Output in each block

Modifier Matrix

- A total of nine external and eleven internal Modifiers can be tied to numerous parameters – up to twenty connections can be made simultaneously
- External Modifiers can be controlled via MIDI controllers, Velocity, Pitchbend, Aftertouch, Note-onkey etc
- Internal Modifiers include: 2 envelope followers, 2 ADSR generators, 2 LFO's, a Pitch detector as well as a Free Form modifier that uses a sequencer that lets you design your own modulation waveform to create a rhythmic sequence

User Interface

- The twelve effect block keys on the front panel are used for bypassing or muting the active effect blocks
- Navigation is made simple via Left/Right arrow keys, parameter and value wheels as well as Enter and Exit buttons
- The Alpha Modulation wheel provides realtime control of a number of parameters in the current preset simultaneously
- All pertinent info is clearly layed out on the 56 x 128 dot backlit LCD display
- Left/Right LED input level and gain reduction meters are also provided



Tempo Control

- All tempo based parameters can be connected to the global tempo
- The global tempo can be tapped, dialed in, or MIDI clock based
- Each parameter can be set from 16 bars through 1/1 to 1/32 straight, dotted or triplet subdivisions of the global tempo

Inputs and Outputs

- Balanced XLR analog inputs and outputs with 24-bit A-to-D and D-to-A converters as well as AES/EBU, S/PDIF, Toslink and ADAT digital I/O
- Wordclock input (RCA)
- External control input
- Analog or digital I/Os can be used as an insert loop making it possible to connect an external device

Algorithms

Dynamic –	Expander/Gate, Soft Compressor Hard Comp/Limiter
Vocode –	Vocoder, Ring Mod. Internal, Ring Mod. External
Formant –	Formant
Pitch –	Single Voice, Dual Voice
Filters –	Resonance, Bandpass, Phaser, Resonator, Resochord
Distort –	Drive, Cruncher
Synth –	Curve Generator, Chaos generator, Noise Generator
Reverb –	Simple, Advanced
Chorus/Flanger –	Classic Chorus, Advanced Chorus, Classic Flanger, Advanced Flanger
EQ –	Fixed EQ, Modifiable EQ
Delay –	Stereo, Dual, Dual three-tap One-tap, Six-tap, Reverse
Panner –	Simple Tremolo, Advanced Tremolo Simple Panner, Surround Panner, Stereo Enhancer

TC ELECTRONIC

INTONATOR

Vocal Intonation Processor



The Intonator is a high resolution pitch correction processor aimed at reducing the often tedious task of doing vocal re-takes. Intuitive user interface gives you a choice between manual and automatic pitch correction with complete control over scale, correction amount and additional parameters that allow you to preserve the artist's unique vibrato and initial intonation while providing correction of targeted notes. Besides offering comprehensive pitch correction, the Intonator offers a variety of valuable vocal-specific processing tools including a De-esser and a special Adaptive LoCut filter (ALC) that automatically adjusts in relation to the current pitch. There is also TC's world-renowned DARC 3-chip technology, 96kHz processing and premium 24-bit A-D and D-A converters for an ultra-transparent signal path worthy of your most precious vocal tracks. To ensure compatibility with any audio system, the Intonator is equipped with XLR-balanced inputs and outputs as well as a full range of digital I/O and a word clock input.

FEATURES

Transparency

- Proprietary DARC 3 chip technology incorporates 96kHz internal processing and true 24 bit resolution that provides an ultra-transparent signal path
- Balanced XLR inputs and outputs – analog input and output gain controls allow you to take full use of the dynamic range available by the premium 24-bit converters
- Fully integrated AES/EBU, S/PDIF, Tos-link and ADAT digital I/Os with TPDF output dithering (8, 12, 16, 20, 22 or 24 bit)
- 96kHz, 88.2kHz, 48kHz and 44.1kHz compatible on both digital and analog I/O's

Pitch Correction

- Use the "Do-not-process-anything-but-this-note" function to correct only a single note or type in your personal scale
- The Window knob determines how close to the target note, ±200 cent, the source should be in order to be corrected (100 cent is 1 semitone)
- The Attack knob controls how fast the pitch is corrected to the "correct" note
- The Amount knob sets the amount of correction added to the signal
- The Note Hold key is used to hold the current corrected pitch

Display and Keyboard

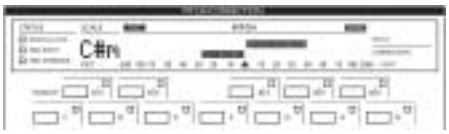
- The Display shows you the current Root note and Scale. While processing you can see both the incoming signal and the amount of correction added to the source, creating a visual confirmation of the process taking place
- The one octave keyboard on the front panel gives you instant access to changing the Root note of the current scale or the notes allowed to be corrected – it also indicates the incoming note, making it possible to recognize the note about to be processed

Audio-To-MIDI

- You can monitor and edit vocals externally by tracking the incoming pitch history to a MIDI sequencer, making the necessary corrections and play back the pitch corrections via MIDI back to the Intonator
- You can also use a MIDI keyboard for real-time manual pitch correction simply by hitting the notes to be processed

Manual Pitch

- The Manual Pitch control allows you to add pitch correction by hand
- The wheel can also be remote controlled by keyboard modulation wheel or even automated when locked to a sequencer



De-Esser

- The De-esser uses a relative threshold technique allowing the De-esser to adapt to the incoming signal and maintain the same threshold to the average level regardless of the input signal

LoCut Filter

- The LoCut filter can be used either in a conventional (fixed) mode or an adaptive mode where the LoCut is controlled via pitch – ensuring a perfect LoCut that never removes too much low end

Two Signal Setups

- In Normal mode, the ADIOS (Analog Dual I/O's) configuration lets you to make simultaneous recordings of processed and unprocessed vocals, making comparisons a breeze
- In Dual mode you have the option of pitch-correcting the signal in one channel while using the dynamics, De-esser and LoCut, on the signal in the other channel

REVERB 4000

The Stereo Incarnation of System 6000





The Reverb 4000 delivers the massive System 6000 Reverb Palette to you in a single rack space true stereo processor. Reverb 4000 is powerful enough to be your main reverb delivering the depth, localization, spaciousness and character demanded by world class recording facilities. It utilizes the latest hardware technology and software based upon 30 man-years of development. Its broad palette spans from new state-of-the-art Reverbs to world-renowned classics, and includes the best stereo reverbs and presets from System 6000 and M5000. A full complement of I/O protocols is supported including balanced XLR analog I/O, AES/EBU, S/PDIF and ADAT digital I/O. This ensures that the Reverb 4000 will integrate seamlessly into any production environment — from computer based DAWs to large-format mixing consoles. The extremely easy to use instant access interface provides instant access to key parameters, plus you are given the option of connecting the Reverb 4000 to your computer via USB where allowing you to access using the intuitive ICON software editor for Mac/PC.

FEATURES

- 44.1 to 96kHz sample rates and processing
- 24 bit A-to-D and D-to-A converters as well as 44.1 to 96kHz Sample Rates
- One engine, massive SRAM, no compromise design.
- VSS-4 allows you to create realistic stereo spaces from two discrete sources.
- Digital and analog wide dynamic range topology
- Over 150 preset and user programs

The Ultimate Reverb Palette

- True Stereo Reverbs taken from the renowned System 6000
- New pristine Stereo Reverb
- Favorite presets and algorithms from the M5000
- Realistic Environments from a closet to a canyon
- VSS-4, Source based Reverb providing Rooms with Character
- Classic Reverbs providing Polished Sustain
- Vintage Reverb emulations including EMT 250 plate.

Instant Access User Interface

- The front panel provides instant access to key parameters - locating presets from System 6000, M5000 and vintage devices is a breeze with the dedicated Search function. Switch between Character, Glossy and Vintage Reverb effects with just a single touch of a button.
- 56 x 128 dot backlit graphic LCD display.
- PCMCIA card slot supports up to 2 MB SRAM for storing additional programs.

ICON Software Editor

- The included ICON software editor for Mac/PC provides easy access to the Reverb 4000 functions with a complete overview for detailed editing purposes.
- The USB connector on the rear of the Reverb 4000 connects directly to the ISB port of your Mac or PC for accessing the ICON software.
- Ideal for live sound applications as well as integrating with computer-based DAWs.

Inputs and Outputs

- Balanced XLR analog inputs and outputs
- AES/EBU, S/PDIF, TOS-Link and ADAT digital I/Os.
- The MIDI In, Out and Thru connectors are provided for controlling any parameter from a sequencer or from a remote midi controller.
- Word Clock input (RCA) ensures accurate sample rate synchronization.



TC HELICON

QUINTET

Affordable, Pro-Quality Vocal Harmony and Reverb for Stage and Studio



The Quintet is an affordable, professional quality vocal harmony and reverb processor for both live and recorded vocal applications. It combines TC Helicon's renowned harmony algorithms and TC Electronic's famous Reverb technology along with a high quality microphone preamp. The Quintet is also fast and easy to use — editing and control of all harmony and effect parameters either from the front panel or via MIDI. Using the Quintet live is like bringing four professional backup singers to your gig. The studio quality mic preamp and 24 bit A-to-D conversion alone justifies the Quintet's place in your rack. Add to that the ability to create high quality, solo-able harmony vocals and you see what makes the Quintet an indispensible tool for your studio.

FEATURES

- HybridShift harmony generation delivers a smooth, natural sound.
- Four harmony voices plus lead voice double-tracking with control over timing and Pitch Humanization of harmony voices.
- Five distinct harmony Modes:
  Scale (Scalic) Mode harmonies use key and scale information to create musically correct, diatonic harmonies.
  Chordal Mode harmonies take your chord information, that you input in realtime via MIDI, to create intelligent, diatonic harmonies based on your voice.
  Shift Mode takes the pitch of the lead voice and creates harmonies a set number of semitones away, based on that pitch.
  Thicken Mode presets do not produce harmony voices, instead they produce voices in unison with the input voice. The unison voices have pitch and time humanization applied to them to make them sound as if sung by another person.
  In **MIDI Notes Mode** you provide specific MIDI note information to determine the pitch of the harmony voices.
- Selectable equal temperament or just tuning for harmonies.



User Interface

- Fast and easy to use with straight forward and simple editing with control of all harmony and effect parameters either from the front panel or via MIDI.
- 50 Presets are provided as a starting point. Each preset can be modified to suit your needs and overwritten.
- A/B program switching allows bulletproof program changes during live performance.

TC Electronic Reverb

- The reverb section adds depth and space for blending harmonies. A variety of high quality acoustic reverb simulations are provided including Living Room, Chamber, Club, Classic Hall, Concert Hall, Large Cathedral, Vocal Studio, Vocal Room, Vocal Hall, Ambience, Live Reverb, Plate1, Plate2, and Spring.

Inputs and Outputs

- High-quality balanced XLR mic preamp with variable gain and a 20dB pad.
- Mono 1/4″ TRS balanced/unbalanced line level input.
- 24-bit processing with 44.1/48 kHz
- 24-bit Coaxial S/PDIF digital output
- MIDI In, Out and Thru connectors allow you to use MIDI CC to 'echo' the front-panel controls.
- 1/4″ footswitch input compatible with the TC Helicon Switch-3 footswitch controller.

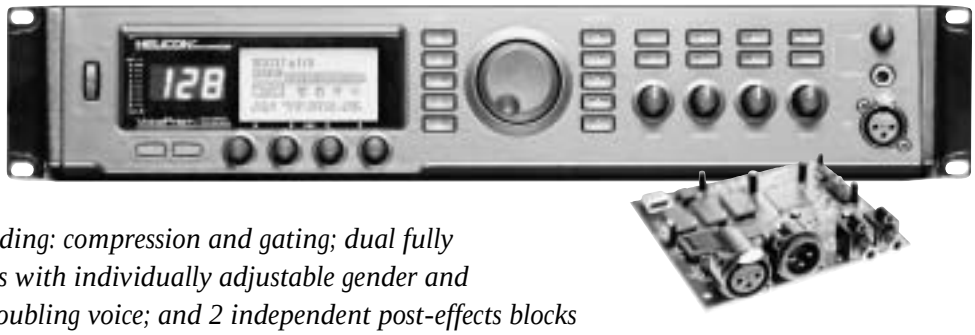
Switch-3 Footswitch

- Three-button footswitch is the standard footswitch for controlling the Quintet, VoicePrism, VoicePrismPlus, VoiceOneand VoiceWorks.
- Provides remote control of such features as, Bypass, Harmony Mute, Preset Up, Preset Down and more...
- Durable all-metal construction, making it durable and portable. It also includes the connnector cable.

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VOICEPRISM PLUS

Vocal Formant Pitch Processor



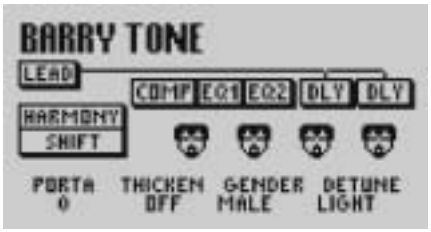
The VoicePrism Plus is an entire vocal channel that provides the ability to go directly from a phantom powered mic into a high quality mic preamp where it can access a variety of fully editable onboard vocal processors including: compression and gating; dual fully parametric EQ; four harmony channels with individually adjustable gender and humanizing parameters), a 5th lead-doubling voice; and 2 independent post-effects blocks with a choice of chorus/flange, delay and reverb. The included VoiceCraft card provides a number of unique voice modeling algorithms that allow realtime resynthesis and reshaping of the human voice including the ability to add breath, growl, rasp, head and chest resonance, inflection or vibrato.

FEATURES

- Contains a complete complement of tools designed to expand what is possible to achieve with the most beautiful of all instruments, the human voice.
- The VoicePrism has a full range of lead and backing channel vocal processing under preset control.
- 24-bit A-to-D and D-to-A converters

Processing Stages

- The **Front End** features a high quality mic preamp with switchable phantom power that provides direct access to a variety of onboard dynamics, harmony and effects processors.
- The **Pre Effects** section provides compression and Gating and a dual fully parametric EQ that can be assigned to the lead channel, the harmony channel or both.
- The **Harmony Channel** provides up to four-voice harmony formant corrected pitch shifters to create very natural harmony voices from your vocals. Each voice also features independent humanizing parameters including: Gender, Vibrato, Timing, Randomizing and Scooping.
- The **Lead Channel** has a thickening algorithm to create double tracking effects.
- Using the two **Post Effects** blocks you can apply to your Lead and Harmony voices including Chorus, Flange, Delay and Reverb with Pre-delay to create interesting and unusual spatial effects.



Intuitive Interface

- 128 fully editable effects and harmony preset libraries are easily accessible via an intelligent browser with context sensitive help menus, as well as the ability to audition sampled human voices for preset previewing.
- Dedicated edit buttons allow you to jump directly to multi-page edit menus – Vocals, Effects, Comp/EQ, Mix, Step.
- The 128 x 84 LCD Display shows preset information as well as menus where you can view your editable features.
- Four multifunction softknobs, underneath the LCD display, can be pushed and turned to quickly navigate through parameters and modify their settings.
- The Data Wheel provides additional navigation and editing control.
- A numeric preset number display and 10-segment LED level meters are provided for additional operating status.

Inputs and Outputs

- Balanced 1/4″ TRS line inputs and outputs.
- A balanced 1/4″ TRS balanced auxiliary input allows you to integrate additional audio material into the final effects blocks and mix them there with the vocals.
- Input sensitivity for the line and aux ins are switchable for +4dBu or -10dBV operation.
- Coaxial S/PDIF digital output at 44.1kHz
- 1/4″ headphone output with level control on the front panel.
- Can be configured to use a 1 or 3 button footswitch. You can set the number of buttons and assign button functions in the FOOTSWITCH menu, accessed by pushing the UTILITY button.
- MIDI In, Out and Thru allows realtime parameter control via MIDI CC.

VoiceCraft Card

- The optional VoiceCraft card, included with the VoicePrism Plus, adds human voice modelling DSP algorithms that provide realtime resynthesis and reshaping of the human voice including the ability to add breath, growl, rasp, head and chest resonance, inflection or vibrato. The Card also adds AES/EBU and S/PDIF I/O.

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TC HELICON

VOICEONE

Voice Pitch Shifting, Correcting And Modeling Tool



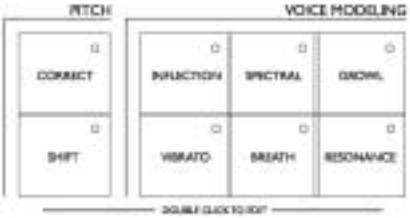
The VoiceOne is a powerful production tool that offers a unique range of voice-specific pitch shifting and VoiceModeling algorithms designed to open up a whole new world of possibilities for enhancing, harmonizing, transposing, correcting and otherwise transforming vocal performances. Using VoiceOne's next generation VoiceModeling algorithms you can sculpt a vocal to have unique elasticity, breathiness, growl, rasp and resonant characteristics. Ideal applications for VoiceOne include: adding strength, resonance and overall character to weak vocal performances; adding breathiness and more head-tone to a darker voice; re-pitch or pitch-correct an existing vocal while maintaining the natural formants; re-phrase an existing vocal to change the timing and feel of a vocal line; create convincing harmony and doubling; create new and original vocal effects. Vocal modeling effects can be manipulated in realtime allowing you to enhance particular notes or phrases and create lush and vibrant background vocals from a single solo performance. 24-bit processing and A-to-D, D-to-A converters as well as XLR balanced analog and S/PDIF digital I/O ensure that the integrity of the source material will be maintained throughout the signal path.

FEATURES

- Combines superior voice shifting and correction (±2 octaves) with award-winning, second generation VoiceModeling technology to provide an amazingly creative tool for processing vocals.
- The vocal-trained (voice-specific) **Pitch Recognition** algorithm employs a highly stable pitch detector that provides extremely accurate handling of vowels, hard consonants and sibilants thus ensuring that shifted vocals stay natural.
- **Intelligent harmonization algorithms** – harmonies can be controlled via MIDI, internal scale, or you can custom write your own scales.
- **Hybrid Pitch shifting algorithm** combines standard 'instrument' pitch shifting (suitable for small intervals) with formant-corrected pitch shifting which is more apt at preserving the voice characteristics over large intervals. Combining the two provides unlimited flexibility and control for achieving both natural and supernatural sounds.
- The enhanced pitch recognition is also a key component to VoiceOne's superior Pitch Correction providing fewer errors and artifacts and faster results when fixing intonation problems. Using the adjustable pitch correction/ re-pitching window (0-600/infinite cents), and the hybrid shifter, you can correct and even re-pitch vocal lines keeping them natural sounding or extreme – even over wide intervals.

Next Generation VoiceModeling

- Next-generation advanced VoiceModeling allows yo to enhance or even transform vocals using a variety of algorithms
- The **Flextime Inflection** algorithm seamlessly stretches and shrinks the pitch and time of a vocal line (up to 500ms) allowing it to breathe. Ideal for doubling and harmonies as well as providing you with control for radically altering phrases.
- The independently adjustable **Breath** and **Growl** algorithms provide a wide range of natural to extreme sounds.
- The **Real Vibrato** algorithm features controls and styles based on real singers from all styles of music
- The **Spectral** algorithm is a Voice Optimized Equalizer that allows you to intelligently dial up a high-boost curve for a vocal without boosting hard consonants and esses.
- The **Resonance** algorithm allows you to create, remove or enhance chest tones and head tones, as well as set the apparent length of the vocal tract.



- Custom Main LCD display giving feedback on key data with a flexible and simple to use parameter editing interface
- 100 Factory/50 User presets
- Balanced XLR inputs and outputs with 24 bit A-to-D and D-to-A converters with 24 bit internal processing driven by a powerful 100 million instruction per second DSP engine.
- S/PDIF digital I/O is also included for a pure all digital, 24-bit signal path supporting both 44.1K and 48K rates.
- MIDI In/Out/Thru provides pitch to MIDI conversion as well as extensive MIDI control of effect parameters echoing most front panel functions
- The 1/4″ footswitch connector supports momentary single or three-button external footswitches for remote control of select parameters

VOICEWORKS

Harmony, Pitch Correction and Effects Processor



The VoiceWorks combines multi-voice harmonies, fat automatic double-tracking and transparent pitch correction, along with superb TC Electronic effects – all in a one rack space unit. VoiceWorks utilize TC Helicon's state-of-the-art humanizing and HybridShift functions that give you complete control over the sound and allow you to create natural sounding harmonies that sound authentic even when soloed. VoiceWorks also employs TC Helicon's latest innovation the HarmonyHold function which lets you sustain back-up harmonies while you sing over them. The critically acclaimed TC Electronic Compression, EQ, Tap-Tempo Delay and Reverbs allow you to add that finishing touch to your processed vocals. Audio can be input to the VoiceWorks via the studio-quality mic pre-amp, balanced/unbalanced line input or the coaxial 24-bit S/PDIF digital input. The processed signal can be output via either the coaxial S/PDIF digital output or the balanced/unbalanced 1/4″ line outputs. Full MIDI implementation allows you to control virtually every parameter of the VoiceWorks remotely including the ability to automate parameters from your favorite sequencing application. Designed for live and recording situations – anywhere superior vocal processing is required.

- HybridShift harmony generation for smooth, natural sound
- Four independent harmony voices accessible from four distinct harmony Modes: Scalic, Chordal, Shift and MIDI Notes.
- 2 Lead-voice Thickening Voices - one voice above and one voice below
- Humanization of harmony voices utilizing: FlexTime, Vibrato, Inflection, Pitch randomization and Portamento
- Selectable Equal-tempered and Just tuning for harmonies
- HarmonyHold lets you freeze harmonies while continuing to sing over them.
- Fully adjustable gender and voicing per voice

Pitch Correction

- The built-in scale-based Pitch Correction function listens to the pitch of the incoming voice, compares it to the selected correction scale, and then applies pitch correction in realtime. Besides scale type, you have control over the amount of pitch correction to be applied, attack (or speed) at which pitch correction is activated as well as a window parameter which controls how out of tune a note needs to be before pitch correction is applied.
- Suitable for studio and live performance.

Presets

- 100 presets are provided as a starting point. These presets can be easily customized and overwritten.
- Song Mode for live performance allows you to save internal "sequences" of up to 50 songs with 30 user presets per song.

Effects

- The flexible TC Electronic 3-band EQ provides frequency adjustable high and low shelving bands as well as a fully parametric band with Q frequency and Gain controls. EQ can be applied to the Lead, Harmony or both the Harmony and Lead channels.
- The TC Electronic Dynamics section allows you to apply compression and gating to the Harmony, Lead or both the Harmony and Lead. The compressor has a variable threshold with a range of 0 to -60 dB and a ratio variable from 1.1:1 to 64:1. The compressor also features an auto makeup gain circuit. The gate has a variable threshold with a range of -70dB to 0dB.
- The TC Electronic effects section provides a variety of Reverb and Tap-tempo Delay processing. Individual aux sends for each effect allow you to control the amount of each effect for the Harmony and Lead channels independently.

Mic Input

- High quality mic preamp with 48V phantom power, variable gain and a 20 dB pad.

Inputs and Outputs

- 24-bit A-to-D and D-to-A converters
- Balanced XLR mic input and a 1/4″ TRS balanced/unbalanced line level input.
- Balanced left and right 1/4″ TRS outputs
- Coaxial S/PDIF digital I/O
- MIDI In, Out and Thru provides access to MIDI CC control over all parameters
- The 1/4″ Footswitch input is compatible with both single and 3-button footswitches (including TC Helicon's Switch-3) and provides remote control of Bypass, Harmony Mute, Preset Up/Down and more.



Additional Features

- Easy to read and comprehensive 16 character by 2 line backlit LCD display
- 5-segment LED level meter as well as independent input and output clip LEDs.

TL AUDIO

5001

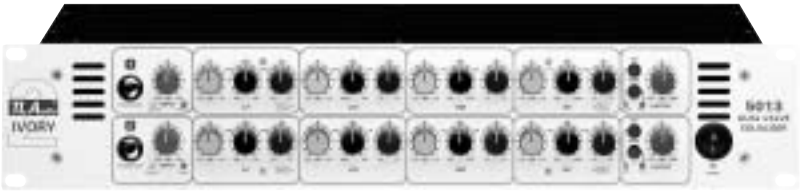
4-Channel Tube Mic Preamp



In addition to offering four top quality tube mic preamps in a single 2U package, the 5001 also features a discrete mic preamp stage, 30dB pad, instrument DI inputs, improved metering and optional 24-bit digital output. The combination of sound and facilities is guaranteed to bring out the best in any microphone. Separate input and output level controls allow the tube stages to be driven harder if necessary for increased warmth, while still allowing the overall output level to be regulated - absolutely essential if recording direct to tape. The output fader permits anything from complete attenuation through to +15dB of extra output gain - making it ideal for driving today's high level digital recorders.

- Continuously variable input and output gain controls
- 48V phantom power, phase reverse switch
- 30dB mic pad; 90Hz low cut filter
- Drive & Signal level LEDs assist with gain setting
- Instrument DI inputs
- Balanced XLR mic input
- Balanced XLR line output (+4dB) duplicated on unbalanced jack connector (-10dB)
- Optional four channel 24-bit digital output

5013 *2-Channel Tube Parametric EQ*



Offering the finest combination of performance and affordability, there is simply no EQ in this price range to touch it. And now the 5013 features a peaking/ shelving option on both LF and HF bands plus a 'Fat' contour switch on both channels. Like all TL Audio equipment, the interfacing options couldn't be more flexible - separate input and output level controls allow easy level matching with any other equipment, and a choice of balanced and unbalanced line inputs/outputs plus instrument DI inputs means you can get yourself up and running in seconds.

- Continuously variable input and output gain controls (-20dB to +20dB)
- Four bands of fully parametric EQ per channel: LF: 30Hz to 1kHz, LM: 100Hz to 3kHz, HM: 1kHz to 12kHz, HF: 3kHz to 20kHz
- EQ bypass switch on both channels
- Each band offers 15dB of cut or boost and variable Q (bandwidth) between 0.5 and 5
- Shelving mode available on LF and HF bands (by pulling out Q control)
- Fat' contour switch provides a gentle LF/HF lift combined with a mid cut, for extra warmth
- Front panel instrument input
- Drive and Peak LEDs
- Balanced XLR line inputs/outputs duplicated on unbalanced jack connectors
- Optional stereo 24-bit digital output

5021 *2-Channel Tube Compressor*



The 5021 provides first class audio quality, lots of flexibility and a price tag so modest, you'll think it was a mistake. Features include hard and soft knee modes, four attack and release times, an improved optical gate design and an optional digital output. It also uses TL Audio's own special transconductance amplifier design that avoids the use of VCAs - ensuring maximum audio transparency.

5021

- Each channel provides continuously variable control of: Input Gain, Threshold, Ratio, Gain Make-Up and Output Gain
- Four attack times: from 'Fast' (0.5mS) to 'Slow' (40mS)
- Four release times: from 'Fast' (40mS) to 'Slow' (4S)
- Hard/soft knee compression modes
- Automatic 'Hold' facility reduces LF distortion
- Optical gate with variable control of threshold (from -10dB to 'off')
- 'Bypass' and 'Stereo link' modes
- Switchable VU meters provide indication of output level or gain reduction
- Drive and Peak LEDs
- Front panel instrument inputs
- Balanced XLR line inputs/outputs duplicated on unbalanced jack connectors
- Sidechain insert points
- Optional 24 bit stereo digital output

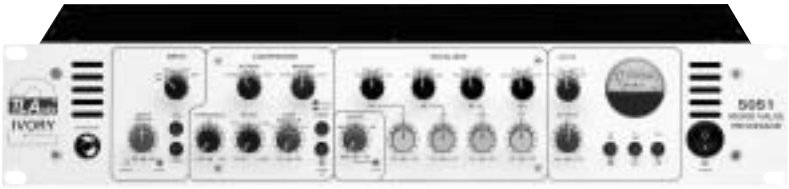
5050 *Tube Mic Preamp and Compressor*



A simple yet great sounding tube front end, the 5050 boasts the winning combination of a mic/line/instrument preamp and mono compressor– making it the most cost-effective way of processing your sources. Features include a 30dB pad, hard/soft knee compression modes, a fully balanced line output and an optional digital output.

- The Preamp stage provides variable input gain, +48v phantom power, 30dB mic pad and 90Hz low cut filter
- Compressor section provides control of Threshold, Ratio, Gain Make-Up and Bypass
- Automatic "Hold" facility reduces LF compressor distortion
- Attack time switchable between 'Fast' (0.5mS) and 'Slow' (20mS)
- Release time switchable between 'Fast' (40mS) and 'Slow' (2S)
- Hard/soft knee compression modes
- Output fader permits complete attenuation through to +15dB of extra output gain
- Two 8-segment LED meters provide indication of output level & gain reduction
- Front panel instrument DI input
- Balanced XLR mic input
- Balanced 1/4″ line input/output
- Optional 24-bit digital output

5051 *Tube Channel Strip*



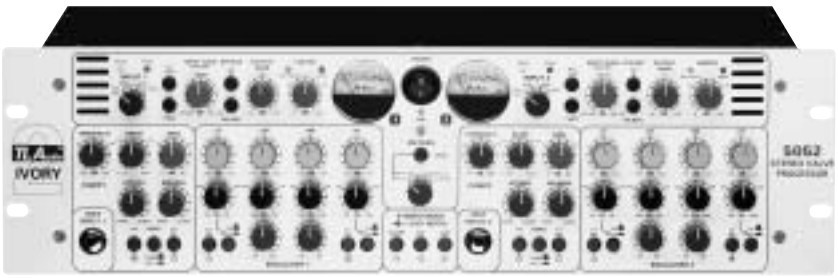
The single channel voice processor format is a great way to get the most out of any sound source. The 5051 provides the perfect front end for any system by offering a combined preamp, compressor, EQ and noise gate in one easy-to-use package, plus an optional digital output. The 5051 compressor section is identical to that of the 5021 and includes hard/soft knee modes, and the four band EQ section is modelled after the legendary TL Audio EQ-1 —yielding superb results instantly, with an improved optical gate for removing unwanted source noise.

- Preamp stage: input gain, +48v phantom power, 30dB mic pad and 90Hz low cut filter
- Compressor section: control of Threshold, Ratio, Gain Make-Up and Output Gain
- Hard/soft knee compression modes
- Four switchable Attack times: 0.5mS to 40mS
- Four switchable Release times: 40mS to 4S
- Optical gate with variable control of Threshold (from -10dB to 'off')
- Instrument input; Balanced XLR mic input
- LF EQ band gives ± 12dB shelving at 60Hz, 120Hz, 250Hz or 500Hz (12dB/octave)
- LM EQ band gives ±12dB peaking at 250Hz, 500Hz, 1kHz or 2.2kHz (Q=0.5)
- HM EQ band gives ±12dB peaking at 1.5kHz, 2.2kHz, 3.6kHz or 5kHz (Q=0.5)
- HF EQ band gives ±12dB shelving at 2.2kHz, 5kHz, 8kHz or 12kHz (12dB/octave)
- Switchable VU metering of input/output level or gain reduction; Drive and Peak LEDs
- 'EQ Pre' switch allows the EQ section to be placed ahead of the compressor section.
- 'Link' switch for stereo operation (two 5051s)
- Balanced XLR line input/output duplicated on unbalanced jack connectors
- Sidechain insert point
- Optional 24 bit digital output

TL AUDIO

5052

Dual Tube Channel Strip



The 5052 provides everything you would expect from a high quality 'channel strip' product, and then doubles it - making it the ideal system front end and the perfect stereo mixdown and mastering device. While most channel strip products provide a single mono channel made up of preamp, dynamics and EQ stages, the 5052 provides two channels of each with independent stereo linking of the compressor, EQ and limiter sections. Recording a stereo source to hard disc through the 5052 is simplicity itself, yet when it comes to mixdown the 5052 come into its own by offering full stereo linked EQ and dynamics, allowing unparalleled processing of the stereo mix buss signal while recording to 2-track.

Each channel of the 5052 boasts a tube preamp stage, a tube compressor, a four-band tube EQ section and a peak optical output limiter, with the following features:

- Preamp stage accepts mic, line and instrument inputs with variable input gain, 90Hz filter, 30dB pad and phase reverse.
- Compressor section offers variable threshold, ratio, attack, release and gain make-up controls, with hard and soft knee modes.
- Tube EQ section features swept LF and HF bands switchable between peaking and shelving modes, and two fully parametric mids. The EQ can be switched 'pre' or 'post' compressor, and can also be switched into the compressor sidechain for frequency conscious compression.
- Optical output limiter provides transparent brick wall limiting of the output signal, and provides a variable threshold control.
- VU metering is provided for input/output levels, and the gain reduction of the compressor and limiter stages can be separately monitored. A '+10dB' setting attenuates the meter reading to allow high input/output levels to be accurately tracked.
- The compressor, EQ and limiter stages can be individually stereo linked (channel A becomes the master, channel B the slave).
- Has an output level control, and a 'system bypass' switch allows A-B comparison of the original and processed signals.
- Full connectivity is built in, with balanced mic & line ins/outs (duplicated on unbalanced connectors), balanced inserts, compressor sidechain inserts and front panel instrument inputs.
- Optional DO-2 digital card provides 24-bit A-D conversion of the unit's main outputs, with switchable 44.1/48kHz sample rate and a word clock input.

VP-1 *Tube Channel Strip*



For those requiring an even higher quality channel strip, the VP-1 combines features from the TL Audio's classic PA-1 pentode preamp, C-1 compressor and EQ-2 parametric EQ. The VP-1 also contains an expander/gate, a de-esser and an optical peak limiter. The expander/gate, de-esser, compressor and EQ sections are all bypassable for easy A/B comparison. An optional 24-bit/96kHz digital output is also available.

Preamp stage accepts mic, instrument and line level signals. The mic input is transformer balanced and offers a choice of tube or class A signal paths. Switchable phantom power, phase reverse, 30dB pad and variable high pass filter are provided. Input gain is variable in 10dB steps with a 15dB trim control, and Drive and Peak LEDs indicate the level of tube drive and clipping respectively.

Expander/Gate offers precise and responsive VCA control of signal dynamics, and features fully variable adjustment of threshold, attack and release. A four segment bar graph meter indicates the degree of gain reduction.

De-esser employs high quality VCAs to allow effective but natural reduction of sibilance. De-esser depth, frequency and bandwidth are all fully variable. There is a bypass switch and status LED. Four segment bar graph meter indicates the amount of de-esser gain reduction.

Super-smooth compressor stage comes with both tube and optical compression circuits, and offers control of threshold, ratio, attack, release and gain make-up. Both hard and soft knee characteristics are provided, and a switchable 'Hold' circuit reduces LF distortion.

Sweetness and warmth comes via the 4-band tube EQ section, which utilizes four frequency LF/HF shelving as per the original EQ-1 equalizer, with two fully parametric mids identical to the EQ-2. The EQ can be switched 'pre' or 'post' the compressor stage.

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PRECISION 8

8-Channel Precision Mic Preamp with M-S Decoding



With the emphasis shifting to digital audio platforms and modular configurations, software "plug-ins" are replacing many outboard signal processors. Other than the microphone itself, the mic preamp is the most significant analog device remaining in the signal chain. Designed to provide the detailed, transparent sonic performance necessary for the highest quality direct tracking and live sound applications, Precision 8, with its sonically pure design, simple-to-use features, and unparalleled cost/performance value, is the ultimate solution for state-of-the-art digital recording studios and project studios.
Use as a complete input system for MDM's, HDR's, and DAW's, Precision 8 is also ideal for location recording due to its 8-channel single rack unit format, smooth continuous gain controls, useful level metering, and easy interconnect. The integral M-S decoder provides creative spacial image control that is particularly useful in attaining an exciting stereo image with minimum effort and microphone repositioning.

FEATURES

- Eight highly transparent solid state mic preamps in a 1 unit rack configuration
- Challenges the recognized leaders in sonic accuracy and performance
- Two sonically accurate discrete FET instrument inputs (DI's) are available for recording or live sound applications in which realistic, unaffected sound quality is desired.
- Built-in M-S (Mid-Side) decoding for creative spacial image control
- Smooth continuous gain controls
- 5-segment peak-hold level metering with selectable peak reference allows rapid optimization of program levels between the Precision 8 and devices to which it is interfaced.
- Easy interface to MDM's, HDR's, DAW's, or consoles
- Dual DB25/TRS outputs for flexible interconnection
- Elegant, sturdy physical design
- Exceptional cost per channel value

P2ANALOG *2-Channel Microphone Preamp*



A 2-channel microphone preamp designed to provide detailed, transparent sonic performance for the most critical direct tracking and sound reinforcement applications, the P2ANALOG features selectable M-S decoder, active DIs with impedance modification capability, dual gain range selection, selectable high-pass filters, and relay-switched signal routing. It also offers a Stereo Phase Correlation display that assists in optimizing microphone placement for stereo recording.

Modeled after the Precision 8, the P2ANALOG's unique combination of functions make it a complete input system for standalone or PC-based recording systems. Hybrid, totally balanced, dual servo, high-voltage design provides top-notch transient response, headroom, imaging and noise performance. In addition to the acclaimed mic inputs, two instrument direct inputs (DIs) offer incredible articulation and control for electric bass, detail and smoothness for stringed instruments and punchy, clean sound with keyboards.

	Precision 8	P2ANALOG
Gain (microphone)	+16 to +64 dB	3.5 to 52dB and 15.5 to 64dB
Gain (direct input)	-4 to +44 dB	-16 to 32dB and -4 to 44dB
Frequency Response (gain=40 dB)	1.5 Hz to 500 kHz (+0/-3 dB)	1.5 Hz to 500kHz (-3dB)(gain=40dB)
Maximum Input/Output Level	+15/31 dBu	+15/31 dBu
Input Impedance	5.5 kΩ(mic); 2 MΩ (direct input)	5.5 kΩ(mic); 2.5 MΩ (direct input)
Noise (Rs=0 Ohms)	-132 dB e.i.n.	-132 dB e.i.n.
Slew Rate	> 40 V/uS	> 40 V/uS
CMRR (CMV=+10 dBu):	85 dB	85 dB
Crosstalk	-130 dB	
THD (+26 dBu, 100 kOhm):	.0008%	.0008%

L2 – Ultramaximizer

24-bit / 96kHz Hardware Limiter/Converter



The L2 Ultramaximizer is a 2U 19˝ rackmountable limiter and converter that combines refined look-ahead brick wall peak limiter algorithms and IDR (Increased Digital Resolution) dithering technology from the award-winning L1 software with 24-bit A-to-D and D-to-A converters, 48-bit internal processing and support for sample rates up to 96kHz. The L2's balanced XLR and RCA analog inputs and outputs, AES/EBU and S/PDIF digital I/O and a dedicated Word Clock input makes it compatible with a wide variety of audio applications, from tracking and mixing to mastering and concert sound. The L2's limiting capability can significantly increase the average signal level of typical audio signals without introducing audible side effects but is equally capable of extreme limiting, intentional pumping, and vintage dynamic processing effects. Additionally, L2's Increased Digital Resolution word length reduction system uses ninth-order noise shaping which can increase the perceived sound by as much as 18dB when dithering down from high resolution formats.

FEATURES

- Peak Limiter with 48-bit internal processing significantly increases the average signal level of typical audio signals without introducing audible side effects.
- Supports 44.1, 48 kHz, and x2 88.2 and 96 kHz sample rates at 24, 22, 20, 18 or 16 bit resolution.
- Linked stereo and dual mono operation.

Inputs and Outputs

- 24-bit balanced XLR & unbalanced RCA analog inputs and outputs
- AES/EBU (XLR) and coaxial S/PDIF digital inputs and outputs
- Digital sync to AES/EBU, S/PDIF or from the dedicated Word Clock input (BNC)

L2 Limiter

- The use of look ahead technology allows the L2 to anticipate peaks before they happen, thereby minimizing the possibility of artifacts and can be used with absolute confidence in situations where brick wall limiting is critical.
- Dedicated front panel rotary controls accompanied by numeric displays include:
  – Threshold with a range of -30.0 to 0.0dB in 0.1dB steps.
  – Output Ceiling with a range of -30.0 to 0.0dB, in 0.1dB steps.
  – Variable Release Time (ARC control bypassed) with a range of 0.01 to 1000ms (logarithmic)
- ARC (Auto Release Control) dynamically controls release times to fit the human ear's expectations. This allows a greater amount of limiting and level maximizing without distracting artifacts.
- Dedicated bargraph meters for input, output and attenuation with infinite peak hold and peak reset buttons are also provided.

IDR Requantize, Dither and NoiseShaping System

- IDR (Increased Digital Resolution) is Waves' proprietary word length-reduction (quantization), dither and noise shaping technology which preserves and even increases the resolution of digital signals. For example, 16 and 20-bit masters can have the perceived resolution of 20 and 24-bit signals, respectively.
- 24-bit data can be re-quantized to 22, 20, 18, and 16-bits.
- There are two types of Dither available:
  – Type 1 provides no nonlinear distortion;
  – Type 2 exhibits lower dither (hiss) level;
  – Dither can also be switched off.
- Noise shaping options include: Moderate; Normal; Ultra; and off.

Applications

- For mastering applications, the combined effort of the L2's Ultramaximizer and IDR processing enhances the perceived sound quality by as much as 24dB.
- The limiter eliminates the danger of clipping when recording hotter signals – ideal for tracking and mixing as well as live sound or broadcasting.
- Requantize 24-bit input signals to 16-bit or 20-bit signal suitable for a wide array of today's recording equipment, such as DAT, 20-bit MO, CDR, etc.
- Additional applications include:
  – A-to-D and D-to-D mastering;
  – Digital recording 16 bit dithered / 24bit
  – Analog limiter insert;
  – Digital limiter insert,
  – A-to-D and D-to-A conversion

Technical Specifications:

Analog to Digital:	
Frequency Response:	20 to 20k Hz (+0/- 0.01 dB)
Noise:	-112 dBFS (18.3 bits)
THD: 3rd Harmonic	-130dBFS, (@1kHz, -10dBFS)
Analog to Analog:	
THD just below Clipping:	< or = 0.0060%
Crosstalk:	99.5 dB (with full scale 1kHz source)
96kHz Freq. Response:	20 to 24.6k Hz (+0/-.5 dB) -3 dB @ 45.2 kHz
44.1kHz Freq. Response:	20 to 20k Hz (+0/-.3 dB) -3 dB @ 21.2 kHz

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MAXXBASS 101 / MAXXBASS 102

Psychoacoustic Bass Extension Processors





MaxxBass 101 and MaxxBass 102 are bass enhancement hardware processors based on the MaxxBass software plug-in that has become a standard processing tool for computer based DAWs. Both MaxxBass processors can add up to one and one-half octaves of bass extension without adding any distortion or stress to your sound system. In just a few minutes the system can be tuned for optimum bass performance for a wide range of speaker types from desktop multimedia and museum installations to sophisticated club and house of worship sound systems. Both products provide a complete MaxxBass solution with a single set of input trim, frequency and intensity controls that adjust the left and right channels. The MaxxBass 101 is a 1U rack-mounting unit with balanced line level (+4dBu) analog I/O on XLR and TRS connectors and terminal blocks and the MaxxBass 102 is a half-rack unit with RCA connectors operating at -10 dBV.

FEATURES

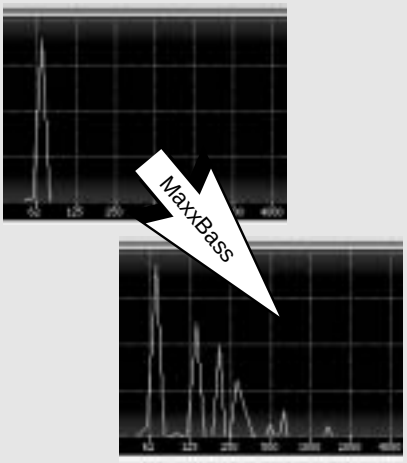
- The MaxxBass processor is inserted in your system's signal chain between the audio source and amplifier
- A single set of controls are provided controlling both the left and right channels
- An Input Trim knob is provided for attenuating an incoming audio signal that is too hot. The 101 features a clipping indicator for the left and right channels. The 102 adds a signal present indicators.
- The Frequency knob allows you to tune the MaxxBass processor to the specific speaker system being used over a range of 25 to 100 Hz.
- The Intensity knob affects the amount of effect being applied allowing you to tune in the optimum bass level for your specific speaker system. As the Intensity knob is raised, the relative (to the original source frequency) level of harmonics increases and their decay rate becomes slower and more effect is mixed into the signal.
- The Bypass button, with status LED, allows you to turn the MaxxBass processing on and off for easy A/B comparison
- Once you you have tuned your system with the parameter settings, you can disable the frequency and intensity controls and preserve the optimized settings using the system's fail-safe Lock button, accessed through a small pinhole. A Lock LED lights up to indicate that the unit is locked.
- The MaxxBass 101 is a 1U rack-mounting unit with balanced line level (+4dBu) analog I/O on XLR and TRS connectors and terminal blocks. The MaxxBass 102 is a half-rack unit with RCA connectors operating at -10 dBV
- Stereo analog inputs and outputs
- Both MaxxBass units are ideally suited for sound reinforcement and commercial sound installation applications including PA, clubs, houses of worship, restaurants, and retail stores, delivering richer, deeper bass without requiring larger, more sophisticated, or more expensive amplifiers and speakers.

MaxxBass Technology

MaxxBass is a patented psycho-acoustic bass frequency extension technology that provides perceived bass frequency response below the physical speaker cutoff in an audio system. MaxxBass utilizes the principle of the missing fundamental, which creates the sensation of low frequencies by generating a carefully calculated series of harmonics designed to simulate the auditory experience caused by the missing fundamental pitch. These harmonics extend the virtual frequency response up to 1½ octaves or two-thirds below the physical speaker cutoff frequency without perceived distortion or increased power consumption.

MaxxBass At Work

In the example below, MaxxBass is allowing the listener, using a multimedia speaker system, to perceive a 70 Hz tone even though the speaker is physically not capable of reproducing this frequency. The bass signal is analyzed by MaxxBass, which creates a carefully calculated series of harmonics of these low frequencies. The dynamics and the loudness of the original bass are duplicated in these harmonics to create the most natural-sounding enhancement of the original bass. High frequencies are merely passed to the output and added back to the signal.



YAMAHA

REV100

Digital Reverberation Unit with 99 Programs



The Yamaha REV100 is an affordable, true stereo effects processor that offers a selection of 99 editable reverb programs in a single rack space. It offers remarkably smooth, natural reverb sound as well as a range of other outstanding studio-quality effects including Stereo Reverb, Gated Reverb, Reverb plus Flanger, and Delays. The REV100 is designed to be super-simple to use — each of the 99 programs offers control of three parameters, including decay, delay time, and balance of wet and dry signal to allow you to build your own custom programs. The REV100 features dual 1/4˝ inputs and outputs and 16 bit A-to-D and D-to-A conversion at 44.1kHz. The REV100's MIDI In connector allows you to change effects programs using MIDI Program Change commands.

FEATURES

- Dedicated input level and wet/dry Mix controls.
- Peak LEDS make it easy to set the optimum input level.
- 99 high quality editable presets including true stereo reverb.
- The three most critical parameters for all effect programs can be adjustable via dedicated rotary panel controls
- Edited effects can be easily registered in memory via the stored button.
- Unbalanced 1/4˝ stereo (left and right) inputs and outputs.
- 16-bit A-to-D and D-to-A converters and a 44.1 kHz sampling rate provides full 20 Hz to 20 kHz frequency response with exceptionally low noise and distortion.
- Compact "1U" rack mount dimensions.
- Easy-to-read 2 digit LED display

Advanced MIDI Features

- You can select effect programs via MIDI rather than the panel controls. You can also use MIDI control change commands to actually change effect parameters.

99 Editable Presets - Exceptionally Easy Programming

The REV100 is basically a no-fuss signal processor - just select one of its 99 top-quality preset effects and you're ready to go. When you need something specially tailored to the occasion, however, the REV100 makes editing extra easy. The three most critical parameters for all effect programs are adjustable via rotary panel controls. No button-tapping to find or change settings - just rotate the appropriate controls until the sound is right. With reverb programs, for example, the panel controls give you direct access to the initial delay, reverb time, and high-frequency damping parameters - all you really need for full control. Input level and dry/wet balance controls are also provided. And when you create a sound you want to keep, just press the STORE button to register it in the REV100 memory. Of course, you can recall the original Yamaha presets whenever you like - either all at once or individually. Each of the parameter controls also has an LED that lights when the control setting matches the preset parameter value. The presets include reverb, delay, and modulation effects, plus a range of combinations so if you need reverb and flanging, for example, it's right at your fingertips.

Effect Categories

- Reverb Programs 1 - 20 – Includes essential reverb effects such as realistic hall, room, and plate reverb simulations
- Stereo Reverb Programs 21 - 40 – True stereo reverb effects that let you add ambiance to stereo signals without sacrificing the stereo image
- Gate Reverb Programs 41 - 50 – A range of "gated" reverb effects that add warmth and ambiance while maintaining a tight sound
- Delay Programs 51 - 60 – A selection of mono and stereo delay programs including straightforward one-shot repeats and complex bounce effects.
- Delay/Reverb Programs 61 - 70 – Combinations of delay and reverb that can add a little more life to your sound than delay or reverb alone.
- Reverb/Modulation Programs 71 - 99 – Reverb combined with a range of modulation effects including flange, symphonic, chorus, and tremolo.

Specifications

Frequency Response	20 Hz-20 kHz
Dynamic Range	80 dB (typical)
Distortion	< 0.1% (1 kHz, max. level)

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REV500

Digital Reverberator





The REV500 is an affordable, high-quality, and easy-to-use digital reverb processor, well suited for home and project studio recording or live sound reinforcement applications. The REV500's true stereo processing capabilities delivers natural and spacious stereo reverb, as well as combinations of reverb with a variety of other effects. The 20-bit A-to-D and D-to-A converters and 32-bit internal processing ensure superb sound quality with a wide frequency response. The REV500 comes complete with 100 preset programs that exploit the potential of the high-performance DSP, plus there's room for 100 customized user programs. The REV500 is also flexible and simple to use. Both 1/4˝ phone and XLR input and output connectors are provided with the option of +4 dB input and output levels. Four rotary controls allow quick and easy editing of the basic program parameters. And MIDI In and Out terminals allow remote program selection, remote data storage, as well as realtime control and automation of program parameters.

FEATURES

True Stereo Processing

- 20-bit (64x oversampling) A-to-D and 20-bit (8x oversampling) D-to-A converters with a sampling rate of 44.1 kHz and the same 32-bit internal Digital Signal Processing (DSP) as the PROR3 provides superb sonic performance.
- True stereo reverb processing, with independent left and right channels.

Organized Program Libraries

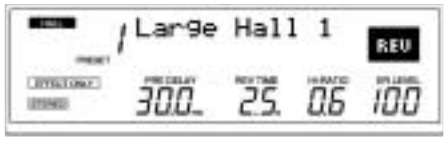
- 100 Preset programs and 100 User programs are available divided into four easy to find categories of 25 presets each:

  Hall – Small, Medium and Large Halls, Churches, Concert Halls, Reverbs w/ Chorus and Flange, Gated Reverbs, Slapback and more.

  Room – Small, Medium and Large Rooms, Bathroom, Garage, Bright, Ambience, Tunnels, Opera, Cathedral, Live, Live Gate, Wood and more.

  Plate – Basic, LA Plate, Percussion, Long, Short, Vocal, Delayed Spring, Live, Cave, Fat, Thin, Light, Gated Plate, Plates w/ Symphonic, Chorus and Flange.

  Special – Variations of Flanging, Chorus, Dynamic Filters, Tremolo and Echo with Reverb.



Intuitive User Interface

- An input level control accompanied by two four-segment LED level meters with clip indicators simplify level setting for optimal performance.
- A large custom multi-purpose LCD display provides information about the currently selected program and basic parameters being edited, as well as other messages related to utility functions.
- Dedicated rotary controls provide realtime control of the four most critical reverb parameters – Pre Delay, Reverb Time, Hi Ratio and Early Reflection Level.
- Internal editing parameters, specific to each effect algorithm, are easily accessed via the front panel Utility button and the four rotary controls.

Inputs and Outputs

- Left and Right balanced XLR and 1/4˝ inputs and outputs allow simple hookup to pro or semipro gear.
- Input and output levels for each channel are independently switchable between -10dB and +4dB.

MIDI

- The four basic parameter controls plus the output volume can be controlled in real-time via MIDI Control Changes.
- Program selection via MIDI is also supported as is MIDI Bulk Dump for user program backup.
- An optional footswitch with a 1/4˝ connector can be used to mute the reverb effect or advance through reverb programs.

Conveniences

- An Audition button to triggers a built-in snare and rim shot sample allowing you to instantly hear the effect of programs and edits directly from the front panel.

Specifications	
Frequency Response	20 Hz to 20 kHz (+1.0 dB, -2.0 dB)
Dynamic Range	96 dB typical
Distortion Less Than	0.03% (1 kHz, max level)

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YAMAHA

PRO R3

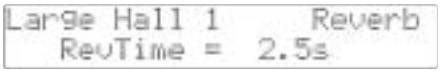
Digital Reverberator



The Yamaha ProR3 Digital Reverberator is a true stereo effects processor. It utilizes Yamaha's third-generation 32-bit digital signal processors (DSP) and high-performance 20 bit linear A-to-D and D-to-A converters to provide rich, high resolution reverbs, with breathtakingly dynamic impact. The high quality analog input and output circuitry achieves a wide dynamic range of 110 dB, providing clean and natural sounding reverb right down to the final tail. The ProR3 is divided in two processing stages. The first stage features a choice of 10 primary effects including reverb, early reflection, room simulation, and combinations of reverb with echo, chorus, flanging, pitch change, and auto pan. The secondary stage features a chain of additional processors for fine tuning the primary effect. These include dynamic filter, compressor, EQ, gate, level, and balance stages. A pre-effect EQ is also provided for shaping the input signal. The ProR3 ships with 90 preset programs with room for an additional 90 user programs for storing customized settings. Both balanced XLR and 1/4˝ inputs and outputs are provided for connection to a variety of equipment. The ProR3 is an ideal choice as main or secondary reverb for any professional application.

FEATURES

- Third generation 32-bit DSP power has been coupled with 20-bit linear A-to-D and D-to-A converters to deliver unprecedented density, superior resolution and astonishing impact with a noiseless decay and a dynamic range of 110dB.





90 Presets and 90 User Memory Locations

- 90 factory presets are provided covering a wide range of applications – presets can be fine-tuned or completely re-programmed to create totally new effects. Edited effects can be stored in any of 90 user memory locations and instantly recalled.

Inputs and Outputs

- Electronically-balanced XLR and 1/4˝ input and output connectors – -10dB/+4dB level selectors on both the ins and outs provide level-matching flexibility.

Effects Stages

- Programs consists of two main stages; the primary effect and the secondary effects.
- Primary effects can be selected from one of the ten types divided into single effects and compound (multi) effect types, depending on the program that you select.

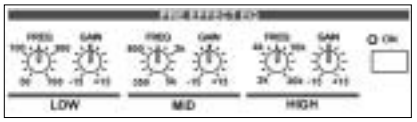
  Single effect types include: Reverb, Early Reflection and Room Simulator
  Compound effects types include: Reverb+Echo, Reverb+Early Reflection, Reverb+Chorus, Reverb+Symphonic, Reverb+Flange, Reverb+Pitch, Reverb+Pan – you have control over the order in which compound effects occur (i.e. whether reverb comes before or after echo
- There are 5 secondary effects provided for each program designed to shape and optimize the potential of the primary effects: DFL (dynamic filter), COMP (compressor), EQ, GATE, and LEVEL/BAL.

Dynamic Filter

- Versatile dynamic filter features low-pass, high-pass and band-pass modes with programmable center frequency, resonance, sensitivity, decay and shift direction.

Digital Pre-and Post-Reverb EQ

- 3-band sweepable pre-verb EQ controls with ±15db boost/cut are available right on the front panel for fast, easy access.
- The programmable Post-reverb EQ stage offers even greater control and versatility – 3 sweepable bands with a much broader frequency range, ±15db boost/cut and variable Q (bandwidth) parameters. The low and high bands can be each switched for shelving or peaking response.



Built-In Dynamics

- The noise gate not only provides a simple threshold control, it also features a programmable trigger delay (0.1 to 100 ms), as well as adjustable hold and release times. Level detection can be carried out at the gate itself or at the pre-reverb signal – the gate can also be triggered via MIDI.
- A precision compressor is provided with variable threshold, output level, ratio, attack and release time parameters as well as selectable knee (soft, normal or hard).

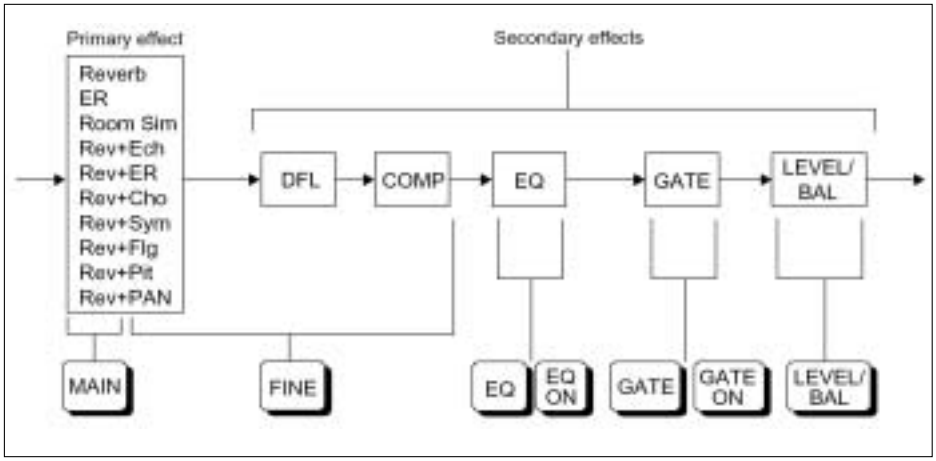
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PRO R3

User Interface

- The large 24-character x 2-line backlit LCD displays multiple related parameters at once for easy programming and operation including the name of the selected program and the program parameter values
- A dual concentric knob allows you to adjust the input level independently for the left and right channels
- Two 8 segment LED meters indicate accurate, pre A-to-D input levels, from -36dB to Clip, for each channel
- A two-digit seven-segment display indicates the currently selected program number – two LEDs indicate whether programs are loaded from Preset or User locations
- The Parameter keys provide direct access to the current effects' parameters, the post-EQ and Gate processors, level/balance parameters and MIDI controller assignments. Each time a key is pressed, it will cycle through the parameter pages
- The Data keys (INC) (DEC) allow you to modify the value of the selected parameter while the Cursor keys are used for navigating to parameters displayed in the LCD
- The Numeric keypad is used to directly input the a program number for recall, or direct input of parameter values
- The Main key accesses an effects most common parameters (Rev Time, Pre Delay and Hi and Lo Ratio), while the Fine Key accesses a program's more detailed parameters like density and liveness etc...
- The Compare function allows instant A/B comparison between the original and an edited sound
- The Infinite key, when pressed, will dramatically lengthen the RevTime parameter producing the effect of a "freeze-frame" or "stop-motion" sound
- MIDI In/Out/Thru connectors support program change and program dump – up to 2 different effect parameters can also be controlled via MIDI in real time – if MIDI data is being received from an external device connected to the MIDI In connector the MIDI indicator will light
- There are four user-selectable input modes – Stereo, LR/Mix, R-Mono, L-Mono

Signal Flow of the ProR3's Primary and Secondary Effects Stages



Reverb Types

- Reverb parameters are based on 2-stage reverb processing designed to achieve extremely smooth, natural reverb and allows true stereo reverb if required. There are a total of 9 reverb algorithms available including: Small Hall, Large Hall, Vocal plate, Perc.Plate, Spring, Echo Room, Strings, Snare and Reverb Flange.
- Early Reflections parameters include 6 preset early reflection types and 4 "user" types that can be created with customized individual reflection levels and pan positions.
- Room Simulator parameters are essentially basic reverb programs, configured with a range of parameters which allow you to control the dimensions and characteristics of a simulated room.

Compound Effects

- All compound effects are available in three patches configurations: series effect to reverb, series reverb to effect and parallel effect with reverb.
- Echo + Reverb parameters are one of the most commonly combinations used in recording and live sound applications
- Early Reflections + Reverb Parameters are ideal for producing exceptionally natural-sounding ambience effects
- Chorus + Reverb; Symphonic + Reverb; Flange + Reverb Parameters combine outstanding reverbs with chorus, symphonic and flange type modulation, making it easy to add a subtle touch of animation to the reverb sound or produce wild effects
- Pitch Change + Reverb Parameters — Pitch change capability can be extremely useful for creating artificial harmonies, or adding a subtle extra touch of high or low end to the basic sound. The pitch change portion of this program allows two pitch variations to be created in addition to the original sound. The series patches allow the reverb sound to be pitch-changed, while the parallel patch allows the original sound to be pitch-changed independently from the reverb effect.
- Pan (Auto-Pan) + Reverb Parameters form a useful combination for many applications

Specifications

Frequency Response	20 Hz - 20 kHz (+1.0 dB, –1.5 dB)
Dynamic Range	110 dB (typical), not less than 104 dB
Hum and Noise	< –80 dB
Distortion	< 0.007% (1 kHz, maximum level)

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KS-T2000

Compact 3-channel Mixer



The KS-T2000 is a compact 3-channel mixer that is small and light enough to go anywhere. This product is made and guaranteed by Kamesan and is ideal for all audio applications in professional broadcast and film work. Camcorders are shrinking but that doesn't mean audio requirements are getting any less demanding. Despite being one-third smaller and about one pound lighter than any comparable mixer, the Kamesan KS-T2000 offers all the flexibility you'll need to engineer precise location sound. True 'Pack and Go' design. At first sight, a 3-in / 2-out configuration does not appear to be a very comprehensive mixer specification. But on location, ease-of-use and adaptability count as the most valued features. You need to interface with varied signal levels, provide multiple feeds, and cope with awkward working conditions. Attention to smallest design details in the KS-T2000 has resulted in a mixer that can be relied on to excel in every situation.

FEATURES

Accommodating Inputs

- 3 switchable XLR-balanced mic/line inputs with 48v and A-B (12V) phantom powering.
- A variable input sensitivity control adjusts the input amplifier gain to optimize its amplification for any level, from a hot condenser to a low output ribbon mic.
- Inputs will match a wide variation in line signal levels. This means that the normal physical position of the main channel faders can be 'trimmed' to be at 2/3 rotation – providing adequate rotation range to 'ride' actual variations in signal levels.
- Switchable 160Hz high pass filters eliminate most low-frequency noise problems.
- Separate left / right output switches route the signal to the main faders and outputs.

Versatile Outputs

- To achieve such a compact size, a single meter is provided though the mixer has full stereo capability.
- The transformer-balanced XLR outputs, can drive at a professional line level (+4dBm) or can be switched down to a low mic signal level. (-60dBm).
- Two RCA connectors for connection to a consumer camcorder's mic input.
- Low channel crosstalk and output isolation allow the mixer to be configured for multiple feeds as the situation demands.
- 1kHz tone oscillator provides a useful calibration signal.

Versatile Monitoring

- Both headphone output and metering can be switched to left or right outputs, or to a mono mix of both. When used for multi-mono applications, one of the main outputs can be designated as a separate live feed and monitored independently.

Exceptional Ease Of Use

- The side panel reveals just how convenient operation is. Only the three main and monitor faders have tall knobs. It is virtually impossible to 'knock' any of the other controls inadvertently during operation.
- Even the main faders have been located on the side of the mixer allowing the operator to set and forget – a useful feature when inexperienced operators use the mixer.
- Wide knob spacing means that the channel fader provides for an easy grip under all conditions.
- An all-weather case allows the mixer to be slung around the neck or shoulder and operated beneath a transparent cover under all conditions.
- The KS-T2000 runs five hours on four AA batteries. Higher voltages can provide higher output levels.
- Alternatively, the KS-T2000 will accept and regulate any external DC supply in the range of 10 to 15-volts.



KS-107 *Lip Checker*

A unique product, the KS-107 helps to track and eliminate broadcast signal time differences between audio and image signals and also multiple audio signals. At the receiving site, audio is easily captured via a microphone or direct line connection. Video bursts are detected directly from a monitor screen using an exclusive image probe, which is manually pointed at the flashing image on the monitor. The calculated delay is displayed on the unit's large LCD to an accuracy of within one millisecond, allowing the operator to make necessary adjustments.

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KS-342

4-Channel Compact Mixer

The KS-342 is a no compromise high-performance mixer that adapts to your needs—since no two location projects are ever the same, and often, they have demands beyond the traditional sound mixing techniques. Ideal for all audio application in professional broadcast and film work.



FEATURES

Input Facilities

- Selectable mic/line inputs providing 12- or 48-volt phantom power as required.
- Level trim at both sensitivities means the main channel faders can be set at optimum rotation for normal working.
- Massive 30dB pre-fader headroom plus continuously variable hi-pass filtering ensure signals never get out of hand.
- Channels can be ganged for true stereo operation and switched to the left or right mix busses.

Outputs

- Thumb-knob master level controls prevent accidental adjustment and the signal passes through switchable compressors before the final output stages.
- Fast attack, compound release circuitry provides a comfortable safety margin for distortion free operation.
- Independent as well as ganged settings ensure maximum flexibility for stereo and dual-mono setups. This design philosophy is carried through to the isolated outputs, which are not a luxury but a necessity in many situations. With a choice of levels (left/right independent), symmetry and connectors, four isolated two-channel feeds are provided.

Comprehensive Monitoring

- Backlit, jam-proof, level meters offer a clear visual indication of the main output signal.
- The headphone circuit can be switched to monitor any combination of the outputs in mono or stereo.
- Additionally, an indispensable PFL function, which appears on the right meter as well as at the headphone output, is included.

Small and Light and Expandable

- Smaller and lighter than any comparably featured mixer, the front and side panels are uncluttered and easy to use.
- The most-important channel faders feature latex knobs for all-temperature comfort and are easily gripped, even in gloves. Kamesan's trademark bar-grip knobs are used for all presets and never obstruct easy operation.
- Runs 8 hours on NP-1 battery in a quick-change compartment, which also accepts a battery pack using 8 AA batteries. Plus up to 16v DC can be applied to the mixer power circuit, which also provides an auxiliary regulated output for powering receivers, etc.
- The nature of location recording is that there will always be situations that require more channels or more processing. The KS-342 features a multi-way link socket that can be used to cascade a stereo input from another mixer without sacrificing channels.
- Secondly, a discrete connector and sturdy locks attach and integrate a range of specialized modules that expand capabilities instantly. Currently two variations are available; the KS-6001, a four channel sub mixer with identical features, and the KS-6002, a block of four parametric equalizers and compressors.

	KS-342	KS-T2000
INPUT		
Mic Input Level (3kΩ)	-70 ~-30dBm (Balanced)	-70 ~-30dBm (Balanced)
Line Input Level (600Ω)	-20 ~ +4dBm (Balanced)	-20 ~ +4dBm (Balanced)
Aux Input Level (10kΩ)	-20 ~ +4dBs (Balanced)	
Ext Mon Input Level		0dBs/600Ω x 2
Low Cut Filter	20 to 200Hz (Cont. Var/12dB/octave)	160Hz (12dB/octave)
Headroom	> 34dB (Pre-fader input)	> 30dB (Pre-fader input)
OUTPUT: Output level		
Main L/R (600Ω, Trans. Output)	Selectable +4/0/-20/-60dBm	Selectable +4/-60dBm
Sub L/R (600Ω Load, Elec. Bal.)	Selectable -20/-60dBm	–
Unbalanced Output	-60dBs (>10kΩ at full load)	-60dBs (600Ω ~10kΩ)
Monitor(Max Load, 8Ω, Stereo)	0dBs/50Ω	-10dBs/50Ω
OUTPUT: Maximum Output		
Main (600Ω Load, +4)	+24dBm (at 12V), +20dBm (at 8V)	+23dBm (at 6V)
Sub L/R (600Ω Load, -20)	+8dBm (at 12V)	
Monitor(50Ω Load)	+6dBs (at 12V)	+5dBs (at 6V)
AES/EBU Digital Output	Selectable 44.1/48/96 kHz (20 bit A-to-D)	
Compressor	No	Yes
AUDIO Frequency Response	30Hz~40kHz ±1.0dB	50Hz~15kHz ±0.5dB
Distortion (T.H.D)	<0.1% (+4dBm output 50Hz~15kHz)	<0.1% (+4dBm output 50Hz~15kHz)
POWER: Internal Battery	'AA' x 8 (BP-3/8 Case Included) NP-1 NiCd Battery (opt. Sigma case)	'AA' x 4 (Holder Included)
Operating Time (Continuous)	>8 hrs (Alkaline Batteries)	>8 hrs (Alkaline Batteries at 77°F)
Dimensions (WDH)	8.46 x 6.99 x 2.64″	6.26 x 5.67 x 1.85″
Weight (no case or batteries)	Approx 4.4 lbs.	Approx 2.2 lbs.

ALPHAMIX

Portable 4-Channel Audio Mixer



The AlphaMix contains all of the features required by today's most demanding ENG recordist. Do you regularly do two camera shoots, work with demanding producers or have complicated IFB feeds? No problem. The AlphaMix features four switchable mic/line inputs, 12T or 48v phantom power, low noise pre-amplifiers with continuously variable gain trims, active low cut filters, balanced line level outputs per input channel, pre-fader-listens and simple three LED meters per input channel.



The AlphaMix uses NP-1 style batteries and is equipped with selectable low battery monitoring. In addition, it is equipped with four Hirose 4-pin power output connectors that allow it to power up to four wireless receivers. And unlike other external battery distribution boxes, each of the power outputs is individually filtered for quiet wireless operation. The AlphaMix was also designed from day one to feed and monitor audio returns from two cameras. Outputs include special, custom made, dual isolated output transformers that are the largest in the industry. Finally, the AlphaMix is the only mixer to offer two-camera support, and it utilizes a custom-made peak reading LCD meter that emulates most of today's cameras. For further convenience, it is equipped with a boom pole mounted remote control that can be used to control the gain of the first input channel, thus allowing you to boom with both hands.

- Four line or mic level switchable inputs
- 12T and 48v phantom powering
- Active 12dB/octave low cut filters.
- Retractable gain trims on each pre-amp
- Retractable pan pots
- Individual balanced outputs per input channel
- Individual 3 LED meters per input channel
- Pre-fade listens per input channel
- Boom pole mounted remote control slide fader allows booming with two hands
- Custom made (properly sized) dual isolated output transformers
- Two Hirose 10 pin connectors for dual camera support, line/mic level switchable
- Two 3.5mm stereo auxiliary outputs, line or microphone level switchable
- Two XLR balanced outputs
- Output limiters, switchable separate or ganged
- Additional output that provides balanced mono feed
- Built in 4-way power distribution for wireless receivers
- Built in slate microphone
- Built in reference tone oscillator
- Built in battery monitoring, switchable calibration for all NP-1 battery types
- Switchable headphone monitoring left, stereo, right, mono, MS as well as tape returns from 2 cameras.
- Aircraft aluminum housing with epoxy powder coat finish.
- Lexan overlays with wear-proof subsurface silk-screening.
- Includes Porta Brace case and strap

MJR *2-Channel Mixer*



Perfect for use with today's small DV camcorders, DAT and Minidisc recorders, the PSC MJR Mixer offers the features of full size mixer in a small, portable format. The MJR Mixer contains two high quality mic preamps, each with switchable line or mic inputs, high pass filters, channel assignment pan switches, and 48v phantom power. It also offers a slate microphone, reference tone oscillator, headphone amplifier with tape/direct switch, sunlight readable LED meters and a multi-pin connector for use with PSC's BetaSnake cables. All of this housed in a rugged aircraft aluminum case barely 4.75 x 6˝ and weighing less than 1.5 lbs. In addition, the entire mixer can be mounted to the bottom of a small video camera via the supplied 1/4-20 thumb screw. And the whole mixer/camera combination can be mounted to any standard tripod as well via 1/4-20 threaded attachment point. Runs on a single 9-volt battery for more than five hours.

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M3

3-Channel ENG Audio Mixer

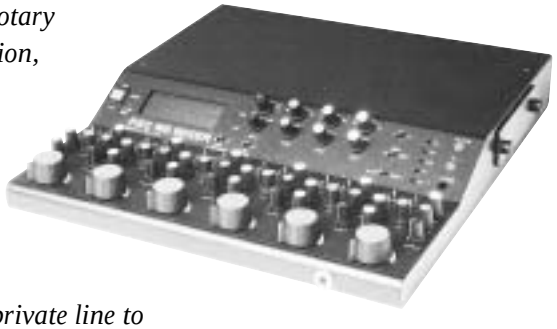


A cost effective, flexible portable mixer that combines rugged construction, high quality componentry, and ease of use ergonomics in a compact package, the M3 is perfect for ENG and EFP applications, digital and desktop recording and editing. The M3 features 3 high-quality, transformer-balanced XLR inputs and 2 (L/R) stereo paired XLR outputs capable of both mic and line level operation. Output limiters eliminate transient peaking caused by signal overload. The dual limiters can either be used independently or "gang" them together for applying the limiter to stereo signals. The M3 is powered via AA batteries offers 48-volt phantom power, and has a dual multi-pin connector for a single interface between the M3 and Betacam cameras. A convenient on board slate microphone is standard and a supplied boom pole remote control allows engineers the freedom of adjusting record levels while handling the boom pole.

- Three high quality transformer-balanced XLR inputs (with switchable "T", Dynamic or 48v phantom powering) and outputs for pristine audio with low noise. All inputs and outputs are mic/line switchable and have a 3-way input pad: Dynamic 0dB Pad, Condenser -15dB Pad, Line Level -50dB Pad.
- Additional outputs include two 10-pin Hirose connectors for use with Beta snake cables (mic or line switchable) for conveniently connecting to two cameras, and two Auxiliary outputs for transcription or RF feeds (mic or line switchable). The 10-pin connector provides audio signal flow as well as tape return signal flow for monitoring.
- Ganging input and output connectors allow you to use two M3 mixers tied together.
- Channels 3 and 4 can be linked for stereo recording using 2 mics in either M-S or X-Y configurations. The "Gang" switch links inputs 3 and 4 and both input levels are then controlled by Pot 4.
- "Separate" and "Ganged" switches allow independent or linked operation of both limiters.
- The LCD Peak meters emulate the peak reading attributes found in many Betacams. This ensures the meters accuracy no matter how the mixer and camera are connected.
- Low cut filters eliminate low frequency noise and rumble.
- The momentary "Slate" button activates a built in slate microphone for recording notes and information related to editing.
- Headphone monitoring selections include Left to both ears, Stereo, Right to both ears, Mono, Pre Fade Listen of channels 1,2 and 3 and MS Stereo as well as Tape returns from two Betacams.
- A handy boom-pole remote control attaches to any boom pole. This allows sound engineers to adjust recording level without reaching for the mixer.

M6

6-Channel Portable Mixer for Location Sound Recording



The M6 Mixer was designed for extreme field operating conditions (utilizes rotary pot faders because of their immunity to dirty and dusty conditions). In addition, it offers provisions for direct mounting of a DAT or DVD recorder. This allows for a simple and convenient means of recording in the field whether you are working on a sound stage, in an insert car, or at some other remote location.

The M6 Mixer offers six inputs each with switchable mic powering, line or mic inputs, phase reversal, 2-way EQ, pan pot and two aux sends. Its output section contains two large, easy to read LCD peak meters, slate microphone, private line to boom operators, full duplex boom communication, recorder remote rolls, reference oscillator, four selectable headphone feeds, two main outputs, two aux outputs, and a selectable video assist output.

DMX-P01

Digital Portable Field Mixer



Effective audio quality is now more important than ever for ENG and EFP applications, because even with high-quality video, lackluster audio performance will result in a mediocre production. The DMX-P01 is Sony's answer to these high-quality audio requirements. Not only does it offer outstanding audio quality, it is also one of the most user-friendly mixers in the ENG and EFP markets. With full 24-bit processing and 48kHz or 96kHz sampling rate, the DMX-P01 provides sound quality comparable to high-end production mixers. Its front panel is engineered to allow fast, easy, and accurate setting adjustments - essential when working in the field. It also includes other useful features, such as panel-lock and parameter-lock functions, selectable meter scales, camera-audio return-level check, memory function, digital cascade capability, and digital outputs. All of this functionality and versatility is packaged into one sleek, compact, and lightweight body that can be used effortlessly in field productions.

Digital Benefits of the DMX-P01

Excellent Sound Quality

- In order to provide outstanding sound quality, the DMX-P01 offers full digital-audio processing. Its 24-bit A/D and D/A converters provide a high level of linearity for analog inputs and outputs. Internally, the unit utilizes 32-bit digital processing for maximum throughput. In addition, the DMX-P01 has a sampling rate that is selectable from either 48kHz or 96kHz.

Digital Limiters/Compressors

- By using digital limiters and compressors, the DMX-P01 can provide extremely high-quality sound in one small package. Limiters are available at the input, and both limiters and compressors are available at the output for maximum flexibility.

Full Parameter Controls

- The DMX-P01's front panel puts all of its controls at your fingertips with an organized and logical layout. The DMX-P01 processes audio digitally, so parameters that are used less frequently are stored internally and accessed only when needed. Using the front-panel controls and easy-to-read backlit LCD allows full control of every parameter without the need to remove the unit from its audio-organizer case.

Panel-Lock

- One major concern for engineers in the field is the accidental bumping of controls, which can change the sound settings. The DMX-P01 safeguards against this with its panel-lock feature, which can be set to secure all of the control settings, or selected individual control settings.
- In addition, a Parameter-Lock feature also avoids inadvertent parameter changes.

Flexible Meter Scales

- Being digital, meter calibrations can be easily changed from one type to another without the need to replace the entire meter. Six easy-to-change meter scale sheets are supplied:
  VU, PPM1 (BBC-type), PPM2 (DIN-type), PPM3, PPM4 (IEC-type1) and dBFS.
  Simply insert the desired scale sheet and select the right meter type. The DMX-P01 will display the audio level according to the scale selected.

Camera-Audio Return Check

- The DMX-P01 enables users to visually verify that the mixer's audio level matches the level recorded to the camcorder tape. This is done using the camera return-level mode, available in the setup menu.

Digital Output

- The DMX-P01 is equipped with a digital output, which can be used to send audio to digital equipment such as DAT recorders. AES/EBU and S/PDIF coaxial interfaces are available.

Memory Function

- Easily store and recall parameters from the setup menu. There are two memory functions:
  – Power-On Memory Recall - When the DMX-P01 is powered on, the system can recall parameters in three ways: default factory settings, the last settings used, or with the parameters of one specific scene memory.
  – Scene Memory Recall - This allows users to recall up to ten different user-defined parameter settings or the factory default settings. Invaluable in situations where a single unit is required to serve multiple users or multiple shooting scenarios

Digital Cascade

- When additional inputs are required, DMX-P01 mixers can be cascaded using a digital connection between mixers. Best of all. sound quality isn't degraded, as it would be when cascading analog mixers.

DMX-P01



Side views of the DMX-P01



Inputs

- Four XLR-balanced mic/line inputs
- +48 V power for each mic input
- Digital cascade input with phono connector
- Microphone/line gain-level control
- Level control knobs with stereo-link facility
- Selectable sampling rate (48 kHz or 96 kHz) for A/D converters

Outputs

- 2 balanced outputs on XLR-type balanced connectors
- Digital AES/EBU output (stereo) on XLR-type balanced connector
- Coaxial output connector for mix-bus output (for cascade) or S/PDIF digital output (selectable)
- Stereo tape output on unbalanced 3.5 mm TRS jack
- Switchable output mode: stereo or monaural
- Selectable output-level control for L/R master outputs and camera send
- Selectable sampling rate (48 kHz or 96 kHz) for D/A converters

Panning

- Variable pan controls

Low Cut Filters

- Adjustable (50 to 200 Hz) cut-off frequencies for 2 user settings (A/B)
- Quick parameter-recall switch with OFF/A/B positions

Limiters/Compressors

- Digital limiters on both inputs and outputs
- Digital compressors on outputs
- Precise parameter control on threshold and ratio value, attack and release time
- Link function (ON/OFF switchable)
- LED indicators for output limiter/compressor operation

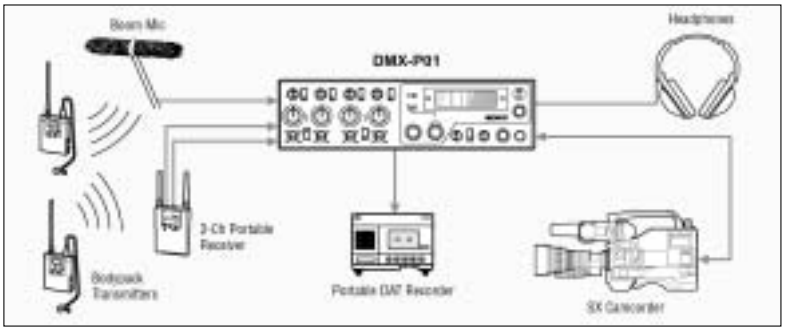
Link/M-S Operation

- Links input levels, LCFs, and PAN controls for channels 1/2 and 3/4
- Links output levels and limiter/compressor settings for master left/right outputs
- Decodes M-S microphone inputs, and links the input levels of channels 1/2 and 3/4
- Phase reverse on channels 2 and 4 (M-S decode)

LCD Panel

- Various level-meter displays: VU, PPM1 (BBC-type), PPM2 (DIN-type), PPM3 (NORDIC-type), PPM4 (IEC-type1), dBFS
- Displays setup menus and allows various parameter settings
- Three quick-recall memory settings for immediate mixer setup
- Ten user-scene memory settings (each including level meter, LCF, limiter/compressor, and link status)
- Six scale sheets supplied for different level-meter calibrations
- Back light
- Heated LCD for low-temperature conditions

SYSTEM EXAMPLE



Monitoring

- 2 outputs: 1/4″ phone jack and 3.5-mm mini jack
- Six monitoring modes: left output, right output, stereo output, left/right-mixed monaural, M/S decode and camera return
- Level-control knob

Camera-Audio Send/Return-Level Control

- Stereo return from a camcorder via 12-pin Tajimi balanced connector
- Precise level control on LED with auto-evaluation function for return level
- Monitoring capability with headphones

Oscillator/Talkback (selectable)

- Oscillator: 1-kHz pilot-tone signal into all outputs
- Talkback: slate into all outputs
- Momentary and alternative modes for both oscillator and talkback

Power

- External DC 10 to 15v input with 4-pin XLR connector
- External DC 12 V input with jack connector
- DC 12 V output on 4-pin Hirose connector
- 8 internal AA-size (LR6) alkaline batteries for 5 hours of continuous operation

SHURE

FP-33

Portable 3-Channel Stereo Mixer



The standard by which portable mixers are measured, the FP-33 is used all over the world for remote audio recording, ENG/EFP applications and location film production. Built upon the benchmark FP32 and FP32A field mixers, the FP33 is light enough, small enough and rugged enough to take anywhere. It features three XLR-balanced mic/line inputs and two outputs, center detented pan pot for each input channel, oscillator, two headphone jacks, 48v/12v phantom and 12v T (A-B) power, tape out jack, and a monitor input for the headphone circuit. Exceptional low noise design and wide dynamic range make the FP-33 ideal for use with digital transmission links or digital video/audio media, including DAT and recordable CD.

FEATURES

Inputs

- Three XLR-balanced inputs; switchable to low-impedance mic or line level.
- Phantom or A-B (T) power for condenser mics is available at each mic input.
- Built~in tone oscillator for level checks or line tests.
- Slate microphone with automatic gain control (AGC) for take identification or for emergency use. Slate tone for identifying take locations during editing.
- Stereo monitor input allows headphone monitoring of external sources without interruption of mixer functions.
- Link switch couples mixer inputs 2 and 3 into stereo pair

Outputs

- Left and right XLR-balanced outputs are mic/line switchable.
- Left and right tape outputs are available to feed tape recorder inputs or other unbalanced aux-level inputs.
- 1/4″ and 3.5 mm headphone jacks are driven by a stereo headphone power amp with separate level control.

Controls and Indicators

- Center detented pop-up pan pots on each input. Color-coded, soft touch rubberized knobs with tactile position indicators
- Active, feedback~type input gain controls permit direct input of high-level sources without input attenuators.
- Dual clutched Master gain control for individually adjusting left and right levels at line/mic and tape outputs, as well as tone oscillator and slate mic levels.
- 150Hz (6dB/octave) Lo-cut filters at each input reduces extraneous low~frequency interference.
- Built-in limiter with adjustable threshold prevents output clipping of mixer or input overload of amp or tape deck.
- LED indication of input level, peak output level, limiter action, and battery.
- Illuminated VU meters for left/right channels. Preset for 0 VU = +4 dB, adjustable for other levels.
- Headphone monitoring mode switch and headphone MS matrix.

Performance

- Wide, flat response with extremely low distortion and up to +1dBm output level for studio-quality performance.
- Small and lightweight, the FP-33 offers sealed input potentiometers and a steel chassis, making it extremely rugged and durable as well.
- Mix bus jack for connecting additional FP33 or FP32A mixers
- Internal DIP switches provide over 4,000 different set-ups

Power

- Powered by two standard 9v batteries (up to 8-hours under normal conditions) that can also supply 48v or 12v phantom power to condenser microphones.
- Can also be externally powered from any 11 to 30v DC source such as battery belt pack or car battery.
- A third 9v battery can be used for condenser mics that require A-B power.

FP-33 with carrying case, shoulder strap and mix bus cable.....**1199.95**

Applications Include:

◆ Remote audio recording ◆ Electronic field production (EFP) ◆ Electronic news gathering (ENG) ◆ Film production

FP-410

Portable Automatic 4-Channel Mic Mixer with IntelliMix



A portable automatic mixer, the FP-410 is equipped with Shure's patented "IntelliMix" to deliver flawless automatic microphone mixing. The remarkable operating concept behind the FP-410, "Intellimix" combines three unique functions: Noise-Adaptive Threshold, MaxBus and Last Mic Lock-On, to provide greater gain before feedback, reduce pickup of ambient noise and eliminate comb filtering effects. Ideal for video production, broadcast, conference recording and field production, the FP-410 is also useful for convention facilities, hotels and sound installations.

Noise-Adaptive Threshold:

Distinguishes between constant background noise (such as air-conditioning) and rapidly changing sound (such as speech). This function continuously adjusts the activation threshold so only speech levels that are louder than background noise will activate an FP-410 channel.

MaxBus:

Eliminates the poor audio quality that results when a speaker is picked up by more than one microphone. It does this by controlling the number of microphones that may be activated for a single sound source. With MaxBus, one talker will activate only one FP-410 channel, even if multiple microphones are "hearing" that talker.

Last Mic Lock-On:

Maintains a seamless audio mix by keeping the most recently activated microphone open until a newly activated mic takes its place. Without this function, a long pause in conversation might cause all mics to turn off and sound as if the audio signal has been lost. With Last Mic Lock-On, background ambience is always present.

"IntelliMix"—How it Works

Multiple miking situations—with a number of talkers participating—have always presented problems for the audio technician. If too few mics are used, the coverage of each talker may vary, with one talker (nearest the mic) being louder and clearer than the next. Talkers farthest from the mics will sound "echoey" and reverberant, as very little of the direct sound from their mouths reaches the microphones. If too many mics are used, there's more background noise and reverberation pickup, as well as less gain before feedback if a sound reinforcement (PA) system is used.

It's somewhat like having multiple video cameras all focused on the same subject. If the camera signals are combined, the result is a blurred image. When multiple microphones are open for a single talker, the result is a blurred audio signal. But it's often not practical for someone to turn mics on when they are needed and off when they are not. The answer is the FP-410.

The FP-410 automatically attenuates any microphone not being used, greatly reducing excess reverberation and feedback problems. When a new talker starts, the FP-410 immediately selects and silently activates the most appropriate microphone. "IntelliMix" electronic processing enables the FP-410 to provide clear, natural voice pickup. The FP-410 significantly reduces the problems of "boomy" or "muddy" sound, insufficient sound level (because of feedback or "howling"), and operator errors. In fact, errors are virtually eliminated because the FP-410 doesn't need an operator or technician for continual adjustment—once set up, it is completely self-sufficient.

The FP-410 has numerous applications in video production and audio recording, broadcasting, and sound reinforcement. In any speech pickup application with multiple microphones, the FP-410 dramatically improves audio quality. Switching from manual to automatic operation allows an individual's voice to rise above background noise and reverberation to become clearer and more intelligible.

- Four XLR-balanced inputs and two XLR-balanced outputs. Each can be individually set for microphone or line-level signals.
- Any high quality, low-impedance, balanced mic (dynamic or condenser) can be used, including wireless and shotguns
- Additional FP-410 mixers can be interconnected. Linked systems can contain over 25 mixers and 100 microphones.
- Front-panel channel gain and master controls operate as in conventional mixers.
- Selectable hold time keeps microphones on during short pauses in speech.
- Automatic gain adjusts as additional microphones are activated.
- Wide, flat frequency response and low distortion up to +18 dB output.
- LED indication of mic channel mix levels, output level and limiter action.
- Automatic muting prevents annoying thumps and loudspeaker damage when the FP-410 is turned on and off.
- Separate monitor input and tape output (aux-level) jacks.
- Front panel headphone monitor jacks with level control.
- 48-volt phantom powering for condenser microphones.
- Operates on AC or two 9-volt batteries.
- Includes optional bumpers (feet) for use on horizontal surfaces; a short cable for linking two FP-410's; and a rack mount kit for installation in a 19˝ rack.

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SHURE

FP-24

Portable 2-Channel Stereo Preamp and Mixer

The FP-24 is a studio-quality, two-channel, portable stereo microphone mixer, designed for active use in demanding broadcast environments. Features including assignable L-C-R inputs, built-in slate microphone, 1kHz tone oscillator, and headphone monitoring render this mixer extremely adaptable in application. The remarkable audio performance and comprehensive attributes of the FP-24 make it an appropriate choice for studio or field production engineers. Due to its compact and rugged mechanical and electrical construction, the FP-24 excels in no-compromise settings such as radio, television and film production.



- Two transformer-balanced microphone inputs with left, center, and right position mixing capabilities
- Unique 7-segment peak output meters, with three selectable levels of LED brightness—readable even in direct sunlight
- High current balanced output drivers provide signal integrity over long cable runs
- Headphone preamplifier enables monitoring of program audio or external tape return
- "Unclippable" input peak limiters with adjustable threshold (each input)
- Selectable 15v and 48v phantom power
- Switchable low-cut filters with 80 and 160 Hz corner frequencies, 6dB per octave
- Powered with two AA alkaline batteries for 11-12 hours) or 4-14v DC power (PS20)
- The FP-24 has the ability to link with the Shure FP33 mixer using the optional A33LK cable kit to create a highly portable, flexible, and cost-effective 5 x 2 audio production system. (The A33LK kit can also be used with the Shure FP23 to create a 4 X 2 setup.)



M367

Portable 6-Channel Mic/Line Mixer

An industry standard, the M367 is a six-input mono mic/line mixer/remote preamplifier specifically designed for professional applications. A complete and compact console, the M367's excellent performance, versatility and features make it ideal for studio, remote, video deposition and sound reinforcement applications as well as an add-on mixer for expanding existing facilities. Built to meet the requirements of the most demanding field production applications.

- Six switchable XLR-balanced mic or line level inputs with individual gain controls and low-frequency roll-off switches
- Two XLR-balanced outputs; one selectable mic/line output and one dedicated line output
- Metal XLR connectors on both inputs and outputs, detachable AC cable
- Feedback-type input gain controls for maximum clipping levels and dynamic range
- Built-in switchable peak limiter cuts output overload distortion and adapts to power supply voltage
- LED indicator shows limiter operation or overload with limiter defeated
- Externally adjustable limiter threshold (-4 to +18dB)
- Wide, flat frequency response (20Hz to 20 kHz) and extremely low distortion up to +16dBm line level output
- VU meter is calibrated for +4 and +8dB with range switch. Meter is also illuminated during AC operation
- Phantom power for condenser mic operation
- Front-panel headphone level control and monitor jack; can drive almost any stereo or mono headphones
- Headphone output level is high enough to be used as an auxiliary unbalanced line feed to drive a tape recorder or power amplifier
- Automatic muting prevents speaker damage during power on/off
- Highly stable, low-distortion tone oscillator provides for line test and level checks
- Rear panel Mix Bus jack facilitates stacking multiple M367's for additional input capability without losing any inputs. Two M367's connected, provides two independent master gain controls and two isolated line amplifiers with eight individually controlled inputs
- Selectable 120 or 240 v AC operation as well as portable DC capability (three 9v batteries required)

M367 .....**524.50**

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EQUIPMENT LEASING AVAILABLE

FP-42

Portable 4-Channel Stereo Mic Mixer

The FP-42 is a four-input, two-output, compact self-contained stereo mixer with all the features and ruggedness that have made Shure mixers the industry standard. Perfect for mixdown in video editing suites, it integrates all the operating features and reliability of a professional studio mixer in a single unit that is small and light enough for location use.



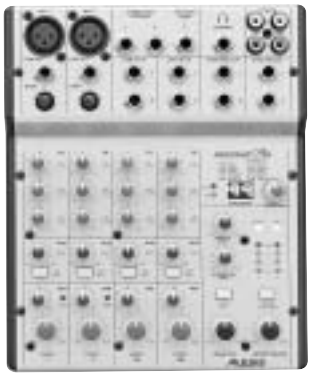
- Four XLR-balanced inputs, each mic/line switchable with low-cut filters and cueing options
- Left and right XLR-balanced outputs with mic/line and mono/stereo switches
- Active feedback-type input gain controls for high level signals without input attenuators
- Pull-pot cueing on all inputs provides channel previewing
- Built-in tone oscillator permits level checking and line testing
- Parallel stereo headphone jacks (1/4″ and mini) with level control
- Adjustable-threshold limiter with left and right channel peak indicators
- VU meters with range switch and battery check function
- Left and right channel master level controls and ganged headphone level control
- Powered by three 9v batteries or via internally selectable) 120 or 240v AC power

	M367	FP-33	FP-42	FP-410	FP-24
Transformer-Balanced Inputs and Outputs	Yes	Yes	Yes	Yes	Yes
Active-Balanced Outputs	—	—	—	—	Yes
Low-Z Mic-Level Input	Yes	Yes	Yes	Yes	Yes
Line Level Input	Yes	Yes	Yes	Yes	—
Aux Level Input	—	—	—	—	Yes
Mic Level Output	Yes	Yes	Yes	Yes	—
Line Level Output	Yes	Yes	Yes	Yes	Yes
Phono Jack Aux Level Output	—	—	—	—	Yes
Headphone Output	Yes	Yes	Yes	Yes	Yes
Phantom Power	48v	48v	Yes	48v	Yes
VU Meter	Yes	Yes	Yes	Yes	—
Peak Meter	—	—	—	Yes	Yes
EQ	Yes	Yes	Yes	Yes	Yes
Tone Oscillator	Yes	Yes	Yes	Yes	Yes
Linkable	Yes	Yes	Yes	Yes	To FP33
Slate Mic & Tone	—	Yes	—	—	Yes
Limiter	Yes	Yes	Yes	Yes	Yes
Stereo Operation	—	Yes	Yes	—	Yes
Power Options	AC/DC	DC Only	AC/DC	AC/DC	DC Only
TYPICAL APPLICATIONS					
Automatic Microphone Activation	—	—	—	Yes	—
Church Sound	Yes	—	—	—	—
Broadcast	Yes	Yes	Yes	Yes	Yes
Field ENG/EFP	Yes	Yes	Yes	Yes	Yes
Basic Microphone Mixing, Basic PA	Yes	—	—	—	—
Distance Learning	Yes	—	—	Yes	—
Video Editing Suite	Yes	—	Yes	—	Yes
Conference Rooms, Round Table Discussion	Yes	—	—	Yes	—

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MULTIMIX SERIES

Alesis' MultiMix series is a line of powerful, portable, affordable mixers designed for a wide range of operating environments and applications. Available in 4 different tabletop mixers, plus a powerful rack mixer, all feature the high quality and high performance that Alesis is known for. Each mixer features 99 on-board 24-bit Alesis digital effects, for incredible control and creativity, as well as an intuitive interface design with superior feel. And each mixer has been designed from the inside out to provide the level of performance users expect from Alesis, at incredible prices.






MultiMix 6FX

A super compact analog mixer for performing or recording applications, with digital effects. An incredible value (under $100), the MultiMix 6FX includes 2 mic/line inputs, 2 stereo line inputs, a 3-band EQ on each channel, plus 99 high quality 24-bit Alesis digital effects.

MultiMix 8FX

A highly compact analog mixer, designed for live applications such as piano bar, single musician, duets, etc., or for simple recording applications such as live to 2-track, etc. Delivering powerful mixing capabilities at a very low price, the MultiMix 8 FX includes 99 high quality 24-bit Alesis digital effects, and is small enough to fit in a "gig bag." Featuring 8-input, 2-buss analog mixing with 4 mic preamps with switchable 75Hz high-pass filters, 2 stereo line channels, a 3 band EQ on each channel, and more.

MultiMix 12FX

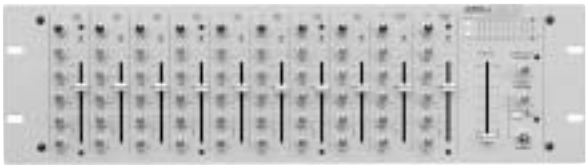
A portable analog mixer that can handle a wide range of PA and recording applications, such as live performances for small bands, ensembles, and general PA, or recording applications such as live to 2-track, simple multi-track, etc. The low cost, yet powerful MultiMix 12FX includes a 12-input, 4-buss analog mixer featuring 4 mic preamps with switchable 75Hz high-pass filters, 60mm linear faders on each channel, 4 stereo line channels, a 3-band EQ on each channel, 99 high quality 24-bit Alesis digital effects, and much more.

MultiMix 12FX D

The MultiMix 12FXD is the same as the MultiMix 12FX plus it adds a 44.1kHz digital S/PDIF digital output—making it ideal for those seeking a low-cost, yet powerful mixing solution with digital output.

MultiMix 12R

The world's most affordable 12-input, 2-buss analog rack mixer featuring 8 mic preamps with 2 stereo line channels, all in a compact, 3-U rack mount configuration. The MultiMix 12R has 2-band, fixed frequency EQ on each channel, 1 pre-fader aux send and 1 post-fader aux send per channel, and 60mm linear potentiometers (faders) for master level of each channel. The master section features an external stereo aux return level, stereo LED bar graph meters, stereo master L/R buss 60mm fader, and a separate phones/monitor level control.



- 12-channel analog mixer in compact 3U rackmount design
- 8 XLR mic inputs with phantom power
- 2 aux sends (pre-and post-fader) for monitor returns and effects
- Insert points on 8 channels, plus 2 channels of stereo line inputs
- 2-band shelving EQ at 80Hz and 12 kHz
- 60mm linear faders on each channel

M-3 TUBETRACKER

8/2 Tube Mixer

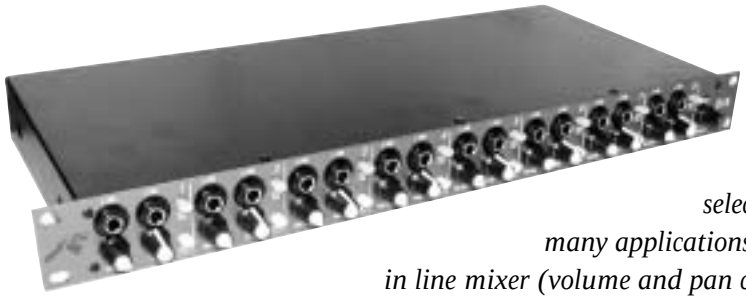
The M-3 is an 8-channel compact tube mixer, ideal for all tracking and mixing applications. Each channel features discrete tube mic preamps, a 4-band EQ with swept mids, 2 aux sends and a post-fader direct output for multitrack recording. The master section has tube stages in the main mix bus, together with 2 aux returns, monitoring, metering and an optional 24-bit/96kHz stereo digital output. When linking multiple M-3 mixers, the stereo, PFL and aux busses are also linked, providing an unlimited number of channels.



- Balanced XLR mic and 1/4" jack line inputs
- Balanced XLR L/R mix outputs
- Optional AES, S/PDIF and TOSLINK optical digital outputs
- Sovtek ECC83/12AX7A tubes
- Phantom power, 90Hz filter, phase reverse and mic/line switch
- 100mm faders with pan, mute and solo
- Drive and Peak LEDs
- Balanced internal signal busses
- Stereo main fader with VU metering
- Unbalanced 1/4˝ jack inserts
- Word Clock input
- Switchable level direct outputs
- Link facility via 15-way D-type connectors
- Headphone output
- External linear PSU
- 200v DC stabilized supply
- 19˝rack-mountable or table-top use

SM PRO AUDIO DI4v/DI8

4- and 8-Channel DI/Line Mixers



Multi-function direct injection boxes and stereo line mixers in one single unit rackmount chassis, the half-rack SM DI4v and full-rack SM DI8 feature 4 and 8 channels of unbalanced to balanced converters with 10dB pad and earth lift selector provided per channel. They are suitable for many applications right out of the box but it is the addition of a built in line mixer (volume and pan control for each channel and master volume), that makes them unique problem solvers. If you are a keyboard player, electronic drummer, PA/Studio owner, pro musician or a hobby player, the SM DI4v or SM DI8 will find a place in your rack. Imagine running your 4 keyboard or modules on stage directly onto them and being able to send off balanced signal to the front of house rig while providing your own dedicated monitoring mix...all on a single rackmount direct injection box/line mixer! SM DI4v includes a headphone amplifier so you can monitor your own signal, the SM DI8 includes a LINK function so that multiple units can be linked to a single stereo output. So if you have more than 8-16-24.... inputs of unbalanced signal they can all be routed and monitored through the one output pair!

- 4- and 8-channel DI Box
- 4- and 8-channel Mixer
- Each input has a balanced output
- Earth lift per channel
- 10dB pad per channel
- Volume control per channel
- Pan control per channel
- Master Volume control
- Stereo Line Out
- Headphone Amp (SM DI4v)
- Stereo Link Input (SM DI8)
- 110v-240v external power supply

UB SERIES

Ultra Low-Noise Eurorack Mixers

By continuously responding to the suggestions of their customers, Behringer has developed a range of mixers that offer more bang for the buck without compromising on audio quality. Built to exacting standards, they are rugged and reliable, and offer the flexibility and sonic integrity demanded by the most discerning end users. Ultra-low noise circuitry and transparent audio, balanced mic/line inputs, invisible mic preamps, solid steel chassis, sealed potentiometers make the UB Series Euroracks ready for broadcast production as well as MIDI rigs, home/project studios, video editing suites, and live sound reinforcement. And with ten mixers in the line, there is sure one to meet your requirements.

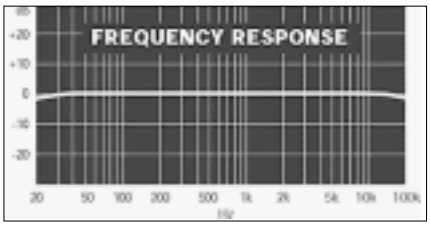
With the UB Series, Behringer takes mixing to the next level. From the miniscule UB502 to the largest UB2442FX-PRO, all mixers combine the same high-quality components and innovative features with affordable pricing. They all feature premium IMPs ("Invisible" Mic Preamps) and ULN (Ultra Low-Noise) circuitry design, and with a feature set of impact-resistant pots and first-rate faders from ALPS, and low-tolerance components, the UB Series provide uncompromising quality under their sleek and sturdy roadproof skin. In addition, the PRO models feature an internal autorange (80-240v) switch-mode power supply and all FX PRO models feature award-winning 24-bit digital and 24-bit/46 kHz internal processing Virtualizer technology with 99 breathtaking stereo effects.

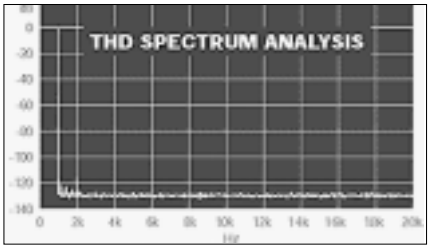
The state-of-the-art, studio-grade IMPs provide 130 dB dynamic range for 24-bit, 192 kHz sampling rate inputs, ultra-wide 60dB gain range and +30dBu line input capacity for crystal-clear audio and ample headroom. With their extreme bandwidth and amazingly neutral, noise- and distortion-free circuitry, the mic preamps deliver crystal-clear audio for awesome sound without any coloration. ULN low impedance circuitry design coupled with high-quality 4580 op amps provides maximum headroom, minimal noise and stunningly transparent audio.

THEY ALL FEATURE:

Invisible Mic Preamp (IMP)

- Five years in the making, the Invisible Mic Preamp (IMP) is a superior mic preamp that delivers crystal-clear audio with incredible reproduction of the slightest nuances. It matches or exceeds the performance of outboard mic preamps. You get extreme bandwidth of 5Hz to 100kHz coupled to unusually neutral, noise- and distortion-free circuitry.





Ultra Low-Noise (ULN)

- ULN stands for the Ultra low-noise design of their circuitry. It is based on extremely low-impedance components that keep thermal inherent noise and crosstalk at an absolute minimum. The result: your console is considerably less noisy and crosstalk between neighboring channels is virtually non-existent.

Switch-Mode Power Supply (SMPS)

- All PRO series mixers (UB1204-PRO and up) use a state-of-the-art, over-sized, switch-mode power supply with lots of headroom. The advantage, compared to conventional circuits, is that a switch-mode power supply automatically adapts to AC voltages between 100 and 240 volts. Furthermore, due to its much greater efficiency, it consumes much less energy than a conventional supply unit.
- Even more important is the fact that conventional transformers always induce 50 Hz hum (check it out on other mixers when you turn up all volume controls). Behringer's switch-mode supply operates at very high frequency and therefore keeps the mixer "dead" quiet.

High-Quality Construction

- High quality ALPS sealed rotary controls and faders keep out dust and moisture to ensure long lasting, reliable performance. Steel chassis and horizontally mounted fiberglass circuit boards are used for their ruggedness, whether on the road or bouncing from studio to studio.

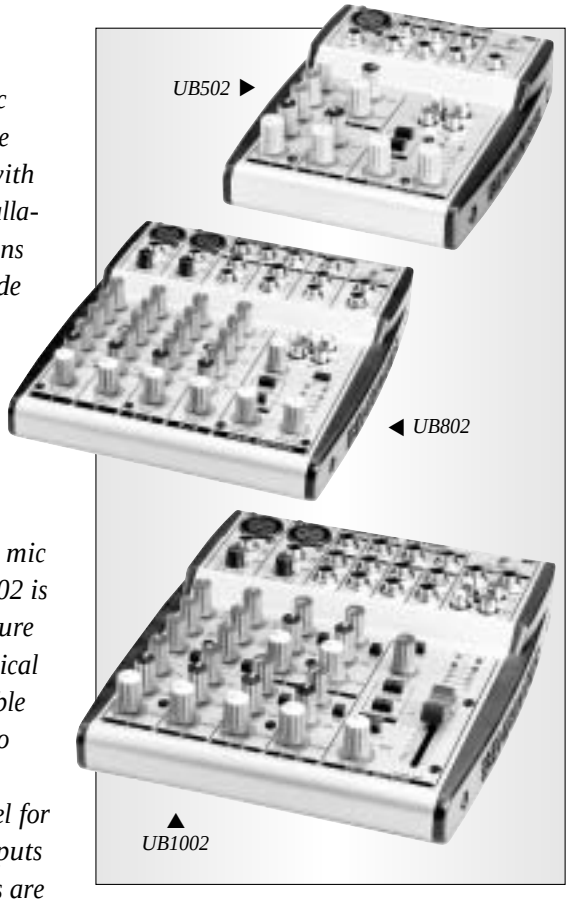
24-bit / 46kHz

- Digital recordings done at 16-bit depths break up the dynamic range between the loudest and quietest portions of a signal into 65,536 steps. Behringer uses 24-bit resolution (dividing the signals' dynamic range into 16,777,216 steps) allowing for far more accurate rendering of the quieter parts of the audio.
- Audio signals change constantly, so a digital device must capture these changes as faithfully as possible. The higher the sampling rate, the better. UB Series mixers use a sampling rate of 46kHz (46,000 times per second). Since the sampling rate should be twice that frequency, 46 kHz is more than adequate for capturing all signals in the audible range—the human ear can not hear sounds whose frequency is higher than 20 kHz.

B&H PHOTO - VIDEO - PRO AUDIO EQUIPMENT LEASING AVAILABLE

UB502 • UB802 • UB1002 • UB1202

5-, 8, 10- and 12-Input 2-Bus Mixers



It may be small, but the UB502 features the same state-of-the-art Invisible Mic Premap (IMP) and the same Ultra low-noise (ULN) design for highest possible headroom as its larger brethren. Its one mono channel plus 2 stereo channels with a 2-band EQ on the mono channel make it ideal for use with fixed audio installations as well as keyboards, samplers and computer application. Hobby musicians and video makers will also find lots of use for this ultra-compact mixer. A shade larger than the UB502, the UB802 features 2 mic preamps, 6 balanced high-headroom line inputs, 1 post fader FX send per channel for external FX devices and 1 stereo aux return for FX applications or as separate stereo input. There is also an extremely musical 3-band EQ on all channels. The UB802 is well-suited for use with fixed audio and video installations as well as keyboards, samplers and computer application.


UB1202

The 10-input UB1002 features two mic preamps), while the 12-input 1202 is equipped with four. Both feature an effective, extremely musical 3-band EQ plus switchable low-cut filter on all mono channels. There is 1 post fader FX send per channel for external FX devices, main mix outputs plus separate control room, headphones and stereo tape outputs. Tape inputs are assignable to main mix or control room/phones outputs. These are extremely versatile mixers with possibilities ranging from connecting a DAT recorder and monitor speakers to running permanent video and audio installations.

FEATURES

- One (UB502), two (UB802, UB1002) or four (UB1202) XLR-balanced discrete inputs featuring studio-grade "Invisible Mic Preamps" with:
  - 130dB dynamic range for 24-bit, 192kHZ sampling rate inputs
  - Ultra-wide 60dB gain range
  - Lowest distortion 0.0007% (20Hz -20kHz)
  - A bandwidth ranging from below 10Hz to over 20kHz for crystal-clear reproduction of even the finest nuances
- Ultra-low noise ULN design, highest possible headroom, ultra-transparent audio
- Effective, extremely-musical 3-band EQ (2-band on the UB502) and peak LED on mono channel
- 5 (UB502), 6 (UB802) balanced high-headroom 1/4″ line inputs
- Main mix outputs plus separate headphones and stereo tape outputs
- Tape inputs assignable to main mix or phones outputs
- External power supply for noise-free audio and superior transient response

UB 802 Step-up Features:

- Switchable +48v phantom power for condenser mics
- 1 post fader FX send per channel for external FX devices
- 1 stereo aux return for FX applications or a as separate stereo input
- Main mix outputs plus separate control room, headphones and stereo tape outputs
- Tape inputs assignable to main mix or control room/phones outputs

UB1002/UB1202 Step-up Features:

- 10 (UB1002) or 12 (UB1202) balanced high-headroom 1/4″ line inputs with +4/-10 level selection on all stereo channels
- 3-band EQ plus switchable low-cut filter on all mono channels
- FX to control room function helps to monitor effects signal via headphones and control room outputs
- Long-wearing 60mm logarithmic-taper ALPS master fader and sealed rotary controls

UB1204-PRO • UB1204FX-PRO

12-Input 2/2-Bus Mic/Line Mixers



These are mixers whose small size belies their incredible versatility and audio performance. You get 8 balanced high-headroom line inputs, 4 Invisible Mic Preamps, 2 aux sends per channel (1 pre fader for monitoring applications and 1 post fader for external FX devices). There are 2 subgroups with separate outputs for added routing flexibility and 2 multi-functional stereo aux returns with flexible routing round off the list of high-quality features. The UB1204FX-PRO has all the same features plus a 24-bit digital stereo FX processor with 99 great-sounding VIRTUALIZER presets including reverb, delay, chorus, compressor, tube distortion, vinylizer and more, plus a 1kHz test tone generator. There are also solo and PFL functions on all channels as well as a pre/post fader switchable aux send for monitoring/FX applications.

STEP-UP FEATURES (FROM THE UB1202):

- Only 8 balanced high-headroom 1/4″ line inputs (not 12) with +4/-10 level selection on all stereo channels
- 2 aux sends per channel:
  1 pre fader for monitoring applications
  1 post fader for external FX devices
- Peak LEDs and mute/alt 3-4 function on all channels routes signal to subgroup instead of main outs
- Tape inputs assignable to main mix or control room/phones outputs
- 2 subgroups with separate outputs for added routing flexibility
- 2 multi-functional stereo aux returns with flexible routing
- Control room/phones outputs with multi-input source matrix
- Balanced main mix outputs with gold-plated XLR connectors plus separate control room, headphones and stereo tape outputs
- Internal switch-mode power supply for maximum flexibility (100-240v), noise-free audio, superior transient response plus lowest possible power consumption
- Includes rackmount brackets

UB1622FX-PRO

16-Input 2/2-Bus Mic/Line Mixer and 24-bit Multi-FX Processor



You get all the important elements right here in a 19″ form-factor. This compact mixer offers 12 balanced high-headroom line inputs, 4 Invisible Mic Preamps, 2 aux sends per channel (1 pre/post fader switchable for monitoring/FX applications and 1 post fader for internal FX or as external send). You also get an extremely musical 3-band EQ with semi-parametric mid-band plus switchable low-cut filter on all mono channels. The integrated 24-bit digital stereo FX processor offers 99 great-sounding VIRTUALIZER presets as well as a 1kHz test tone generator. There are 2 subgroups with separate outputs for added routing flexibility and 2 multi-functional stereo aux returns with flexible routing.

STEP-UP FEATURES (FROM THE UB1204 FX-PRO):

- Inserts on each mono channel for flexible connection of outboard equipment
- 3-band EQ with semi-parametric mid-band plus switchable low-cut filter on all mono channels
- 12 balanced high-headroom 1/4″ line inputs with +4/-10 level selection on stereo channels
- Peak LEDs, mute, main mix and subgroup routing switches
- Main mix outputs with jack and gold-plated XLR connectors, separate control room, headphones and stereo tape outputs

UB1832FX-PRO

18-Input 3/2-Bus Mic/Line Mixer and 24-bit Multi-FX Processor



If you are looking for a capable compact mixer with a breathtaking 3D XPQ stereo surround effect and a 9-band stereo graphic EQ, this is your machine. Of course, you also get 14 balanced high-headroom line inputs, 6 "Invisible" Mic Preamps, and 3 aux sends per channel and extremely musical 3-band EQs with semi-parametric mid bands plus switchable low-cut filters on all mono channels. There are 2 subgroups with separate outputs for added routing flexibility and 2 multi-functional stereo aux returns with flexible routing are also there for your convenience. You will also find the integrated 24-bit digital stereo FX processor with 99 great-sounding VIRTUALIZER presets.

STEP-UP FEATURES (FROM THE UB1622 FX-PRO):

- 6-XLR-balanced discrete inputs featuring studio-grade "Invisible Mic Preamps"
- 14 balanced high headroom 1/4″ line inputs with +4/-10 level selection on stereo channels
- 9-band stereo graphic EQ allows precise frequency correction of monitor or main mixes
- Breathtaking XPQ 3D stereo surround effect widens the stereo image and adds life and transparency to your sound—an easy way to put a unique final polish on your music and turn your performance into an incomparable experience.
- 3 aux sends per channel (Pre fader for monitoring, Pre/post fader switchable for monitoring/FX applications, Post fader for internal FX or as external send).
- Balanced TRS and gold-plated XLR main mix outputs, separate control room, headphones and stereo tape outputs

With their extensive and carefully thought-out routing possibilities, the FX-PRO mixers are perfect for both live and studio use. The prime suspects are high-quality recording studios as well as MIDI-studio applications, small-size PA applications (solo entertainers, duos to quartets, small bands), video edits and similar setups.

UB2222FX-PRO



22-Input 2/2-Bus Mic/Line Mixer and 24-bit Multi-FX Processor

The second-largest mixer in the UB Series, the UB2222FX-PRO has 16 balanced high-headroom line inputs, 8 Invisible Mic Preamps, 3 aux sends per channel, 3-band EQs with semi-parametric mid bands plus switchable low-cut filters on all mono channels, and integrated 24-bit digital stereo FX processor with 99 great-sounding VIRTUALIZER presets. There are 2 subgroups with separate outputs for added routing flexibility as well as 3 multi-functional stereo aux returns with flexible routing.

STEP-UP FEATURES (FROM THE UB1832 FX-PRO):

- 8-XLR-balanced discrete inputs featuring studio-grade "Invisible Mic Preamps"
- 16 balanced high headroom 1/4″ line inputs with +4/-10 level selection on stereo channels
- 3 multi-functional stereo aux returns with flexible routing

BEHRINGER

UB2442FX-PRO

24-Input 4-Bus Mic/Line Mixer and 24-bit Multi-FX Processor



This is the big cajuna of the whole UB Series. You get all the features you'd expect in an ultra high-quality compact mixer: 16 balanced high-headroom line inputs with dedicated gain controls on stereo channels 13-16, 10 studio-grade IMP "Invisible" Mic Preamps, and an effective, extremely musical 3-band EQ with semi-parametric mid band plus switchable low-cut filter on all mono channels. There is also an integrated 24-bit digital stereo FX processor with 99 great-sounding VIRTUALIZER presets.

STEP-UP FEATURES (FROM THE UB2222 FX-PRO):

- 10-XLR-balanced discrete inputs featuring studio-grade "Invisible Mic Preamps"
- 16 balanced high-headroom 1/4″ line inputs with dedicated gain controls on stereo channels 13-16
- 2 headphone outputs
- Channel inserts and direct outputs on each mono channel plus main mix inserts for flexible connection of outboard equipment
- 4 aux sends per channel: 2 pre/post fader switchable for monitoring/FX applications; 2 post fader (for internal FX or as external send
- 4 subgroups with separate outputs for added routing flexibility
- 4 multi-functional stereo aux returns with flexible routing
- BNC connector for standard gooseneck lamps

	UB502	UB802	UB1002	UB1202	UB1204 PRO	UB1204 FX-PRO	B1622 FX-PRO	UB1832 FX-PRO	UB2222 FX-PRO	UB2442 FX-PRO
Channels	1 mono+ 2 stereo	2 mono+ 2 stereo	2 mono+ 4 stereo	4 mono+ 4 stereo	4 mono+ 2 stereo	4 mono+ 2 stereo	4 mono+ 4 stereo	6 mono+ 4 stereo	8 mono+ 4 stereo	10 (8 + 2) mono + 4 stereo
Mono mic/line channel inputs	1 (XLR + 1/4″ TRS)	2 (XLR + 1/4″ TRS)	2 (XLR + 1/4″ TRS)	4 (XLR + 1/4″ TRS)	4 (XLR + 1/4″ TRS)	4 (XLR + 1/4″ TRS)	4 (XLR + 1/4″ TRS)	6 (XLR + 1/4″ TRS)	8 (XLR + 1/4″ TRS)	8 (XLR + 1/4″ TRS)
Stereo line channel inputs	2 (1/4″ TRS)	2 (1/4″ TRS)	4 (1/4″ TRS)	4 (1/4″ TRS)	2 (1/4″ TRS)	2 (1/4″ TRS)	4 (1/4″ TRS)	4 (1/4″ TRS)	4 (1/4″ TRS)	2 (1/4″ TRS & 2 (XLR + 1/ 4″ TRS)
Channel inserts (pre EQ, pre fader)	—	—	—	—	—	—	channel 1-4	channel 1-6	channel 1-8	channel 1-8
EQ mono channels	2-band	3-band	3-band + Low cut		3-band + Low cut		3-band mid-sweep + low cut			
EQ stereo channels	—	3-band	—	—	3-band	3-band	4-band	4-band	4-band	4-band
Phantom power (+48V)	—	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Main outputs	1/4″ TRS connectors		1/4″ TRS connectors		XLR connectors		XLR & 1/4″ TRS connectors			
Headphones output*	✓	✓	✓	✓	✓	✓	✓	✓	✓	2
2-Track I/O (RCA connectors)	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
Control room outputs*	—	✓	✓	✓	✓	✓	✓	✓	✓	✓
PFL/solo section	—	—	—	—	—	✓	✓	✓	✓	✓
FX unit**	—	—	—	—	—	✓	✓	✓	with separate FX output	
Switch-mode PSU	—	—	—	—	✓	✓	✓	✓	✓	✓
Rack-mount kit inc.	—	—	—	—	✓	✓	✓	✓	✓	✓
Dimensions (W x D):	5¼ x 7″	7⅛ x 8⅔″	7⅛ x 8⅔″	9½ x 8⅔″	9 11/16 x 13 5/32″	9 11/16 x 13 5/32″	11⅞ x 13⅞″	16 1/16 x 14 1/16″	16 1/16 x 14 1/16″	16½ x 17¼″
Height (front/rear):	1½ x 1¾″	1½ x 1¾″	1½ x 1¾″	1½ x 1¾″	3 1/16 x 5″	3 1/16 x 5″	3 x 5″	3 x 5″	3 x 5″	2¾ x 5½″
Weight	1⅛ lbs.	2⅛ lbs.	2⅜ lbs.	3 lbs.	5⅝ lbs.	5¾ lbs.	7¼ lbs.	10⅜ lbs.	10⅝ lbs.	13 lbs.

** 1/4″ TRS connector ** with footswitch connector*

DDX3216

Fully Automated 32-Channel, 16-Bus Digital Mixing Console



The DDX3216 offers a feature set you won't find in any other product in its class. 32 channels, 16 internal busses and 8 aux sends. Each channel features a fully parametric 4-band equalizer, sweepable high-pass filter, gate, compressor/ limiter and phase inversion—all simultaneously operable. The first 16 channels also offer a delay function. Add four on-board effects processors that put just about any imaginable effect type at your fingertips, in top quality — and simultaneously accessible. Then full-fledged dynamic and static automation functions, extensive synchronization options and MIDI implementation, a comprehensive internal I/O patchbay, full digital compatibility and functionality. Plus unusually flexible connectivity and expandability, supported via two expansion slots for ADAT, TDIF, AES/EBU interfaces and more. All based on 32-bit floating-point processing (40-bit for the EQ's).



This all comes in a rack-mountable case and with an intuitive, logically laid out user interface based on excellent software and effective control elements, like motorized ALPS faders, assignable channel controls with LED rings, push-and-turn master controllers and a large, backlit LCD display. Ideal for almost any application including recording, live sound, multimedia, audio for video, broadcasting and installations.

FEATURES

- 32 full-fledged channels and 12 high-end ultra low-noise "Invisible" Mic Preamps with phantom power deliver crystal-clear audio with incredible reproduction of the slightest sound nuances and ample headroom.
- 16 busses, 8 aux sends and internal input/output patchbay for comprehensive routing options
- Ultra high-resolution 24-bit AKM A/D and CRYSTAL D/A converters provide optimum audio quality bound to please even the most discerning user.
- High-power floating point DSP technology (32-/40-bit) ensures virtually unlimited internal dynamic range
- Fully featured dynamic and snapshot automation
- 4-band fully parametric EQ, compressor, gate, sweepable high-pass filter and phase inverter on all 32 channels — with additional delay function on the first 16 channels — provide effective dynamic processing.
- Four (simultaneously operable) ground-breaking effects processors, accessible from all 32 channels, provide first-class algorithms, such as chorus, delay flanger, LFO filter, phaser, pitch shifter, reverb, tremolo, and many more.
- 17 precise, low-noise 100mm motorized ALPS faders
- Additional compressor/limiter (switchable pre/post) and EQ for stereo main mix
- Freely configurable built-in level meters on all channels and channel controls with LED rings facilitate editing of any of nine selectable parameters per channel.
- Four freely assignable analog outputs on balanced 1/4" TRS and MIDI plus RS232 connectors round out the I/O options by allowing interfacing with a wide variety of external devices.
- Extensive MIDI implementation capability enables program and control changes, MIDI sysex and MMC.
- Six master controllers with comfortable push-and-turn functionality
- Analog feel, intuitive user interface
- Large, easy-to-read LCD display with adjustable contrast
- Synchronization via SMPTE, MTC or internal clock
- Dither, word length and noise shaping adjustable for digital main outputs
- PCMCIA slot for saving/loading various libraries and other settings
- Routing of complex signal configurations is accomplished by an internal input/output patchbay.
- Two extension slots for installation of optionally available digital interfaces for virtually unlimited connectivity:
  - ADT1616 16-channel ADAT interface
  - TDF1616 16-channel TDIF interface
  - AES808 8-channel AES/EBU interface

MACKIE

1202-VLZ PRO • 1402-VLZ PRO

12 x 2 and 14 x 2 Compact Mixers

Providing maximum performance in minimum sizes, the 1202-VLZ Pro and 1402-VLZ Pro are the mixers of choice for tens of thousands of musicians and sound engineers who need equipment to serve "double duty" in studios and on the road. With 12 or 14 balanced input channels divided between 4 (1202) or 6 (1402) mono mic/line inputs and 4 stereo line input channels, these mixers excel in the field or in the studio. Video suites doing dialog recording, voiceover or sound effects and project studios will find the range of bussing flexibility a blessing. For live gigging, the rugged steel chassis will stand up to the abuse of the road and the 2 aux sends are perfect for monitor and or effects sends.



FEATURES

The1202 and 1402-VLZ PRO are the standard for small-to-medium video post production suites. They're the most popular mixers for home project studios. They're in touring racks of performers. They record location sound for major Hollywood movies. They're on duty at major-league football games, recording the crunch of bodies and the sportscasters' breathless play-by-play. They are also used as live field mixers by the news departments of every major network.

Input Channels

- 4 mono balanced XLR mic/ 1/4″ line inputs with inserts, trim and rotary level controls.
- 4 balanced 1/4″ stereo line inputs with rotary level controls.
- 2 Aux sends per channel can be used as effects sends or for creating monitor mixes and offer 15dB of extra gain above Unity
- Each input has 3 bands of EQ at 80Hz, 2.5kHz and 12kHz with a ±15dB boost/cut.
- Inputs have a -18dB/oct. 75Hz Lo Cut filter that eliminates stage rumble, wind noise, P-pops and other low frequency noise
- The Mute/Alt 3-4 buttons located on each input channel serve two functions: muting the input channel from the main mix, and signal routing, where they act as your gateway to an extra stereo bus
- Constant-loudness pan pots on each input channel keep sound at a consistent volume when panning between left, center and right positions
- PFL Stereo in-place Solo on each channel

Aux Inputs

- 2 balanced 1/4″ stereo returns with EFX to Monitor switch
- Aux 1 master control with Monitor/Post assign switch
- Stereo RCA 2 track tape return

Outputs

- Main stereo outputs are XLR-balanced switchable from +4dBu to mic level
- 1/4″ balanced alternate outputs (alt 3/4) are assignable from mute buttons on input channels
- 2 balanced 1/4″ auxiliary sends
- 1/4″ TRS balanced stereo control room outputs
- Stereo RCA (2 track) tape outputs
- Multi-input source matrix with level control lets you route any combination of Main Mix, Tape In and Alt 3-4 to the Control Room/Phones bus. Routings can be used for submixes, monitoring or tracking and can be subsequently routed to the main mix

Additional Features

- Gain knobs have Unity gain detents, when input levels are properly set, the highest headroom and lowest noise floor is at unity gain
- 12 segment LED's give accurate level status of main outputs or soloed channels
- 48-volt phantom power for condenser mics
- Built-in power supply–no wall wart!
- Rude Solo Light LED bluntly advises that a channel is soloed
- Tape input level control and Tape To Main Mix switch
- 3-year parts and labor warranty.
- Optional RM1202VLZ rackmount ears

1402-VLZ PRO *Step-up Features*



- 6 mono balanced XLR mic/1/4″ line inputs instead of 4
- Switchable AFL/PFL Solo
- Optional RM1402VLZ rackmount ears
- 60mm long-wearing log-taper faders ensure a consistent, smooth accurate response through the length of the fader travel

B&H PHOTO - VIDEO - PRO AUDIO

ONE HOUR FREE PARKING
AT 349 W. 34th STREET (with purchase of $100 or more)

MACKIE MIXER TECHNOLOGY

Mackie revolutionized the compact mixer market by applying features and audio quality previously reserved for high-end, large format mixers. From the smallest to the largest of their models, Mackie puts their best effort into every phase of board design. For example, XDR (Extended Dynamic Range) mic preamps offer over 130dB of headroom, exceeding the performance of outboard preamps costing $1000 to $2000 per channel. Project studios, video suites, multimedia authors, live venues, broadcast facilities and houses of worship seeking uncompromising sonic quality and maximum flexibility will find what they are looking for in Mackie's VLZ PRO mixer line.

Very Low Impedance

- Very low impedance is achieved by scaling down resistor values by a factor of three or four, resulting in a corresponding reduction in thermal noise.

XDR Mic Preamps

- 130dB dynamic range and extremely flat frequency response allows the XDR mic preamps to handle inputs from 192kHz workstations without added coloration.
- DC pulse transformer reject RFI without attenuating frequencies of 15kHz and above.
- Ultra-high-speed, large-geometry input diodes protect the XDR mic preamps from hot-patching and direct short circuits in cables carrying phantom power.

EQ

- Positioned at 12kHz, the Hi Shelving EQ adds sheen and presence to instruments and vocals, enhancing the textures of sounds without contributing to aural fatigue.
- Positioned at 80Hz, the Low Shelving EQ emphasizes the fundamental lows of kick drums, and other bass instruments while allowing you to fatten up male vocals and instruments like guitars.
- Mid-range frequency is placed at 2.5kHz (1202/1402-VLZ PRO), harmonics of vocals and instruments are enhanced without becoming strident or fatiguing.
- Sweepable Mid-Band EQ (1604/1642-VLZ PRO) from 100Hz to 8kHz for specific EQ treatment of a broad range of frequencies.

Mix Amp Headroom

- Negative gain mix amplifier architecture sets standard mixing levels at -6dB. This offers a greater amount of headroom, allowing up to 4 times as many hot signals to be summed at the main mix bus without clipping

Impact Resistance

- Knobs ride just thousandths of an inch from the metal surface of a mixer chassis.
- Brutal knob impact is absorbed by broad pressure on a tough circuit board
- Ultra-tight lip seal design provides a continuously-sealed barrier against dust and liquid
- A braced, horizontal circuit board and shock-absorbing structure eliminates force transferred to the circuit board.

1642-VLZ PRO *16 x 4 x 2 Compact Mixer*



Bridging the gap between the 1604-VLZ Pro (next page) and the smaller 1402-VLZ Pro, the 1642-VLZ Pro gives you 4-bus mixing flexibility, ten XDR mic preamps and four stereo line-level channels—fewer mic preamps and more line-level inputs at your fingertips. Yet it also has live-sound features like EFX to monitor, AFL/PFL solo, pre-fader aux sends, and built like a tank construction. The ideal mixer if you record with lots of keyboards, samplers and drum machines. Perfect for home and project studios as well as video post-production applications.

All The Features of the 1402 VLZ Pro ***PLUS—***

Input Channels

- 16 input channels total— channels 1-8 are mono and channels 9 - 16 are stereo.
- Each channel features input trim, 4 aux sends (aux 1 & 2 assignable pre/post), -20dB pad, 75Hz HPF, Solo (PFL and Solo-In-Place), mute, pan, bus 1/2, 3/4 and L/R assignment buttons and overload LEDs
- Channels 1 - 8 feature balanced XLR mic/ 1/4″ line with inserts, direct outputs and 3 band EQ
- 2 stereo balanced XLR mic/ 1/4″ line inputs, 2 balanced stereo 1/4″ line inputs
- 4-band active EQ on stereo channels

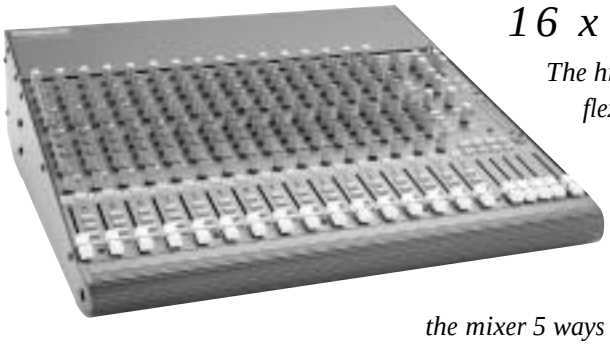
Outputs

- Four 1/4″ balanced subgroup outputs have 60mm faders, switchable L/R assignment to the main mix bus.
- Two 1/4″ stereo headphone outputs
- Four 1/4″ balanced aux sends. Aux send 1-2 master section with level controls and solo
- 1/4″ balanced mono output with level control

Additional Features

- Four 1/4″ balanced stereo aux returns with level controls and solo.
- Switchable AFL/PFL Solo with level control
- Optional rack ears are allow mounting in a standard 19″ rack.

1604-VLZ PRO



16 x 4 Compact Mixer

The highest headroom and lowest noise of any mixer in its class, the 1604's flexible 4-bus architecture is ready for just about any project recording studio or live venue. Post production and broadcast facilities will also appreciate the myriad routing possibilities. VLZ design ensures quiet mixes with all 16 channels open during mix down, while the XDR mic preamps reveal all of the subtleties and nuances of your favorite condenser mic. Mackie's rotopod design and supplied rackmount hardware allows you to configure the mixer 5 ways by allowing you to rotate the input and output jacks to the top, back or front of the mixer with or without rackmounting. With unprecedented sonic performance, bandwidth, dynamic range and versatility, the 1604-VLZ meets the demands of the most sophisticated digital audio workstations —while remaining totally affordable.

ALL THE FEATURES OF THE 1642 VLZ Pro *PLUS*—

Input Channels

- 16 mono balanced XLR mic/ 1/4″ line inputs with inserts, input trim, 6 aux sends/4 per channel (aux 1 & 2 assignable pre/post), -20dB pad, Solo (PFL and Solo-In-Place), mute, pan, bus 1/2, 3/4 and L/R assignment buttons and overload LEDs
- Each input channel features 3-band active EQ with sweepable midrange (12kHz high shelving, mid eq sweep between 100Hz - 8kHz, 80Hz low shelving eq ±15dB) and 75Hz HPF
- Input channels 1-8 feature 1/4″ balanced direct outputs

Outputs

- 1/4″ balanced main outputs with inserts
- 6 balanced 1/4″ auxiliary send
- Single 1/4″ headphone output with level control

Additional Features

- 5 physical configurations via rotating I/O pod
- Rack mount kit included

DFX-6 / DFX-12

Compact Live Sound Mixers with EFX

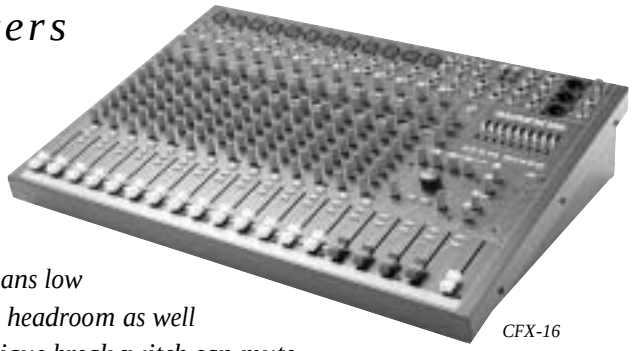


Combining Mackie's sound quality with an extensive feature set, the DFX-6 and DFX-12 are designed specifically for solo or small performing groups that don't have the luxury of a front-of-house sound technician. All knobs are color-coded and can be easily identified at a glance. Channel faders are white. Effects and Monitor Sends are red. Master level faders are blue. The jack field is on the top surface so connections are immediate and every channel has a level set and overload LED that can be easily seen on a dark stage. Both include a combination of mic/line and stereo line inputs, silky smooth 60mm faders, and Mackie's proprietary 32-bit EMAC custom digital effects processor with 16 effects. Each also offers low-noise, low-distortion, studio-grade mic preamps wrapped in a box rugged enough to withstand years of on-the-road wear and tear.

- Low noise, high headroom mic preamps with switchable phantom power
- 5-band stereo graphic equalizer with 12dB boost/cut, Bypass and Aux/Main Assign
- 2-band channel strip shelving EQ at 80Hz and 12kHz
- 75Hz Low Cut filters and inserts on mono mic/line channels
- 32-bit Digital EFX Processor with 16 effects
- Input trim controls with LED level set indicators
- Separate Aux Send and Effects Send for each channel
- 2 stereo Aux Returns
- Master Aux 1, Aux 2/EFX and CD/Tape Return faders
- Effects to Monitor feature with separate rotary level control
- Tape/CD inputs and Tape outputs
- Vocal Eliminator switch for "Karaoke" use
- Break switch for playing music between sets
- Headphone output with level control
- Bal./unbal. XLR and 1/4″ main stereo outputs

CFX-SERIES

12-, 16- and 20-Channel Audio Mixers with Digital Effects



CFX-16

Available in 12 (CFX-12), 16 (CFX-16) and 20-channel (CFX-20) configurations, the CFX mixers incorporate digital effects and a 9-band graphic EQ to offer all-in-one compact mixing solutions. Sounding clean and realistic, the effects processor rivals many outboard processors, while the graphic EQ on the output section is of "audiophile-quality" which means low noise, free of phase distortion. The mixers also feature low noise and high headroom as well as balanced I/Os and multiple busing options. For live performance, a unique break switch can mute the entire mixer (except for the two-track return input) as well as the effects and graphic EQ sections.

Input Channels

- Level-setting LED for each channel.
- Low-noise, high-headroom mic preamps with 50dB of gain; 48v phantom power
- +30dB of line level gain and a full 15dB of attenuation to "pad" hot signals.
- Two Aux Sends with balanced 1/4″ outs, switchable to pre-fader for monitor use and post-fader for use as effects sends
- One external and one internal EFX Send.
- Pan, Mute, PFL solo on each channel
- Pre-Fader Solo switch allows channel monitoring via headphones
- 3-band EQ (12kHz shelving HF, ultra-wide 100Hz to 8kHz bandwidth sweepable peaking Midrange EQ and 80Hz LF).
- Inserts and 100Hz 18 db/octave low-cut filters on all mic/line channels
- 4-band EQ on stereo line channels

Subgroup

- Four Subgroup buses with 1/4″ balanced direct outputs and L/R assign "collect" channel signals assigned to them so you can submix vocals, drums and other audio signals for monitoring or tracking.

Effects

- 32-bit digital FX with 9 reverbs, 4 delays, phaser, chorus, flange; 2 parameter controls and an EFX wide spatial expander and bypass switch

Master Section

- Studio-quality, 9-band stereo graphic EQ
- Tape/CD inputs with level control assignable to Main Mix via Break switch
- RCA tape/CD inputs and tape outputs.
- Two stereo effects returns with balanced 1/4″ inputs

Outputs

- Balanced/unbalanced 1/4″ TRS and balanced XLR outputs for mains with inserts.
- Headphone jack w/level control.
- XLR balanced subwoofer output from built-in 18dB/oct. 75Hz crossover!
- Extra 1/4″ balanced utility stereo outputs with level control (post-Main Fader).

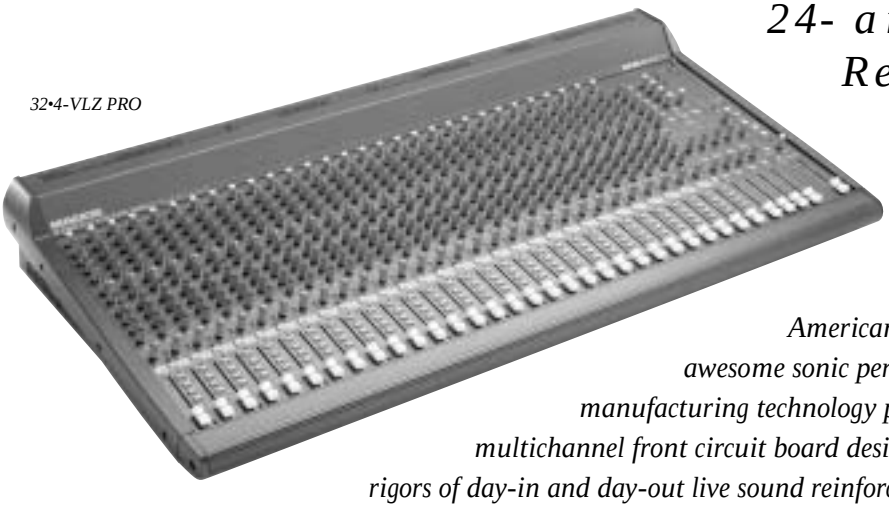
Conveniences

- Break Switch mutes all channels and routes Tape input to the mains while you're on break so you don't have to worry about feedback during breaks
- Long-wearing 60mm logarithmic taper faders give you smooth, linear control throughout the fader's entire travel.
- Rude Solo Light alerts of any current soloed channels

	1202-VLZ Pro	1402-VLZ Pro	1642-VLZ Pro	1604-VLZ Pro	CFX-12	CFX-16	CFX-20
Total Channels	12	14	16	16	12	16	20
Buses	2	2	4	4	4	4	4
Mono Inputs	4 Mic/Line	6 Mic/Line	8 Mic/Line	16 Mic/Line	8 Mic/Line	12 Mic/Line	16 Mic/Line
Stereo Inputs	4 Line	4 Line	2 Mic/Line, 2 Line	–	2 Line	2 Line	2 Line
XDR Mic Preamps	✓	✓	✓	✓	–	–	–
EQ Mono Channels	3-Band (Fixed)	3-Band (Fixed)	3-Band (Mid-Sweep)	3-Band (Mid-Sweep)	3-Band (Mid-Sweep)	3-Band (Mid-Sweep)	3-Band (Mid-Sweep)
EQ Stereo Channels	3-Band (Fixed)	3-Band (Fixed)	4-Band (Fixed)	–	4-Band (Fixed)	4-Band (Fixed)	4-Band (Fixed)
Aux Sends/Channel	2 Mono	2 Mono	4 Mono	6 Mono	2 Mono	2 Mono	2 Mono
Aux Returns	2 Stereo	2 Stereo	4 Stereo	4 Stereo	2 Stereo	2 Stereo	2 Stereo
Direct Outputs	–	–	8	8	–	–	–
Main Outputs	XLR, 1/4″ Bal.	XLR, 1/4″ Bal.	XLR, 1/4″ Bal.	XLR, 1/4″ Bal.	XLR, 1/4″ Bal.	XLR, 1/4″ Bal.	XLR, 1/4″ Bal.
Mono Output			1/4″ Bal.	1/4″ Bal.	XLR Bal.	XLR Bal.	XLR Bal.
Internal Effects	–	–	–	–	✓	✓	✓
9-Band Graphic EQ	–	–	–	–	✓	✓	✓

MACKIE

24•4-VLZ PRO/32•4-VLZ PRO



32•4-VLZ PRO

24- and 32-Channel Sound Reinforcement Consoles

First introduced almost five years ago (but updated many times since), Mackie's SR24•4 and SR32•4 have totally redefined the 24-and 32-input/4-bus mixer. Feature-packed, extremely affordable, American-made mixing consoles, complete with awesome sonic performance, they incorporate the very latest in manufacturing technology plus Mackie's own proven horizontal, multichannel front circuit board design. They are also designed to stand up to the rigors of day-in and day-out live sound reinforcement, yet still have the required dependability and pristine audio quality for professional studio applications as well.

The SR24•4 and SR32•4 provide superior mixing and recording. They feature major headroom, thanks to Mackie's VLZ circuitry. Main and submix amplifiers use distinctive negative-gain mix architecture to double the mix headroom of conventional designs. Ultra-quiet high-headroom mic preamps on 20 (SR24•4) or 28 (SR32•4) mono channels mean great-sounding vocals or instruments. Low-cut filter (on mono channels) centered at 75Hz allows you to cut out mic thumps and wind noise. Four buses make live mixing a breeze by letting you control the overall level of those channels with one fader. Six aux sends per channel offer the flexibility to handle multiple chores at once. Two aux sends are switchable to pre- or post-fader so you can have 4 pre-fader aux sends, or 4 post-fader aux sends. They also feature a sweepable midrange EQ (on mono channels) with midrange centering anywhere from 100 Hz to 8 kHz.

FEATURES

- They combine premium XDR mic preamps with 60dB gain range, 130dB dynamic range for hot 24-bit digital signals, massive +22dB input handling capacity, 192kHz bandwidth and 0.0007% THD with the best RFI resistance of any compact SR mixer in the galaxy thanks to advanced DC-pulse transformers.
- 20 (SR24) and 28 (SR32) XDR mic preamps are impedance-independent, so the frequency response remains constant whether the mic preamp is presented with an extremely high- or low-impedance load. Additionally each channel has its own switchable low-cut filter (18dB/octave at 75Hz), allowing you to cut out mic thumps and wind noise that can enter your mix and rob you of amp power.
- They have 20 (SR24) and 28 (SR32) mono mic/line input channels, with XLR mic inputs and 1/4″ TRS line inputs, and 2 stereo line input channels with 1/4″ TRS inputs.
- Each channel strip has an input trim, 6 aux sends (2 prefader, 2 post-fader and 2 switchable pre or post), 3-band EQ with sweepable mid-range (4-band EQ on the stereo channels), pan control, and mute, solo and bus assign switches. A 60mm fader provides output gain for each channel.
- The mono channel EQs provide a range of ±15dB at the following frequencies: 12kHz shelving high-frequency EQ, 100Hz to8kHz sweepable peaking mid-frequency EQ, and 80Hz shelving low-frequency EQ.
- Stereo channels (21-24) provide ±15dB of boost and cut at the following frequencies: 12kHz shelving high-frequency EQ, 3 kHz peaking high-mid frequency EQ, 800Hz peaking low-mid frequency EQ, and 80Hz shelving low-frequency EQ.
- You can mix any combination of channel strips down to a single submix bus—or any combination of the four submix buses. Channels can be assigned to buses 1-2, 3-4, and Main Mix L/R, and the 4 sub can be assigned to Left and Right Main Mix, or fed directly to a multitrack recorder. This is incredibly helpful, not only in a live situation, but in the studio as well. Furthermore, each bus is "double-bused" providing 8 eight outputs that can be connected to an 8-track recorder without repatching anything.
- Each sub out (1-4) and main out (L/R) has a TRS insert jack. Furthermore, Mackie's unique "AIR" EQ circuit is included on each of the four subgroups. This peaking EQ circuit, centered at 16kHz, enhances guitars, vocals, percussion, whatever, giving a gentle "lift" to the extreme high end. The result is extra detail and fidelity in the high end.

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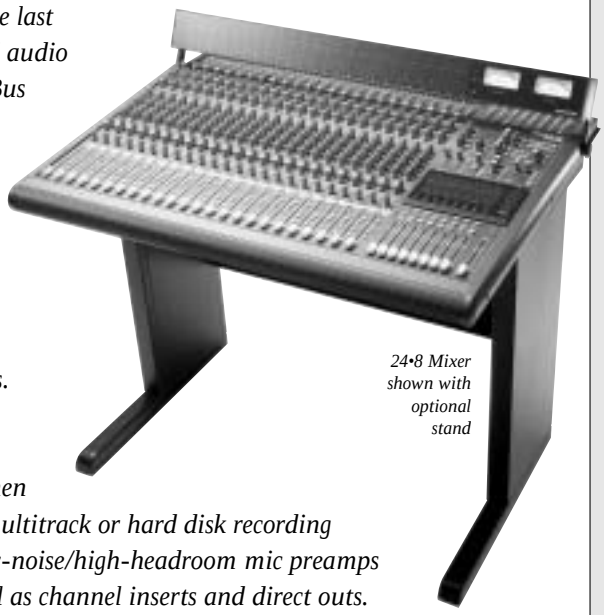
EQUIPMENT LEASING AVAILABLE

24•4-VLZ PRO/32•4-VLZ PRO

- Outputs include XLR and 1/4″ TRS line outputs for the left and right mains, 1/4″ TRS line outputs for subs 1-4, and an XLR mono main output. The mono main out has its own level control so a mono mix can sent to another zone and adjusted accordingly.
- The Phones/Control Room switch and level control are connected to stereo headphone outputs and left and right Control Room output, allowing the stereo Tape Return, Left/Right Main Mix, and Solo to be monitored.
- A stereo playback device can be monitored via the Tape Return inputs. Tape Return to Phones/C-R routes the tape playback signal into the monitor system and meters, and the Tape Return knob adjusts the level of playback, which can be monitored via headphones.
- Tape return signals can also be assigned directly to the Main Mix. This not only routes Tape Returns to Main L/R outputs but also disables all other inputs to the mains. You can play a tape or CD between sets without losing a channel and submaster settings. They also have RCA-style tape outs for output to conventional stereo recording devices.
- Each of the six aux sends has its own individual master send control, driving 1/4″ TRS output jacks. Six stereo aux returns are provided, with 1/4″ TRS input jacks. Two aux returns can be folded back into AuxSends 1 and 2 via their own volume controls to add effects in stage monitors.
- An XLR input is provided for a talkback mic, which can be assigned to the Main Mix (for making announcements over the mains) or Aux 1-2. The talkback mic has a level control in the talkback section.
- Like all Mackie mixers, the SR24•4 and SR 32•4 are designed for rugged 24-hour-a-day use. With their multiple input/output configurations, true 4-bus architecture, 6 aux send and extensive routing capabilities, they can be used in a variety of live sound and recording applications. Their sturdy steel construction houses rugged, double-sided SMT-plated fiberglass circuit boards, and 60mm faders with ultra-tight lip seals for keeping out dust and other contaminants. Impact-resistant knob are mounted so they "ride" just above the steel chassis.

16/24/32•8

8-Bus Recording/PA Mixers



24•8 Mixer shown with optional stand

Designed to eliminate the last barrier between you and audio creativity – Mackie's 8-Bus mixers have recorded more platinum albums and major motion picture soundtracks than any other mixer in their class – in fact, they set the standard for affordable 8-bus consoles. Excellent for project studios and digital multitrack recording, when combined with digital multitrack or hard disk recording systems, they feature low-noise/high-headroom mic preamps on every channel, as well as channel inserts and direct outs. In the channel strip you'll find Mackie's truly musical EQ – with Hi and Lo shelving EQ, parametric Hi-Mid, and sweepable Low-Mid EQ – as well as a Low Cut Filter, which allows you to eliminate mic thumps and room rumble from your mix, and use your EQ for music and vocals. With Mix B, "in-line monitoring" is a breeze. Six aux sends and six aux returns allow you to use all the effects you want. And the 8•Bus's complete talkback and phones level controls make communicating with the talent a snap. And besides being built like tanks, the 24•8 and 32•8 can grow along with your budget and input requirements. Both can be expanded in increments of 24 channels with the 24E Channel Expander Console. You can also add meter bridges, stands, and even a "sidecar" for patchbays.

	32• Bus	24• Bus	16• Bus
Total Channels	32	24	16
Buses	8	8	8
Mix B Inputs	32	24	16
Mic Preamps	32	24	16
EQ	4-band parametric	4-band parametric	4-band parametric
Aux sends/ch.	6	6	6
Aux Returns	4	4	4
Tape Outputs	24	24	24
Channel Inserts	32	24	16
Faders	100mm	100mm	100mm

MACKIE

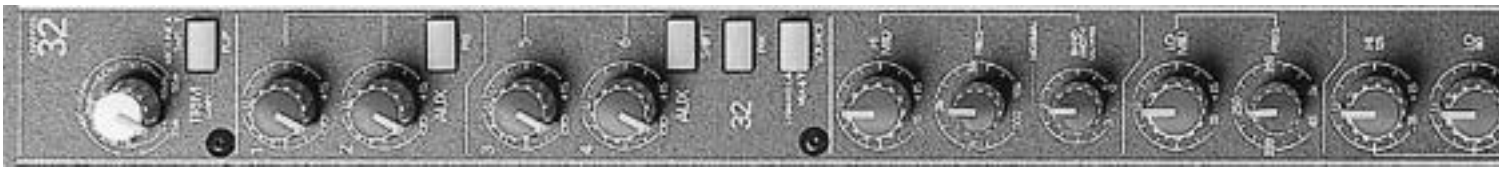
16•8 / 24•8 / 32•8 8•BUS MIXERS

Channel Strip

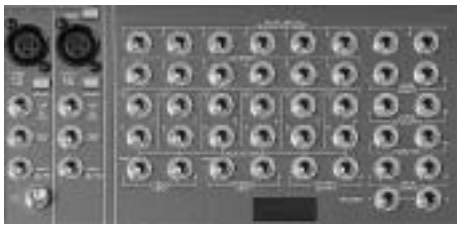
- In-line FLIP reverses tape and mic/line inputs between channel strip and Mix-B/Monitor section
- Each channel strip has 6 mono auxiliary sends with several routing options:
- Auxes 1 & 2 have two independent level controls that share a common Pre/Post switch.
- Aux 3-4/5-6 are accessible from two level controls: A 'shift' button changes Auxes 3 4 to 5-6. A 'source' switch selects signal source of Aux 3-4/5-6 from channel strip or channel's Mix B/Monitor send so you can build and effects mixer (pre or post-Mix-B level) to assign to phones during tracking.
- True parametric, 3-control Hi Mid EQ with ±15dB boost/cut: Ultra-wide 500Hz to 18kHz frequency sweep range; bandwidth can be adjusted from a very wide 3-octave width to a very narrow 1/12-octave width.
- Low Mid EQ with ultra -wide 45Hz to 3kHz sweep, ±15dB boost/cut
- ±15dB shelving Hi (12kHz) and low (80Hz) EQ
- Multi-purpose 18dB/oct. low cut filter cleans up "mix mud", cuts PA rumble, creates a "neo-peaking" bass control when used with LO shelving boosts
- Independent Mix-B (Monitor) section—think of it as as discrete "channel strip within a channel strip"—with its own pan, level, EQ and source capabilities. During mixdown, use as extra pre-fader stereo Aux send or double your inputs
- Mix-B Split EQ assigns Hi and LO EQ to Mix-B
- Mix-B Source selects from flip switch or channel strip (pre-fader) for in-line monitoring while recording or gives an extra stereo aux bus
- Constant power, buffered panpot for rock-solid panning
- +22dB Overload LED (monitors three critical points in the signal chain, and displays the one that's highest at any given time) and Hyperactive -20dB Signal Present LED (can also tell you at a glance what vocal or instrument is on what channel)
- Selectable Solo with Channel Metering allows soloing in full stereo perspective: displays soloed channel operating level on master L/R meters so input trims can be adjusted for optimum levels.

Top of the Channel Strip (not shown)

- DC phantom power for condenser microphones is applied to channel strips in groups of eight. It has a "ramping" function that gradually increases from 0 to 48v power when you turn it on preventing thumps and pops.
- Two BNC sockets for 12v gooseneck lamps
- Balanced mic inputs lead to the VLZ mic preamps. Mackie's VLZ (Very Low Impedance) design provides terrific dynamic range with an EIN of -129 dB and 10Hz to 300 kHz bandwidth.
- Balanced/unbalanced 1/4″ jacks with 40dB range handle everything from hot digital inputs to older keyboards with low output voltages.
- 1/4″ 4dB direct out jack (connected to the output of the channel, post-EQ/post-fader/post-mute switch
- Channel Insert allows you to insert external serial processing equipment (eg. compressor, gate) into the main signal path of the input.



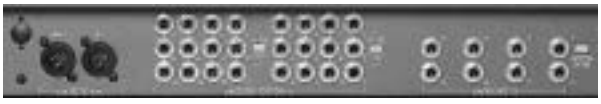
Master I/O Section



- Main and submaster inserts allow you to insert a compressor or EQ into main L/R or any of the 8•Bus submaster circuits.
- Aux Returns are unbalanced stereo with a L&R input for each return channel. If you only have one return signal, plugging it into the Mono/L jack will cause it to be connected t both the left and right return inputs and end up centered in your stereo image.
- Control Room output, Main Mix output, Mix-B output and Studio output
- 2-Tk Input and External Inputs offer you the ability to listen to two stereo sources directly, without patching through input channels.

Rear Panel

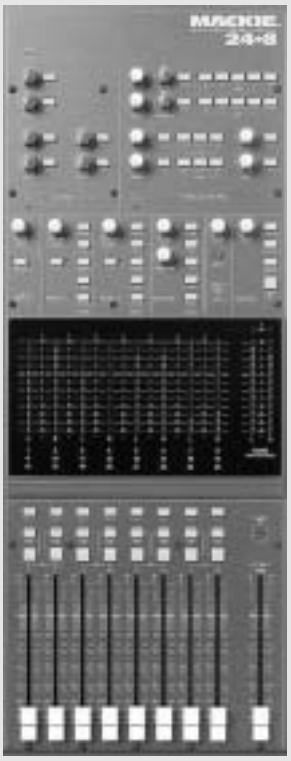
- Three tape output jacks per bus (total of 24). Balanced outputs, switchable from +4/-10.
- +4dBu balanced Tape Returns, switchable to -10dBV in banks of 8 returns
- "Triple busing" allows you to feed a 24-track deck without having to constantly repatch. When you send a signal to Submaster 1 output, for instance, it will appear as Submaster Outputs 1, 9, and 17. If you have a 16-or 24-track deck, simply patch these outputs to the corresponding multitrack inputs. Now the tracks on your multitrack that are in Record mode will accept the signal, while the tracks in Safe mode won't.



16•8 / 24•8 / 32•8 8•BUS MIXERS

Master Section

- Six Stereo Aux Returns. All have 20dB gain, Solo, and can be used in stereo and mono. Aux return 1 & 2 are pannable and bussable. 3 & 4 are assignable to the phones for 'wet' monitoring.
- 6 Aux Sends with Solo and Solo LED
- Talkback assigns to all submasters, main mix, Aux 1, Aux 2 or Phones 1&2
- Solo level adjust and ultra-rude LED
- Monitor section with separate control room and studio levels, source selection between L/R mix, Mix-B, tape and external. Can be switched to mono.
- Two separate headphone sections can be used totally independent of each other. Each features source selection between control room and any combination of Aux 3/4, Aux 5/6, Mix-B or external source. Solo allows control room to hear what musicians are hearing in their headphones.
- Mix-B/Monitor section can be used as an independent stereo out for PA monitor mix, 2-track recording, video/broadcast feed or assigned to L/R mix.
- -40 to +10 bar graph LED displays for each sub-master and Solo/Main (with main L/R +22dB Clip LEDs)
- Expansion consoles let you add channels in banks of 24 to either the 24•8or 32•8. Expanders have their own internal mix amps so the main board only "sees" one extra channel per expansion console.
- Trick Bus Solo switches send odd-numbered buses to the left speaker and even-numbered buses to the right speaker—unless you've pressed the respective Mono L/R button. When a bus has been mono-ed, Solo sends the bus to both speakers
- L Mix/R Mix and Mono L&R buttons assign buses to main L/R stereo bus
- Built-in Talkback Mic with level control lets you scream engineering commands to any combination of Aux Send 1, Aux Send 2, Tape/Submasters (L/R Mix and the eight submasters) and Phones/Studio. When Talkback is engaged, control room outputs are padded by 20dB to avoid feedback.





8•BUS SERIES ACCESSORIES

Meter Bridges

Matching meter bridges (MB•16, MB•24, MB•32) are available for each mixer. Each provides 12-segment LED ladders for each input channel, and lighted VU meters for master Left/Right output. Like the mixers' integral meter sections, the LEDs in the bridges are with Lexan to resist dust and humidity. On each meter bridge, input buttons allow you to globally switch between Tape Return input and Channel Strip post-fader output. Each meter attaches in minutes and can be tilted as desired (up to 90°). They also tilt down fully flat to save you inches of road case depth—and from having to buy another case just for the meter bridge.

The Stand

Any 8•Bus (including the 24•E Expander) can be floor-mounted via the optional The Stand. Made of rugged steel with ultra-tough, 2-part polyurethane coating, The Stand assembles in minutes.

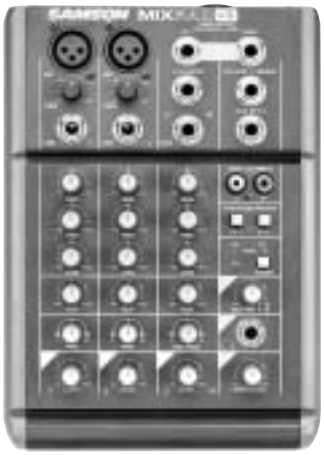
24•E Expander Console (for 24•8, 32•8 only)

The 24•E is essentially another 24-channel 8•Bus mixer—but without a master output section. You get 24 complete input channel strips and 24 more tape returns. The 24•E connects to the 24•8 or 32•8 via a multi-pin connector. Additional 24•E's can be daisy-chained to create 48, 56, 72, 80 or 96 more input channels. The 24•E's outputs are pre-mixed in the expander to reduce line and thermal noise and to maintain maximum sonic quality at the main console. The connecting cables are long enough so that you can place one or Expanders on each side of your main 8•Bus mixer.

SAMSON

MIXPAD 4

Compact 4-Channel Audio Mixer



An incredibly compact 4-input stereo mic/line mixer with exceptional sound and a host of features, the Mixpad 4 also offers AC/DC capability, making it ideal for use in ENG and field recording applications as well as in the studio. Exceptionally professional from its high headroom design (+27dB) to its durable extruded aluminum chassis, the Mixpad 4 also shines as a sub-mixer in live and recording settings.
Features include 3-band EQ and pan/balance and level controls on each channel, a front-panel stereo headphone jack, two balanced mic/line inputs (XLR and 1/4˝ connectors), 1/4˝ stereo line input, CD/tape input, 48-volt phantom power, an Aux send and a stereo Aux return (with level control which can be used as 2 mono aux returns), as well as a main level control and power and peak LEDs.

- Two balanced (XLR and 1/4˝) mic/ line and one stereo input (4 channels total)
- Trim controls on mic input channels
- 48v phantom powered mic inputs
- Independent 3-band EQ for each channel, with ±15 dB of cut/boost for low (80 Hz) and high (12 kHz) frequencies, and ±12 dB of cut/boost for mid (2.5 kHz) frequency
- Independent CD/tape input; signal is routed via channel EQ
- Constant-level pan controls (mono channels) and balance controls (stereo channel)
- One Aux send (pre-fader)
- One stereo effects return with front panel level control
- Peak overload LEDs on left and right main outputs for monitoring
- High slew rate allows it to react quickly to transients and to maintain crisp sound.
- Balanced stereo output with 1/4˝ jacks for connection to amps and other gear
- Front-panel headphone output
- An inline power supply maintains the mixer's low profile while eliminating hum.
- Battery operated, the mixer can run for 12 hours on three 9V batteries (the front-panel Power LED also indicates battery strength).
- Includes a carrying strap for easy portability.

MPL-1602

Rackmount 16x2 Line Mixer



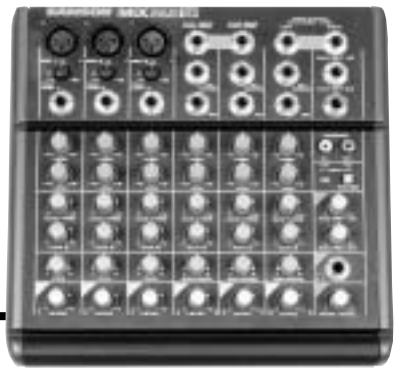
The MPL-1602 provides a 16 x 2 stereo mic/line mixer in an easily installed 2RU high package. Equipped with 2-band EQ and balance and level controls on each channel, the MPL-1602 is ideal for recording, keyboard setups, broadcasting, live concert and contracting applications. It also features 2 main left/right faders, a stereo headphone jack with dedicated level control, 16 electronically-balanced 1/4˝ line inputs, 2 XLR mic inputs with trim controls, 2 pre-fade Aux sends and 2 stereo Aux returns, along with 6 channel insert and 2 main bus insert patch points. A highly practical audio tool for all kinds of professionals, the MPL-1602 is a great way to add more inputs with mixing capability to keyboard setups, small live sound systems, Techno bands and multimedia situations.

- 8 balanced stereo line input channels
- 2 XLR-balanced mic inputs with trim controls on rear panel
- Level and balance controls on each channel
- 2-band equalization on all channels
- 2 aux sends on each channel
- 2 stereo aux returns (which have independent level and balance controls allowing for use as 4 mono Aux returns)
- 2 balanced main L/R bus outs
- Front-panel stereo headphone output with dedicated level control
- 6 channel inserts and 2 main bus inserts for increased flexibility

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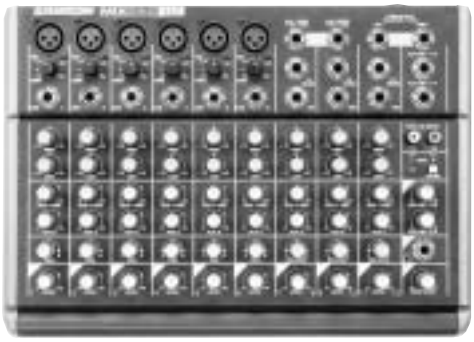
MIXPAD 9 / MIXPAD 12

9-and 12-Channel Compact Audio Mixers



Featuring a high headroom design for matchless audio performance, the deceptively sized Mixpad 9 is small but professional in every respect. Like the Mixpad 4 it offers superior low-noise and low distortion specifications along with an exceptionally low group delay over the full bandwidth for a transparent, open sound. An ideal choice for smaller live sound situations such as clubs, acoustic setups, keyboard rigs and lounge acts, project recording and more.

Remarkably compact 9- and 12-channel mixers, the Mixpad 9 and 12 offer professional audio performance and a wide range of user-intensive features for virtually any type of application. They boast superior low noise and low distortion specifications, include wide-range gain trim controls for both mic and line inputs, provide very transparent, open sound, and they have a very high slew rate—usually found on larger, more expensive mixers. They offer phantom power on all mic inputs for use with condenser mics and an in-line power supply eliminates magnetically-induced hum.



Fully professional, the Mixpad 12's complete feature set and high-quality audio performance allows it to function very effectively in the studio, sound reinforcement, field recording, audio for video and film applications and more. Almost identical except for their inputs, both feature 2-band EQ and pan/balance and level controls on each channel, a front-panel stereo headphone jack, 3 electronically balanced mic/line inputs (XLR and 1/4″ connectors), three 1/4″ stereo line inputs, a CD/tape input, 48-volt phantom power, 2 Aux sends and 2 stereo Aux returns (which have independent level controls and so can be used as 4 mono Aux returns), as well as a main level control and power and peak LEDs.

- In the MIXPAD 12, a total of twelve input channels (including three stereo channels); in the MIXPAD 9, a total of nine input channels (including three stereo channels);
- Mono channels provide electronically balanced inputs that can be used for mic or line-level input, while stereo channels are ideal for line-level sources such as outboard signal processors; CD players; tape or cassette recorders; stereo drum machines; and keyboards and MIDI tone modules.
- An electronically balanced main stereo output for connection to a power amplifier or tape recorder.
- Dedicated Tape/CD input
- Two auxiliary sends and two stereo auxiliary returns (which can be used as four monophonic returns). Aux send 1 is pre-fader (but post-equalizer), making it ideal for use as a headphone or monitor cue mix, while Aux send 2 is post-fader and post-equalizer.
- Independent 2-band EQ for each channel, with 1±5 dB of cut/boost for low (100 Hz) and high (10 kHz) frequencies.
- Constant level pan controls for placing each mono channel in the left-right stereo spectrum, as well as balance controls that let you blend the relative levels of stereo inputs.
- Mic input trims are continuously adjustable from +4 to -50 dB, making it possible to use them with a wide variety of microphones.
- Provide 48 volts of phantom power to all mic inputs
- Peak LEDs for the left and right main outputs, showing you when signal is overloading or near overloading.
- Center detents for all pan, balance, and EQ controls, making it easy to access them in low-light situations like live performance.
- Convenient front-panel Power switch, Power LED and dedicated headphone jack.

	Mixpad 4	Mixpad 9	Mixpad 12
Frequency Response	±3dB 10Hz to 50kHz	±3dB 10Hz to 50kHz	±3dB 10Hz to 50kHz
Signal To Noise Ratio	95dB	95dB	95dB
THD	< 0.01%	< 0.01%	< 0.01%
Dimensions (W x D x H)	6.4x 9 x 2.7″	9.4 x 9 x 2.3″	12.8 x 9 x 2.3″
Weight	2 lb.	5.5 lb.	6.5 lb.

SAMSON

MDR SERIES

Samson's MDR (Maximum Dynamic Range) mixer series of mixers offer transparent sound, low noise and high quality mic preamps at very affordable prices. Each mixer features low impedance circuit design, 60mm faders, combi XLR connectors, phantom power, steel construction and all but the MDR 6 have built in 24-bit DSP. Samson's exclusive Hard Disc Record Mode (MDR 6 and MDR 8 only) eliminates delay between instrument and monitor mix when recording to hard disc.

MDR 6 *6-Input Stereo Mixer*



The compact MDR 6 is a six-input stereo mic/line tabletop mixer in a portable and rugged package. Perfect for a small studio, the MDR 6 features 60mm faders, 3-band EQ, Aux send and pan or balance on each channel. There are four high quality mic preamps, an Effects return, dual 5-segment LEDs and a headphone out and mono out both with level controls. The left and right mix out also feature 60mm faders.

- 60mm faders on each channel and mains
- 3-band EQ on each channel
- Four XLR inputs with mic preamps
- Mono out with level control
- Dual 5-segment LEDs
- Aux send with effects return
- Hard Disc record mode for recording with a PC
- Headphone out with level control
- In-line power supply



MDR 8 *8-Input Stereo Mixer*

The MDR 8 is an eight input stereo mixer with 24-bit DSP featuring 60mm faders, 3-band EQ, two Aux sends and pan or balance on each channel. Included are six high quality mic preamps, two Effects returns Dual 6-segment LEDs, a headphone out and mono out both with level controls. The left and right mix out also feature 60mm faders.

Step-up features from the MDR 6—

- Six XLR inputs with mic preamps
- 2 Aux send with 2 effects return
- 24-bit DSP effects (lush Hall, Room and Vocal reverbs, Chorus and Delay)
- Dual 6-segment LEDs

MDR 10 *10-Input Stereo Mixer*



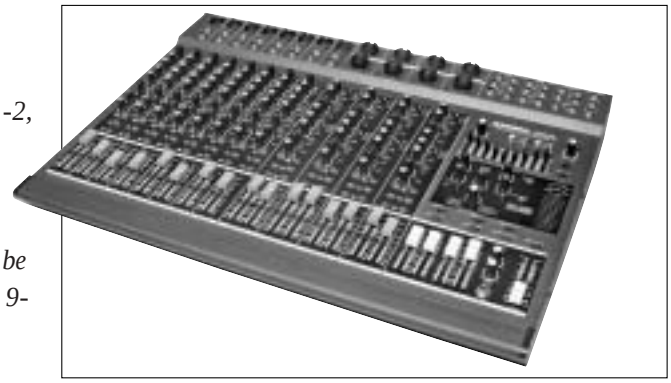
The 10-input MDR 10 packs professional features into a compact package. All 10 channels feature a high quality mic preamp, 60mm faders, a pan control and a 1/4˝ and XLR input (with mic preamp). Channels 1-6 offer a gain control, 3-band EQ with sweepable mids, a low-cut filter, two Aux sends (1 switchable pre/post, 1 to the DSP), Solo and Mute switches and smooth 60mm faders. Channels 7-10 feature 2-band EQ, 1 Aux send (switchable to Aux 1 or 2), Solo and Mute switches and 60mm faders. The outputs and bus 3 and 4 are on 60mm faders and the master section features selectable DSP, headphone out with level control, effects returns, mono out, phantom power and more.

Step-up features from the MDR 8—

- 10-channel stereo mixer with 2+2 bussing
- 10 XLR inputs with mic preamps
- 3-band EQ with sweepable mids and a low cut filter on channels 1-6
- Dual 12-segment LEDs
- I2-track in and out connectors with send to mix switch

MDR 16

16x4 Stereo Mixer



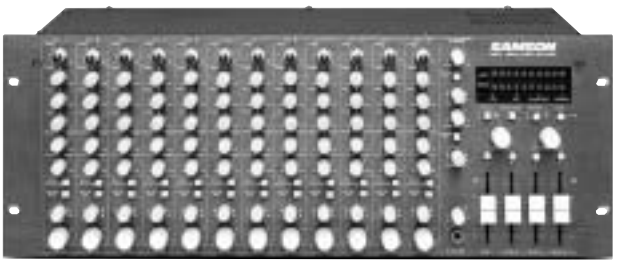
The flagship of the series, the MDR 16, is a true 4-buss mixer. All 16 channels feature a high quality mic preamp, gain control, Solo, Bus 1-2, Bus 3-4 switches and 60mm faders. Channels 1-8 offer 3-band EQ with sweepable mids, a low cut filter, two Aux sends (1 switchable pre/post, 1 to the DSP) and channel inserts. Channels 9-16 have 2-band EQ and a switchable Aux send. The four 60mm bus faders can be routed to the left/right fader. The master section features 24-bit DSP, 9-band stereo EQ, dual 12-segment metering, Solo, PFL, 2-track out, headphone and mono out with level controls and more.

Step-up features from the MDR 10—

- 16-channel stereo 4-bus mixer
- 16 XLR inputs with mic preamps
- 60mm faders on channels, busses and mains
- 3-band EQ with sweepable mids and a low cut filter on channels 1-8

MPL-1204

Rackmount 12x4 Mic/Line Mixer



"MPL" stands for "Mic/Program/Line" and the name describes the broad range of signals which can be handled by this powerful mixer. In fact, the compact design of the MPL-1204 belies its extraordinary versatility. A great all-around audio tool, the 4RU high MPL-1204 is designed for use in live and monitor mixing, keyboard setups, recording and fixed installations, conference rooms, distributed sound and as an extra mixer to expand home recording setups.
A true 4-bus, the versatile MPL-1204 features 12 input channels with mic and line trim pots, 3-band EQ, two Aux sends (1 pre, 1 post) plus solo and bussing switches. There are independent faders for each bus, and stereo Main and Control Room volume controls provide complete flexibility in monitoring and routing.

- Twelve independent channels with balanced XLR and 1/4˝mic and line inputs; also has a stereo CD/Tape input with dual RCA jacks.
- 48V phantom power for all 12 mic inputs
- Constant level Pan controls for each channel that allow you to precisely place each input signal in the left-right plane.
- Independent 3-band EQ for each channel, with 15dB of cut/boost for low (100 Hz) and high (10 kHz) frequencies, and 12 dB of cut/ boost for mid (1 kHz) frequencies.
- Channel inserts for all twelve channels, enable use of outboard signal processors such as EQs, compressors or noise gates.
- Center detents for all Pan and EQ controls make it easy to use in low-light situations.
- Flexible front panel metering system with 10-segment level meter lets you view at a glance the levels of the main Left/Right output, as well as main power, phantom power, and PFL or AFL status.
- Four bus outputs and an electronically balanced main stereo output, with dedicated front panel control. Any channel input can be routed to either of the two bus pairs (1/2 or 3/4), and any of the buses can be routed to the Left/Right mix output with the touch of a button. This flexible design allows you to easily mute selected channels and/or to create submixes within your main mix.
- Independent XLR-balanced Main stereo outputs as well as balanced stereo Control Room outputs with dedicated level control.
- Bus inserts allow external signals to be sub-mixed into any or all of the four buses and also enables linking of multiple mixers.
- Two auxiliary sends on each channel. Aux send 1 is pre-fader, Aux send 2 is post-fader.
- Two stereo auxiliary returns (route to any of the buses). with front panel level controls
- Selectable Pre Fade Listen (PFL) or After Fade Listen (AFL) soloing for each channel. Both are non-mix-destructive in that they do not affect the signal being output either by the Main, Bus, or Control Room outputs.
- Front-panel headphone output with volume control for monitoring of soloed channels.
- Mounts in any 19" rack (4RU), making it easy to integrate into existing systems.

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FOLIO NOTEPAD

Compact Audio Mixer

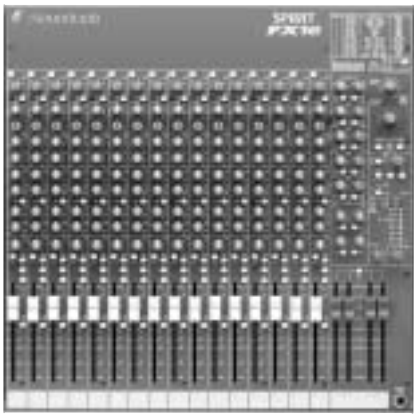


The ultra-compact and very affordable Folio Notepad is packed with enough features to handle a surprisingly wide range of mixing tasks. Less than 10˝wide, it can handle up to 10-inputs, configured to give you the maximum choice of source signals, from condenser mics to turntables. Its high-quality components and advanced design ensure CD-quality sound, while its ergonomic layout makes it very easy to use. Features include separate mix and monitoring outputs, custom-built pots for precision control and carefully-tailored responses, and an external power supply that guarantees hum-free mixes with high RF rejection. Custom controls give you smooth, even response, while high quality pre-amps let you connect mic or line signals without noise or overload. Ideal for video editing, desktop recording, houses of worship, small live venues, DJ's and more.

- 10 inputs
- 4 mic inputs and 2 stereo inputs
- High quality mic preamp inputs
- 2-band EQ on all 4 mono channels
- Post-fade aux send on every input
- Stereo effect return
- Switchable +48V phantom power
- 2-track tape return
- Stereo inputs with switchable RIAA pre-amps for turntables
- Separate mix and monitor outputs
- Peak and VU metering
- Headphone output
- Custom designed rotary controls for consistent, accurate response

FOLIO FX16

16 Mic/Line Input Mixer w/Lexicon Effects



Spirit FX16 is a flexible 4-bus mixer capable of producing Digital sound quality for live and recording applications. It features a specially designed 16 program Lexicon Effects Section with dual effect capability (including Chorus and Reverb, Chorus and Delay , and Reverb and Delay) and fully editable and storable programs and parameters. The console itself includes 16 mic/line inputs and 26 inputs to mix in total (including FX returns and tape return to mix). In addition to the mix outs, two sub-buses allow groups of instruments to be sent to multitrack, to additional speakers, or sub-grouped to mix. FX16's 16 Direct Outs are individually fader pre-post switchable so they are equally useful for recording in the studio or at a gig. In keeping with the multipurpose nature of FX16, both Solo In Place and PFL solo are available, for studio monitoring and channel gain set-up applications respectively.

- All FX16's 16 mic/line inputs are equipped with UltraMic preamp which provides a full 60dB of gain range and +22dBu of headroom, meaning that the FX16 input stage can handle virtually any mic or line device.
- In the EQ section, a "truly British 3-band EQ" with swept mid benefits from custom designed pots which give greater control across carefully chosen frequencies.
- There are 4 Auxiliary Sends, including a dedicated Lexicon effects send, 1 pre-fade send and 2 pre/post-fader selectable send which are equally useful as extra foldback send in monitor-heavy live applications, or as effects sends in studio mixdown situations.
- In addition, there are four Stereo returns, a separate Mono Sum Output and 2 Subgroup Outputs.
- 100mm faders are used throughout, giving you accurate control during complex mixes, and all pots have been custom designed to give even and consistent response around their entire sweep.
- All of these features in a rugged, compact frame which can be optionally rack-mounted into a 10U space via FX16's rotating connector pod which allows leads to be connected conveniently behind the rack.

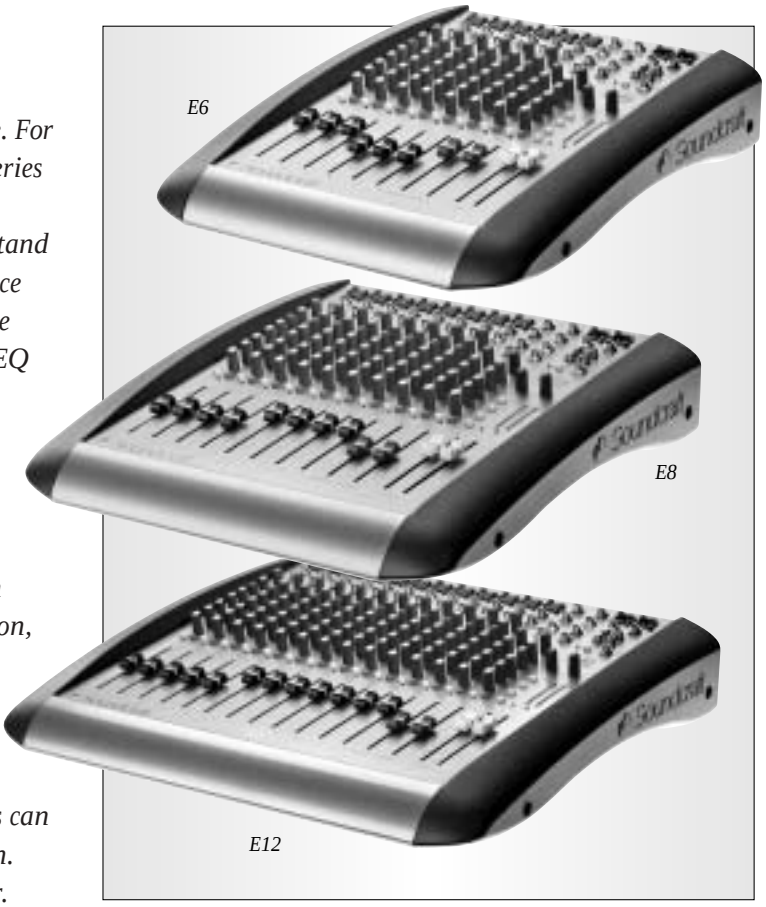
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SPIRIT E SERIES

6-, 8- and 12-Input Mixers

Looking for a simple, easy to use mixer that delivers an exceptional audio performance? The Spirit E Series is here. For recording, live, install or broadcast use there's a Spirit E Series mixer for you.

The emphasis with the Spirit E Series is an easy to understand control surface uncluttered by unnecessary facilities. Surface mount technology is used throughout, using close-tolerance components for high accuracy and repeatable settings for EQ and gain controls. The mic amp features high-resolution adjustment over a wide gain range of 55dB, and provides a stunning +22dB headroom through the console. The E Series is available in 4 standard models: 6, 8 and 12 mono inputs (each model featuring two stereo inputs) and the unique ES. The ES caters for set-ups with multiple stereo sources that require simultaneous connection, such as keyboards, samplers and computers, and has 10 full-function stereo inputs as well as four mono inputs (for microphones and other mono sources). Two of these stereo inputs are equipped with RIAA - equalized inputs on RCA Phono connectors so that record turntables can be plugged straight into the mixer for DJ music production. The ES is ideal for touring keyboard players as a submixer.



FEATURES

- 6, 8, and 12-mono input channel frame sizes, each with two stereo inputs
- ES Version has 10 full-function stereo inputs and 4 mono inputs
- High quality, ultra-linear mic preamps with smooth gain control
- True, professional +48V phantom power for condenser microphones
- 2 aux sends, each globally switchable pre or post fade
- All mono input channels have TRS insert sockets and inserts are also provided on the mix output.
- EQ on the standard mono input is 3-band with a swept mid, while the standard stereo input has two band EQ.
- Close-tolerance surface mount components for high-accuracy repeatable EQ
- Peak LED uses multipoint signal take-offs to watch for overload in several parts of the channel strip (post-mic pre, post-EQ, post-fade), and the LED lights more brightly as the signal approaches the peak point. .
- 100mm faders are used throughout, giving you accurate control during complex mixes
- Professional insert points for external processing
- Each mixer has 2 auxiliary buses, configurable to work the way you do, and all main connectors are professional XLR and 1/4″ metal jack sockets for reliability.
- RCA phono connectors are provided for disc and stereo playback inputs and record outputs.
- Effective grounding eliminates noise and crosstalk
- Main stereo mix has two 10-segment LED meters
- Intuitive and comprehensive solo system and a headphone output
- The monitor output and headphone output work in parallel so that performers can still listen on phones while an engineer is listening on studio monitors.
- The AC mains inlet for the integral universal switched-mode power supply is recessed underneath the console, allowing the mixers to be put against a wall with the AC cable routed under the sculptured side cheeks or mounted in a rack without the AC cable protruding on the top.
- Each mixer can be easily and quickly converted for rack mounting, simply by removing the side cheeks and adding the optional rack rails.

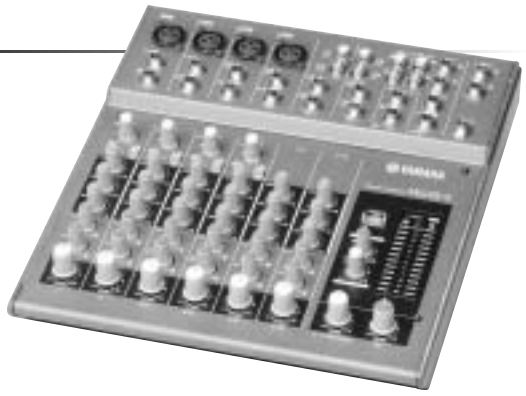
MG SERIES

Analog Mixers



Yamaha's analog MG mixer series cover multiple applications, including PA, project studio, classroom, house of worship, personal monitoring and sub-mixing uses. The MG-series features six models ranging in size from the small MG10/2 10-channel/2-bus unit right up to the very flexible MG32/14FX 32-channel/14-bus type with an impressive selection of built-in effects. There have been no compromises. The mixers are built for great sound, total control, and superior reliability. In fact, they undergo the same rigorous quality and reliability tests as Yamaha's world-class PM-series mixing consoles. Yet, they take full advantage of the latest Yamaha technology and manufacturing techniques to offer more value than you'll find anywhere else. In short, they offer extraordinary performance and mixing power at remarkable prices. If you need a high-performance analog mixer for music production or sound reinforcement, the Yamaha MG Series is the first — and last — place you should look.

MG10/2 Mixing Console



If you simply need to mix a few sources to stereo, but still want the finest, audio quality available, the MG10/2 is an outstanding choice. Compact and easy to use—the MG10/2 can even be mounted on a mic stand (with optional adapter) for totally flexible positioning and easy access. Ideal for demo and music production in your personal studio, for band rehearsal or small sound reinforcement applications.

10 Input Channels

- Channels 1 - 2 provide a choice of XLR mic or 1/4″ TRS line inputs with insert I/O for adding external compressors or EQ.
- Channels 3 - 6 can be configured as two 1/4″ TRS stereo line inputs or as two mono microphone inputs.
- Channels 7 - 10 are configured as two stereo channels with a choice of 1/4″ TRS or RCA inputs.
- The mic preamps use high-performance head amplifiers and switchable phantom power allowing you to bring out the best of both dynamic or condenser microphones.

Aux Sends and Returns

- Two 1/4″ TRS post-fader aux sends are available for creating a monitor mix or accessing an external effects processor.
- 1/4″ TRS stereo aux returns with level control provide the means bringing an external effects unit back into the mix.

3-band Channel EQ & HPF

- Designed for smooth, "musical" response, a 3-band EQ is provided on all input channels. Mono mic input channels also feature a switchable highpass filter for cutting unwanted low-frequencies.

Mains Section

- 1/4″ TRS balanced main outputs with master level control
- 1/4″ TRS balanced control room and 1/4″ headphone outputs with level control are also provided.
- Left and right RCA record outputs as well as 2 track inputs with level control.
- 12-segment LED output meters

Optional Mic Stand Mount

- Mounts on a mic stand with the optional BMS- 10A Mic Stand Adaptor, so the mixer is always within easy reach.

MG12/4 • MG16/4 • MG16/6FX

Extensive Creative Control in the Studio or on Stage

The mid-range MG models go beyond the basics to give you extensive control for a wide range of applications - with the no-compromise Yamaha sonic quality that makes the MG mixers the finest in their class. Whether music is a hobby or profession, these mixers will deliver total satisfaction. If you don't need effects, or already have an arsenal of outboard favorites, the MG12/4 or MG16/4 may offer all the capacity and capabilities you need. But if the idea of having some of the finest effects available built right into the console appeals to you, then consider the effect enabled MG16/6FX.



Step-up Features from the MG10/2

MG12/4 – 12 Input Channels

- Channels 1 - 4 provide a choice of XLR mic or 1/4″ TRS line inputs with insert I/O for adding external compressors or EQ.
- Channels 5 - 8 can be configured as two 1/4″ TRS stereo line inputs or as two mono microphone inputs.
- Channels 9 - 12 are configured as two stereo channels with a choice of 1/4″ TRS or RCA inputs.

MG16/4 – 16 Input Channels

- Channels 1 - 8 provide a choice of XLR mic or 1/4″ TRS line inputs with insert I/O for adding external compressors or EQ.
- Channels 9 - 12 can be configured as two 1/4″ TRS stereo line inputs or as two mono microphone inputs.
- Channels 13 - 16 are configured as two stereo channels with a choice of 1/4″ TRS or RCA inputs.

Stereo Group Buss

- A stereo group bus with 1/4″ TRS outputs adds a convenient way of channel grouping – ideal for creating a discrete signal path for tracking or creating a monitor mix.

Illuminated Switches

- Each input channel features illuminated switches for ON (Stereo buss assign), PFL (Pre- Fader Listen), and phantom power providing visual confirmation of settings.
- Faders are provided for the input channels, stereo group and main stereo bus.
- Separate level controls are provided for each channel's two Aux Sends. Additionally, Aux one can be switched Pre or Post fader.
- Master Send controls for controlling output levels independently for Aux 1/2
- The main stereo outputs feature both XLR and 1/4″ TRS output connectors.
- Supplied with rackmount adapters, use them on a desktop or mounted in a rack

MG16/6FX Mixer with Effects

16 Input Channels

- Channels 1 - 8 provide a choice of XLR mic or 1/4″ TRS line inputs with insert I/O for adding external compressors or EQ.
- Channels 9 - 12 can be configured as two 1/4″ TRS stereo line inputs or as two mono microphone inputs.
- Channels 13 - 16 are configured as two stereo channels with a choice of 1/4″ TRS or RCA inputs.

Internal Digital Effects

- Built-in internal effects processor with 16 superb reverb and delay programs with a variable parameter control.
- An effect send on each input is provided for controlling the amount of effect for each channel.

7-band Stereo Graphic EQ

- A 7-band stereo graphic equalizer with ±12dB boost/cut per band is available for flexible overall response shaping control.

Six Buses (Stereo and Two Group Pairs)

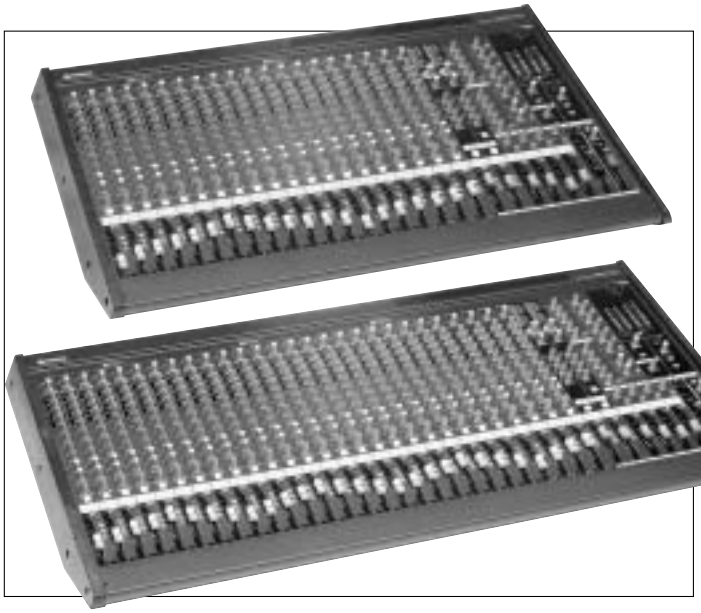
- Two pairs of stereo group buses are provided for convenient channel grouping, in addition to the main stereo bus.

Enhanced Channel EQ

- The mono input channels feature 3-band EQ with a sweepable mid band as well as a High Pass Filter for cutting out unwanted low frequencies on the mic inputs.
- Stereo input channels feature four fixed frequency bands (High, High-Mid, Low-Mid and Low).

YAMAHA

MG24/14FX • MG32/14FX



Serious Capacity for Sound Reinforcement and Installations

If your application is live sound reinforcement you'll want all the channel capacity you can get - just in case. Vocal mics, instrument mics, stereo keyboards, direct-injection feeds, drum mics, and the rest can add up very quickly. With 24 and 32 input channels, respectively, the MG24/14FX and MG32/14FX are ready to handle all but the most ambitious sound-reinforcement setups. And with dual SPX digital effect systems on-board you won't need racks of outboard gear to get the sound you need. There's also a comprehensive range of group and auxiliary busses to make even complex mixes easy.

FEATURES

MG24/14FX – 24 Input Channels

- Channels 1 - 16 provide a choice of XLR mic or 1/4″ TRS line inputs with insert I/O for adding external compressors or EQ.
- Channels 17 - 20 are configured as two 1/4″ TRS stereo line inputs.
- Channels 21 - 24 are configured as two stereo channels with a choice of 1/4″ TRS or RCA inputs.

MG/32/14FX – 32 Input Channels

- Channels 1 - 24 provide a choice of XLR mic or 1/4″ TRS line inputs with insert I/O for adding external compressors or EQ.
- Channels 25 - 28 are configured as two 1/4″ TRS stereo line inputs.
- Channels 29 - 32 are configured as two stereo channels with a choice of 1/4″ TRS or RCA inputs.

Mic Preamps

- All 16 (MG24/14FX) and 24 (MG32/14FX) mic preamps are of exemplary quality featuring low-noise, high-precision head amplifiers with phantom power switchable in 8-channel groups.

14 Buses For Flexible Signal Routing

- In addition to lots of input channels, live sound reinforcement applications usually demand a number of additional mixes - usually in the form of group sub-mixes and aux sends for external signal processing and monitor mixes.
- Four stereo group bus pairs for convenient channel grouping.
- Six auxiliary busses (four configurable for pre- or post-fader operation and two set up as effect sends).
- Two internal effect busses feed the dual high-performance built-in SPX processors.

Dual SPX Digital Effects

- Not one, but two high-performance digital signal processing stages, fed by separate effect buses, are provided using the very latest Yamaha DSP technology. Each stage provides a selection of 16 professional-quality SPX digital effects, including reverb, delay, pitch change, chorus, phasing, vocal doubling, distortion, and more.
- Parameter controls that can be adjusted to tailor the effects to your sonic requirement are also provided and Tap delay allows you to create tempo-synchronized delays.

3-band EQ with Sweepable Mid and HPF

- 3-band EQ with sweepable midband provided on all input channels provide exceptionally smooth, intuitive response, helping you create cleaner, tighter mixes.
- All mono mic input channels also feature a switchable high-pass filter that can be used to cut out unwanted low-frequency noise.

Talkback Input

- Communication capability is important for efficient setup as well as for keeping a show running smoothly. Both the MG24/14FX and MG32/14FX feature a talkback system that allows the FOH engineer to communicate with the monitor engineer, performers, or other staff to keep the team operating at optimum efficiency.

Sweepable LPF for Mono Out

- The XLR balanced mono output, aided by the built-in sweepable low-pass filter, provides an ideal way to tune and drive a subwoofer system.
- An XLR input on the top of the console is provided to accommodate a talkback mic
- Balanced XLR Stereo and Mono Outputs

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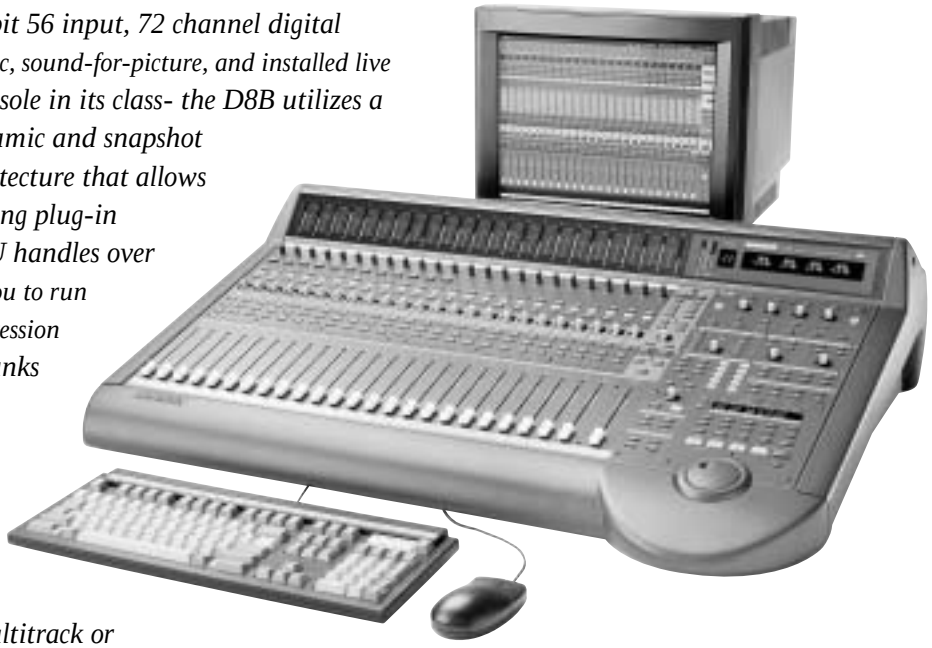
MG SERIES COMPARISON

	MG10/2	MG12/4	MG16/4	MG16/6FX	MG24/14FX	MG32/14FX
Input Channels	10	12	16	16	24	32
XLR Mic Inputs	4	6	10	10	16	16
1/4″ TRS Line Inputs	2	4	8	8	16	16
Stereo Inputs	2	2	2	2	4	4
1/4″ TRS Insert I/O	Ch 1 – 2	Ch 1 – 4	Ch 1 – 8	Ch 1 – 8	Ch 1 – 16	Ch 1 – 16
Phantom Power	+ 48 V	+ 48 V	+ 48 V	+ 48 V	+ 48 V	+ 48 V
Input Gain Control	44 dB variable	44 dB variable	44 dB variable	44 dB variable	44 dB variable	44 dB variable
High Pass Filter	80 Hz 12 dB/Oct	80 Hz 12 dB/Oct	80 Hz 12 dB/Oct	80 Hz 12 dB/Oct	80 Hz 12 dB/Oct	80 Hz 12 dB/Oct
AUX Send/Return						
1/4″ TRS Send	2 (1/Pre, 2/Post)	2 (1/Post-Pre, 2/Post)	2 (1/Post-Pre, 2/Post)	2 (1/Pre, 2/Post-Pre)	6 (1-4 Post-Pre, 5-6 Post)	6 (1-4 Post-Pre, 5-6 Post)
1/4″ TRS Stereo Aux Return	1	1	1	1	2	2
Effect Send	—	—	—	1	2	2
2 Track In	Stereo RCA	Stereo RCA	Stereo RCA	Stereo RCA	Stereo RCA	Stereo RCA
Record Out	Stereo RCA	Stereo RCA	Stereo RCA	Stereo RCA	Stereo RCA	Stereo RCA
Main Stereo Out	1/4″ TRS	1/4″ TRS & XLR	1/4″ TRS & XLR	1/4″ TRS & XLR	1/4″ TRS & XLR	1/4″ TRS & XLR
Main Stereo Inserts	—	—	—	—	1/4″ TRS	1/4″ TRS
Stereo C/R Out	1/4″ TRS	1/4″ TRS	1/4″ TRS	1/4″ TRS	—	—
Stereo Sub Out	—	—	—	—	1/4″ TRS	1/4″ TRS
Mono (Sub) Out	—	—	—	—	XLR	XLR
Mono Out Low Pass Filter	—	—	—	—	80-120 Hz 12dB/Oct	80-120 Hz 12dB/Oct
1/4″ TRS Group Out	—	2	2	4	4	4
1/4″ TRS Group Inserts	—	—	—	—	4	4
Stereo Headphone Output	1/4″ TRS	1/4″ TRS	1/4″ TRS	1/4″ TRS	1/4″ TRS	1/4″ TRS
CH EQ (Mono Channels)						
Boost/Cut per band	±15 dB (Max.)	±15 dB (Max.)	±15 dB (Max.)	±15 dB (Max.)	±15 dB (Max.)	±15 dB (Max.)
High (Shelving)	10 kHz	10 kHz	10 kHz	10 kHz	10 kHz	10 kHz
Mid (Peaking)	2.5 kHz	2.5 kHz	2.5 kHz	0.25 – 5 kHz	0.25 – 5 kHz	0.25 – 5 kHz
Low (Shelving)	100 Hz	100 Hz	100 Hz	100 Hz	100 Hz	100 Hz
CH EQ (Stereo Channels)						
Boost/Cut per band	±15 dB (Max.)	±15 dB (Max.)	±15 dB (Max.)	±15 dB (Max.)	±15 dB (Max.)	±15 dB (Max.)
High (Shelving)	10 kHz	10 kHz	10 kHz	10 kHz	10 kHz	10 kHz
Hi-Mid	—	—	—	3 kHz (Peaking)	3 kHz (Peaking)	3 kHz (Peaking)
Mid (Peaking)	2.5 kHz	2.5 kHz	2.5 kHz	—		
Low-Mid	—	—	—	800 Hz (Peaking)	800 Hz (Peaking)	800 Hz (Peaking)
Low (Shelving)	100 Hz	100 Hz	100 Hz	100 Hz	100 Hz	100 Hz
7-band Graphic Equalizer	—	—	—	(125, 250, 500, 1 k, 2 k, 4 k, 8 kHz)	—	—
Boost/Cut per Band	—	—	—	±12 dB (Max.)		
Internal Digital Effect	—	—	—	1 (16 Programs)	2 (16 Prgs each)	2 (16 Prgs each)
Power Supply	External	External	External	External	Internal	Internal
Dimensions (WDH)	10.2 x 11.3 x 2.5″	12.5 x 16.4 x 4.1″	16.7 x 16.4 x 4.1″	16.7 x 16.4 x 4.1″	33.5 x 21.3 x 5.9″	41.7 x 21.3 x 5.9″
Mounting Capabilities	Mic Stand Mountable	Rack Mountable	Rack Mountable	Rack Mountable	—	—
Option Mic Stand Adapter	BMS-10A	—	—	—	—	—

SONIC SPECIFICATIONS	
Total Harmonic Distortion	Less than 0.1 % (THD+N) 20 Hz – 20 kHz @ +14 dB 600 Ω (ST OUT)
Frequency Response	0 +1, -3 dB 20 Hz – 20 kHz @ +4 dB 600 Ω (ST OUT)
Input Hum & Noise	128 dB 20 Hz – 20 kHz, Rs=150 Ω, Input Gain=Max, Input sensitivity=-60 dB
Crosstalk	-70 dB @ 1 kHz

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D8B (DIGITAL 8•BUS)



The Mackie Digital 8 Bus or D8B is a 24-bit 56 input, 72 channel digital mixing console designed for professional music, sound-for-picture, and installed live applications. The most advanced digital console in its class- the D8B utilizes a separate CPU for controlling limitless dynamic and snapshot automation capabilities and an open architecture that allows software upgrades as well as signal processing plug-in expansion. The pentium-based 32-bit CPU handles over 3 billion instructions per second enabling you to run digital functions such as EQ, gating and compression on forty-eight simultaneous channels. And thanks to the Mackie Real Time OS the system doesn't get bogged down by another operating system such as the Mac OS or Windows. The addition of an SVGA monitor, keyboard and mouse make working the D8B a dream. Whether you are working with analog tape, a digital multitrack or DAW, optional multi channel I/O cards are available to ensure that the D8B will integrate seamlessly into your system. A dedicated DSP processor provides two stereo effects and up to three additional DSP cards can be installed for even greater signal processing power. The D8B's smart ergonomic design, digital patchbay and surround sound mixing capabilities as well as the studio grade 24-bit converters and Apogee UV22 16-bit CD encoding system far exceed the flexibility and sonic integrity of anything else on the market.

FEATURES

- 56 inputs, 72 channels
- Full big-console automation of all parameters including digital effects—recall and edit any mix anytime
- 48 channels of audio with gating, compression and parametric EQ available on all channels simultaneously. 48-channel overview screen
- 12 balanced XLR-mic inputs with switchable phantom power, 1/4″ TRS line inputs with high quality mic preamps and 1/4″ TRS inserts. There is an additional twelve 1/4″ TRS balanced/unbalanced line inputs
- Channels 25 to 48 are accessed from three card slots that accept any combination of optional 8 channel I/O cards
- 12 Aux sends, AES/EBU and S/PDIF digital I/O, 8-channel sub/ surround outputs (25-pin D-sub connector), 2 stereo monitor outputs, balanced XLR and 1/4″ master outs and 3x 1/4″ balanced 2-track returns.
- Built-in meter bridge and Talkback mic
- Speaker select and level controls let you choose between two sets of monitors (eg. near and far-field) as well as two independent, automatable Phones/Cue Mix sections.
- Insert and patch point feature on channels - source either plug-ins, 8 aux sends, 8 buses or 72 channels (pre-DSP only)
- Multifunction display provides tape position in SMPTE /MIDI and lets you mark multiple in and out points.
- Dynamic and snapshot automation
- Event automation track, graphic time line and Event list style automation editing
- MIDI Machine Control
- Assignable MIDI commands on transport and on the D8B master control section
- 999 levels of undo
- Up to 999 samples of time offset (delay) for each channel pre-DSP (dynamics/EQ)
- Inserts can be run across Mains and Busses
- Open architecture compatible with 3rd party plug-ins. Up to 16 simultaneous plug-ins on the first 48 channels, pre or post DSP, pre-fader via up to 4 UFX cards
- Cross patching allows channels from various banks to be substituted with those from other banks
- Enhanced dynamics with input keying and EQ filter, soft-knee compression toggle, linear and exponential option
- Advanced Mix Editor with view sizing arrows, auto-punch, auto-loop SMPTE time code boxes, Loop-in/out, Punch-in/out and Locate markers in the time bars
- Drag and drop file management between File windows and desktop
- Auto Punch can be toggled from the console via MMC.
- 5.1, 6.1 and 7.1 surround-sound capabilities in an adjustable 3-D floating window.

D8B (DIGITAL 8•BUS)

Channel STRIP

- Built-in meter bridge displays the selected channel's input level, including level display for tracking, mixing, effect returns, and sub bussing.
- Trim Level (channels 1-24 only) with quick and easy level setting
- Mic/Line Switch (channels 1-12).
- Record Ready button on each channel strip lets you remotely arm up to 24 tracks of your digital multi-track or hard disk recorder (on recorders that support MMC) without turning away from the console.
- Assign button lets you quickly route channels to buses or any of the 24 tape outputs with the punch of a button or a click of the mouse. It's just like a digital patch bay!
- Write button displays the automation state on the channel strip. For those who want hands-on ability, the D8B's automation can also be run in Auto-Touch mode. As soon as any recordable control is touched and adjusted, the corresponding channel automatically goes into record mode.
- V-Pot gives you hands-on control over all channels' panning as well as all auxiliary send, tape output, and digital trim levels. V-Pots have visual feedback, instant recall and full automation capability.
- Bank LED indicator tells you whether the channels are currently Mic/Line, Tape In, or Effect Return channels. The bank LEDs switch along with the selected fader bank.
- Select function for channel editing, grouping, sending to tape outputs, linking, group or bus assignment, or clipboard functions.
- Direct access Solo and Mute buttons—no scrolling through menus
- 100mm motorized channel faders are silky-smooth when you move them; jitter-free in motorized mode. The motorized action can be globally defeated.

Apogee UV22 Super CD Encoding

Built into the D8B is Apogee's UV22 outboard processor which encodes more bits of detail onto CDs. Audible benefits include more detailed reverb, better tonal balance, wider stereo spread, more precise placement of instruments in the sound stage, and lower noise. Produce fantastic-sounding 2-track masters that will continue to sound fantastic no matter who masters and duplicates them.



Fat Channel – For Hands-on Processing

Perhaps one of the favorite features of the D8B is the legendary Fat Channel. And now it has a new look and many new features that will greatly augment your audio arsenal. Larger views, adjustable dynamics EQ key types, Gate expansion, and external MIDI gate trigger are just a few of the new Fat Channel components that will make your mixing tasks easier and more effective.

- Provides four V-Pots and Previous/Next scroll buttons to quickly find and adjust effects parameters of all twelve auxiliary sends and 4-band EQ, compressor with limiting, and gate with expansion mode parameters for 48 channels
- The Compressor and Gate parameters can be viewed graphically in the expanded dynamics view windows.
- The Dynamic Key provides a selection of four EQ types (Parametric, Shelf, High Pass or Low Pass) that can be applied to the Key input.
- A bright vacuum fluorescent display above the FAT Channel gives you visible read-outs of changing DSP values. (In addition to the optional SVGA monitor.) Although you can tweak the D8B's settings with its analog-style knobs, you may prefer the on-screen control panels for intricate adjustments. To quickly check channels for phase inversion, you can flip the phase of individual channels from the VFD.

Master Section

- Lurking behind the Digital 8•Bus' 100mm motorized faders are four banks of channels, each available at the touch of a dedicated button.
  - —Channels 1-24 are for tracking
  - —Channels 25 to 48 are for tape returns
  - —A third bank for Internal Stereo Effects Returns and four External Stereo Effects Returns
  - —A fourth bank - 8 virtual group master faders, 8 dedicated MIDI controller faders, and 8 bus master faders. When you switch banks, all faders will "snap" to their relative position in the selected fader bank. The SVGA display changes, too, reflecting the selected bank and all controller positions.
- Choose between near-field or main speaker outputs to monitor your mixes. This section can receive source signal from your stereo mastering returns via three sets of analog two-track inputs and/or the stereo AES/EBU or S/PDIF input. Plus, a built-in 'Talkback' mic with 'Dim' button allow communication with studio talent through the Phones/Cue mixes or Studio outputs.
- Each provides an amplified headphone output of either the Control Room mix or either of the 'Cue Mixes' (Aux 9/10, 11/12)
- A unique 'Copy Mix to Cue' function allows you to set up your headphone mixes using the D8B faders, and then copying their relative positions to the Cue Mixes.

External CPU



The D8B includes a separate, external CPU with Pentium processor, complete with 32 MB of RAM, large capacity hard disk, 3.5″ floppy drive (use to load additional software or save sessions), and SVGA video card, keyboard and mouse connections for connecting an optional SVGA monitor (for display of many console functions including fader banks, DSP and effects, surround sound, software library, and file management), keyboard and mouse. On the back is an Ethernet connector. Use it to cascade two D8B's solo control logic, mutes, transports, etc… or connect one or more Digital 8•Bus's to Mac's or PC's via a LAN for transfer of session files and EQ/Effects libraries.

D8B (DIGITAL 8•BUS)

Locator/Transport/Jog

- Once you start using it, you'll discover that you don't have to take your eyes off of the console nearly as much. You can run the transport of any tape or hard-disk based recorder that can slave to MIDI Machine Controlled. You can create song start and stop or loop continuously between two points. And you can dial in any location point in a session right from the console.
- You can also arm up to 24 tape machine tracks directly from the Record Ready button on each D8B channel strip. The time code counter window and location point reader above the transport section reflect the tape position in SMPTE (hrs:mins:secs:frms) or MIDI (bars:beats:ticks), and let you mark multiple location points.

Easy Surround Sound

The future of music is multichannel and the D8B makes it easy. To move a channel from front to back or side to side, just pull up the Surround window and use the mouse to "grab" the sound source and move it around in the Pan window. You can select two different positions for a sound and then "morph" it from one spot to another over a specified length of time. There is also a mind-boggling Surround Screen that shows the front-back/left-right position of up to 72 different channels—all in one view.

- Surround Sound mixing environment with depth of center control, surround LFE gain control for each channel, sound-corrected buss and track assignment.
- Surround front-to-rear pan control via control surfaces or MIDI
- Pan Depth of Center for keeping sound effects and/or music out of the center speaker during LCRS, 5.1/6.1/7.1 sessions.
- The Surround Sound window includes nine faders (eight surround and a master) for adjusting the analog output levels at the Bus 1-8 outputs. This doesn't affect the digital surround output signals at the Alt 1-8 outputs. This allows you to make adjustments in your monitoring levels without affecting the signal being recorded. Once the levels are set, you can lock the monitor level settings by clicking the Menu button and choosing Lock Monitor Levels.

HUI Mode



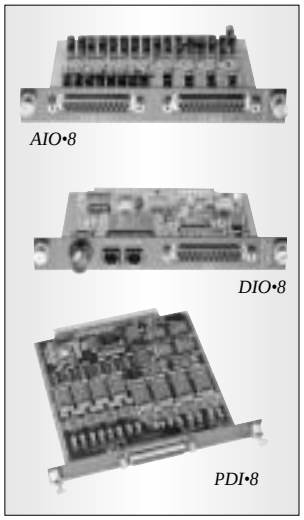
The D8B software includes a HUI mode, which allows the D8B to operate as a HUI (Human User Interface) control surface with DAW software applications that support the Mackie HUI. Turn on HUI mode by clicking Options in the top menu bar and selecting HUI Mode. This creates a fifth fader bank in addition to MIC/LINE, TAPE IN, EFFECTS, and MASTERS, which can be accessed by pressing SHIFT+MASTERS on the control surface or clicking the HUI button above the Master fader on-screen. HUI mode uses the MIDI IN/OUT ports connected to a MIDI interface to control specific functions in the DAW, including fader and transport control.

Open Card Cage Architecture



The D8B can be configured for almost any recording application. It offers analog I/O, digital I/O, word clock I/O and effect processing cards as add-on options. And it doesn't take a degree in electrical engineering to add a card. In fact, you don't even need a screwdriver to install a card.

OPTIONAL 8-CHANNEL I/O CARDS



AIO•8 Analog I/O Card:
8 balanced line level (+4dBu) outputs and returns for connecting the D8B to analog equipment.

DIO•8 Digital I/O Card:
Made by Apogee, this card gives you 8 channels of 24-bit digital tape outputs and digital tape returns in both ADAT lightpipe and Tascam T/DIF formats. The DIO•8's provide direct dubbing capabilities from ADAT optical to T/DIF format, and vice versa.

OPT•8 Low-Cost Digital I/O Card:
The low-cost OPT•8 I/O card provides 8 channels of ADAT lightpipe for under $100!

PDI•8: 24-bit AES/EBU Digital I/O Card;
Ideal for digital audio workstations and supports real time sample rate conversion on input as well as Apogee's UV22 16-bit CD encoding on output.

Apogee Word Clock I/O
If you need to sync your D8B to an external clock reference, the optional Apogee Word Clock I/O card provides a BNC clock input for slaving the D8B. Extremely low-jitter, cost-effective clocking card with vari-speed and pull-up/down capabilities.

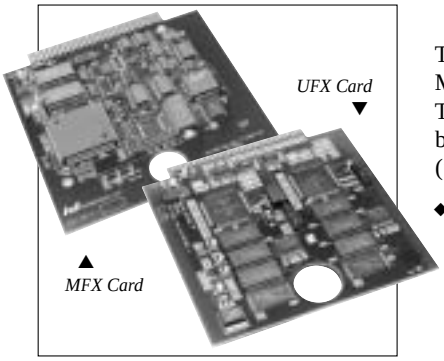
D8B (DIGITAL 8•BUS)

Mackie MFX and UFX Cards and Digital 8•Bus Plug-ins

The D8B's open architecture lets you take advantage of some of the world's most creative software companies. These plug-ins are made possible by the D8B's advanced operating system and optional MFX and UFX cards. The D8B has four card slots in the rear labeled "Digital Effects Cards". The D8B comes with one MFX card comes installed. That leaves three slots to fill up. You can mix and match MFX and UFX cards depending on the plug-ins you want to use. The D8B is the only digital consoles that supports such a wide variety of plug-ins from the world's best processor and effects companies and there's more to come.

MFX Card

The MFX card is the host for Mackie Effects reverbs, choruses, echos etc... The IVL Vocal Studio Demo is included. Adding a second MFX card unlocks the official Vocal Studio plug-in. (One MFX card comes with your D8B.) MFX cards can be mixed with UFX cards, up to a total of four. Removing the MFX card will disable the Mackie Effects, and Vocal Studio.



UFX Card

The UFX Card is the host for Acuma, Antares, Drawmer, and Massenburg plug-ins. Mackie Mono Delay and T.C Electronic's TC FX II (provides reverbs from the renowned M2000) are bundled free with a UFX card. UFX and MFX cards can be mixed (up to a total of four cards).

- UFX cards are 4-in/4-out, which means each card can simultaneously support either: Four mono effects, or two mono and one stereo effect, or two stereo effects
- Up to four UFX cards can be installed in the D8B. This allows you to run up to 16 channels of mono or 8 channels of stereo plug-in effects simultaneously.

Antares
AUTO-TUNE

Pitch Correction for Voice or Solo Instrument

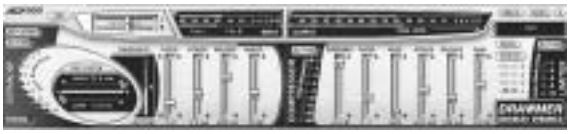


The worldwide standard in professional pitch correction, Auto-Tune corrects intonation problems in vocals or solo instruments, in real time, without distortion or artifacts, while preserving all of the expressive nuance of the original performance. In fact, Auto-Tune offers audio quality so pristine that the only difference between what goes in and what comes out is the intonation.

The precision by which Auto-Tune detects pitch is extraordinary. At a frequency of 400 Hz and a sample rate of 44100, the Auto-Tune DSP algorithm computes the pitch to an accuracy of .0001 samples per cycle, or .0004 Hz. At this resolution, the very question "What is pitch?" becomes relevant. That is, as the pitch of typical performances continuously change, the amount of variation in pitch, even over the time of a few cycles, changes greatly in comparison to the accuracy by which Auto-Tune computes pitch.

Drawmer
ADX100

Frequency-Conscious Gating, Compression, Limiting and Expansion



Acknowledged as master of the analog dynamics processor, Drawmer now offers its expertise to the Mackie Digital 8•Bus.

- Left and Right input levels. -15.75dB to +16dB in 0.25dB increments.
- Input VU meters. -42dB to 0dB.
- Faders for quick parameter control. All thresholds and levels are in linear dBs, but other controls are scaled to a user friendly law which minimizes 'fine tuning' and allows a wide range of operation.
- You can instantly recall two preferred settings.
- Noise Gate is a switch designed for percussive signals, such as drums where it can shape the envelope to create a crisp, well defined signal.
- Gate/Duck switch. When Duck is selected, the operation is reversed, so that when the signal is above threshold, the output is attenuated. Below Threshold, the signal passes unattenuated.

ACUMA LABS PLUG-INS for the D8B

A division of Mackie, Acuma Labs Ltd. develops real-time embedded systems for professional audio applications, specializing in digital signal processors, micro processors, digital audio effects, analog and digital software, real-time operating systems, interfaces, and hardware design. Seven plug-ins altogether, all of their parameters are automated through the D8B's automation engine and can be stored or recalled as a preset.

DSR-1 *Three-band Frequency-Controlled "DeEsser" Dynamic Processor*



The DSR-1 is a highly accurate, frequency controlled, three-band dynamic processor that enables you to quickly isolate and correct unwanted sibilance found in vocal recordings of singers and speakers. The DSR-1 reduces annoying sibilance and popping sounds that are often found in recordings without loosing the crisp top-end clarity. Don't let the simple chicken head controls and the funky retro-look fool you; the DSR-1 is a highly professional tool that is ideal for editing vocals, instruments, and other sources. The DSR-1 is an invaluable D8B plug-in that can save otherwise great recordings by eliminating troublesome "Ess" frequencies.

The DSR-1 employs frequency controlled compression to reduce problematic "Ess" that often occurs in recordings of singers and speakers. Using the Frequency and Width controls to zero in on specific problem areas, an internal keying device known as Listen allows you to solo or (Listen) to the isolated sibilant frequency that you want to compress. The DSR-1 can also help to reduce shrill, high-pitched sounds that may accompany bass guitars, wind instruments, or other acoustic instruments. With a series of useful presets designed for male and female vocals, spoken word and other instruments, the DSR-1 promises to make even the toughest job easy to correct.

FINAL MIX

Real-time Stereo Mastering Processor

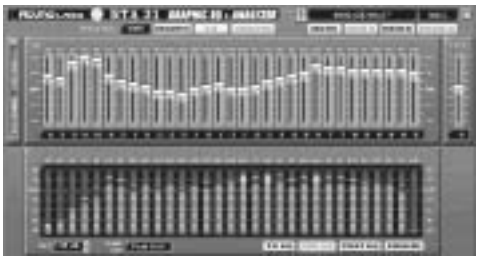


Now master your sessions within the D8B, printing directly to hard disk without having to rely on expensive mastering houses or outboard gear. Final Mix will dramatically elevate the quality of your mixes and help you create your own professional masters.

- 6-band pre-dynamics parametric EQ, 3-band dynamics processor, 6-band post-dynamics parametric EQ
- Graphical (user definable) dynamics band contours
- Adjustable crossover points and slopes for multi-band dynamics
- Node based adjustment of EQ bands and dynamics bands
- Soft Clip feature provides peak overload protection
- Noise gate with threshold adjustment
- DC removal filter automatically removes DC offset noise
- Dynamics bands linking – simplifies use in stereo configurations
- Fully automated; Global enable button
- On/Off for the dynamics section and each EQ
- Separate dynamics and EQ reset buttons
- Plug-in patch load and save
- Memories A and B – compare settings quickly and easily
- Input, output and gain reduction metering

RTA-31

Real Time 31-Band EQ and Spectral Analysis



RTA-31 is both a Graphic EQ and a Spectrum Analyzer that enables you to easily modify the frequency response of any given signal within the D8B or from external live and studio acoustic environments. Quickly view your sound using the on-board spectrum analyzer to zero in on the troublesome highs and lows of even the most difficult rooms. The amazing 'ToEQ' feature lets you immediately identify troublesome frequencies using the Frequency Analyzer while the 31-Band Graph automatically compensates for the highs and lows of your signal.

- Use your mouse as an on-screen paint brush to quickly perform various EQ editing functions.
- Use the 1/3 Octave Spectrum Analyzer (20-20K) in live and studio applications to identify the nature of an acoustic environment.
- Selectable dB reference level (0dB to -60dB) allows you to focus in on the most important range of your signal.
- Factory presets feature useful EQ and Analyzer setups
- Selectable graph types, choose from Bar, Peak Hold, Point, Peak Difference, and Peak Bar for viewing preference.
- Control graph response time to identify quick transients.
- Immediately freeze the analyzer screen to view detailed information.

ACUMA LABS PLUG-INS for the D8B

TIME PAK

Easy-to-Use Time Modulation

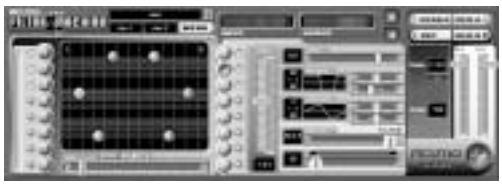


An easy-to-use time modulation offering all of the classic time effects including Chorus, Flange, Auto Pan, Tremolo, combined with radically new and Xtreme settings.

- Use graphic editing sliders to adjust Level, Regen, Pan and Time Modulation, and 4th order High and Low pass filtering.

FILTER MACHINE

Intuitive, Powerful, Sweeping Filter Effects



Acuma's classic, analog styled 4-pole filtering plug-in is an easy and intuitive way to achieve powerful, sweeping filter effects.

- Select from the graphic editing sliders to adjust Pan Modulation, Frequency Modulation, and 4th order High and Low pass filtering.

- Utilizing stereo input and output paths, Time Pak's and Filter Machine's large intuitive display field lets you easily drag and drop any of the 16 nodes (Time Pack) or 16 filters (Filter Machine) to simultaneously adjust Time (Time Pack) or Frequency (Filter Machine) and Output Pan.
- Link left and right nodes (Time Pack) or left and right filters (Filter Machine) together into stereo pairs or fine tune the parameters further using 2 separate LFO blocks that incorporate preset waveforms and envelope followers with adjustable rate, depth, and phase (and volume on the Filter Machine).
- Dial in the beat of the music by selecting the BPM, or tapping the tempo in real time, and quantize the LFO rates to the nearest interval selecting from note, whole note, triplet, etc. All of the filter rates snap to the correct interval with just a click of your mouse.

MASSENBURG DESIGNWORKS



MDW2x2

2x2 High-Resolution Parametric EQ

Headed by George Massenburg, the preeminent designer, MDW (Massenburg DesignWorks) brings you the 2x2 High-Resolution Equalizer which features oversampling and double precision processing for unprecedented clarity and accuracy. Configure the MDW2x2 as two mono 5-band or one stereo 5-band parametric EQ.

DELAY FACTOR

16 Tap Stereo Delay with Modulation



Setting up a stereo, ping-pong, or multi-tap delay has never been easier! Delay Factor has a large, intuitive display field which allows you to drag and drop any of the 16 delay taps, simultaneously editing both the delay time and the output panning.

Fine tuning the delay to match the beat of the song is even easier; just dial in the BPM or tap in the tempo, and select the interval you want to quantize to - note, whole note, triplet, etc. All taps will simultaneously 'snap' to the correct interval.

Adjusting all the other parameters is easy too. All values for a single tap are automatically displayed on easy to adjust sliders when you click on the tap in the display field. Set the level, regeneration, regeneration panning, or the tap's filter. Each tap has a 4th order high and low pass filter.

SATURATED FAT

Mono Distortion & Cabinet Modeling with 7-Band EQ



As its' name implies, Saturated Fat is a mono plug-in that employs a proprietary technology to achieve warm, natural-sounding tube distortion and cabinet modeling. Choose from over 40 preset distortion types to create your "perfect sound" or select from the array of factory presets to find a setting that suits any style of music. And don't stop with guitars — Saturated Fat is great on drum tracks too! Combine the Cabinet Modeler, selecting from a list of small, medium, large, acoustic and bass amps, with the Drive Level and the on-board 7-Band EQ to build custom distortions that can be saved and recalled as User Presets. No more (loud) amplifier setups to contend with, just plug any instrument into the D8B and start playing! Saturated Fat's distinctive distortion types and amplifier models will give you killer sounds instantly.

DMX-R100

Digital Audio Mixer



A professional mixer inheriting the control philosophy of the world-famous Sony OXF-R3 Console, the DMX-R100 is a compact, 48-channel mixer with a comprehensive feature set that includes 25 motorized faders, a sophisticated control panel with touch-screen control, a fully integrated package of automation, a digital routing matrix and machine control. The DMX-R100 delivers excellent sonic performance via its state-of-the-art processing technology, offering 24-bit quality and the ability to operate at both standard and double sample rates (44.1, 48, 88.2 and 96 kHz). With its stunning sound performance, operability and flexibility, the DMX-R100 is ideal for producers, artists and engineers in applications ranging from music studios to post production and audio pre-mastering.

FEATURES

Control Panel Ergonomics

- Although the DMX-R100 is a highly cost effective mixer, it has a fully professional control surface with dedicated control knobs and switches for each individual parameter – emulating the best of traditional console ergonomics. A considerable amount of space is allocated to individual controls for fast, accurate, adjustment and they are laid out in a logical manner that reflects the way that they are used.

High Quality Sound Processing

- The DMX-R100 is designed to deliver the benefits of higher resolution audio signals – greater dynamic range and higher bandwidth. All appropriate inputs and outputs are 24-bit and both standard and double sampling rates are supported.
- The DMX-R100 also processes the full 24 bits of its digital AES/EBU I/Os without any truncation. Internal processing uses precise floating-point calculations to maintain the console's excellent sound quality.

Channels, Returns and Buses

- The DMX-R100 provides 48 input channels and 8 Aux Returns, making a total of 56 channels available for stereo or surround sound mixdown. These channels can be routed to the 8 MTR buses, 8 Aux Send buses, Master L/R Buses or Solo/PFL Buses. As well as EQ and dynamic processing for all 48 input channels, the PGM, Aux Send and MTR outputs also have EQ and dynamics.

Sophisticated Channel Faders

- 25 touch-sensitive motorized faders (24 channel and one PGM fader). The 24 channel faders can be switched in three layers; two layers for the 48 input channels and one layer for master control, including MTR masters, Aux Send masters and Return inputs. The faders are designed to give the operator a very professional 'feel'. Their 10-bit resolution provides precise level adjustment, as well as smooth and accurate replay of automation moves. he The DMX-R100 uses touch-sensitive faders because they allow for excellent operator control of automation and level.

Color SVGA Touchscreen

- The 800 x 600 color SVGA LCD touch screen provides high-quality graphics pages accessed via an intuitive menu structure. These graphics pages include channel processing, input/output routing, automation and mixer setup, and others. For example, the Channel pages gives simultaneous view and control of any one of the 48 channels. Another page, 'AUDIO OVERVIEW' gives a clear view of all 48 channels on two pages so that the operator knows instantly how the mixer is setup. The knobs, buttons, switches and LEDs are displayed on the touch screen with their size and position corresponding to the size and positions of the real controls on the control panel section. This links the visual information provided on the touch screen with the physical controls.
- In addition to numeric indications, large and clear graphic representations of the EQ and Dynamics curves are displayed on the touch screen. Additionally, the graphic/touch screen combination lets you 'zoom in' on specific control panel areas.

B&H PHOTO - VIDEO - PRO AUDIO EQUIPMENT LEASING AVAILABLE

DMX-R100

Channel Strip

The 24 fader strips of the DMX-R100 combine the familiarity of traditional console design with additional features derived from the OXF-R3 console. By default, each fader controls the channel gain, but Select to Fader buttons allow them to control 10 additional level adjustments in the channel path, including the eight Aux Sends, I/P Trim and MTR Sends. Similarly, the Pan rotary control defaults to stereo mix pan but can be switched to provide the same 10 level control functions as Select to Fader. As different signal paths are selected, the faders automatically move to the correct position and a ring of LEDs around each Pan control indicates its current value. This arrangement of two level controls per fader strip provides a simple method of offsetting various channel levels – one of the most common adjustments made during any audio production.

Channel Link Control

Link control of parameters such as Trim, Delay, EQ/Filter, Dynamics, Channel Cut, Channel Fader, MTR Fader, and solo mode. In normal mixing, adjacent odd and even channels can be linked in stereo pairs, and in the case of surround mixing, operators can choose to link groups from the following combinations: 1-6 , 7-12, 13-18, 19-24 channels, L/C/R, LS/RS and so on. In addition, a 'mask' function permits selective exclusion of parameters from the link operation available separately for stereo and surround channels. Surround link groups such as L/C/R, LS/RS and Sub can be selectively linked for surround mixing.

Copy and Zero Reset Function

Copy function allows channel settings of a source channel to be copied to any number of destination channels. The 'mask' function permits selective exclusion of parameters from the copy operation, and moreover, the DMX-R100 supports copying of channel fader mixes to MTR fader/Aux Send faders as well as MTR fader mixes to Aux send faders. This is extremely helpful when creating cue/foldback mixes. Additionally, a zero reset function selectively resets all level controls, faders, and knobs to their default values. EQ curves can also be set flat, and dynamics setting are set to their default values.

Flexible Internal Routing Matrix



The DMX-R100's is its internal audio routing matrix . This provides comprehensive crosspoint switching for virtually every input and output, and avoids the need for a costly external patch bay. The input section of the matrix allows any input signal to be routed to any channel. The same input signal can also be routed to multiple channels. Similarly, the output section allows bus signals to be assigned to any output including those on the four I/O slots and also allows the same signal to be assigned to multiple outputs.

The routing matrix is controlled by two touch-screen pages, one for input signal assignment and one for output signal assignment. Both pages have two levels of access. The first level provides free assignment of inputs and outputs on an individual basis. The second level supports logical groupings of inputs and outputs. For example, the block of signals from one of the optional input boards can be assigned to a group of channels. Similarly, logical groups of mixer buses, Aux Sends for example, can be assigned to a range of mixer outputs. Using this second level to work with these logical groups enables the matrix to be quickly set up. Input and output matrix crosspoint assignments are stored in snapshots. This means that a DMX-R100 can very quickly be reset to different projects by recalling snapshots that include these settings.

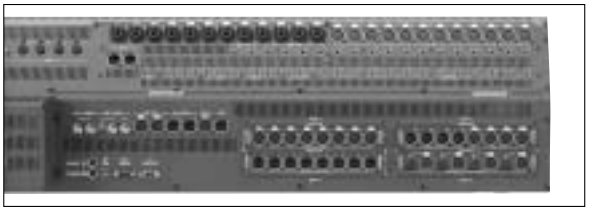
Advanced Snapshot and Dynamic Automation

The DMX-R100 includes 99 snapshots per title, making it possible to memorize and recall the state and values of virtually all mixer functions, including input matrix routing, Delay, Phase, Trim, Input Mode, Filter, EQ, Dynamics, Pan Assign, Cut, Fader and Aux.

The DMX-R100 also offers a ' library' , which is intended for storing repeatedly used EQ and Dynamics settings. Up to 99 can be stored in the library per title for later recall and assignment to any individual channel. In addition, there is comprehensive dynamic automation of Faders, Cuts, Pans, EQ and Filters, Dynamics, and Aux sends. Dynamic automation can be synchronized to both SMPTE and MTC (MIDI time code), and the TC Link function allows snapshots or EQ/Dynamics settings stored in the library to be recalled to programmable time code cues.

A time code offset function is also available to offset the time code relative to that fed from an external source. As dynamic automation is such an important feature, touch-sensitive motorized faders are used. This typical high-end approach greatly simplifies writing and modifying the automation data. The DMX-R100 can be switched between two dynamic automation files (A and B) and automation moves can be written in Absolute or Trim modes. In both modes, the automation moves can be rehearsed before overwriting the previous mix in memory. These high-performance automation features make the DMX-R100 very suitable for complex music and audio post mixing that requires extensive auditioning and scene changes.

Inputs/Outputs



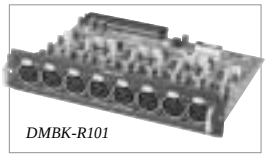
Equipped with 24 analog inputs as well as 8 Aux returns (4 mono analog and 4 mono digital) and 2-track inputs (digital/analog). Analog inputs 1-12 have XLR (for mic inputs and and include 48v phantom power) and 1/4″ TRS connectors (for line level signals). Inputs 13-24 feature neutrik combo connectors (XLR or a 1/4″ TRS. Outputs include stereo program (analog and AES/EBU), Aux Send (8 analog, 2 AES/EBU), control room monitor (6 analog) and studio monitor (2 analog).

DMX-R100

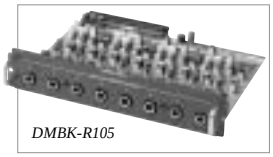
Optional Expansion Boards

For input/output expansion, the DMX-R100 has four option board slots. There are eight optional input/output boards, seven of which allow the user to flexibly expand input/outputs in groups of eight. The eighth optional board, the DMBK-R109 MADI board, allows the addition of 48 inputs/48 outputs via a single board slot. By installing the MADI board in one slot* and the appropriate boards in the three remaining slots, an incredible 72 inputs/outputs become available through the option board slots. As detailed in the chart, the DMX-R100 also interfaces to most popular audio recorders using the appropriate input/output board.

**The DMBK-R109 can only be installed in 'slot-4' of the DMX-R100.*



DMBK-R101



DMBK-R105



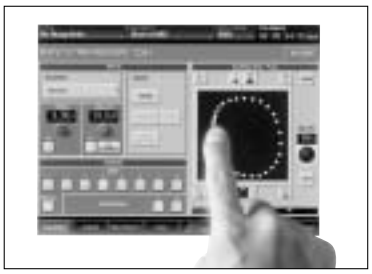
DMBK-R107

Model Number/Description)	Number of Channels	Connector Type
DMBK-R101 (Analog Line-In)	8 inputs	XLR
DMBK-R102 (Analog Line-Out)	8 outputs	XLR
DMBK-R103 (AES/EBU I/O)	8 mono I/O	XLR
DMBK-R104 (Sampling Rate Converter DI)	8 mono ins (x2)	XLR & Optical
DMBK-R105 (Analog Insertion)	8 send/return	1/4″ TRS
DMBK-R106 (ADAT Optical I/O)	8 I/O	Optical x2
DMBK-R107 (TDIF I/O)	8 I/O	D-Sub 25-pin
DMBK-R109 (MADI)	48 I/O	BNC / Optical

Surround Sound Processing

Surround sound is increasingly required for areas such as DVD, film pre-mixing, audio postproduction, etc. As standard, 5.1 surround sound. Furthermore, unique to the DMX-R100, high rate (96 kHz/88.2 kHz) processing is available in 5.1 surround. When high rate processing is selected, the buses and inputs of the DMX-R100 software are reconfigured as follows:

5.1 surround sound at double sample rate	Stereo at double sample rate
24 input channels, 5.1 MTR bus, 2 Aux buses, 4 Aux returns 5.1 CR monitor, 5.1 external source (x 1)	24 input channels, Stereo PGM bus, 4 MTR buses, 2 Aux buses, 4 Aux returns, Stereo CR monitor, Stereo external source, Stereo studio LS



Surround panning is accomplished via the Pan Assign page on the touch screen – by simply tracing the desired panning pattern with a fingertip on the smooth surface of the screen. Alternatively, a PC mouse connected on the back panel of the mixer can be used.

The DMX-R100 has six outputs for the control room monitor, so surround sound monitoring is achieved without sacrificing other outputs and without the need for external monitor switching.

SIU-100 System Interface



The optional SIU-100 is designed for greater system flexibility in applications such as broadcast & post production, and the need for expanded I/O in recording and PA scenarios. Used stand-alone or as an expansion for the DMX-R100, the 5U rackmount SIU-100 accommodates up to eight DMX-R100 style expansion boards, offering a variety of analog and digital inputs and outputs, and high quality microphone preamps.

The SIU-100 interfaces with the DMX-R100 or DAW via the MADI interface, allowing you to position the two units up to 300 meters apart. The Ethernet control capability, gives you the freedom to install the SIU-100 as desired, while also keeping settings at your fingertips.

Additional Features

- No cooling fans are used in the DMX-R100 so it does not generate any acoustic noise.
- Equipped with separate word and video reference input connectors, as well as its own internal reference generator. It also provides multiple machine control with its two 9-pin and MIDI ports. This means that the DMX-R100 can be integrated into any multimedia facility without the need for external synchronization equipment.
- Two DMX-R100's can be cascaded (connected) using the optional DMBK-R109 MADI boards, creating a 96-channel mixer. Since 48 faders become available, this reduces the need for resorting to paging channel faders. Cascade connection at double sample (96kHz/88.2 kHz) rate is also supported.
- The PGM/MTR/AUX buses each have an internal switch to select the bus linkange status (Linked or not) between the master mixer and sub mixer on an individual basis. The AFL bus and solo logic of the master mixer are shared between the master and sub mixer, emulating the operation of a single 96-channel mixer.
- Channel Scribble function allows the entry of names for individual input sources, which then appear on the touch screen next to the channel number. A maximum of seven characters can be entered for each input, and the scribble is displayed in the Channel, EQ and Dynamics GUI screens.

DRE-S777

Sampling Digital Reverberator



The DRE-S777 is a digital reverb device that bridges the gap between "artificial" (synthesizer technology) and "real" to provide a new set of creative tools based on natural-sounding reverberation. This provides audio professionals a totally new approach to sound processing. The DRE-S777 is a technologically advanced digital effects processor that recreates the natural reverberation of concert halls, theaters and sound stages with unparalleled depth, presence and richness. It achieves this breathtaking naturalism by using highly advanced processing that allows audio signals to be combined with sampled data taken from actual acoustic environments. Sony engineers have painstakingly collected sample data from some of the world's most highly regarded concert halls, studios and other acoustic environments, and also from classic analog plate reverb units.The DRE-S777 is supplied with a standard set of sample data, with additional libraries of samples also available.

FEATURES

- Supplied sample CD-ROM contains seven standard presets including data from two different types of concert hall, a recording studio and two churches, plus data from two types of plate reverberator. The DRE-S777 provides for adjustment of parameters such as reverb time, effect/dry balance, EQ and pre-delay. The optional DASK sample discs contain a range of acoustic environments from Europe, America and Japan.
- Nine user caches for quick reverb-program recall.
- The DRE-S777 is capable of a 'self sampling' function, enabling it to capture any acoustic environment. This function requires the optional DASK-704 software module as well as a microphone and speaker system.
- MIDI control
- In addition to stereo reverb modes, it supports multi-channel reverberation. Therefore it is highly suitable for a wide range of multi-channel production tasks, including audio-for-video, and for TV and film post-production applications.
- With the optional DABK-S703 Expansion DSP Board, the DRE-S777 provides a mono input, four-channel output reverb mode. Larger surround sound arrays, for example, 5.1 channel surround, can be provided by using two or more DRE-S777s. The optional DASK software includes reverberation data actually sampled using a five-position mic array, allowing five-channel surround sound effects to be created.
- Reverb mode:
  - Mono in Stereo out (standard)
  - Stereo in Stereo out (with DABK-S703)
  - Mono in 4-channel out (with DABK-S703)
  - Two Mono split in 4-channel out (with DABK-S703)
  - Mono in stereo out at 2Fs (with DABK-S703)
- Direct/Rev: Direct+Reverb/Reverb
- Variable reverb time (0.3-5.5 s max.)
- Sony DSP convolutes an amazing 256,000 sampling points resulting in reverberation that accurately recreates all of the detail of the early reflections and the complexity of the reverberant tails—not only a reverb effect processor, but an ambience emulator!
- Optional DABK-S703 Board upgrades the DRE-S777 for 96kHz sampling, making it compatible with DVD video and audio.
- Optional DABK-S701 (A/D) and the DABK-702 (D/A) converter boards further extend the versatility of the DRE-S777.
- Mixer functions (peak hold, bypass, mixture of dry/wet, muting)
- Four-band parametric EQ
- Factory-presets provided in Memory Stick, as well as set-up data for 92 user-presets
- Operates via jog dial and 4 function keys

Real-World Reverb

To achieve exceptionally natural sounding reverb, it has been necessary to capture the unique "sound" of many different environments; concert halls, churches, studios and so on. The gathering of this data was no mere mechanical process. Rather, it was a series of individual recording projects, each requiring a host of creative decisions familiar to audio professionals. Each project required loudspeakers to radiate the test signals and a microphone array to capture the reverb signature. For each of the locations, multiple samples were recorded using different combinations of loudspeaker and microphone positions. These samples were then combined in the DRE-S777 to provide an extensive range of stereo and surround reverb modes. Sony hardware engineers worked closely with experienced recording engineers to choose microphone types, their directivity and location. Each acoustic space was sampled across a wide range of conditions, and the data supplied on a CD-ROM for use with the DRE-S777.

TASCAM

DM-24

32 Channel 8 Bus Digital Mixer



The Tascam DM-24 is an affordable 32-channel digital mixer that combines 24-bit audio quality, with highly flexible routing capabilities, two powerful onboard effects processors, comprehensive snapshot and dynamic automation as well as dynamics processing and parametric EQ for every channel. The plethora of onboard analog and digital I/O including: 16 mic/line inputs; three 8-channel TDIF I/Os; and 8-channel ADAT optical I/O; along with the wide range of onboard synchronization and machine control capabilities, makes the DM-24 the ideal companion for professional multitrack recording systems such as Tascam's MX-2424 24 track hard disk recorder. The DM-24's 24-bit A-to-D and D-to-A converters and custom 32-bit floating point processor ensures true 24-bit performance throughout the digital signal path.

FEATURES

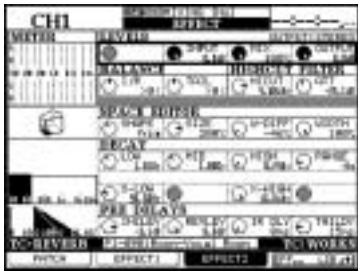
Console Overview

- 32-channel, 8 buss digital mixing console with 6 aux sends at 44.1 or 48 kHz sample rates (16x8x2 at 88.2 or 96 kHz)
- 16 XLR mic, 1/4″ TRS line inputs with inserts and switchable phantom power.
- 24 channels of TDIF I/O, 8 channels of ADAT optical I/O – ideal for any multitrack recording system including Tascam's MX-2424 hard disk recording system.
- Any input can be assigned to any output
- 32-bit internal processing resolution and 24-bit A-to-D, D-to-A converters.
- Each of the 32 input channels is equipped with 4-band fully-parametric EQ, six Aux Sends (assignable pre or post fader) and configurable dynamics section.
- Channels 1-16 also feature a high quality Gate/Expander.

Control Surface

- Sixteen long-throw motorized faders, arranged in three switchable layers, controls up to 32 mono inputs, eight buss sends and six aux sends.
- Each of the 32 inputs can be linked as a stereo pair plus their are eight fader groups and eight mute groups available
- Grouping layers provide further flexibility in grouping arrangements.

Two Fully-Editable Internal Effects Processors



- Processor One gives you the choice of using the TC Works reverb, with over 100 presets, or the Antares Mic and Speaker modeler that emulates the distinctive characteristics of a wide variety of classic and modern microphones, using any standard microphone.
- Processor Two is dedicated to Tascam effects including – Chorus, Delay, Pitch Shifting, Phaser, Flanger, Compressor, Guitar compressor, Exciter, De-esser, Gate and Distortion (with amp simulators)
- Effects can be accessed from the Aux sends 1-6 and returned to any of the 32 channels.
- Effects can also be inserted directly on busses 1-8, the stereo buss or onto one of the 32 channels.
- Effect 1 and 2 can also be run in series (chained) allowing you to run a delay into a reverb, for instance.

Aux Sends

- 6 aux sends, configurable Pre or Post, are provided for each channel and can be freely assigned to the two internal effects processors as well as the 4 assignable sends on balanced TRS jacks, and in pairs to the 2 channel AES/EBU and S/PDIF digital outputs for use with external effects.

Monitoring

- Full control room and studio monitoring facilities are provided, along with an integral talkback microphone and master bargraph meters.
- All popular surround formats are supported including 2+2, 3+2 and 5.1.

Internal Automation – No Computer Required

- The Built-in automation system allows full real time control of almost all mix parameters without the need for connection to an external sequencer.
- Up to eight banks of automation mix files, each containing up to 8000 events, can be stored internally. Up to four banks or 32,000 events can be used per mix.
- Automation is triggered by external MTC, SMPTE time code or from the internal MTC generator.
- You can store and recall 95 user snapshots of the current mixer settings.

DM-24

Input and Master Sections

A. Sixteen XLR mic and 1/4″ TRS line inputs with variable gain and 1/4″ TRS inserts for each channel (phantom power for the mic inputs are switchable in groups of four).

B. Master Section:

– 1/4″ TRS Control Room outputs and Studio Monitor outputs plus two headphone outputs

– 2 Track Input on RCA Connectors

– XLR Main stereo outputs with 1/4″ TRS insert

– Four 1/4″ TRS Sends and 1/4″ TRS Returns can be configured as inserts for tape returns.

Control Surface

C. Each of the sixteen channel strips features a long-throw motorized fader, as well as backlit Record Enable, Channel Select and Mute/Solo buttons and an overload/status LED. The channel strips are arranged in three switchable layers, allowing them to control each of the 32 mono inputs, 8 buss sends and 6 aux sends.

D. Control Room Section:

– Solo and headphone volume controls.

– A volume control, dim switch and mute switch are provided for the control room outputs.

– Three programmable source select switches allow you to designate the desired monitor source.

– 2 x 12-segment LED meters

– The Jog/Data wheel can be used for changing parameter values as well as transport control.

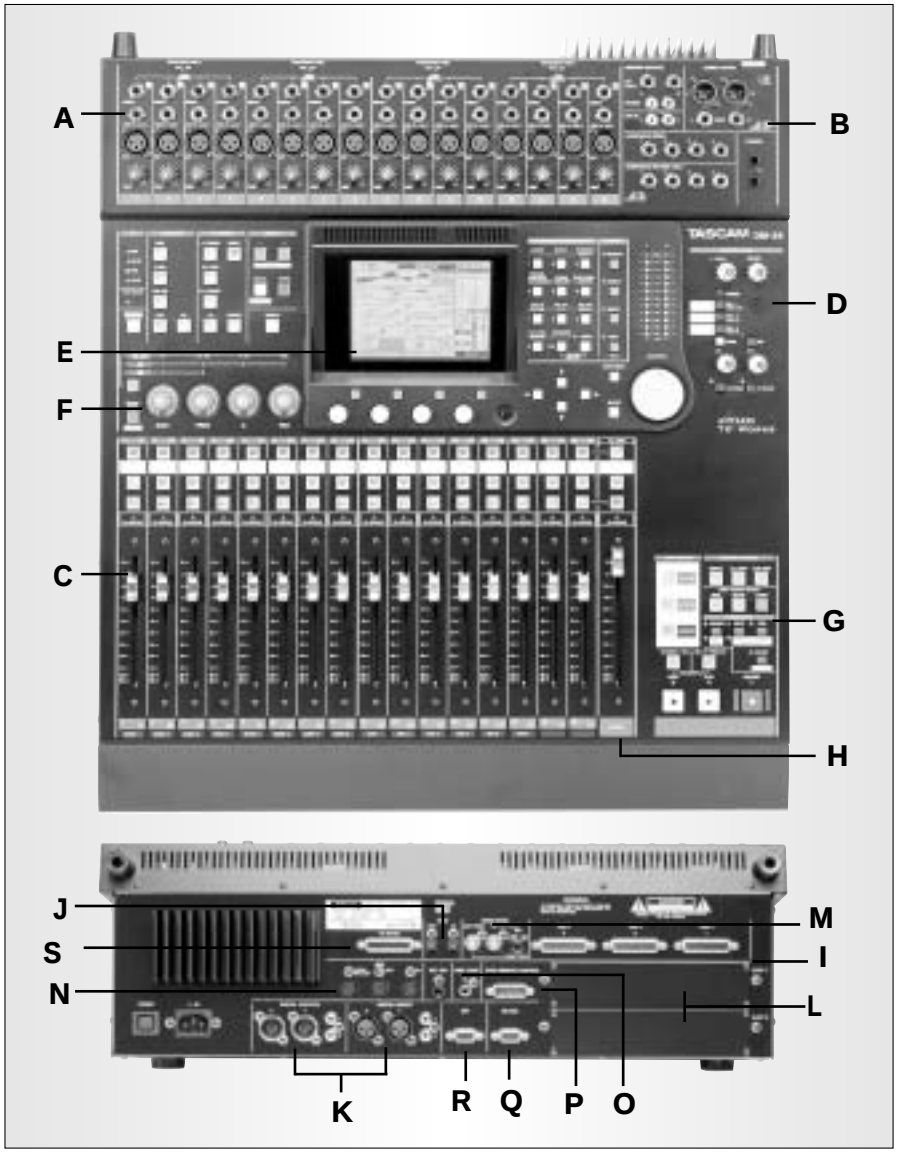
– Built-in talkback mic with level control.

E. The backlit 320 x 240 LCD displays routing assignments, EQ curves, effects parameters, automation data, external control status, synchronization and more. 4 multifunction rotary "POD" controls, located directly underneath the display, provide control over effects, EQ, dynamics, and surround panning parameters and more. The POD controls also transmit MIDI controller data.

F. Four Ring Encoders give instant visual feedback of key EQ, pan and aux send settings.

G. The Transport Section can control a wide variety of external devices using the onboard RS-422, DTRS remote, MMC and GPI ports as well as ADAT Sync by using the optional IF-AD-DM card. Functions include: Play, Record, Stop, FF, REW transport controls; numeric keypad for manual and direct locate; auto punch keys; repeat; memo; edit keys and track arming keys.

H. The dedicated motorized long-throw Master fader is provided for controlling the stereo out buss.



I. Three TDIF I/O interfaces provide 24 channels of buss outs and returns for use with TDIF compatible DAWs and DTRS machines.

J. Eight channels of ADAT Optical I/O provide lightpipe compatible buss outs and returns.

K. Two stereo AES/EBU and two stereo S/PDIF interfaces.

L. Two option slots accommodate additional 8-channel analog and digital interface modules or a cascade module for connecting two DM-24s.

M. Word Sync In and Out/Thru ports (BNC)

N. MIDI In, Out and Thru/MTC Out

O. SMPTE Time Code input (RCA)

P. 15-pin D-sub DTRS compatible Remote output

Q. RS-422 (for Sony 9-pin) 9-pin female D-sub connector

R. GPI 9-pin female D-sub connector allows remote machine start

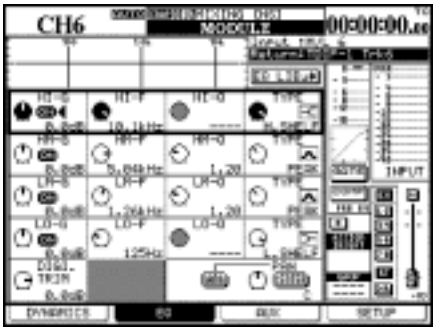
S. 25-pin D-sub connector for use with the optional MU-24/DM Meter Bridge.

TASCAM

DM-24

EQ AND DYNAMICS ON ALL 32 INPUT CHANNELS

4 Band Parametric EQ



High Filter On/Off switchable
Type: Hi-shelving, Peak, LPF
Frequency: 31Hz to 19kHz
Q: 0.27 to 8.65
Boost/Cut: ±18dB, 0.5dB resolution

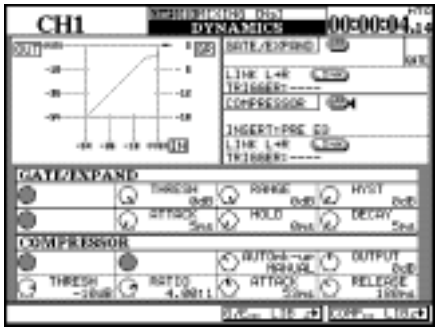
Hi Mid Filter On/Off switchable
Type: Peak, Notch
Frequency: 31Hz to 19kHz
Q: 0.27 to 8.65
Boost/Cut: ±18dB, 0.5dB resolution

Lo Mid Filter On/Off switchable
Type: Peak, Notch
Frequency: 31Hz to 19kHz
Q: 0.27 to 8.65
Boost/Cut: ±18dB, 0.5dB resolution

Low Filter On/Off switchable
Type: Low-shelving, Peak, HPF
Frequency: 31Hz to 19kHz
Q: 0.27 to 8.65
Boost/Cut: ±18dB, 0.5dB resolution

All filters are fitted with "gain flat" switches

Dynamics



Gate:

Threshold	-80dB to 0dB in 1dB steps
Range	60dB to 0dB in 1dB steps
Hysteresis	0dB to 24dB in 1dB steps
Attack time	0ms to 125ms in 1ms steps
Hold time	0ms to 990ms in 100 steps
Decay time	50ms to 5.0s

Expander:

Threshold	–48dB to 0dB, in 1dB steps.
Ratio Values	1:1, 1:2, 1:4, 1:8, 1:16, 1:32, 1:64
Attack	0ms to 125ms in 1ms steps
Release	5ms to 5.00 seconds

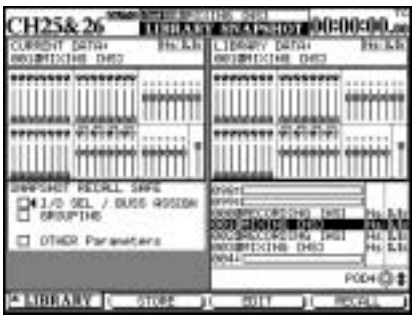
Compression:

Threshold	-48dB to 0dB in 1dB steps.
Ratio	1:1 to ∞:1
Attack time	0ms to 125ms in 1ms steps.
Release time	5ms to 5.0s in 100 steps.
Auto (gain) make-up	switchable
Output gain	-20dB to +20dB in 1 dB steps

Libraries

- The DM-24 allows allows you to store customized settings in libraries. Most of the libraries included a number of locations reserved for read-only presets containing useful points of reference.
- The Library undo/redo function allows a recently recalled library setting to be compared with a previously-loaded setting.

Library Presets	Total	Read-only
Snapshots	100	4
EQ	128	20
Effect 1 (P1)	128	128
Effect 2 (P2)	128	128
User effect library	128	0
Compressor	128	18
Gate/Expander	128	2
Automation	7	–



The snapshot library screen includes a picture of the faders, together with pan/balance settings and mute settings.

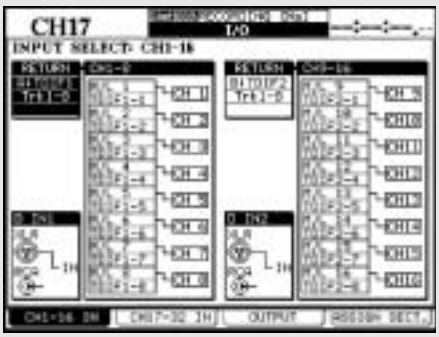
Signal Flow Examples

Tracking:

- Channels 1-16 can be accessed from the 16 mic/line inputs and then assigned to 16 direct outs while channels 17-32 can provide 16 tape returns for monitoring.
- Aux sends 1-6 can be used to create 6 mono or 3 stereo cue mixes.

Overdubbing:

- TDIF returns 1-16 can be assigned to channels 17-32 while channels 1-16 are used for four mic/line inputs, two internal effects returns, and the returns from the third return group. This gives allows you to record a vocal track dry, yet monitor the vocal with reverb in the cue mix.



48 Channel Mixdown:

- The first 24 of the 32 input channels can be accessed by the three TDIF returns.
- The additional 8 input channels are accessed by the two internal effects, a digital stereo pair, and an external send/return.
- The mic/line inputs can also be used, by their direct assignment to Aux 1-2. This means that up to 16 channels of live MIDI sequenced instruments can be added to the 32 input channels.

DM-24

Synchronization and Machine Control

- The DM-24 can be easily configured as a remote controller for a wide variety of devices using the DTRS remote, P2 and MMC protocols. Multiple devices can even be controlled simultaneously.
- MIDI In, Out and Thru/MTC Out allows you to Send MMC, Lock to incoming MTC, Update firmware via Standard MIDI File, offload and upload library and automation data as well as Send/Receive MIDI program changes and MIDI controller data.
- Built-in Mackie HUI emulation provides MIDI-based control surface support for a variety of DAW applications including ProTools, Logic, DP, Cubase and Nuendo.

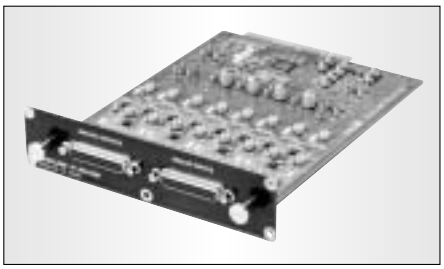
Cascading Two DM-24s

- Two DM-24s can be cascaded together to act as one large console. This requires the IF-CS/DM cascade option card in each mixer.
- Cascading allows the two mixers to share busses 1-8, aux sends 1-6, the solo buss and the stereo buss. Connections to the control room and stereo outputs would be made on the cascade master.
- An AES/EBU option card added to the cascade master can be used for sending the 5.1 mix digitally to your stem recorder while an analog option card could be added to the cascade slave to send the 5.1 mix to the surround monitoring system.

Cascading two DM-24s provides:

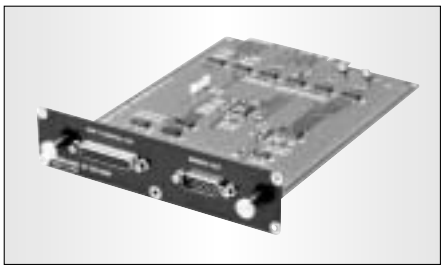
- 64 channels with 32 mic pres
- 48 channels of TDIF I/O
- 16 channels of ADAT lightpipe I/O
- 4 AES/EBU 2 channel digital I/O
- 4 S/PDIF 2 channel digital I/O
- 8 Assignable sends and returns
- 4 Internal effects processors (2 in each)
- 33 Touch sensitive, motorized faders
- The ability to run a 24 track, 24-bit, 96kHz, 5.1 mixing environment.

IF-AN/DM



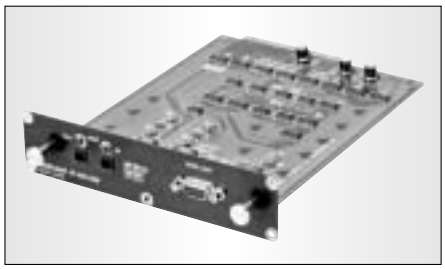
Eight channels of analog I/O on D-25 connectors. Installing two of these cards will give you a total of 32 simultaneous analog inputs.

IF-TD/DM



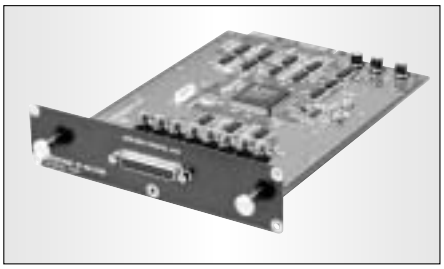
Eight channel TDIF card. The IF-TD/DM is not available for input. It is designed to be used to send the 6 buss outputs to a TDIF stem recorder like a DTRS machine.

IF-AD/DM



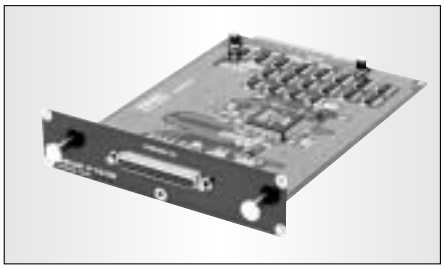
Eight channels of ADAT optical I/O with ADAT Sync. Adding two of these cards will allow you to mix 24 tracks of ADAT. The IF-AD/DM does NOT support 96kHz.

IF-AN/DM



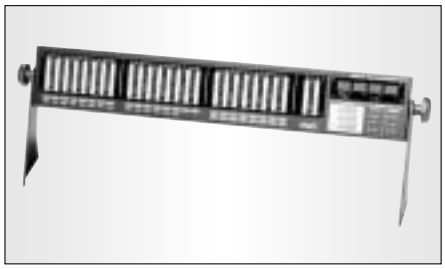
Eight channels of AES/EBU I/O on D-25 connectors. Adding two IF-AE/DM AES/EBU cards provides a total of 20 channels of AES/EBU I/O (four channels come standard on the DM-24.) The IFAE/ DM AES/EBU card supports 4 channels of DUAL LINE or HI SPEED I/O

IF-CS/DM



For cascading two DM-24 digital mixers. To cascade two DM-24s, you must put one cascade card in “SLOT 1” of each DM-24. A cascade cable is included.

MU-24/DM

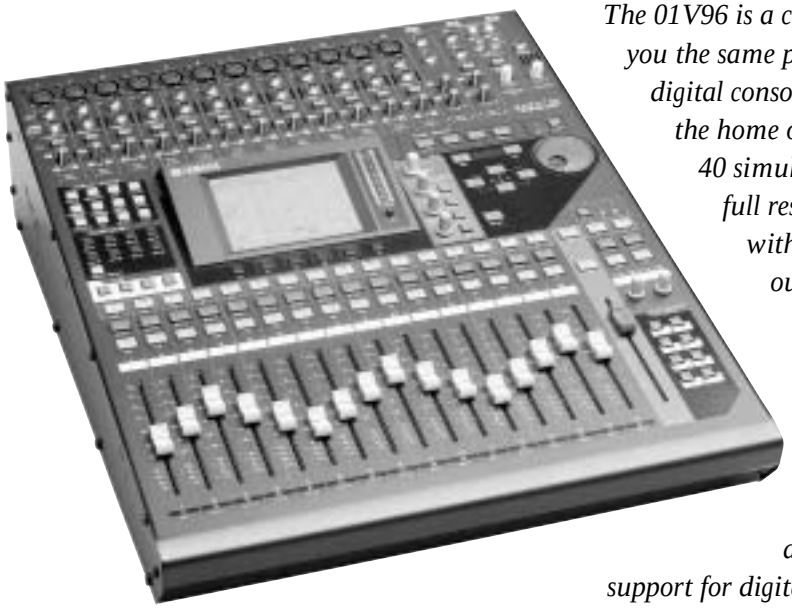


The optional MU-24/DM Meter Bridge provides channel and master metering facilities through LED bargraph displays which are switchable in “layers”.

YAMAHA

01V96

32-Channel 24-bit / 96kHz Digital Mixing Console



The 01V96 is a cutting edge digital mixing console designed to bring you the same performance and reliability of Yamaha's most advanced digital consoles, in a smaller, more affordable format that's ideal for the home or smaller professional production studio. It featuring 40 simultaneous mixing channels (when fully expanded) with full resolution 24-bit/96 kHz audio, a range of stereo effects with 32-bit internal processing and full automation. Right out of the box, the 01V96 provides 24 analog inputs and an 8 channel ADAT optical I/O interface. For further expansion, a single I/O slot will accept Mini-YGDAI digital and analog I/O cards, including third party cards, providing up to 16 inputs and outputs at 48 kHz (8 at 96kHz). The comprehensive control surface allows analog-style hands-on operation along with the added flexibility of providing extensive support for digital audio workstations including ProTools, Nuendo and other popular computer-based recording software.

FEATURES

- 36 simultaneous inputs, 8 busses and 8 auxes, at 24 bit, 96kHZ operation
- Supports 44.1 kHz, 48 kHz, 88.2 kHz, or 96 kHz sample rates
- 24-bit A-to-D and D-to-A converters

24 Analog and Digital Inputs

- 16 analog channel inputs:
  - Channels 1 - 12 will accept balanced XLR microphone signals via high-performance head amplifiers, as well as (bal/unbal) line-level signals via 1/4″ TRS connectors. 48-volt phantom power (switchable in 4-channel groups), trim controls and pad switches are also provided as well as unbalanced 1/4″ TRS inserts that will allow you to insert external processing gear into these channels (pre-A-to-D).
  - Channels 13 - 16 feature 1/4″ TRS (bal/unbal) line inputs with trim controls, that can be configured as four individual inputs or two stereo pairs.
- Eight digital channel inputs (and outputs) are provided via a built-in ADAT optical interface.

Expandable To 40 Inputs

- When you need more I/O or need to configure a system based around a specific protocol, additional I/O can be added via a range of user-installable 8 and 16 channel Mini-YGDAI expansion cards that can simply be plugged into the expansion slot to provide in a variety of formats including ADAT, AES/EBU, TDIF or analog.

All Input Channels Feature

- Independent compressor/limiter and gating/ducking processors on each channel provide complete dynamics control.
- Each channel also features 4-band fully parametric EQ. Each band is fully sweepable from 20 Hz to 20 kHz, with bandwidth variable from 0.1 to 10 and a ±18dB boost/cut range.
- The channel delays have a maximum delay of 452 milliseconds (96 kHz mode). Even the stereo bus, eight mix buses, and eight aux buses have individual compression and EQ!

17 Physical 100mm Motorized Faders

- There are sixteen channel strips each of which features a precision 100mm motorized fader, channel ON/OFF (mute) key, a SOLO key, and a SEL key. The SEL key allows you to assign the channel to the console's Selected Channel section where you have access to detailed controls for panning, EQ and dynamics.
- The 16 channel strips can be instantly switched via four Layer-switching keys to control:
  - Input channels 1 - 16
  - Input channels 17 - 32
  - The Master Layer which controls auxiliary sends 1 - 8 and buses 1 - 8
  - The Remote Layer can be used to control your DAW software including ProTools, Nuendo and more
- Fader Mode keys allow the motor-driven faders to be instantly switched and updated between fader and auxiliary level control.
- The seventeenth motorized fader is reserved for the Master Stereo bus which also features On/Off and Sel keys.

01V96

20-bus Configuration

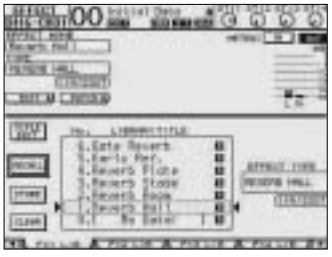
- A wide range of freely configurable signal-routing options are provided to adapt to a variety of tracking, mixing and live sound applications:
  - Main stereo program bus
  - Eight individual mixing buses
  - Two solo buses
  - Eight auxiliary buses

Digital Patching System

- A fast, flexible and easy-to-use digital patching system allows you to assign all of the available inputs, outputs, effects, and channel inserts to any of the console's channels or outputs.
- A direct out function allows the signal from any input channel to be routed directly to any digital or analog output.
- The eight auxiliary buses can also be patched to anywhere in the system.
- The Patch Library allows you to store and instantly recall entire patch setups – ideal for configuring templates for different applications i.e. tracking, mixing, mastering etc.

24-bit Fully Editable Internal Effects

- You can use two stereo effects simultaneously at the 88.2 and 96 kHz sample rates, and up to four stereo effects at 44.1 and 48 kHz.
- Effects can be assigned to an auxiliary bus for send and return operation, or to an insert into any input channel as required.



- An effect libraries with 44 factory and 76 user presets is provided covering a whole host of effects types, including reverbs, delays, modulation-based effects, combination effects, and multichannel effects designed especially for use with surround sound.

Snapshot and Dynamic Automation

- The 99 Scene Memory allows complete console setups to be memorized and instantly recalled via the dedicated SCENE MEMORY controls or via MIDI program changes.
- An external sequencer can be used to record real-time movement of the motorized faders, EQ settings and other parameters, allowing complete mix automation.

Integrated DAW Control

- Extensive support, including full control of mixing and processing parameters, as well as transport/track-arming control and access to editing functions, is provided for Digidesign's Pro Tools system as well as Steinberg's Nuendo DAW. There's also a "General DAW" mode that provides compatibility with other workstations.

Cascade Link

When you really need high capacity – particularly for sound reinforcement applications – the 01V96 offers "01V96 Cascade Link" capability that allows two 01V96 units to be cascaded to create up to an 80-channel mixing system at an unbelievably affordable price!

Rear Panel



- XLR-balanced analog main stereo outputs and 1/4″ TRS (bal/unbal) monitor outputs
- Four 1/4″ TRS (bal/unbal) "Omni" outputs.
- ADAT optical In and Out connectors
- 24-bit coaxial S/PDIF digital 2-track inputs and outputs with on-board sample rate conversion that allows CD players and other digital sources connected to the digital input to be monitored or routed to an input channel without having to be synchronized to the system clock.

Sync and Control Options

- Word clock inputs and outputs (BNC)
- MIDI In, Out and Thru connectors and a USB "To Host" connector are provided allowing computer control via the supplied Studio Monitor software as well as automation via MIDI cc data.
- An expansion slot which will accept a wide range of Yamaha mini-YGDAI expansion cards that can add up to 16 additional channels in a variety of formats.

Expandable Data Libraries

- A number of Data Libraries, each containing an extensive selection of presets, are provided for effects, compression, gating, EQ, I/O patching, and channel setups.
- Presets can be selected and used unmodified, or edited to suit specific requirements. Modified setups can be renamed and saved to the libraries and instantly recalled when needed.

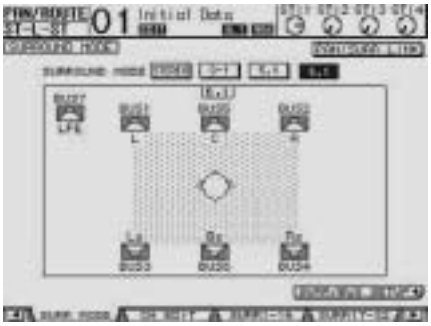
Libraries Presets	Factory	User
Effect libraries	44	76
Compressor libraries	36	92
Gate libraries	4	124
EQ libraries	40	160
Channel libraries	2	127
Input patch libraries	1	32
Output patch libraries	1	32

01V96

Navigation

- The high-resolution 320 x 240 dot LCD panel provides easy visual access to all of the consoles functions and parameters.
- Many of the parameters, such as EQ curves and compression parameters, are displayed graphically so you can see what's happening at a glance.
- Display Access keys determine which type of data will be shown on the LCD panel – a total of 12 selectable categories. This approach minimizes the need to scroll through on-screen lists when you need access to a particular type of data.
- The Selected Channel controls include the hands-on panning and EQ controls for the currently selected channel. Analog-style buttons and knobs provide direct and easy access to the parameters.
- Eight User Defined Keys can be assigned to control any functions you choose. For example, you can use them to recall input patch setups, to arm MTR tracks for recording, or to handle locator functions.
- When the REMOTE layer is selected, the USER DEFINED KEYS are automatically assigned to Pro Tools control functions by default.
- Large cursor, INC/DEC, and enter keys are complemented by a data entry wheel that lets you spin in values quickly and easily.
- The data entry wheel can also be used as a shuttle/ scrub dial for a recorder or DAW.

Surround Panning



Surround is becoming an important part of modern sound production. The 01V96 features 6.1, 5.1 and 3-1 surround panning modes so you can create surround mixes without having to sacrifice features or performance in other areas.

STUDIO MANAGER SOFTWARE (MAC/PC)

SUPPLIED WITH BOTH THE 01V96 AND 02R96

Studio Manager gives you complete access to all parameters for either on-line or off-line control, and the program's visual interface makes it easy to relate on-screen controls to the corresponding console functions. It can also be used to manage an extensive archive of mix data. The Studio Manager includes the following main display pages:

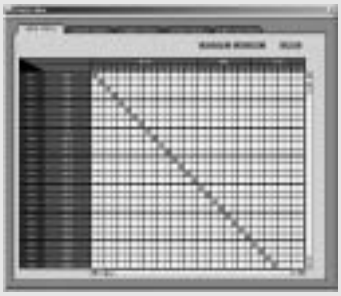
Console Window

Almost a complete virtual mixer, this display shows the primary channel parameters "in-line" as they would appear on an analog console.



Patch Editor Window

When you add several Mini-YGDAI expansion cards to create your system's I/O configurations, input and output patching can become an issue. The Patch Editor makes patch assignments easy with a matrix type visual interface.





Selected Channel Window

Similar to the SELECTED CHANNEL section on the console, this display includes all parameters for the currently selected channel: level, buss assignments, sends, gate, compressor, delay, etc.



Surround Editor Window

When the console is being used in surround mode the Surround Editor display can facilitate surround positioning of individual tracks.

B&H PHOTO - VIDEO - PRO AUDIO

EQUIPMENT LEASING AVAILABLE

01V96 • 02R96 EXPANSION BOARDS

Connection with 96-kHz Recorders and Workstations

Although the 01V96 handles 96 kHz audio as standard, most of the currently available digital recorders and workstations can handle 96 kHz audio only in double channel mode (using two tracks to make one). In this configuration the 01V96 uses one channel per (96-kHz) track, but twice the number of I/O connections must be used. MY8-AT/TD/AE cards work in double channel mode to handle 16 channels of 44.1/48-kHz audio or up to 8 channels of 96 kHz audio in double channel mode. With the latest equipment that handles 96-kHz audio as standard (in double speed mode like the 01V96) you can make standard connections using the MY8-AE96 card. The MY8-AE96 card can work either in double speed or double channel mode.

MY-16TD



16 channel TDIF format I/O

MY-16AT



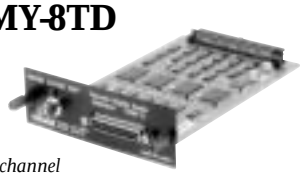
16 channel ADAT format I/O

MY-16AE



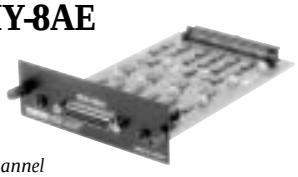
16 channel AES/EBU format I/O

MY-8TD



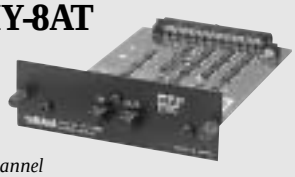
8 channel ADAT format I/O

MY-8AE



8 channel AES/EBU format I/O

MY-8AT



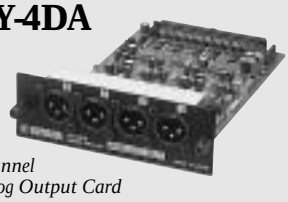
8 channel ADAT format I/O

MY-8mLAN

mLAN interface card



MY-4DA



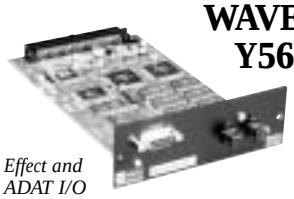
4 channel Analog Output Card

MY-8AE



4 channel Analog Input Card

WAVES Y56K



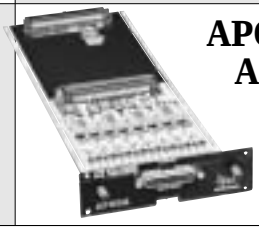
Effect and ADAT I/O

APOGEE AP8AD



A/D Converter

APOGEE AP8DA



D/A Converter

Mfr.	Model	Function	Input	Output*	Format	Resolution	Frequency	Notes
Yamaha	MY16-AT	Digital I/O	16	16	ADAT	24 bit	44.1/48 kHz	2
Yamaha	MY16-AE	Digital I/O	16	16	AES/EBU	24 bit	44.1/48 kHz	2
Yamaha	MY16-TD	Digital I/O	16	16	Tascam	24 bit	44.1/48 kHz	2
Yamaha	MY8-AT	Digital I/O	8	8	ADAT	24 bit	44.1/48 kHz	2
Yamaha	MY8-TD	Digital I/O	8	8	Tascam	24-bit	44.1/48 kHz	2
Yamaha	MY8-AE	Digital I/O	8	8	AES/EBU	24 bit	44.1/48 kHz	2
Yamaha	MY4-AD	Analog In	4	-	Analog	24 bit	44.1/48 kHz	
Yamaha	MY4-DA	Analog Out	-	4	Analog	20 bit	44.1/48 kHz	
Yamaha	MY8-AD24	Analog In	8	-	Analog	24 bit	44.1/48 kHz	
Yamaha	MY8-AD96	Analog In	8	-	Analog	24 bit	44.1/48/88.2/96 kHz	
Yamaha	MY8-DA96	Analog Out	-	8	Analog	24 bit	44.1/48/88.2/96 kHz	
Yamaha	MY8-AE96S	Digital I/O	8	8	AES/EBU	24 bit	44.1/48/88.2/96 kHz	3
Yamaha	MY8-AE96	Digital I/O	8	8	AES/EBU	24 bit	44.1/48/88.2/96 kHz	
Yamaha	MY8-mLAN	mLAN Interface	8	8	IEEE1394	24 bit	44.1/48 kHz	Max. 5 nodes
Waves	Y56K	Effect & I/O	8	8	ADAT	24 bit	44.1/48 kHz	
Apogee	AP8AD	Analog In	8	-	Analog	24 bit	44.1/48/88.2/96 kHz	4
Apogee	AP8DA	Analog Out	-	8	Analog	24 bit	44.1/48/88.2/96 kHz	4

***** *Selectable from Stereo/Bus/Aux/Direct Out/Insert Out/Cascade Out (STEREO, BUS1-8, AUX1-8, SOLO). Details depend on each interface card.*

2. *Can Handle 24 bit/96 kHz using double channel mode* ***3.*** *Sampling Rate Converter for input* ***4.*** *4ch @fs = 88.2, 96 kHz*

YAMAHA

02R96

24-bit/96kHz Digital Mixing Console



The 02R96 Digital Mixing Console is the long-awaited, and fully revised, successor to the revolutionary 02R — the industries first truly professional, affordable digital console. In fact, the 02R96 packs more than five times the processing power into the same footprint size as the original 02R. This power allows the 02R96 to provide 56 channels of full resolution 24-bit/96 kHz audio, full automation of virtually all console parameters with quarter-frame accuracy, surround monitoring, as well as comprehensive range of 96kHz compatible stereo effects with 32-bit internal processing. Out of the gate, the 02R96 provides 24 analog inputs including 16 mic preamps using the same high quality head amplifiers as Yamaha's top-of the-line DM2000 digital mixing console. Thirty Two channels of I/O can be added via four I/O slots that accept a new range of 24-bit/96kHz-capable Mini-YGDAI digital and analog I/O cards. This allows you to configure the 02R96's I/O to suit your specific production environment.

FEATURES

Sonic Spec

- Transparent, full resolution 24-bit/96 kHz audio, with true 24-bit, 128-times oversampling 96kHz A-to-D and D-to-A converters, 32-bit internal processing and 58-bit accumulators.
- 20 Hz–40 kHz (0.5, –1.5 dB) frequency response at 96 kHz sampling rate.
- 105 dB typical dynamic range (A-to-D Input to Stereo Out).

I/O Architecture

- Up to 56 simultaneous input channels each with direct outputs
- 24 balanced analog inputs including sixteen XLR mic, 1/4″ TRS line inputs featuring high-performance head amplifiers derived from Yamaha's top-of the line DM2000, plus eight 1/4″ TRS line inputs.
- An additional 32 inputs (and outputs) can be derived from four rear panel 24-bit 96kHz compatible mini-YGDAI expansion card slots. These slots accept a wide variety of analog and digital (ADAT optical, TDIF and AES/EBU) plug-in cards that will allow you to create an input/output configuration to perfectly suit your system's needs.
- Each input channel has access to 8 aux sends; independent compression and gating / ducking processors; 4-band parameter EQ and up to 453ms. of channel delay.
- 8 freely assignable 1/4″ TRS Omni outputs.
- Dedicated studio monitor and control room monitor outputs.
- AES/EBU digital I/O and two Coaxial 2-track digital I/Os – A sampling rate converter is provided on the digital ins allowing easy integration of 44.1/48 kHz digital audio equipment such as a CD player.
- XLR and 1/4″ TRS stereo outputs plus two analog 2-track inputs.

Superlative Analog Head Amplifiers & Converters

The head amplifiers are derived from the acclaimed DM2000 - some of the finest analog mic preamps available in any console, anywhere. The on-board 24-bit/96-kHz converters ensure that you get an excellent digital representation of the warm, transparent output from these remarkable mic preamps.

Digital Patching

- All available inputs, outputs, effects, and channel inserts can be assigned to any of the console's channels or outputs via the easy-to-use digital patching system.
- Each of the 56 input channels can be routed to their direct outputs; subgrouped to the eight bus outputs; and to the main stereo outputs – simultaneously if need be.
- The eight aux busses can be patched to anywhere in the system, and patch setups may be stored in the patch library for instant recall at any time.
- Input and output ports can be named for easy identification and patches can be stored in the I/O Patch libraries.

EQ

- 4-band parametric EQ is provided on all Input and Output Channels with a choice of a "type I" EQ algorithm or a newly developed "type II" EQ algorithm.

Dynamics

- Independent compression and gating / ducking processors are provided for all 56 Input Channels. Additionally, compression is provided for each output.

02R96

– Control Surface–

A. Variable gain, signal present and peak LEDs are provided for all of the analog inputs plus independent switchable 26dB pad and phantom power on the first 16 analog inputs as well as on/off switches for the inserts I/O.

B. The 320 x 240 fluorescent backlit LCD display is enhanced by twelve Display Access keys that allow you to instantly switch the display to specific editing, utility and metering functions such as Automix, Channel Groups and Pairs, I/O Patching, MIDI and Remote setups and more.

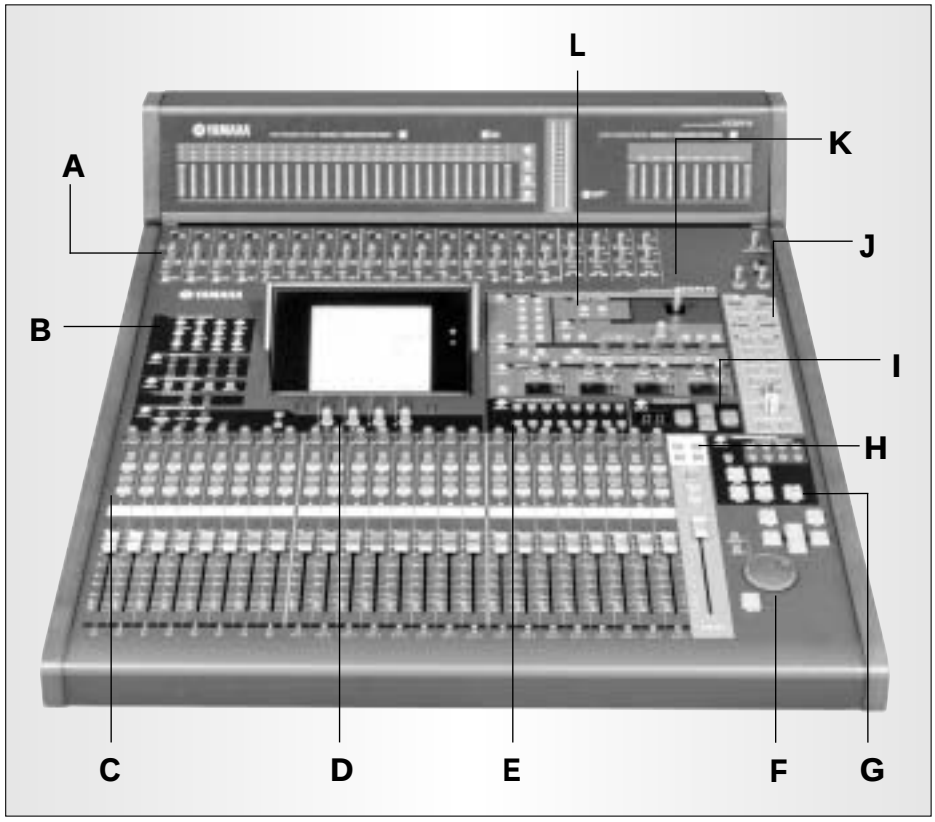
C. Each of the 24 channel strips feature: A touch-sensitive 100 mm motorized fader; a rotary encoder for controlling Pan, Aux Send levels, or user assigned parameters; a channel On/Off key and Solo key; and an Auto key to turn mix automation on or off for that channel. A SEL key allows you to assign a channel strip to be represented on the LCD display and allows you to control the dynamics, EQ, buss assignment and panning for that channel via the dedicated Selected Channel controls.

D. Editing the four internal effects processors is made easy with the parameter up/down selection keys and four rotary encoders. Four keys, located above the encoders switch between menu tabs to reveal additional editing pages.

E. Up to 16 functions from a list of over 150 can be assigned to the User Defined Keys. Setups can be stored in any of the four available banks. Example functions include track arming, scene recall and muting surround monitor outputs.

F. A large Parameter wheel is provided for editing parameter values and scrolling through Scene and library lists. Shuttle and Scrub mode buttons allow you to convert the Parameter wheel to be used for machine control of your DAW or MMC controlled device.

G. Standard transport buttons (Stop, Play, Rec, FF and Rew) as well as as well as 8 locate keys are provided for controlling your DAW via MMC commands.



H. Four Layer switching keys let you access all 56 inputs, the 8 AUX sends and 8 busses as well as a Remote Layer for controlling your DAW via the 24 channel strips.

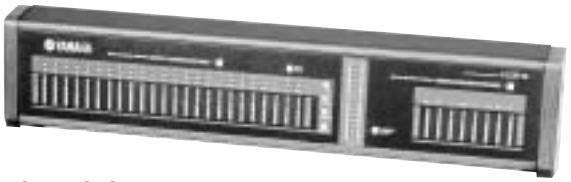
I. A numeric display right next to the Store, Recall, and Up/Down keys shows the current scene number - 01 through 99. Additional scene memories can be managed via a computer running the supplied Studio Manager software.

J. The monitor section features source select keys as well as independent Control Room, Studio, Headphone and Surround level controls. A talkback mic with level control, on/off and Dim switches is also provided.

K. The joystick can be used for surround panning, normal panning as well as parameter control for the 5.1 Reverb effect.

L. Detailed control of dynamics, EQ, buss assignment, panning and surround positioning is available via the Selected Channel controls.

The Optional MB02R96 Meter Bridge Features:



– Twelve 12-segment level meters that can be used to display pre-EQ, pre-fader, or post-fader input channel signal levels.

– An additional eight meters displays the levels for the console's eight busses.

– A 32-segment stereo meter is also provided for the main stereo program.

02R96

– Rear Panel–

A. Sixteen balanced XLR mic and 1/4″ TRS line inputs with 1/4″ TRS inserts (+4dB unbalanced).

B. Channels 17 - 24 feature balanced 1/4″ TRS line level inputs.

C. Analog Master I/O Section

– XLR stereo outputs (+4dB balanced)

– Left and right 1/4″ balanced TRS control room monitor outputs (+4dB balanced)

– Dedicated left and right 1/4″ TRS studio monitor outputs (+4dB balanced)

– 1/4″ TRS studio (+4dB balanced) and RCA (-10dBv unbalanced) 2-track inputs.

– RCA stereo outputs (-10dBv unbalanced)

D. Eight freely assignable balanced 1/4″ TRS Omni outputs can be independently accessed from the Bus Outs, Aux Sends, the Stereo Out, Insert Outs, Direct Outs, or Surround Monitor Channels.

E. AES/EBU digital I/O (XLR) and two coaxial digital I/Os.

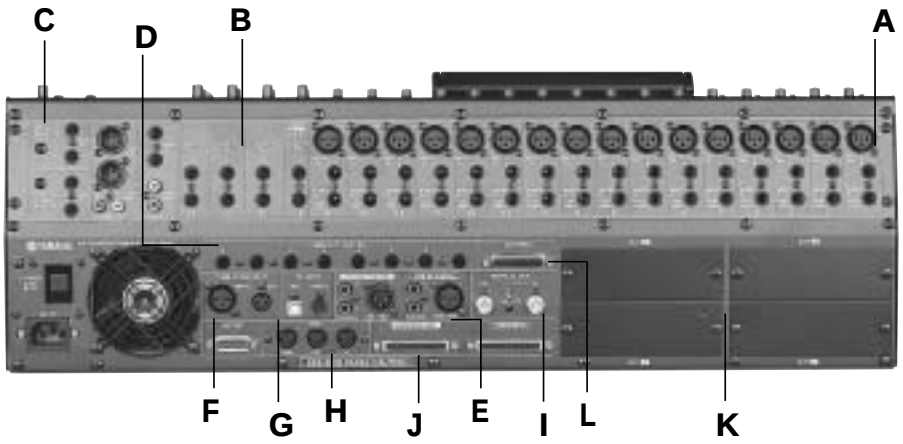
F. A balanced XLR SMPTE timecode input and a dedicated MTC (MIDI Time Code) input are provided for synchronizing the Automix functions with an external device.

G. A USB "To Host" port and 8-pin mini serial DIN "To Host" port allow MIDI communication (including MTC) between the 02R96 and your host computer. The "To Host" ports are ideal for integrating the included Studio Manager Software.

H. MIDI IN, OUT, and THRU ports can be used to send/receive program changes for scene recall, control changes and parameter changes for real-time parameter control, Bulk Dump for data storage, MIDI Clock, MTC, and MMC.

I. Word Clock In and Out is provided by 75Ω BNC connectors – termination can be switched on and off.

J. The 64-pin Cascade In and Out ports can be used to cascade up to four 02R96s to create a multiple-unit mixing system with up to 224 input channels.



K. Four mini-YGDAI slots support up to 32 additional 24-bit/96kHz-capable inputs and outputs (64 I/Os at 44.1/48kHz) via optional analog and digital I/O cards including AES/EBU, ADAT lightpipe, TDIF, and mLAN.

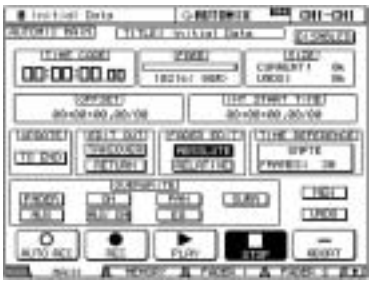
L. A 25-pin D-sub GPI connector allows external equipment to be triggered from specified faders or User Define Keys.

Machine Control

- Both Sony 9-pin (P2) and MMC protocols are supported for external machine control. Control can be switched between MTR and master target machines.
- You can control the transport and locate functions of up to eight external recorders that support MMC (MIDI Machine Control). Machines that support MMC can be controlled by connecting them to the 02R96's MIDI, SERIAL, USB, or with the optional mLAN I/O Card installed in expansion Slot #1.

Automix Functions

- Allows dynamic automation of virtually all mix parameters, including Levels, Mutes, Pan, Surround Pan, Aux Sends, Aux Send Mutes, EQ, effects, and Plug-Ins.
- Events are recorded in real time and can be edited either offline, with high resolution 1/4 frame accuracy, or by re-recording with punch in/out



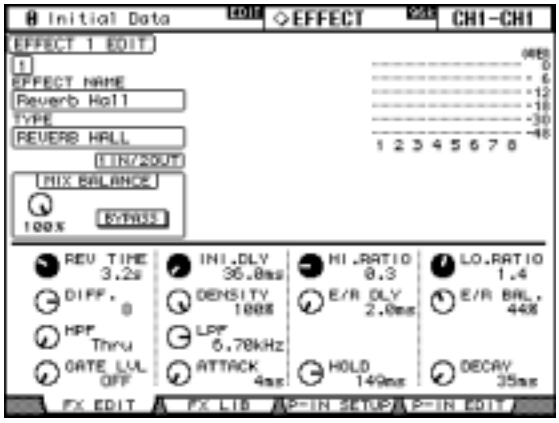
- The smooth and quiet touch-sensitive, 100mm motorized faders make writing and updating automation fast and intuitive.
- You can specify which parameters will be recorded, and punch channels in and out of recording on-the-fly.
- User Defined Remote Layer operations, and scene and library recall operations can also be automated, combining snap shot and dynamic mix automation
- Automix can be synchronized to an external timecode source or to the internal timecode generator
- Up to 16 Automixes can be stored in the Automix library – They can also be stored to an external MIDI device, such as a MIDI data filer, by using MIDI Bulk Dump

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02R96

Four Multi-Effects Processors

- Effects types include reverbs, delays, modulation-based effects, combination effects, and multichannel effects designed especially for use with surround sound.
- Effects processors 2–4 feature assignable stereo inputs and outputs while processor 1 features eight assignable inputs and outputs for use with surround effects.
- An Effects library is provided with 52 preset and 76 user memories.
- Effects processor inputs can be fed from the Aux Sends, Input and Output Channel Insert Outs, or the outputs of another effects processor
- Effects processor outputs can be patched to the Input Channels, Input and Output Channel Insert Ins, or the inputs of another effects processor allowing you to chain processors together in series.
- Joystick control of early reflections and reverb with the Reverb 5.1 effect
- User defined plug-ins for external effects control via MIDI, with Learn function



MULTI-EFFECTS LIBRARY FOR 01V96 • 02R96

Reverbs

#	Preset Name	Description
1	Reverb Hall	Concert hall reverberation simulation with gate
2	Reverb Room	Room reverberation simulation with gate
3	Reverb Stage	Reverb designed for vocals, with gate
4	Reverb Plate	Plate reverb simulation with gate
5	Early Ref.	Early reflections without the subsequent reverb
6	Gate Reverb	Gated early reflections
7	Reverse Gate	Gated reverse early reflections

Delays

#	Preset Name	Description
8	Mono Delay	Simple mono delay
9	Stereo Delay	Simple stereo delay
10	Mod.delay	Simple repeat delay with modulation
11	Delay LCR	3-tap (left, center, right) delay
12	Echo	Stereo delay with crossed left/right feedback

Modulation-based Effects

#	Preset Name	Description
13	Chorus	Chorus
14	Flange	Flanger
15	Symphonic	Proprietary Yamaha effect that produces a richer and more complex modulation than normal chorus
16	Phaser	16-stage stereo phase shifter
17	Auto Pan	Auto-panner
18	Tremolo	Tremolo
19	HQ.Pitch	Mono pitch shifter, producing stable results
20	Dual Pitch	Stereo pitch shifter
21	Rotary	Rotary speaker simulation
22	Ring Mod.	Ring modulator
23	Mod.Filter	Modulated filter

Guitar Effects

#	Preset Name	Description
24	Distortion	Distortion
25	Amp Simulate	Guitar amp simulation

Dynamic Effects

#	Preset Name	Description
26	Dyna.Filter	Dynamically controlled filter
27	Dyna.Flange	Dynamically controlled flanger
28	Dyna.Phaser	Dynamically controlled phase shifter

Combination Effects

#	Preset Name	Description
29	Rev+Chorus	Reverb and chorus in parallel
30	Rev->Chorus	Reverb and chorus in series
31	Rev+Flange	Reverb and flanger in parallel
32	Rev->Flange	Reverb and flanger in series
33	Rev+Sympho.	Reverb and symphonic in parallel
34	Rev->Sympho.	Reverb and symphonic in series
35	Rev->Pan	Reverb and auto-pan in series
36	Delay+ER.	Delay and early reflections in parallel
37	Delay->ER.	Delay and early reflections in series
38	Delay+Rev	Delay and reverb in parallel
39	Delay->Rev	Delay and reverb in series
40	Dist->Delay	Distortion and delay in series

Others

#	Preset Name	Description
41	Multi.Filter	3-band parallel filter (24 dB/octave)
42	Freeze	Simple sampler
43	Stereo Reverb	Stereo reverb
44	Reverb 5.1	6-channel reverb for 5.1 surround
45	Octa Reverb	8-channel reverb for 7.1 surround
46	Auto Pan 5.1	6-channel auto pan for 5.1 surround
47	Chorus 5.1	6-channel chorus for 5.1 surround
48	Flange 5.1	6-channel flanger for 5.1 surround
49	Sympho. 5.1	6-channel symphonic effect for 5.1 surround
50	M. Band Dyna.	Multi-band dynamics processor
51	Comp 5.1	Multi-band compressor for 5.1 surround
52	Compand 5.1	Multi-band compander for 5.1 surround

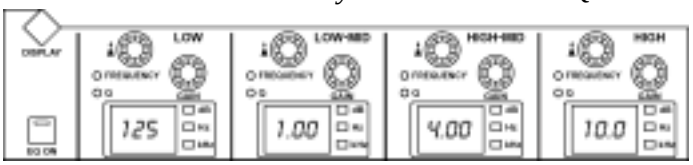
YAMAHA

02R96

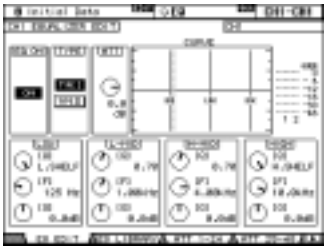
DIGITAL MIXERS

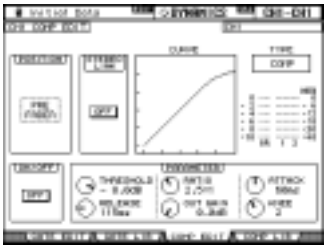
412

Four Band Fully Parametric EQ



Boost/Cut:	±18.0 dB in 0.1 dB steps
Frequency Range:	21.1 Hz to 20.0 kHz (120 steps per 1/12 octave)
Q (Low Band):	HPF, 10.0 to 0.10 (41 steps), L.SHELF
Q (Low-Mid Band):	10.0 to 0.10 (41 steps)
Q (High-Mid Band):	10.0 to 0.10 (41 steps)
Q (High Band):	LPF, 10.0 to 0.10 (41 steps), H.SHELF



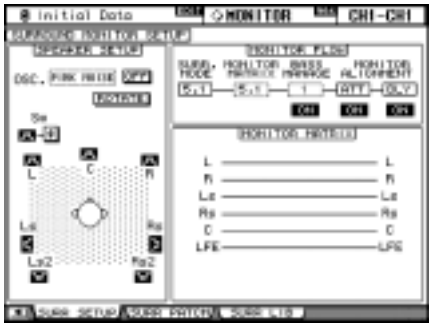


Dynamics



- Independent compression and gating processors are available on all input channels; compression is also provided on all output channels.
- The Selected Channel Dynamics [GATE/COMP] button allows you to switch the dedicated dynamics section to control the Compression or Gating.
- Variable Threshold, Ratio, Attack, Release, and Gain controls are provided for the Compressor.
- Variable Threshold, Range, Attack, Decay, and Hold controls are provided for the Gate.

A Complete Surround Solution



- Everything you need for 5.1 surround processing, panning and monitoring is provided without the need for any external equipment or the need to reassign channels for surround use.
- Multi-channel surround pan/position displays allow you to see where multiple channels sit in the surround mix at a glance.
- The joystick provided for surround panning in the console's SELECTED CHANNEL control section has high 128 x 128 step resolution providing exceptionally smooth control.
- A divergence parameter can be used to adjust the hard/phantom center ratio for each channel.
- The monitor processing section includes individual bus (speaker) muting, attenuation and delay, overall level control for all monitor outputs, stem mix monitoring via independent source selection capability for each expansion slot, bass management for subwoofer delivery, and independent delays for precision speaker tuning.

Remote Control

- Control and manage your 02R96 from your Mac or PC using the bundled Studio Manager software.

Integrated DAW Control

- Mixing and processing parameters, as well as transport control and editing functions of leading digital audio workstations and computer-based recording systems can be controlled directly from the 02R96 control surface. Extensive support is provided for Digidesign ProTools, Steinberg's Nuendo, as well as Emagic's Logic Audio.

Expandable Data Libraries

Setting up EQ, compression, and other parameters for a mix from scratch can be a daunting task, so Yamaha has provided an extensive selection of presets in a range of "libraries" that can simply be selected and used unmodified, or edited to suit specific requirements. Libraries are provided for effects, compression, gating, EQ, I/O patching, and more. Of course, your own setups can be added to the libraries for instant recall whenever they are needed.

- An extensive selection of presets in a range of "libraries" may be selected and used unmodified—or edited to suit specific requirements. Libraries are provided for effects, compression, gating, EQ, I/O patching, and more; user setups can be added to the libraries for instant recall.

	Preset	User
Effect library (Effect 1–4)	52 (Effect 2–4: 44)	76
Presets		
Compressor library	36	92
Gate library	4	124
EQ library	40	160
Channel library	2	127
Surround Monitor library	1	32
Input patch library	1	32
Output patch library	1	32
Bus to Stereo library	1	32

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The 5 Ws of mLAN

What is it ?

mLAN is the enabling technology for creating an intelligent, managed local area music network (LAN) using Firewire. mLAN not only carries multi channel digital audio and MIDI over IEEE1394 Firewire, it includes the connection management so you can easily manage your entire network.

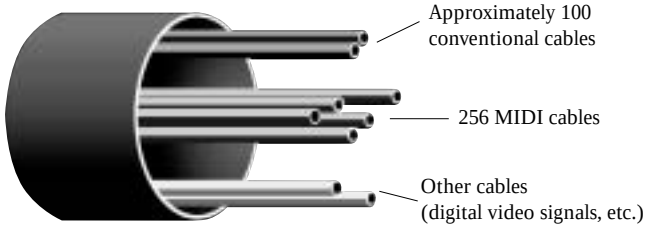
One of the reasons for the complexity and inflexibility of many of today's music production systems and studios is the sheer number of connections involved. All the MIDI and audio signals are routed separately, and each MIDI cable can handle up to 16 channels of data. Audio usually requires a separate cable — whether analog or digital — for each channel of audio for both input and output.
Yamaha's innovative mLAN networking system neatly overcomes all these problems by allowing many channels of digital audio and MIDI music data to be transferred bi–directionally via a single Firewire cable. Extremely powerful music systems can be quickly and easily configured using mLAN-compatible musical instruments, computer components and interface devices. You don't need to worry about the order you connect the devices in, and mLAN ports are "hot pluggable" so you can unplug and plug in devices without having to power-down or reset the system.

When is it?

First generation mLAN products that use the 200 mbps PH-1 chipset are currently available. They allow 8 channels of 48 kHz digital audio i/o and 128 2 Ports (times 16 Channels) of MIDI.
Second generation products are now in development and will be released using the latest PH2 400 mbps chip set. They will be capable of up to 16 x 64 channels of 96 khz audio i/o (or 32 x 128 channels at 44.1 or 48 khz) along with 256 MIDI channels (8 ports times 16 Channels).

Using the software patch bay application provided with all mLAN products, you can easily reconfigure your system – connecting and disconnecting devices as required – without having to physically plug, unplug, or re-route any cables at all.

IEEE 1394 CABLE CONCEPTUAL DIAGRAM



Although computer control can be an advantage in mLAN networks, mLAN allows music and audio devices to be networked without a computer. This allows you to configure an mLAN system for live performances systems.

The other problem that exists in today's market is proprietary formats. Some of these proprietary systems use a Firewire connector, but you can only connect equipment from the manufacturer of the proprietary format which severely limits your choices of gear that you integrate into your studio.

Who is it ?

mLAN was developed by Yamaha, but we realized that for mLAN to be successful, it needed broad participation from other manufacturers. It is available to other manufacturers as a royalty free (no-cost) license. Currently there are over 40 manufacturers who have signed on as mLAN licensees and 8 manufacturers have developed first generation products. Those with current mLAN products include **Apogee**, **Korg**, **Kurzweil**, **Otari**, **Presonus**, **Swissonic**, **Terratec** and **Yamaha**.

Where is it?

mLAN works in Windows XP and there are both ASIO and WDM multi-client drivers available for the PC platform. mLAN is also included in Core Audio as part of OSX 10.2.3 so, you will not even need a driver for Macintosh computers.

Why is it?

So what are the advantages of mLAN over other Firewire audio devices?

Firstly, because other devices use proprietary formats, you can only connect devices from that particular maker to your system. With mLAN, you can hook up an Apogee converter with a Presonus mic pre and a Yamaha synth in one intelligent, well-managed networking system. Because it uses a standard 1394 connection, you can even run video and other data on the same Firewire cable without affecting your mLAN network.

Second, mLAN doesn't require a computer. You can setup your mLAN system at home using a computer and then take the gear to a live gig and hook it together. MLAN will remember the pre-configured setup and re-establish the network and all it's connections.

Third, mLAN handles word clock arbitration so you can run different devices at different sampling frequencies on the same network. Anyone whose tried to setup a digital audio studio knows how important word clock is for a successful studio setup.









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The Professional Audio SourceBook

Section 4

Players & Recorders

PORTABLE 2-TRACK

Cassette

Sony 416

Marantz 416-417

MiniDisc

Sony 418-419

HHB .....420 **Marantz.....421**

Memory Card/Hard Disk

Denon 422

Marantz.....423

Soniflex.....424

Zaxcom Audio 425

HHB 426-427

DAT

Sony 428-429

Eco-Charge ..430 **Porta Brace ..430**

Tascam .....431 **Fostex 431**

CD

Superscope.....432-433

SONY

TC-D5 PROII

Portable Stereo Cassette Recorder

A compact recorder designed for rigorous field use, the TC-D5 PROII provides stable, high performance stereo recording with complete lightweight portability. Measuring 9⅝ x 11⁵⁄₁₆ x 6⅝″ and under 4 lbs. with batteries, the TC-D5 PROII lets you slip onto the scene easily and come back with high quality stereo sound. Features include record-level limiter, large VU meters and XLR-balanced inputs for direct connection with professional microphones.



Basics

- Stereo low impedance XLR mic inputs
- Stereo RCA line outputs
- Dolby B noise reduction
- Built-in speaker and stereo 1/4″ headphone jack let you check your recordings immediately.
- Record-level limiter eliminates clipping, while a microphone attenuator suppresses excess input signal levels by 20dB to prevent distortion.
- Runs for 5-1⁄2 hours with two “D” size alkaline batteries. Also connects to a car battery via a car battery cord.

VU-Metering

- Large, easy-to-read backlit VU meters. One switch illuminates the VU meters and at the same time indicates the battery condition on the L-channel meter. Light shuts off after 10 seconds to save power. LED peak indicator illuminates at +7 dB.

Disc-Drive System

- Featuring a coreless design, the motor is as small as an “AA” size battery but delivers strong torque. Reliable transport mechanism results in increased accuracy and stability and contributes greatly to the exceptional compactness of the TC-D5 PRO II.

Specifications

Frequency Response (CrO2): 40Hz–15kHz

Winding Time: 150 sec. with C60 cassette

Headphones: Stereo phone jack x 1

Speaker: 5cm (2″) diameter, 200mW

Power Requirements: Two “D” size batteries or DC-12v car battery with DCC-127A cord

Dimensions (W x H x D): 9⅝″ x 11⁵⁄₁₆″ x 6⅝″

Weight: 3 lb. 12 oz. with batteries

TC-D5PRO II
With carrying case and shoulder strap...**00.00**



MARANTZ

PMD-101/201/221/222/430

Portable Professional Cassette Recorders



The world standard for field recording, the PMD line is also the value leader. They all feature RCA line input/outputs, 1/4″ headphone jack, built-in speaker, pause control, audible cue and review, tape counter, full auto shut-off and low battery indicator. Used professionally by journalists, law enforcement agencies, aural historians and musicians.

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ONE HOUR FREE PARKING
AT 349 W. 34th STREET (with purchase of $100 or more)

PMD-101/201/221/222/430

- Record limiter and Automatic Level Control (ALC) are great in news gathering. Connect a mic, press record, and get the story while the recorder takes care of levels.
- They run up to 7½ hours on three 'D' cell batteries, or up to 5½ hours with the optional RB430 rechargeable battery.
- Built-in electret condenser microphone is a great back-up in a pinch.
- Modular telephone jack lets you record phone interviews and gives you the ability to play back over a telephone line as well.
- Microphone attenuation is adjustable in three settings from 0 to -20dB
- 3-position ANC (Ambient Noise Control) switch, (flat, low-cut, or low-and-high cut.
- All except the PMD-430 have 1/2 speed playback/record capability. With 1/2 speed playback, musicians can slow down complicated passages for analysis. And when played back at 1/2 speed, the pitch is lowered by exactly one octave, so the notes are still musically correct —great for figuring out complicated solos or picking patterns.
- At 1/2 speed, a three hour meeting can be recorded on a single tape. Built-in speaker makes transcription convenient.
- 1/2 speed recording is great for churches, because 90 minutes can be recorded on a single side—no interrupting your recording to flip the tape over. Line inputs make it easy to use and connect to your existing sound system.

Optional Accessories

CLC221 Vinyl Carrying Case: Cutouts and flaps allow complete operation while in the case, and provides protection against scratches and dents. Fits all except the PMD-430.

WPC221 Weather Resistant Case: Fits all PMD recorders. Cutouts and flaps allow for complete operation while in the case.

RB430 Nicad Rechargeable Battery: Provides 3 hours of playback or 2.5 hours of recording on a 8 hour charge. Can be recharged in the unit.

CA221 Car Adapter: Plugs into a standard automotive cigarette lighter, and powers any of the PMD portable cassette recorders.

General	PMD-101	PMD-201	PMD-221	PMD-222	PMD-430
Stereo/Mono	Mono	Mono	Mono	Mono	Stereo
Heads	2	2	3	3	3
			Inputs/Outputs		
Mic Input	1/4-inch	Miniplug	Miniplug	Mini/XLR	1/4-inch
Condenser Mic	Built-In	Built-In	Built-In	Built-In	—
Remote Jack	—	Yes	Yes	Yes	—
Modular Telephone Jack	—	Yes	Yes	Yes	—
External Speaker Jack	—	Yes	Yes	Yes	—
			Record Controls		
VU Meters	—	1	1	1	2 (Illuminated)
2-Speed Recording	Yes	Yes	Yes	Yes	—
Dolby B NR	—	—	—	—	Yes
dbx NR	—	—	—	—	Yes
Mic Attenuation	—	0,-10dB, -20dB	0,-10dB, -20dB	0,-10dB, -20dB	0,-15dB, -30dB
Ambient Noise Cont.	—	Yes	Yes	Yes	
MPX Filter	—	—	—	—	Yes
Manual Level Control	—	Yes	Yes	Yes	—
Limiter	—	Yes	Yes	Yes	Yes
ALC	Yes	Yes	Yes	Yes	—
Peak Indicator	—	—	Yes	Yes	—
			Playback Controls		
Pitch Control	±20%	±20%	±20%	±20%	±6%
Bias Fine Adj.	—	—	—	—	Yes
Tone Control	Yes	Yes	Yes	Yes	—
Half-Speed Playback	Yes	Yes	Yes	Yes	—
Memory Rewind	—	—	Yes	Yes	Yes
Price	**199.95**	**279.95**	**319.95**	**339.95**	**459.95**

SONY

MZ-NE410

High-Speed Net MD Walkman Recorder

The MZ-NE410 is a high-speed Net MD Walkman Recorder that lets you record your MP3s or CDs at up to 32x normal speed and playback ATRAC3 Audio, MP3, WMA, and WAV files. Enjoy up to 56 hours continuous playback on one AA battery (LP4 Mode), and record up to 5 hours of music on one 80 minute disc. Skip-Free G-Protection Technology enables a smooth listening experience, and the "Easy Skip" Group/Folder Function allows for easy navigation between multiple file folders. Finally, take advantage of the supplied music management software including SonicStage Software and the Net MD Simple Burner Application.



FEATURES

Recording

- MDLP (MiniDisc Long Play) lets you record 2X (LP2) or 4X (LP4) longer on a standard 80-minute MiniDisc for up to 320 minutes (over 5 hours) of music.
- Records from various sources as well as multiple Internet audio formats (MP3, WMA, WAV files) for flexible transfer
- Select from three recording modes: standard mode gives the richest sound quality or two different long play modes for the most effective use of space on a single MD.

Playback

- Playback modes include: Normal and Repeat (1-track, All-tracks) and Shuffle.
- Digital sound presets let you recall the sound quality settings you like for listening to different discs
- Skip-Free G-Protection for perfect playback without skipping, even while running or jogging. Provides shock recovery up to 10-times faster than conventional systems.

High Performance

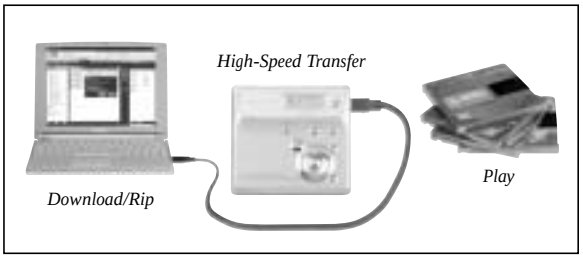
- MiniDiscs are re-recordable, scratch-resistant, low-cost high-capacity media
- Up to 56 hours playback (16 hours recording) on a single AA battery
- Internal battery compartment holds one “AA” type battery
- "Easy Skip" Group/Folder Function allows for easy navigation between multiple file folders transferred from playlist

Net MD Technology

- Breaks the speed limit with a direct USB connection that transfers 80 minutes of music from PC to MD in less than three minutes.
- Bundled SonicStage Software enables you to rip, import and transfer your digital music files (MP3s or CDs) to the Net MD recorder at up to 32x speed from your PC The title/track information is automatically transferred to the Net MD Recorder.
- Bundled Net MD Simple Burner application enables you to dub your CD tracks directly your CD-Rom drive. This gives you all the music you want without storing music on your hard drive
- Compatible with RealOne Player—a music management system which lets you download, create playlists and transfer music to your Net MD recorder

Conveniences

- Fits a jacket pocket for entertainment on the go. Includes MDR-027LP headphones
- Optional digital cables are available to connect to CD/DAT or home systems
- LCD displays remaining battery power and time left on the disc
- Easy pop-up eject mechanism makes it fast and easy to insert or remove MiniDiscs
- Automatic End Search Recording Function informs you how much unrecorded time remains on a disc and begins additional recording from that point,



Download music files from the Internet or rip your CDs. Net MD technology lets you perform high speed transfer from your PC to MD, providing up to 32x speed transfer directly to the recorder.

The supplied SonicStage software makes it a simple task to manage your music and create custom playlists. The SonicStage software also allows you to choose your recording level quality among standard LP2 or LP4 Mode, which allows you to store over five hours of music on one 80 minute disc. Best yet, you can transfer Multiple Audio Codecs to your NetMD recorder. They also play back ATRAC3 and support MP3/WMA/WAV files.

MZ-N510CK / NF610 / NF810CK

MZ-N510CK

The all-blue MZ-N510CK is the same as the MZ-NE410 with the added convenience of an editing remote control, and a Bookmark function. It also comes supplied with a car kit to connect the player to an in-car audio system, as well as an AC adapter.

MZ-NF610

The all-blue MZ-NF610 doesn't include a car kit, but adds a AM/F/TV Weather Band tuner and a NH-7WMAA rechargeable battery. A car kit (CPA-9C) is available as an option.

MZ-NF810CK

The all-silver MZ-NF810CK steps up with an external mic jack, supplied car kit and a battery charging stand. The stand lets you access the MZ-NF810CK while simultaneously charging the battery by simply placing the recorder in the stand.

It is also equipped with a 6-band Graphic EQ (4 presets/2 custom) gives you more flexibility in choosing which sound setting best suits the music. The four presets are Heavy, Pop, Jazz and Walkman Unique. You can also create different listening environments by digitally adjusting sound resonation. The four presets are Studio, Live, Club and Arena.

SIMPLE BURNER INTERFACE



The bundled Sony NetMD Simple Burner/Quick Rip CD dubbing application gives you all the music you want without storing or managing music on your hard drive.

Sony's NetMD Simple Burner provides an easy to use CD dubbing application that makes dubs at accelerated speed. It gives you the freedom to dub straight from CDs to a NetMD recorder. Easy to use, simply insert the CD and double click the NetMD Simple Burner icon on the desktop. This opens the dubbing application with the CD tracks displayed on the left. All that's left to do is select the the tracks to be transferred to the NetMD recorder, select recording mode (LP2, LP4) hit the record button, and the selected tracks are transferred to the recorder.

With the Simple Burner, Quick Rip CD dubbing application, transferring music from CD to MD is faster and easier than ever.

MZ-N10

64x Net MD Walkman Recorder



The pinnacle of the Net MD Walkman line, the MZ-N10 packs a wealth of features into a compact silver magnesium body that weighs just 3oz. Besides recording MP3s and CDs from your PC at up to 64x speed, the MZ-N10 also makes creating custom music mixes even easier with a backlit LCD editing remote and a three-line dot matrix display right on the unit that enables you to view nine characters per line for easy song searches and edits. Sony's Jog Dial navigator scrolls through the list vertically to search for songs and groups. By simply rotating the dial with one finger the pushing it, the desired point is accessed. The MZ-N10 is supplied with a matching USB Cradle for direct connection to the SonicStage jukebox software.

Same features as MZ-NF810CK (no car kit) PLUS—

- USB Cradle connects the MZ-N10 directly to the PC while at the same time recharging the internal battery.
- 3-line dot matrix display features track number, MDLP recording mode, elapsed time, song title, artist name, play mode, depending on the mode chosen.
- Backlit LCD remote with editing functions provides direct access to the recorder's functions and is easy to see, even in low light.

MDP-500 'PortaDisc'



Professional Portable MiniDisc Recorder

An advanced portable professional MiniDisc recorder, the MDP500 'PortaDisc' is designed and engineered without compromise to handle the most demanding field recording applications. Built on a rigid steel chassis and based around a rugged, professional MD drive, the MDP-500 combines advanced sound quality with a full complement of professional features, comprehensive connectivity and flexible powering options. Ideal for journalists, broadcasters and sound recordists in radio, TV, film and music recording applications, the MDP-500 offers XLR balanced mic inputs with switchable phantom power, digital I/O and even a USB interface which allows realtime recording of audio into a laptop or desktop computer for later editing.



FEATURES

High Quality

- Advanced V4.5 ATRAC recording algorithm combines with high performance mic preamps, precision DACs and on-board limiters and bass roll-off to deliver recordings of exceptional quality.
- For maximum reliability, the PORTADISC uses a dependable, professional MiniDisc drive - not a low-cost consumer device
- 40-second memory buffer ensures consistent, glitch-free recording on the move
- Pre-buffer continuously records 6 seconds of audio before the Record switch is pressed to ensure that you never miss a take.
- Comprehensive range of connectivity includes balanced XLR Mic/Line inputs, unbalanced RCA/phono line outputs, and both coaxial and optical S/PDIF digital I/Os.



- Mic inputs feature switchable attenuation, (0dB, -15dB, -30dB, Bass Roll-off (Off, 75Hz, 150Hz), Limiter (On, Off, Ganged, AGC) and 48V phantom power.

High Performance

- Auto Start feature with variable threshold makes the PORTADISC ideal for use in noise monitoring applications.
- Recording from digital sources is easy with a built-in sample rate converter accepting 32 and 48kHz signals and auto sync recording with CD and DAT IDs converted automatically to MD tracks.
- In addition to disc and track naming, basic editing lets you go directly to a particular point in a track, combine two tracks together to make one, divide one track into two, move and erase tracks.
- The recorder is housed in a tough 1.2mm steel chassis with transport keys and function buttons protected by a durable rubber molding.
- Tracks can be numbered manually during recording with the Mark button, or automatically with the threshold level variable from 30dB to 60dB in 10dB increments. Tracks can be accessed directly using the AMS Forward and Back keys.
- In the event of a microphone or cable failure, the Portadisc even includes an on-board microphone for back-up.
- Headphone monitoring is available in Stereo, Mono-L, Mono-R and Both modes, with an additional, mono internal speaker provided for non-critical monitoring.

USB Interface

- Unique to the PORTADISC is a USB interface enabling the transfer of files to and from a PC. This allows suitably equipped journalists and sound recordist to record and edit finished programs and news pieces in the field and transfer them via e-mail or write them to CD.

Ease of Use

- Using a simple, intuitive menu you can configure and store up to 5 User Set-Ups selecting input sources, mic limiting /bass roll-off, phantom power, headphone monitoring formats, time /date display formats, auto track increment, threshold levels, etc.
- The main LCD display is large and clear with switchable illumination and can be viewed from a wide range of angles. 19 segment metering is provided with switchable peak hold and a margin indicator accurate in 1dB steps to -60dB.
- Record level controls are lockable, the phone level control retracts and a Keyhold facility prohibits accidental mis-operation during recording.
- Handles perfectly in the field. Transport keys, record level controls and all major functions are ideally located for quick, easy access and menu structures are straightforward and logical.

B&H PHOTO - VIDEO - PRO AUDIO

PMD-650

Professional Portable MiniDisc Recorder



Offering the same basic features as their legendary portable cassette recorders while providing the benefits of a sophisticated digital recorder, the PMD-650 is ideal for interviews, music, multimedia, broadcast video and film production. It includes two XLR-balanced mic/line inputs, a built-in mic, and a monitor speaker. It supports 74 minutes of stereo and 148 minutes of mono record and playback, and has a host of practical features that make quick work of any high pressured production environment. For example, when recording from one mic/line source onto two tracks, the PMD-650 incorporates a dual mono mode that sets the level of the 2nd track 15dB lower than the first, to make sure that a backup track is available in the event of a problem or digital overload.

FEATURES

Versatile Recording

- Editing capabilities including divide, combine, move, erase allows tracks to be trimmed and combined in order to create custom sound files.
- Time/Date stamp marks every track recorded on the PMD650 providing a reliable reference for archiving.
- Pre-Record Audio Cache (up to 2 seconds) provides protection for late starts.w
- Either one or two mic/line sources can be recorded for 148 minutes onto 1 (LP) or 74 minutes on 2 (SP) tracks.
- Three record level controls: manual, manual with limiter, and automatic
- Programmable Level Sync Recording (LSR) automates recording and saves disc space by only recording when there is sound present. LSR parameters include sound threshold level (-10dB to -60dB), time below threshold level (2-10 seconds), and automatic or manual track increment.
- One touch recording with separate Rec-Pause button

High Performance

- Two position noise cancel filter
- 40 second (20 sec. 2-track) audio buffer for shock resistance
- 48v phantom power
- Variable mic attenuator (0, -15, -30dB)
- The PMD650 secures all recordings by writing a backup of the Table of Contents (TOC) at the beginning of the track (Pre-UTOC). Even if a recording is interrupted by a power failure, all recorded information can be retrieved.

Conveniences

- Built-in microphone and speaker
- Backlit LCD display
- A-B loop playback, Repeat Playback (all or one track)
- 3-way power: AC adapter, 8 AA batteries, or optional RB-1100 Nicad battery. Has a Low-Battery Alarm
- Control of SCMS (on or off)

Inputs/Outputs



Full complement of professional inputs and outputs allow total integration with professional broadcast studio and field equipment.

- Stereo XLR mic/line inputs
- Stereo RCA analog line outputs
- S/PDIF coaxial digital input with sample rate converter and XLR digital output (AES/EBU)
- Remote control (RC-5) input
- Headphone jack with level control

Comparison:	MDP 500	PMD 650
Rugged Pro transport	Yes	No
Disc autoload	Yes	No
MD ATRAC version	4.5	4
AES/EBU	None	Output
S/PDIF (Coaxial)	In/Out	Input
S/PDIF (Optical)	In/Out	None
Bidir. USB interface	Yes	No
Auto start/cut with adjustable level	Yes	Yes
Pre record buffer	6 sec.	2 sec.
User presets for ALL system settings	Yes 5	Limited system only
Split level mono record (2nd track -15dB)	No	Yes
MD edit - divide, combine,move,erase	Yes	Yes
Adjustable display contrast	Yes	No
Meter number of segments	19	14
Numerical margin indicator	Yes	No
Time/date stamp	Yes	No
Input monitor without disc present	Yes	No
Input level lock	Yes	No
Limiters switchable ganged for stereo	Yes	No
Backspace function (last record erase)	No	Yes
Remote control	8 pin mini-din	Marantz RC5 serial
Batteries supplied	Rechargeable	None
Case	Supplied	option
Record time with rechargeable battery	> 3 hours	2.5 hours
SRC Switchable on/off	No	Yes

DENON

DN-F20R

Portable IC Recorder

Weighing just over two pounds and offering a full list of professional broadcast features, the compact DN-F20R is ideal for ENG applications. A portable IC recorder that uses IC memory cards as the recording medium, the DN-F20R does away with cumbersome mechanisms and enhancing reliability. Two card slots provide recording flexibility. Recordings on the DN-F20R are created as sound files in MPEG format which can be easily ported to a PC for editing and manipulation.



FEATURES

- Use of an IC memory card as the recording medium does away with moving parts and ensures high reliability. Has 2 slots for CompactFlash cards.
- The IC memory card complies with the CompactFlash Type 1 standard and supports large memory capacity of 256MB. The card can be read and written to by a PC equipped with a USB CompactFlash adapter or an ATA adapter for a laptop.
- Inputs include two XLR mic jacks and two RCA line inputs, as well as a parallel remote mini-jack input for external control. Outputs include two RCA line outs and a 1/4˝ headphone jack with adjustable levels.
- Audio may be recorded in a choice of MPEG2 Layer2, MPEG1 Layer 2 or Linear PCM (WAVE file) formats, in either stereo or mono modes.
- Sound quality is also user-selectable (in recording bit rates of 16 to 768 kbps per channel) to suit the application, enabling efficient use of the cards, compression and high sound quality.
- Concentric rotary controls are provided to adjust the DN-F20R's recording levels. Level metering can be monitored on the unit's backlit LCD display, which also shows recording settings, file names and other various information at a glance.
- Offers switches for a low-cut filter, limiter and attenuator.
- To prevent accidental turn-on or misuse while recording, a 'key hold' switch can be activated to lock out the operating buttons.
- Compact, easy-to-carry A5 size even with XLR terminals.
- Can be powered by 6 AA batteries or using the included AC adapter.
- Third-party hardware and software such as Digigrams's MPEG board and Xtrack editing software can be used to edit sound data waveforms in their compressed form after they have been recorded by a DN-F20R.

SanDisk CompactFlash Cards for Denon and Marantz Recorders





Capture more music and voice recordings with SanDisk flash memory cards, then quickly transfer your data to your computer with an ImageMate drive or CompactFlash adapter. About the size of a matchbook and weighing half an ounce, the cards were designed based on the popular PC Card (PCMCIA) standard and are fully compatible with the Denon DN-F20R and Marantz PMD-680/690 portable recorders.

- Highest transfer rates for fast copy and download—up to 1.2 MB per second.
- High storage capacity
- Operating shock rating of 2,000Gs, equivalent to a 10´ drop to the floor
- Unfazed by drastic weather conditions ranging from blistering heat to arctic cold
- Non-volatile solid-state; no moving parts maximizes battery power. Data isn't lost when power is turned off

32MB CF Card (SACF32MB).....**CALL**
64MB CF Card (SACF64MB).....**CALL**
128MB CF Card (SACF128MB).....**CALL**
192MB CF Card (SACF192MB).....**CALL**
256MB CF Card (SACF256MB).....**CALL**
512MB CF Card (SACF512MB).....**CALL**
1GB CF Card (SACF1GB) .....**CALL**

PMD-680/PMD-690

Portable PC Card Recorders



Compact and portable, the PMD-680 (mono) and PMD-690 (stereo) are next generation digital recorders that bridge the gap between real-time audio recording and computers — while keeping the size and functionality of Marantz's renowned portable cassette and MiniDisc recorders. The PMD-680 or PMD-690 record high-quality digital audio files to a CF (Compact Flash) compatible card or IBM Microdrive, where they can be immediately transferred to a computer for archiving, editing, or uploading to the Internet. Designed as a field acquisition tools, the PMD-680 and PMD-690 can record audio as compressed MP2 digital audio files stored as raw MP2 (.mp2) or the Broadcast Wave standard (.bwf), as well as uncompressed using 16-bit/48kHz Pulse Code Modulation (PCM) saved in the true wave (.wav) file format.

FEATURES

- Their Type III PC Card slot is compatible with both compact flash (using an adapter) and ATA-size PC Cards. The slot also accepts cost-effective IBM MicroDrives.
- Highly reliable, flash memory PC Cards contain no moving parts and are impervious to temperature extremes and shock which might damage recordings.
- Compressed MP2 digital audio can be stored as raw MP2 (.mp2) or Broadcast Wave (.bwf). files. Uncompressed recording (PCM) can be saved in Wave (.wav) format.
- They offer three ways to control the recording level: Manual, manual with limiter, and automatic (ALC) record level control
- Built-in Time and Date generator marks each recording
- Band or high-pass ANC (Ambient Noise Cancel) filter lets the recorders filter out wanted background noise.
- Pre-recording memory buffer records 2 seconds of audio before the record button is pressed. This prevents missed or delayed starts of a recording.
- A silent skip mode (1 to 5 seconds) can be activated based on parameters set in the menu. Sound levels for silent skip can be set from -50 dB to -10 dB in 5 dB steps.
- Remote input for pausing and un-pausing during recording or playback
- They can be powered three ways: via the supplied AC adapter, rechargeable Ni-Cad battery pack (optional) 8 AA batteries. l
- Inputs and outputs include XLR and 1/4″ mic inputs, RCA line output and coaxial RCA digital output (S/PDIF). They also feature a headphone output, built-in microphone and built-in speaker.
- Each recorder includes an AC adapter, battery holder, and carrying strap.

Two Different Recording Formats

The MP2 (MPEG1 layer2) file format that is supported by the PMD-680/690 is a world-wide standard for compressed digital audio storage and transmission. Many playback and editing systems are available commercially and through the Internet.

PCM (Pulse Code Modulation) is the most widely used format for coding uncompressed digital audio.The PCM system is used on CD players, DAT recorders, and on computer editing programs that support Wave (.wav)files. Recording in the PCM format will provide the most universally accepted storage format but comes with the limitation that it uses a lot of memory very quickly.

Non-Destructive EDL System

- They use an on-the-fly system for marking tracks during recording. EDL marking and playback allows either a custom playback sequence to be programmed or the audio between two specific points in the card to be repeatedly played back.
- During recording EDL marks are placed on the card at the beginning of each new track; when REC/MARK button is pressed; or when the Auto Mark function is active. When Auto Mark is active, an EDL mark is placed every time a recording is paused, manually or by the Silent Skip system.
- The maximum number of EDL marks per card is 255. EDL marks can be edited by changing the mark types, erasing specific marks, or erasing all the marks.

PDM-680 Only

- Monaural audio recording and playback
- Has XLR and 1/4-inch mic input with -15/-30 dB attenuator, and RCA line input.
- Telephone jack for recording and playing back phone conversations. This jack also passes through live recording signals to a telephone line.

PDM-690 Only

- Stereo audio recording and playback
- Two XLR mic line inputs with -15/-30 dB attenuator. +48v phantom power is available for use with condenser mics.

COURIER

Portable Hard Disk Audio Recorder



The Courier is a compact (3 lb.) battery-powered digital audio recorder designed for in-the-field or on location applications. Using industry-standard PCMCIA cards to store only high quality digital audio, the Courier can be used while moving or running. The cards can be changed quickly, and are capable of storing up to 18 hours of audio. Two inputs are available for either a single stereo source, or two mono sources, which can be at mic or line level. Each input has a level control, which can be used separately or ganged together - the record level control is recessed so that it can't easily be knocked. Phantom power and a high pass filter, together with a limiter, can be switched in.

When recording in the field you often only get one chance, so it is imperative that you can rely on your recorder. Courier has a confidence monitor - it is able to replay material recorded to disk while still recording. Once you are recording, the 'Autolock' function is activated which disables all other controls, except record levels, so you can't accidentally stop your recording. Courier also features on-board audio editing which allows full edit and audition of clips before sending back to the studio using a number of flexible methods. These include built in ISDN support, software for transfer via standard telephone line or over a digital telephone network via mobile phone or audio playback. Courier is powered via standard camcorder batteries. There is an audible warning when the battery is running low and the battery can even be changed without interrupting the recording process. Includes a carrying case, nicad battery, and universal AC adapter.

FEATURES

Control At Your Fingertips

- Controls are designed for ease of use. Large, readily identifiable keys are used with added benefits such as 'feel in the dark' operation and accidental switch-off prevention.
- A confidence monitor confirms that your recording is safe and a backlit LCD display gives you vital information on the recording levels, battery and disk time remaining.
- A software controlled limiter can be switched in together with a high pass filter for use in noisy surroundings and there are "over level" LEDs for an instant visual check on recording levels. These let you concentrate on your job not your recorder.

Live Audio

- Provide your listeners with all the excitement of live commentary. Connect the COTBU telephone balance unit and you're ready to broadcast down a telephone line, either by using the mic/line inputs or by playing pre-recorded cuts.

USB Connection

USB connection allows fast downloads to the PC to be made. The USB port works in addition to the RS232 serial port and the optional ISDN connection.

Record in 'Style'

- Courier's superior versatility lets you define 'Record Styles' to make your report perfectly compatible with the majority of editing and broadcast playout systems. Simply define parameters such as sample rate, file format, compression used and whether mono or stereo, then give your recording style a name such as 'Radio' or 'CD'.
- Up to 20 personalized Record Styles can be configured. Once you've selected your chosen style and finished recording, plug the PCMCIA disk or flashcard straight into your editing or playout system for instant editing or playback. No need for audio dubbing - what could be simpler?

Editing and Transmission

- Courier has sophisticated non-destructive accurate editing capabilities. By using the jog shuttle wheel, and watching the waveform on the graphical display, you can place multiple marks, and perform complex editing simply. A playlist can be compiled by taking pieces from any number of cuts, which can be saved in a number of different formats.
- Once you've finished recording, simply connect Courier to a modem and sit back and relax as it sends data down the phone line. Or you can send your report over a digital phone network

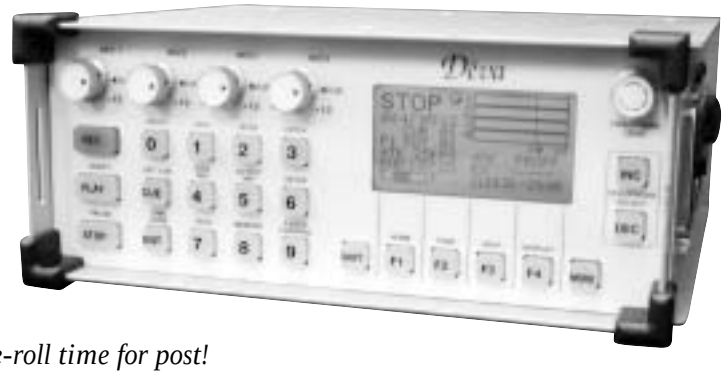
Portable Power

- The internal rechargeable battery gives over 1 minute of hot-swap time so you can change the main battery. An optional cigarette lighter adapter recharges batteries on your way to the next assignment. Includes a worldwide AC adapter.

B&H PHOTO - VIDEO - PRO AUDIO EQUIPMENT LEASING AVAILABLE

DEVA II

Portable 4-Channel Hard Disk Recorder/Player/Mixer



Deva II a portable 4-track, 24-bit uncompressed hard disk recorder, digital mixer and timecode generator/ reader that stands alone as the most reliable way to record in the field. Used for countless feature films and TV shows, you can use it as a replacement for all your current audio field recorders, bringing 48kHz studio-quality recording capabilities into the realm of location production. Pre-Record function records up to 10 seconds of 4 channel audio from just before the record key is presses. That means no more missed cues and plenty of pre-roll time for post! And with an input dynamic range of 110dB, Deva II provides the highest quality field recording available today.

FEATURES

- Using the optional 40GB removable IDE hard drive provides a maximum record length of 68 hours (mono), 34 hours (stereo), or 17 hours (4 channel). The 125 G-force rated disk is immune to dust and humidity changes, and operates over a wide range of temperatures. An optional 4GB drive with 8 hour record capability is available as well.
- Zaxcom's Mobile Audio Recording Format (MARF) is designed for the specific requirements of multi-channel disk recording, providing the ultimate in data integrity. So robust is MARF's design, Deva II can flawlessly recover audio from a disk that may have had its directory re-initialized.
- Sophisticated error-checking circuitry provides a fool-proof means to flawless recordings. During recording, Deva II performs a constant Read/Verify routine. If an uncorrectable error should occur, audible (to the headphones only) and visual warnings are generated. It is virtually impossible to unknowingly produce a bad recording.
- Unlike mechanical movements whose calibration and response may change with temperature, metering, Deva II's LCD display is always calibrated.
- Zaxcom's exclusive NeverClip technology provides precise input level detection, making it extremely difficult to overload the A-D converter. NeverClip provides smooth, automatic gain manipulation, assuring virtually distortion-free recordings.
- Deva II incorporates a state-of-the-art digital mixer. Any of the 4 inputs may be mixed to any of the 4 disk channels. Similarly, any of the 4 inputs may be mixed to any of the 4 line outputs – a mix that may be completely different from the disk mix! And unlike analog field mixers, Deva II's pots are immune from the "scratch noise" syndrome.
- Besides the front panel controls, Deva II's Sony BVW75 emulation permits frame-accurate control via an edit system, telecine, or synchronizer. Deva II also provides a time code chase facility that has a lock time of under one second.
- Full-function internal timecode reader/ generator supports 24, 25, 29.97 (NDF and DF), and 30 (NDF and DF) frame rates. The generator can even automatically tag consecutive recordings with ascending User Bits. Nothing is more flexible and efficient at helping you locate the "buy take."
- No matter what the monitoring requirements, Deva II's 4 channels of "line level" analog and AES/EBU outputs, along with a user-configured stereo headphone connection, means you always hear what you want.
- Accepts a wide range of microphones, offering phantom power, when required.
- High-quality 20-bit A/D and D/A converters on both the mic and line inputs ensure wide dynamic range, while the 24-bit interface guarantees transparent transfers from digital sources.
- A standard Deva II gives you 48, 48.048 and 47.952 kHz operation as standard. If required, 44.1 is available as an option.
- 4 channels of 20-bit A/D converters and 24 bit record depth provide 16 times the resolution of DAT in a package that is inherently more reliable and flexible.
- Deva II remembers up to 100 cue points per disk for instant access to "buy takes" or pre-produced playback audio.
- Deva II will output files to DVD drives in up to 6 file formats. Sound Designer 2 and Broadcast Wave are both fully supported. External disks can be recorded in the background so at the end of the recording day there is no transfer time.
- Deva II's internal software allows for replay of audio data even when the disk directory has been erased. It also means that Deva II's software is crash proof. Even if power is lost while recording; Deva II will seamlessly playback audio to the time of the loss
- What happens to if Deva II is accidentally dropped while recording? Probably nothing! A unique mechanical shock-sensing circuit automatically shifts recording to a 24MB RAM buffer. Once Deva II senses that conditions are favorable, the buffer is automatically transferred to disk.
- Powered via NP1-type camera battery for over three hours of continuous operation. Deva II may also be powered via a separate 12v supply – even a car battery.

PDR-2000 PortaDrive

8 Track Location Sound Recorder



With 8 tracks of pristine 24-bit/96kHz recording, flexible on-board mixing and comprehensive timecode facilities, the PDR-2000 PortaDrive sets the standard for location sound recording. The PDR-2000 records audio onto a compact, robust, shock resistant removable 2.5˝ hard drive in industry-standard BWF or SDII formats. This allows full compatibility with Mac and PC-based digital audio workstations, connecting via an optional drive docking station. A comprehensively equipped 6 into 2 digital mixer is built-in, enabling the simultaneous recording of a stereo mix alongside 6 discrete inputs. Microphone inputs are of the highest quality with 6 high-gain, ultra-low noise balanced XLR Mic/Line inputs with individual phantom powering, 'gangable' limiters, attenuation, high pass filter, delay and phase reverse. The 2-channel return input and both the main and auxiliary analog stereo outputs are balanced. Digital connectivity is equally comprehensive with 8-channels of AES I/O on an industry standard 25-pin D-sub connector, S/PDIF input and AES and S/PDIF stereo digital outputs.

FEATURES

8-Track Location Sound Recorder

- Rugged, all in one, portable 8-channel hard disk (HD) audio recorder
- Audio is recorded in industry standard BWF (Broadcast Wave) and SDII (Sound Designer II) formats onto an included 20GB 2.5˝ hard drive housed in a compact, robust, shock resistant removable caddy
- Over 2 hours of uncompressed 8-channel 24-bit/96kHz or over 9 hours of 4-channel 24-bit/48kHz recording on removable HD
- Metadata can be stored and transferred with audio files
- FAT32, HFS hard drive formatting ensures compatibility with Mac and PC-based digital audio workstations
- Session based recording using either AES31 ADL or Pro Tools V5 formats, simplifying production workflow
- 10 second pre-record buffer
- Built-in slate mic and tone generator
- Advanced power management

Extensive Analog and Digital Inputs and Outputs

- 6 high gain XLR balanced mic/line input channels with individual phantom powering, ‘gangable’ limiters, input pads, HPF, delay and phase reverse
- 2-channel balanced line level analog return input using a single 5-pin connector
- Balanced XLR analog main and auxiliary outputs
- Stereo AES/EBU digital output (balanced XLR) and coaxial SPDIF digital I/O
- 1/4˝ balanced jack headphone socket
- 8 channels of AES digital inputs and outputs on a 25-pin D-sub connector

6x2 Digital Mixer

- The built-in 6 x 2 digital mixer enables the simultaneous recording of a stereo mix alongside 6 discrete inputs
- M/S decoding is available on inputs and outputs
- Complete routing flexibility allows you to easily record from analog and digital sources simultaneously
- Extensive headphone monitoring capability including designated 6-2 mixer

Timecode and Sync

- Reads and generates timecode in all popular international frame rates – a 5-pin LEMO socket is used for interfacing
- Supports tri-level synchronization used with emerging HDTV technology
- Dedicated word clock output on a BNC connector
- Sync input (word clock, video, tri-level) on BNC connector
- Comprehensive remote control via RS422 (Sony P2) and parallel remote socket



Data Communication Interfaces

- A SCSI interface is provided for data transfer to and from external storage device
- USB 2.0 and Ethernet ports support high speed data transfer to/from a computer as well as software upgrades
- A PS2 port allows you to connect and ASCI keyboard to ease logging and labeling

PDR-2000 PORTADRIVE

Front Panel

- 6 multi-function rotary encoders with collar LEDs allow displayed values to be altered
- A 5-position rotary transport control allows you to initiate Record (with or without lockout; Stop (with and without lockout); a Review function allows the user to check the last audio take and a Mark button enables easy identification of points within a session
- Four Primary Mode (PMODE) buttons allow rapid access to the control, metering and monitoring of all audio signals
- A large transreflective LCD displays the menu associated with the currently selected PMODE and the 6 rotary encoders
- Undo, Jam TC, Slate, Tone, and Group functions above the display are protected from accidental use with an 'Enable' button
- A push-lock rotary level control prevents accidental adjustment of the headphone output and built in speaker

Metering

- 18-segment high-visibility LED meters for each channel have selectable ballistics allowing variable peak hold time as well as a variety of display modes for power economy
- Digital clip indicators, limiter activity and track record arm status are also provided
- A tri-color LED indicates data write errors, TC/sync errors and low battery level

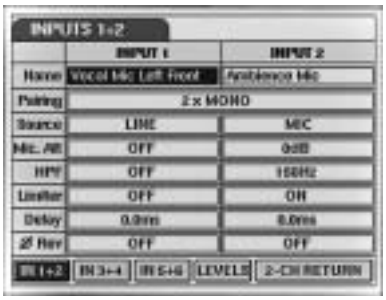
Top Panel Functions

- A highly intuitive, hierarchical navigation system is accessed by 8 dedicated menu function buttons – Input, Busmix, Track and Output functions mirror those on the front panel; Disk, Session, TC and Setup functions allow recording sessions to be configured to suit your workflow
- Four cursor keys and a large data wheel allow you to navigate and alter information displayed on the large transreflective LCD
- Further pages within the selected menu are listed along the bottom of the LCD and are easily accessed through the row of function buttons immediately below the screen
- Functions can be disabled to avoid mis-operation and to conserve battery life
- **Large transport controls** – besides the usual Play, Stop and Cue functions, a Locate button allows you to access audio by take, marker or timecode value

Easy To Navigate Menu System

INPUT

- Set-up of each input, including name, pairing, source, 48V phantom powering, attenuation, HPF, limiter, adjustable delay and phase reverse



- Adjust input levels enable the 2-channel return with optional limiter

BUSMIX

- Create a 2-channel mix of inputs 1-6 with pan, level adjustment and M/S decoding
- Route the mix to 2 disk tracks – ideal for exporting rushes at the end of a session

TRACK

- Source select for each disk track, M/S decoding as well as setting of the number of recorded tracks for each take

OUTPUT

- Configure the output mix with Source, nominal level and M/S decoding settings for the Main and Aux outputs
- Digital options include source select, sample rate, bit depth, etc.

DISK

- Controls formatting, file and folder creation/deletion and naming on each disk

SESSION

- Specify disk format, sampling rate, bit depth, number of tracks, TC/sync
- Sessions can be named, stored and recalled – information is continuously saved

TC

- Timecode menu with internal generator, chase, jam and user-bit capability
- Frame rates include 23.976, 24, 25, 29.97df, 29.97nd, 30df and 30nd. FPS, with T.O.D., Rec Run, Free Run and external timecode modes

SETUP

- Controls general system parameters including tone generator, slate, power management, LCD contrast, meter and time / date settings

- Powered by an industry standard 50 watt hour Lithium-Ion NP1 battery or an A/C adapter (both supplied)
- When fully charged, the battery provides a minimum 2 hours of constant operation
- When the PORTADRIVE is not in use, the A/C adapter doubles as a charger for the battery





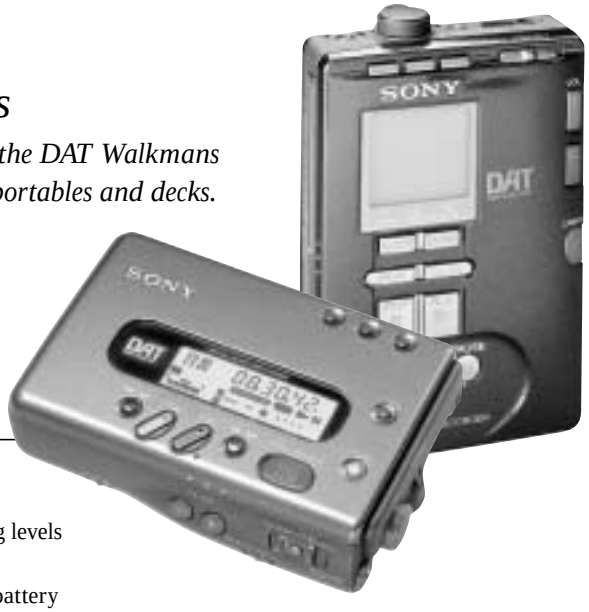
- An optional 5.25″ Docking Station allows the removable HD caddy to be plugged directly into a computer drive bay or external drive enclosure

SONY

TCD-D8 / D100 / PCM-M1

Portable DAT 'Walkman' Recorders

Ultra-portable digital audio recorders that fit in the palm of your hand, the DAT Walkmans by Sony feature the same pristine CD sound quality found in high end portables and decks. They feature selectable sampling rates of 48kHz, 44.1kHz and 32kHz. Analog and digital I/O allow them to integrate seamlessly with most hard disk recorders, sound cards, digital samplers and non-linear video editing systems. Ideal for live music, sound effects and dialog recording, they deliver superb audio at an affordable price.



FEATURES

- 16-bit record mode for CD quality sound (12-bit in Long Play Mode)
- Selectable sampling rates of 48kHz, 44.1kHz and 32kHz for both analog and digital recording.
- Dynamic range over 87dB
- Digital input and output.
- 1/8″ (3.5mm) stereo mic/line input
- 1/8″ (3.5mm) stereo line output
- Long play mode for up to 4 hours of recording on a single 2 hour tape.
- Automatic and manual recording levels are switchable
- Large backlit display with level, battery status, time & date and program info
- ID subcodes can be set either automatically or manually while recording.
- Hold switch prevents unwanted operation of transport functions.
- 100x high speed AMS (Automatic Music Sensor) search function finds and plays tracks, skips forward or back up to 99 tracks at 100 times speed.
- Automatic Volume Limiter System (AVLS) maintains volume output at levels that are below distortion threshold for cleaner sound reproduction.
- Automatic date function records the date and time for easy track identification.
- SCMS (Serial Copy Management System)

Optional Microphones for DAT/MiniDisc Recorders

ECM-MS907 *One-Point Stereo Mic*

Incorporating a sensitive electret condenser mic and Mid/Side switch, the ECM-MS907 is designed for high-quality instrument recording with Minidisc and DAT recorders. Features Mid/Side (MS) capsules for natural stereo panorama; the Mid capsule picks up mono sound while the Side capsule picks up left/right difference sound. Mid/Side (MS) switch selects pickup angle between left and right channel; choose 90° for a single voice or instrument or 120° to pick up many voices and instruments, arranged across the stage. Supplied with stand/holder, windscreen and carrying case.





ECM-717 *One-Point Stereo Mic*

An ultra-compact, one-point stereo mic, the ECM-717 is ideal for use with portable DAT and MiniDisc recorders– in table-top or clip-on applications for hands-free recording of music performances, meetings, lectures and interviews. One-point stereo recording design is like having two mics in one: Stereo unidirectional (cardioid) pickup pattern is more sensitive to sounds from the front, less sensitive to sounds from the side; rejects extraneous noise and emphasizes what you want to record.

TCD-D100 and PCM-M1 Step-Up Features

- Smallest, lightest DAT Walkmans ever
- Built-in switchable limiter prevents digital clipping of the audio signal to tape
- Mic preamps use same components as SBM-1 Super Bit Mapping Adapter
- On the PCM-M1 (only) SCMS is defeatable

DAT Walkmans and Accessories

TCD-D8 with carrying case, cleaning cassette and AC adapter.....**599.95**

TCD-D100 with two rechargeable batteries and battery charger, carrying case, cleaning cassette, AC adapter, headphones and remote control.....**Call**

PCM-M1 with two rechargeable batteries and charger, carrying case, cleaning cassette and AC adapter.....**679.95**

RMD-100K Optical/coaxial digital interface with remote control and timer.....**179.95**

SBM-1 Super-Bit Mapping Adapter **419.95**

TCD-D10 PRO II

Professional DAT Recorder

Designed for professional recording, the TCD-D10 PRO II provides XLR balanced mic/line inputs, AES/EBU digital I/O and has a 4-motor Direct Drive (DD) transport to ensure consistent and reliable performance. Ideal for field recording, it also features a pistol grip with integrated wired remote control that allows you to mount a microphone while simultaneously controlling the transport of the machine. A built-in speaker allows playback without headphones, and a comprehensive self-diagnostic system monitors vital signs for worry-free operation.



FEATURES

Professional Design

- Precise 4-motor direct drive tape transport controls the head drum, capstan and reels individually
- Dual independent analog to digital converters offer improved channel separation
- Available sampling rates are 48kHz, 44.1kHz and 32kHz for playback and digital recording and 48Khz for analog input.
- Built in mic limiter and 20dB pad enable audio capture at extremely high SPLs without causing overload or distortion.
- Multi- functional backlit LCD provides 20 segment peak level metering with overload indicators, 5 segment battery indicator, absolute time, program time, time remaining and a built in calender and clock.
- Sophisticated self diagnostics warn of tape trouble due to problems with the head drum, capstan and reels as well as transport and load/unload time. Also has a sensor warning for excessive moisture.

Inputs/Outputs

- Stereo XLR balanced mic/line inputs with independent L/R level controls variable from -60dB mic level to +4dB line level.
- Unbalanced stereo RCA line output and AES/EBU digital I/O interface.
- Monitoring is available via 1/4″ headphone output and built-in 5cm speaker

Supplied Accessories

- Includes a carrying case, shoulder belt, two NP-22H rechargeable batteries, charger, AC adapter, digital I/O connecting cable and RMT-D10P remote control.

	TCD-D8	TCD-D100	PCM-M1	TCD-D10
Sampling Frequency	48kHz, 44.1kHz, 32kHz	48kHz, 44.1kHz, 32kHz	48kHz, 44.1kHz, 32kHz	48kHz, 44.1kHz, 32kHz
Dynamic Range*	> 87dB	> 87dB	>87dB	>85dB
Quantization Bits	16-bit Linear	16-bit 12-bit non-linear (long time mode)	16-bit (standard); 12-bit non-linear (long time mode)	16-bit Linear
Signal To Noise Ratio*	> 87dB	>87dB	>87dB	>85dB
Total Harmonic Distortion**	< 0.008%	<0.008%	<0.008%	<0.006%
Wow and Flutter	Below Measurable Limits			
Dimensions (WHD)	12¼ x 1½ x 3½″	1⅛ x 3⅛ x 4⅝″	3.15 x 4.6 x 1.15″	10 x 2.16 x 7.5″
Weight	1 lb. 15 oz.	13⅛ oz.	10.2 oz.	4 lbs. 7 oz.

@ 1 kHz IHF-A 22 kHz LPF Line in* * (@ 1 kHz 22) kHz LPF Line In*

SBM-1 *Super Bit Mapping Adapter*

An all-in-one package for superior analog and digital recording, the SBM-1 is designed for Sony's DAT and MiniDisc Walkman line of portable digital recorders. 20-bit analog to digital converters ensures more natural sound reproduction with lower noise than standard 16-bit A-D converters.



- 20-bit quantization, supporting sample rates of 48kHz, 44.1kHz and 32kHz
- Manual record level for each channel
- Line inputs via 2 x phono (RCA) jacks
- Mic inputs via stereo mini jack with plug-in power and 2 x 1/4″ phone jacks
- Connects to any compatible digital interface via standard optical and coaxial S/PDIF I/O
- Compatible with the entire line of DAT Walkman and MiniDisc recorders
- Supplied accessories include; carrying case and AC adapter

ECO CHARGE

EC-50/EC-90

6- and 12-volt Lead-Acid Battery Systems for Portable Recorders

The 6-volt EC-50 and 12-volt EC-90 are the first sealed lead acid batteries with a mind of their own. Their advanced design offers a host of features to provide a new level of reliability, flexibility and convenience. They provide two 4-pin XLR output connectors so you can power two recorders from the same battery without the hassle of Y-cables. Each output is protected with automatic short circuit protection for maximum safety. Far more reliable and easier to maintain than any nicad battery pack, there are no inherit memory effects so you never have to fully drain them before recharging. The rugged carrying case offers belt loops for situations when you want to wear the battery.



FEATURES

- Reliable and affordable Sealed Lead Acid technology with no memory effects—less hassle and maintenance
- Dual XLR output connectors for powering two devices from a single cell
- Confidence is achieved with the built in LED battery level meter. At the push of a button, the amount of power remaining is indicated with a 4-position LED meter.
- Both the EC-50 and EC-90 use industry-standard Neutrik XLR connectors to provide the ultimate reliability and convenience. Eco Systems offers a selection of power cables to connect with today's most popular devices.
- They incorporate an internal smart charger with global (110- 230v) power supply allowing them to cost substantially less without sacrificing quality.
- They are have features to maximize battery life. The smart charger design monitors battery requirements and automatically adjusts to provide the optimum charge cycle. In addition, the intelligent circuitry monitors battery voltage and automatically shuts off power when it's time to recharge.
- For portability and protection, each battery includes a Cordura heavy duty protective case with belt loop and shoulder strap.
- The 5.5 lb. EC-90 is available in 3 different voltage combinations:
  - ➢ Dual 12-volt; 12 and 7.5-volt; 12 and 6-volts
  - ➢ The 3 lb. EC-50 is available in dual 6-volt or 6- and 4.5-volt configurations

Optional Power Cables for the EC-50/EC-90

Available for: Apogee MINI-ME; LunaTEC V3; Tascam DA-P1; Sony TCD-D8/D100 and PCM-M1; E.A.A. mic preamp; 6v XLR, 12v XLR

PORTA BRACE *Audio Recorder Cases*



The custom-tailored, foam-padded Audio Recorder Case is the ideal case for professional portable recording equipment. The flaps, clear vinyl windows and protective coverings provide access to cassettes, batteries, cables, dials and switches. The RM-Multi case is the second component to help organize the most essential accessories. It holds spare cassettes, batteries, wireless microphones, and other small items. A comfortable brown suede shoulder strap is provided.

AR-222 For Marantz PMD-201, 221, 222, 420, 430 ..... **173.95**
AR-D10 For Sony TCD-D10 Pro **173.95**
AR-PD4 For Fostex PD-4 (Special pocket no RM-Multi) **193.50**
AR-D5 For Sony TC-D5 ..... **173.95**
AR-DAP1 For Tascam DA-P1 ..... **173.95**

DA-P1

Portable Pro DAT Recorder



A full-featured portable DAT recorder, the DA-P1 is built to handle the most rigorous field applications. It has a dual direct drive motor transport for unprecedented reliability, and it offers XLR-balanced mic/line inputs with switchable 48v phantom power for use with condenser mics. When recording from an analog source, sampling rates of 44.1 or 48 kHz can be selected. Coaxial S/PDIF digital I/O allows you to output directly to compatible digital equipment, while the digital input will automatically adjust to sample rates of 32, 44.1 and 48 kHz.

- Rotary 2 head configuration and dual motor transport for reliable operation.
- Built in a well-constructed hard shell, that makes ideal for field use but also at home in any studio.
- A-D converters allow analog recording with selectable sampling rates of 48kHz and 44.1kHz. Digital recording rates of 32, 44.1 and 48 kHz are also supported.
- Independent L/R level controls
- A switchable mic limiter and 20dB pad ensures audio capture without distortion.
- Easy to read backlit LCD displays input and playback levels, ABS (Absolute Time), sampling rate, program number, ID select functions and record margin.
- A hold switch prevents accidental operation of the transport functions.
- SCMS (Serial Copy Management System) free recording, allowing unlimited digital tape cloning.
- Includes 2-hour, long-life life rechargeable battery (charges in 30 minutes) as well as a AC adapter and shoulder belt.

Inputs/Ouputs

- XLR balanced mic/line inputs accepts a broad range signals from -60dB to +4dB.
- RCA unbalanced line inputs and outputs
- Coaxial S/PDIF digital input and ouput
- 1/4″ headphone jack with level controls

Optional Accessories

BP-D1 Rechargeable Battery.....**84.95**
CB-D1 Battery Charger.....**69.95**
CS-D1 Carrying Case.....**109.95**

FOSTEX PD-4M

Professional Portable Timecode DAT Recorder



Used in award-winning movies and television shows throughout the world, the PD-4M is a complete, all-in-one portable field recorder that meets all the requirements of the demanding professional on location. Rivaling a studio production DAT recorder in quality and capability, the PD-4M features three XLR-balanced mic/line inputs, confidence monitoring via the four head transport, a unique "anti-jam" transport mechanism, variable frame rates and sampling frequencies including 48.048kHz for digital video. There is also full timecode recording, on-board three channel mixer with mic attenuation, phantom power and a headphone matrix. Weighing less than 7 lbs. the PD-4M is designed for the most demanding field work—whether in film, video, live music or ENG applications.

- Rotating 4-head drum with off-tape confidence monitoring enables you to hear what is being recorded directly off of the tape not from the input signal.
- 3 into 2 XLR-balanced mic/line mixer with independent gain control, -20 dB attenuation pads, 3-position pan switches, 48v phantom power and high pass filter variable between 20Hz to 250Hz for each channel and a master output control.
- Sampling frequencies are selectable between 44.1kHz, 48kHz and 48.048kHz
- A switchable stereo limiter prevents transient peaks from overloading tape.
- Choice of monitoring using 1/4″ headphone output or the built-in speaker
- Edit date in Rec/Pause or Stop Mode
- XLR-balanced AES/EBU and S/PDIF digital I/O
- Operates for two hours on a single NP-1B battery. Can also be powered by a power belt or optional AD-15 AC adapter.

Timecode

- Timecode reader/generator supports all frame rates including 24, 25, 29.97 DF, 29.97, 30 DF and 30 frames per second.
- TC modes including external run, free run, record run and 24 hour clock.
- Jam Sync is selectable for accepting external references and will output external timecode for maximum flexibility.
- Reel number editing is available in 24 hr Run Mode

SUPERSCOPE

PSD-220/PSD-230

Portable CD Players



Portable, professional-quality CD players designed especially for the performing arts, the PSD-220 and PSD-230 are ideal for music and dance practice, with controls so intuitive that even beginners can operate them. Both are capable of increasing or decreasing the tempo of a song on CD without affecting the pitch, while the PSD230 can also change the key without affecting the tempo. Add to that the ability to loop a segment of music infinitely for repeated listening and practicing, and you have what could be considered the definitive practice tool. Use them for transposing, mastering a solo, rehearsing a vocal harmony, transcribing lyrics—even choreographing a dance step or aerobics.

FEATURES

- Simple button controls let you quickly increase/decrease tempo by 50% in 1 % increments without changing the key (musical pitch). Slow down a music track as a valuable practice aid.
- 10-second anti-shock buffer to prevent skips and jumps in the audio when the unit is moved or shaken while playing.
- Read and playback finalized CD-RW discs.
- 3-band EQ with bass, mid, and treble controls
- Single Track Play feature plays one track at a time without having to create a custom play program. While in this mode, the players will cue a selected track at the beginning of audio (cue to music), rather than at the true start of the track.
- A-B Repeat provides even more control during practice by allowing a segment of a track to be selected and continually replayed (looped) until mastered. Simply select a start and end point with the touch of a button. Makes no difference if you are working on smaller parts of a song or solo, whether a two-bar phrase or an A section
- Both offer analog stereo line output (RCA) as well as a digital coaxial (RCA) output
- Built-in speaker allows you to listen or practice with a CD without having to connect to external speakers. They also have a 1/4″ headphone jack if private listening is desired.
- Three-way power capability includes the supplied AC adapter, three 'D'-size alkaline batteries or the optional RB-430 nicad battery. Both have a battery life indicator.

PSD230 Step-up Features

- Change the musical key ±1 octave in 12.5 cent steps without changing tempo. This makes it easy to play or sing along with a popular song in the key of one's choice.
- Tuning control on the PSD230 allows an audio CD to be tuned ±2% in 0.1% increments to your musical instrument—rather than tuning the instrument to the CD player.
- The PSD230 includes a special processor that can reduce the volume level of the lead vocal track on a CD. The voice reduction system allows a practicing musician to better hear the instrumentals or a vocalist to sing along as the new lead.





- Mic/line input (1/4-inch or XLR) input allows a microphone or music instrument (line level) source to be mixed together with the CD sound for practice purposes.
- Mix/Split switch determines how a signal from the Mic/Line input is mixed with the sound from the CD. The MIX position will mix (combine) the input signal with both the left and right channels of the CD playing back. The SPLIT position will separate the input signal and the CD playing back by sending the input signal to the right channel and both channels of the CD to the left channel.
- Built-in CD-TEXT displays information regarding the disc title, artist name, etc.,

Optional PSD220/PSD230 Accessories

FT-200 Play/Pause Foot Pedal—
used to start and stop the CD for music practice or transcription purposes. Offers easy control when your hands are occupied with an instrument **19.95**

MIC300 Wired Handheld Microphone .... **Call**

CA200 Rugged Carry Case—
With storage for adapter, cables & CDs ... **89.95**

RC300 Miniature IR Remote with Tempo Control—
Includes wristband, necklace and clip .. **69.95**

RB430 Nicad Rechargeable Battery ..... **Call**

PSD-300

Dual-Drive CD Recorder + CD-R/RW Player for the Performing Arts



The PSD-300 is the world's first CD recording system for the performing arts that combines a professional CD-R/RW recorder with a CD-R/RW player featuring innovative CD manipulation controls. This portable stand-alone system is designed specifically as a valuable portable music rehearsal and performance tool. In addition to its recording drive, the unit's CD player features the same performing arts controls found on the PSD230 portable CD player. These special controls allow musicians to practice with their favorite artist or a specialized music accompaniment CD and manipulate the key or tempo of that music on the fly.

FEATURES

- The PSD300 can adjust the tempo of any music CD as much as -33% to +50% without changing the music's original key (unlike standard pitch control found on some tape and CD machines).

– It can instantly transpose the music's key in musical half step increments or micro tunings without affecting tempo.

–It can significantly reduce a lead vocal from a song at the touch of a button.

–The unit can also simultaneously change key and tempo.

–Portions of music can be easily slowed down and programmed to continuously loop until a riff has been mastered.

- This means that using only the PSD300, musicians can play their instruments or sing along to music on CD, manipulate that music dramatically in real-time, and then record their live performance blended with the prerecorded music to blank CD-R or CD-RW discs. Superscope has made it easy for musicians to record ideas and practice sessions and immediately play them back to hear how they sound.
- To record, simply place the PSD300 on a table-top, plug it in and record live - with programmable mic/line EQ, noise filtering, and manual or automatic level control--using either the internal microphone or external microphones that attach to stereo XLR or 1/4˝ mic/line inputs.
- No external mixer or other equipment is needed. The recordings can be played back on any CD player that reads finalized CD-R or CD-RW data or music discs.

Applications

- For music transcription, the PSD300 can convert a music CD to a half-speed CD-R, CD-RW copy, essentially slowing tempo by 50% and lowering every note by exactly one octave.
- The PSD300 also functions as a 2x speed stand-alone duplicator, for making perfect digital copies of any disc, including CD-R/RW discs recorded on the machine itself. A music teacher, for example, might record an ensemble's practice session and provide CDs for each student to take home.
- Plug mics in for stereo recording without the need for a separate mic preamp and mixer. Or use the built-in microphone.
- Play any music CD and manipulate the key and tempo, create practice loops, or reduce lead vocal tracks all on the fly.
- Plug in instruments or mics and mix a live musicians with music CD accompaniment (taking advantage of innovative CD manipulation controls). Record the mix to blank music or computer CD-R/RW discs. Track titles can be input using CD-Text.

	PSD-220	PSD-230	PSD-300
CD-R/RW Playback/Recording	Yes/No	Yes/No	Yes/Yes
Shock Protection (10 sec.)	Yes	Yes	Yes
Tempo Control (1% increments)	+50%, -33%	+50%, -33%	+50%, -33%
Key Control		Yes	Yes
Tuning Control (0.1% steps)		Yes	Yes
Voice Reduction		Yes	Yes
3-Band EQ	Yes	Yes	Yes
Single Track Play	Yes	Yes	Yes
Cue-to-Music	Yes	Yes	Yes
Repeat 1, ALL, A-B Loop	Yes	Yes	Yes
Tempo/Key Lock	Yes	Yes	Yes
CD-TEXT Display		Yes	Yes
Display Backlight	Yes	Yes	Yes
RCA Line OUT	Yes	Yes	Yes
RCA Digital OUT	Yes	Yes	Yes
RCA Digital IN (SPDIF)			Yes
Instrument/Mic IN with Mixing		Yes	2 inputs
1/4˝ Headphone Out/Built-In Speaker	Yes	Yes	Yes
Foot Pedal IN (1/4˝)	Yes	Yes	Yes
Built-In IR Receiver	Yes	Yes	Yes

B&H
PHOTO - VIDEO - PRO AUDIO

The Professional Audio SourceBook

Section 4a

Players & Recorders

Desktop & Rackmount 2-Track

Alesis **435**

HHB **436**

Superscope **437**

Marantz **438-446**

Philips **447**

Sony **448-455**

Fostex **456-461**

Denon **462-469**

Tascam **470-483**

Telex **484-489**

Echo Star **490**

Microboards **490-493**

MASTERLINK ML-9600

High-Resolution Master Disc Recorder



The standard for high-resolution two-track audio, the MasterLink ML-9600 is a standalone system that integrates hard disk recording, digital signal processing, audio editing and CD creation/playback. By bringing together all these capabilities, MasterLink provides an affordable, all-in-one solution for high-resolution mixdown, mastering and CD-burning. Fully compatible with todays CD and digital audio formats plus offering the sonic quality you'll need for tomorrow, the MasterLink ML-9600 is a uniquely versatile and affordable solution for everyone from large commercial audio facilities to project studios and recording musicians.

Head and shoulders above any CD burner, MasterLink lets you capture mixes to the internal hard-disk recorder at up to 24-bit/96 kHz resolution via analog (XLR balanced, RCA unbalanced) or digital (AES/EBU and S/PDIF coaxial) inputs. Once tracks are on the hard drive (up to 25 hours of stereo 16-bit/44.1 kHz audio can be stored), they can be assembled into playlists. There is room for 16 playlists, each of which can contain up to 99 tracks.

Once in a playlist, a track can be edited and processed in preparation for burning to CD. When you've processed and edited the tracks in the playlist to your liking, simply insert a blank CD-R and then hit the Create CD button. The playlist and the tracks are written to a special area of the internal hard drive as a CD image, then the image is burned to the CD in either the Red Book standard or at up to 4-bit, 96 kHz resolution using Alesis' revolutionary CD24-AIFF technology.

FEATURES

Hard Disk Recording

- Most everything begins when you record two-track audio to the high-capacity 20GB internal IDE hard drive. There is a full complement of connections for bringing in analog or digital audio from your mixer, DAW and other sources. You can even the use the integrated CD player as an audio source.
- During hard disk recording, choose from four sampling rates(44.1, 48, 88.2 or 96 kHz) and three word lengths (16-, 20- and 24-bit). For mixdown, delivery and archiving, choose 24-bit/96kHz as the primary format.
- State-of-the-art 24-bit digital converters offer better sonic performance than any other affordable mixdown format. For creating standard audio CDs, MasterLink's sample rate conversion and noise shaping technology is specially designed to keep your audio sounding its best at every resolution.

Track Editing and DSP Finishing Tools

- After recording a track on hard disk and assigning it a name, you can set precise start and end points to each each song using MasterLink's cropping features.
- Built-in mastering toolbox includes non-destructive DSP capabilities: real-time compression, equalization, limiting and normalization. These functions can be adjusted and combined, allowing you to optimize the timbre and dynamics of each individual song, as well as establishing a professional sonic consistency from track to track.

Multiple Custom Playlists

- With MasterLink's multiple playlist capability, you can create, name and edit 16 different playlists containing up to 99 songs in each. Each playlist offers full control of song order, track gain, fade-ins and fade-outs, and length of time between tracks. With MasterLink you can store 16 CD masters worth of music with polished, professional quality and presentation.

CD-24

- After fine-tuning your playlist on the MasterLink hard drive, CD creation is easy. A one-button command records your master onto CD-R using the 16-bit/44.1kHz Red Book standard (compatible with CD players worldwide) or Alesis' 24-bit/96kHz CD-24 format (perfect for high-resolution archiving ro delivering mixes to mastering studios. The high-resolution discs can also be accessed by Mac and Windows computers. Since CD-24 uses the ISO 9660 CD-ROM disc format and the standard AIFF audio file format, the discs can be used by digital audio workstations at professional facilities worldwide.

CDR830/CDR830 PLUS

'BurnIT' CD-R/CD-RW Recorders



Housed in a durable, 19˝ rack-mounting chassis, the CDR830 BurnIT and BurnIT PLUS feature very high-quality 24-bit A/D and D/A converters and an advanced laser assembly that ensures consistently accurate recordings compatible with a wide range of CD players.

The CD-Text facility, enables disc, artist and track names to be stored and displayed, and an SCMS-free digital input and built-in sample rate converter making synchro recording easy from CD, DAT, MD or hard disc. A unique facility for recording direct from dynamic microphones removes the need for additional pre-amplification equipment thereby increasing the recorder's versatility and range of applications. The addition of a digital record gain control allows level adjustment of digital input signals. The CDR830 BurnIT PLUS steps up with balanced XLR analog inputs and outputs with line/mic input gain switching, balanced XLR digital input and output, word clock input (enabling sync playback at 44.1k or 48k sample rates) and parallel remote input – a range of connectivity that will satisfy the needs of most analog and digital recording environments.

FEATURES

- Precise design of the CD laser assembly means that every single CD-R and CD-RW disc is exceptionally accurate, providing greater compatibility with other CD players and recorders.
- Five CD-RW erase modes for flexibility when compiling recordings. These include erasing individual tracks, several tracks, all tracks, the table of contents or entire disc.
- Built-in SRC (sampling rate converter) accepts signals from 32kHz to 48kHz so you can record from other digital sources.
- Three digital and analog synchro recording modes for recording and finalizing CDs : One/All Tracks and All Tracks with Finalize.
- Left/right digital balance control enables the balance of a CD, DAT, MD or any other source, digital or analog sources to be adjusted before they are recorded.
- CD Text display input and editing enables you to store disc, track and artist names with every recording. Entries can be up to 120 characters in length, with the title scrolling if it is too long to fit in the display.
- Fade in and out times can be adjusted from 1 to 12 seconds
- Digital balance control lets you control the left/right balance of digital or analog sources before they are recorded.
- IDs can be triggered on both analog and digital recordings by adjusting the auto track increment level threshold from -24 to -78dB in 6dB steps.
- Signal levels can be monitored during input, recording or playback using the LCD meters
- SCMS-free digital input lets you record from any CD, DAT, MD or hard disk. Plus, you can set the SCMS status when burning CDs (Copy Prohibit, Copy Once, Permit).
- Both the coaxial and optical digital inputs have their own digital gain control
- Digital input gain control can be adjusted using one of the five available scaled values:

+12dB to -12dB in 0.5dB steps
+20dB to +12dB in 1dB steps
-12dB to -24dB in 1dB steps
-24dB to -48dB in 3dB steps
-48dB to -84dB in 6dB steps)

Warning! Not all discs are the same

HHB produces a wide range of high performance CD recording media, designed specifically for professional audio use.

CDR74 Gold: 74 minute audio-optimized CD-R with gold reflective layer. Secure archival life in excess of 100 years.

CDR74 Silver P: 74 minute CD-R with printable surface for use with inkjet printers

CDRW74: High performance 74 minute rewritable disc. 1000 erase/rewrite cycles

CDR74 Silver: 74 minute audio-optimized CD-R with silver reflective layer. Secure archival life in excess of 100 years.

CDR80 Silver: Orange Book compatible 80 minute CD-R with an archival life of 200 years

CDRW80: 80 minute rewritable disc with exceptional direct over write performance

CDR80 Silver Bulk: Bulk-packaged 80 minute CD-R discs. Orange Book compatible. Suitable for use with thermal printers

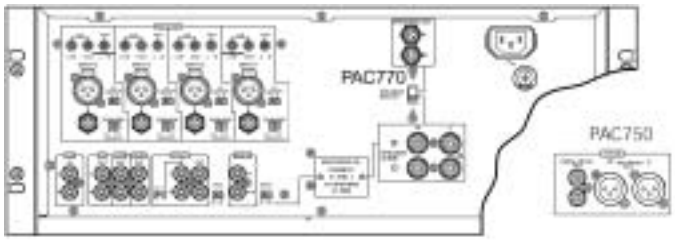
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PAC-750/PAC-770

Professional Integrated Sound Systems

The 45 lbs. PAC770 (built-in 110-watt stereo amp) and 27 lbs. PAC750 (pre-amplifier only) provide a remarkably easy to use, feature-rich, and cost-effective installation solution for applications requiring music and public address. In one durable, commercial-grade product taking up just 4U of rack space) the PAC770 combines four quality system components: CD player with music playback features specially designed for the performing arts, a full-featured cassette recorder, an elegant mixer that any novice can handle with ease, and a powerful and reliable amplifier. Ideal for small-to-medium sized venues like houses of worship, schools and meeting rooms, they are also portable on an A/V cart or in an optional case.





FEATURES

CD Player

- Playing speed can be adjusted from -33% to +50% without changing the key (Tempo Control). This is done in 50 intervals (1% from previous tempo setting).
- The key of a CD can be altered by ±1 octave, in 1/16th note increments (Key Control) without changing tempo.
- Audio loops can be defined within a CD track utilizing the A-B button.
- Single or multiple tracks can be programmed for custom repeat playback.

Inputs/Outputs

- 4 mic/line inputs (XLR=1/4″)

- 48v phantom power
- Pan control and 2-band EQ per channel

- Aux Line In (RCA x2)
- CD Line Out (RCA x2)
- Tape Line Out (RCA x2)
- Master Effect In/Out (RCA x2)
- Unbalanced Line Pre-Out (RCA x2)
- Balanced Line Pre-Out (XLR x2) PAC750
- Binding Post Speaker Out
- 1/4″ Headphone Out
- RC5 Remote In/Out (AMX/Crestron)

7-Channel Mixer

- Simultaneous control of 4 XLR mic/line inputs, tape, CD, and stereo aux inputs.
- The amount of signal each input sends to the mixer can be easily adjusted using each input's gain control knob. Easier still, the auto gain feature will set an optimal level for the mic/line inputs.
- Master output volume, EQ, and balance are simply controlled from the front panel, while individual mic/line EQ, panning, phantom power, and input attenuation are designed to be "set-once" on the back panel.
- Parametric EQ and speaker processing can looped through the unit.
- Voice reduction circuit can reduce the lead vocals from CD, tape and aux sources.
- For paging applications, the unit can be set to automatically duck music when any mic/line input is active, either eliminating levels or reducing them by -15dB.

Cassette Recorder

- Quick Auto Reverse –optical sensing of leader reverses the direction of tape transport, for seamless double-sided recording.
- ±12% pitch (tempo + key) control
- Dolby B/C/HX Pro noise eduction
- Manual and automatic (ALC) record levels
- Memory point; music search; normal speed dubbing

Amplifier or Preamp Only

- The PAC750 is built with a preamplifier only, enabling flexible installation. The PAC770, however, comes with a high performance amplifier. It includes:
- Stereo or Mono Bridge Modes – The unit allows stereo or monaural amplification.
- Power Protection Circuit – The amplifier is designed to protect itself from running at levels that can cause damage.

Optional Accessories

RC5PMDSW Infrared Remote Kit (includes handheld IR remote, IR receiving eye). Can control tempo remotely.....**Call**

WRC200 MKII Wired Remote Control.....**Call**

RC300 Miniature Wearable IR Remote (includes a wristband, necklace and clip).....**69.95**

MIC300 Wired Handheld Mic .....**Call**

MARANTZ

PMD-501/PMD-502

Single-Well Cassette Decks



Versatile and reliable, both the entry level PMD-501 and the full featured PMD-502 are workhorse cassette decks, ready for demanding everyday use in commercial recording and broadcast facilities, project studios, houses of worship and A/V rental houses. The easy to use PMD-501 has a host of intelligent features that are found across the Marantz cassette deck line including: optically sensing quick auto reverse, Dolby B noise reduction, remote capabilities, auto tape type select and more.
The PMD-502 adds Dolby "C" noise reduction with HX Pro headroom extension, optional XLR balancing kit, ±15% pitch control, defeatable automatic level control and front panel microphone/auxiliary line level inputs. Of course they both offer the world-class sonic integrity that Marantz products are famous for.

Marantz's line of professional single and dual well cassette decks combine thoughtful engineering with uncompromising audio quality. From their entry level single well PMD-501 to the top of the line PMD-520 dual deck, they offer problem solving solutions for the most demanding mastering, dubbing and installation systems. All are housed in a 3RU high chassis and are easy to use.

- Optically sensing quick auto reverse detects leader at the end of a tape and automatically flips the tape with a minimum of audio loss or, in the case of a dual deck, will activate the second deck when the first has completed recording on both sides.
- Automatic tape type selector detects the type of tape formulation such as CrO2 or normal bias cassettes and adjusts the bias for optimum recording performance.
- Dolby B noise reduction
- Large transport control buttons with LEDs
- Recessed record balance control
- Marantz CD player equipped with RC-5 interface can control the start and stop of a PMD-series cassette deck placed in the REC/Pause mode (CD synchro recording).
- One-piece 3RU high chassis

Inputs/Outputs

- Unbalanced RCA inputs and outputs
- Headphone output with level control
- Marantz RC-5 remote control I/O. This remote interface also allows seamless integration of any PMD cassette deck with an AMX or Crestron control system.
- Optional wired and wireless remotes allow simultaneous control of multiple units for synchronized dubbing and other remote capabilities.

PMD-502 Only

- Output level control
- Pitch control (±15%)
- Front panel microphone/auxiliary inputs
- Digital linear time-based counter
- Memory rewind
- Front panel input select switch
- Dolby C and HX Pro
- Fluorescent peak holding bargraph meters
- Defeatable automatic level control (ALC) overrides the manual input level control and sets the input to its optimum level.
- MPX FM filter eliminates the 19kHz pilot tone associated with FM radio signals — essential for proper Dolby B encoding.
- Optional XLR502PMD balanced-XLR kit

Accessories for PMD-Series Cassette Decks

RC-5
Infrared Remote Control

- RC-5 can control multiple cassette decks simultaneously using RCA cables connected serially in between units. Ideal for realtime duplication systems.
- Two mounting options include a 1 unit rack panel and a single gang wall plate.
- Infrared receiver can be mounted up to 30 feet away; ideal for situations where decks need to be heard and not seen such as boardrooms and houses of worship.
- Includes handheld infrared remote controller, infrared receiver and AC adapter (RCA connection cable optional).

WRC
Wired Remote Control

- Hard wired rugged table top remote control for PMD series recorders and players.
- Includes 6 foot RCA to RCA patch cable to interface with RC-5 interfaces.
- Yellow and green LED indicates current control over CD or tape. Ideal for duplication systems with decks serially connected via RC-5.

XLR Balancing Kits

- Add +4dB balanced XLR inputs and outputs to the PMD-series cassette decks.
- Kits consist of rear panel assembly and ribbon cables for solderless installation.

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ONE HOUR FREE PARKING
AT 349 W. 34th STREET (with purchase of $100 or more)

PMD-510

Independent Dual-Well Cassette Deck



The PMD-510 is one of the most comprehensive and flexible dual well cassette decks ever made for professional use. Each of the two wells is a completely independent cassette deck (simultaneous playback on well A while recording on well B or vice versa), with their own distinct set of stereo inputs and outputs, pitch control, stereo bargraph meters, linear time counter, Dolby B/C selector, and large transport buttons. In addition to independent operation, the two wells can work together to record from one input.
An extension remote port allows the connection of several PMD-510s for long term continuous recording and playback. All auto-reverse, well-switching, and cascade operations are optically sensing for minimal audio loss in continuous record and play modes. The PMD-510 also features a headphone output that is switchable between well A, well-B, or both wells. The remote port is compatible with Marantz (RC-5), AMX, and Crestron control systems.

Same features as the PMD-502 (except no ALC) PLUS—

Dual Deck Operation

- Each deck features independent large transport buttons with LEDs. Each deck also has level, balance, pitch and noise reduction controls as well as independent LED bargraph meters and linear (minutes and seconds) time counters.
- Perform synchronized dubbing from deck to deck at normal and high speed (2x). Normal and high speed programmed dubbing of up to 16 tracks in any order is also possible.
- Pressing Play button twice on either well sets the current track to loop continuously.
- Up to 15 tracks can be programmed to play back in any order.
- Continuous (serial) play or recording from deck A to deck B or deck B to deck A.
- Simultaneously record to both decks from an external source (parallel recording)
- Independent inputs allow recording on each deck from different sources.
- Link several PMD-510s together in series for continuous playback or recording using the cascade (EXT) connectors.
- All auto reverse functions for well switching and cascade operations are optically sensing for a minimum of audio loss during record and playback modes.
- You can record on one well while playing back on the other.
- Independent outputs allow simultaneous playback of a tapes in both deck A and B.
- QMS (Quick Music Search) will fast wind to the beginning of the next or previous track. QMS can advance up to 15 tracks.
- AMS (Auto Music Scan) mode fast winds and plays the first ten seconds of the next track.
- While recording, Quick Rewind function automatically returns the tape to its initial record position and enter the stop mode.
- Optional XLR510PMD balanced-XLR kit

PMD-505



Marantz's most economical dual-well cassette recorder, the PMD-505 features auto play/record on power up, normal and high speed dubbing, auto tape select, record mute with auto- pause, music search (QMS), Dolby B/C/HX Pro noise reduction, record balance control, ±10% pitch control (compared to ±15%) on the other PMD-500 series decks) and timer record/play function.
It is equipped with RCA line input/output, RC5 remote input/output, headphone output (without level controls) and unlike the other Marantz dual-well cassette decks, it doesn't offer an optional balanced-XLR kit.

MARANTZ

PMD-520

Independent Dual Well 3-Head Cassette Deck



The top-of-the-line PMD-520 has all the capabilities of the PMD-510 plus adds three uniquely distinct features. It has three stationary heads allowing direct off-tape (confidence) monitoring in realtime. This is useful for tracking master recordings and live audio capture and is an efficient way of spot checking tapes during extended duplication runs. The next two features are unique for any deck. Direct tape insertion/removal eliminates tape doors for easier access to cassettes while the tape mechanisms can be removed allowing servicing of installed decks with a minimum of fuss. The cascade recording feature is also taken to the next level by allowing a three minute overlap when recording from one well to another. The loop through outputs are perfect for cascading multiple decks for duplicating or serial recording. Also offers a 25-pin connector for customized remote control in addition to the Marantz RC-5 interface.

Same features as the PMD-502 PLUS—

- Stationary 3-head mechanisms in each deck provides confidence monitoring. The monitor button switches between tape and source (input) for comparison of the input versus off tape signals.
- Direct tape insertion/removal eliminates mechanical problems (no eject mechanisms or hinged doors). Makes reloading more efficient during large duplication runs.
- Innovative construction allows each tape mechanism to be completely removed for service if needed, while the other remains available for use.
- When cascading multiple decks (up to 100 units can be chained together) for duplication, the stereo Loop-through outputs carry the input signal out to the next deck in the chain, freeing up the standard outputs for monitoring (eliminates the need for a distribution amp) and ensuring that the input to each deck in the chain is unaffected by the previous decks electronics.
- To prevent any loss of audio during long format recording, an overlap series recording feature automatically engages the next deck cascaded in the chain 3 minutes before the current deck runs out of tape.
- To ensure optimum performance, a microcomputer controlled automatic tape calibration adapts the bias, record gain, mid-frequency gain, high-frequency gain and high frequency peak of the deck to the specific tape(s) being used.
- Individual bias reset button returns the bias settings to their factory default.
- Recessed, front panel ±3% fine speed control allows you to fine-tune motor speed without having to unrack the unit.
- 25-pin contact closure control port to control transport functions and bias control.
- Stereo LED meters with peak hold
- Auto rewind when the tape ends
- Tape stop alarm uses a buzzer to indicate a tape has stopped playback or recording
- Tape time button selects the length of tape being used (60/90/120/10/20/30/46 minute) ensuring that the counter display will show accurate information.
- Optional XLR520PMD balanced-XLR kit



PMD-511

Dual Well Cassette Deck

The PMD-511 is one of the most versatile and functional dual well cassette decks ever made for professional applications. In addition to all the features expected from an independent dual-well cassette deck, it includes stereo microphone inputs and a built-in recording mixer. Mics input to the PMD-511 can be mixed with an incoming line level source and then recorded onto either tape well. A front panel Mix Level control adjusts the amount of mic signal that is mixed with the line signal. The PMD-511 also features a high (2x) speed mode, allowing either high-speed duplication or ultra-high quality recording, a sophisticated ALC (Automatic record Level Control) system for worry-free recording, and 25-pin GPI port that allows contact closure control of the PMD-511.

PMD-351

CD with MP3/Cassette Combo Deck

Marantz combines the strongest attributes of their CD player and cassette recorder/reproducer technology in the 3RU PMD-351 combo deck. Designed to offer maximum flexibility for everything from mastering and dubbing applications to broadcast, installation and DJ rigs, the PMD-351 is equally competent as an independent CD player and tape deck, as it is an integrated combo deck.



CD Features

- Cue to music feature allows instant start of program material.
- A-B rehearse function loops audio between any two points in a track.
- Program play up to 20 tracks in any order
- Repeat function (all tracks on the disc or in the current program list)

Cassette Deck Features

- Optically sensing quick auto-reverse
- Defeatable automatic level control (ALC) as well as manual level control with a recessed left/right record balance control
- Dolby B, C and HX Pro noise reduction
- Memo button stores a specific point on tape allowing you to FF or REW to that point from anywhere on tape.
- Automatic tape type selector sets the bias and equalizer levels for the tape (Normal, Chrome, Metal) you are using.

Rear Panel

- Separate and mixed RCA tape and CD outputs
- Left and right 1/4″ high impedance mic inputs
- Coaxial digital output
- Multiple Marantz decks (equipped with RCA-EXT cascade connector) can be looped together for extended playback or recording. When the first deck completes both sides of a tape, the next deck starts.
- MPX Filter switch eliminates sub and ultra sonic frequencies transmitted during FM broadcasts that can interfere with Dolby noise reduction operation.
- Independent fader start jacks - when receiving a dry switch contact closure the CD will switch between Play/Pause and Play, and the tape deck between play and stop mode
- Marantz RC-5 Remote I/O allows basic remote control for a single unit or synchronized control of multiple units.

Front Panel Functions

- One-Touch Automatic Dubbing (from CD to cassette) function scans the CD to find the peak level, sets the optimum recording level using the Automatic Level (ALC) Control and commences recording.
- Multi function display indicates the operating status of the main functions for both the CD and cassette deck.
- Large transport buttons with LED inset
- Separate ±12% pitch controls for tape and CD
- Headphone output with source selector switch; CD, Tape or Mix

Additional Features

- One-piece 3U rackmount chassis
- Optional balanced XLR kit (XLR350PMD)
- Optional wired or infra-red controls
- AMX and Crestron compatible

	PMD-501/PMD-502	PMD-510	PMD-505	PMD-520	PMD-511	PMD-351
Number of Heads	2	2	2	3	2	
Head Type	Hard Metal Alloy	Hard Metal Alloy		Hard Permalloy	Hard Metal Alloy	
Number of Motors	2x DC Servo	2x DC Servo	2x DC Servo	2x DC Servo	2x DC Servo	
Wow and Flutter (JIS weighted)	0.07%	0.07%	<0.08%	<0.07%	<0.07%	0.07%*
Output Level	500mV / 2.2k Ω	500mV / 2.2k Ω	460mV / 47k Ω	540mV/47 Ω	500mV/3k Ω	500mV/1.5k Ω*
Input Level	100mV / 47k Ω	100mV / 47k Ω	100mV / 50k Ω	100mV/100k Ω	100mV / 47k Ω	100mV / 47k Ω*
Power Requirements	120V AC, 60 Hz	120V AC, 60 Hz	120V AC, 60 Hz	120V AC, 60 Hz	120V AC, 60 Hz	120V AC, 60 Hz
Power Consumption	22W	22W	0.13A	30W	35W	25W
Freq. Resp. Metal (Type IV) tape	20-18kHz (±3dB)	20-18kHz (±3dB)	20-18kHz (±3dB)	20-18kHz (±3dB)	20-18kHz (±3dB)	20-18kHz (±3dB)
Freq. Resp. CrO2 (Type II) tape	20-17kHz (±3dB)	20-17kHz (±3dB)		20-17kHz (±3dB)	20-17kHz (±3dB)	20-17kHz (±3dB)
Freq. Resp. Normal (Type I) tape	20-16kHz (±3dB)	20-16kHz (±3dB)		20-16kHz (±3dB)	20-16kHz (±3dB)	20-16kHz (±3dB)
No Noise Reduction[1]	56dB	56dB		58dB	58dB	56dB
Dolby B[1]	66dB	66dB		67dB	67dB	66dB
Dolby C[1]	72dB *(PMD502 only)*	72dB	74dB	75dB	75dB	76dB
Dimensions (H x W x D)	5.2 x 18.9 x 11.0″	5.2 x 19 x 13.2″	5⅝ x 17⅛ x 12⅝″	5.2 x 19 x 15.4″	5.2 x 18.9 x 13.2″	5.3 x 19 x 11″
Weight	11.5 lbs.	15 lbs.	8.58 lbs.	15.4 lbs.	18.1 lbs.	13.4 lbs.

[1] *Signal-to-noise ratio "A" weighted* * *Tape section*

MARANTZ

CDR-300

Portable Desktop CD Recorder with Built-in Mic Preamp



Designed for stand-alone portable recording, the CDR-300 is a professional CD recorder that offers the functionality of a professional tape recorder—while recording directly to blank CD-R/RW media. Once finalized, recorded discs will play back in virtually any CD player

To record, simply place the compact and lightweight CDR300 on a table-top, power it up via 120v AC or 12v DC power and record live using either the internal mic or external microphones that attach to stereo XLR or 1/4˝ mic/line inputs. Because it has a built-in preamp, an external mixer or mic preamp is not required. A built-in speaker or headphone jack conveniently allows recordings to be monitored. High and low EQ and high and band pass filtering can be independently adjusted for each mic/line input in the preset menu. 48v phantom power is available for condenser microphones. The CDR-300 also provides manual or automatic level control, limiter, and digital level meters.

FEATURES

- The CDR-300 records to both music and computer CD-R and CD-RW discs. Track titles can be input using CD-Text. It can make a perfect digital copy of any CD.
- Built-in mic preamp with phantom power and onboard mixing capabilities means you can plug quality microphones straight into the XLR or 1/4˝ mic/line inputs for stereo recording of live events. No need for a separate mic preamp and mixer.
- Capture ambient room sound with a pair of mics and mix in the stereo feed from a larger sound system to make a great live CD recording of any musical event.
- Record CDs from any analog or digital source including LP, DAT, cassettes, or MiniDisc.
- Internal AC power supply, plus a 4-pin DC power input that enables it to be used with external 12v battery packs.
- Portable design and easy operation enables convenient direct-to-CD recording anywhere - from churches, band and practice rooms, to classrooms, auditoriums and outdoor venues.
- For CD playing you can add (mix in) your own microphone or instrument input, connect to an amplifier/speaker system, connect to an external CD recorder, connect an additional CD player (or other line input) to the CDR-300 and mix that input
- Built-in microphone and full range speaker allows listening to a CD without having to connect to other equipment for amplification.
- Tone controls include:

–TREBLE High frequencies (6 kHz)

–MID Middle frequencies (2 kHz)

–BASS Low frequencies (150 Hz)

- 3-band master EQ
- Programmable mic/line EQ
- Programmable mic/line high and band pass filtering
- Manual or automatic level control (ALC) for recording
- On-board switchable limiter
- Stereo XLR & 1/4" mic/line inputs
- Computer CD-R/RW disc compatible
- 48V Phantom Power
- RCA line out
- Stereo RCA Aux input
- Digital S/PDIF coaxial input and output
- Includes full function IR remote control
- An optional foot pedal (FT200) can provide Play/Pause control for the CDR drive



Portable design and easy operation enables convenient direct-to-CD recording anywhere. And it is designed to be as easy to use as a portable tape recorder. Just place the CDR-300 on the table at your next meeting, record using the built-in condenser mic and automatic record level control, and capture every word with clarity. You can start and stop recording just like a tape, and you can even erase the disc or last track if you use rewritable discs.



CDR-500/CDR-631

CDR-631 *CD-R/CD-RW Recorder*



Offering superb flexibility and reliability in a rugged rack-mount design, the CDR-631 is a next-generation digital recorder that lets you create your own custom CDs from any audio source. Powerful feature set includes a rich assortment of I/Os including balanced XLR analog and AES/EBU digital for connecting to other analog and digital gear. Also offers a sample rate converter, memory buffer, complete SCMS programmability, remote control input and the ability to use a variety of CDs, including computer discs.

CDR-500 & CDR-631 Both Feature

- CD-Text function lets you save the CD title, artist and track names via the menu using simple input controls.
- For a wider choice in media with greater flexibility and additional cost savings, they accept CD-R and CD-RW discs, including 1X-speed discs for computer use.
- Sync recording mode for for fast and easy duplication of CDs, MiniDiscs, and DATs. Track increments are automatically detected, and for analog source material, a silence of 2.7 seconds is automatically detected as a track increment.
- Automatic sample rate conversion handles any source from 11 kHz to 56 kHz and converts it to the 44.1 kHz. CD standard (44.1 kHz sources bypass the converter).
- They have a memory buffer that prevents the beginning of tracks from getting cut off
- Auto Stop Control allows dynamic sources (eg. classical music) to be recorded effectively
- 2x speed disc finalization
- Program up to 99 tracks to play in any desired sequence
- Coaxial digital input includes loop-out for unprocessed connection to other digital equipment.
- RC5 remote control input for installed (AMX/Crestron compatible) controllers
- Bypass SCMS copy protection when duplicating a disc. Or select among copy protection options when recording your own discs.

CDR-500 *CD-R/CD-RW Recorder and Player*



CD-R/RW recording and playback has never been easier. The CDR500 features two independent drives: one for CD-R/RW playback and recording and a second drive for playback only. This means it can operate as a stand-alone 2X CD-R/RW duplicator; make a recording from any analog or digital source; program a playlist that selects tracks from both CD drives; play from both drives at the same time.

	CDR-631	CDR-500
CD Recorder/Player Drive	•	•
CD Player Drive		•
High Speed 2x Dubbing		•
Disc-at-Once Dubbing		•
Pro & Consumer CD-R/CD-RW discs	•	•
Audio Buffer	•	•
Programmable Copy Protection	•	•
Digital & Analog Record Level	•	•
Record Balance Control	•	•
Record Mute	•	•
Manual Track Increment	•	via remote
Auto Stop On/Off	•	•
CD-TEXT Display + Encoding	•	•
10-key Pad	•	via remote
Program Play	•	•
Repeat 1/ALL	•	via remote
Random Play	•	via remote
Rackmount Height	2U	2U
INPUTS & OUTPUTS		
RCA Line IN/OUT	•	•
XLR Line In	•	•
Coaxial Digital IN/OUT	•	•
Coaxial Digital Loop-OUT	•	•
Optical Digital In	•	•
XLR Digital IN (AES/EBU format)	•	•
Sample Rate Converter (SRC)	•	•
Headphone OUT (1/4″) with level	•	•
RC5 Remote IN/OUT	•	•
Included IR Remote	•	•
Power Consumption	19W	21W
DIMENSIONS		
Width	19.0″	19.0″
Height	3.9″	3.9″
Depth	13.4″	13.4″
Weight	10.4 lbs.	11.7 lbs.

MARANTZ

PMD-325

Industrial CD Player



The PMD-325 offers great versatility and reliability for music playback, background music, and other applications. In addition to playing CDs it will play finalized or unfinalized CD-R (write-once) and CD-RW (rewritable) discs, as well as MP3 files on CD-R/RW. And, equipped with RS-232 serial control, will revolutionize music playback for your professional A/V system. Customization options for audio playback give you full control over how the music plays—single-track, randomly, or by time without requiring a monitor or touch panel display. In addition to RS-232 and standard CD functions, other features include pitch control, direct track access, A-B loop, auto cue, CD text, XLR with adjustable trim and GPI contact closure control of many features.

FEATURES

- Built for installation, the PMD325 can receive AMX or Crestron compatible control signals via its rear panel RS-232C serial port or RC5 port.
- It also features a GPI port with fader start. The fader start trigger input enables playback and pause control from an external switch or fader from a mixing desk.
- While many DVD players play MP3 files, they require a TV monitor or touch panel display to navigate the files, and they are not designed to fully control how the music plays. The PMD325, however, provides single track, random, and timer play for both music CDs and MP3 files.
- Plays CD-RW discs without a hitch, and also display CD-TEXT for identifying CD and song titles at a glance.
- ±12% pitch control in 0.1%. increments. A Calibration button is used to calibrate and return the CD player to normal pitch.
- Single track and program play. Program up to 30 tracks for playback in any order while a time display calculates the total accumulated program time of the selected programs.
- Remote control with 10-key pad (0 - 9) and rotary knob for direct track access and selecting tracks in program mode.
- Auto cue can be set to an adjustable threshold
- Time button switches the displayed time between elapsed time and time remaining of current track, total length of disc and time remaining on disc.
- Three repeat modes include 1-track, all-tracks, A-B repeat
- Equipped with unbalanced RCA and balanced-XLR analog outputs with level control, and XLR, optical and coaxial digital S/PDIF outputs, enabling easy integration into professional A/V systems.
- One piece 2U rackmount chassis

With four models to choose from, the Marantz family of single-well professional CD players provides a solution for every need – from the demanding broadcast studio environments where dependability and precise control are key, to corporate installations, theaters, houses of worship, and other applications that require a CD player with professional features and ease of use.





PMD-331/340

Industrial CD Players

Marantz audio products have long been recognized for having exceptional sound quality. And the PMD-331 and PMD-340 are no exception. Adhering to Marantz's rigorous audio standards, they are not only built with the highest quality D/A converters, but they employ state-of-the-art digital and analog filters and noise sampling technology to maintain pure signal clarity. It's a difference you can hear.

PMD-331/340

PMD-331

All the features of the PMD-325 except no MP3 playback, RS-232 serial port or infra-red remote control

- Up to 25 different functions (presets) can be individually programmed using the intuitive front panel control. Items include end of track warning, tray lock, fade-in/out time, default time display, disc start mode and end monitor time.
- CD mechanism is designed to handle rackmount installations at up to a 45° angle.
- 10-second anti-shock buffer
- Jog wheel enables especially quick pitch change as well as faster searches.
- In addition to ±12% pitch in 0.1% increments, pitch bend function allows quick ±8% adjustments in pitch.
- Outstanding cue point handling gives you the ability to search within a track for the exact point for playback to start. Precise manual cue points are set using the FF and REW buttons to advance the track one audio frame at a time.
- Frame-accurate searching allows you to hear CD frame as an audible signal, and zero in on a recognizable note of music or word syllable.
- High contrast LCD displays information at a glance, such as time remaining (shown graphically or numerically), end of track monitor, track pitch, scrolling CD-TEXT display, and current play mode.
- 25-pin D-sub GPI Port

PMD-340

Step-up Features

- Has the same features and functions as the PMD-331, however, the PMD-340 is specially designed to hold up in the most demanding environments such as broadcast, recording/duplication studios and mastering facilities.
- One of the most durable and reliable professional CD players available, the PMD-340's extremely durable and reliable CD mechanism includes a die-cast frame, a highly durable, brushless spindle motor, and a high-resolution, heavy-duty precision laser for outstanding tracking.

	PMD-320	PMD-321	PMD-330	PMD-331	PMD-340
High-Performance Drive			Yes	Yes	Yes
Heavy-Duty Mechanism					Yes
CD-RW Playback			Yes	Yes	Yes
Shock Protection (10 sec.)				Yes	Yes
CD-TEXT Display			Yes	Yes	Yes
Instant Start				Yes	Yes
AutoCue		Fixed level	Adjustable	Adjustable	Adjustable
Fade-In/Fade-Out			Yes	Yes	Yes
Index Search			Yes	Yes	Yes
Pitch Control (±12%)	Button	Button	Button	Jog Wheel	Jog Wheel
Pitch Bend (±8%)				Yes	Yes
Cue Point			Yes	Yes	Yes
10-Key Pad	Yes	Yes	Yes	Yes	Yes
Single Track Play			Yes	Yes	Yes
Program Play	Yes	Yes	Yes	Yes	Yes
Repeat 1, ALL	Yes	Yes	Yes	Yes	Yes
A-B Repeat			Yes	Yes	Yes
Random Play			Yes	Yes	Yes
Programmable Presets			Yes	Yes	Yes
Rackmount Height	2U	2U	2U	2U	2U
RCA Line Out	Yes		Yes	Yes	Yes
XLR Line out		Yes		Yes	Yes
Coaxial Digital Out	Yes	Yes	Yes	Yes	Yes
Optical Digital Out				Yes	Yes
XLR Digital Out (SPDIF)		Yes		Yes	Yes
Headphone OUT (1/4″)	Yes	Yes	Yes	Yes	Yes

PMD-371

5-Disc Rotary CD Changer



The PMD371 can receive AMX or Creston compatible control signals, and play up to six hours of uninterrupted music on 5 discs, during which 3 discs can be changed. It also feature easy programming, the ability to edit a play list for fitting tape length, Introdisc scan (play 10 seconds of every song) and a custom rackmount kit for smooth installation integration.

- Full programming and program editing, random play, single or multiple track repeat
- Change discs while playing
- Audible Cue and Review
- Analog RCA outputs
- Marantz RC-5 Remote Input/Output AMX and Crestron Compatible
- Removable 3U rackmount handles

MARANTZ

PMD-910/PMD-930

Professional DVD Players



PMD-930

The PMD-910 features 3:2 pulldown progressive scan, Dolby Digital and DTS compatibility, and is both NTSC and PAL compatible. In short, it's ideal for corporate and institutional users who demand high definition picture quality and exceptional sound. One rack space installation is possible only with an 1/8˝ bottom clearance on mid-section. Otherwise two rack spaces are needed. The PMD-930 is a global-compatible DVD Player with an RS-232 port offering maximum flexibility. It accepts all DVD discs, including DVD-RW and DVD+RW discs that are rapidly becoming the preferred media for edited video. So DVDs and Video-CDs created by a corporate office or university in another part of the world will play back problem-free in the PMD-930. It also plays MP3 audio files on CD-R and CD-RW discs. Moreover, it's operating system can be easily flash upgraded via CD-ROM in the DVD drive to accept future DVD formats.

PMD-910



PMD-910

- The PMD-910 will display a range of formats, from DVDs, Super Video CDs, and video burned to DVD-R, to MP3 files. Because timing is critical for presentations, the PMD-910 provides controls for rapid title, track, and time searches and fast file opening.
- Delivers stunning Dolby Digital and DTS multi-channel compatible sound. It boasts 96kHz/24-bit audio D/A converters and a four special 3D sound mode for playback of music CDs and MP3 files.
- Can be tailored to an installation's requirements. Audio and video outputs defaults can be changed for bit-stream, downmix, dual mono mode, and 96k sampling rates. Aspect ratios can be set for 4:3 or 16:9. Front key lock prevents unauthorized usage.

PMD-930

- The PMD930's RS-232 serial port provides enhanced functionality, allowing 2-way remote control feedback of all functions. It also provides an easy way to defeat on screen displays and icons, or to fly in bitmap background images such as corporate logos and school crests that can appear upon disc start up or during custom sleep modes.
- The PMD-930 is built to last, offering three times the Mean Time Between Failure (MTBF) of a standard DVD player—45,000 hours. For presentations, the unit provides fine tune controls for color, saturation, brightness and contrast, eliminating the need to adjust a monitor or projector when alternating between multiple video sources such as DVD, VHS, or teleconferencing equipment.
- The PMD-930 offers an extensive array of navigation and picture search modes, such as zoom, shuffle, repeat, fast-motion search, and time search capability. It's possible to select A-B points to repeat specific portions of a video.
- Allows favorite DVD titles and chapters to be stored in memory. Navigation menus that are hidden by other players can be readily accessed and navigated by the PMD-930.

DV7110P

Professional DVD Player w/RS-232



For any A/V system under wired remote control the DV7110P boasts an RS-232C port enabling complete control and feedback. It is packaged with a custom rackmount kit to ensure smooth integration into installations. Not only does this DVD player provide exceptional picture quality and 5.1 Dolby Digital and DTS surround sound, but it also plays DVD-R and DVD-RW discs, which are quickly becoming the preferred media for corporate and educational video.

CDR-795

	PMD910	DV7110P	PMD930
45000 MTBF			•
Global Compatible			•
DVD and Video CD	•	•	•
NTSC / PAL Conversion	•		•
Super Video CD	•		•
DVD-R, CD, CD-R, CD-RW	•	•	•
DVD-RW, DVD+RW User Upgradable			•
DVD-R Formats		•	
MP3 Compatible	•		•
Progressive Scan	•	•	
Random / Shuffle Play		•	•
Program Play		•	•
Bar Code Capable			•
Bitmap Upgradable Startup Screen			•
Time Search Direct Play			•
Playback Control for Use with VCD & SVCD			•
Bitrate Indicator			•
Video Shift			•
Fine Tune Color, Saturation, Brightness & Contrast			•
Black Level Shift Adjustment			•
Screen Saver (dim 75% after 15 min.)			•
Hide OSD and Icons via RS232			•
INPUTS / OUTPUTS			
RS232 Control		•	•
Optical Digital Output	•	•	•
Coaxial Digital Output	•	•	•
RC5 Input / Output	•	•	
RCA Analog Ouputs	•	(x2)	•
Composite Video Ouput	•	(x2)	•
Component Video Outputs	•	•	•
RGB Output on 4 BNC			•
S-video Input	•	•	•
Multi Voltage (100v-230v)			•
Shipping Weight	11 lbs.	13 lbs.	13 lbs.

Dual Deck CD Player/CD Recorder



Make your own compilation CDs with the CDR795. This system allows 4x recording and playback of CDs, CD-Rs, and CD-RWs. CD Text capability enhances the display of your CDs (remote control has CD Text capability for further ease). Want to program in advance ? Program up to 99 tracks. A Sample Rate Converter keeps digital audio consistent when recording.

- 4x speed precision recording from built-in CD player
- Simple "COPY CD" button for making a quick copy
- Audio Core drive with Bit Sync technology for true copies
- Plays computer CDs with MP3 music (for over 12 hours of music)
- CD Text display, edit and recording
- 2 disc changer mode for long listening
- Archive old cassettes and LPs on CD
- Digital, optical and analog inputs allow recording from any source
- Enhanced Auto Sync Start recording from external analog sources
- One Touch Timer recording from external sources
- Smart Finalizing reminds you to finalize only when needed
- Headphone output with level control

CDR-820

CD Recorder + Integrated 3 CD Changer



Easy to set-up and easy to use, the CDR820 makes recording CDs a breeze. The 4 disc changer mode plays CDs, CD-Rs, CD-RWs, and MP3 discs. Record your discs at 4x speed in less time. Automatic Level Control allows you to record compilations all at the same volume level. Add CD Text to your recorded CDs or from original CDs on the recorder or use the standard PC keyboard input for easy editing.

- Make your own compilation CDs
- Precision 4x recording from internal CD Changer
- Plays computer CDs with MP3 music
- Record compilations at the same volume level with Auto Level Control
- Mix and record your own voice on CD with stereo microphone input
- Add CD Text to your recorded CDs or copy from the original CD
- PC keyboard input for easy editing
- Digital & analog record level control for correct recording volume
- Headphone jack
- Remote control with CD Text input possibility
- Advance and easy programming (99 tracks) for recording compilations
- Sample rate converter for recording from multiple digital audio sources
- Audio buffer for perfect track starts
- Automatic CD Synchro Recording start from all analog & digital sources
- Digital coaxial and optical inputs & line input and outputs
- Direct Line Recording for perfect digital copies (for true bit recording)
- Instruction video tape included

MINIDISC TECHNOLOGY



The incredibly unique aspect about MiniDisc is that it is your music. No other audio format allows you to make your own custom, digital recordings—with the ease and flexibility — that MiniDisc gives you. And the mixes you record are distinctively your own. The titles you create can be as open as your imagination. Create and re-create mixes as often as you want—you can continually re-record on a single MiniDisc up to 1 million times without any loss of quality. MiniDisc hooks into everything. Record from your favorite digital (CD, DAT) or analog (LP, cassette, radio) music source to make your mixes. Some models even offer high-speed dubbing from CD-to-MD. Record live sound from dictation or, at concerts and lectures using a portable with mic input. MiniDisc also connects to your PC. You can record music files straight from your hard drive or the Internet. Convert and playback any standard audio format, including MP3, Real Audio, Windows media audio, ripped CD tracks, and Liquid Audio to make your mix complete. Indulge in complete mastery over your music. There isn't a sound around that MiniDisc can't add to the mix. No other portable, recordable digital player can do this and still sound so good.

The Perfect Format for Creating and Editing Music Mixes

Easy Editing:

MiniDisc records in non-linear fashion. You can edit the order of tracks with the touch of a button, a feat that is impossible on recordable cassette and somewhat difficult with recordable CDs. You can even store a single song as several fragments if you want to be able to jump right to your favorite part.

Portable:

MiniDisc is tiny (2.5″) and is encased in a protective cover, so you can carry lots of discs with you and enjoy hours of music. On the other hand, CDs aren't nearly as easy to carry around. You can fit four or five MiniDisc in your shirt pocket.

All Digital:

MiniDisc recordings have great, clean sound with wide dynamic range, incredible frequency response, low distortion, and no hiss. In addition, MiniDisc has great editing features which allows you to customize your music selections.

Durable Construction:

MiniDisc machines are practically unshockable. Their special system for storing information allows your player/recorder to pinpoint the exact position of the laser, even when suddenly bumped. It reacts with incredible speeds to readjust the laser so you hear sweet music instead of sweet music.

Flexible Playback:

As the MiniDisc line continues to expand, you are not limited to playback on the go. You can record and playback at home, in your car, and even from your PC.

Affordable Media:

MiniDisc media is really affordable when compared to the cost of other standard flash media used in most MP3 players. The average price of a 80 minute MD is under $2, while a 64 MB of flash media (which holds approximately 2 hours of music) is over $60. You don't have to be so cautious when deciding what to save on MiniDisc; you can build your recorded music library affordably.

Features	MiniDisc	CD-R (Recordable)	CD-RW (Re-writable)
Sound Quality	Digital	Digital	Digital
Recordable	Yes	Yes	Yes
Re-recordable	Yes (up to 1,000,000 times)	No	Yes (up to 1000 times)
Ease of Recording	Easy	Complex	Complex
Rearrange, Add, Delete and Move Tracks	Yes	No	No
Disc Plays in Existing CD Players	No	Yes	No
Protected Disc	Yes	No	No
Portable Recorders	Yes	No	No

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MXD-D40

CD Player/MiniDisc Recorder



With the amazing MXD-D40 you can dub from CD to MiniDisc—at 4x normal speed. That means you can record a three minute song in just 45 seconds, or a 74-minute CD in less than 20 minutes. The MXD-D4 has two transports, one for the CD you want to copy, and one for a blank MD. And it's easy to use. It features dual front-panel jog dials (one for CD and one for MD), plus a large two-line display that shows titling information for CD (if it's a CD Text disc) and MD. Digital record level control lets you match volumes from song to song when making a "mix MD" from multiple CDs. Of course, you also get all the classic MiniDisc editing functions: titling, instant erase, move, divide, combine, etc. A sampling rate converter lets you make direct digital recordings to MD from non-44.1 kHz sources (like DBS or DAT) via the optical digital input.

- MDLP-compatible, the MDS-JE470 can record 74 minutes (Standard Mode), 2.5 hours (LP2 Mode), over 5 hours (LP4 mode) on a single disc
- PC Link terminal for full computer control with the optional PCLK-MN10
- Analog and digital recording levels ensure better control over your recordings
- Two line fluorescent display shows title information for both CD and MD
- Automatic CD Text and Custom File transfer via Control A-1 II interface
- Scale Factor Edit enables you to adjust the volume of each track after recording, without disturbing the music itself. Ideal for compiling MDs, discs recorded at different times and from different sources.
- Optical digital input, headphone output

MXD-D5C

CD Changer/MiniDisc Recorder



Load five CDs, a blank MiniDisc, and you're set! The MXD-D5C gives you tremendous functionality and versatility by integrating multi-CD playback, MD recording, editing and playback, and internal high-speed 4x dubbing from CD to MiniDisc— in one compact deck.

- Very similar to the MXD-D40, the MXD-D5C has all the same features except it doesn't have a PC Link terminal and cannot record in MDLP mode. It does add a PC keyboard input allowing use of any standard PC keyboard for typing in disc names, artist names and song titles.
- Dual front-panel jog dials (one each for CD and MD) and a direct-access remote keep you in control, while a large two-line display keeps you informed. Enter disc/track titles for your MDs; you can even plug in your PC's keyboard for easy and fast title entry. Additional features include digital record level control so you can match volumes from song to song when you're mixing from a batch of CDs, and Scale Factor Edit allows you to adjust MD track volumes after they're recorded.

MZB-100

MiniDisc Business Recorder

Lightweight and easy to transport and use almost anywhere, the MZB-100's built-in stereo flat mic provides convenient recording versatility in a variety of situations.



- Stereo flat mic records separate left and right channels
- Voice Operated Recording (VOR) activates the recorder only when audible sound is present, saving tape and eliminating soundless passages
- Record up to 320 minutes using standard 80 minute MiniDiscs—record business meetings or press conferences without worrying about recording capacity.
- Record 14 hours or playback 45 hours on one AA battery
- Other features include -20%/+10% playback speed control, multiple track marks for later reference, large 3-line dot matrix LCD, built-in front speaker, easy search function, and an external microphone jack.

SONY

MDS-JE470 / MDS-JB940

MiniDisc Recorders



Digital, portable, recordable and virtually unshockable, MiniDisc records music your way. MiniDisc stores your music as computer files so you can Move, Combine and Divide tracks to your heart's content. No FF or REW to search for tracks. Save disc name, track names and artists names. The MDS-JE470 is a basic recorder, while the JB-940 adds a host of extra features including Scale Factor Edit to create professional sounding fade-ins and fade-outs, coaxial digital inputs and outputs, playback pitch control and PC keyboard input for disc and track names.

FEATURES

Highest Quality

- MDLP-compatible, the MDS-JE470 can record 74 minutes (Standard mode), 2.5 hours (LP2 mode), over 5 hours (LP4 mode) on a single disc.
- PC Link terminal for full computer control with the optional PCLK-MN10
- ATRAC Type 'R' –the heart of MD recording technology, features error evaluation, a second stage of bit re-allocation and superior processing accuracy to yield smooth, more clearly delineated sound.
- 24-bit A-D converter establishes high precision in the record mode, with a wide dynamic range. Forms the input for Sony's Wide Bit Stream technology. which maintains 20-bit processing throughout the recording process, for clearer reproduction of delicate, low-level music.

Recording/Playback

- Time Machine recording with 6-second frame buffer lets you hit the record button up to six seconds late–and still capture every note. Ideal for recording live events or radio broadcasts.
- CD Synchro Record simplifies making tapes from compatible Sony CD players by releasing from Record/Pause when the CD player goes into Play.
- Sampling Rate Converter lets you record from sources with different sampling frequencies, such as DAT. Automatically converts them to MiniDisc's 44.1 rate.
- Smart Space function provides a uniform three-second spacing between all songs.
- Play modes include Continuous, Shuffle and Programming of your favorite songs in your own selected sequence.

Editing

- Move, divide and combine tracks—shape your music even after you've recorded it. For example, you can combine Track 1 with Track 5 for seamless, uninterrupted play. Perform editing functions like A-B erase—ideal for deleting interruptions like radio commercials. Undo lets you reverse your last move, erase or combine.

Conveniences

- Jog dial for track selection and text entry. There is storage space for hundreds of characters of text, so you can store disc names, track names and artists names.
- Record without first searching for the end of a recorded section—the deck automatically finds blank spots on the disc and records on them without interruption.

PCLK-MN10 *PC Link Kit*



The optional PCLK-MN10 kit (consists of USB audio device, M-crew CD-ROM, optical and USB cables) allows you to use your PC to remotely operate all transport functions on the MDS-JE470 (and MXD-D40), play audio CDs in your CD-ROM drive and tune into Internet radio stations. You can record music CDs or .WAV files to MiniDisc. You get full editing control of the MiniDisc in the deck. Go through it track by track entering names or other information. Keep track of played or recorded music via a Disc Library, access related websites to the tracks you're playing, while you're playing them, set recording and playback timers and even automatically print out MD labels.

MDS-JB940 Step-up Features (no PC Link)

- Scale Factor Edit enables you to adjust the volume of each track after you've recorded it, without disturbing the music itself. Ideal for compiling MDs, discs recorded at different times and discs recorded from different sources.
- Variable Coefficient Digital Filter allows adjustment of the audio tone to complement your audio system, listening environment, and the type of source being played. Customize the music the way you want, by changing the slope characteristics of the filter. Operates at 24-bit word length for superlative precision.
- PC Keyboard input enables you to plug in a standard PC keyboard and type in disc names, artist names and song titles.
- Playback Pitch Control lets you change the playback (pitch) speed. Tone rises at higher pitches, and falls at lower pitches. Pitch Control is ideal for speeding up or slowing down foreign language tapes or lecture notes. Also enables you to "tune" playback sharper or flatter, to match your instrument. (You can lower the pitch by up to 36 steps or 3 octaves. Additionally, you can control the playback speed in 0.1% increments from -87.5% to 0.0%.)

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MDS-E10 / MDS-E12

Professional MiniDisc Recorders



Elegantly housed in a 1RU high enclosure, the MDS-E10 and MDS-E12 are affordably priced professional MiniDisc player/recorders with an impressive list of editing and programming features. They offer pitch control, non-destructive editing and "Hot" starts which allows the beginning of up to 10 songs to be stored in memory for instant start capability. They also include a full function remote control and standard PS/2 keyboard input allowing even easier editing and programming capabilities. With their compact design and remote interface capability they are well-suited for use in variety of mobile A/V, live sound (contractor) and broadcast applications. The MDS-E12 has all of the features and functions of the MDS-E10 plus adds control interface capability (RS-232 and Parallel), rec/play relay control I/O and balanced XLR analog I/O.

FEATURES

Highest Quality

- High-quality 24-bit A-D and D-A converters assure accurate sound reproduction
- Sony's Wide Bit Stream technology handles signals during processing at 20-bit resolution providing a dramatic improvement in the accuracy with which low-level signals are processed. Results in higher recording quality, with a wider dynamic range.
- Record levels are adjustable on both digital and analog inputs:

Conveniences

- Machine status, audio levels, menus, disc and track info are displayed with a large, easy-to-read fluorescent display.
- The TOC of a disc, even a protected master disc can be copied to RAM and edited without changing the TOC data on the disc.
- Time Machine Recording function continuously stores 6 seconds of audio data in its memory buffer ensuring no program material is lost because of missed cues.
- Engaging the Music Sync button puts the recorder into Record/Pause mode waiting for the source deck to begin playback, at which time recording will start.
- Auto Space feature inserts 3 seconds of silence in between tracks.
- Auto Repeat function with four modes: all tracks, current track, shuffle and program.

High Performance

- Full-featured TOC (Table Of Contents) editing includes undo, divide and combine.
- ±12.5% vari-speed playback in 0.5% steps in standard mode and +0 to -12.5% when playing back an MDLP2/4 recording
- The first few seconds of up to 10 tracks are memorized to provide an instant, hot start, from any of these tracks. This is useful for radio/DJ and live theater production.
- MDLP (Long Play) stereo recording mode is useful for recording meeting minutes, logging, etc. Two stereo modes are available: x2-for up to 160 mins & x4-for up to 320 mins. (using 80 min. MD media).
- When recording a single, long track, the track to be divided into segments of identical duration This makes it easier to locate material during editing and production. Tracks can be marked automatically at timed intervals (from 1-10 minutes) or via a level sync mode which detects the start of music after a long pause.
- Scale Factor Edit allows relative levels of various tracks to be changed after recording. This prevents wildly different playback levels, which can occur in productions done from various sources. Fade-in and fade-out can also be added to recorded tracks.
- Up to 1,700 characters can be used to title a disc and individual tracks.

Input/Output

- Headphone output with level control
- S/PDIF coaxial and optical digital I/O
- RCA unbalanced analog I/O
- Automatically converts 32 or 48 kHz audio to MiniDisc's 44.1kHz sample rate.
- Frequency response of 5Hz to 20kHz with 97dB dynamic range
- Their front panel is equipped with a standard PC keyboard interface, which allows easier access to remote and track/disc titling functions, plus hot starts. Provides the same functions as the wired/IR remote.
- Supplied dual-mode RM-DR1E Remote Control operates as a conventional IR remote or wired remote when connected to the Control-S jack via supplied 5m cable.

MDS-E12 Step-up Features

- Connect several MDS-E12s together with a stereo 3.5mm mini plug for uninterrupted sequential playback and recording.
- RS-232C machine control interface
- Coaxial S/PDIF digital I/O (no optical)
- Switchable XLR balanced and RCA unbalanced inputs and outputs Balanced outputs are switchable (+4dBu to -10dBu)
- 9-pin parallel remote interface allows external control and monitoring of machine functions including fader start.

SONY

PCM-R300/500/700

DAT Recorders



The PCM-R300 is a semi-professional rackmount DAT machine well suited for the home studio and desktop video suite. Equipped with standard RCA (phono jacks) and S/PDIF digital I/O (both optical and coaxial), the PCM-R300 easily integrates into most any recording environment. It also incorporates the same Super Bit Mapping 20-bit analog to digital converters as Sony's higher-end decks for superior audio reproduction.



The PCM-R500 and PCM-R700 are designed with the professional audio and video studio in mind. They are built around the stability of 4-motor direct drive transports, and a flexible number of analog and digital I/O terminals are available for connecting to almost any audio hardware. Sony's 20-bit Super Bit Mapping technology enables recordings rivaling near 20-bit resolution.

FEATURES

- Analog recording supports 16-bit 44.1 and 48 kHz sample rates (standard mode) and 12-bit/32kHz Long Play Mode (4 hours recording on a 120-minute tape).
- Digital input recording accepts sample rates of 32, 44.1 and 48 kHz.
- 20-bit Super Bit Mapping analog to digital converter gives you a S/N spec close to that of an actual 20-bit recording.
- RCA (Phono) line inputs and outputs
- Coaxial and Optical S/PDIF digital I/O
- Headphone output with level control
- Input monitor lets you audition audio without engaging transports. This saves the tape and heads from wear and tear.
- AMS (Automatic Music Sensor) lets you easily shuttle to any track on a tape.
- Program Number and Start, and ID subcodes can be written manually or automatically for direct track number access via remote control or AMS function.
- LED display shows level and margin, transport status, sample rate, input source, program number, tape running time, absolute time, playing time of the track, remaining time on the tape, date and time of recording and current date and time.
- Counter mode display switchable between absolute time, program time, remaining time on tape, error rate and menu
- Displays total drum operating time up to 9999 hours in 1 hour increments
- Includes RM-D757 wireless remote and rackmount adapter.

Menu Operation

- Menu settings are stored internally and memorized after power down
- Record mute duration can be set between 0.5 to 9.5 seconds in units of 0.5 seconds.
- Settable Serial Copy Management System (SCMS) allows multiple copies, one copy or no copies of a DAT tape.
- When recording digitally, start ID and skip ID detection is settable to on or off.
- Level sync threshold sets the reference input level for recognizing start IDs. Set from -12 to -60 dB in 1dB increments.
- The ability to recognize CD-Q codes (track numbers) when recording from a CD is settable to on or off.
- Level sync blank time sets how long the audio signal must stay below the sync threshold before a new start ID is written.

PCM-R500 Step-Up Features

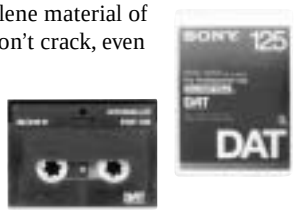
- 4 Direct Drive motor transport ensures a reliable and extremely stable tape path
- Independent record level controls for left and right analog inputs
- Automatically or manually writes Program Numbers, Start Id's, Skip Id's and End Id's subcode information.
- Jog dial/Shuttle wheel - shuttle wheel allows audible cue and review ±0.5 to ±8 times normal speed (±1 to ±8 times in Long Play mode), while the jog dial is used as a data entry device for menu options and AMS (Automatic Music Search) Program Number access
- Erase start-ID and skip-ID function
- Repeat function plays a single track or entire tape continuously
- Renumbering function searches all start-Ids from the beginning of the tape and writes consecutive program numbers
- Specific absolute time locations can be temporarily stored and recalled using the Mark and Locate function. Stored locations are lost once a tape is removed or the deck is turned off.

PCM-R300/500/700

- Rehearsal function ensures accurate positioning of subcodes. Once a general location for the desired subcode is found and the rehearsal button is engaged, a 3 second section of tape will repeat 8 times while the FF and REW buttons allow fine adjustment of the locate point in 0.3 sec. steps. Subcodes can then be written normally.
- Existing start IDs can be accurately repositioned as much as 2 seconds using the rehearsal function.
- XLR balanced and RCA (Phono) unbalanced analog I/O for compatibility with professional and consumer equipment
- Switchable between XLR balanced AES/EBU and coaxial S/PDIF Digital I/O
- 8-pin parallel port is available for custom wired remote control of transports.
- Serial remote input for optional RM-D750 wired remote
- Includes wireless remote commander

5 Reasons to Choose Sony PDP Series DAT Tapes

- The PDP Series employs ultrafine "Chrystal Art" magnetic particles, which deliver high retentivity and outstanding output.
- Sony's HD&R Binder Systems allows for higher densities and better durability, while the special backcoating ensures a tape with low error rates, even after 1000 cycles.
- The anti-static resin lid shuts out dust and other unwelcome particles. Also, the high-precision heat resistant shell prevents warping due to excessive heat.
- Polypropylene material of the case won't crack, even when dropped.
- "Dust-free" lid guards against dust being generated during openings and closings. A stronger lock mechanism and roundish edge construction protect against shocks from falls while also preventing tape slack.



PCM-R700 Step-Up Features

- 4 heads allow confidence monitoring (monitoring off of tape while recording)
- Automatic Fade in & Fade out adjustable between 0.5 to 9.5 seconds
- Key protect function disables certain front panel controls to prevent accidental erasing or manipulation of a tape.

PCM-7040 *Digital Audio Recorder*



Supported by a wide range of essential features for DAT recording and editing, the PCM-7040 is the ideal 2-track digital recording solution for professional post production and broadcast. Features such as an onboard timecode reader/generator, digital I/O, memory start, an RS-232 interface and edit memory are standard. Using the internal timecode reader generator, editing, chase synchronization and conversion between the SMPTE/EBU/FILM and DAT timecode formats is possible. Both drop and non-drop frame rates are supported as well as the ability to reference to composite video, composite sync and blackburst signals or in the case of a digital audio configuration, word clock I/O is also supported. The memory start function stores 3 seconds of stereo audio, overcoming the inherent delay found in a normal rotary head DAT system, thus allowing the instart capabilities required by broadcast facilities.

- 4-head rotary system provides two recording modes: RAW (Read-After-Write) and RMW (Read-Modify-Write). In RAW mode, real-time off tape monitoring is provided. In the RMW mode, the PCM-7040 allows enhanced punch in and punch out operation with crossfading at the punch in and out points.
- Precision-controlled transport allows ±12.5% variable playback speed. Varispeed recording is also possible allowing "pull up" and "pull down" ±1% recording sample rates (for example, 48.048kHz or 47.952kHz). Pull up or pull down recordings permit automatic ±0.01% speed change during playback for film/video post production when necessary.
- Memory Start Function (3 seconds) makes the PCM-7040 suitable for on-air applications and audio post production.
- Time Code Reader/Generator, Digital I/O, Memory Start and Edit Memory are all standard.
- Automatic time-stamping of recording
- Start ID level Sync Function enables the PCM-7040 to write a Start ID automatically each time the audio level rises from period of silence and exceeds a selectable audio threshold.
- RS-422 (9-pin) serial remote and RS-232C for convenient control from external equipment and computers.

SONY

CDP-D11

Professional CD Player



An incredibly compact—only 1RU high—professional CD player, the CDP-D11 offers a host of state-of-the-art features including instant start of tracks, variable speed operation, a dual mode IR wireless/wired remote controller and connectors for both RS-232C and programmable parallel remote control. Superb audio performance is guaranteed by high density linear digital to analog converters. The CDP-D11's space saving design includes a slot-in loading mechanism designed for high reliability and with resistance to shock and to high temperatures. A three-second Advanced ESP anti-shock memory is also standard. Outputs include balanced-XLR and unbalanced RCA connectors as well as coaxial and optical digital outputs. Ideal for commercial and professional applications including recording studios, theatres and A/V presentations.

FEATURES

- Instant start allows immediate playback from start of any track
- Mark and locate feature allows user defined spot as cue point.
- Advanced electronic shock protection with 4MB D-RAM buffer (2.972 msec) virtually eliminates unwanted "skipping" due to vibration or dirty CDs.
- Search modes include AMS (Automatic Music Sensor) dial for fast and convenient access to CD tracks and menus, Direct Play and Music Scan
- Three search speeds: Low speed search (2x), standard search (20x), and high speed search (80x)
- Play and repeat modes include Program and Shuffle; A-B repeat is also available
- Auto cue pauses track at the start of music. Sound level (threshold level) can be selected from five levels (-72dB, -66dB, -60dB, -54dB and -48dB).
- ±12.5% pitch control in 0.5% steps)
- Last Memory Mode retains the last settings after the unit is turned off and returns to them when power is turned on.
- Connect with an external timer to start or stop playback at a specified time.
- Long-duration sequential playback can be achieved using several CDP-D11 units connected via the Relay Play input and outputs (using a stereo mini plug cable). After a first unit ends, a control signal goes through through the jack and makes a second unit start to playback automatically.
- Remote control via RS-232C or parallel remote control ports. The parallel remote can be user-assigned to control and monitor several functions of the machine.
- A special "fader play"mode is provided on the parallel interface to automatically play tracks in order as the fader start switch is opened and closed.
- Headphone output with level control.
- Flexible interfacing includes balanced XLR and unbalanced RCA connectors for analog output and coaxial and optical S/PDIF (IEC 958 type ll) digital outputs.
- Supplied dual-mode wired/IR wireless remote controller connects to the 3.5mm Control-S jack with a supplied 5m cable. Wired remote control is ideal for use in installations with multiple players. Or simply operate as a wireless remote if desired.

CDP-3100



A professional CD player with superb sound quality, the compact CDP-3100 features 18-bit 8x oversampling D-A converters, fader start function, end-of-track alarm, AES/EBU digital output, balanced analog outputs, and headphone and monitor outputs for previewing of cue point. It also features extensive transport control and location functions such as Jog Search, CD frame-accurate location, "Instant Start", selectable timer modes (Remaining, Actual and Set Time), single play mode and index mode, and ±12.5% variable speed playback when equipped with the optional CDS-3100 remote control or DABK-3101 Memory Board.

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ONE HOUR FREE PARKING
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CDR-W33/CDR-W66

CD-R/CD-RW Recorders



Even among professional CD recorders, the CDR-W33 and CDR-W66 stand apart. Three DSP functions fine-tune your sources. There's a limiter with adjustable ratio. A 3-band EQ. And Super Bit Mapping circuitry captures near 20-bit quality on industry-standard 16-bit CDs. Physically similar and sharing many features, both the innovative CDR-W33 and the CDR-W66 feature 24-bit A/D and D/A converters, easy-to-read fluorescent display, sample rate converters, CD-Text support and are rack mountable (2RU). Remote transport control can also be accessed via Control-S or a PC keyboard. In addition, they include a dual-mode wireless/wired remote control.



Designed for budget-conscious users, the CDR-W33 is equipped with RCA analog as well as coaxial and optical digital (S/PDIF) I/O. The CDR-W66 adds balanced XLR analog and AES/EBU digital I/O, Word Clock interface, RS-232 and parallel (GPI) control ports, and 2x speed duplication link for dubbing audio titles (using 2 CDR-W66 units), making it ideal for mid-to high-end recording studios and TV/radio broadcast production. On the CDR-W66, DSP functions are available on digital inputs as well as analog.

FEATURES

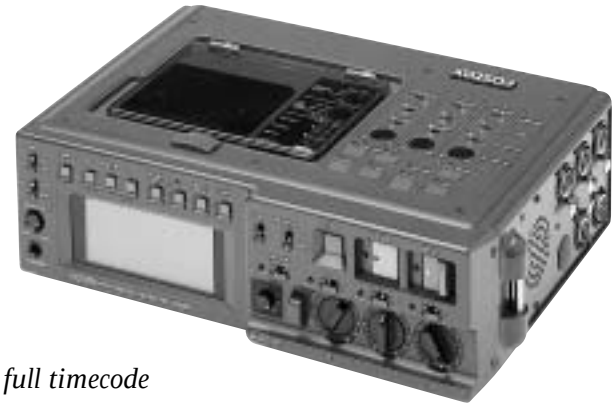
- 24-bit A-D converters work together with the Super Bit Mapping (SBM) filter to provide natural sounding 20-bit quality when converting 24-bit data to 16-bit data.
- Digital limiter lets you maximize recording levels without introducing distortion caused by hard clipping. The limiting ratio is adjustable up to 100%–at which the gain is increased by 6 dB.
- Built-in digital 3-band (Bass/Mid/Treble) parametric EQ provides up to ±6 dB of correction (in 1 dB steps) allowing you to change sound character
- Record disc and track name in CD-TEXT format. You can record up to 23 characters each for disc name and track name.
- For convenience, it offers synchro-recording with a similarly equipped Sony CD Player
- 32-48 kHz sampling rate converter lets you record not only CD and MiniDisc 44.1 kHz signals, but also 32 kHz and 48 kHz signals from digital sources such as DAT, satellite digital broadcasts and DVD.
- Skip setting function lets you create discs that skip unwanted tracks (when playing back) from among those that have been recorded. (Requires compatible player).
- Record level adjustments is available on both digital and analog inputs.
- CD-TEXT function allows disc/track names to be displayed and entered from either the front panel AMS controller, the supplied remote control or an optional PC keyboard.
- Equipped with useful recording functions such as fader (adjustable between 1 and 8 seconds), mute recording (4 seconds or as long as long as you press the mute button), music sync recording (automatically starts recording, depending on the input signal), and auto track marking.
- Playback functions include shuffle and program mode, auto-pause, music scan (play the start of tracks one after another), and repeat play (1 track/all tracks/A-B repeat)
- Equipped with both analog (RCA) and digital (optical and coaxial) inputs and outputs.
- Supplied dual-mode wired/IR wireless remote controller connects to the Control-S jack with the supplied 16-ft. cable. Wired remote control is ideal for use in installations with multiple players. Or simply operate as a wireless remote. A PC keyboard can also be used to control transport functions.

CDR-W66 Step-up Features

- All DSP functions including Super Bit Mapping (SBM), digital limiter and EQ can be applied to both the analog and digital AES/EBU inputs. (CDR-W33 only enables DSP functions on the analog input.)
- Unique 32-to- 96 kHz sample rate converter allows compatibility with higher resolution music recording formats.
- RS-232C and 9-pin parallel connectors allow the CDR-W66 to be externally controlled from a PC or remote-control options such as GPI triggers (fader starts).
- Duplication bus link allows 2x speed CD dubbing from one CDR-W66 to another.
- Adds XLR-balanced analog inputs/outputs plus AES/EBU digital I/O.
- Word Clock input assures accurate sample clock synchronization to other digital audio sources.
- User-selectable SCMS (via menu) allows you to select the level of SCMS copy-prohibit data written to the disc. CDs may be written with the maximum level of protection or none at all.

PD-4M

Professional Portable Timecode DAT Recorder



A complete, all-in-one portable field recorder, the PD-4M combines a professional four head timecode-enabled DAT recorder with a three-channel mixer in one lightweight but rugged package. Used in award-winning movies and television shows throughout the world, the PD-4M meets all the requirements of the demanding professional on location, in a package that actually rivals a studio production DAT recorder in quality and capability. The PD-4M features three XLR-balanced mic/line inputs, confidence monitoring via the four head transport, variable frame rates and sampling frequencies including 48.048kHz for digital video, full timecode recording including JAM sync, on-board three channel mixer with mic attenuation, phantom power, low-cut filter, limiter on each channel, and a headphone matrix. Powered by on-board NP-1B type batteries (or via an optional AC adapter), weighing less than 7 lbs and equipped with a unique "Anti-Jam" transport mechanism, the PD-4M is designed for the most demanding field work—whether in film, video, live music or ENG applications.

- 3 into 2 XLR-balanced mic/line mixer with independent gain control, -20 dB pads, 3-position (left, center, right) pan switches, 48v phantom power and high pass filter variable between 20Hz to 250Hz for each channel and a master output control.
- Sampling frequencies are selectable between 44.1kHz, 48kHz and 48.048kHz (for digital video).
- XLR-balanced AES/EBU and S/PDIF digital I/O
- A switchable stereo limiter prevents transient peaks from overloading tape.
- Choice of monitoring using 1/4″ headphone output or the built-in speaker.
- Subcode functions include PNOs (Program Numbers) and Start IDs
- Comprehensive illuminated display with transport direction tallies, timecode, level and subcode status.
- Panel lock feature prevents unwanted operation of the transport and false triggering of keys out in the field.
- Operates for two hours on a single NP-1B battery. Can also be powered by a power belt or optional AD-15 AC adapter.



D-5

Affordable Full-Featured DAT Recorder

An affordable full featured DAT recorder, the D-5 features balanced XLR analog I/O and AES/EBU digital I/O, three selectable sampling rates, and +4 or -10dBu compatible analog inputs. It also offers a unique jog/shuttle function that makes audible cue and review of a tape a snap.

- Supports 16-bit recording at 48, 44.1 kHz sampling rates and 12-bit/32kHz in Long Play mode. Long Play doubles the amount of recording time available on tape for long recording sessions.
- 3x and 5x speed audible CUE/Review in playback, 300x search speed (under 60 seconds for a 120 minute tape) and Automatic Index search for locating Start ID's.
- Auto recording of Start-IDs and Program-Numbers (PNOs)
- Margin display with reset key, numerically references the highest input level in dB, aiding in setting optimum input levels.
- Recognizes CD-Q codes (track numbers) via the S/PDIF input. This ensures accurate Program Number (PNO) encoding when digitally recording from CD.
- SCMS free recording allows unlimited duplication of DAT tapes
- AES/EBU and optical S/PDIF digital I/O as well as XLR-balanced analog I/O. Inputs are switchable between +4dB and -10 dB.
- GPI input for external transport control, ID search and fader starts.
- 1/4″ headphone jack with level control

D15 / D15TC / D15TCR

Professional Timecode DAT Recorders



A very affordable timecode-capable DAT recorders, the D-15 series offers XLR-balanced analog and AES/EBU digital I/O, jog/shuttle wheel and instant start capabilities, essential for professional audio applications. A 4-motor transport ensures rock solid performance as well. Upgradable or available with the ability to read and output timecode (D-15TC) at any frame rate or RS-422 machine control (D-15TCR).

FEATURES

- Supports 16-bit recording and playback at 44.1 and 48kHz sample rates.
- 18-bit, 64X oversampling A-D and 20-bit, 8X oversampling D-A converters
- Efficient 4-motor transport (2-DD) for stable performance and reliability. Up to 200x search speeds
- Jog/Shuttle allows you to audibly cue or review tape at 1/2 to 2x speed in jog mode and 1/2 to 15x in shuttle mode.
- Input reference levels can be set to -12, -18 or -20dB. Default setting is -12dB.
- Instant start without pre-loading into RAM or you can use the 10 second RAM buffer for locating specific start points.
- 100 locate points can be stored in memory for editing complex tapes.
- Auto Record when used in conjunction with memory locate points 00 and 01 allows punch-in/ out recording with a 5 second pre-roll. Locate points can be rehearsed before executing the punch.
- Skip ID mode allows three selectable options; stop, play or off.
- Auto Cue mode enables automatic recording of Start-IDs and PNOs. Auto Cue mode is executed when the initial sound threshold level is exceeded. Cue levels are settable to -55, -40, -30 or -20dB.
- Analog and digital inputs can be made active without engaging transports.
- Individual front panel trim pots for reference level calibrating. Allows control of input levels in ±2dB increments.
- Level trim controls adjust the balanced and unbalanced analog outputs.
- Front panel lockout function.

Comprehensive Displays

- Displays time, date, program number, level meter, margin level in 0.1dB steps, absolute time, CH-1 and CH-2 level and error rate.
- Preset display indicates status of ID mode, TOC, emphasis, sample rate, reference level, chase lock (TC, TCR versions only), PCM error and PCM mute.
- Peak reading digital bargraphs with 5 settings, can be held for closer viewing.

Inputs/Outputs

- XLR balanced and RCA unbalanced I/O
- XLR balanced AES/EBU and optical S/PDIF digital I/O
- 37-pin interface for external control of transport functions, RAM scrubbing and ID writing/searching.
- GPI input and output for control of the transport functions and RAM scrubbing
- Headphone output with level control

D-15TC and D-15TCR

Intended for professionals who need to sync to video sources, the D-15 when outfitted with the optional timecode expansion card is the lowest price professional timecode capable DAT recorder available. The D-15 has two expansion slots in the rear. The 8335 External Sync card can be installed for video reference facilitating striping of the tape with externally generated timecode and output of LTC or Absolute Time (ABS). Absolute Time can also be converted to LTC giving you the option of referencing externally to Video or World clock.
The second expansion slot may be used for the addition of the 8336 RS-422 protocol Serial Card (with the exception of vari-speed command). The 8336 board allows you to have your controllers shuttle the machine to desired locations. (Note, the 8336 must be used in conjunction with the 8335 Sync Card)

- Reads, writes and chases externally generated timecode. Can input and output LTC through its XLR or RCA I/O.
- Supports 24, 25, 29.97 and 30 frames per second (drop and non-drop frame)
- Timecode can be offset against master machine
- Can reference to external video or word clock (BNC connectors)
- 9-pin RS-422 control
- 8335 option card adds TC capability to standard D-15
- 8336 option card adds 9-pin RS-422 port to D-15TC

PD-6

Portable DVD-RAM Location Recorder with Timecode



When Fostex started out to design a successor to their PD-4 portable timecode DAT machine, they had set for themselves no easy task. Having invented both timecode and portable timecode DAT, Fostex location recorders had become, and still are, the most widely used and respected DAT machines for film & television sound production around the world, helping their users win many coveted international awards. That's why they knew it was important to follow the design and philosophy processes that had made their previous machines the first choice of sound mixing professionals. Namely, to prove the technology in a stationary, stand-alone format and obtain comprehensive and wide-ranging user-feedback. The result of this process is the PD-6. A portable location recorder which offers six audio tracks, timecode-locked DVD-RAM recording plus full-function integrated mixing, industry-standard BWF file format, IEEE1394 interface, pre-record memory buffer and extensive time code facilities—all in a rugged, lightweight and proven design.

FEATURES

- BWF recording file format (interleave 1 file mode) in either 2, 4, 5 and 6 track modes along with simultaneous two file recording modes of 1+5 and 2+4 tracks for guide track audio.
- UDF (Universal Disc Format) on DVD-RAM allows discs to be mounted and read instantly on both PC and Mac platforms
- On-board timecode generator with ± 1ppm accuracy and built-in backup offers all frame rates including 23.967 and 29.97 plus drop and pull-up/down. Generator can be 24H, Rec Run, Free Run or external complete with jam, while output can be repro or generator derived.
- BWF files recorded on the PD-6 can also be imported directly into Avid Film Composer (depending on software version) along with information about Scene, Take, Reel Number, Event, etc. being transported and read from the files Metadata. (The Metadata area is user-editable for future expansion).
- Circle Take to mark files/takes for easy identification leading to quick imports and EDL references
- Alphanumeric keypad for quick file/take and track naming, EDL manipulation, location, etc. Attach full-size keyboard via USB for easy alphanumeric entry and keyboard shortcuts.
- Up to 100 cue points per file/take
- Option of SDII and AIFF file recording modes via software update
- EDL management built-in to create & edit multiple ALE (Avid Log Exchange) compatible edit decision lists per DVD-RAM.
- IEEE1394 interface for fast back-up and restore. When interfaced to PC, PD-6 DVD-RAM discs can be mounted on desktop.
- Pre Record enables PD-6 to constantly buffer up to 10 secs of audio (no more missed takes).
- 128 x 64 back-lit dot matrix display offers various display modes including alternative level meter resolution indication modes and provides for high visibility with low power consumption.
- Digital I/O selectable between AES/EBU and S/PDIF has auto input sensing, overriding individual analog inputs as required
- Auxiliary bus input/output provides for remote camera working and monitoring (e.g. HD cam) and is switchable between +4dBu, -10dBu and –60dBu to cater for most camera manufactures requirements. Provided on industry standard 10-pin Hirose connector.

DVD-RAM: REVOLUTIONARY REMOVABLE MEDIA

Location recorders need a removable media, it was one of the requests that Fostex heard from the field again and again. Part of the success of the portable timecode DAT format was the fact that at the end of the shoot, the days work was in your hand. Fostex knew that the normal DVD-RAM drives featured in the DV40 were not built for the rigors of location recording and certainly couldn't be hung around the neck! But they've solved it: enter the new 8cm DVD-RAM drive. A drive specially developed for the rigors of portable use. Designed from the ground-up to provide vibration stability, high speed access and most importantly, high reliability. Having performed extensive testing with this drive unit, Fostex based the PD-6 around it. Recording to 8cm DVD-RAM cartridges, each side provides 1.46GB offering, for example, 126 minutes per disc of 4 track recording at 48KHz/16-bit.

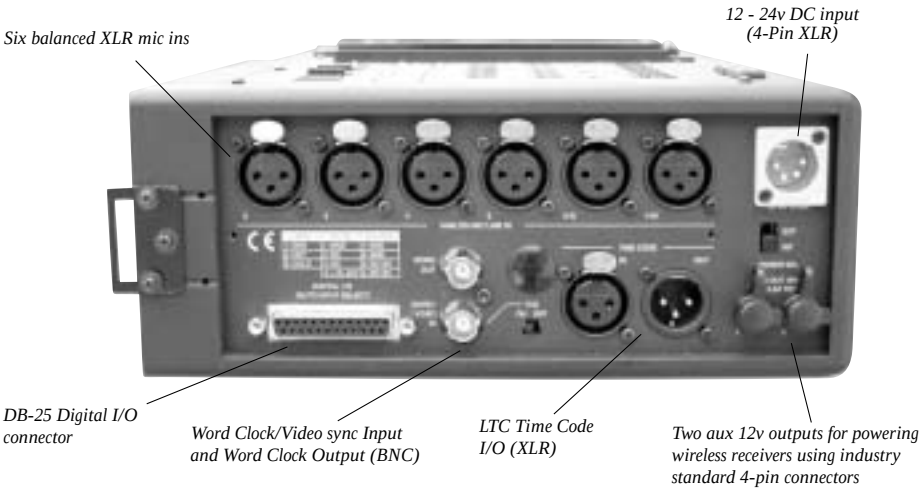
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PD-6

- Slate mic and tone generator on-board
- Flexible monitoring through headphones and/or built in speaker & amplifier. All track combinations can be monitored post-disc either individually or in summed mono and stereo modes. MS monitoring as standard. Aux return and stereo bus live monitoring is provided plus individual channel PFL. All accessed through just two rotary and one toggle switch.
- External power supply input works within 12-24v range for greater flexibility (use the optional AD-15B AC power supply).
- NP-1 type battery provides approximately 2 hours of operation
- Two auxiliary 12-volt outputs on standard Hirose 4-pin connectors provide power for radio mic receivers, etc. to allow for truly self-contained operation on location
- External word & video sync inputs (auto selection) and word output plus parallel remote connector
- Optional AATON connector complies with ASCII & LTC I/O specifications for external loggers, synchronizes, etc.
- Automated file/take naming routine speeds set-up time between takes complying to US and Euro standards. Individual track naming rolls through to next file/take until changed by user, if required
- Comprehensive software selectable UBIT output format combinations including Scene, Take, Event no., etc.
- Propriety expansion connector allows the PD-6 to dock with Fostex rack-mounting full-size DVD-RAM & power supply unit for extended recording time on location.
- An optional back-up hard drive is available to mirror the PD-6's main DVD-RAM drive.

Applications

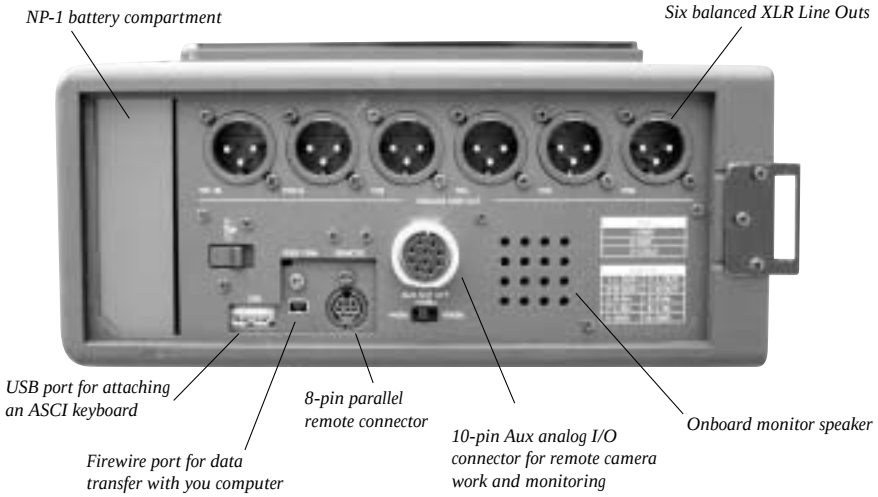
- Post Production, Archiving, Logging
- Stereo mixdown, Final Master sequencing
- Field Dialog and Music recording
- Live concert recording
- Broadcast playback
- Theme Park playback
- ADR (Automatic Dialog Replacement
- Commercial background music source



Full-Featured On-Board Mixer

The six channel mixer accepts either microphone level, with both 12v T-Power and 48v phantom power, or line level. Phase reverse is provided on the three even channels while each channel features adjustable input gain; a generous defeatable and variable high pass filter; and a digital limiter with user adjustable characterizes via software.

The most ingenious part of the mixer however is the routing capability: working in conjunction with each channel's level control mounted on the face of the PD-6, the 'Disc Feed' switch allows the recording section of the machine to derive its audio 'feed' PRE the channel level control, meaning levels are set once with the earlier trim pot, leaving the front level controls for bus mixing; POST of the channel level control providing for regular level adjustment on the easily-accessible controls; or from the Stereo Bus, routed by the PAN switch providing not only simple stereo mixing of all six channels, but also by using a combination of the modes, parallel mix (guide) tracks can be made while recording of either five tracks (mono guide track) or four tracks (stereo guide track) Fostex calls these "1+5" and "2+4" recording modes as separate BWF files are created. Additional features such as PFL monitoring and peak led indication on each channel complete this very flexible mixer.



DV40

DVD-RAM Master Recorder



Working hand-in-hand with post production facilities, sound mixers, broadcast professionals and recording houses, Fostex have developed a mastering machine ready to handle every application and exception in today's, and tomorrow's, constantly changing and challenging audio environments. In short, you can now move from Timecode DAT to 'Timecode DVD-RAM' with supreme confidence.

The future of post production recording, the DV40 is designed to replace (tape) DAT technology in a mastering format by recording audio data directly onto a DVD-RAM disc. The DV40 employs the UDF file format mode that establishes file compatibility between computer-based work stations. You can select either SDII or BWF (.WAV) as the desired audio format and either 24-bit or 16-bit resolution. This means you can take your media out of a DV40 and load it right into a Pro Tool based computer editing system without doing any file format conversion.

FEATURES

DVD-RAM Recording Format

- The successor to DAT tape, DVD-RAM has already established itself as the film industry's preferred media format due to its built-in error correction and longevity. This security is enhanced by the use of 'Verify/Write' technology which constantly examines the recorded data being stored on DVD-RAM, in real-time, for total error-free recording.
- To ensure the widest range of compatibility throughout the industry and maximizing the DVD-RAM technology, the DV40 can record up to four channels of simultaneous audio in a vast number of permutations.
- Offers two recording modes: New File and Insert. The Insert mode offers Time Jump UNDO functions supported by time and date. In Tape mode (destructive mode) data security for accidental power loss is secured up to the point where the power loss occurred. For back up and archiving a Disc duplicate mode is available by installing an optional hard drive, while Audio file duplicate modes are possible on board.
- DVD RAM incorporates Defect Sector Management to prevent data being recorded to defective sectors on a disc. DVD-RAM also offers extensive error code correction with more resolution than DVD-R/RW.

Universal Disc Format

- As the DV40 employs a UDF (Universal Disc Format) file layout, its DVD-RAM media can be removed, (or accessed via the house media network), and instantly read by Mac and PC workstations running a variety of operating systems. This can be a huge time saver as it alleviates the costly process of having to convert audio data through third-party programs beforehand.
- DV40 discs be read by any operating system (Windows, Mac, Linux, etc.) which makes it easy to transfer audio files to and from the DV40 for external editing.
- Records in industry standard SDII (Sound Designer II) or BWF (Broadcast .wav File) formats (more in development). Both formats can exist together on the same DVD-RAM disc and have been enhanced by the ability to include Fostex User Bit (UBIT) information in the file format.
- DV40 files can be set-up as 4-track (Multi), stereo or mono files in 24- or 16-bit resolution with a vast choice of sampling frequencies: 44.1, 48, 88.2, 96, 176.4 and 192kHz. A 0.1% pull up/pull down can also be applied to the selected frequency. (Optional HD installation required for 88.2 & 96kHz/4-track and 176.4 & 196kHz recording.

Editing

- Once files have been recorded or imported, they can be processed with built-in non-destructive editing functions such as Copy, Paste, Insert, Cut and Erase with a virtually unlimited number of Undos.
- Editing functions are made easy with the combination of a high precision tracking jog/shuttle wheel and a clear and concise FL display. The large high-resolution (200 pulses per rotate) jog/shuttle dial and clever digital tracking technology result in incredible analog audio scrubbing.
- File backup and duplication functions are also provided, either to the same DVD-RAM disc, or via the fitment of an optional internal hard disc drive.
- The DV40 offers internal file conversion between four-track or stereo files and multi-track mono files. File editing modes offer non-destructive edit functions such as Copy, Paste, Insert, Cut and Erase.
- Editing can also be enhanced with the installation of the optional VGA I/F graphics card. The plug-in card enables easy file management and waveform editing via an external VGA monitor. There is also a back panel port to interface to an ASCII 2 keyboard for inputting file names, TC locations, etc.

DV40

Superb Audio

- Delivers superbly transparent, natural sounding audio recording and playback. It produces up to 110dB of clear, quality audio with an incredibly wide 20Hz - 80kHz frequency range.
- The recording reference level can be set in software to -12dB, -18dB or -20dB with fine recording level adjustment on each channel ensuring a good match with the incoming audio feed.

User-Friendly Ergonomics

- The DV40 offers the most user-friendly interface ever seen on a professional recorder. The clear and concise front panel features a large multi-function fluorescent display, oversized illuminated transport keys, a large high-resolution (200 pulses per rotate) jog/shuttle dial and the kind of logical button and switch layout which makes any new user feel instantly at home.
- For example, cue point store and recall functions are based on the BWF format. Direct location of internal memory points and other locate functions such as Last Record Start/End points and Last Play Start point can easily be accessed via simple key presses.
- A front-panel alphanumeric keypad makes for quick and simple file naming (up to 255 characters) and direct timecode entry, etc. while the added advantage of being able to add a standard PS/2 keyboard and pointing device4 via front-mounted sockets will be appreciated in busy edit suites.
- In addition to UDF PD-6/DV40 files, DV40 can also read files created on other manufacturers recorders in FAT16, BWF Interleaved format complete with timecode information.

Optional Accessories

5050 2-CH: With the optional 5050 2-CH (6-channel playback kit), the DV40 can read the 6 track files created by the PD-6, simply by putting the disc in the DV40's tray. (A software update is provided allowing the meters to be switched between all six channels.

VGA: VGA video output option

HD OPTION: Internal hard drive kit

Timecode

If the audio quality and editing facilities offered by the DV is spectacular, it's the implementation of the timecode facilities which really sets the DV40 apart. Here is a machine which elegantly addresses the challenges presented by today's 'random access' requirements. And it is here that Fostex's expertise from a lifetime in timecode DAT design for post production facilities to sound mixers through machines such as the D-20, D-10 and portable PD-4, really comes into its own.

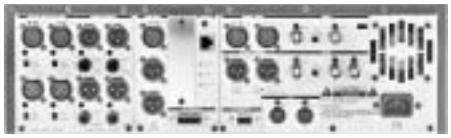
- Includes all the standard clock references of word and video and has the ability to resolve to incoming timecode and finish with a regenerated video sync output. That means the DV40 not only reads timecode and syncs to any external sync reference, but will also generate and regenerate timecode in all the industry standard formats.
- Built-in timecode generator will generate, regenerate and externally synchronize via 24 hour run, record run, free run and external run modes at all standard frame rates including the 23.97 HD camera mode.
- Timecode formats and LTC offset can be individually set for each new file and being able to switch between timecode and audio formats (with pull-up and pull down), helps solve tricky standard problems which can occur when using high-definition video.
- Offsets can be captured and adjusted on the fly while the display can be scrolled to indicate generator timecode, absolute time, disc timecode, external timecode and remaining time all with simple key presses. Plus with 1/3rd to 100x play speed, the reader will always keep up with external equipment.

DVD RAM offers some outstanding features over the other DVD technology families:

- Can store data safely for 30 years.
- DVD-RAM discs can be played without their cartridge if desired. The Cartridge provides an extra layer of protection and also stabilizes the disc in DVD-RAM drives.
- You can have upwards of 100,000 record and erase cycles on a DVD-RAM disc
- Uses redundant addressing and timing information, making the chance of losing data, rare.
- DVD RAM offers the Wobble Land and Groove recording format. This improves clocking and the actual writing (deep burning) of the data.

Interfacing Capabilities

The DV40 features an incredibly comprehensive set of analog and digital I/O interfaces on the rear panel—as one would expected from a machine of this caliber.



- Each of the four audio channels feature balanced XLR inputs individually switchable between +4dBu and –10dBV and the highest quality 24-bit/192kHz AD converters.
- The outputs are provided on both balanced XLR again with 24-bit/192kHz DA converters as well as 1/4″ unbalanced phone jacks for easy monitoring hook-up.
- Digital input and output for all four channels via the AES/EBU format (doubling up for 176.4/192kHz) on XLR connectors.
- Flexible headphone monitoring is provided on the front panel with switchable source selection for Track 1+2, Track 3+4, Track 1+3/2+4 and each track individually.
- Interoperability and system integration is taken care of via 9 and 15 pin RS-422 connectors, the latter also providing a 12V 400mA power source for future remote control applications. Both interfaces comply with the 9-pin protocol for VCR emulation and Fostex's own extended instruction set.
- The standard complement of timecode, video sync and word clock inputs and outputs are via BNC style connectors.
- 10Base-T/100-BaseTX Ethernet (RJ-45) port and a firmware-selectable GPI for remote control, fader starts and S-ID location rounds off the interface capabilities. The ethernet port allows the DV40 to transfer files via FTP in the media network.

DENON

DN-M991R • DN-M991RM

Continuous Duty MD Cart Player/Recorders



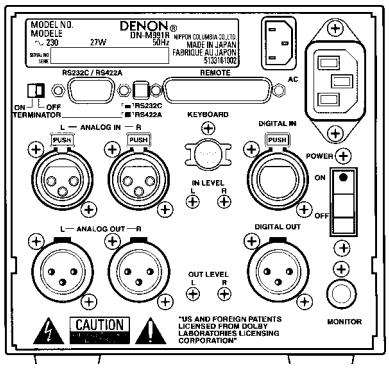
The DN-M991R and DN-M991RM are sleek MiniDisk decks that slide into the same 1/3- rack space as the professional Denon CD and cart decks, even using the same audio and remote connections. Ideal for broadcasting, features include 10-track hot start function, a buffer to prevent skipping and ±8% pitch control. They also have a pre-UTOC system that prevents loss of data in case of power failure, and they can be set to start a second deck while recording if the first disk becomes full. Identical in every respect, the DN-M991RM adds five on-air and production features; Visual E.O.M. record and mode lock out, five-minute skip search, high-speed title scroll and long title text.

FEATURES

Record Functions

- Record 74 min stereo, 148 min mono
- Record up to five cue points per track.
- Relay recording allows continuous recording to another DNM991R
- SCMS on/off
- In Auto Level Record function, they automatically start recording when an audio signal from a connected unit exceeds the preset level (-36 to -66dB, 6 steps).
- Denon's pre UTOC system is capable of detecting a power failure during recording and can play back the portion that was recorded immediately prior to the failure.

Inputs/Outputs



- Stereo or mono output
- Balanced XLR input/outputs (adjustable)
- AES/EBU digital input/output
- Switchable RS-232/422 serial port
- 1/4″ headphone output
- Optional RC-650 wired remote control

Editing Functions

- Basic editing functions include: Divide, Erase, Combine, Move, Track and Disc Name. Cue signals can also be edited (erased, rewritten, or added to later). There are two levels of undo and redo

Playback Functions

- Rotary track selector knob
- Next track reserve; allows a seamless transfer to your next selected track
- ±8% pitch control in 0.1 steps
- Fader start when connected to a mixer via its 25-pin parallel remote terminal.
- Play modes include single, continuous and Program (up to 25 tracks). Finish modes include Stop, Next, Rescue and Repeat.
- Cue search (up to 5 cue points per track)
- Auto Cue pauses where audio starts, not where the track starts. (Levels can be set between -36 and -72dB or turned off).
- END MON lets you instantly preview the end of the track, to ensure perfect “outs”. Monitoring starts can be set within a range of 5 to 35 seconds in 7 steps).
- End of Message (EOM) flashes at the end of a track. Warnings can be set within a range of 5 to 35 seconds in 7 steps).

Hot Start Functions

- Up to 10 tracks or cue points can be instantly played on demand. Ideal for sound effects, commercial spots or music.
- Tracks can be loaded into Hot Start by detecting the Auto Cue Level (selectable -72, -66, -60, -54, -48, -42 and -36 dB)
- Sound can be loaded into Hot Start from any track location.
- Loaded tracks can be replaced with new tracks.
- Seamless Loop can be used during Hot Start playback.
- Hot Start operation is possible using parallel remote, serial remote or a keyboard

Additional Features

- Sampling rate converter allows digital inputs of 32 and 48 kHz as well as 44.1 kHz.
- Automatic or manual track increment. In auto, they detect the silent portion of the program material and automatically increase the track number. Silent portion can be set within -36 to -66dB in 6 steps.
- Playback, recording, program input, editing and Hot Start via an optional keyboard.
- Slim chassis, 3 units fit in a 19″ rack shelf

DN-M991RM Step-Up Features

- Displays a visual end of message (EOM) allowing messages of next track cue, phrase cue, etc.
- Airlock prevents accidental recording even if the lockout tab on the disc is left enabled
- 5 minute Skip Search through long single track field recordings save editing time
- High speed title scroll for quick track and disc name recognition
- Long Name Text/Rescroll lets you leave the important part of the track name on the display for visual reinforcement. Also allows for easy rescrolling of disc and track names.

DN-M1050R

Professional MiniDisc Recorder



The DN-M1050R is a rackmountable MiniDisc recorder designed for all professional applications. Engineered for studio and broadcast environments, it includes a comprehensive list of editing functions and can be controlled by an external keyboard. Cue signals can be recorded (up to five points per track) at desired points on a disc for quick referencing. A cue tally signal is output from the parallel port at cue signals during playback and can be used to control other devices. Also, A-B sections can be erased or inserted at any point within a track. An optional Hot Start card allows up to 20 tracks to be loaded into memory locations for instant playback at the touch of a key on a parallel remote, serial remote or external keyboard. And an optional SMPTE card provides sync capability with VCR controllers—ideal for post-production applications mixing video with audio.

FEATURES

Record Functions

- Record 74min stereo, 148 min mono
- SCMS on/off
- Record up to five cue points per track.
- Auto Level Record function automatically starts recording when an audio signal from a connected unit exceeds the preset level (-36 to -72dB, 7 steps).
- Automatic track increment detects the silent portion of the program material and adds a new track with adjustable level sensitivity.
- U-TOC (User Table Of Contents) recording is selectable (Auto or Manual).

Playback Functions

- Instant start—playback starts less than 0.01 seconds after play button is pressed
- Play modes include single, continuous and Program (up to 25 tracks). Finish modes include Stop, Next, Rescue and Repeat.
- Cue search (up to 5 cue points per track)
- ±9.9% pitch control in 0.1 steps
- Auto Cue pauses where audio starts, not where the track starts. (Levels can be set between -36 and -72dB or turned off).
- END MON lets you instantly preview the end of the track, to ensure perfect "outros". Monitoring starts can be set within a range of 5 to 35 seconds in 7 steps).
- End of Message (EOM) flashes at the end of a track. Warnings can be set within a range of 0 to 35 seconds in 7 steps).

The DN-M1050R makes a perfect match when combined with the DN-C680 CD Player. Since its basic design is the same as the DN-C680, the DN-M1050R lends easily to the construction of a uniform operating environment.

Additional Features

- Basic editing functions include: Divide, Erase, Combine, Move, Track and Disc Name. Cue signals can also be edited (erased, rewritten, or added to later). There are two levels of undo and redo.
- Playback/record, program input, editing and Hot Start via an optional keyboard.
- Jog/shuttle dial for fast scans and frame accurate searches

Inputs/Outputs

- Unbalanced RCA inputs/outputs
- Balanced XLR inputs/outputs (adjustable)
- S/PDIF coaxial digital input/output
- AES/EBU digital input/output
- Switchable RS-232/422 serial port
- 25-pin parallel remote
- 1/4″ headphone output with level control

Optional Cards

ACD-26HM Hot Start Kit

- Up to 10 tracks or cue points can be instantly played on demand. Ideal for sound effects, commercial spots or music.
- Tracks can be loaded into Hot Start by detecting the Auto Cue Level (selectable -72, -66, -60, -54, -48, -42 and -36 dB)
- Sound can be loaded into Hot Start from any track location
- Loaded tracks can be replaced with new tracks
- Seamless Loop can be used during Hot Start playback
- Hot Start operation is possible using parallel remote, serial remote or a keyboard

ACD-25FSM
Sampling Rate Converter

- Digital input and output is possible at 32 and 48 kHz as well as 44.1 kHz

ACD-27MS
SMPTE Time Code Sync Unit

- Conforms to LTC time code format
- Time Code chase search
- Video Sync or Word Sync is selectable in preset mode
- Word Sync signal supports 44.1 kHz

DENON

DN-780R

Twin Output Auto-Reverse Cassette Deck Recorder



With its dual auto reverse decks, the DN-780R excels at a variety of dubbing and continuous playback applications. Each deck features amorphous heads, non-slip reel drive, Dolby B/C noise reduction and 12% pitch control. Extremely versatile, the DN-780R's twin output design makes a host of configurations possible, while an Auto-Standby mode ensures you never miss a second recording when cassettes reverse direction or when recording moves from deck A to deck B. Housed in a standard 19˝, 3RU chassis with an aluminum front panel.

FEATURES

- Quick Auto Reverse and non-slip reel drive components and permalloy heads
- Simultaneous recording on both decks
- Normal and double speed dubbing modes
- Independent return-to-zero, precision music search, tape counters, ±12% pitch control, auto tape selector, and fluorescent peak hold level meters for each deck
- Dolby B and C noise reduction system, Dolby HX Pro headroom extension
- An MPX filter eliminates the 19kHz pilot tone present in FM broadcast sources

Inputs and Outputs

- Unbalanced RCA I/O for each deck
- Two unbalanced 1/4˝ mic inputs on the front panel, with switchable Manual/Auto Gain Control, can be monitored with or without recording to tape
- Headphone output with selectable Deck A, Deck B, or Mix monitoring

Continuous Record and Playback Functions

- Independent dual deck functionality, coupled with twin inputs and outputs, allows for advanced playback and recording capabilities including:
  - ➢ Playback or record of two separate signals at the same time;
  - ➢ Playback on one deck while recording the same or a separate signal to the other deck;
  - ➢ Use the Relay Play/Record feature to cycle through both sides of deck A and then both sides of deck B for four sides of uninterrupted playback or recording
  - ➢ Multiple DN-780R's connected via the Cascade In/Out terminals' provide the ability multi deck relay recording and playback
  - ➢ Multiple synchronized recordings can also be made using one or more DN-780R's

Optional ACD-780 I/O board

- Provides independent XLR inputs and outputs for each deck
- Separate Input Select switches per deck facilitate selection between balanced and unbalanced XLR inputs
- A Mono Output switch gives you the option of +18dBu (CD) and +4dBu (Tape) mono output from the unit's Left channels or -10dBu mono output (CD and Tape) from the unit's Right channels

Remote Features

- Built-in Serial D-Sub 9 pin (RS-232C) connection; Parallel D-Sub 25 pin connection; and Mini TRS connection, support remote Play, Fast Forward, Rewind, Stop, Pause, and Record/Record Mute functions
- Remote control options include the RC-U620 hand held IR remote, the RC-41 wristwatch IR remote, and the RC-770TW wired remote

DRW-585P

Double Cassette Deck



The DRW-585 is an attractive and affordable dual-well, high-speed dubbing cassette recorder. It offers many convenience features that also enhance the performance. While only one transport is capable of recording, there are separate counters for both decks, and both are equipped with Dolby B and C noise reduction as well as HX-Pro to improve playback. Loaded with playback options, the DRW-585 has a replay button that, when activated, plays through both sides of the A deck, then automatically plays the B deck. Normal and double-speed recordings benefit from a manual bias adjustment that lets you set the bias according to your own preferences. Both transports are backlit so it is easy to see how much tape is remaining, or when you're passed the leader tape at the beginning of a recording.

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EQUIPMENT LEASING AVAILABLE

DN-T625/DN-T645

CD/Cassette Combo Decks



The DN-T625 and DN-T645 are professional combo decks that combine the functionality of a single well of the DN-T820R advanced quick auto reverse cassette deck with a Compact disc player that offers direct track access and 10 second shock memory. The CD player on both units are capable of CD and CD-R(W) playback with the DN-T645 adding the capability of MP3 playback. Both units feature a wide range of flexible I/O including unbalanced (RCA) and balanced XLR I/O as standard along with coaxial SPDIF digital output multiple external control options including RS-232C control that allows the units to to be controlled by remote systems such as AMX and Crestron.

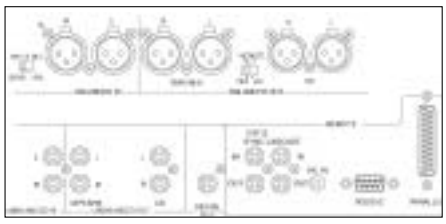
FEATURES

General Features

- Duplicates CD to cassette internally
- Cascade In/Out terminal set can be used to link several units together for extended continuous play
- Relay Playback function for continuous play between the CD and cassette decks

Both CD Players Feature

- Playback support for CD-R/RW discs
- Instant start / Cue to music; Stop, Next and Recue finish modes; Next Tr Reserve mode
- ±12% Pitch control in 0.1 steps
- Single and Continuous playback modes, and Program Play (up to 99 tracks)
- 3 repeat modes (Single Track, All Tracks, A-B) for continuous playback, as well as Random mode playback
- An 11 key keypad and a rotary knob are provided for direct track access and programmability as well as Frame Search and Back Cue functions
- Able to display CD encoded text through the use of a highly visible FL tube display
- 10 second Shockproof memory
- Adjustable user presets



Both Cassette Decks Feature

- Auto Reverse Full-Logic mechanism featuring a two motor quick reverse mechanism
- ±12% Pitch control
- Music search
- Dolby B and C noise reduction
- Single or Selectable playback cycles
- Memory rewind
- The REC Input source is switchable between internal CD (with CD deck sync recording) and external line input
- Cueing enhancements such as Memory Rewind (Return To Zero) and an Auto Reverse Full-Logic mechanism featuring a two motor quick reverse mechanism

Inputs/Outputs

- Stereo balanced XLR and unbalanced RCA inputs with selector switch for tape deck
- Stereo balanced XLR and unbalanced RCA outputs for the CD player
- Stereo balanced XLR and unbalanced RCA outputs for the the tape deck (switchable as a CD/tape mix output)
- Coaxial S/PDIF output
- Headphone output has a level control and source selector switch determines the output signal (CD, tape or mix)
- XLR Balanced Mono Output switch, which gives the option of +18dBu (CD) and +4dBu (Tape) mono output from the unit's respective Left channels or -10dBu mono output (CD and Tape) from the unit's respective Right channels

External Control

- AMX & Crestron compatible Serial D-Sub 9 pin RS-232C control port
- Stereo mini hard wire control compatible with Phillips RC5 control
- Options include the RC-U620 handheld IR remote, the RC-41 wristwatch IR remote, and the RC-620 wired remote

Additional Features

- A Power On Playback function can be used for automatic playback at a specified time when using an external timer
- Housed in a standard 3R unit with an attractive aluminum front panel with independent FL tube displays for both CD and cassette decks
- Optional AMC-22 lens cleaner

DN-T645 Adds

CD Player

- MP3 Playback
- MP3 Folder Search by ID3 Tag; File, Title, Artist & Album

Cassette Deck

- Automatic Gain Control recording
- HX PRO headroom extension
- Stereo 1/4″ TRS Mic inputs

External Control

- RS-422A
- DB 25 Parallel

DENON

DN-C635

Industrial CD/MP3 Player



An affordable CD player with an amazing feature set, the DN-C635 is equipped to take on the most demanding tasks in broadcast, DJ, production or installation applications. MP-3 capability, proprietary Super Linear Converter with 18-bit, 8x oversampling digital filter prevents left/right timing errors and zero cross distortion. Three customizable presets allow you to store over 25 of the DN-C635's functions for instant recall. Balanced-XLR and unbalanced RCA line outs as well as a coaxial digital output get you connected, while the large, easy to read fluorescent display will keep you on top of transport status and the multiple programming functions.

FEATURES

- 18-bit digital filter with 8X oversampling offers clear, crisp sound.
- Index Search, A-B repeat
- Instant start (within 0.01 sec from PLAY)
- Search accurate to one frame (1 frame = 1/75th sec.)
- MP3 playback (folder, artist, album, title)
- ±12% playback speed including MP3
- Auto Cue pauses a track where audio starts not where the track starts. The level at which sound is first detected can be set between -36 and -72 dB in 7 steps or off
- Three program modes (up to 25 tracks) and two power-on playback modes
- Play modes include continuous (whole disc) or single (track). Finish modes included Stop (after playing a track or disc), Next (standby at the beginning of the next track) and Recue (standby at the beginning of the same track)
- Rotary Track Select Knob allows easy track selection by 1x or 10x normal speed
- ±9.9% long stroke variable speed (pitch) fader is adjustable in 0.1% increments. Speed can also be adjusted ±3% in increments of 0.2% using a preset function.
- A bar graph display indicates playback points on the track being played. Elapsed time and remaining time displays are switched using the TIME button.
- End Monitor lets you instantly preview the end of the track, assuring a perfect "end point". Monitoring can be set from 5 to 35 seconds in 7 steps prior to track end.
- Sleep function stops the player if there isn't an entry for 30 minutes
- As a track nears the end, an EOM (End of Message) is displayed. When the warning begins can be set from 0 to 60 seconds in seven steps prior to the end of the track.
- 'Next Track Reserve' lets you select and reserve the next track to be played back. When the current track finishes, the unit plays back the reserved track.

Inputs/Outputs

- Unbalanced (RCA)and adjustable XLR balanced stereo line output. Also offers level controls for L & R balanced outputs on the back panel.
- Digital coaxial (RCA) output
- Stereo headphone jack with volume control
- 9-pin serial and 25-pin parallel connectors allows fader start operation from a mixer

DCM-280 *5-Disc Carousel CD Changer*



The DCM-280 is a very affordable CD changer that allows you to change four discs while one is playing. To ensure high quality, the DCM-280 is equipped with a Digital servo laser mechanism, 8x oversampling digital filter and multi-level DAC ensure high quality. Has a headphone jack and includes an infra-red remote control and rackmount kit (3RU high).

- 20-Selection Music Calendar Display
- 3-mode random playback (Full, Program and Disc Sequential Random)
- Combine up to 32 tracks from the five or fewer CDs which are loaded.
- Disc Skip button can rotate the carousel tray clockwise or counter-clockwise when searching for a disc. (Counter-clockwise rotation is operable only by remote controller.)
- Large, Easy-to-use-read FL Display
- 8x oversampling digital filter makes possible an extraordinary degree of attenuation and reduces frequency irregularities in the audio range to an absolute minimum.
- Advanced Multi-level Noise Shaping DAC removes jitter, and because it is built into the output amp, a clean analog output with suppressed high-frequency interference can be directly obtained.

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DN-C680

Industrial CD Player



The DN-C680 represents the best in CD technology developed by Denon for studio and broadcast production applications. A step up from the DN-C630, the DN-C680 adds Auto Edit and Auto Space functions, jog/shuttle dial and switchable RS-232 or RS-422 control (as well as parallel remote control). A sampling rate converter and SMPTE time code sync unit are also available as options. The DN-C680 also makes a perfect match when combined with the DN-M1050R MiniDisc recorder. Since its basic design is the same as the DN-M1050R, the DN-C680 lends easily to the formation of a uniform operating environment.

Same as the DN-C630 PLUS—

- Large fluorescent display and illuminated rubber buttons
- Jog/shuttle for frame accurate searches and fast, easy scans
- Switchable 9-pin RS-232/RS-422 serial remote for external control
- Auto Space function provides 4 seconds of silent space between tracks
- When recording CD to tape, the Auto Edit function automatically divides the total recorded time on the disc in half and rearranges the tracks so that they fit neatly onto the A and B sides of the cassette.

Optional Accessories

ACD-25FSC Sampling Rate Converter– converts digital output to32 kHz or 48 kHz as well as outputting 44.1 kHz.

ACD-27CS SMPTE Time Code Kit– Conforms to LTC format, Time Code Chase Search, selectable Video Sync or Word Sync



DN-C615

Industrial CD/MP3 Player

The DN-C615is a heavy duty, rackmountable single CD player with MP3 capability, ±12% pitch control, repeat functions, 4-second auto spacing, Auto Edit, and timer play. Ideal for aerobics and dance lessons, it can also be operated remotely via the optional RC-41 Wristworn Remote Control.

- 20-bit digital filter with 8x oversampling and noise shaper offer clear, crisp sound.
- MP3 Folder Search; File, Title, Artist & Album
- Pitch Control (±12%, 0.1% step)
- Auto space function provides 4 seconds of silent space between tracks
- Direct track selection via 11 numerical keys on the front panel
- Program Play (20 tracks), 20-track music calendar display
- Wen the DN-C615 powers up, it automatically plays the first track of a disc
- When recording CD to tape, the Auto Edit function automatically divides the total recorded time on the disc in half and rearranges the tracks so that they fit neatly onto the A and B sides of the cassette.
- Repeat function includes the entire disc or programmed selections
- A-B repeat function lets you repeat the designated starting (A) and ending point (B). Once you set the repeat phrase via the front panel, you can control Play/Pause with the optional RC-41 wristwatch.
- Time Counter - elapsed/remaining/total
- Unbalanced stereo output (RCA)
- Digital coaxial output
- Headphone output - fixed level
- 19″ rack mountable (2RU)

RC-41

Wristwatch Remote Controller

The optional RC-41 “Wristwatch” Remote Control is designed for aerobics instruction, dance lessons and other activities where the instructor needs remote control but can’t use a conventional wired remote. Operates seven commonly-used functions: Play, Pause, Pitch ±, Cancel Varied Pitch, Next Track and Previous Track.



DENON

DN-951FA/DN-961FA

Professional Cart/Tray CD Players



The standard for broadcast CD players, the DN-951FA and DN-961FA combine professional grade construction with a full complement of features. Identical except for their tray loading systems, they serve as workhorse CD players in thousands of radio stations and other professional installations around the world. The DN-951FA is the CD Cart version, offering top reliability when the CD media is subject to rough handling (in the DJ booth, control room, etc.) Housed in the secure cart, CDs are protected from mishandling, especially surface scratches. The DN-961FA is the tray loading version, featuring quick load/unload, just like a conventional CD player, the choice wherever the CD media isn't subject to abusive handling.

- Startup time is .01 second or less. A delay start of .1, .2, or .3 second is available via user preset. Startup via track flag or program material beginning is possible.
- End Monitor function allows tracks ending to be monitored. End monitor start position can be user-set within a range of 5 to 35 seconds from the end of the track.



DN-961FA

- A pulse encoder-type rotary selector is used for the track search operation. The tracks can also be selected in units of 10 by pressing the selector while turning it.
- Auto Cue and Back Cue functions:
  - — Auto Cue: Pickup is cued to the position within the track where the sound starts and then paused. The sound detection level can be set from -36 to -72 dB.
  - — Back Cue: When standby (STDBY/CUE) button is pressed during playback, the standby mode is set with the pickup at the position where playback was last started.
- Single and continuous playback modes
- Variable Pitch (0 to +3% in 0.2 second increments)
- Displays EOM (End of Message) indicating the track is about to end. Warnings can be set within a range of 5-35 seconds from the end of the track.
- Up to 28 presets to customize operation.
- Can read CD-R discs without final TOC, such as multi-session CD-Rs that have not been finalized
- Play start input connector and delay start function for use with mixing console fader.
- 9-pin RS-422A serial port for external control plus a 25-pin parallel port for the optional Denon RC-680 remote control.
- Auto Track Select (DN-951A Only) reads bar code info off the disc caddy for automatic "pick" track selections.

DN-H800

5-CD Changer with Tuner



Ideal for installation in bars, restaurants and clubs, the DN-H800 is a professional 5-CD changer with an AM/FM tuner. Features include RCA analog and balanced XLR outputs for CD and Tuner, CD pitch control (±12%, 0.1 steps), 5-way repeat, selectable timer switch operation, Intelligent Disc Scan and the ability to remove CD without interrupting current playback. In addition, the AM/FM/FM Stereo Tuner adds RDS (Radio Data System) functions including PS-program service name, CT-clock time, PTY-program type and TP-traffic program searches.

- Three way operation: via front panel, optional RC-H800 remote control, wired control terminal on rear panel.
- Three random playback modes (all disc random play; disc sequential random play or program random play) and 5-way repeat: single track; entire disc; all loaded discs; stored program sequence; random playback
- Intelligent Disc Scan allows the carousel tray to rotate clockwise or counter-clockwise when searching for a disc.
- Up to four discs can be exchanged while the fifth disc is playing

DN-C550R

Professional Dual Drive CD-R/RW Recorder



Now it is easier than ever to create CDs in one, all-inclusive package. A professional, rack-mountable CD and CD-R/RW deck the DN-C550R offers a wealth of playback and recording options. Using a mixer, you can play back both of the unit's two drives at once, or program tunes from each drive to be played in whatever sequence you choose. Equipped with professional features such as balanced XLR and coaxial S/PDIF inputs (SCMS defeatable), the DN-C550R is the perfect choice for high-quality CD recording.

FEATURES

◆ The analog XLR inputs feature full 24-bit A/D conversion for highly accurate recordings from analog sources, while the digital inputs use a digital sampling frequency converter to convert 32kHz and 48kHz rates to the CD standard 44.1kHz.

◆ Text input feature allows disc and track names to be entered for display on the easy-to-read front panel for instant recognition and simple documentation of discs.

◆ The DN-C550R is packed with features allowing its interface with many of the compact disc formats available today:

—Drive 1 can be used as a dedicated playback deck, accepting both pro and consumer CDs, CD-Rs, and CD-RWs, as well as supporting playback of HDCD encoded discs via the built-in HDCD decoder.

—Drive 2 functions as either a CD-R/RW recorder, or as a second playback drive same playback functions as Drive 1).

—Both drives are also PC/data disc compatible, while both analog and digital inputs of Drive 2 are SCMS defeatable.

◆ Dubbing can be done in real-time or at 2X speed for fast and seamless duplication (HDCD dubbing compatible).

◆ Several different recording modes are also available:

—Scan Rec Dubbing Mode allows selection of specific tracks for recording, while Disc Dubbing Mode records all information, including HDCD data and user-input text, from one disc to the other.

—In Synchronized Recording function, recording begins as soon as a signal is seen.

◆ A wide variety of playback modes are also incorporated into the DN-C550R.

—Repeat Mode lets you select one or all tracks from Drive 1 and/or 2 for repetitious playback.

—The Relay Play Mode gives automatic, continuous play from Drive 1 to Drive 2.

—Mix Program Play Mode supports programmed playback of one or both discs, and the Mix Random Play Mode plays random selections from Drive 1 and/or 2.

—Dual Play mode lets you play discs simultaneously in both drives. Since sound can be output independently from each drive, you can connect a mixer to enjoy mixing the sound.

◆ Selectable Copying (Dubbing) Mode

–All Track dubbing: Entire disc or programmed tracks are recorded (manual finalization)

–Make CD dubbing: Entire disc or programmed tracks recorded (auto finalization)

–One Track dubbing: Records selected track (manual finalization)

–Scan Rec dubbing: Tracks selected in Intro Scan are recorded (manual finalization)

◆ Selectable Synchro Recording Mode

—"Disc Sync": Recording is automatically triggered to copy entire disc when the external source player is started.

—"Make CD": Recording is automatically triggered to copy entire disc when the external source player is started.

—"1 Tr Sync": Recording is automatically triggered to copy selected single track when the external source player is started.

DENON DP-DJ151 ANALOG TURNTABLE

The DP-DJ151 allows you to record directly to CD-R, MiniDisc or hard disk through it's exclusive coaxial digital output. The output maintains a constant 16-bit/44.1 kHz signal, regardless of the rotational speed setting. This provides a quick and easy way to archive your vinyl digitally or to capture "samples" for studio remixing.



With the built-in RIAA Phono Equalizer, the DP-DJ151 allows for direct connection to any device that requires a "line level" RCA stereo input, like a cassette deck, sampler or any other audio device that doesn't have a built-in "phono" input.

MD-301MKII

MiniDisc Recorder



An affordable but durable MiniDisc recorder, the MD-301MKII provides maximum flexibility and reliability. Balanced XLR and unbalanced RCA ports offer easy integration with PAs or studios of any level. Digital S/PDIF optical I/O allows digital transfers, while another optical input on the front panel makes it easy to connect portable units without crawling into the back of the rack. For further convenience, there is a front panel keyboard input for easier track naming, offers TOC editing functions and includes a wireless remote control with 10-key for direct track access.

- 20-bit A-D and D-A converters and ATRAC v4.5 compression offer top sound quality
- Play modes include program (up to 25 tracks) and random; Repeat modes include one track, all tracks and A-B repeat.
- Monitor the input signal of the component connected to each input—before recording.
- In sync record, the deck starts recording when the input audio exceeds a certain level and pauses when the audio drops below a certain level for 4 seconds or more.
- When simple recording of long material is required, the MD-301MKII can be flipped to a 148 minute mono recording mode.
- Overwrite function lets you record music by erasing a previously recorded track
- Track information (start address, end address, title, etc.) is recorded in the U-TOC (User Table of Contents) area. Tracks can be edited by simply rewriting the data.
- Timer function lets you start/stop recording at a preset time (with optional timer)
- Equipped with balanced XLR and unbalanced RCA inputs and outputs as well as two S/PDIF optical digital inputs and one output. Has a headphone output with level control, and a front panel keyboard input.
- The front panel PS/2 port allows you to plug in an inexpensive keyboard for easier track naming, TOC (Table of Contents) editing functions and remote control.
- Includes wireless remote with numeric 10-key and is rackmountable (2RU high)



MD-350

Live MiniDisc Recorder

The MD-350 has all the features of the MD-301MKII plus it incorporates advanced performance functions, making it the best value for live venues, houses of worship, or budget conscious broadcasters. The MD-350 incorporates version 3 ATRAC encoding allowing up to 320 minutes of record time on a standard 80 minute MiniDisc, and it offers pitch control with a clear on/off switch on the front panel. Finally, cueing functions like Auto Cue and Auto Ready, as well as a complete complement of analog and digital I/O including a digital optical port on the front, make the MD-350 a must have for any theater or cue playback intensive situation.

Same features as the MD-301MKII PLUS—

- 12% pitch control function with a clear on/off switch on the front panel
- In LP4 mode, the MD-350 can record and playback for up to 320 minutes on a standard 80-minute Minidisc
- Auto Cue positions the machine to the first audio in a MD track, rather than the actual ID location. This eliminates dead space in the beginning of the track, allowing tighter cues for live shows or on-air environments.
- Auto Ready pauses the machine at each track ID allowing an engineer to cue in one track and forget about the machine - it automatically sets itself at the next cue. Also prevents the machine from accidentally rolling into the next cue on the MD.

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MD-801RMKII

Professional MiniDisc Recorder



The ultimate MiniDisc recorder, the MD-801RMKII is designed to meet the most demanding needs for theater, recording studios, and broadcast applications yet still offer the flexibility to integrate with simple installations. The MD-801RMKII provides all the amenities needed in live environments with advanced cueing functions, and a separate cue output for checking a signal without sending it to the main output. Plus, the optional BU-801 RAM buffer can be added for true instant start functionality. All the I/Os needed to make the connections are built in. The MD-801RmkII can be controlled via keyboard, RS-232 port or 37-pin parallel port. There are two wired remotes optionally available.

FEATURES

- Uses highest-quality 20-bit A-D and D-A converters and ATRAC v4.5 compression to offer top sound quality
- Offers thorough editing capability via the front panel buttons, or with a PC keyboard connected to the PS/2 port
- ±9.9% pitch control
- On the digital side, Word Clock In and Thru allow you to clock to another source on playback. This is key for integration into larger digital set-ups with a central clock source.
- Soft mute eliminates clicks when pausing and restarting
- SCMS Copy ID modes (copy prohibit, one generation limit, no copy protection)
- Built-in clock for time stamped recordings
- Optional BU-801 RAM buffer for true instant start—turns the MD-801RMKII into a powerful tool for assembly editing or sound cue drops.
- Balanced XLR main I/O with fine calibration controls; unbalanced RCA I/O; digital AES/EBU and S/PDIF coax digital input/output
- Analog output offers the additional ability to split the balanced XLR and unbalanced RCA outputs into “cue” and “main” outputs, so you can check your cues easily before sending them to your mix.
- PS/2 keyboard port for easy editing and remote control
- Headphone output with level control

◆ *Cueing functions on the MD-801RmkII make live productions easier:*

– Auto Cue positions the machine to the first audio in an MD track, rather than the actual ID location. This eliminates dead space in the beginning of the track, allowing tighter cues for live shows or on-air environments.

– Call returns the machine to the cue where play began. This is useful for setting cue points in the middle of a track.

– Auto Ready pauses the machine at each track ID. Ideal for live environments, this lets the engineer to cue in one track and forget about the machine - it automatically sets itself at the next cue. Also prevents the machine from accidentally rolling into the next cue on the MD.

– Incremental Play will automatically advance to the next track when play is stopped.

◆ *Multiple control ports allow the MD-801RMKII to be controlled by a variety of standards:*

– The PS/2 keyboard port allows transport control and editing functions from a standard PC keyboard.

– The RS-232 port allows serial control from any number of automation/control systems.

– The parallel port allows for simple remotes and tallies, including Fader Start for integration with broadcast consoles.

– Optional RC-8 (simple wired) and RC-801 (full-function) remotes

	MD-301MKII	MD-350	MD-801RMKII
Recording Time	74 minutes (Stereo)		74 minutes (Stereo)
Frequency Response	20Hz-20kHz ±0.5dB	20Hz-20kHz ±0.5dB	20Hz-20kHz ±0.5dB
Signal to Noise Ratio 94dB (A filter)	102dB Playback	>96dB Record (1kHz A-filter)	96dB Record (1kHz A-Weighted)
Wow/Flutter (WRMS)	Unmeasurable		Unmeasurable
Pitch Adjustment			±9.9%
Playback Rise Time			0.1 sec from Ready
Analog Inputs	XLR Balanced RCA unbalanced	XLR Balanced RCA unbalanced	CalibXLR Balanced RCA unbalanced
Digital Inputs	SPDIF Coaxial; SPDIF Optical	SPDIF Optical	AES/EBU; SPDIF Coaxial
Analog Outputs	XLR Balanced	XLR Balanced;	XLR Balanced;
	RCA unbalanced	RCA unbalanced	RCA unbalanced*
Digital Outputs	SPDIF Optical	SPDIF Optical	AES/EBU; SPDIF Coaxial
Dimensions	19 x 3.5 x 11.5″	19.3 x 3.5 x 11.5″	19 x 5.2 x 13.9″
Weight	12.1 lbs.	12.2 lbs.	17⅝ lbs.

RCA acts as separate monitor output for engineer cueing purposes

DA-20MKII

Master Quality DAT Recorder

An economical master DAT recorder, the DA-20MKII delivers solid, reliable performance with true CD sound quality. Well suited for home studios,"b" rooms and video suites, the DA-20mkII's S/PDIF digital and unbalanced RCA connectors allow integration with most of today's desktop video editors and digital audio workstations.



- Supports 16-bit recording at 48, 44.1 kHz sampling rates and 12-bit/32kHz in LP (Long Play) mode. LP doubles the amount of recording time available on tape for long sessions like sermons or conferences.
- Capable of SCMS free recording
- Start ID and PNOs (Program Numbers) can be written automatically or manually.
- Subcode functions include: Absolute and Program Time, Start ID, PNO, Skip Id, End Mark, lead-in area and renumbering.
- Program Numbers can be renumbered if accidentally inserted out of sequence.
- Auto ID level switching offers four selectable audio levels (-30/-40/-50/-60dB) for easier, more accurate start ID encoding.
- Sampling Monitor displays a visual confirmation of analog or digital source signals.
- Margin display with reset button continuously displays maximum peak level in dB.
- Titling and other text (take #, version #) of up to 60 characters per track.
- Self-diagnostic function warns of potential problems like condensation, irregular tape tension, dirty heads or a damaged tape.
- Last memory function retains auto ID and input selector after powering down
- Includes a wireless remote control

Inputs/Outputs

- Unbalanced RCA inputs and outputs, coaxial digital I/O, single stereo input level control with left/right balance

DA-40/DA-45HR *Professional DAT Recorders*



Found in recording studios around the world, the DA-40 and DA-45HR offer balanced and unbalanced analog and digital inputs/outputs to ensure compatibility with a wide range of audio gear as well as video editing equipment. Well constructed rackmount chassis, large easy to read display, and comprehensive setup menus complement their sonic excellence. And like the DA-20MKII they feature text editing capability, allowing tape and program titles to be inserted in the subcode of the tapes for cataloging and identification. The DA-45HR has all the features and functions of the DA-40—except it is equipped with 24-bit record capability—making it the ideal mixdown deck for today's high-end digital audio workstations (DAWs) and mixers.

All the features of the DA-20MKII PLUS—

- Shuttle wheel (up to 12x speed) for audible cue and review and refining locate points. Data wheel selects program numbers and changes menu and parameter settings.
- Margin display is measured in 1dB increments between 40dB and 20dB and 0.1 dB increments between 19dB and 0dB.
- Counter mode switches between Absolute Time, Program Time and time remaining.
- Two assignable locate points (with selectable 0 to 5 second pre-roll) memorizes precise locations for quick access to any point on a tape.
- Displays Program Number, Margin Mode, Digital Audio Frames mode and Block Error Rate mode.
- Auto ID level detection is selectable from -48, -56, -60, -66 and -72dB.
- Auto ID time setting has a range of 0.5 to 3 seconds in 0.5 second increments allowing precision Start ID placement.
- Analog input source selector allows either uncalibrated (allows variable level control) or calibrated (defeats variable level control with a constant +4 or -10 dB setting).
- SCMS is selectable to allow no copies, one copy or unlimited copies.

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DAT RECORDERS

- Record Mute can be set from 2-7 seconds.
- End ID markers can be set to be automatically written when recording stops. Auto-rewind can be activated when an End ID marker is recognized.
- Programmable repeat function lets you loop audio over specific points. Loops can be set to play 10x or indefinitely.

Inputs/Outputs

- XLR-balanced and RCA unbalanced analog inputs and outputs.
- Separate left and right input level controls
- Balanced outputs have a trim control ensuring compatibility with the widest range of analog audio equipment.
- XLR-balanced AES/EBU and coaxial S/PDIF digital I/O connectors. XLR-balanced and coaxial digital outputs can be reversed allowing S/PDIF signals to be output via the XLR output or vice versa (AES/EBU via the coaxial output).
- 15-pin parallel interface for automated control environments, fader starts and edit controllers.
- Optional RC-D45 wired remote control

DA-45HR Step-up Features

- Two record modes: High Resolution 24-bit (record up to 60 minutes on a 120 minute tape) and Standard 16-bit mode.
- 24-bit A to D and 20-bit D to A converters with dither provides accurate reproduction of the original source material.
- Programmable repeat function settable from 2 to 15 repeats to unlimited looping
- Digital out word length is selectable to 24-bit or 16-bit.
- Record mute time is selectable from 1 to 8 seconds in .5 second intervals.
- Matches I/O reference levels of analog equipment (+4dBu, -10dBV) to digital reference levels; -20 (SMPTE), -18 (European) and the default -16dB
- Word Sync In and Thru using BNC connectors ensures sample accurate sync between compatible digital devices.

DA-60MKII

Professional Timecode DAT Recorder



You won't find a better 4 head synchronizable DAT value than TASCAM's DA-60 MKII. With complete Sony P2 support, the DA-60 MKII behaves like a VTR and can be integrated with edit controllers and hard disk systems, making it the perfect DAT for post-production environments. More importantly, its onboard synchronizer and improved servo system means virtually instant lock-up times. And with advanced A/D and D/A converters, the DA-60 MKII delivers extremely high dynamic range and very high signal to noise ratios.

DA-302

Dual DAT Recorder/Duplicator



A unique dual mastering DAT deck with high speed dubbing and continuous recording capabilities, the 3RU high DA-302 can make an exact clone of any DAT tape including all subcode information. Flexible and easy to use with exceptional sonic quality and reliability, it is ideal for archiving, mixdown, duplicating and installation. Cascade multiple DA-302's to work in tandem.

- Two DAT decks work independently or in tandem. Recording on both decks simultaneously, allows you to create a master and back-up master in one pass.
- High speed (2x) and real time dubbing lets you clone any DAT including subcode and timecode information
- Continuous long format recording from Deck 1 to Deck 2 at 32kHz, records up to 8 hours on two 120 minute tapes
- Append dubbing allows dubbing from any position on the master tape
- Loop function when enabled starts the master deck after the last slave deck finishes continuous play/record operation
- Seqtime sets the start time for Deck 2 during continuous record. This feature extends to continuous recording when cascading multiple DA302s. Settable in 1 or 15 minute increments between 0H15M00S and 2H30M00S
- Independent S/PDIF digital and RCA analog I/O for each deck. An optional balanced analog I/O kit (LA-D302) offers stereo inputs and 2 sets of stereo outputs
- External control for multiple unit operation with analog and digital cascade outputs
- Optional WR-7000 synchro cable allows DAT clones from one DA-302 to another

TASCAM

102MKII

Professional Cassette Recorder

A precision built, rack mountable cassette deck with excellent sonic performance, the 102MKII offers outstanding value for musicians, system contractors, studio operators, schools and theaters. A three space rackmount design (rack ears included) makes it easy to integrate into any studio setup.



- Dolby B/C noise reduction and Dolby HX PRO extend high frequency performance up to 6 dB and minimize distortion
- HX Pro improves recording quality at high frequencies without increasing noise or distortion. Benefits any tape, regardless of the playback deck
- CPS search function finds the track you want by scanning up to 20 tracks at high speed until 4 seconds or more of blank tape is detected, then enters play mode
- Automatically chooses optimal settings for any type of tape — Normal, Metal or Cr02
- Bias Fine Tuning lets you customize high frequency response
- Left and Right balance control and master record level control
- Multi-counter with both 4-digit tape counter and realtime modes
- Record/mute autospacer automatically inserts 4 seconds of silence between tracks
- Intro Check gives you a 15 second preview of each selection before advancing to the next track.
- Multi function display indicates transport mode, tape counter position, Dolby system, tape type and peak level metering.
- Unbalanced RCA input/output
- 3.5mm CD/ Synchro-Dub jack
- Headphone output

202MKIII/302MKII

Dual-Well Auto Reverse Cassette Decks



The 202MKIII and 302MKII have all the great sound and convenience of the 102MKII plus they add bi-directional dual deck capability making them ideal for multi-purpose recording and dubbing as well as studio applications. Both their decks can record, so you can make two masters of the same source at the same time. You can also use double deck capability for extended recording/playback applications. And two mic inputs make them convenient for recording of groups —great for music teachers or business meetings. The 302MKII steps up with individual audio I/O for each well, pitch control, optical leader sensing, a wired remote control. For long format recording or playback, cascade audio and transport logic is included to automatically trigger multiple machines in succession.

All the features of the 102MKII in a dual deck design

- The decks can be used separately or in tandem during recording and playback.

– Play deck 1 while deck 2 records in either real time (1-7/8 IPS) or high speed (3-3/4 IPS) dubbing.

– Record simultaneously on both decks from an external master in Parallel Mode.

– Continuous mode lets you play/record both sides of Tape 1 and 2 in sequence (repeatable up to 5 times).

– Rev Mode determines when play/record stops: at the end of one side of a tape, after both sides of a tape are played, after both sides of both tapes are played or recorded.

- Teac compatible CD Synchro Dub jack
- Auto Reverse function reverses tape direction during playback and record.
- Synchro Reverse dub reverses the playback and record deck at the same time. Ideal when using different cassette lengths.
- Blank Scan skips long portions of tape (10 seconds or more) and fast forwards to the next section of audio
- Infinite looping during playback.
- Pre-program playback or recording, when used with a standard commercial timer.
- Independent L/R stereo LED meters
- Dual high impedance 1/4″ mic inputs on front panel allow live stereo recording

3-Head Cassette Deck



A 3-head cassette recorder that puts unprecedented value into your studio, the ruggedly constructed 130 virtually guarantees pristine recording and playback results with years of dependable operation and uncompromising sound. Features such as confidence recording, ±10% pitch control, return to zero (RTZ) function and auto tape select give you the extras you need, so you can work quickly and confidently.

Same features as the 102MKII (no CD Synchro-Dub Jack and CPS Search) *PLUS*—

- 3-heads provide confidence monitoring so you know exactly what is being recorded.
- Auto Monitor function allows signal monitoring to match the play/record status of the deck. During playback, tape is monitored. In Rec/Pause, input source is monitored. Record mode monitors off-tape.
- MPX filter eliminates pilot tones during FM broadcasts that can interfere with Dolby noise reduction.
- ±10% pitch control assures accurate playback speed of cassettes recorded on other decks. Makes matching pitch with other instruments a snap.
- RTZ (Return-To-Zero) fast winds the tape to the tape counters zero reference point.
- Timer record/playback capability
- Headphone output with level control



322

Independent Dual-Well Auto Reverse Cassette Deck

In applications where long-capture or playback is essential, nothing doubles your performance better than the 322. Two fully independent cassette decks housed in a 3U rack-mountable enclosure, the 322 elegantly adds a suite of dubbing and duplicating features to your arsenal. Ideal for musicians and contractors in a wide variety of applications including music studios, boardrooms, house of worship, live venues and more.

Same features as the 202MKIII *PLUS*—

- Separate RCA inputs and outputs, transport functions including RTZ function, level controls and metering for each deck.
- RCA line inputs and outputs for deck 1 can be used as common I/O for both decks
- Normal (1x) and high (2x) speed dubbing
- Optical leader sensing for quick auto reverse. When the tape approaches the leader tape from the cassette, it immediately flips the tape. This minimizes the time lost switching sides, making the 322 ideal for logging or long-form recording.
- ±10% pitch control for each deck
- RS-232 port allows serial control of the deck from various control systems. This is ideal for board rooms, lecture halls, or advanced home theater systems.
- Headphone output with level control and source selector (Deck 1, 2 or mix)
- Supplied RC-322 remote offers complete control of both decks.
- Optional LA-322 kit adds balanced XLR balanced inputs outputs to a deck.
- Up to ten 322s can be connected in series via the stereo RCA cascade outputs and 3.5mm external control I/O (with optional WR-7000 synchro cables) allowing:

— Sequential playback of multiple cassettes

— High speed simultaneous dubbing to multiple cassettes

— Recording an external source to multiple cassettes simultaneously

— Record separate sources to multiple cassettes simultaneously

112MKII / 112RMKII / 122MKIII

Master Cassette Decks

Production workhorses, Tascam's Master Cassette Decks are known throughout the industry for their outstanding performance, stability, and reliability in a wide range of applications. They each feature a parallel port, Dolby B&C noise reduction with HX Pro, an advanced Servo control system and more.

Ideal for production mastering and mixdown, the 112MKII features a parallel port for external control and with the optional LA112 balanced connector kit can be integrated into any studio. Stepping up with Auto reverse and a 3-head design that allows off tape record monitoring the 112RMKII provides exceptional playback for engineers and contractors.

The industry standard for production and broadcast facilities, the top-of-the-line 122MKIII features balanced and unbalanced input/output, gear and clutched input level controls precision servo direct-drive capstan motors, and a built-in oscillator adjustment signal with front panel bias and level calibration controls.



Top to bottom: 112MKII, 112RMKII, 122MKIII

They All Feature

- Input mode selects the input signal to be monitored at all times.
- Automatically selects the input signal to be monitored in Rec/Rec Pause modes or tape to be monitored in play mode. The 112RMKII and 122MKIII also allow off tape monitoring in record mode.
- LOC 1 and LOC 2 buttons each store a locate point giving you one button access to any two points on a tape.
- Repeat button allows looping between two points with one-touch convenience.
- Precision VU meters with peak LEDs
- ±12% pitch control.
- Front-panel 1⁄4″ phone jack line inputs take priority over the rear-panel inputs.
- Rec mute with auto spacer function.
- Dolby B/C noise reduction and Dolby HX Pro headroom extension system
- MPX filter removes 19Hz and 38kHz tones used in FM broadcasts that can interfere with Dolby B and C operation
- Return-to-Zero (RTZ) function

Inputs/Outputs

- 25-pin parallel ports provides external control of transport functions when used with the optional RC-134 Remote (122MKIII and 112MKII), or RC-112R (112RMKII)
- The 112RMKII and 112MKII can be upgraded for balanced operation with the optional LA-112 expansion kit
- Headphone jack with level control

112RMKII Only

- Auto Reverse has two modes; one-time mode plays both sides of a tape once, while continuous reverse mode plays both sides of a tape in a continuous loop.
- Direction button switches tape to other side for playback or recording
- Tape-end tally signal allows multi-deck operation for continuous recording or playback with two or more decks
- Compumatic Program Search goes directly to the beginning of a track by sensing 4 seconds of blank space in between 2 tracks.

122RMKIII Only

- Three-head transport with separate high-performance record and playback head.
- Equipped with balanced XLR I/O as well as unbalanced RCA I/O ensuring compatibility with all types of recording equipment.
- 400Hz/10kHz calibration oscillator and front-panel record level and bias trimmers. Bias and level are independently adjustable for the left and right channels.
- Hysteresis Tension Servo Control (HTSC) maintains consistent back tension on the tape all through the reel, eliminating wow and flutter.
- Precision FG servo direct-drive capstan motors provide the highest standards of reproduction quality and performance.

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ONE HOUR FREE PARKING
AT 349 W. 34th STREET (with purchase of $100 or more)

CD-160

CD Player



The CD-150 is a solid, straightforward, rackmountable CD player that easily integrates into almost any environment. A host of convenient features such as Auto Cue, 6% pitch control, timer play function, keypad and programmable playback, make the CD-150 ideal for audio and radio production, as well as DJ, fitness clubs and PA installations.

- Program and shuffle play modes. Program up to 32 songs to playback in any order. Visually review the program to ensure its accuracy. Repeat modes include one track, all tracks, A-B repeat.
- Auto Cue locates track's first audio–bypassing intro or subcode information. This allows a songs to start instantly.
- ±6% Pitch Control with clear on/off switch
- Intro Check plays only first 10 seconds of each track on the disc
- Time counter display is switchable between total disc playing time, elapsed time and time remaining of current track and total remaining time of disc.
- 10-key numeric keypad for direct access to any track
- Analog stereo (RCA) outputs—includes stereo RCA cable
- Headphone output with level control
- Supplied with wireless remote control
- Rackmountable (2RU high)



CD-450

CD Player

With great sound, rugged construction and a broad range of programming features, the CD-450 is designed for use in professional applications, including live theater, radio broadcast, production, DJ and sound contractor environments.

- ±12% Pitch Control
- Numeric keypad for direct track access
- Repeat modes include one track, all tracks, A-B repeat.
- Single Play mode stops the CD player at the end of the current track
- Program mode (up to 30 tracks)
- Auto Cue function cues any track up to the first frame of audio for immediate playback. Auto Cue threshold level is settable to -72, -66 -60 or -54dB
- Auto ready function, upon completion of the current track, cues the player to the start of the next track and enters pause mode ready for the next cue
- Call key returns to the last Ready cue position stored in memory
- Built-in timer allows unattended playback and stop control
- Incremental play begins playback of the next track every time the play key is pressed or will enter Auto Ready mode if the stop key is pressed
- Program material can be searched by track number, index number or time.
- Selectable display shows elapsed time, remaining time or total time remaining
- Can automatically trigger an external machine to commence playback facilitating smooth playback transitions across multiple machines
- Fader and Event start for recording applications such as flying in sound effects. The CD-450 will begin playing when it receives an audio trigger from an open fader or other audio event.
- Fader stop ceases playback from the CD-450 and automatically cues the player to the next track
- End Of Message indicator lights when nearing the end of a track and is settable from 5 to 35 seconds
- Back-up memory retains full operating status of the repeat, single and increment play, display mode, auto cue and search functions
- 2U rackmountable

Outputs

- RCA unbalanced analog outputs
- S/PDIF digital output
- Tally jack for connecting LEDs for monitoring the operating status of the deck
- Remote control output
- Optional LA-450 Balancing Kit gives you XLR balanced analog outputs with 12dB maximum trim control and balanced AES/EBU digital output.

CD-305

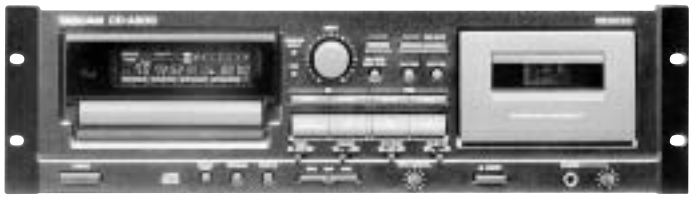
5-Disc CD Player



Multi-disc capability makes the CD-305 perfect for applications that require continuous playback or quick access to multiple discs. Discs can be played back in sequential, random, or program order with a 32 track memory. Ideal for use in commercial sound installations or production environments, equipped with balanced XLR and S/PDIF digital outputs for connection to professional, and studio equipment.

- Rotating carousel accommodates up to five CDs and is solidly built with a robust transport mechanism
- Bypass any disc in the tray (disk skip), you can also remove any unused disc while another disc is playing
- Programmable (up to 32 tracks), random and repeat playback modes
- 3-beam laser tracking system and CIRC error protection circuitry ensures exact reproduction of the original CD
- XLR-balanced and RCA unbalanced analog outputs as well as a coaxial S/PDIF digital output
- Includes wireless remote control with 10-key pad for direct track access

CD-A500/CD-A700 *CD/Cassette Combo Decks*



The CD-A500 and CD-A700 combine a CD player and cassette deck in one very affordable package. The two decks can act as totally separate units with individual outputs, or take advantage of the combination with direct dubbing, long play (combo) capabilities, and an input-saving cascade output mode. Both include unbalanced RCA inputs and outputs for the CD player and cassette recorder and each includes a wireless remote control. The CD-A700 adds XLR inputs and outputs, S/PDIF coaxial digital output (from the CD player) and 15-pin paralleled port for external control.

Cassette Deck Features

- 15-program music search
- Dolby B and C noise reduction
- ±12% pitch control
- Auto reverse mode allows continuous looping of both sides of a tape, play the tape once through or play one side only
- Record mute button inserts 4 seconds of blank space in between program material
- Two directional play buttons let you pick which side of a tape to play or record
- Counter reset and return to zero function

CD Player Features

- The CD player offers program (up to 20 tracks) and shuffle play, and continuous play of one track, or all tracks. Repeat playback includes 1-track/all-tracks/A-B
- Intro check function plays the beginning of each track for 10 seconds.
- A Side/B Side program dubbing feature allows monitoring the accumulated time of programmed material for even distribution to both sides of a cassette
- Auto spacing inserts 4 seconds of blank space in between program material.
- CD Sync recording allows one touch CD to cassette recording. Once the cassette is set to record, the CD player will start counter reset and return to zero function

Inputs/Outputs

- CD outputs can be shared as a common out for CD and cassette.
- Headphone output with level control
- RCA unbalanced inputs/outputs for cassette deck and input for the CD player

Combo Features

- Continuous play from CD-to-tape and tape-to-CD for long playback in background music installations
- Separate CD and tape function displays
- Includes RC-A500 IR remote control
- Timer engages a function (play CD/ tape, record tape) when the deck is powered up
- Rackmountable (3RU high)

CD-A700 Step-Up Features

- Parallel control port
- S/PDIF coaxial digital output from the CD player
- Independent pitch adjustment (±12%) for both CD and cassette
- XLR-balanced (and RCA unbalanced) I/O for cassette and inputs for the CD player

CD-A630

Triple-CD/Cassette Combo Deck



The CD-A630 builds on the CD-A500's feature set, adding a triple tray CD player with pitch control, parallel port, and optical digital output. This added flexibility makes it perfect for live shows, exercise studios, restaurant installations, and simple PA systems. The CD-A630 can operate continuously CD-to-tape and tape-to-CD, and discs can be changed individually without interrupting the disc that is playing. Uniquely, a 12% pitch control is provided for both CD and cassette playback to adjust tempo in dance and exercise studios.

Same features as the CD-A500 *PLUS*—

- Separate trays for each CD makes it very clear where a CD is loaded. This is great for live shows when switching between intermission music and show cues.
- Separate drawers also allow discs to be changed individually without interrupting the disc that is playing.
- Multiple tray operation provides more source material for playback—a key for public gathering areas,
- 12% pitch control on the CD player (as well as on the cassette deck) is key for performers, allowing minor adjustments in pitch and tempo to CD or cassette.
- Like the CD-A500, it is equipped with unbalanced RCA inputs/outputs for the cassette deck, inputs outputs for the CD player. Also offers S/PDIF coaxial digital output from the CD player
- Can be controlled from the supplied remote control or via its parallel port

CC-222 *CD Recorder /Cassette Combo Deck*



The CC-222 is the world's first CD recorder and cassette recorder combo deck, bringing the two most popular formats in one very affordable package. The two decks can be used independently (each have their own discrete set of unbalanced inputs/outputs) as well as for internal dubbing from cassette- to-CD and CD-to-cassette. The CD recorder adds S/PDIF coaxial (RCA) and optical digital I/O and as a bonus, a phono input is provided with the proper preamps and RIAA EQ circuits built-in. This allows direct connection of a turntable for creating CD or cassette masters from your favorite vinyl LPs.

Cassette Deck Features

- Dolby B noise reduction
- ±12% pitch control
- Auto reverse mode
- Counter reset and return to zero function.
- Record mute button (4 seconds)

CD Recorder Features

- Built-in sample rate converter (32-48kHz)
- Digital direct recording
- Sync One and Sync All CD record modes
- Erase and un-finalize CD-RW discs
- Adjustable Auto ID Sensitivity Level
- Selectable Copy ID (Free, Prohibit, SCMS)
- Adjustable auto track/Sync Rec level -72 to -24dB in 6dB steps
- Auto track increment (S/PDIF or Audio level)
- Adjustable fade-in/out (1 to 24 seconds in 1 second increments)
- Play modes include single, continuous, random, program (up to 25 tracks) and 3-way repeat (one, all and A-B) modes
- Timer play function engages the CD deck into play when the machine is turned on.
- Intro check function plays the beginning of each track for 10 seconds.

Combo Features

- Cassette to CD-R/RW or CD to cassette dubbing
- Rackmountable (3RU high)
- Includes wireless remote control
- RIAA phono input for recording vinyl LPs to CD or cassette.

Inputs/Outputs

- Headphone output with level control for selectable monitoring of tape, CD or mix
- Independent unbalanced RCA inputs/outputs for cassette deck and CD recorder
- S/PDIF coax and optical digital I/O (CD)

CD-RW4U

Compact CD-R/RW Recorder with USB Interface



The CD-RW4U is perfect for those who want to produce fantastic sounding CDs without sacrificing quality and durability. It is also space conscious. This allows it to be installed almost anywhere including personal studios and mobile systems, directly into a podium on stage, or in a restaurant or club.
The CD-RW4U features unbalanced RCA inputs/outputs, S/PDIF coaxial digital interface, wireless remote, digital level adjustment, Auto ID with adjustable sensitivity level, fades, and input monitor capability. It is also uniquely equipped with a USB interface. This allows direct connection to a Mac or PC and used as a CD-RW drive for archiving data to disc.

High Quality

- On-board 20-bit A-D and D-A converters for high quality audio capture
- Can also record to less expensive computer data CD-R and CD-RW media. Doesn't require "For Audio Only" CDs.
- Fade-in/fade-out (from silence) provides smooth beginnings and endings to tracks. Fade times can be set from 1 to 24 seconds in one second increments
- Auto ID function with adjustable threshold control. This allows the deck to identify the breaks between tracks without adding extra IDs in soft passages.
- To assure durability, the CD-RW4U uses a TEAC CD-RW computer mechanism instead of a modified consumer CD player mechanism. The drives are extremely robust, carrying a mean time between failure rating of 100,000 power-on hours.

High Performance

- USB connector allows direct connection to a Mac or PC for burning audio and data CDs. Bundled with Windows and Mac authoring software.
- Built-in 2MB RAM buffer for accurate sync starts and auto ID modes—ensures that even the first milliseconds of a track are recorded.
- Record mute function records four seconds of silence on the disc
- Sync record function allows recording to begin when a signal is received, and stopped when the signal ends. A signal level can be set between -24dB and -72dB in nine steps.
- Can override SCMS codes implemented by consumer players. This means you'll be able to make copies of your work.
- Three modes of copy protection (Free, Copy Once, Copy Prohibited) can be applied on a track by track basis to new discs.

Conveniences

- Playback modes include program (25 tracks), shuffle and repeat
- A digital gain adjustment is available for boosting low-level digital recordings,
- Track divisions can be entered manually, or produced automatically as a response to the input signal level.
- Monitor the input signal without entering record mode (Input Monitor function).
- In-line sample rate converter seamlessly accepts digital inputs from 32kHz to 48kHz. Connect to DATs, MiniDiscs or DVDs.
- Includes wireless remote control

Inputs/Outputs

- Unbalanced RCA analog I/O
- S/PDIF coax digital I/O
- Headphone output w/ level control

CD-RW700 *Professional CD-R/CD-RW Recorder*



A step-up from the CD-RW4U, the rackmountable CD-RW700 features 24-bit A/D and D/A converters for even higher resolution and offers a "Digital Direct" mode which bypasses sample rate conversion to give you an exact copy of the original audio. It also adds a S/PDIF optical I/O and a Timer Record/Play function. Like the CD-RW4U, the CD-RW700 can record to standard data CDs, override SCMS codes, and uses TEAC's extremely robust CD-RW computer drives.

Same as the CD-RW4U (no USB) *PLUS*—

- 24-bit digital-to-analog and analog-to-digital converters provide even higher recording and playback quality for the CD recorder.
- The CD-RW700 can use its internal clock to start and stop playback or recording at pre-determined times.
- "Digital Direct" mode

CD-RW2000

Professional CD-R/CD-RW Recorder



The CD-RW2000 is more than just an advanced CD recorder, it also serves double duty as an advanced CD player, making it ideal for live theater, radio production and broadcast as well as the recording studio.
Like the CD-RW700, the CD-RW2000 uses advanced 24 bit DACs, the same robust TEAC CD-RW drive, and offers the same simplicity of operation. However, the CD-RW2000 also offers a word clock input, has a complete complement of I/O ports, and features several cueing functions to make live productions easier.

Same as the CD-RW700 *PLUS*—

- Balanced and unbalanced analog I/O, SPDIF coax and optical digital I/O, plus AES/EBU digital I/O covers virtually every 2-track audio format available.
- Word clock input is provided for integration into large digital environments, allowing the CD-RW2000 to chase external word clock while playing a CD.
- Offers a parallel control port for transport controls, including a fader start function.
- Includes RC-RW2000 wired remote control
- *Several cueing functions on the CD-RW2000 make live productions easier:*
  - — Auto Cue positions the machine to the first audio in a CD track, rather than the actual ID location. This eliminates dead space in the beginning of the track, allowing tighter cues for live shows or on-air environments.
  - — Auto Ready pauses the machine at each track ID. This is ideal for live environments, allowing the engineer to cue in one track and forget about the machine - it automatically sets itself at the next cue. Also, for presentation and on-air applications, this is ideal since the machine won't accidentally roll into the next cue on the CD.
  - — Call is used to return the machine to the cue where play began. This is useful for setting cue points in the middle of a track. Just press PLAY to check the cue, then press CALL to go reset back to that cue point.

	CD-160	CD-450	CD-305	CD-A500	CD-A700	CD-A630	CC-222	CD-RW4U	CD-RW700	CD-RW2000
Frequency Response	20Hz-20kHz	20Hz-20kHz	20Hz-20kHz	10Hz-20kHz	20Hz-20kHz	20Hz-20kHz	20Hz-20kHz	20Hz-20kHz	20Hz-20kHz	20Hz-20kHz
Signal to Noise Ratio	>85dB	>90dB	100dB*	>90dB	>90dB	>90dB	>88dB	>88dB	>98dB	>98dB
THD	<0.02%	<0.006%	<0.03%	<0.02%	<0.02%	<0.02%	<0.012%	<0.012%	<.004%	<.004%
Pitch Adjustment	±6%	±12%	–	–	±12%	±12%	–	–	–	–
Analog Outputs										
Unbalanced RCA	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
Balanced XLR	–	optional*	✓	–	✓	–		–		✓
Digital Outputs										
S/PDIF (Coaxial)	✓	✓	✓	–	✓	–	✓	✓	✓	✓
S/PDIF (Optical)	–	–	–	–	–	✓	✓	–	✓	✓
AES/EBU XLR	–	optional*	–	–	–	–	–	–	–	✓
Analog Inputs (CD)										
Unbalanced RCA	N/A	N/A	N/A	N/A	N/A	N/A	✓	✓	✓	✓
Balanced XLR	N/A		N/A	N/A	N/A	N/A	–	–	–	✓
RCA Phono Inputs	N/A	N/A	N/A	N/A	N/A	N/A	✓	–	–	
Digital Inputs (CD)										
S/PDIF (Coaxial)	N/A	N/A	N/A	N/A	N/A	N/A	✓	✓	✓	✓
S/PDIF (Coaxial)	N/A	N/A	N/A	N/A	N/A	N/A	✓	–	✓	✓
AES/EBU (XLR)	N/A	N/A	N/A	N/A	N/A	N/A	–	–	–	✓
Dimensions	19x4x12	18.9x4.1x12.2″	19x15¼x5.2″	19x5.2x10.6″	19x5¼x10⅜″	19x5¼x10⅜″	19x5.4x11.6″	8.5x12.2x3.3″	19x12½x3⅞″	19x12.5x3⅞″

** requires optional LA-450*

TASCAM

CD-RW402

Professional Dual Tray CD Player/Recorder/Duplicator



A professional dual tray CD player/recorder/duplicator, the CD-RW402 combines playback, recording and duplication—in a single compact, rack mountable machine. Record, edit and duplicate CDs without compromising the performance of any of the CD applications. In fact, the CD-RW402 offers reliable functionality and great sound quality for a price that is much less than the cost of three separate machines.

The CD-RW402 offers advanced CD player functions like call, auto cue, and auto ready on both trays. A stutter scrub function allows for cue points to be set frame-accurately. The player deck also adds pitch control capability and two locate points. The record side includes traditional functions like record mute, digital gain adjustment, digital fade in/out, digital direct mode, and auto ID.

So go ahead. Capture a mix on CD-RW, edit the track, add CD-TEXT titles with a keyboard, dump the finished product to CD-R, and duplicate all in the same box. Nothing else can do this as easily, effectively and affordably as the CD-RW402.

FEATURES

High Performance

- 24-bit digital-to-analog and analog-to-digital converters provide excellent recording and playback quality for the CD recorder.
- Incorporating industrial-grade CD-ROM and CD-RW mechanisms, the CD-RW402 is a 1:1 duplicator of audio or data CDs at selectable 1x, 2x and 4x speeds, so you can quickly and easily make multiple copies.
- Disk At Once CD (4x speed disc duplication) can be used for Red Book glass-mastering and duplication, avoiding expensive pre-mastering work at the CD pressing plant.
- Combining the player and recorder together allows you to edit their original tracks. The A-B edit function uses the locate points on Deck 1 to define specific areas for copying to the record deck. A-B exclusive edit can be used to delete gaps in the middle of tracks.
- Powerful software emulates MiniDisc TOC (Table Of Contents) editing—virtual edits can be performed to CD1 and then executed to CD2 at up to 4x speed. Make custom mastering adjustments for your duplicated disks with controls to join, split and remove tracks. You can also adjust the incoming volume level of both analog and digital sources, create custom fade ins/outs (1-24 seconds), and mark indexes the way you want.

Front Panel Operation

- Operation is by means of the front panel, featuring an easy-to-use menu system, and the included wired remote control unit.
- Front panel includes independent displays for each deck offering individual level meters and 12 character text displays.
- Independent displays keep clear which deck is being monitored - a great feature for live and broadcast applications. Front panel also has 1/4˝ TRS headphone jack with source selection and level controls.

Inputs/Output

- Features a host of independent audio inputs/outputs for each deck. Includes unbalanced RCA and balanced XLR analog inputs/outputs as well as S/PDIF optical and coaxial digital inputs and outputs.
- A parallel control offers remotes and tallies with custom control applications. Also includes RC-RW402 wired remote control.
- Entering CD-TEXT information is simple–just plug a standard PS/2 keyboard in the front for track naming and editing
- If a common output is desired, a dedicated pair of unbalanced RCAs offers a common audio feed.
- Monitor the input signal without entering record mode (Input Monitor function)

Additional Features

- ±9.9% pitch control on CD player. Pitch effects can be made on the transfer as well, committing the effect to the new disc.
- Sampling frequency convertor allows the recording of CDs from digital audio sources with frequencies other than the CD standard sampling frequency of 44.1 kHz.
- Disc track divisions can be entered manually, or can produced automatically as a response to the input signal level.
- Built-in RAM buffer ensures that even the first milliseconds of a disc track are recorded.
- Independent jog dials on each deck. Jog function with preview allows precise setting of cue points
- Cue function allows instant return to the previous start point. Stutter scrub function allows for cue points to be set accurately.
- A multi-function “multi dial” is used to set and confirm parameter settings.
- Playback functions include continuous play, shuffle play, program play (25 tracks)
- Advanced cueing functions including call, auto cue and auto ready (see description on previous page).
- Three modes of copy protection (Free, Copy Once, Copy Prohibited) can be applied on a track by track basis to new discs.

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EQUIPMENT LEASING AVAILABLE

CD-D1x4

High-Speed 1x4 CD Duplicator



The CD-D1x4 is an affordable solution for musicians, independent record labels or anyone else that requires multiple copies of their audio CDs or CD-ROMs. It implements sturdy TEAC CD-ROM and CD-RW drives for years of reliable performance.

Using the CD-D1x4 couldn't be more simple. Just pop the disc you want to duplicate into the master drive on top, then insert your blank CD media into the slave drives on bottom. In just a few button pushes, you can set the duplication speed and start burning discs! Plus, the CD-D1x4 is capable of duplicating more than just music CDs. CD-ROMs with software and other computer files can be easily replicated. For applications that require larger duplication runs, two CD-D1x4s can be cascaded to act as a single 1x8 replicator.

- Creates four exact duplicates of a CD at once
- Burns CDs at up to 16x regular speed for high-speed duplication runs
- Disc Analyze and Disc Verification features
- Duplicates audio CDs and CD-ROMs using CD-R, CD-RW, CD-R-DA, and CD-RW-DA media.
- Uses reliable, industrial grade TEAC CD-ROM and CD-RW mechanisms
- Cascade port allows two CD-D1x4's to be chained for duplicating up to eight discs at at a time
- Standard 19″ 4U rackmount design

DV-D6500

RS-232 Controllable DVD Player



The DV-D6500 is an affordable RS-232 controllable DVD player, ideal for residential, commercial and institutional installations. Its wide range of readable media and programming capabilities makes this useful in numerous applications.

In Video DVDs, Super Video CDs and regular Video CDs, the DV-D6500 offers crystal clear picture quality. The true Progressive Scan output option far surpasses imaging of "line doubling" DVD players, with higher resolution and jitter reduction. RS-232 control on the DV-D6500 offers extensive transport control, machine set-ups, counter data, and disc format data. Includes the RC-6500J wireless remote control.

- RS-232 Bi-directional serial control port
- Plays DVD Video Discs
- Plays Super Video CDs and audio CDs
- Reads CD-R, CD-RW and DVD-R media
- Composite, component and S-Video outputs
- Unbalanced RCA analog outputs
- Coaxial and optical digital outputs
- Format adjustments for 4:3 and 16:9
- Progressive scan for improved image quality
- FVPP Image Processing (Gamma, Sharpness, etc.)
- 2-U Rack Mountable DVD Player
- Slow Motion Playback
- Frame-by-Frame Playback
- Program & random play (Audio & Video CD Only)
- Repeat Play Mode (All Formats)
- RC-6500J Wireless Remote Included

RS-232 Control Features:

- Transport control & current status info
- Disc format information
- Counter information (Track, Disc)
- Direct Track Number Access

TELEX

COPYETTE EH SERIES



1x or 3x Mono and Stereo Duplicators

One of the best values in cassette duplication, the popular Copyette EH series produce high quality, low cost cassettes in large quantities at nearly 16x normal speed. This means you can reproduce both sides of a C-60 tape in less than two minutes. Available in two versions, the Copyettes are capable of duplicating either one cassette (1-2-1) or three (1-2-3) at a time. In addition, each are available in both mono and stereo configurations.



They couldn't be easier to use. You simply insert the cassettes, press the START switch and they do the rest. They rewind all tapes to the beginning, copy, then rewind to the beginning again before stopping. The whole process can be stopped at any time by pressing the CYCLE button. Side Select feature allows you to set them up to copy one side of a tape or both sides at once. Copyette EH duplicators provide the quality and reliability you demand, and the speed you need, all at the affordable price you want. They offer one touch copying (tape rewinds to the beginning, copies, then rewinds again), erase heads (allows cassettes to be reused, saving you money), as well as the ability to copy both sides of up to three cassettes in mono or stereo at 16x normal speed. In addition, all Copyette EH duplicators include a track select feature, optical non-reflective end-of-tape sensing system (for gentle tape handling), a carrying handle for portability, a removable power cord and a protective hard cover for storage.

Stereo Copyette 1•2•1

Weighing only 9 lbs. this unit has a durable, impact resistant housing and includes a removable power cord, handle and protective cover. It also has an optical, non-reflective end-of -tape sensing system that provides gentle tape handling. A mono version is available.

Mono Copyette 1•2•1.....**419.95** **Stereo Copyette 1•2•1** **679.95**

Stereo Copyette 1•2•3

This duplicator copies both sides of three cassettes at once, yet is as small as the 1•2•1. It weighs only 14 pounds and includes a hard cover for protection when not in use. It uses all DC Servo motors for the ultimate in reliability . A mono version is also available.

Mono Copyette 1•2•3.....**1049.95** **Stereo Copyette 1•2•3** **1649.95**

TASCAM 3000

High Speed Cassette Duplicator

A must have in any studio or house of worship requiring high quality cassette duplication, the T-3000 provides 4x double sided duplication (effectively 8x duplication) of two cassettes within a single unit. For larger duplication setups, up to ten T-3000s can be slaved together.

- 1-to-3 high speed cassette duplicator
- 4-track/4-channel, ferrite record/play head and a 1/2 track/2 channel inline, ferrite erase head
- Auto switchable between Normal Type I and CrO2 Type II tapes
- Tape speed accuracy of ± 1 %



- 7.5 ips or 4x speed duplication of both sides, simultaneously copying a C-60 cassette in 7.5 minutes
- Up to ten T-3000s can be cascaded for larger duplication setups
- Well-3 can act as master or slave in larger duplication set-up
- Rugged design with easy access to mechanisms for cleaning and demagnetizing
- Uses standard RCA and MIDI plugs to connect units together

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ONE HOUR FREE PARKING
AT 349 W. 34th STREET (with purchase of $100 or more)

XGEN SERIES

Mono and Stereo Audio Cassette Duplicators



Designed for high performance and high production, the XGEN series also offers easy maintenance and unsurpassed ease of use. Available in two-channel mono or four-channel stereo versions each produces 3 copies from a cassette master at 8x or 16x normal speed and each can expand up to 67 copy positions (with additional copy modules).With extra modules duplicate up to 67 copies of a C-60 original in under two minutes. And they copy both sides at once. The XL Series feature "Extended Life" cassette heads for increased performance and wear characteristic as well as improvements in wow and flutter, frequency response, S/N ratio and bias. Additionally the ACC4000 XL allows for either chrome or ferric cassette duplication. All models feature microprocessor controls with direct drive motors in all positions, Ferrite erase heads, a C-60 rewind speed of 30 seconds, track select, modular cassette pockets, end of tape sensing, auto or manual operation, cleaning cycle, audible tone end of copy and two-year limited warranty.

FEATURES

- One Touch Copying
- Individual rotary level controls
- Peak reading LED indicators for quick and accurate monitoring of audio levels
- Side A or A/B select button
- Stop all tapes instantly, at any point during the copy or rewind cycle.
- Automatic or manual selection of rewind and copy operation:
- Slanted work surface and unique "heads-up" cassette platform allow less oxide build up on the heads and makes cassette loading and unloading much easier.
- Short tape indicators alert you if a tape stops or jams before the original ends
- Each cassette position has a three point tape guidance system to eliminate skew problems. Plus, when a tape is inserted, each cassette position is activated to prevent unnecessary wear and tear on the tape head mechanism.
- Audio and bias, along with head adjustments, are done from the top of the unit and a switch on the back engages the head and pinch roller for convenient cleaning.
- Microprocessor-controlled design means fewer moving parts and direct drive motors means quiet operation, more accuracy and no gears to break down.
- Two year limited warranty

Copy Module Features

- Each module has four copy positions with erase heads and controls for side select
- LED displays indicate end-of-tape status for each pocket

Master Duplicator Features

- Produce copies from a master at 30 ips (16x normal speed)
- Up to 67 simultaneous tape copies with optional copy modules
- Track select, short tape indicators, auto/manual operation
- Erase heads in the copy positions erase existing audio as new material is being recorded
- Two master stations can be linked together for added flexibility (i.e. additional 3-pocket copying)

PRICING

XGEN Mono Master Duplicator **1499.95**

XGEN Mono Master Copy Module.....**1499.95**

XGEN XL Mono Master Master Duplicator
With XL (Extended Life) Heads **2135.00**

XGEN XL (Extended Life)
Mono Master Copy Module.....**2135.00**

XGEN Master Stereo Duplicator.....**2209.00**

XGEN Stereo Slave Module **2209.00**

XGEN XL Stereo Master Duplicator
With XL (Extended Life) Heads **2899.00**

XGEN XL (Extended Life)
Stereo Slave Module **2899.00**

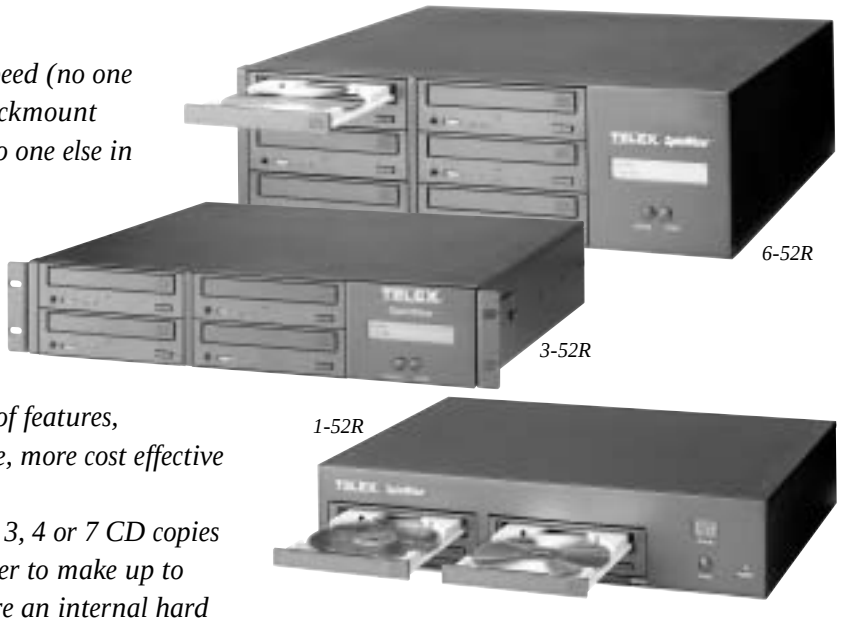
REPLICA *Mono Duplicator*



At 7 lbs., the Replica is the smallest, fastest copier you can buy. It copies both sides of a cassette in mono at 16x normal speed, and offers a full set of functions including stop, rewind, copy and interrupt, all operated by a single lever. The Replica features durable, impact-resistant housing and includes a recessed handle and removable power cord. An optional carrying case is available as well.

SPINWISE SERIES RACKMOUNTS

52x CD Copiers



Telex's SpinWise Series of certified 52x recording speed (no one is faster) CD Copiers includes high performance rackmount and tower units designed for ease of use at prices no one else in the industry can duplicate.

SpinWise Rackmount models offer the ability to make 1, 3 or 6 CD copies at once, at 52x recording speed. That's enough power to make up to 180+ copies an hour! The 6-52R features an internal hard drive, while the 1-52R and 3-52R do not. The 6-52R offers a full complement of features, while the 3-52R and 1-52R provide an easier to use, more cost effective alternative to copiers with internal drives.

SpinWise Tower models offer the ability to make 2, 3, 4 or 7 CD copies at once, at 52x recording speed. That's enough power to make up to 215+ copies an hour! The 3-52H and 7-52H feature an internal hard drive, while the 2-, 3-, 4- and 7-52NH do not. The 3-52H and 7-52H provide a full complement of features, while the 2-, 3-, 4- and 7-52NH offer an easier to use, more cost effective alternative to copiers with internal drives.

All Rackmount and Tower SpinWise CD Copiers Feature

- 52x recording speed
- Disc at Once support for Audio
- Support for multiple CD formats
- Multiple fans for maximum cooling
- User-friendly interface for immediate integration into any environment
- Adjust recording speeds of writers and system to 4x, 8x, 16x, 24x, 40x, 48x and 52x
- Update the firmware in the system by loading a CD into the master reader drive; the system will automatically update itself
- Support all business card CD format sizes

SpinWise CD Copiers with Internal Hard Drives (Only) Add

- Support for CD+G (Compact Disc + Graphics) and Karaoke CD formats
- Audio track extraction for music disc compilation
- Menu (Status) Commands:
  Copy, Copy and Compare, Verification, Emulation, and Load to Hard Drive
- Sub Functions:
  Setup, Hard Drive Partition, CD to CD, Start Writers, Write Method, Write Speed, CD ROM Drive, CD DA Load Speed, Eject Disc, Buzzer, Edit, Copycomp Eject, Skip Read Error, CD+ G, Startup Command, Read CDDA Index O
- Original does not have to be loaded onto the hard drive. If you leave original in the 6-52R's or 7-52H's reader drive, for example, they will act as a one to five or one to six CD copiers, if desired.

- The 1-52R offers one to one disc duplication at 52x speed. Designed for low volume applications (30+ copies per hour), it supports multiple CD formats, as well as Disc at Once support for audio. Approximate recording time is fewer than two minutes per full disc. Includes rack ears.
- The 3-52R features one to three disc duplication at 52x speed. 19″ rackmountable configuration with space saving 2U height. Supports multiple CD formats, including CD+G Karaoke, as well as Disc at Once support for audio. Approximate recording time is under two minutes per full disc. Includes rack ears.
- This 6-52R features an internal hard drive, with six copy drives, a 52x recording speed, and the ability to make 185+ copies an hour. 19″ rack-mountable configuration/3U, Disc at Once support for audio and support for multiple CD formats including CD+G, Karaoke. Audio track extraction for music disc compilation. Copy, Copy and Compare, Verification, Emulation, and Load to Hard drive are supported. User-friendly interface for immediate integration into studio/production environment.

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EQUIPMENT LEASING AVAILABLE

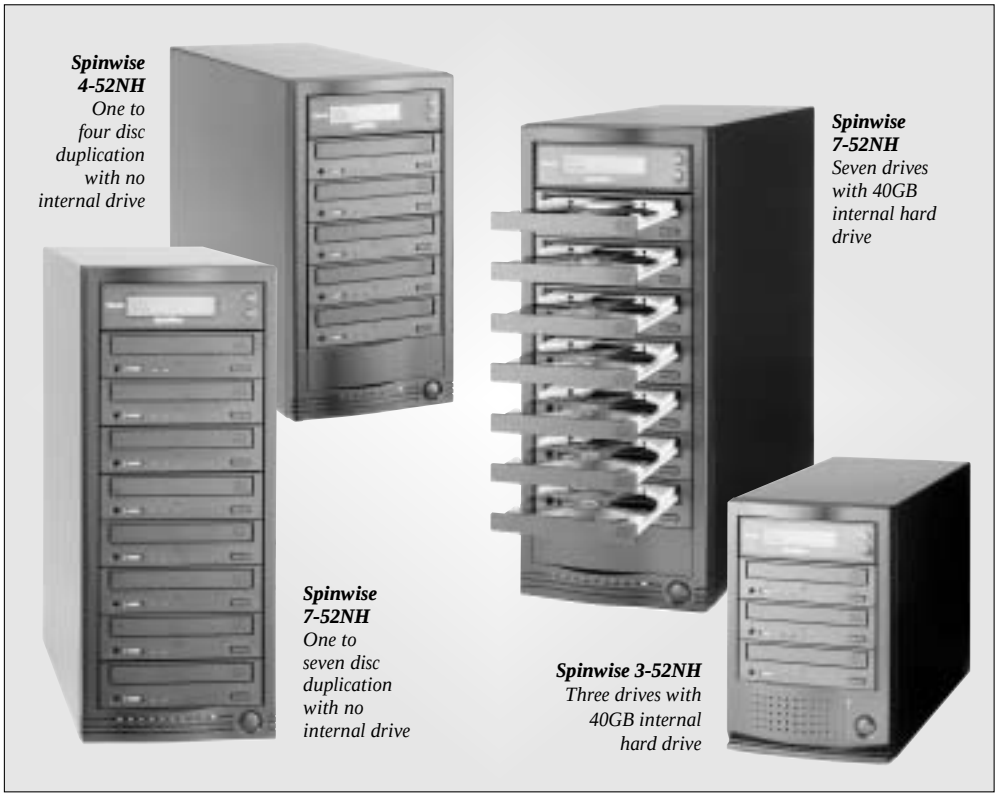
SPIN WISE SERIES TOWERS

Spin Wise Tower CD Copiers without Hard Drive

These towers feature 2, 3, 4, 5 or 8 drives, a reader/ master drive and 2, 3, 4 and 7 copy drives, for one to seven disc duplication at 52x speed. Makes up to 215+ copies per hour. They support multiple CD formats, including Disc at Once support for audio.

Spin Wise Tower CD Copiers with 40GB Hard Drive

These towers feature an internal hard drive, with three or seven drives, a 40x recording speed, and the ability to make 120 or 215 copies an hour. They support multiple CD formats, including Disc at Onc, CD+G, and Karaoke. Audio track extraction for music disc compilation. Copy, Copy and Compare, Verification, Emulation, and Load to Hard drive are supported. The original disc can be left in the reader drive rather than loaded to hard drive, to utilize the unit as a one to two or one to six disc copier if desired.



About CD Recording Time:
As music is sampled at a continuous rate for recording to CD, a typical CD (650 MB) can contain 74 minutes of audio. Even though many recordable CDs are used for data-type purposes, most people measure record speed in multiples of 'real time'. 1x record speed means that it takes 74 minutes to record ("burn") 74 minutes of audio. 12x speed means: 74 / 12 = 6.166 minutes.

How do you make MP3 files from CD audio tracks?
You need to get conversion software or an encoder of some kind. These programs allow you to take the files from the master CD, encode them to the specifications of your choosing, and store them on your hard drive for later use.

What is finalizing and what does it do?
When you burn a CD-R, you have a choice to either burn the disc as an open session or a closed session. As an open session, you are able to add to the disc at a later time. However, open session discs cannot be played on an audio CD player. For playable audio, you need to close the disc, a process that is called "Finalizing." Once closed or finalized, you cannot add any more information to the disc. When you finalize, you are creating the TOC (table of contents) within the lead in. The lead out is also created, thus closing the disc. You are also finalizing the disc when you use disc-at-once recording.

RackMount Copiers with Internal Hard Drive

SpinWise 6-52R: 52x recording speed, 180+ copies/hour 19˝ rack-mountable configuration/3U 40GB HD 26 Partitions, 900 MB each

RackMount Copiers without Internal Hard Drive

SpinWise 3-52R: 52x record speed, 90+ copies/hour 19˝ rackmountable configuration with slimline 2U height. Supports multiple CD formats

Tower Copiers with Internal Hard Drive

SpinWise 3-52H: 52x record speed, 92+ copies/hr tower configuration, supports multiple audio and data formats. 40 GB HD, 44 partitions, 900 MB each.

SpinWise 7-52H: 52x record speed, 215+ copies/hr tower configuration, support for multiple audio and data formats. 40 GB HD 26 Partitions, 900 MB each.

Tower Copiers without Internal Hard Drive

SpinWise 2-52NH: One to two CD copier. 52x speed, 61+ copies / hr.

SpinWise 3-52NH: One to three CD copier. 52x speed, 92+ copies / hr.

SpinWise 4-52NH: One to four CD copier. 52x speed, 123+ copies/hr.

SpinWise 7-52NH: One to seven CD copier. 52x speed, 215+ copies/hr.

SPINWISE CDR100A

Automated CD Copier



A standalone robotic system, the SpinWise CDR100A features 52x record capability, 100 disc capacity, and the ability to reload media on the fly. Supports multiple CD formats, as well as Disc at Once support for audio. Easy one button operation includes start, pick and sequence commands. Reliable pick and place design prevents blank CDs from dropping. Low cost, high performance, and ease of use makes the SpinWise CDR100A ideal for SOHO (Small Office/Home Office) or production/studio environments.

- Best price, throughput and input capacity
- 52x record speed for maximum throughput
- Standalone automated robotic system (no computer required)
- Disc to Disc duplication with one button operation
- 100 disc capacity, with ability to reload media on the fly
- High performance fan for maximum cooling
- Sleek desktop footprint for virtually any environment, including front office
- Support for multiple CD formats
- Pick and place design with proprietary arm configuration prevents CD from dropping
- Unlike gravitational feed designs, pick and place lets you work with multiple disc manufacturers. The only true constant between manufacturers is the circumference of the inner hub of a CD-R disc, which makes the pick and place feature more user-friendly.
- Bad media/CDs are automatically ejected for unattended operation

SPINWISE DVD



DVD Tower Copiers

SpinWise DVD Tower models offer the ability to make from two to seven CD or DVD copies at once. DVD copies are made at 4x speed, CD-R copies are made at 16x speed. All units are standalone towers with a 40 GB hard drive. Why wait up to 2-3 weeks, and pay up to $40 for a DVD copy, when it's this easy to own your own copier? Ideal for weddings, memorial services, church services, videographers, and corporate training seminars—just some of the applications where DVD can simplify production and distribution.

- 4x speed for DVD copies
- 16x speed for CD-R copies
- 40 GB internal hard drive
- Backwards compatible (i.e. can copy DVD-R, CD-R, CD-ROM, CDR-W)
- Multiple industrial fans provide maximum cooling and protect your data/video
- User friendly interface for immediate integration into any environment
- An easy way to back up your current computer's hard drive...a great storage tool
- Original does not have to be loaded onto the hard drive. If you leave original in the 7-416D's reader drive, for example, if will act as a one to six DVD copier, if desired

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PRINTWISE 1500 & 4500

4800 dpi Automated Print/Duplication Systems



Fully self-contained CD or DVD print and record stations. The ability to send multiple jobs, walk away, and come back to a fully printed and recorded set of CD/DVDs, makes these units a must have. You stack them, they do the rest. You can now ask to do 10 copies of Sundays sermon with graphics, then ask for 10 of Thursdays Choir Practice, 50 backups of internal documents, and so on... the PrintWise 1500 and 4500 do it all. Need two or four copies done simultaneously, no problem, they have data streaming to make the drives mutually exclusive of one another. PrintWise 1500 offers configurations up to two CD or DVD recorders, the 4500 up to four CD or DVD recorders. You even can have two CD and two DVD recorders in the same device!

FEATURES

- Batch Mode lets you send several jobs to the machine at one time. For example, if you want to do 20 CD/DVDs of one graphic and data, then do 50 of another, 25 of another...you would set up each in the software and the unit would automatically switch to the new job when the time came. Being the input capacities are 100/120 respectively, you could set up jobs to match those inputs.
- Networking capability allows multiple users from different computers within the company.
- Network in conjunction with the ability to "Data Stream" —sending individual jobs to a set of drives or single drive. This unique feature allows two users to use the machine at one time. One sending a job and using drives 1 and 2, the other using drives 3 and 4, as long. Most commonly used in the system that houses both DVD and CD drives, where some want to run a DVD job and CD job simultaneously.

PRINTWISE 4800

4800 dpi CD DVD Ink Jet Printer



Put 24-bit graphics directly onto your CD or DVD in amazing time. Immediate dry times, and immaculate results, the PrintWise 4800 gives you everything you have been waiting for in a CD or DVD printer. The same printer used in the PrintWise 1500/4500 series, the PrintWise 4800 lays down less ink than it's predecessors which means faster throughput, less ink consumption, and more compatibility with various types of printable media which results in faster drying times.

PRINTWISE SPECIFICATIONS

	CD Copy Speed	DVD Copy Speed	Input/Output Capacity	CDs Per Hour	DVDs Per Hour
PrintWise 1500	48X	4X	100/150	50+	8+
PrintWise 4500	48X	4X	120/120	100+	16+

QD-2/QD-123



16x CD-Recorder, 1:1 and 1:3 Duplicators

The QD-2 (Quick Disc-2) and QD-123 are low cost, high performance, one to one and one to three CD Recordable/Duplication systems. With their user friendly one button operation, sleek desktop footprint, and office environment aesthetics, the QD-2 and QD-123 are perfect for any user requiring easy copying of audio, video, or data CDs. Superior technology allows them to ship without a hard drive; duplication of CDs is done on the fly. CD+G Karaoke support. LED indicators provide you with the status of the duplication process, minimizing the number of faulty discs. Current high speed CD recording is 16x.

ECHO STAR ES-18CD-T *CD/DVD Shredder*

The ES-18CD-T safely and quickly destroys CDs and DVDs. The system reduces a disc to hundreds of thin plastic shreds. Removable hopper holds shredded CDs and is easily emptied. Heavy-duty electric motors assure smooth, trouble-free operation. A single disc can be destroyed in less than five seconds. Measures 12 x 8 x 14˝ and weighs 12 lbs.



COPYWRITER *Live*

Designed for recording live events to CD, the Copywriter Live easily integrates with your sound and video system to record a feed directly from your board. This allows for high quality mixed sounds to be recorded directly to CD as the event is taking place.



- You can record from practically any audio or video source, including live feed from a microphone, cassette deck, video camera, mixer, and many others.
- Make a high quality CD on the first try. Make backup copies, or archive your video to CD for playback in most DVD players. Record and pause to create multiple tracks for playback.
- Equipped with two CD recorders and disc spanning capability, the Copywriter Live allows virtually unlimited and easy recording of live audio or video. As the disc in drive one becomes full, drive two automatically starts recording the second disc.
- A setting in the intuitive menu interface allows for creation of either an audio disc or VideoCD on the fly. Playback of a VideoCD can be done on the CopyWriter Live itself, or on most DVD players. Each CD-R can hold approximately 60 minutes of video.
- The Copywriter Live is also a 1 to 1 (12x) CD duplicator with automatic detection and support for most CD formats. Simply drop your master in one drive and a blank in the other, and hit “copy”.
- As a rack mount (rackmount kit included) or set-top device, the system integrates easily with existing audio/video equipment.
- If you need multiple copies fast for immediate sales, it can be combined with a high speed duplicator for high volume just-in-time production.
- Combining the flexibility of cassette tapes with the quality and longevity of CDs, in one low-cost, easy to use unit, the CopyWriter Live is ideal for churches, schools, home or office in seminars, conferences, lectures and any other special event.
- Inputs/Outputs:
  3x RCA inputs (2 rear/1 front)
  1x XLR inputs (rear)
  1x RCA outputs (rear)
  1/4-inch mic input (front)

ORBIT II / ORBIT DVD



Standalone 50-Disc Automated CD and DVD Duplicators

The Orbit II is a low cost, high performance CD recordable duplication system with a sleek ergonomic design and 40x record speed that is perfect for any office environment. Orbit II's exclusive kernel autoloading technology, small desktop footprint, and ease of use offers even novices very fast, consistent performance. A choice of copy, copy/verify or verify only is easily selectable through the two button/LCD interface. Orbit II is ideal for a wide range of applications including pro audio, software development, government, desktop publishing, and many more. Combining DVD-R and CD-R duplication capabilities, the Orbit DVD gives you all the functionality of the Orbit II with DVD recording at 4x and CD recording at 16x.

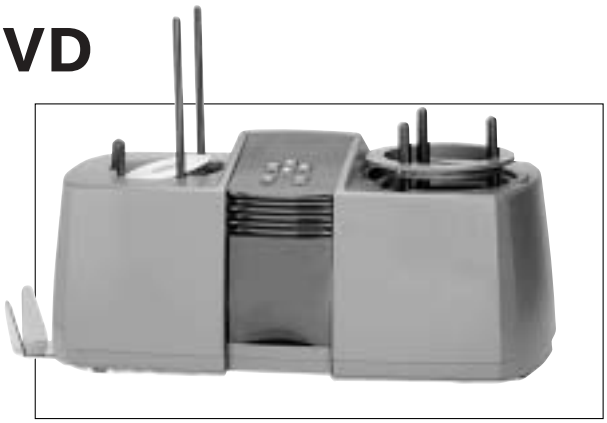
- Low cost /entry level
- Completely standalone, no PC required
- 50 disc input capacity
- "Insert and Duplicate" buttonless operation
- High performance autoloading mechanism
- Small desktop footprint
- Support for almost all CD formats
- Variable Record speed control (up to 48x)

Orbit DVD Only

- Copy from DVD-R/RW, DVD+R/RW and DVD Authoring Media
- 16X CD burning and 4X DVD Burning

ORBIT PRO / ORBIT PRO DVD

Standalone 150-Disc Automated CD and DVD Duplicators



The Orbit Pro is a sleek, standalone-automated CD duplicator with powerful features. The next generation of the award-winning Orbit II, the Orbit Pro touts two 40x recorders for ultra high-speed CD duplication. And with an Easy Copy Mode for the novice and Expert Mode for the advanced professional, the Orbit Pro fits perfectly in any production or studio environment. A Batch Mode allows unattended duplication of multiple master CDs as the system automatically detects the format of each master CD—a great timesaver. The Orbit Pro also includes a hard drive for multiple CD image archival and track extraction for custom audio CD compilations. To handle the 150-disc capacity of the system, a removable output hopper has been designed to unload the media onto a spindle, minimizing the handling of the discs. To ensure recording stability, the recording speed can be slowed for media not certified for high-speed duplication. Combining DVD-R and CD-R duplication capabilities, the Orbit Pro DVD gives you all the functionality of the Orbit II with DVD recording at 4x and CD recording at 16x.

- Sleek, charcoal black, molded enclosure fits any environment - desktop, audio and more!
- HDD for audio track extraction and image archival
- Disc to Disc and HDD to Disc copy modes
- Copy, Copy + Verify, and Verify only modes
- Batch Mode for multiple-master duplication
- Selectable recording speeds for up to 40x duplication
- 4-line lit display with intuitive menu navigation and detailed error reporting
- 150 disc input
- Output tray for rejects
- Removable output bin for easy disc unloading

Orbit Pro DVD Only

- DVD recording at 4X and CD recording at 16X (Orbit Pro DVD only)

DSR & COPYWRITER TOWER SERIES



Standalone, Manual Load 40x and 52x Duplicators

The 40x DSR and 52x CopyWriter Tower series are low-cost, easy to use CD recordable duplication systems that utilize superior technology from Microboards. With a compact footprint, and multiple configurations available (up to 8 CD recorders), they can be used disc to disc, or disc to multiple disc for higher volume requirements. No hard drive is required, because caching image files is done on-the-fly, to multiple writers simultaneously. LED indicators detect errors during the record process, preventing bad disc duplication, and monitoring the status of the recording session.

- Automatic detection of virtually all CD formats (CD-ROM Mode 1/2, Form 1/2, Red Book Audio, CD1, Mix Mode, Multisession.
- Selectable recording speeds up to 40x or 52x for recording stability
- No hard drive is required, because caching image files is done on-the-fly, to multiple writers simultaneously.
- Copy directly from the source CD to writer(s) simultaneously
- One-touch operation for simple CD copying, allows novice users to operate it effectively
- Test Mode lets you copy from the source CD to the blank disc as a test. No data is burned to the disc.
- Simulation mode ensures recording stability
- Easy to read LED display for status and error recording

DSR & COPYWRITER PRO SERIES

Standalone, Manual Load 40x and 52x Duplicators w/Hard Drive and Control Module



All the features of the DSR and CopyWriter series, the 40x DSR PRO and 52x CopyWriter PRO adds the benefits of a hard drive for image storage, including Bit-to-Bit Data Verification and Audio track extraction, as well as a moveable ergonomic interface module for access to advanced features.
They provide high-speed duplication at 40x and 52x speed with configurations of 2 to 16 recorders. An intuitive, menu driven display and simple 2-button control interface are integrated into a sleek remote module. The cabled remote module allows flexible positioning in different ergonomic locations. For example, you can keep the tower tucked under a desk while keeping the remote on top of the desk for easy control and status access.
The powerful interface provides simple one-touch operation for novices while giving pros access to advanced features. The built-in 20GB hard drive allows for CD image archival and track extraction for audio compilations. Recording speeds can be adjusted (slower) when using media not certified for high-speed duplication.

- Remote module interface allows for flexible, ergonomic positioning
- Built-in 20GB hard drive for CD image archival and track extraction for audio compilation CDs
- One touch start for CD to CD duplication
- Read Check ensures all copied discs are readable
- Selectable recording speeds up to 40x and 52X for recording stability
- Bit to bit verification of data formats
- Simulation mode to ensure recording stability
- Easy-to-read, two line display for status and error reporting

PRINT FACTORY



Automated CD-R/DVD-R Inkjet Printing Solution

Print Factory combines Microboards' patented autoloading system with HP's new inkjet printer engine to give users the highest output possible, both in image resolution and print speed. Print Factory has a 50-disc capacity, making the task of printing large numbers of discs fast, easy and hands-free.

- Print Factory is an autoloading printer, so no calibration of robotics or manual feeding of discs is required.
- Print Factory connects by USB or parallel interface to the computer, and comes with label design software.
- Uses standard HP color and black ink cartridges which have a very high yield at a competitive cost. This allows for printing costs of a few pennies per disc.
- Print Factory is capable of producing up to 200 discs per hour.
- With the included design software, users can enjoy its benefits whether making a few discs or a few hundred.
- Fast and flexible for any application. While Print Factory was designed to fill a need in the high-volume arena, it also has the ability to print at 4800 x 1200 dpi on both DVD and CD media.

Print Speed

Depends upon percentage of coverage and print mode selected:

- About 108 seconds for 100%, full-color coverage, Photo Normal Mode
- About 20 seconds for 100%, full-color coverage Text & Graphics, Best Mode
- About 12 seconds for 20% coverage Text & Graphics, Normal Mode

DSR DVD *CD/DVD Duplicators*

The DSR DVD Series is a multi-drive DVD-R/CD-R duplicator employing the use of general purpose DVD-R recorders. It supports both standalone and computer-based operation and allows you to copy from multiple DVD formats. Configurations of 1 to 16 recorders are available to meet individual production requirements. An intuitive, menu-driven display and 2-button control interface provide simple one-touch operation for the novice user while giving access to advanced features for the professional. A hard drive comes standard with the system, allowing you to archive an image or perform CD track extraction for CD audio compilations.



- Stand-alone DVD-R / CD-R duplicator
- One Touch duplication
- Speed selectable (1/2/4x for DVD-R and 4/8/16x for CD-R)
- Copy from DVD-R/RW, DVD+R/RW and DVD Authoring Media
- Direct DVD/CD mastering with optional Firewire (IEEE1394) interface for PC or Mac. (Windows software included with the interface).
- Copy and Verify
- Ergonomic placement of remote interface module
- Built-in hard drive for DVD/CD image archival and CD track extraction for audio compilation CDs
- Simulation mode to ensure recording stability
- Easy-to-read, two-line display for status and error reporting
- Updates easily installed through DVD/CD reader

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The Professional Audio SourceBook

Section 4b

Multitrack Recorders

Alesis **495-497**

Fostex **498-499**

Mackie **500-503**

Tascam **504-511**

LX-20

ADAT Type-II 20-bit Digital Multitrack Recorder



Built with the same 20-bit encoding and expansion capabilities as the rest of the ADAT Type-II line, the LX-20 provides an exceptional value in professional multitrack tape recorders. Also featured on the LX-20 is the 9-pin ADAT Sync interface which allows multiple ADATs to be synced together and the ADAT fiber optic, multi-channel digital interface that has become a standard digital interface found across the ADAT line as well as on a wide range of digital mixers and soundcards. The LX-20 makes the concept of an affordable, all digital tape based recording studio a reality.

ADAT Technology

- ➤ Uses standard S-VHS tapes to provide more than enough bandwidth to record 8 tracks of 20-bit digital audio.
- ➤ ADAT tapes are formatted with a proprietary time code with single sample accuracy (up to 1/48,000th of a second) that is much more precise than SMPTE Time Code (1/30th of a second).
- ➤ ADATs are ideal mixdown decks for surround and theatrical sound applications that require more than two-channels.
- ➤ Because time code is written into the subcode area of the tape you can synchronize ADATs without giving up a track. Up to 16 ADATs can be synchronized together for a total of 128 tracks—with single sample (20 microsecond) accuracy.

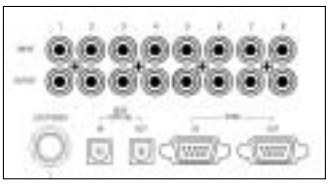
Recording Functions

- ◆ Eight record enable buttons control the input monitor status.
- ◆ Input monitoring is controlled with the Auto Input button which automatically monitors record-enabled tracks. The All Input button monitors all inputs regardless of the record-enable status.
- ◆ Pitch control depends on the current sample rate. +300/-200 cent pitch change is possible when using the 48kHz sample rate, ±200 cent at 44.1kHz.
- ◆ Uses crossfading when punching in and out to make the transition from what's recorded on tape to what is being punched, as smooth as possible. There are four crossfade times available -11, 21, 32 and 43 ms.
- ◆ Analog and digital input select buttons determine how analog and digital signals are routed. When recording from the analog inputs, there are three input modes available to accommodate the different bus architectures of mixing consoles.

Auto Locate

- ◆ There are 5 Locate buttons. Locate buttons 1 through 4 are used for shuttling to specific points on a tape. 0 returns the tape to the zero position. Locate buttons used in conjunction with Auto Play, Auto Return and Auto Record facilitate a number of automated transport functions.

Inputs and Outputs



- ◆ 8 unbalanced RCA inputs and outputs
- ◆ 1/4″ TRS jack for supplied LRC (Little Remote Control) or punch pedal
- ◆ ADAT 9-pin female sync input and output allows synchronization and machine control between multiple ADATs and optional BRC remote control.
- ◆ ADAT optical lightpipe I/O carries 8 channels of digital audio between ADATs and compatible devices. Bounce audio from one ADAT to another, mix and edit tracks completely in the digital domain.

Transport Controls

- ◆ Transport controls include REW, FF, stop, play and record. Also offers audible cue and review at 3x normal speed.

Display

- ◆ Time counter shows the current location of the tape in hours, minutes, seconds, hundredths of a second or optionally frames (1/30th of a second).
- ◆ Eight 13-segment meters show either input levels or the levels already recorded on tape. Eight red Record indicators located directly beneath the meters show a track's record status while blue indicators show input monitor status.
- ◆ Three blocks on the lower right hand corner of the display show the active parameter for clock (internal, external, digital) and sample rate (44.1 or 48 kHz), input monitor (all or Auto) and input source (analog, digital or track copy).
- ◆ Status indicators show the bit depth of the current tape (16 or 20-bit), whether various locate functions are enabled, and if a cassette is properly inserted.
- ◆ An interpolation indicator warns that error correction is taking place within the LX-20 and the heads should be cleaned.

ALESIS

HD24/HD24XR

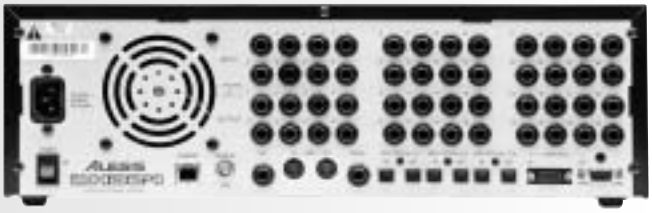
24-track, 24-bit/96kHz Hard Disk Recorders



Designed to fit comfortably into any standard ADAT environment and matching ADAT's legendary performance and affordability, the HD24 and HD24XR are hard disk recorders built from the ground up exclusively for the purpose of recording music instead of data — resulting in remarkable stability and performance. And yet, thanks to their use of standard, low-cost IDE computer drives, the recording media costs no more than traditional ADAT tape on a per track basis. Two hot-swappable media bays provide convenient access to the recording drives, and allow you to backup data in just minutes. You can edit tracks internally or easily transfer to computer via built-in Ethernet. They're the only HD recorders on the market with standard ADAT Optical and ADAT Sync, enabling them to work efficiently with existing systems. Essentially the same, the HD24XR incorporates state-of-the-art 24-bit A-D and D-A converters capable of greater than 110dB dynamic range and providing ultra-low distortion, even at nominal sample rates of 44.1kHz and 48kHz. It is also capable of recording and playing back twelve channels of audio at nominal sample rates of 96kHz and 88.2kHz, as well as 24 channels of audio at standard sample rates of 44.1kHz and 48kHz.

FEATURES

- The HD24 provides 24 tracks of high-resolution 24-bit recording at standard sample rates of 44.1 and 48 kHz, and supports 12 tracks at 88.2 and 96 kHz when slaved to external digital products. The HD24XR provides 88.2 and 96 kHz analog I/O right from the back panel.
- Standard connections include 24 analog inputs and outputs (+4dBu, 1/4″ TRS), plus 24 channels of ADAT Optical I/O. Also includes MIDI in/out, MTC out, Ethernet, ADAT Sync I/O and external Word Clock input to slave to external clock sources.
- Internal editing capabilities include cut, copy, paste, and track-move with undo can be done from the front panel, or remote control. Or just send the files (AIFF format) to a computer DAW via the Ethernet ports.
- Dual recording bays utilize standard low-cost, removable IDE drives so media costs don't break the bank. With custom drive caddies and protective storage cases, affordable IDE drives are hot-swappable, and offer fast back-up (a few minutes for an entire drive, is possible between the two front-panel drive bays)
- The HD24/HD24XR include a 20GB hard drive which yields 90 minutes of recording time at 24-bit/48 kHz.
- Instant random access to audio using ADAT/FST (File Streaming Technology). Hit REW or FF and you're there in a flash. The ADAT FST drive format establishes removable drives as the most reliable, easy-to-use multitrack media for pro applications.
- The HD24/HD24XR offer the digital audio and synchronization connections of the existing ADAT for easy integration into an existing system or to synchronize multiple HD24's to create a larger system. (Up to 5 units can be connected and synchronized to make a 120-track system, simply be daisy-chaining 9-pin sync cables.)
- The rear panel of HD24/HD24XR includes 24 channels of ADAT optical inputs and outputs and ADAT synchronization I/O connectors to allow them to sample-accurately synchronize with other ADATs or a BRC remote controller. When connected to a BRC, the HD24 "looks-like" three ADATs. (Comes with an Alesis LRC remote control, the BRC remote control is available as an option).



A rear-panel Ethernet port allows the HD24 to be connected as a stand-alone FTP server with its own IP address that can be accessed from a computer network, even over the Internet! Songs appear as folders on any Web browser, containing individual .WAV or AIFF files for each track. You can download and upload files from any HD24 connected to a network. Files can then be processed using computer-based editing applications and moved back to the HD24.

HD24/HD24XR ACCESSORIES

ADAT/FST *File Streaming Technology*

The HD24 and HD24XR uses a proprietary method of writing to hard disk to provide 24 tracks of 24-bit audio on low-cost IDE hard drives, and drop-in compatibility with over 150,000 ADAT systems worldwide.

- To give hard disk the same level of utility as tape, Alesis engineered a new method of writing on hard drives, specifically designed for music recording. Unlike the writing schemes employed by computer-based systems, this new method dramatically reduces fragmentation of data and the required "seek time", providing a much greater level of stability in recording and playing back data.
- Using this method of writing, seek and play functions are much faster than existing systems –under 100ms – and data fragmentation, which can cause crashing, is greatly diminished. As a result, very low-cost, low RPM hard drives can be used with exceptional results. Entire 24-track projects can be stored on a single removable drive, and for the first time the cost of the hard disk storage medium equals that of ADAT tape in cost per GB – less than five dollars.

COMPARING MEDIA COST @48kHz

	Media Unit Price	Recording time per media unit	Qty. req'd for 24 track, 45 min. project	Recording Cost Per track minute	Recording Cost Per 24-track minute	Recording Cost Per Project
ADAT Tape (unformatted)	$15.00 ea.	42 min. x 8 tracks x 20 bit)	3 tapes	4 cents	$1.00	$45.00
Standard IDE Hard Drive 5400 rpm	$99.00/ 30 GB average	135 min. x 24 track, 24 bit	1/3 of a drive (10 GB)	3 cents	73 cents	$33.00

HD24 FIREPORT 1394

Easy Connection of HD24 Hard Drives to PC Using Fast "Firewire Interface



The HD24 FirePort 1394 is an affordable interface which allows lightening-fast transfers of audio data from the ADAT HD24 or HD24XR to PC. The FirePort 1394 allows you to connect the HD24's ADAT/FST hard drives to a computer using a Firewire (IEEE1394) interface, where the audio files can then be incorporated or manipulated using a range of computer software.

- Comprised of two elements: a hot-swappable docking station for HD24 caddies with an IEEE1394 connection to the PC, and FST/ Connect software that allows you to manage ADAT/FST-formatted hard disk drives on the PC. The software not only allows easy audio data transfers to and from the PC, but gives you the ability to manage the file system on the target drive, i.e., creating, deleting, copying songs, and more.
- The hot-swappable FirePort module allows easy connect/disconnect of HD24 caddies in a convenient and portable desktop design. Once connected, take advantage of up to 400Mb/ second transfer rate up. An inline switching power supply eliminates the need for cumbersome "wall warts."
- With the HD24/HD24XR, the FirePort 1394 offers a complete solution for reliable and expandable cross-platform digital recording.

AI-4

8-Channel AES/EBU to ADAT Optical Interface



The AI-4 is a digital audio interface converter capable of converting AES/EBU and ADAT optical data streams at sample rates of 44.1kHz to 96kHz. Housed in a rack mountable box (1RU high) , this powerful converter is the perfect complement for the ADAT owner who needs AES/EBU input, output and conversion. Its simple interface makes this the ideal companion for any ADAT user.

- The AI-4 is designed to satisfy the requirements of the high-end studio ADAT owner, yet priced so it is just as accessible for the home studio recording engineer. There are many studio applications that require 24-bit ADAT optical to AES/EBU conversion (and vice versa); the AI-4 facilitates the interconnection of systems employing these two very popular digital interface formats.
- The AI-4 features 8 channels of AES/EBU to ADAT optical as well as 8 channels of ADAT to AES/EBU format conversion. Includes 4 XLR inputs and outputs capable of simultaneous 8-channel full-duplex single-wire transmission and reception. There's separate clocking for each format conversion, BNC wordclock input, and flexible clocking options for optical to AES conversions.

FOSTEX

D2424LV

24-Track Hard Disk Recorder



A computer-based digital multitrack set-up might be fine for studio use, where you can work around limited inputs and complex operational issues, but when it comes to live recording they simply don't cut it. There's only one chance to capture the performance. One chance to capture the magic. You need a solution which is tough, reliable, proven, rack-mountable, expandable and with enough inputs for the whole band—in short, you need the Fostex D2424LV.
Simply the most capable, most flexible, best sounding, easy to use, and affordable digital multitrack ever, the D2424LV is a rackmount digital recorder/editor with removable hard drive and six selectable recording formats, including 24-bit 96kHz. Designed for the live recording/reproduction, the D2424LV features 24-track simultaneous recording capability via 24 switchable balanced/unbalanced 1/4″ TRS connectors, 32 virtual tracks, program chain play mode, .WAV file import/export, and non-destructive 99-time Copy/Paste/Move across programs (songs). It also offers MTC/MMC & FEX full implementation, MIDI clock with song position pointer, multiple tempo and signature changes per song, and 'all input'/'all ready' control. Connections include ADAT I/O (S/PDIF switchable) Word I/O, MIDI in/out/thru, ethernet, SCSI-2 and RS422 ports. RS-422 I/O is complemented by the optional Timecode/Sync card, allowing the D2424LV to be used in studio and installation applications. A second drive bay is available for simultaneous back-up via optional second hard drive or DVD-RAM disc, making the D2424LV ideally suited for remote/live recording.

FEATURES

Recording Formats

- 24 tracks of simultaneous recording and playback is available at 16-bit and 24-bit using sample rates of 44.1 and 48 kHz
- 8 tracks of simultaneous recording and playback is available at 24-bit using sample rates of 88.2 and 96 kHz
- 32 additional 'ghost tracks' are available at the 44.1 and 48kHz sample rates for capturing and editing multiple performances
- 48 additional 'ghost tracks' are available at the 88.2 and 96 kHz sample rates
- ±6% vari-pitch function by 0.1% step.

Twin Drive Bays

- Two 3.5″ ATAPI BUS compatible drive bays are available for mounting a removable E-IDE hard drive for recording and playback plus an optional 2nd hard drive or DVD-RAM drive for backup and archiving – an optional caddy with mounting kit lets you use the 2nd bay with a removable drive
- A 50-pin SCSI-2 port is available for backing up recordings to an external hard drive.
- WAV file import and export is also possible to and from a DOS formatted FAT16 disk via the SCSI-2 interface

Non-Destructive Editing

- Non-destructive Cut, Copy and Paste editing with multiple chronological UNDOs (limited only by available disk space)
- Graphical preview function, using level meters, allows you to intuitively pinpoint a desired editing point
- Tracks can be named and exchanged
- Program duplication function creates an identical copy of a song program without using additional disk space
- Editing can be executed across different song programs
- Jog/Shuttle wheel lets you FF or REW through a recording with full audio monitoring—without altering the pitch. The inner wheel and the audio scrub continuously loops around a very small section of audio allowing you to pinpoint an edit start point precisely.

Comprehensive Editing Functions





D2424LV

Locate Memories

- 6 point edit memory and up to 99 locate memories can be stored and recalled using the jog/shuttle wheel or by pressing the Prev/Next key
- The Mark/Stop functions allows you to have the transports automatically stop at a preset Mark point
- Program Chain Play allows for a compilation of programs to be played back in a certain order

Optional Accessories

9043 Empty Hard Drive Caddy
Robust empty caddy for housing 3rd party IDE hard drives (see Fostex website for the latest drive recommendation)

9044 Drive Caddy & Mounting Kit
Complete assembly for adding a 2nd caddyheld hard drive for back-up and archive

9046 DVD-RAM Drive
4.7GB DVD-RAM drive (for back-up and archive) fits into the 2nd internal bay

8346 Timecode/Sync Card

- Sync to external VIDEO/WORD (10 preset sync patterns)
- Slave to external LTC running across 0h point. Offset can be set to external timecode
- Timecode can be recorded from internal or external sources to a 'virtual timecode track' keeping all recording tracks free for audio.
- Internal timecode address can be output as LTC
- Timecode converted from ABS or recorded timecode can be output as LTC
- Independent timecode function generates timecode from 24hr clock or any designated timecode address. FORCE JAM also possible. Generated timecode via the output.
- 24, 25, 30ND, 30DF, 29.97ND & 29.97DF timecode rates
- Sync to video feature for system integration with various video editors compliant with the P2 protocol



Front-Panel Remote Control

The D2424LV's entire front panel is removable for convenient positioning on the work surface, while the body of the main unit remains in the rack. The front panel controller with its large fluorescent display, transport and locate functions and Jog/Shuttle wheel can be removed and setup up 10 meters from the main frame using the optional 8551B extension cable.

Synchronization and Control

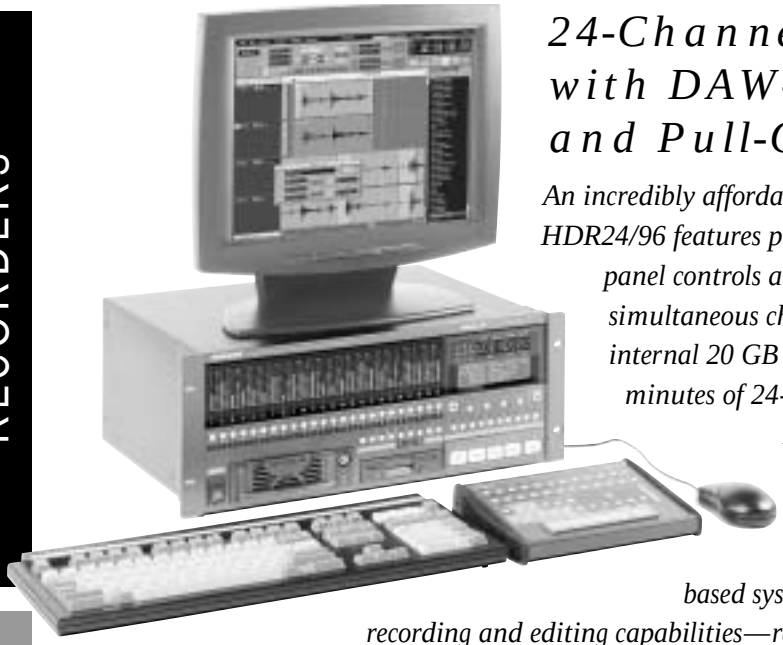
- MTC/MMC & FEX implemented for external MIDI control
- MIDI In, Out and Thru supports MTC and MMC as well as MIDI clock with song position pointer supporting multiple tempo and signature changes per song
- There is the choice of three time bases (ABS, MTC & bar/beat/clock), up to 64 tempo and signature changes per song,
- 'All Input' and 'All Ready' functions allow you to check all input signals and make all tracks ready with single button pushes.
- Word I/O (BNC) is standard for keeping multiple-linked digital devices 'in sync'
- RS-422 D-sub 9-pin remote in and thru connectors
- The "Virtual Timecode Track" feature allows recording of timecode from internal or external sources to a 'virtual timecode track' keeping all tracks free for audio.
- Optional Timecode, Ethernet and timecode cards are available
- Footswitch control for start/stop as well as punch in/out functions.

Inputs and Outputs



- 24 balanced 1/4″ TRS inputs and outputs using 24-bit 128x oversampling A-to-D and D-to-A converters designed by Asahi Kasei
- The reference level for the analog I/Os are selectable between -12dB and -20dB.
- Three 8 channel ADAT Optical I/O's (switchable to S/PDIF) are provided for interfacing with a digital mixer, computer-based DAW or any number of multichannel mic preamps with A-to-D converters.
- Back up recordings can be to external SCSI-2 media, DAT, ADAT or to an optional secondary internal hard drive or optional DVD-RAM drive

HDR24/96



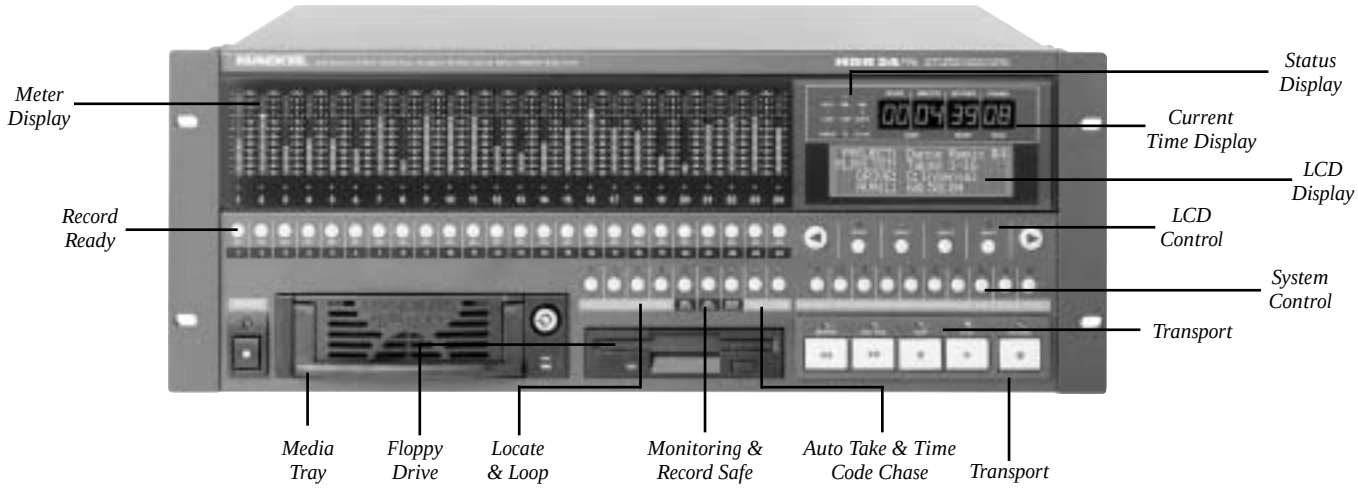
24-Channel 24-Bit Hard Disk Recorder with DAW-Style Editor and Pull-Out Media

An incredibly affordable 24-bit/24-track non-linear hard disk recorder, the HDR24/96 features pristine sonics, pull-out media, intuitive analog-like front panel controls and rugged construction. It has processing power for 24 simultaneous channels at 24-bit/48kHz or 12 channels at 24-bit/96kHz. An internal 20 GB IDE Ultra-DMA hard drive provides storage for up to 90 minutes of 24-track recording while an on-board drive bay accommodates Mackie's Media M-90 pullout disks. Perfect for recordists, as a replacement for older tape-based digital recorders, or in multi-deck, multi-room operation.

Front panel controls are fashioned after familiar tape-based systems. But unlike them, the HDR24/96 offers true, non-linear recording and editing capabilities—record multiple versions of a track or track segment without destroying the original—nothing is lost. All edits are non-destructive as well.

Full-featured DAW-style editing software is accessible by adding an SVGA monitor, two-button mouse, and a standard PC keyboard. No external computer is required to take advantage of the HDR24/96's rich graphic operating interface and editing software. Rear panel ports let you plug the monitor, mouse and keyboard directly into the recorder. The intuitive editing software includes 999 levels of undo, non-destructive drag-and-drop crossfades, regions and super regions, track slipping, audio phase inversion, normalization, pitch shifting, time compression and expansion, audition and scrubbing modes, quantization, one-to-multiple replacement, 192 virtual takes (8 per track), track and take bouncing, and many other familiar features. The SVGA display also provides complete control over recorder functions and features a meter display, virtual transport controls, time code display, scrolling waveforms with 1x/2x/4x/8x/24x display, time bar with user-defined resolution, Punch, Loop, Cue and Tempo Change markers, snap functions, locate, loop and rehearse.

Designed for project studios, live recording, video post production and professional dubbing facilities, the HDR24/96 connects to any console—large or small, analog or digital—using your choice of four optional 8-channel I/O cards including analog, ADAT, TDIF, and AES/EBU (the same ones used for the D8B). It also syncs to SMPTE, MIDI, video black burst, NTSC and PAL clock sources with no additional hardware required, and offers an optional RS-422 (9-pin) card for integration into professional video systems.



HDR24/96

- 24-track/24-bit non-linear, non-destructive editing
- 12-channel 24-bit recording at 96kHz
- 192 virtual tracks (8 virtual takes per track for overdubs and additional data storage
- Use with ANY analog or digital mixer; optional 8-channel cards can be mixed and matched—no converter boxes needed. Choose from ADAT, TDIF, and AES/EBU.
- Integrates seamlessly with the Mackie d8B for extra functionality
- 24 individual record ready buttons with LED; transport controls, Track Arm buttons, Locate and Loop
- Built-in internal IDE hard drive for up to 90-minutes of 24-track recording, plus pullout bay for extra removable Mackie Media hard drives
- Front panel cut, copy, paste editing with 999 undo's
- Full meter bridge
- On-screen graphic interface mirrors front controls and meter bridge
- Sync to SMPTE, MIDI, Word Clock and video black burst
- 4 x24 LCD shows operating info, prompts, and selects options via paging buttons
- 100Base-T Ethernet port for easy file transfer to digital audio workstations
- The HDR24/96 is an FTP-server as well. Transfer .WAV and AIFF audio files via FTP and Ethernet into a DAW for additional processing and/or editing.
- 3.5-inch floppy drive bay for importing tempo maps and software upgrades
- With the optional Remote 48 Controller, you can run 48 simultaneous channels of HDR-24/96s and be assured that all 48 channels will start and stop in unison.
- Built-in mouse, keyboard and monitor ports—no external computer needed
- Full-feature editing software built - in

–Non-destructive cut, copy and paste

–999 undo's with real time history list

–Continuously scrolling track waveforms

–2x, 4x, 8x, 12x & 24x track displays

–Drag and drop crossfades and rack slip

–Instant drawing and fade in and fade out

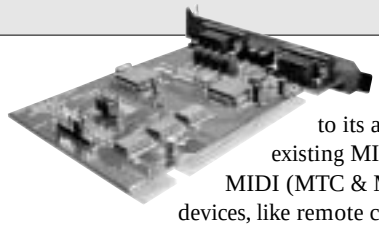
Mackie Media M90



Ideal for studios saving clients projects - or home studios needing to back-up a lot of data, one Mackie Media M90 removable hard drive gives you over 90 minutes of pre-formatted 24-track, 24-bit audio. Consisting of a 20GB UltraDM33 IDE drive, you can back-up from the SDR or HDR24/96's internal drive - or simply record directly to a Media M90 and skip the back-up process altogether. Each Media M90 comes in its own padded storage case, complete with track sheet and project labels for both drive and case.

- Pre-qualified, pre-formatted, pre-tested
- Mackie Media drive tray for easy removal
- Back up entire internal drive at real time
- Includes padded case with ID label and Mackie track Sheets

Serial•9 RS-422 Card



The Serial•9 is an optional PCI card that installs in the HDR24/96 and adds RS-422 control capabilities to its already-rich feature set. The Serial•9 replaces the existing MIDI card seated inside the HDR24/96 and offers both MIDI (MTC & MMC) and RS-422 communication to/from external devices, like remote controllers and digital audio workstations.

Sony 9-pin Device Protocol

- Slave the HDR24/96 to a wide range of remote controllers
- Have all positional reference in one simple connection
- Expands the applicability of the HDR24/96

MTC and MMC Communication

- Slave the HDR24/96 to a DAW
- Control MMC-compatible devices with the HDR24/96's transport
- Distribute to multiple machines for expanding track count

HDR Pro

With the optional HDR Pro package you can add even greater functionality to the HDR24/96. Using the HDR Pro software you can export HDR Project Files as Digidesign Pro Tools Session files (for 3.x, 4.x, 5.x Mac versions only), and allows Pro Tools Session files to be directly into an HDR Project. In addition, HDR Pro comes with a single-bay Firewire drive that is fitted with a Mackie Media M90 drive frame for direct communication between the M90 and the Mac desktop. This makes backing up projects or swapping to and from the HDR24/96 and Pro Tools a breeze. Easy project storage, too, since you can swap the hard-drives in the Mackie Media M90 drive frame when you need to.



HDR Pro is THE solution for anyone who wants to record with the HDR24/96 and import audio files and session data into Pro Tools.

HDR24/96, *continued*

Built-in Editing Software

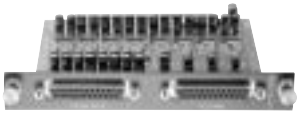
No expensive Mac or software-laden PC needed. Simply plug an SVGA monitor, PS2 mouse and PC keyboard into the HDR24/96's rear panel ports - and you're ready to go. If desired, the video can be output to a projector or flat screen plasma display. The HDR24/96's software will unleash your creativity allowing you to easily perform the most intricate tasks such as slipping, trimming, looping, fading and cross-fading tracks. In addition to lots of on-screen navigation tools and keyboard short-cuts, the HDR24/96 has an exclusive "Dive Key". Position the mouse anywhere on the track overview and press the 'Z' key on your keyboard. You are instantly zoomed to where you can immediately select, nudge, trim and edit.



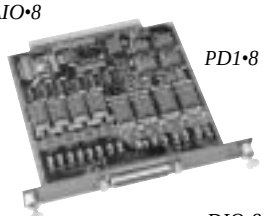
- Front panel controls are on the screen, and you'll rarely see a menu. 24 track meters display 'plasma-style' peak/hold and averaging levels at the same time. Large 'LED' display shows current time, and the transport controls are right there.
- Ultra-fast ATI Rage Pro video card allows real-time waveforms to scroll past a fixed 'now' line (instead of jerky page jumps). Select a color palette. View all 24 tracks. Narrow your scope down to a stereo pair. Zoom to view tracks right down to the sample, in real-time - without effecting the HDR24/96's performance.
- Easily set Cue points, Locates and Punch In/Out points. Capture points on-the-fly from the GUI, keyboard, front panel, or remotes. A pop-out window displays all cue points as you add them. Simply double-click or use the keyboard to locate any point in a project instantly.
- Easy and powerful drag 'n drop editing. Pull a region out of the region list and drop it onto the track view. Snap the region to Hrs:Min:Sec:Frames or Bars:Beats:Ticks. Pull a region from one track and join it to another to create a new region. Slip regions, trim their beginning or end points. Turn on 'Loop' and drag a region's start or end points as far as you need.
- Context-sensitive mouse changes from one type of tool to another based on where you move it. The most complex editing tasks become a drag 'n drop breeze. Pull the top right or left corner of a region to create fade-ins and fade-outs. Right click to select the fade curves. Push one region into another and instantly create a cross-fade with your selection of nine fade curve combinations. The HDR24/96 re-draws all waveforms in real-time as you edit.
- Automate the volume of each region by creating a region volume envelope. Click to add as many volume points as you need. Click again to drag them into position, creating as complex a mix as you can imagine. It mutes, too.
- Scrub all 24 tracks of audio on the HDR24/96 from the jog/shuttle wheel of Mackie's D8B digital mixer, Remote 48, or any other MMC wheel. But scrubbing gets really cool with the HDR24/96's mouse! Listen to what you're doing as you drag to spot edits and set markers. It's all real-time and designed to make you more creative and productive.

I/O Options for the HDR24/96

Each HDR24/96 holds your choice of four 8-channel I/O cards (the same ones as used in the D8B), as well as a 24-channel card. Mix and match for a wide range of input options.



AIO•8



PD1•8

DIO•8



Opt•24: 24-Channel Digital I/O
Unlike the other I/O cards, the OPT•24 provides 24 channels of ADAT formatted digital audio I/O via three sets of TOSlink optical connectors. Because of it's unique feature set and design, the OPT•24 is not used in the standard I/O card cage - it is installed in one of the vacant PCI slots inside the HDR24/96 and provides a permanent 24 channel digital I/O platform that can be used in tandem with whatever other I/O cards may be installed - which means you can interface optical and analog/AES formatted signal together without purchasing and swapping I/O cards.

AIO•8: Analog I/O Card
8 balanced line level (+4dBu) outputs and returns for connecting the D8B to analog equipment

DIO•8: Digital I/O Card
ADAT lightpipe and TDIF with format conversion (16/24-bit). Format convert between ADAT and TDIF

OPT•8: Low-cost Digital I/O Card
The low-cost OPT•8 I/O card provides 8 channels of ADAT lightpipe for under $100!

PDI•8: 24-bit AES/EBU Digital I/O Card
24-bit AES/EBU with sample rate converter on stereo input pair. Plus it has a 96kHz I/O for audiophile-quality recording via esoteric outboard D/A converters

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HDR24/96, *continued*

Remote 48 Controller

Professional hard disk recorders require professional remotes. But Mackie went far beyond just transport controls and record arm buttons when they designed the Remote 48. It fully duplicates the entire front panel of our HDR24/96 Recorder/ Editor —including LCD and LED displays, all function buttons and entry keypad. In fact, the Remote 48 adds features including a weighted jog/shuttle wheel, additional Locate Time display, 10-key keypad and more. Use the Remote 48 to control two HDR24/96s. When connected to an HDR24/96,the Remote 48 controller is fully interactive with the SVGA graphic user interface.



FEATURES

- Status LEDs show sample rate (44.1, 48 or 96kHz), Bit Depth (16- or 24-bits), Sample Clock, and Time Code status information.
- LCD display is identical to the one on the front panel of the HDR24/96)
- Toggle between SMPTE and BBT on the Remote 48 and HDR24/96 front panels
- Edit Time button lets you edit specific fields within the Locate Time display. Pressing the button once selects the Frames/Ticks field, a second time the Seconds/Beats field, etc.
- Access LCD display menus such as the Track, Project, Backup, Disk Utility, System, Digi I/O and sync menus
- Record ready buttons for tracks 1-48
- Copy the instantaneous value of Tape Time into the Located Time
- Numeric keypad enters time values or cue numbers into the Locate Time or Cue displays
- Copy the values of the Locate Time or Tape Time displays into cues, numbered locates, pre-roll time, and loop and punch lengths
- Record Safe disarms all tracks and disables all recording controls. When enabled, Record Safe prevents you from arming tracks and entering record.
- Jog/Shuttle/Data Wheel is used for data entry in certain Autolocator operations, such as scrolling through the Cue List or changing time values in edit time operations, in addition to controlling Jog and Shuttle motion.
- Jog gives precise control over transport cueing by playing audio both forwards and backwards at non real-time speeds in proportion to how the Wheel is rotated. Jog play speed varies continuously between 0x and 1x of normal play speed.
- Shuttle allows you to play audio both forwards and backwards at non real-time speeds in proportion to how far the Wheel rotated. Shuttle play speed is quantized to 1/32, 1/16, 1/8, 1/4, 1/2, 1x, and 2x the normal play speed.
- PUNCH automatically punches into Record at the punch in point, and punches out of Record at the punch out point.
- REHEARSE allows you to practice punching without recording anything to disk, and automatically stores the punch points to the Auto Punch in and out points.
- AUTOPLAY automatically starts playback when you've jumped to LOC or Cue point.
- AUTO TAKE automatically increments the active Virtual Take of all armed Tracks at the beginning of each record pass.

Remote 24 *Small Remote*



Designed to work with the HDR24/96, the Remote 24 provides remote control of the most commonly used functions.

- Arm 24 tracks for recording
- Select Record Ready, Auto Input or Rec Safe
- Set up to four locate points
- Select Loop, Punch and Rehearse modes
- Select 8 virtual takes for each of 24 tracks
- Select Auto Take Mode for repetitive virtual track recording
- Delete last record pass button
- 12-segment LED meter for selected track
- Full-size transport controls
- Punch in/out foot switch jack
- Accepts standard Atlas AD-11B flange for mic stand mounting
- Single cable connection to recorder for power and control data

TASCAM

PORTASTUDIO

Since their introduction in 1979, Tascam has sold over 1,000,000 Portastudios, making them some of the most popular tools ever to be embraced by musicians all over the world. Tascam's four models of analog, cassette-based Portastudios—the ultra-affordable MF-P01, the easy-to-use Port02MKII, the 414MKII and the advanced 424MKIII—each offer respective advantages for several levels of musicians and recording hobbyists. However, what they all share is the inexpensive, readily-available cassette tape recording medium, and the integration of simple mixers and recording devices in one portable package.

MF-P01



The simplest and most inexpensive Portastudio ever made, the MF-P01 is ideal if you're just getting into recording your own music. Using the MF-P01 couldn't be easier ... just pop in a standard cassette tape, plug in your instrument, select the track you want and hit "Record". You have four tracks to work with, so recording your drum machine, guitar, vocals and keyboard is no problem.
The MF-P01's easy-to-use recording approach is based on the unit's single Mic/Line input and no buss requirement. To make new songs, simply select the track on which you wish to record by pushing a single button and then adjusting the input level. A 4-dot LED meter allows you to easily identify input signal levels and overloads. During playback, the four-channel mixer section allows users to mix down their songs by simply adjusting each track's output level. To monitoring recordings, the MF-PO1 comes complete with both a 1/4″ headphone output and an RCA L/R line output. Its small 12 x 8″ footprint and light-weight design (a mere 2.5lbs.) means the MF-P01 can be easily carried to and from music lessons, jam sessions or band practice.

- Perfect for musicians and novice recordists
- Uses inexpensive standard cassette tapes
- Records one track at a time with classic warm analog sound
- Includes 1/4″headphone output and an RCA L/R line output.



Porta 02

A cost-effective self-contained unit designed for the home recording musician, the Porta 02 features two Mic/Line input channels each with a dedicated linear fader, a four channel mixer with LEVEL and PAN controls for each channel, a 4-track cassette recorder with 2-track simultaneous record capability, a head-phone monitor output with level control and Mono switch and a Master L-R Line output. Sporting an entirely blue color scheme, the Porta 02's design appearance is especially appealing for today's young musician.
The Porta 02 has a solid and smooth feel to all the knobs and faders - inspiring confidence at every step of the creative process. 2-track simultaneous record capability makes it the perfect tool to capture live performances, as well as multitrack recording. With its mono capability, the headphone monitor output makes it easy to check for phase discrepancies - thus ensuring quality recordings.

- 4 track 4-channel cassette recorder section
- Two 1/4″ mic/line inputs for keyboards mics, guitars, drum machines and most other sources
- Up to 2 tracks simultaneous recording
- Simple 4-track recording with assignable input-to-track routing
- 4-channel Cue mixer for mixing down
- 4-stage LED meters for each track
- Headphone monitor out with level control
- RCA unbalanced line outputs

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EQUIPMENT LEASING AVAILABLE

414 MKII



Packing high-end features in a convenient, compact and economical package, the 414mkII Portastudio is ideal for newcomers or those experienced in multitrack recording. Use it for demos, as a portable sketch pad, or for practicing recording techniques. A high-speed transport, combined with Hi & Low EQ and dbx noise reduction means great sound. Four full-function input channels, each with mic/line capability and trim control (two feature XLR inputs and one is a dedicated guitar input). High and low EQ make it easy to capture quality 4-track recordings in any setting. Two effects sends per channel, together with two stereo effects returns, give you plenty flexibility when using effects and external audio sources. Also includes a smooth-action 60mm master fader, 12% pitch control, discrete sync output, and a musician-friendly price.

- 4-track, 4-channel format
- Simultaneously 4-track recording
- 4-mic/line inputs with trim control, supplemented by dedicated stereo inputs
- Dedicated guitar input with adjustable trim control
- 2-band High and Low EQ on each input channel
- ±12% pitch control
- Zero return function
- 2 Aux sends for effects processing, one of which is switchable to Tape Cue
- Separate main and monitor outputs allow mastering and monitoring without repatching Recorder section
- dbx noise reduction with Sync on/off select
- 3-3/4 ips tape speed (twice normal speed) for superior reproduction quality
- Stereo Sub input patches stereo sources, such as mixing consoles direct to L/R busses
- Sync out terminal for easy MIDI-tape synchronization
- Optional C-30P foot-switch for hands-free punch in/out control



424 MKIII

A real personal recording studio with a professional mixing console and a logic-controlled cassette deck in a single easy-to-use unit, the 424MKIII is Tascam's finest four track cassette . The 424MKIII is equipped with powerful musical EQ on all six input channels, flexible auxiliary and monitoring systems, and a comprehensive range of inputs and routing options. Advanced feature set includes 2 independent auxiliaries so you can add two separate effects to the same channel. The 3-band sweepable mid band EQ gives you the flexibility to focus on the sweet spot of the sound to help give it a little more life. The 4 XLR mic preamps keep your inputs quieter, and will save you from the hassles of finding the XLR to 1/4˝ transformers.

- 4-track 4-channel recording system
- Simultaneous recording up to 4 tracks
- 4 assignable XLR mic inputs
- Logic controlled 4-track tape transport
- Dedicated stereo inputs (channels 7-8)
- 6-full-function input channels with Mid-sweepable 3-band EQ
- 2 Aux sends for effects processing, one of which is assignable to Tape Cue
- Dual tape speed (3-3/4 and 1-7/8 ips)
- dbx noise reduction
- ±12% pitch control
- Return-to-zero and 2 locate points
- Repeat function
- Auto punch I/O with rehearsal function
- Stereo Sub Input
- Separate main L-R and monitor outputs
- Dedicated 4 tape outputs
- Easy-to-read fluorescent display
- Optional C-30P foot-switch for hands-free punch in/out control

DA-78HR/DA-98HR/DS-D98

High Resolution Modular DTRS (Digital Tape Recorders)



The DA-78HR is a 24-bit tape-based 8-track modular digital multitrack recorder. Based on Tascam's DTRS (Digital Tape Recording System) technology it provides up to 108 minutes of pristine 24-bit or 16-bit digital audio on a single 120 Hi8 video tape. Designed for project and commercial recording studios as well as video post and field production, it features built-in SMPTE Time Code Reader/Generator, Word Sync In/Out/Thru, MIDI Time Code synchronization. A digital 8 x 2 mixer with pan and level controls and a coaxial S/PDIF digital I/O allows pre-mixed digital bouncing and playback monitoring within a single unit, or externally to another recorder or even a CD recorder. Additional features include 8-channel TDIF-1 digital I/O, pitch control, test tone generator, electronic patchbay, machine offset in sub-frame or sample rate accuracy and Auto Punch I/O in 0.1 frame accuracy. LCD display, set-up data memory on tape, seamless punch I/O with digital cross fade, adjustable peak hold release time level meter and an A/B head playback error rate-display function. Up to 16 DTRS machines can be synchronized together for simultaneous, sample accurate control of 128 tracks of digital audio.



The DA-98HR has all the features of the DA-HR78 plus it adds the audio quality and critical features required for high-end post production and commercial recording facilities. As well as 44.1 and 48 kHz sampling frequencies, the DA-98HR can record and playback 24-bit or 16-bit recordings at 88.2/96kHz (4-tracks) and 176.4/192kHz (2-tracks) sampling frequencies. In fact, different sampling frequencies can be recorded simultaneously on different tracks on the same tape, providing the ultimate in flexibility. Additional step-up features include confidence replay mode, allowing off-tape monitoring while recording is in progress and 9-pin RS-422 control. XLR time-code input/output terminal with an onboard SMPTE synchronizer, Video sync allows the DA-98HR to be used while synchronized to other recorders so that backup tapes can be constructed as the session progresses. Multi-channel AES/EBU digital I/O provides instant integration into digital patchbays, DAWs, and other systems. Also offers a user-friendly setup menu with comprehensive LCD display and direct menu access soft keys.



Identical to the DA-98HR, the DS-D98 is designed to support the Sony Direct Stream Digital format for Super Audio CD (SACD). The DS-D98 can be configured in both a DSD two-track and PCM multitrack format— the only tape-based machine capable of SACD recording as well as the only tape-based system designed for high resolution DSD audio recording that allows multiple units to be synchronized. In DSD mode, the DS-D98 functions as a 2-track recorder/player with synchronization of up to 16 units and to other DTRS recorders and features Sony's SDIF-3 digital I/O. PCM recordings are backward compatible with existing DTRS recorders.

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DA-78HR/DA-98HR/DS-D98

Transport Controls

- Standard transport controls include Play, Record, Fast Forward, Rewind and Stop
- Pitch is variable ± 6% in 0.1% steps
- Precision rotary shuttle knob advances or rewinds tape at 1/4 to 8 times normal speed. While shuttling tape, the monitor output level is automatically attenuated by 12dB of the normal playback level.
- Shuttle monitor button mutes the over-bearing, high pitched squeals associated with high speed shuttling.
- Fast frame accurate tape location and positioning; end to end winding of a "120" tape is 80 seconds

Synchronization

- Full SMPTE/EBU operation with on-board time code reader/generator and support for all frame rates including 29.97 DF/NDF 30, 25 and 24 fps.
- ABS (Absolute Time) can be converted to SMPTE time code. SMPTE time code can also be offset from the ABS time code.
- When time code is on part of a tape and you want to stripe the rest, an assembly time code function will read a few seconds of the time code and jam sync the generator and stripe the rest of the tape.
- Re-chase time code is achieved monitoring the internal (Absolute Time Code) and external code and then slowing down or speeding up the transport to ensure consistent sync or it can run freely once the initial sync has been achieved, ignoring incoming time code. The re-chase window settable to 1 or 2 seconds, determines the amount of drift that is tolerated before re-chasing is necessary.
- Time code errors of 10 or 30 frames can be set up to be bypassed and ignored.
- The park position of a slaved DTRS machine is the pre-roll time needed for the machine to lock to timecode. Park position can be preset or automatically set to the optimum park position.
- MTC (MIDI Time Code) support allows synchronization between the DTRS machine and a MIDI compatible device and can be output when the DTRS is playing, in FF, rewind or when stopped.

High Resolution DTRS Technology

- ➤ Selectable 24-bit and 16-bit recording at 44.1 or 48 kHz sampling rate
- ➤ 24-bit, 128x over-sampling analog to digital and digital to analog converters
- ➤ 1 hr. 48 min. recording time on a single 120 Hi8 tape
- ➤ 20Hz - 20kHz frequency response and 104dB dynamic range
- ➤ Fully compatible with DA-88 machines.
- ➤ When synching multiple DTRS units, it is sometimes necessary to offset the time between one or more slaves in relation to the master deck. All DTRS units have frame accurate offset capabilities independent of SMPTE time code functions.
- ➤ Track Advance and Track Delay from -200 to 7200 samples in 1 sample steps can add subtle nuances to a musical groove or tighten the synchronization of a sound effect.
- ➤ Tape formatting process writes subcode information such as ABS (Absolute Time) on the non-audio portions of the Hi-8 tape. This ensures machine to machine compatibility of tape with no potential for loss of timing information due to tape stretching or fraying.
- ➤ Exclusive, interleaving write after read 4-head system provides superior drop out protection and yields significantly more area read on tape.

Editing

- Digital Track Copy function allows assemble or composite editing between machines or within a single unit. For example, copy tracks 3 and 4 to tracks 5 and 6 within one machine or copy tracks 3 and 4 of one to tracks 5 and 6 of another machine.
- Highly accurate and reliable transport mechanism features DD brushless motors
- Built-in digital patchbay allows routing any input to any track digitally. For example input 1 can record to track 2 and input 2 can record to track 1.
- Offset times between master and slave units(s) can be set on the fly while machines are in the play mode, or by entering a specific offset time

On-Board Stereo Mixdown Functions

- Level and pan controls for each channel (controllable via MIDI or from the front panel of the unit)
- They have a read before write feature that allows tracks 1 through 8 to be bounced to tracks 7 and 8
- Digital direct-track copying within a single unit or between multiple units
- Coaxial S/PDIF digital I/O accommodates direct mixdown from the DA-78HR multi-track to a compatible digital recorder such as DAT, CDR or DAW

Auto Locate And Punch

- The points at which recording is started and stopped are automatically stored in memory as punch points.
- Auto Punch allows hands-free punch in and out recording of a section of audio. A rehearsal function allows setting and auditioning the in and out points before executing the punch.
- Two Auto Locate points allow one button access to any two positions on a tape, or use for a beginning and end marker for continuously looping a section of audio.
- The exact position of locate and punch points can be set by min/sec/frames
- Pre-roll times for locate and punch points are independent and can be set from 0 seconds to 59 minutes and 59 seconds.
- When executing an edit or punch, the crossfade time between the signal on tape and the input signal to be recorded, can be set from 10 to 90ms (milliseconds)
- In Auto Play (after rehearsal is complete) the tape automatically rewinds to the pre-roll position before the punch in location

DA-78HR/DA-98HR/DS-D98

Conveniences

- All Input Monitor Button automatically sets all outputs to monitor the audio inputs, regardless of the transport status. This function is primarily used for alignment purposes and is equivalent to pressing all of the input monitor switches.
- Auto Input/Auto Monitor switch automatically activates the inputs when shuttling, fast forwarding or rewinding or in stop mode. This is useful for allowing talent to communicate with the engineer when no audio is being played.
- They can display how many hours are on the drum heads. This is useful for determining when routine maintenance such as head cleaning may be necessary.

Synchronization I/O

- MIDI in/out/thru allows MIDI timecode (MTC) synchronization as well as MIDI Machine Control (MMC)
- Remote/ Sync In and Sync out used for connection with optional RC series remote controls as well as allowing the unit to be used in a master/slave system.
- Word clock in/out and thru (via BNC connector) ensures that digital audio transferred between machines via T/DIF or S/PDIF I/O is synchronized.

Inputs and Outputs

- Eight unbalanced analog RCA inputs and outputs
- Two 25-pin connectors provide eight channels of balanced analog inputs and outputs
- 25-pin T/DIF (Tascam Digital Interface) connector is used for transferring up to 8 channels of digital audio to and from units conforming to the T/DIF format.
- Remote punch in/out connector for use with the optional RC-30P footswitch
- RCA in/out allows all SMPTE/EBU timecode frame rates to be read or generated



Additional Features

- All functions are front panel accessible including, machine ID (used in multi machine configurations), trim and tape locate positions, test tone oscillator and defining pre-roll time.
- 12-segment LED peak reading level displays
- Internal digital sine wave generator provides signals of A-440 and 1kHz at the nominal operating level of +4dBu for tuning and meter calibration
- When recording from a 20- or- 24-bit word length to a 16-bit length, dithering can be set to add noise to low level signals to prevent unwanted quantization noise to the signal.
- There are three dither settings available: No Dither, Triangular and Rectangular which gives 3dB better S/N ratio than triangular but may add noise modulation to the signal.

Optional Remote Control Units for the DA-78HR/DA-98HR/SD-D98



RC-808 Remote Control
Controls all of the basic transport functions including, record, auto-locate, monitor switching, track arming and punch **179.95**

RC-828 Remote Control
For controlling up to four DTRS machines with transport control, track arming, 12 locate memories, jog/shuttle and more **529.95**

RC-898 Remote Control
Provides accurate control of up to 6 DTRS machines including all basic transport functions, input monitoring and track arming. It also controls the transport functions of external units conforming to the Sony P2 protocol using the RS-422 interface, as well as parallel and GPI controlled devices **1199.95**



- No power supply, draws power from DTRS units.
- Up to 99 location memories can be stored and edited with frame accuracy.
- Time code values are input with a positive feel numeric key pad.
- Convenient menu system with 20-character by 4-line backlit LCD display.
- Twenty frequently accessed menu settings can be assigned to function keys for quick and easy recall.
- All time code formats are supported as well as pull up/pull down.
- Up to 10 setup configurations of the RC-898 and up to 6 DTRS units can be stored in memory for instant recall.
- Location memories and configuration settings are stored in battery-backed memory.
- Includes remote/sync cable and remote/sync terminator

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ONE HOUR FREE PARKING
AT 349 W. 34th STREET (with purchase of $100 or more)

DA-78HR/DA-98HR/DS-D98

DA-98HR and DS-D98 Step-up Features

Not only can the DA-98HR record 8-tracks of 24-bit audio at 44.1 and 48 kHz, it can also record 4 tracks of 24-bit audio at 96kHz and 2 tracks of 24-bit audio at 192kHz making the DA-98HR the ultimate mixdown deck. The DA-98HR builds on the DA-78HR and adds critical features for post production including the availability of AES/EBU digital I/O for instant integration into digital patchbays, DAWs, and more! The DA-98HR can also be controlled with standard DTRS remotes, or via 9-pin serial.



STEP-UP FEATURES FROM THE DA-78HR:

- Confidence replay mode allows direct monitoring off of tape while recording
- Input monitor mode allows channel by channel source monitoring regardless of tape transport status
- Simplified source/tape monitoring functions with automatic switching
- Edit and punch crossfade times are selectable between 10 to 200ms
- There are three user setup memory banks for storing setup profiles
- Menu settings are displayed on a LCD display with cursor and Enter/Escape keys
- 10-soft key location memories in addition to the basic memo 1 and memo 2 locate points used for looping and punches
- Pull up and pull down functions are provided to adjust sampling frequencies to drop frame rates. For example, since the frame rate of film (24 FPS) and the frame rate for video (29.97 FPS) don't conform to a simple mathematical ratio, audio sampling frequencies would be affected on transfer back to film
- Ten most commonly used functions can be assigned to soft keys
- 15-segment peak level meters have selectable fall ballistics and hold time ranging from 0 to 9 seconds and continuous peak hold
- Reference levels for the analog I/Os are selectable between three standard values: Tascam (-16dB), SMPTE (-18dB) and European (-20dB)

Inputs/Outputs

- Four stereo XLR balanced AES/EBU digital I/Os
- RS-422 port allows the DA-98HR to be controlled from an edit controller
- Control I/O allows parallel control of the DA-98HR
- XLR balanced time code In/Out provides connections for SMPTE time code sync functions DA-98
- Video In/Thru BNC connectors carry video frame sync signals when the DA-98HR is used with professional video equipment

DS-D98



The DS-D98 is the only tape-based recorder that supports Sony's Direct Stream Digital (DSD) for Super Audio CD (SACD) creation. DSD recording uses a 2.8224 MHz sampling frequency for the original 1-bit data to allow direct recording of a Super Audio CD master, producing possibly the highest fidelity audio available from today's digital technology.

The DSD-98 can be configured in both a DSD two-track and PCM multitrack format: In DSD mode, the unit functions as a 2-track recorder/player with synchronization of up to 16 units and to other DTRS recorders and features Sony's SDIF-3 digital I/O. In PCM multitrack mode, the DS-D98 features a selectable 24-bit or 16-bit recording format with 44.1, 48, 88.2, 96, 176.4, 192kHz sampling frequencies. (44.1/48kHz: 8-tracks, 88.2/96kHz: 4-tracks, 176.4/192kHz: 2-tracks). Recordings made on the DS-D98 are backward compatible with existing DTRS recordings. Additional features include: +/- 6% pitch control, an onboard test tone generator, electronic patchbay, TDIF-1 digital audio interface and 25 pin 8-channel AES/EBU digital I/O.

Otherwise identical to the DA-98HR, the 4U rackmountable DS-D98 features a confidence monitoring function, selectable reference levels, a LCD display on the front panel, set-up data memory on tape, seamless punch I/O with digital cross fade, adjustable peak hold release time level meter and an A/B head playback error rate display function. It also supports RS-422 control (P2 protocol). Its synchronization capabilities include XLR time-code IN/OUT terminal with an onboard SMPTE synchronizer, Word Sync In/Out/Thru, Machine offset in sub-frame or sample rate accuracy, and Auto Punch I/O in 0.1 frame accuracy.

TASCAM

MX-2424/MX-2424SE

24-bit 24-Track Hard Disk Recorder/Editors



One of the most popular 24-track recorders ever made, the MX-2424 offers the winning combination of excellent fidelity, sturdy construction, ease of use and digital audio file compatibility for serious musicians, engineers, studio owners and other production professionals. Extremely powerful, the MX-2424 is capable of 72 simultaneous tracks of throughput (playing back 24 tracks of audio, recording 24 tracks and doing 24 tracks that are crossfaded together at the punch points) without a glitch. Its hard disk formats and audio file types make it the most compatible standalone hard disk recorder for interoperability with leading DAW applications like Pro Tools and Logic Audio. It features a 9GB internal hard drive as well as a SCSI Wide port that supports external hard drives from up to 40 feet away. And optional analog and digital I/O cards are available so the MX-2424 can be configured to suit any environment. SMPTE synchronization, Word Clock, MIDI Time Code and MIDI Machine Control are all built-in for seamless integration into any studio. Finally, the MX-2424 is more than just a recorder. Bundled with Tascam's MX-View graphic editing software it is also a sophisticated audio editor with the editing capabilities of a digital audio workstation. Exactly the same, the MX-2424SE includes an 18GB removable front-panel SCSI drive.

FEATURES

- Record up to 24 tracks of 24-bit 44.1/48 kHz audio (or 12 tracks at 88.2/96 kHz) simultaneously using any combination of digital and analog I/O.
- Supplied 9GB internal drive allows 45 minutes of audio across all 24 tracks.
- SCSI connector in the back allows you to add external SCSI Ultra2 Wide hard drives or extended recording times and backup. There is also a front panel 5-1/2″ bay available for installing an additional drive, or an approved DVD-RAM drive for back-up, using affordable DVD-RAM media.
- Since every hard disk recorder writes a particular type of audio file format to its particular format of hard disk, Tascam chose two of the most popular formats for use in the MX-2424: Sound Designer II for HFS/HFS+ Mac formatted disks and FAT-32 Broadcast Wave (compatible with .wav files) for PC disks. Because they support time stamping, these formats make it easy to move your audio files from the MX-2424 into audio editing systems like Pro Tools, Digital Performer, Cubase and more.

Transport Controls

Whether you're a novice recordist or an engineer at the top of a commercial recording facilities, you'll appreciate the ways the MX-2424 lets you navigate through a project.

- MIDI In, Out, and Thru ports are built-in for MIDI Machine Control.
- A footswitch port lets you use a punch-in pedal, or an Alesis LRC remote control.
- 100 locate points
- Pre-roll and post-roll functions for auto-punching
- Jump ahead and jump back a few seconds with an easy key press
- Play through, from or to a captured In or Out point for easy edit point previews.
- Numeric keys that let you enter a time code value and press locate.
- Last button locates to where you last hit Play and begins playing
- Scrub/shuttle wheel lets you feel and sound like there are, even scrubbing in reverse

Editing

- Built-in editing capabilities include cut, copy, paste, split and ripple or overwrite
- 100 levels of undo from the front panel
- Offers two recording modes. And while both offer 999 virtual tracks and lightening fast punching performance, each mode provides its respective advantages.

–Loop Mode is a non-destructive recording mode (similar to a DAW) that continuously records new takes without erasing the previous version. This mode is ideal for those who want to take advantage of the extensive onboard editing tools and 100 levels of record/edit Undo.

–TL Tape Mode is a destructive recording mode that rewrites directly over the existing track or tracks during loop recording (just like a tape-based multitrack with a big fat 2″ reel of tape.). This mode maintains a predictable amount of disk space while creating single audio files per track that can easily be imported into your Mac or PC DAW and conserving as much disk space as possible.

MX-2424/MX-2424SE

Integration

- Any system compatible with SDII or Wave files can import MX-2424 audio files. If that system supports time stamped audio files, they can be placed at their original time code location with sample accuracy.
- Using a format compatible with digital audio/sequencing programs and digital audio workstations called Open TL, these systems can read and write to/from the disks used in the MX-2424 to offer playlist compatibility and recognition of virtual tracks without have to load, reposition and trim each digital file.

Comprehensive On-Board Synchronization

Designed in conjunction with TimeLine, the acknowledged world leader in synchronization tools, the MX-2424 is able to offer the best sync tools of any hard disk recorder today. With exceptionally low jitter and its capability of frame-accurate lock to incoming SMPTE time code, the MX-2424 leads the way in the integration of standalone hard disk recording into the all-digital studio.

- Word Clock In, Out and Thru ports ensure that the sampling rate of connected equipment is properly aligned.
- SMPTE synchronizer for generating or chasing SMPTE time code. This allows the MX-2424 to lock to any device that receives or generates SMPTE time code.
- MIDI In, Out and Thru for controlling the MX-2424 via MIDI Machine Control and MIDI Time Code.
- Video In and Thru: Allows the MX-2424 to lock with a blackburst signal from video.
- TL-Bus synchronization allows up to 32 machines (768 tracks at 48kHz or 384 tracks at 96kHz) to be locked with sample accuracy without any external synchronizer.
- AES/EBU (with selectable input sample rate conversion) and S/PDIF I/O to automatically clock the MX-2424 or transfer audio to digital devices
- Updates are available via a Smart Card slot in the front panel, or, downloaded directly from Tascam via the ethernet port.
- Ethernet port also lets you transfer audio files to your computer and back, as well as letting you back up entire projects.

MX-View Editing Software for the MX-2424



Offering a sophisticated graphic editing interface comparable to full-featured digital audio workstations, MX-View lends powerful editing capabilities to the MX-2424, making it the most comprehensive affordable yet professional recording/editing system ever made.

Running on both Mac and Windows, MX-View is a quick and responsive editor, with the ability to view waveforms down to the sample level. You can repair clicks and pops with the pencil tool, select and nudge audio events to the correct beat, drag and drop audio files with the hand tool, fade, crossfade and trim events on the fly.

The MX-View interface also functions as a complete system interface for as many MX-2424s as you wish. Multiple MX-2424s can be displayed and edited on the screen simultaneously… no need for a separate monitor for each machine. Plus, each machine's 24-track on-screen meters can be clearly seen from the MX-View windows, making it a thorough remote control interface.

Buttons and windows can be hidden as needed, and color-coded edit groups makes editing across multiple tracks a snap. Keyboard command sets can be configured, and windows like the transport, locate markers, and level meter can be moved around freely and stored as a user preference.

24-Channel I/O Options

A variety of optional 24-channel audio interface options including analog and a variety of digital protocols such as TDIF, ADAT optical and AES/EBU, allow you to connect the MX-2424 to any digital or analog console as well as popular digital audio recording/editing system. But that's only half the story…on the MX-2424, you can use both the analog and digital interfaces simultaneously. Using this versatile routing, you could record through the MX-2424's analog converters and monitor via a digital console. And with the exception of the ADAT Optical Interface, all MX-2424 I/O cards support 96kHz sampling rates.

- IF-AN24: 24-channel, 24-bit 96/48kHz A-D/D-A
- IF-TD24: 24-channel TDIF digital I/O
- IF-AD24: 24-channel ADAT digital I/O
- IF-AE24: 24-channel AES/EBU digital I/O

RC-2424 *Remote Control for the MX-2424*



Offers complete transport control and system set-up with the familiar front panel layout of MX-2424 (controls up to 6 MX-2424 machines

- 8 user-definable macros for frequently used keystrokes. Additional functions for controlling the TL-SYNC
- Status lights offer immediate indication of important functions
- Additional keys dedicated to editing functions, like capturing and auditioning in/out points for punch-ins or edits, allow you to work faster and more comfortably.









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The Professional Audio SourceBook

Section 5

All-In-One Workstations

Akai **514-516**

Fostex **517-519**

Korg **521-523**

Roland/Boss **524-532**

Tascam **533-535**

Yamaha **536-541**

Zoom **542-545**

DPS24

24-Track Digital Personal Studio

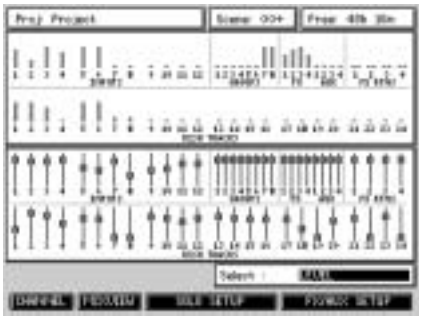
The DPS24 is a self-contained digital studio that takes the professional recording, mixing, effects and editing capabilities of the DPS16 to an even higher level. For starters, the DPS24 combines 24-tracks of uncompressed 24-bit/96kHz random access recording and a 44 channel, 20 bus digital mixer with 100mm touch sensitive moving faders that support onboard dynamic and scene automation. A wide range of inputs and outputs are available right out of the box including 28 analog inputs and an MPLP (Multi-Purpose Light-Pipe) optical interface which can be used for 8 channels of ADAT lightpipe or stereo S/PDIF I/O. 12 rotary encoders allow you to adjust a channel's entire parameter set at the touch of a button. The system's user interface can be further enhanced with the included ak.Sys Track View utility which allows real-time track displays to be viewed on any Mac or PC computer with a USB connection.



FEATURES

24-Bit/96kHz Digital Recorder

- Supports 24-tracks of linear (uncompressed) 16/20/24-bit recording and playback at 44.1/48/88.2/96kHz
- Up to 20 tracks can be recorded simultaneously without compromise to the systems performance



- 24-track recording is supported via a special transfer mode designed to allow you to make digital transfers from other sources in a single pass
- A total of 256 virtual tracks, limited only by available disk space, can be freely assigned to any of the 24 physical tracks for playback

44-Channel Digital Mixer

- 44-channel digital mixer consists of 12 input channels, 24 disk tracks and 4 stereo effects returns
- Each channel features —
  - ➢ Level, Pan, Solo and Mute
  - ➢ Compressor/Limiter/Expander
  - ➢ Noise Gate
  - ➢ 3-band EQ (hi/low shelving, sweepable mid w/variable Q)
  - ➢ L/R routing, Group 1 – 8 routing
  - ➢ Four pre/post FX sends for accessing the internal FX
  - ➢ Four pre/post FX sends for accessing external effects and/or Studio outs for talkback monitoring

12 Faders

- Twelve faders and Five fader banks
  - ➢ Bank 1 controls the levels of the 12 inputs
  - ➢ Bank 2 controls tracks 1-12
  - ➢ Bank 3 controls tracks 13-24
  - ➢ Bank 4 controls the Group/FX levels (Faders 1-8 control the Group Outs while faders 9-12 control the stereo FX returns)
  - ➢ Bank 5 is a USER bank that allows the 12 faders to be customized by the user

Built-in Automation

- Both dynamic (moving fader) and scene automation are supported
- The twelve 100mm Q-Touch touch-sensitive moving faders provide full dynamic automation of level and pan, channel on/off, FX/AUX sends and FX select, with UNDO/REDO
- The Q-Touch technology makes recording and editing mix automation data as easy as touching and moving a fader. You can easily correct any mistakes made when recording automation — simply grab the fader, make the adjustment and the original control movement will be overwritten
- Automation parameters such as levels, mutes and pans can be enabled/disabled during automation recording allowing you to capture only what you want
- A RECORD SAFE mode allows you to protect automation on individual channels from being accidentally overwritten
- In EDIT mode, when you're moving bits and pieces of audio around, you have the option of including any automation data associated with the specified audio region This allows you to maintain the integrity of the mix for that region

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DPS24

Analog Inputs

- Twenty four inputs divided into two banks provide a choice of –
  - ➢ (Bank A) Twelve balanced mic/line inputs using XLR-1/4″ TRS combo jacks
  - ➢ (Bank B) Twelve balanced line inputs using 1/4″ TRS jacks
- The desired input type can be individually selected with a push button switch located at the top of each channel strip, virtually eliminating the need for continual re-patching by allowing several devices to be connected at any once
- A Hi-Z instrument input on the front panel that overrides input 12 when a guitar or bass is connected
- Two RCA tape inputs are provided for patching cassette deck or CD player as well as two 1/4″ AUX inputs for bringing in a sub-mixer or external effects processor
- Switchable phantom power is available for input channels 1-4
- Input channels 1 - 4 also offer separate balanced 1/4″ TRS Send and Return jacks that can be used as
  - ➢ Channel inserts for patching external dynamics and EQ processors
  - ➢ Direct channel outputs
  - ➢ Direct ADC inputs — When using external mic-preamps, the ADC inputs allow you to bypass the DPS24's input stage, and route signals directly to the 24-bit A/D converters

Non-Destructive Editing

- Non-destructive Copy, Cut, Erase, Insert, Paste and Move functions are supported, with 256 levels of undo/redo, using an edit "clipboard" in much the same way as computer-based applications
- Audio regions can be copied within the same project, or between different projects
- Up to two tracks can be viewed in the waveform screen at once, making it easier to edit stereo tracks

DSP Functions

- DSP functions include phase-coherent Timestretch, Pitch Shift, BPM Match, Varispeed, Reverse and Normalize. As with the other edit functions, the user has the option of including the automation parameters when manipulating the audio

Analog Outputs

- Separate 1/4″ Main, Nearfield and Mono monitor outputs, accessible via a front panel switch, are provided for reference checking a mix
- Two 1/4″ stereo headphone outputs, with a dedicated volume control, are provided on unit's front panel

Synchronization

- MIDI In/Out/Thru connectors support MIDI Time Code (MTC), MIDI Clock
- Wordclock I/O is provided using a single BNC connector

Digital I/O

- Coaxial (RCA) stereo input and output
- A MPLP (Multi-Purpose Light-Pipe) optical interface switchable between 2-channel SPDIF mode and 8-channel ADAT mode is also provided

Optional I/O Interface Cards

- Three interface slots, on the rear panel, support a variety of I/O expansion options
  - ➢ SMPTE time code reader/generator card
  - ➢ Firewire (IEEE 1394)
  - ➢ 68 pin SCSI (50 pin SCSI is standard)
  - ➢ 16 channels of ADAT In/Out with sync

Akai's ak.Sys Control and Networking PC/MAC Software



- The USB interface provides a realtime SVGA TrackView display using ak.Sys software installed on a Mac or PC as well as networking capabilities with other Akai products, and software updates
- Using the optional 1394 Firewire interface board, ak.Sys will support VST plug-ins, allowing the ak.Sys computer to be used as a realtime effects engine for the DPS24

Dedicated Controls

- A generous complement of dedicated function keys, transport buttons and rotary encoders are available
- Multi-function rotary encoders located above the faders have a collar of LEDs around them that let you see their values at a glance
  - ➢ Hit the Q-STRIP keys and use the 12 encoders to adjust pan or effects sends
  - ➢ Hit the Q-CHANNEL key and use the 12 encoders to adjust a channel's entire parameter set
- A dedicated multi-track jog wheel for 'reel-rocking' and you begin to get the feeling that it's you that's in charge, not the machine's arcane operating system

Media & Storage

- Equipped with a large capacity 60 GB internal IDE drive that provides over 5 hours of audio on 24 tracks at 48kHz, (12 tracks at 96kHZ) — Additional IDE drives are available at low cost
- The built-in SCSI interface allows you to link up to seven additional external SCSI drives providing the flexibility of recording cuts of any length, limited only by the size of the selected storage medium
- A built-in 5.25″ drive bay supports an optional CD-R/CD-RW drive, providing a cost-effective means of project archiving as well as allowing users to produce professional quality audio CDs



Conveniences

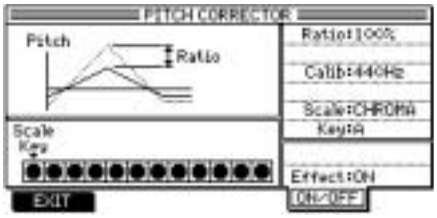
- 5.1 surround mixing will be supported using analog the Stereo/Main/Near or digital (MPLP) outputs
- ASCI keyboard input for track/project naming
- The 1/4″ footswitch port supports the Alesis ADAT LRC (Little Remote Control)

DPS24

Internal 4-Bus Effect Section

- 4-bus 56-bit multi-effects processors offer 44 types of high quality, fully editable effects including a wide range of reverb, modulation, delay, EQ, dynamics processors and combination effects as well as a multi-band dynamics processor with 4-bands of compression or expansion, adjustable cross-over points, adjustable slopes and reduction metering
- Effects can be inserted to any input or playback channel or routed (pre or post fader) from each individual channel via AUX send / return 1–4

Realtime Pitch Corrector



- The Pitch Corrector algorithm allows you to correct the pitch of monophonic tracks, such as a vocal, in real-time
- Parameter adjustments are provided for controlling the amount of pitch correction, vibrato, root note and tuning calibration from 415Hz – 466Hz (default is A440Hz)
- The Scale parameter allows you to choose from 13 different scales plus two user-programmable scales
- An external MIDI keyboard can be used to control the target pitch in real-time

Effects Types

Reverb Effects
- Small Room
- Big Room
- Small Hall
- Big Hall
- Studio
- Live House
- Medium Hall
- Bright Hall
- Non-Linear
- Reverse

Modulation Effects
- Mono Chorus
- Stereo Chorus
- X-Over Chorus
- Mono Flanger
- Stereo Flanger
- X-Over Flanger
- Pan Flanger
- Mono Phaser
- Stereo Phaser
- X-Over Phaser
- Pan Phaser
- Pitch Shift
- Rotary Speaker
- Auto Pan
- Trigger Pan

Delay Effects
- Mono Delay
- Ping Pong Delay
- Panning Delay
- Stereo Delay
- X-Over Delay
- Tape Echo
- Multi Tap Delay

Dynamics Processing
- Compressor
- Limiter
- Expander
- Noise Gate

EQ
- Digital EQ
- Autowah
- Touch Wah
- Enhancer

Combination Effects
- Chorus / Delay
- Flange / Delay
- Phaser / Delay

Other Effects
- Distortion
- Pitch Corrector

What is Non-Destructive Editing?

When a section of audio is copied from one location to another, that audio is re-written on the disk and exists in two separate locations on the drive. This is an inefficient use of disk space and makes your recorder work hard much than it needs to when accessing audio. With non-destructive editing there is no duplication, or re-writing, of the audio on the disk.

Instead, non-destructive editing uses pointers to reference audio on the disk. So when you "copy" the audio you are only adding a second set of pointers that reference the exact same audio on the disk, not re-writing the same audio to another location on the disk. This makes edit functions much faster than on many portable studio products, as well as making the most efficient use of your disk space

Q-Link Navigation System

- The Q-link Navigation System is a row of assignable encoders, above the channels, faders designed to provide quick and direct access to all major mixing functions including EQ, Aux send levels, and effects parameters, without the need to wade through complex menu layers. Encoders can be quickly assigned to make adjustments to FX/AUX Send levels, or adapted to become a full Q-Channel channel strip providing access to all parameters of an individual channel
- The large 6-inch diagonal (320 x 240) graphic LCD display, provides precise feedback of how a track is shaping up with a graphic representation of audio cues on a scrolling track sheet

Enhanced Q-Link Navigation

- Q-Link features make searching through complex layers of menus a thing of the past by providing direct access to all major Record, Edit, Mix, and Effect functions
- 6 Q-Knobs located on the large tiltable LCD display offer easy realtime access to effects parameters and system control.

Editing

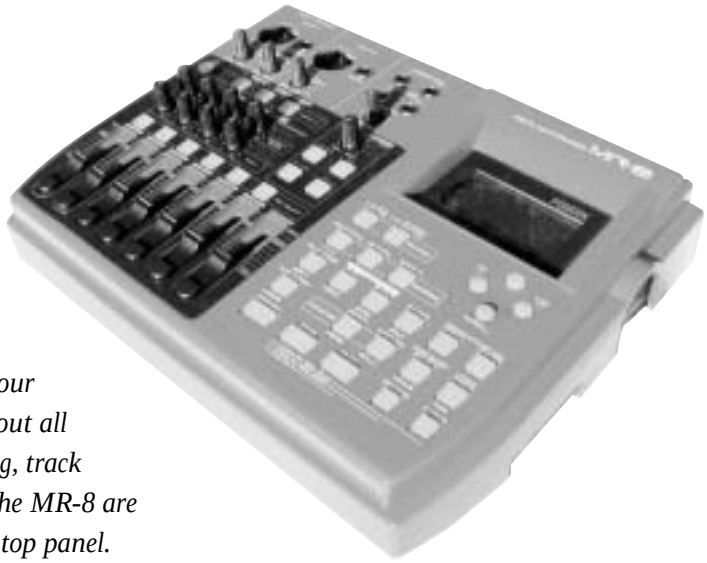
- Graphic waveform editing is provided down to the sample level with up to 256 levels of UNDO
- A number of editing functions can be applied to single or multitrack sections of audio including — copy-overwrite, copy-insert, cut-overwrite, cut-insert, insert-silence, cut-discard, cut-move
- A choice of 54 preset timestretch algorithms provide an unprecedented variety of timestretch options each of which are capable of delivering phase coherent processing of stereo recordings
- You are also able to normalize and change the levels of audio regions up or down

Locate Features

- Up to 100 locate points in a song can be named and stored. A "Direct Locate function" that assigns locate points to the keys on the front panel is also available
- Both Manual Punch In/Out (using the transport buttons or a footswitch) and Auto Punch In/Out (at specified In and Out points) are supported

MR-8

8-Track Digital Multitracker



The Fostex MR-8 is an affordable, ultra-portable eight track digital multitracker which records and plays back 8 tracks of uncompressed audio using a CompactFlash media card. Audio can be recorded at high quality 44.1 kHz/16 bits or 22.05 kHz/16 bits in the extended mode. The MR-8 also features an 8-channel digital mixer, digital effects including the delay/reverb, mastering effects and guitar/bass amp and mic preamp simulation insert effects. The four track cassette-style user-interface allows you to intuitively carry out all the process for digital multitrack recording including overdubbing, track bounce and mixdown – nearly all of the functions available on the MR-8 are accessed directly from physical knobs and buttons located on the top panel.

FEATURES



- 8-tracks of recording and playback using readily available Compact Flash cards (a 128MB card is included)
- 16-bit Delta Sigma A-to-D and D-to-A converters and high quality sound uncompressed linear 16-bit/44.1 kHz digital audio recording and playback (normal mode).
- 2-track simultaneous recording and 8 tracks playback
- Approximately 25 track-minutes of recording time is available using 128MB Compact Flash Card in normal mode (16-bit/44.1kHz). Extend Mode (16-bit/22.05 kHz) doubles the available recording time

Inputs and Outputs

- Two analog input channels –

  Channel 1 gives you a choice of a balanced XLR mic input or a 1/4″ unbalanced line or Hi-Z guitar input Channel 2 Gives you a choice of a balanced XLR mic input or a 1/4″ unbalanced line input
- 1/4″ TRS main/monitor outputs
- Two 1/4″ stereo headphone outputs with level control
- Optical S/PDIF digital output

Effects

- Digital effects include delay and reverb using a newly developed algorithm for the MR-8
- Guitar Amp simulation (Brit Stack, U.S. Metal, and 60'S Combo) with analog style distortion on input A at "Guitar" position) and Microphone simulation (Dynamic, Condenser, Tube) on input A at "Line/Mic" position) as insert effect
- A mastering effect is provided on the stereo bus that allows you to process a song after it has been bounced or during mixdown.

Intuitive Interface

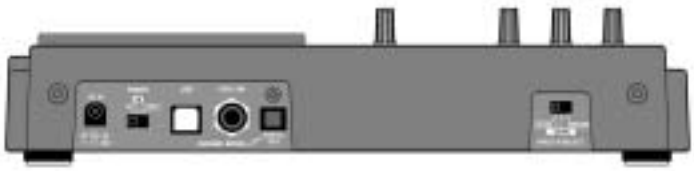
- Intuitive, four track cassette-style operation is ideal for new comers to the world of digital multitrack recording.
- 128 x 64 dot-matrix LCD and self illuminating keys show current operating status
- Using the built-in USB port, on the rear panel, and the free MR-8 WAV manager software allows you to transfer WAV file data between the MR-8 and a Windows PC (XP/2000/ME/98SE) for data editing, CD burning (normal mode) and backup

Editing

- Two locate points are provided for executing a variety of editing functions including auto-punch in and out recording
- Erase, Copy, Paste, Move and Exchange mono or stereo track data with one level of undo/redo.
- Part editing allows you to Erase, Copy, Paste, Move and Exchange data from within two locate points of a track. Part editing also allows one level of undo/redo.

Conveniences

- Powered by 6 AA alkaline batteries (provides just over 2 hours of normal use) or with the included AC adapter
- A built-in microphone allows instant access for recording ideas or events.
- The metronome function generates guide clicks according to the tempo map.
- MIDI output allows you to transmit MTC or MIDI Clock for synchronization with another MIDI compatible device such as a drum machine or computer.

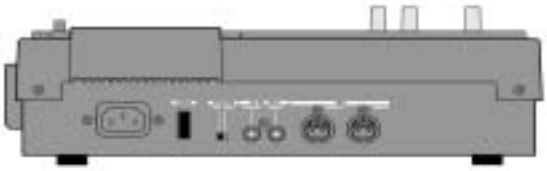


FOSTEX

VF80 • VF80CDR

Digital Personal Studio

The Fostex VF80 is an affordable all-in-one digital multitracker that provides 8-track digital recording, an intuitive digital mixer, A.S.P.+ digital effects with mic and guitar amp modeling, 99 scene memory, waveform display, XLR inputs with phantom power and an optional internal CD-R/RW burner. Fostex's proprietary FDMS-3 technology enables more efficient disk and CPU usage allowing you to obtain a greater amount of recording track minutes per drive without data compression, and faster editing and DSP operations.





FEATURES

- 8 playback tracks plus an additional 16 ghost tracks that allow you to record multiple takes and choose or "comp" the best performances.
- 20-bit A-to-D converters and 24-bit D-to-A converters.
- Internal 3.5″ 20GB IDE hard drive provides 64 total track hours of uncompressed 16-bit digital audio at a sample rate of 44.1kHz.
- CD-1A internal CD-RW drive (optional in the standard VF80) for mastering, data back-up and WAV file import and export.

Digital Mixing

- Eight 60mm faders including six mono faders (channels 1 -6), one stereo fader (channels 7 and 8) and a Master fader for the stereo bus.
- Two-band shelving EQ with ±18db boost and cut (at 100 and 10k Hz) on channels 1-6 with access to a 36 preset EQ library.
- The 99 mix scene memory, with title edits, allows you to store an entire mixes fader positions, EQ and effect settings for later recall or automated scene changes during final mixdown.

Effects

- A.S.P.+ digital channel effects provide 44 types of effects including reverb, panning, hall, reverb, chorus, flange, and pitch shift effects as well as mic modeling, guitar amp simulations and distortion.
- The Bounce function allows audio from tracks 1 through 6 to be bounced to tracks 7 and 8 at the touch of a button.
- Mastering Mode allows you to apply one of 10 preset Mastering Effects, including EQ, compression and effects to tracks 7 and 8 during your final mix to the internal CD-RW drive or an external recorder.

Editing

- Non-destructive Copy, Move, Paste and Erase audio editing with undo/redo.
- Find precise editing points using the waveform display and audio scrubbing using the onboard jog wheel.
- Clipboard data can be automatically repeated up to 99 times.
- Auto punch in and out.
- ±6% varipitch control.

Inputs and Outputs

- Two simultaneous inputs,with trim controls, on/off switches and peak LED indicators, including two XLR inputs with phantom powered and two unbalanced 1/4″ instrument inputs.
- Left and right 1/4″ main outputs and a 1/4″ stereo headphone output with level control.
- Coaxial S/PDIF digital I/O.
- 1/4″ footswitch input provides hands-free manual punch I/O.

Training Mode

- Center signal cancellation and full control of audio speed (tempo) without altering the pitch can be applied to tracks 7 and 8 allowing you to practice along with a stereo track.

Synchronization

- MIDI In and Out connectors allows synchronization and control using MTC, MMC and MIDI clock as well as song position pointer.

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VF160CDR

Digital Personal Studio



The VF160 is a 16-track digital recording, non-destructive editing, mixing and mastering with an internal CD-RW drive included as standard equipment. Up to sixteen tracks can be recorded simultaneously to the included 20GB internal IDE hard drive. The waveform display and 16x zoom feature provides precise editing. The digital mixer features 60mm faders, 3-bands of EQ,solo-in-place as well as two effect and two aux sends for all 16 channels with a 99 mix scene memory. Other features include two independent multi effect DSPs packed with a preset library of great sounding algorithms. And not only are there 8 analog inputs, including two XLR mic inputs with phantom power, there is also an 8 channel ADAT optical I/O that will interface with a host of computer-based DAWs, digital mixers as well as a host of outboard processors.

FEATURES

- 16 playback tracks and 8 track simultaneous record plus 8 additional ghost tracks for multiple takes (16-track simultaneous a recording is available when using ADAT interface in addition to analog inputs).
- 7 on the fly marker points, 7 Edit Memory & 99 Event Memory.
- Internal CD-R/RW drive.
- .WAV file export and import.
- 99 programs (songs).
- 3.5" IDE hard drive bay includes a 20GB drive and is upgradeable to accommodate larger drives.

Digital Mixing

- 17 (60mm) input faders – 8 x analog/track inputs, 8 track inputs and 1 x Master fader (stereo buss).
- Fader pairing allows channels to be grouped in stereo sets.
- Solo, 3-band EQ and plus a Low Pass Filter on all 16 channels.

ADAT Mixer Mode

- Use the VF160 as the front-end, mixer and control surface for your PC/Mac-based audio/midi workstation equipped with an ADAT card and a MIDI interface.
- Two channel compressors assignable to channels 13/14 or 15/16 with EQ off).
- Master compressor and 3-band EQ.
- Two effect sends and two pre/post switchable AUX sends.
- Two built-in Fostex A.S.P. digital effects.
- 100 mix scenes per program.

Inputs and Outputs

- Eight balanced/unbalanced 1/4″ TRS inputs (channels 1 - 8).
- Channels 7 and 8 also give you the option of two XLR inputs with phantom power or 1/4″ instrument inputs plus 1/4″ TRS inserts for injecting outboard processors into either signal.
- Unbalanced 1/4″ L/R Monitor outputs and a 1/4″ stereo headphone outputs.
- Two AUX Sends via single 1/4″ TRS connector (TIP: send 1 / RING: send 2).
- Unbalanced RCA L/R outputs.
- Optical digital I/O switchable between eight channel ADAT and S/PDIF.
- 50-pin SCSI-2 connector supports additional storage devices including external CD-RW drives.

A.S.P. (Advanced Signal Processing Technology)

The A. S. P. is Fostex' proprietary digital effect processing technology designed to provide maximum efficiency from a small amount of DSP power. The H. F. A. (Harmonic Feedback Algorithm) achieves a rich high density Early Reflection sound and wonderfully smooth High Dump response while the H. D. L. P. (Hi-Density Logarithmic Processing) delivers an elaborate reverb simulation with clear sounds without the graininess caused by numerous integrated delay module interference.

KORG

D1200CD / D1600 V40

MULTITRACK RECORDERS

520

Digital Recording Studios



The D1200 and D1600V40 are all-in-one, digital audio workstations that deliver uncompressed 16- and 24-bit direct to hard disk multi-track recording, automated mixing, powerful built-in REMS effects processing, a wide range of I/O and routing flexibility, as well as intuitive non-destructive editing capabilities. The D1200 offers 12 tracks of audio playback at 16-bit (6 tracks at 24-bit) and the D1600V40 provides 16 tracks of playback at 16-bit (8 tracks at 24-bit). The D1200's 16 channel, 4 bus digital mixer and the D1600V40's 24 channel, 8 bus digital mixer each feature level, pan, mute, solo, 3-bands of EQ as well as effects inserts and 3 effects sends on each channel. Mixer, EQ and effects settings can be stored within each song as scenes and even controlled in realtime via MIDI and external controllers. Balanced XLR and 1/4˝ TRS analog inputs with high-performance balanced preamps will allow a variety of audio sources to be connected directly to the systems, ranging from mic level to a +16 dBu level. A dedicated 1/4˝ Hi-Z input is also provided for direct connection of a guitar or bass. The internal CD-R/RW drive, included with both units, makes it possible to backup/restore data, or to create audio CDs in a single integrated package — Go from recording to effect processing to mixing down to CD while staying completely in the digital domain.

PASE (Pure Audio Sound Engine)

- Both units offer uncompressed 16-bit and 24-bit record and playback at 44.1kHz
- The D1200 offers 4 simultaneous record tracks and 6 playback tracks at 24 bit or 4 simultaneous record tracks and 12 playback tracks at 16-bit
- Each of the D1200's 12 tracks offers 8 virtual tracks, for a total of 96 tracks per song
- The D1600V40 offers 4 simultaneous record tracks and 8 playback tracks at 24 bit or 8 simultaneous record tracks and 16 playback tracks at 16-bit
- Each of the D1600V40's 16 tracks offer 8 virtual tracks for a total of 128 tracks per song

The Virtual Track Concept

- Allows you to record multiple passes of instrumental or vocal performances which can then be compared, allowing you to choose the best one or the best phrases from each take can be edited to create the perfect take
- Choosing virtual tracks as the recording destination when bounce-recording allows you to combine the data of all tracks, including effects and EQ into two tracks without losing any data

Digital Mixer Section

- The D1200 features 16 mixer channels which are divided into four input channels and 12 playback tracks. The D1600V40 has 24 mixer channels divided into 8 simultaneous input channels and 16 playback tracks
- The D1200 has 9 channel faders (6 mono and 3 stereo) with pan controls and a master fader that provide smooth, precise level adjustments over individual channels and the overall output level. The D1600V40 has 16 channel faders with pan controls
- Each channel has its own Level, Pan, Solo and Mute controls, Phase (polarity) reverse, a 3-Band EQ as well as an effects insert, two Master effects sends and an Aux send for accessing an external effects processor
- During mixdown, input channels on can be used for submixing external audio sources such as sequenced MIDI tracks as well as outboard effects processors
- A Pairing function allows adjacent channels to be grouped as a stereo pair – ideal for stereo synths and drum loops

Serious EQ

- Three bands of EQ, each with a boost/cut range of ±15dB, are provided for each analog input and mixer channel
  - ➢ High shelving band at a fixed frequency of 10kHz and a Low shelving band at a fixed frequency of 100Hz
  - ➢ Sweepable Mid band with a frequency range from 100Hz to 20kHz
- Because EQs are separate for the input and mixer channels you are able to assign different EQ settings for recording and playback

Mix Automation

- Mixer settings such as fader, EQ, pan and effects can be stored in a "scene" — Up to 100 scenes can be stored per song
- The Scene playback function allows you to assign scenes for automatic recall at the time locations you specify within the song
- Automation data can be sent and received in realtime, as controller data via a MIDI compatible sequencer, keyboard or computer

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D1200CD / D1600 V40

Analog Inputs

- A variety of analog audio sources can be connected directly to the mixer and instantly routed to any channel, ranging from mic level to a +16 dBu level.
- All inputs feature high-quality balanced preamps with individual Trim controls and overload LEDs.
- The D1200 has two 1/4″ TRS and two XLR / 1/4″ TRS combo connectors.
- The D1600 has four balanced XLR / 1/4″ TRS mic/line inputs and four balanced 1/4″ TRS inputs.
- Individually switchable +48V phantom power which is provided for each of the D1600V40's four XLR mic inputs.
- Both units also provide an unbalanced 1/4″ Hi-Z input which allows you to plug a guitar or bass directly into the units.
- Analog inputs and the 24-bit S/PDIF input can be utilized as Sub Inputs that can be used to send external audio sources such as keyboards, samplers or effects processors directly to the Master bus.

Non-Destructive Editing

- Operations such as copying, pasting, or erasing phrases are as easy as specifying the IN, OUT, and TO locations.
- The Scrub function lets you listen to the recorded sound of a track as if you were manually rocking the reels of an open-reel tape recorder, giving you precise control when editing tracks and for finding the exact punch points.
- 99 levels of undo/redo makes it easy to compare complex edits and arrangements.
- To make the most efficient use of hard disk space, you are able to select between 1, 8 or 99 levels of Undo.
- Four Locate points and 100 Mark points with names) are available per song providing quick and easy access to playback and editing locations.
- The Normalize function lets you boost the overall level of track to the maximum volume without clipping.
- The Time Expansion/Compression function makes it possible for phrases of differing tempo to be matched after they have already been recorded, or to make a phrase fit into a specific time.



Scrub Mode

- Optimize Track Data creates one piece of audio from several "fragmented" regions, resulting in perfect track playback and better conservation of hard disk space.
- Reverse playback can be applied to single or multiple tracks and copied to new audio regions or tracks.
- Single and multiple tracks can be faded in and out or even crossfaded using one of six fade curves.

Outputs & Monitoring

- Separate L/R Master and Monitor Outputs via unbalanced RCA phono jacks on the D1200 and 1/4″ jacks on the D1600V40.
- A 1/4″ Auxiliary Output accessed by the Aux send on each channel can be used for connecting an external effects processor.
- A 1/4″ stereo headphone output with level control is provided for both units.
- The Monitor function lets you create a monitor cue mix for all tracks that includes level and pan settings that are independent of the mix routed to the Master outputs.

Effects Section

- Each song can contain up to eleven simultaneous effects programs —
  - ➢ Up to eight insert effects, including guitar, bass and vocal multis, can be inserted into an any analog input or mixer channel.
  - ➢ Two master effects can be applied to the send from each channel.
  - ➢ A final stereo effect can be applied to the master L/R as the last stage.



Guitar Multi

- Each effect program is created with one of 98 different algorithms using Korg's REMS (Resonance structure and Electronic circuit Modeling System) technology.
- 192 preset effects (128 insert effect presets, 32 master effect presets, and 32 final effect presets) created by professional musicians and studio engineers.
- Preset programs can be editing and stored in 192 user program locations.
- An expression pedal, available separately, can be connected to control an insert effect in realtime.

Digital I/O

- 24-bit optical S/PDIF I/O includes a built-in sample rate converter that automatically converts 48 or 32 kHz sources to 44.1kHz.

CD Burning

- The internal CDRW-1X8 slim-line CD-R/RW drive makes it possible to backup and restore data, to create audio CDs, as well as playback and record directly from an audio CD.
- A CD burner with the same capabilities is included with the D1600V40.



CD-R/RW

D1200CD / D1600 V40

Media & Storage

- Both units include an internal 40 GB hard drive providing up to 62 hours of total audio data at the 16-bit resolution
- The D1200 has a built-in USB interface that allows direct connection with a computer. A 2GB partition of the 40GB hard disk is reserved to allow simple drag and drop data transfer between the D1200 and the computer using the .wav file format
- The D1600V40's user-installable 3.5″ hard disk cartridge/bay system supports affordable, readily available IDE hard disks that can be exchanged when switching between projects. A 50 pin SCSI-2 port supports a variety of fixed and removable drives providing additional record, playback and data back-up options
- Data backed up from earlier D-Series digital recording studios can be restored into the D1600V40, allowing easy transfer of data. Playable D12, D1200 and D16 drives can also be played back by the D1600

Metronome

- A built-in Metronome with 96 metronome patterns and 215 PCM-based drum patterns covering many different genres of music provide inspiring tempo guides when laying down tracks
- Drum patterns can even be recorded to tracks and used within your songs

Control Surface

- The comfortable size and logical layout of the control surfaces make navigation on both units easy and intuitive
- Both units provide a full range of transport controls including scrubbing as well as dedicated buttons that provide instant access to system functions
- A large 240 X 64 pixel backlit display provides clear access to setup and editing operations, including precision graphic waveform editing
- The D1200's display flips up for easy viewing
- A number of dedicated LED indicators provide visual feedback of system activity
  - ➢ Hard disk / CD-R / MIDI activity
  - ➢ Track status (play, input, rec, mute)
  - ➢ Transport status

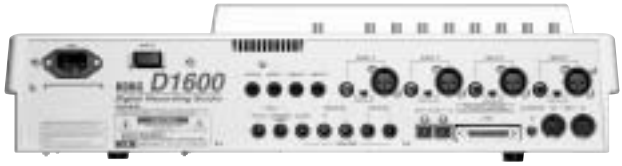


TouchView Display

- The D1600V40 features a 240x64 pixel, backlit TouchView LCD display system which allows you to select pages, tabs, and parameters, and set their values by pressing their related objects on the screen. Also displayed, during recording or playback are level meters and time locations as well as various other parameters

Synchronization & Tempo Control

- MIDI In/Out connectors allow you to:
  - ➢ MTC (MIDI Time Code) and MIDI Clock can be sent or received allowing the units to synchronize with an external sequencer
  - ➢ Use MMC (MIDI Machine Control) to allow Stop/Play/FF/Rew/Record/Locate functions to be sent and received between the units and an external sequencer
- Up to 200 tempo map events, that can be dialed in numerically or created using Tap Tempo or MIDI Clock, are available per song, allowing tempo, time signature and rhythm patterns to change during recording and playback





Conveniences

- The Trigger Recording function allows recording to begin automatically in response to an audio input
- The Auto-Save Feature ensures that song data will be automatically saved automatically when selecting new songs and on power-down
- Built-in auto chromatic tuner – In addition to tuning a guitar or bass, you can even check the tuning of a playback track
- Auto and manual punch-in/out recording makes it easy to re-record just the desired portion of a performance
- Loop recording and playback lets you set specific in / out points within a song and loop the audio. This allows you to:
  - ➢ Rehearse intricate parts to be recorded
  - ➢ Use the punch-in function while looping, and add new data as the loop plays
  - ➢ Use the loop playback feature to audition specific regions of recorded audio
- The Program Play function allows you to playback multiple songs in any order, from the internal drive and external drives

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EQUIPMENT LEASING AVAILABLE

PXR4

Digital Recorder



The PXR4 delivers 4 tracks of high-quality digital recording and mixing, pro-level editing functions along with great sounding digital effects in a compact, ultra-portable package. The PXR4 records directly to readily available SmartMedia cards from 4MB to 128MB — a 16MB card is included. A full compliment of onboard I/O is provided including a dedicated guitar input, stereo mic/line inputs, stereo line and headphone outputs are provided as well as a USB port that allows you to transfer files between the PXR4 and your computer in stereo MP3 format. The PXR4 can be powered by two AA batteries or the included AC adapter. There's even a built-in stereo mic ready to capture your musical ideas at a moment's notice.

FEATURES

Recorder Section

- Record 4-tracks of high-quality digital audio directly to SmartMedia cards from 4 to 128 MB - a 16MB card is included
- Three recording modes afford maximum flexibility by enabling you to select the one that's right for the task at hand depending on the recording quality and time you need.
  - ➢ Standard mode provides an excellent balance between recording quality and memory use
  - ➢ High-quality mode captures your performance with impeccable clarity
  - ➢ Economy mode delivers maximum recording time

Professional Editing

- Eight virtual tracks per track allow you to record multiple takes of the same track and select your favorite one
- Equipped with everything you'd expect from a digital recorder many times larger including easy-to-use punch in/out, bounce, copy and delete functions even time compression and expansion

Intuitive Interface

- Five faders, one per track, plus a master fader as well as dedicated controls are provided for all key features making parameter selection and editing easy
- Clear, backlit display uses an intuitive icon-based user interface



Inputs & Outputs

- 1/4″ input with a hi/low impedance switch
- Stereo mic/line input
- 1/8″ stereo output jack, plus a stereo headphone output
- Using the built-in USB port, you can transfer song data to and from your computer in stereo MPEG format

Digital Effects

- 77 different built-in effects, based on Korg's proprietary REMS technology deliver ultra-realistic models from guitar amps and pedals to world-class microphones as well as studio essentials like reverbs, delays, modulation effects and more
- 100 preset and 100 user multi-effects programs with up to five simultaneous effects

Metronome / Rhythm Patterns

- 55 different rhythm patterns plus 36 metronome patterns, using high-quality PCM sounds are provided for use as a recording guide, or to create the actual rhythm track for your song

Included Accessories

- Model PXRC-00S 16MB SmartMedia card
- AC Adapter
- Soft Case

Application Note!

Not only is the PXR4 a great multitrack recorder, it is also an outstanding tool for practicing guitar —

- ➢ Hook up a CD player to the line inputs and record a song you want to learn
- ➢ Plug in your guitar and dial in the appropriate amp model and effects
- ➢ Use the Time Compression/ Expansion to slow down the tempo of a solo by half without altering its pitch
- ➢ Use the rhythm and metronome patterns to practice your rhythm playing and scales

BR-532



Portable Digital Studio

The BR-532 is a compact and affordable battery powered 4-track digital workstation designed to give the guitarist and songwriter "on-the-go" the freedom to record whenever and wherever inspiration strikes. Up to four tracks of simultaneous playback is provided along with 32 Virtual Tracks per song. Audio is recorded directly to affordable SmartMedia cards (a 32MB card is included). Plug your dynamic mic, guitar, bass or keyboard directly into the BR-532 and record using the high quality COSM amp modeling effects without the need for external processing gear. There's even a built-in microphone. A separate Rhythm Guide track keeps the groove going with a choice several stereo drum patterns available in a variety of styles. The four playback tracks and rhythm guide can be mixed (bounced) down to a stereo file along with onboard effects.

FEATURES

- Four simultaneous playback tracks and two simultaneous record tracks
- A total of 32 Virtual Tracks (8 per track) are available for capturing multiple takes
- Audio is recorded to ultra-thin 3.3V SmartMedia cards — cards up to 128MB are supported — a 32MB card is included
- When more tracks are needed, you can digitally bounce all four tracks with effects to a single or pair of virtual tracks
- Digital non-destructive track editing tools include Copy, Paste, Move, Erase with one level of Undo/Redo
- Three locator points provided instant access to specific edit points and Auto punch In and Out points
- Manual punch In/Out is also provided
- 24-bit A-to-D and D-to-A converters and 32-bit internal processing

Mixer and Transport Controls

- There are four volume faders for each channel and a master fader
- A dedicated pan button lets you control the L/R position of each playback track
- Besides standard Play, Stop, Record FF and Rew controls, a Time/Value dial is provided that lets you shuttle to the desired song position as well as adjust the values of various editing functions

APPROX. TOTAL RECORDING TIMES IN MINUTES			
Recording Mode	**HiFi**	**Standard**	**Long**
32MB Card	16	19	24

Inputs and Outputs

- Balanced XLR and 1/4″ TRS mic inputs allow you to connect a dynamic mic without adapters
- A dedicated Hi-z 1/4″ instrument input is provided for plugging a guitar or bass
- Left and right unbalanced RCA line inputs
- Left and right unbalanced RCA line outs are available for mixdown or monitoring
- A 1/4″ stereo headphone output
- A optical digital output lets you record directly to a computer, standalone CD recorder or other digital device
- A MIDI output allows you to sync the BR-532 with your drum machine, groove sequencer or computer

Two Types Of Editable Stereo Effects

- The first effect is an **Insert Effect** that can be applied during tracking with a wide variety of COSM amp models and effects patches designed specifically for recording guitar, bass, vocals and keyboards without relying on external amps and processors
- The second effect type is a **Loop Effect** which allows you to apply spatial effects like reverb, chorus, flanging and doubling to individual playback tracks during mixdown. Each playback track has its own send control for adjusting the depth of the effect for that track

Truly Portable

- A built-in microphone allows you to capture ideas anytime, anyplace even when you don't have a mic with you
- Up to 7 hours of on-the-go recording using 6 "AA" alkaline batteries (included) – an optional PSA-120 AC adapter is available

Rhythm Guide

- The built-in Rhythm Guide, with tap tempo button and dedicated volume fader, provides 8 drum kits using real PCM-based sounds and patterns, in a variety of styles, that can be used during recording and even bounced along with the four playback tracks
- Multiple patterns can be combined to create a full rhythm track

Additional Features

- 20 character by 2 line backlit LCD display
- An Auto Sub Mix switch on the rear panel, allows you to monitor the line inputs while recording from the guitar or mic inputs
- The Phrase Trainer lets you slow the tempo of guitar licks without changing pitch
- The Center Cancel function removes lead vocals from a mix
- A built-in chromatic guitar and bass tuner is available via a dedicated tuner button
- A 1/4″ footswitch jack accommodates start and stop playback, punch in/out recording and effects bypass

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BR-1180 / 1180CD



Digital Recording Studios

The BR-1180 is a an affordable digital workstation that offers ten playback tracks and 80 Virtual Tracks plus a stereo mixdown track — all recorded to an internal 20GB hard drive. Using the optional CD-RW drive (included with the BR-1180CD) for burning your music to CD, you've got a powerful system with true all-in-one flexibility. Other features include a large icon-based user interface, phantom power for the XLR mic inputs, MIDI controllable faders and recallable mixer/effects scenes. The built-in rhythm guide lets you assemble great sounding drum parts using high quality PCM sounds. You can also import .Wav loops and automatically match them to the tempo of your song, making it easier than ever to have a great sounding rhythm section.

SAME FEATURES AS THE BR-532 *PLUS*—

- Two track simultaneous record and 10 playback tracks – tracks 1 - 8 are mono tracks, tracks 9 and 10 are combined as a stereo pair
- A dedicated stereo Master Track delivers easy digital mixdown and track bouncing
- 80 Virtual Tracks (8 per track) for working out different arrangements and capturing the perfect solo
- The internal 20GB hard drive supports 16-bit linear uncompressed recording as well as Roland's proprietary modes

Internal CD-RW Drive

- The optional user-installable CD-RW drive (included with BR-1180CD) provides high speed data backup and audio CD creation of your mixes. You can also use the CD-RW drive to import .WAV files such as drum loops to be used within a playback track or the rhythm track.

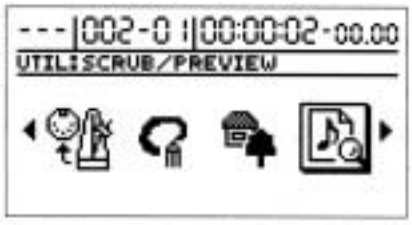
Enhanced Mixing and Transport Functions

- Each playback track has its own volume fader (channels 9 and 10 share a single fader) as well as pan and two bands of sweepable EQ
- You can store and recall up to 8 mixer and effects setting configurations per song
- MIDI input allows you to control and automate fader movements from an external controller or sequencer
- 1000 markers help indicate you instantly navigate thru your song along with audio scrubbing function make it easy to do quick and precise edits.

Input and Output Flexibility

- Two XLR and 1/4″ TRS mic inputs with switchable phantom power available to the XLR inputs
- 1/4″ expression pedal input (compatible with the Roland EV5) allows realtime control of effects parameters such as Wah-Wah

Icon-Based User-Interface



- 64 x 40mm icon-based backlit LCD interface simplifies the recording, bouncing and mixing process by providing concise visual information of the current status

Rhythm Guide and Loop Phrase Sequencer

- The Loop Phrase Import feature allows you to create and edit loop-based audio phrases within an audio track. You can also import loop-based phrases (including .Wav files imported via CD-RW) into the rhythm guide for a fast way to arrange them to fit your song's form and tempo.
- Additional rhythm patterns can be loaded via MIDI as Standard MIDI Files.

Effects

- One Insert Effect and two Loop (send/return) Effects processors – one for chorus/delay and the other for reverb
- COSM Speaker modeling

Mastering Tool Kit



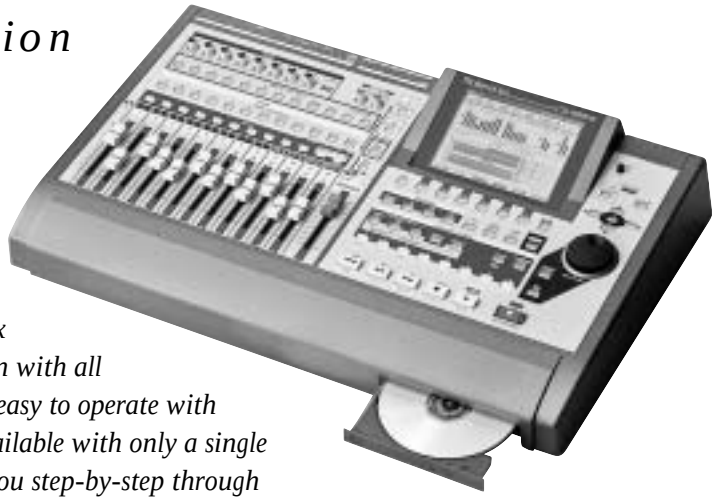
- The Mastering Tool Kit provides nine fully-editable processing stages including EQ, multi-band dynamics and limiting designed specifically to help you create the perfect final mix
- The signal first enters the EQ section which includes a five band parametric EQ, Low Cut Filter and Enhancer. The signal is then split into three frequency bands (Low, Mid, High) each with its own level control. Each frequency band is then sent to its own Expander and Compressor and then into a mixer section which allows you to readjust the volume of the three frequency bands. Finally the signal is sent to a limiter and a soft clip algorithm for achieving level maximizing for your mix without distortion.
- The final signal can be dithered between 8 and 24 bits.

APPROX. TOTAL RECORDING TIMES IN MINUTES			
Uncompressed (16-bit): 60 hours			
MT1:	120 hours	**MT2:**	160 hours
LV1:	190 hours	**LV2:**	240 hours

ROLAND

VS-1824CD

24-bit Digital Studio Workstation



The VS-1824CD is an affordable all-in-one 24-bit virtual studio with 18-track playback, 28-channels of automated digital mixing, onboard effects (including COSM guitar amp, microphone and speaker modeling), precision editing functions and built-in CD burning capabilities. Up to eight tracks can be recorded simultaneously using a wide range of high quality inputs including two XLR mic inputs with phantom power, six 1/4˝ TRS mic/line inputs and a dedicated 1/4˝ guitar input. Even with all of this power and sophistication, the VS-1824CD is surprisingly easy to operate with many of the system's most important and innovative features available with only a single button push and Roland's own EZ-Routing functions that take you step-by-step through the process of configuring recording, mixing, bouncing and mastering set-ups.

FEATURES

True 18-Track Recording

- 18 simultaneous playback tracks, 8 simultaneous recording tracks and a total of 288 Virtual Tracks (16 per track)
- Dedicate any pair of tracks to create your final two-track mix
- Internal 10GB hard drive provides enough space for an album's worth of songs.
- 24-bit A-to-D and D-to-A converters combined with the 24-bit MT Pro recording mode provides true 24-bit signal path.

28 Channel Mixer with AutoMix Automation

- 18 playback channels plus 10 input channels available for tracking or for adding external synths, samplers and effects alongside your audio tracks during mixdown.
- Choice of 3-band EQ on up to 16 channels or 2-bands EQ on all 28 channels.
- All mixer settings including fader levels, pan, EQ and effects can be stored as scenes and easily recalled. There are eight mixer scenes available per song.
- The onboard Auto-Mix function makes it possible to automate level changes and panning in realtime.
- Automix data can be edited using an event list and copied to other locations.

Track Editing

- Time Compression / Expansion can be applied to single or multiple tracks by as much as 75 to 125%.
- Graphic waveform display and scrubbing (with a resolution as fine as 1/3000 second) allow you to find precise editing points.
- 999 levels of Undo/Redo

Inputs and Outputs

- Balanced XLR Mic inputs with switchable phantom power (channels 1 and 2)
- Balanced 1/4˝ TRS mic/line inputs (channels 3-8) and a 1/4˝ Hi-Z guitar input
- Four Aux sends using RCA connectors.
- Coaxial and Optical S/PDIF digital I/O
- Independent L/R Master and Monitor outputs using RCA connectors
- 1/4˝ headphone output
- A 25 pin SCSI connector is provided for adding external storage devices
- MIDI In, Out/Thru connectors allow you send and receive MTC (MIDI Time Code) or MIDI Clock for synchronizing with an external MIDI sequencer or even a second VS recorder
- A 1/4˝ foot switch input can be assigned to allow manual punch in/out, transport start/stop, mark point selection and more.

Effects

- A VS8F-2 effects board provides two stereo effects processors (four mono) with 240 preset and 200 user effects patches based on a wide variety of effects types including reverb, delay, chorus, flanger, dynamics, EQ, Voice Transformer, COSM guitar amp and mic modeling and more.
- A second board can be added for a total of 4 stereo/8 mono simultaneous effects.
- The Mastering Tool Kit includes a multiband compressor and offers several templates for creating polished mixes.

Built-In 24x4x4 CD-RW Drive

- Create audio CDs and backup data as well as import drum loops and other CD audio data using the CD Capture function.
- Audio CD images can be created and saved to the hard disk which significantly speeds up the CD writing process.

Recording Time (at 2 GB, 1 track)

Rec. Mode	48.0 kHz	44.1 kHz	32.0 kHz
MTP	742 min.	808 min.	1,114 min.
CDR	370 min.	404 min.	556 min.
MAS	370 min.	404 min.	556 min.
MT1	742 min.	808 min.	1,114 min.
MT2	990 min.	1,078 min.	1,484 min.
LIV1	1,188 min.	1,292 min.	1,782 min.
LIV2	1,484 min.	1,616 min.	2,228 min.

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VS-1824CD

LCD Display

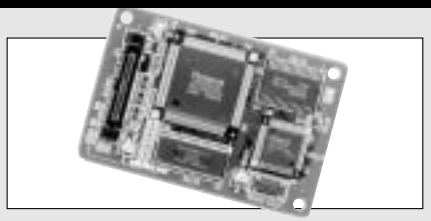
- The large, angled 320 x 240 dot backlit LCD display uses icons, event lists, text and metering to provide simple navigation and and a visual indication of the unit's current status
- Six function keys, under the display, switch between screens and execute commands

EZ-Routing

- EZ-Routing Step Editing functions provide a step-by-step guide that walks you through the required setup procedures required for recording, mixing, bouncing and mastering
- Quick Editing allows you to jump directly to specific parameters when preparing to record, mix, bounce and master
- 4 preset EZ-routing templates are provided for you to edit and then store to any of 29 user templates locations for later use

MORE ABOUT THE VS8F-2 EFFECTS BOARD

- The VS8F-2 is a user-installable dual stereo (four mono) effects board that provides the VS-1824CD and VS-2480(CD) with 250 Preset and 200 User effects patches based-on 36 of Roland's best effects algorithms, configured in single and multiple algorithm chains
- Single algorithm patches include many studio essentials like reverb, chorus and delay; while multiple algorithm chains include guitar and vocal multis that use a combination of COSM modeling and standard effects
- A number of processing effects have also been included to provide a broad creative palette for which to process tracks such as Lo-Fi processing, a Voice Transformer and Vocoder
- Each V-Studio includes a single VS8F-2 effects board plus you can add a second board to the VS-1824CD and up to seven more boards into the VS2480/VS-2480CD



COSM Speaker Modeling

- COSM Speaker Modeling simulates the sound of a variety of well known reference monitors (when used with Roland's 24-bit Digital Reference Monitors) for creating mixes that translate in a variety of listening environments

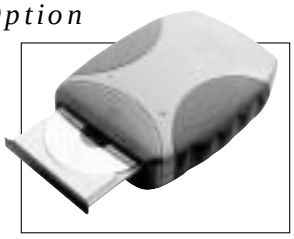
Onboard Mastering Tools

- A Mastering Tool Kit with split-frequency compression, enhancers, limiters and parametric EQ designed to give your mixes more punch and attitude.

ACCESSORIES

VS-CDR3 *CD Recorder Option*

- The VS-CDRIII is an external multi-session 40x read/16x write/10x rewrite CD recorder option available for the VS-2480 as well as many of Roland's legacy V-Studios.
- Includes Toast (Mac) and EZ CD Pro (PC) CD burning software, a SCSI cable and blank CD-R and CD-RW discs.

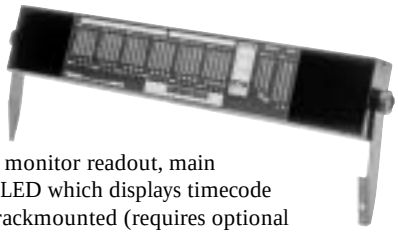


DR-10 / DR-20 *Dynamic Mics*



- Professional-quality dynamic mics with cardioid polar pattern
- Frequency response from 60 - 15k Hz
- Talk switch for on/off control
- Includes hardshell carrying case with mic clip and cable

MB-24
Level Meter Bridge



Gives VS-2480 users super-accurate, 10-segment LED metering for up to 94 channels (24 simultaneously), plus a monitor readout, main readout, track status indicators, and an LED which displays timecode and bar/beat information. Can also be rackmounted (requires optional VS-24MBA adapter for use with VS-2480/2480CD).

ADA-7000 *8-Channel A/D-D/A*



A 2U rackmount analog I/O expander for R-BUS-ready products like Roland's VS-2400CD and VS-2480 digital workstations. It features 24-bit and up to 96kHz A-to-D and D-to-A conversion via 8 balanced XLR/TRS inputs with RCA and balanced XLR outputs.

AE-7000 *AES/EBU Interface*



A 1U rackmount box that converts signals between R-BUS and AES/EBU for up to 8 channels of simultaneous digital I/O. It is the perfect solution for integrating products like Roland's VS-2400CD and VS-2480 digital workstations with other professional equipment adhering to the AES/EBU format, such as ProTools. Word Clock is also supported.

The **RBC-1** (3´) and **RBC-5** (15´) R-BUS cables for exchanging up to eight channels of two-way, 24-bit digital audio between the ADA7000 or AE7000 and the VS2400CD and VS2480(CD).

ROLAND

VS-2400CD

Digital Studio Workstation

The VS-2400CD is a Digital Studio Workstation that delivers professional 24-track recording, 48 channel automated mixing, effects and CD burning at an affordable price. Many of the VS-2400CD's features have been inherited from Roland's flagship workstation, the VS-2480CD including premium analog components, motorized faders, mouse-based "Drag and Drop" editing and a VGA Monitor output that provides powerful software control. The VS-2400CD even has some unique features of its own such as RSS 3-D panning and V-LINK that allows the VS-2400CD to integrate with Edirol video products thus allowing you to trigger video images via MIDI events. One VS8F-2 effects board is already installed and there's room for one more for a total of four simultaneous stereo effects. An internal, low-profile CD burner comes standard, giving you everything they need to burn CDs or even import .WAV files directly into a song.



VS-2400CD shown with optional VGA monitor

ALL OF THE FEATURES OF THE VS-1824CD *PLUS*—

24-bit/96kHz Sound Quality

- 24-track playback up to 24-bit (12 tracks at 96kHz) and 384 Virtual Tracks (16 V-tracks per track) for "comping" the perfect lead vocal or solo.
- Eight simultaneous record tracks are available right out of the box. You can achieve 16-track simultaneous record tracks when interfacing with the built-in 8 channel R-Bus port.
- 24-bit A-to-D and D-to-A converters and 56-bit internal processing provide lots of headroom and dynamic range

Flexible 48-Channel Mixing with Motorized Faders

- The VS2400's 48-channel, fully automated digital mixer consists of the 16-channel Input mixer, 24-channel Track mixer and eight effect returns.
- Thirteen motorized faders instantly update to their proper level position as soon as you select the channel group you want to control.
- Scene memories of all mixer parameters can be stored and recalled at the push of a button. There are 100 mixer scenes available per project.
- Dedicated 4-band EQ (2 shelving and 2 peaking) and dynamics on 32 channels.

Inputs and Outputs

- Eight XLR mic, 1/4″ TRS line inputs with switchable phantom power that allow condenser mics to be used with the mic inputs.
- Channel 8 can be switched to accept an unbalanced 1/4″ Hi-Z instrument.
- Coaxial and optical S/PDIF digital inputs and outputs.
- Independent left and right balanced 1/4″ TRS Monitor and Master outputs.
- Left and right balanced 1/4″ TRS Aux A and Aux B outputs.
- An R-Bus port allows you to add 8 more input channels via XLR / 1/4″ TRS (ADA-7000) or AES/EBU (AE-7000) R-BUS interfaces.
- MIDI In, Out/Thru connectors not only provide MTC, MIDI clock and MMC functionality, they also allow the VS-2400CD to act as a MIDI control surface or be operated from an external device.



Simultaneous Record / Playback Tracks

Sample Rate	Recording Format	Record	Playback
48/44.1/32kHz	M24, M16, CDR	16tr	16tr
48/44.1/32kHz	MTP, MT1/2, LIV, LV2	16tr	24tr
96, 88.2, 64 kHz	M24, M16, CDR	8tr	8tr
96, 88.2, 64 kHz	MTP, MT1/2, LIV, LV2	8tr	12tr

VS-2400CD

Mouse-Based Editing

- "Drag-and-drop" Copy, Move, Exchange, Insert, Cut and Erase editing using the included mouse and the LCD display or use an optional VGA monitor which serves as a the ultimate graphic "Information Display" for navigating playlists, mixer settings, and EZ Routing
- Graphic waveform display and scrubbing features makes finding edit pints a snap
- 999 levels of Undo to back you up
- A second PS-2 input accommodates an optional ASCII keyboard for quick and efficient naming of tracks and phrases.

Onboard Effects

- Two stereo effects processors expandable to four via additional VS8F-2 Effects Expansion Boards.

RSS 3-D Panning

- The RSS Panning function provides dedicated 3-D panning without complicated editing. Using this function, you can create a 3-D sound field with up to six mono sources and write it into the Automix.

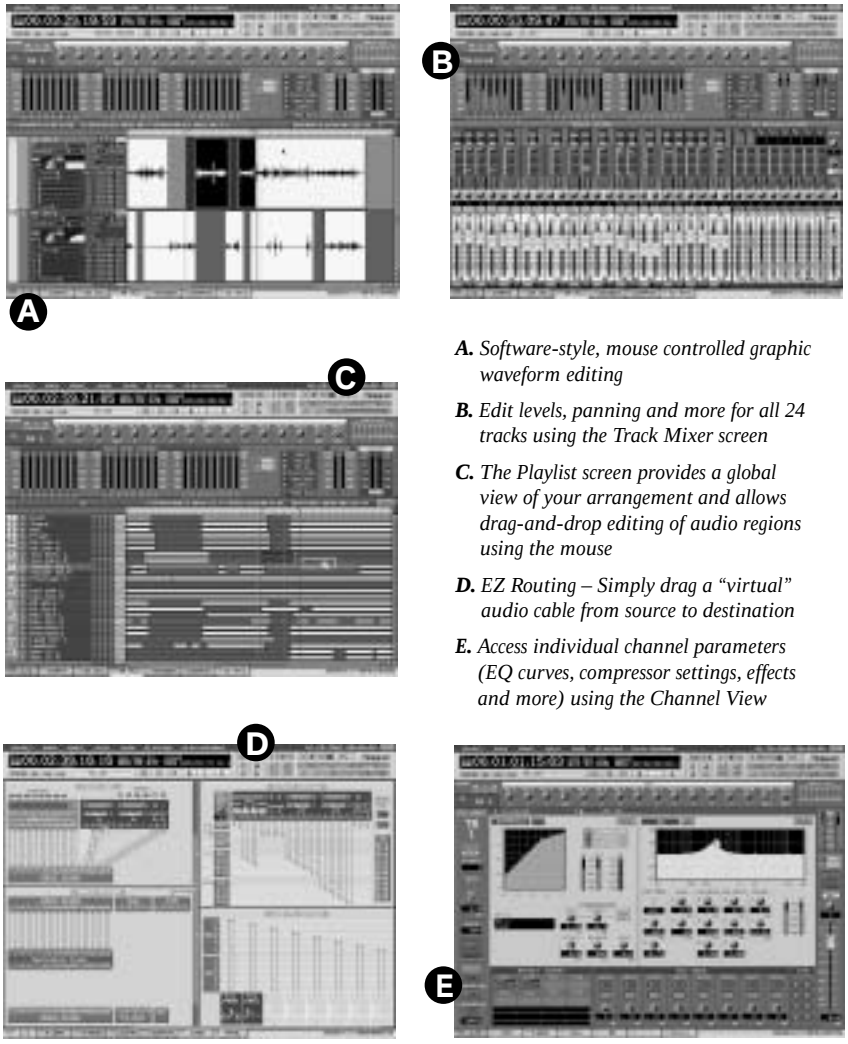
R-BUS Options

- ADA-7000 8 channel 24-bit A-to-D and D-to-A converter with balanced XLR/ 1/4˝ TRS inputs for connecting additional microphones or instruments
- AE-7000 AES/EBU Interface
- An 8-channel R-BUS port allows you to expand the I/O in a variety of analog/ digital formats and provides the ability to record up to 16 tracks simultaneously. You can also use the R-BUS port to exchange eight channels of 2-way audio and MIDI data between your V-Studio and a PC.

V-LINK

- V-LINK is a video synchronization and control protocol that is a built-in component of many of the latest generation of products from Roland including the VS-2400CD.
- V-LINK allows you to trigger video clips from MIDI events, or you can use the V-Fader function to control parameters such as color balance and brightness.

VGA MONITOR SCREENS



A. *Software-style, mouse controlled graphic waveform editing*

B. *Edit levels, panning and more for all 24 tracks using the Track Mixer screen*

C. *The Playlist screen provides a global view of your arrangement and allows drag-and-drop editing of audio regions using the mouse*

D. *EZ Routing – Simply drag a "virtual" audio cable from source to destination*

E. *Access individual channel parameters (EQ curves, compressor settings, effects and more) using the Channel View*

APPROXIMATE RECORDING TIMES

(at 10GB Partition, conversion in 1 track, unit: minutes)

Recording Mode	Sample Rate					
	96kHz	88.2kHz	64kHz	48kHz	44.1kHz	32kHz
M24	662	721	993	1,324	1,442	1,987
MTP	1,989	2,165	2,983	3,977	4,329	5,966
M16	994	1,082	1,491	1,989	2,165	2,983
CDR	994	1,082	1,491	1,989	2,165	2,983
MT1	1,989	2,165	2,983	3,977	4,329	5,966
MT2	2,651	2,885	3,976	5,302	5,771	7,953
LIV	3,181	3,462	4,772	6,362	6,925	9,543
LV2	3,977	4,329	5,966	7,955	8,658	11,932

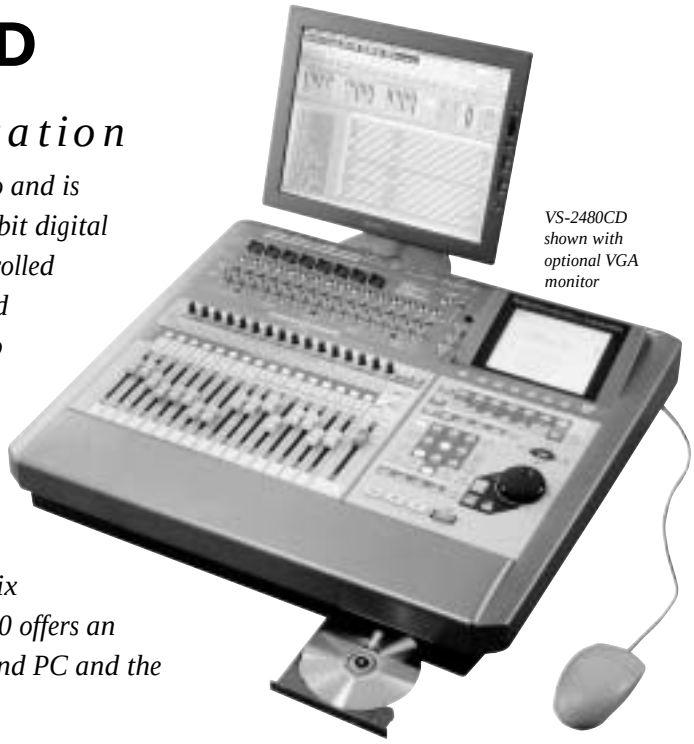
**In "CDR" recording mode, two tracks are always used in a pair (channel link is on), so recording time is half the above-listed.*

ROLAND

VS-2480HD / VS-2480CD

24-Track Digital Studio Workstation

The VS-2480 is Roland's top-of-the-line, self-contained V-Studio and is Roland's first recording workstation to offer 24 tracks of true 24-bit digital recording, with an unprecedented 64 channel digital mixer controlled y 17 motorized faders, dynamics for 48 channels, 4-band EQ and filtering for each channel and two stereo effects expandable up to 8. Analog inputs included 8 XLR mic/line inputs with phantom power and 16 balanced 1/4″ mic/line inputs. 24 Phrase Pads allow you to trigger and sequence samples directly from disk. All of these advanced features are greatly enhanced by full mouse-based editing capabilities using the large LCD display or an optional VGA monitor. You can even mix in 5.1 surround sound — ideal for DVD production. The VS2480 offers an optional CD-RW drive with CD burning software for the Mac and PC and the VS-2480CD adds an internal CD-RW drive.



VS-2480CD shown with optional VGA monitor

ALL OF THE FEATURES OF THE VS-2400CD *PLUS*—

- 16-track simultaneous recording makes it possible to capture a full band in a single take.
- The VS-2480CD ships with a fixed 80GB internal hard drive and is compatible with large-capacity 3.5″ IDE hard drives up to 128GB.
- The VS-2480HD includes a 40GB internal IDE hard drive

64 Channel Fully Automated Mixer

- 64 channel mixer – 24 input channels, 24 playback tracks, 8 Aux master channels and 8 FX return channels
- 48 channels of dynamics processing and 4-band EQ plus Filter per channel
- 16 rotary knobs can be switched between controlling a channel's panning and Aux sends 1-8 or as a single channel strip for direct control over a channel's dynamics and EQ
- 17 motorized faders can be layer switched for controlling all available channels.
- Route any signal just about anywhere you want— such as to internal or external effects

Expandable I/O

- 16 simultaneous inputs (standard) including eight XLR connectors with phantom power and 16 balanced 1/4″ TRS inputs not to mention the ubiquitous dedicated 1/4″ guitar input. Each input has a sensitivity knob that can accommodate mic and line level signals and a switchable 20dB pad
- Eight balanced 1/4″ TRS jacks assignable as 8 direct outputs or in pairs as dedicated Monitor, Master, Aux A and Aux B outputs
- Dual 1/4″ TRS headphone outputs
- Dual R-Bus ports allow you to add 16 additional inputs via 8 channel XLR/ 1/4″ TRS or AES/EBU R-BUS interfaces

Effects

- Includes one VS8F-2 effects board which provides access to two high quality stereo effects processors (4 mono).
- You can add optional VS8F-2 Effect Expansion Boards for up to eight stereo or 16 mono effects

Phrase Pad Sequencer

- The Phrase Pad function allows you to trigger up to 24 different audio phrases such as one-shot SFX or vocals directly from disk
- Each phrase is assigned to one of the 24 Track Status/Phrase buttons located above each fader
- Phrase Pad performances can be sequenced and played backed alongside audio tracks and bounced onto audio tracks

Additional Features

- Word Clock input (BNC) slaves to a master clock for jitter free recording
- A SMPTE input makes it possible to sync the to video.
- Equipped for 5.1 Surround Mixing – panning is made easy using the included mouse or using the optional VE-7000 Channel Edit Controller's built-in joystick.
- A built-in Realtime Spectrum Analyzer helps you tune your EQ to compensate for the room's dynamics and create more accurate mixes.

Rear view of the VS-2400CD



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EQUIPMENT LEASING AVAILABLE

DS-30A • DS-50A • DS90A

24-bit/96kHz Digital Reference Monitors

Roland's 24-bit Digital Nearfield Reference Monitors were designed to be the perfect match for the BOSS and Roland digital studios as well as any computer-based or standalone digital workstation. Each monitor employs a bass reflex cabinet and a 2-way bi-amplified design. An active crossover system feeds each amplifier delivering an ultra-flat response that is sure to aide in creating accurate mixes. The DS series monitors 24-bit/96kHz digital inputs are a no-brainer for the all-digital production environment and help eliminate the noise and hum problems associated with analog cabling. The DS-30A are the most affordable way to add pristine 24-bit/96kHz digital monitoring to your studio. They are ultra-compact monitors that utilizes a 30-watt bi-amp design with a 5″ propylene low frequency driver and a 1″ soft dome tweeter — ideal for any application where a smaller high-quality monitor is needed. The DS-50A feature a more powerful 50W bi-amp design with a 5″ polypropylene woofer and .75″ soft dome tweeter. The DS-50A are the ideal nearfield reference monitor if you need a monitor system with a small footprint and their wider frequency response allows them to take advantage of the COSM Speaker Modeling features found in the Roland and Boss workstations. The DS-90A are Roland's most powerful nearfield monitors and are the reference standard monitors for use with COSM Speaker Modeling. They employ a 90W bi-amped design with a 6.5″ polypropylene cone and 1″ soft dome tweeter. If you're looking for a premier monitoring system, with the widest possible frequency response, for your V-Studio workstation, look no further than the DS-90A.



DS90A



They All Feature

- Compact 2-way near-field reference monitors with built-in bi-amplifier design
- Specially designed bass-reflex cabinets deliver an exceptional flat frequency response with minimal phase shifts and distortion
- Custom active crossover circuitry delivers crystal-clear sound and flat frequency response.
- 24-bit/96kHz coaxial and optical digital inputs
- Analog balanced XLR connections and TRS 1/4″ input
- Magnetically shielded drivers for use in proximity to CPU monitors.
- Adjustable Input, Low Frequency, and High Frequency levels

The DS-30A Features

- 5″ polypropylene cone LF driver and a 1″ soft dome HF driver.
- 78 Hz to 20 kHz frequency response ±3 dB
- Third order active crossover at 2.3kHz
- 30W bi-amplified design – 20 W LF amplifier, 10W HF amplifier
- 11⅛ x 6⅛ x 9⅞″(HxWxD)
- Weighs 13 lbs. 8 oz.

The DS-50A Adds-

- Compatible with COSM Speaker Modeling found in select Roland V-Studios and V-Mixers
- 5″ polypropylene cone LF driver and a .75″ soft dome HF driver.
- 68 Hz to 22 kHz frequency response ±3 dB
- Third order active crossover at 2.3kHz
- 50W bi-amplified design – 30 W LF amplifier, 20W HF amplifier
- 12¼ x 7¾ x 10½″(HxWxD)
- Weighs 17 lbs. 11 oz.

The DS-90A Adds-

- 6.5″ polypropylene cone LF driver and a 1″ soft dome HF driver.
- 48 Hz to 20 kHz frequency response ±3 dB
- Fourth order active crossover at 2.6kHz
- 90W bi-amplified design – 60 W LF amplifier, 30W HF amplifier
- 14⅝ x 9 x 15″(HxWxD)
- Weighs 33 lbs. 1 oz.

COMPARISON CHART

	BR-532	BR-1180	BR-1180CD	VS-1824CD	VS-2400CD	VS-2480HD	VS-2480CD
Recording							
Simultaneous Recording	2	2	2	8	8 (16 w/ R-Bus)	16	16
Simultaneous Playback	4	10	10	18	24	24	24
Recording Media	Smart Media	20GB HD	20GB HD	10GB HD	40	40GB HD	80GB HD
Approx Max Record Time (Minutes)	90	4000	4000	4,000	6,500	6,500	13,000
Virtual Tracks	32	80	80	288	384	384	384
Phrase Pad Playback with Sequencer	–	–	–	–	–	✔	✔
Digital Mixer							
Channels	6	11	11	28	48	64	64
Scenes	0	8	8	8	100	100	100
Dynamics/EQ per channel	–	2	2	No/3 bands	Yes/4-band	Yes /5 bands	Yes/5 bands
Fader Grouping, Channel Copy	–	–	–	✔		✔	✔
Total Inputs	4			10	20	36	36
Total Mic Inputs	2	2	2	8	8	16	16
XLR's with Phantom Power	1/No	2	2	2		8	8
Guitar Input	✔	✔	✔	✔	✔	✔	✔
Total Outputs	4	4	4	12	22	32	32
8 Channel R-Buss Digital I/O Ports	–	–	–	–	1	2	2
EZ Routing	–	–	–	✔	✔	✔	✔
Word Clock Input	–	–	–	–	–	✔	✔
Automation							
Real-Time	–	via MIDI	via MIDI	✔	✔	✔	✔
Snapshot (Scene)	–	✔	✔	✔	✔	✔	✔
SMPTE Timecode Input	–	–	–	–	–	✔	✔
Effects							
On-Board (Stereo)	1 insert/ 1 loop	1 insert/ 2 loop	1 insert/ 2 loop	2	2	2	2
Optional Effects (Stereo)	–	–	–	2	2	6	6
COSM Amp Modeling	✔	✔	✔	✔	✔	✔	✔
COSM Mic Modeling	–	✔	✔	✔	✔	✔	✔
Speaker Modeling	–	✔	✔	✔	✔	✔	✔
Mastering Tool Kit	–	✔	✔	✔	✔	✔	✔
RSS Panning	–	–	–	–	✔	–	–
User Interface							
Mouse-based control	–	–	–	–	✔	✔	✔
ASCI Keyboard Output	–	–	–	–	✔	✔	✔
VGA Monitor Output	–	–	–	–	✔	✔	✔
Editing							
UNDO	1	999	999	999	999	999	999
Markers	0	100	100	1000	1000	1000	1000
Locate Points	3	100	100	64	100	100	100
CDR Capability							
CDR Support	–	Optional	Built-In	Built-In	Built-In	Optional	Built-In
Track at Once	–	w/ Option	✔	✔	✔	w/ Option	✔
Disc at Once	–	w/ Option	✔	✔	✔	w/ Option	✔
CD Track Edit	–			✔	✔	w/ Option	✔
Miscellaneous							
.WAV File Import/Export	✔	✔	✔	✔	✔	w/ Option	✔
Built-in Guitar Tuner	✔	✔	✔	–	–	–	–
PCM Rhythm Guide	✔	✔	✔	–	–	–	–
Free Logic VS Software	–	–	–	✔	–	–	–
Discrete Drums Loop CD	–	✔					

POCKET STUDIO 5

Compact Digital Multitrack Recorder

The Pocketstudio 5 is a low-cost portable digital multitrack recorder that allows you to playback up to four tracks of digital audio, recorded to readily available Compact Flash cards (a 32MB card is included). An internal MIDI tone generator allows you to arrange and playback the patterns of 100 onboard Standard MIDI Files for use as backing tracks for your compositions. Other features include computer interfacing via a built-in USB port over 100 internal effects, a built-in condenser mic and the ability to run for 2 hours using six "AA" batteries.



FEATURES

- Four audio playback tracks available for recording vocals, guitars and other instruments
- 16 bit A-to-D and D-to-A converters and 24-bit internal processing at a sample rate of 44.1 kHz
- All information (audio tracks, MIDI files) is stored on readily available Compact Flash cards between 8 MB and 128 MB – a 32 MB card is included
- Six faders provide volume control of the four playback tracks, the tone generator and the master output
- Each of the four playback tracks as well as the two inputs have Bass and Treble EQ with sweepable frequency control and ±12dB boost and cut
- Pan controls are also available for the 4 playback tracks
- Final stereo mix down is created using the MP3 file format
- Runs for two hours using six alkaline AA batteries (an AC adapter is also included)
- A built-in condenser mic and a headset microphone deliver true portability

Inputs and Outputs

- Two simultaneous analog inputs with trim controls and overload indicators
  - ➢ Input A is a 1/4″ guitar/line level input
  - ➢ Input B has a choice of an 1/8″ mic input or a 1/4″ mic/line level input
- 1/8″ stereo line level output
- 1/8″ stereo headphone out with level control

Editing

- Eight user-defined marker points allow you to quickly access the different sections of a song
- Two additional marker points can be used for Auto punch in and out and repeat playback Manual Punch in and out is also supported
- Cut, Copy and Paste editing of audio and MIDI tracks (with undo) using bars/beats
- ***Two Track Bouncing Modes –***

  Mode 1 (Bouncing) four playback tracks (with or without the tone generator)

  Mode 2 (Bounce +) allows you to combine the two inputs along with four playback tracks (with or without the tone generator)

Internal Synth Module

- An internal high-quality General MIDI compatible tone generator, with 16 multi-timbral parts, 128 instruments, 5 drum kits, as well as chorus and reverb, can be used to play back Standard MIDI file sequences alongside the four audio tracks. Each of the 16 parts has independent control over level, pan as well as chorus and reverb send levels
- 100 standard MIDI files are available in a variety of musical styles to provide a solid foundation for building backing tracks for your compositions
- Each MIDI file contains six standard song components or patterns (Intro, Verse 1, Verse 2, Fill 1, Fill 2 and end)
- The key and running order of each pattern, as well as the tempo and instrumentation of the songs can all be changed



Over 100 High Quality Effects

- Two multi-effects are available during recording (one effect for each of the two inputs) —
  - ➢ The effect for Input A is primarily for use with guitars (either electric or acoustic) and bass and includes a variety of amp simulators and distortion effects that are combined with reverb, chorus, flange or delay
  - ➢ The effect for Input B is primarily meant for use with vocals and other instruments including drums and percussion
- A separate reverb effect is also available for use with the audio tracks during mixdown

Conveniences

- A built-in USB port allows you to transfer files between the Pocket 5 and a Macintosh or Windows computer
- MIDI input connector allows you to control the internal tone generator using a MIDI keyboard
- Large LCD display navigated by a multifunction cursor pad and data wheel
- Tape-style transport controls
- Built-in guitar tuner

TASCAM

788

8-Track Digital Portastudio



The 788 brings together 8 tracks of direct to hard disk recording and editing, a full featured and flexible digital mixer, two assignable effect processors and optional CD burning capabilities with the ease of use and affordability of a Portastudio. The hard disk recorder offers 250 virtual tracks per song, and non-destructive editing with 999 levels of undo and redo giving you plenty of room to experiment without fear of over doing it. The mixing section offers complete routing flexibility with the ability to control 18 simultaneous inputs at mixdown — 8 playback tracks, 6 analog inputs, and 2 stereo effects. The powerful effects can be configured as stereo effects, dynamics, or even as a multi-effect processor for guitar effects. Once you've created the perfect mic, you can use an approved SCSI CD-R or CD-RW drive to burn audio CDs directly from the 788.

FEATURES

Recording Section

- Audio is all recorded with Tascam's own hard disk recording engine that provides 8 tracks of uncompressed 16-bit or 24-bit recording and playback at 44.1kHz
- Includes an internal 7.5 gigabyte hard drive
- A SCSI-2 connector is provided allowing you to record and playback to/from an optional, external hard drive up to 64GB
- 250 Virtual Tracks are provided per song allowing you to record multiple takes and choose the overall best performance or create the perfect take by editing together the best phrases from different takes

Punch In/Out

- Executing overdubs and manual punching is as simple as a single button push or, for hands free punching you can use an optional RC-30P punch pedal
- The Auto Punch feature lets you rehearse and punch in and out at exact locate in/out points. Auto Punch points can also be created using the last Manual Punch points
- The Auto punch will allow you to loop record up to 99 takes that can be auditioned and selected at a later time
- The Last Take Load function allows you to recall your last Auto-Punch session so you can audition them all over again

Digital Mixing Section

- 8 full featured channels, 6 input submixer channels and an 8 channel cue mix
- Seven channel faders are provided for controlling six mono channels and a stereo pair of channels assigned from the input connectors, or from the recorded tracks
- Each of the 8 mixer channels feature —
  - ➢ Input-to-channel assignment (input or track), Digital pad/gain, 3-band EQ, Level, Pan, Solo, Effect and Aux sends
- The Effect and Aux sends are assignable pre- or post-fader allowing optimal setup and routing flexibility
- The pre-EQ, Digital pad/gain control allows you to attenuate signals up to 42dB and increase signals up to 6dB, in 6dB intervals
- Channels may be linked in stereo pairs, allowing them to share control settings
- The six submixer inputs are available during mixdown for combining synths, samplers and drum machines with the 8 audio tracks

3-Band EQ

- High and Low sweepable shelf bands and a parametric mid band with selectable "Q" are provided for each of the 8 mixer channels –

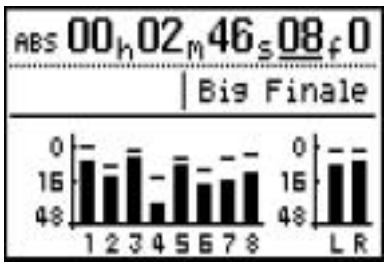
Boost/Cut (per band)	±12 dB
Low frequency range	32 Hz–1.6 Hz
Mid frequency range	32 Hz–18 kHz
Mid Q Control	0.25/0.5/1/2/4/ 8/16
High frequency range	1.7 kHz–18 kHz

Routing & Scene Memory

- You can store and recall up to 128 commonly used input assignment (routing) patterns to a user-library available to all songs.
- Each song has a 10 scene library where you can store and recall all of the mixers routings and settings. Saved settings include — Input assignments, EQs, Effect levels, settings and assignment, Aux send levels, Pan, Digital pad/gain, as well as Cue mixer, Fader and Submixer settings.

Pre-Master Function

- Contents of the recorded tracks, as well as inputs from the sub-mixer can be mixed down to the stereo outputs, which can be fed to the analog or digital outputs for mixdown cassette or DAT
- Alternately, you can "pre-master" or bounce a stereo mix that includes all 8 tracks, the 6 inputs, plus the two effect processors to the hard disk. This will allow you to audition and trim a mix before committing it to tape or disc using the optional CD burner



Inputs & Outputs

- 24-bit A-to-D and D-to-A converters
- 6 Analog Inputs —
  - ➢ Four 1/4″ TRS (balanced or unbalanced) analog mic/line inputs – the fourth input can be switched to accept a high impedance guitar input that can access the internal effects
- A 1/4″ stereo Aux Input is available for plugging in a drum machine, synth or effects processors
- Any input can be easily assigned to any mixer channel by simultaneously pressing the buttons for the input and the channel
- Separate outputs are provided for the monitor and stereo mix sections, allowing you to monitor at different levels without affecting the recording or mixdown levels
- The stereo output is also duplicated as a S/PDIF digital output, allowing you to mix down digitally to a DAT, MD or standalone CD recorder
- A stereo Aux Send output can be used with an external effect processors or for additional headphone mixes



MIDI I/O Synchronization

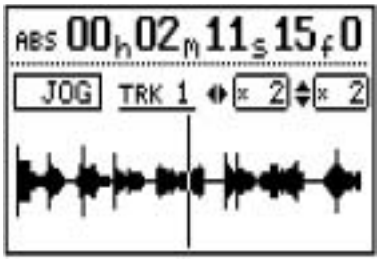
- Generate and chase MIDI Clock, Song Position Pointer or MIDI Time Code (MTC) for locking up drum machines, sequencers and other studio devices including another 788
- A tempo map containing tempo and time signature information can also be transmitted and received over MIDI

CD-Recording

- An optional external CD-RW burner (32x12x10) allows you to burn CDs from mastered tracks - and provides a cost effective way of backing up valuable data.
- You can also import and export .Wav files using ISO-9660 formatted discs
- Audio can be recorded one track at a time, leaving the CD open for more songs to be added later - or you can choose multiple songs to be recorded in one complete pass, making a 100% Red Book-compliant CD ready for mass duplication

Non-Destructive Editing

- Track editing functions include —
  - ➢ COPY->PASTE, COPY->INSERT, MOVE->PASTE, MOVE->INSERT, OPEN, CUT, SILENCE, CLONE TRACK and CLEAN OUT
- The most common editing functions (COPY, MOVE, and SILENCE) are all located on the surface while some of the more advanced functions are accessible in the Track Edit menu
- Besides editing tracks individually, It is also possible to edit across tracks, or on multiple tracks
- 999 levels of undo / redo with an undo history list allow you to freely experiment with different edits without fear of losing anything
- Track scrubbing and waveform displays (with multiple zoom controls) help you easily find precise edit points



Waveform / Jog View

Locate Functions

- Controls for locate points are provided on the surface of the unit, along with standard project navigation keys like Return to Zero
- Up to 999 location marks can be assigned in each song. These markers are stored in a list that can viewed by time or the location marker title
- Timing for locate points can be expressed in Absolute Time, MTC or in musical terms (Bars and Beats)
- Direct location lets you use the cursor keys and the dial to locate a specified position
- The Last Recording Position (LRP) function allows you to return instantly to the point where recording last started — ideal for playing back a track you just recorded, retaking a bad performance or if you need to overdub different tracks at the same position

Built-in Effects

- Two high quality effect engines capable of producing a wide range of high quality effects including reverb, delay, chorus, pitch shifter, flanger, phaser, and reverb + gate
- A library of user definable presets allows you to store up to 128 customized patches for recall at any time
- **Effect 1** can serve as a stereo effect processor, accessed from individual channel sends, or as a multi-effect processor, assigned to a specific input
- Multi-effects can consist of up to five simultaneous effects – although tailored for guitar and bass they can also be used for other instruments as well
- An independent library with 128 user definable presets is also provided for the multi-effect processor
- **Effect 2** can also serve as a stereo effect processor, or it can be assigned as independent dynamic processors across every channel on the mixer or as a single stereo dynamic processor on the stereo output
- The dynamics also have a 128 user definable preset library
- Effect and Aux Sends can be assigned as pre or post fader, allowing them to be configured for effect and monitor routing purposes
- Separate routing and scene memories can store and recall your favorite settings

Conveniences

- A repeat playback function allows you to loop a section of a song based on selected IN and OUT points
- The Slow Speed Audition (SSA) function allows you to play back a pair of tracks at the same pitch, but at a slower tempo. This is an ideal rehearsal tool when combined with repeat playback function
- A varispeed control allows you to play back and record at speeds ±6% of the original – ideal for overcoming differences in tuning
- A built-in metronome provided a useful tempo guide. The metronome sound can be generated by it's own internal source or an external MIDI device

YAMAHA

AW16G

16-Track Digital Audio Workstations



Although the AW16G is Yamaha's most affordable digital audio workstation, it still must be considered an all out professional recording system – sacrificing a few of the frills but not audio quality! Features include 16 playback and 8 simultaneous record tracks of uncompressed 16-bit audio at 44.1kHz, a 36 channel mixer controlled by 13 (45mm) faders, dual stereo effects processors plus mastering effects and dynamics and 4-band parametric EQ on every channel (a unique feature indeed at this price point). Eight mic/line inputs are provided including two XLR inputs with phantom power and a 1/4″ Hi-Z instrument input. The included 20GB internal hard drive drive provides 4,000 total track minutes of audio storage (enough room for over 50 five minute songs while the built-in CD-RW drives audio CD and data backup capabilities complete the all-in-one workstation picture.

FEATURES

- 8 simultaneous record tracks, 16-track playback with 144 virtual tracks (8 per track)
- 16-bit (uncompressed) audio resolution at a sampling frequency of 44.1kHz (± 6%)
- 24-bit A-to-D and D-to-A converters and 32-bit internal processing
- Built-in 2.5″ 20GB IDE hard disk

36 Channel Mixer

- 36 simultaneous inputs, controlled by 13 (45mm) faders, are available at mixdown including 16 playback channels, 8 mic/line inputs, 4 internal effect returns and 4 stereo quick loop sampler pads
- Each channel (except for the effect returns) has 4-bands of fully parametric EQ, dynamics (compression, expansion, gate and ducking), bus assign and pan as well as channel on/off, pad and phase reverse
- The 8 bus outs include two busses, two external Auxes and two stereo effect sends
- A 96 Scene Memory allows you to store all mix parameters for all channels and recall them at the touch of a button. You can create different scenes for each section of a song or A/B compare mixes
- You can also save the entire configuration of a single channel's settings to the dedicated 64 Preset Channel Library
- Independent Dynamics and EQ Libraries each containing 40 Factory and 88 User Presets tailored to specific applications.

Effects

- Two onboard digital effects processors using the same DSP found in other AW series workstations have access to 40 Preset and 88 User patches ranging from Reverb, Delay, Modulation, Distortion, Dynamics to Multi-effect combinations



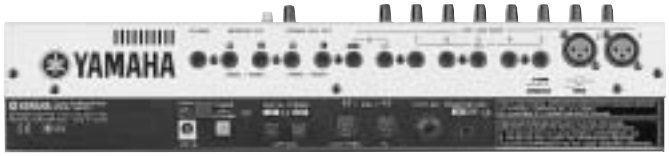
- An Input Channel library, with 40 presets, offers a variety of amp and speaker simulators along with many traditional and not so traditional effects designed specifically for capturing professional sounding vocals, acoustic and electric guitars, and bass tracks
- 13 types of studio-quality mastering effects are available on the main outputs for adding presence and punch to your final mix

Quick Loop Sampler

- Up to 44 seconds of stereo audio samples can be assigned to 16 flash memory locations, independent of the 16 audio tracks, and triggered via 4 stereo sample pads – ideal for recording drums, loops and sound effects
- A CD-ROM with over 250MB of of professionally produced drum kits, loops and other stereo samples is included
- The Slice function will automatically sync loops to match the BPM of the current song without changing their pitch

Inputs and Outputs

- Eight analog inputs are available for recording or submixing synths and effects during mixdown including — two phantom powered XLR inputs, six 1/4″ TRS mic/line inputs and a hi-Z instrument input
- A pair of unbalanced 1/4″ outputs are provided for stereo monitoring and a second pair of unbalanced 1/4″ stereo/aux outputs can be used for accessing external effects and headphone mixes a 1/4″ stereo headphone output is also provided
- Optical S/PDIF digital I/O



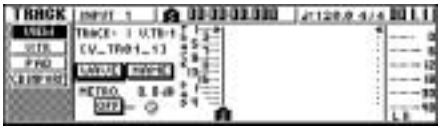
AW16G

Easy To Set Up and Use

- Route input sources to the tracks you want to record to by pressing the respective Input and Track buttons. The graphic display shows the connections you've made
- Display 240 x 64 dot backlit LCD

Track Editing

- Non-destructive track editing functions include erase, delete, insert, copy, move and exchange



- A number of destructive editing capabilities are also available including time compression and expansion in a range of 50% to 200% and pitch change ± one octave
- There is also one level of Undo
- Locate points include Start, Out RTZ, A/B, Last Record In/Out
- 99 Marker points can be assigned and quickly accessed using locate commands
- Manual and Auto punch I/O

Built-In CD-RW Drive

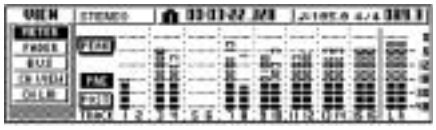
- Burn final mixes directly to inexpensive CD-R media or back up your recording session data directly to the highspeed CD-RW drive
- A Song Marking feature that lets you split a single song files into separate tracks – invaluable for indexing long jam sessions
- Compatible with the backup file formats of both the AW4416 and the AW2816
- You can also import from audio CD and import/export audio in WAV file format

MIDI In, Out and Thru

- Synchronize with synths, drum machines and sequencers using MTC, MIDI clock and MMC commands
- Use faders as a control surface for Cubase, Logic and Sonar
- Remote mode allows you to create remote programming templates by sending MIDI control data from an external device into the AW16G. Controller assignments can then be used for realtime parameter control (i.e. the filter cutoff of a synth.

Sound Clip – Idea Sketchpad

- Sound Clip function allows you to capture and experiment with musical ideas at a moments notice, without having to configure time-consuming input routings, track assignments and other setups
- Up to 180 seconds of looped recording time is provided using special memory allocation, separate from the 16 record tracks – Just plug in, hit record, and go
- Phrases can then be copied to a track, or assigned to a sample pad



Additional Features

- Built-in metronome follows an internal tempo map
- Magnetically shielded, so you can use them near computer monitors and speakers without having to worry about electromagnetic interference

ACCESSORIES FOR THE AW16G, AW2816 and the AW4416

FC5
Punch Pedal **14.99**

ADP25H
Hard drive tray to use
multiple hard drives **49.95**

AW4416 Quick Start Video.....**24.95**

ADAT Cable
15´ ADAT optical cable
(2 required for input and output).....**Call**

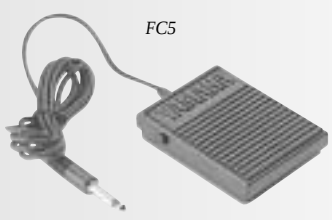
TDIF Cable – 15˝ Tascam I/O cable ...**Call**

RH5MA
Professional monitor headphones.....**49.95**

AW16G Bag
Canvas carry bag for the AW16G **44.95**

AW2816 Case
Custom Case for the AW2816 .....**179.95**

AW4416 Case
Hard carry case for the AW4416 .....**199.95**



FC5



RH5MA



AW2816 Case



AW16G Bag

YAMAHA

AW2816 / AW4416

16-Track Digital Audio Workstations



The AW2816 and AW4416 are compact console-style Digital Audio Workstations that deliver fully professional multi-track recording, mixing, non-destructive editing, effect processing as well as an internal CD-RW drive option for CD mastering and archiving — Complete music production systems, right at your fingertips! Both units offer sixteen tracks, plus a stereo mixdown track, of uncompressed 16- and 24- bit recording and playback directly to/from an included internal 20 gigabyte hard drive. The AW2816's 28 input x 18 bus mixer and the AW4416's 44 input x 20 bus mixer each provide features that meet or exceed the power and performance of the now-legendary Yamaha 02R Digital Mixer including EQ and Dynamics on every channel, fully automatable motor-driven faders and an intuitive all-digital patching system. The state-of-the-art 24-bit A-D and D-A converters employed on each of the unit's inputs and outputs as well as the 32-bit internal mix processing (44-bit for the 4-band parametric EQs) ensure that the pristine sound quality of the 16-track-plus-stereo 24-bit hard disk recorder is maintained throughout the production process. Both units also feature the same extensive non-destructive audio editing capabilities at the song, track. The two available stereo multi-effect processors, based on the Yamaha ProR 3, can be individually assigned as channel inserts or in a send/return configuration. A number of optional I/O interface cards are available for each unit that provide direct connectivity with all types of digital and analog gear – the AW2816 has one expansion slot while the AW4416 has three. The AW4416 also features 8 built in polyphonic sample pads and a special recording track which can be used to trigger and sequence loops, sound effects or even for flying in background vocals.

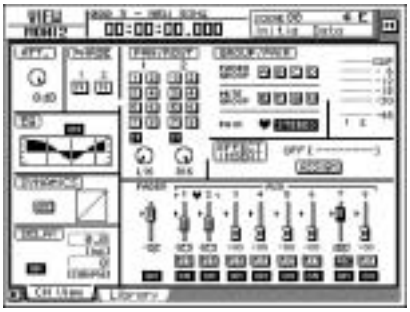
FEATURES

16 Track Hard Disk Recorder

- Sixteen tracks of uncompressed audio recording and playback with selectable word lengths of 16-bit or high resolution 24-bit and sample rates of 44.1 or 48 kHz
- 16 independent record/playback tracks each with up to 8 virtual tracks plus a stereo mixdown/bounce track provide a total of 130 tracks
  - ➢ Virtual tracks can be used to record alternate takes of a part. The best take can then be selected or the best phrase of multiple takes can be edited together to create the final track
  - ➢ The stereo mixdown/bounce track allows all 16 tracks to be mixed down directly to disk, including effects and automation, as a single stereo track
- 8 tracks can be recorded simultaneously on the AW2816 or 16 tracks on the AW4416

Mixer Sections

- The AW2816's 28 inputs are comprised of
  - ➢ 8 freely assignable input channels derived from 8 analog inputs, digital stereo inputs and 8 channel Option I/O slot inputs
  - ➢ 16 hard disk playback channels
  - ➢ 2 stereo effect return channels



- The AW4416 adds 16 freely assignable inputs derived from 8 analog inputs, digital stereo inputs, (3) 8 channel Option I/O slot inputs and 8 sampling pad inputs for a total of 24 input channels and 44 total inputs
- The AW2816's offers 18 mix busses 8 group buses, 6 AUX buses, the stereo bus and the stereo SOLO bus – The AW4416 adds two more aux busses for a total of 20 mix busses
- The input channels and the 16 playback tracks feature individual Attenuation, Phase (polarity) Reverse, Channel Delay, Dynamics, Bus Assign, 4-band Parametric EQ, Pan and Channel ON/OFF (mute)
- Mixer channels and busses are controlled via smooth and responsive 60mm motorized faders (8 on the AW2816 and 16 on the AW4416) that represent actual levels when using automation or switching between mix layers or scenes

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AW2816 / AW4416

Analog Inputs & Outputs

- Eight balanced high quality analog inputs are provided right out of the box –
  - ➢ Channels 1 & 2 feature two balanced XLR / 1/4″ TRS mic/line inputs with switchable phantom power
  - ➢ Channels 3-8 offer balanced 1/4″ TRS jacks for microphone or line input
  - ➢ Channel 8 also offers an unbalanced 1/4″ Hi-Z input that allows direct connection of an electric guitar or bass
- Unbalanced RCA L/R main outputs and balanced 1/4″ TRS L/R monitor outputs are provided as well as a 1/4″ TRS headphone output with level controls
- Four unbalanced 1/4″ TS Omni outputs, assignable from the 4 aux sends on each mixer channel, are ideal for an external effects processor or for creating alternate monitor mixes for headphones

24-Bit Coaxial Digital I/O

- Allows you to mix and transfer audio files between mastering decks, samplers and computer-based audio systems

Open-ended Expandability

- Optional Yamaha Mini-YGDAI cards can provide up to 8 channels of additional I/O in a wide range of formats including mLAN, ADAT lightpipe, T-DIF, AES/EBU and Analog
- Performance enhancing cards are available from 3rd party companies including the AP8AD and AP8DA AD/DA converter cards from Apogee as well as the Y56K DSP card from DSP developer Waves which provides on-board access to industry-standard processors such as the L1 mastering limiter and TrueVerb high resolution reverb



AW2816 rear panel

Scene Memory and Automation

- Up to 96 individual scenes, containing complete control and parameter setups, can be saved within each song
- Scenes can be fully automated to change on the time axis within the song
- Fader movements and parameter changes can be recorded and reproduced in real time using the built-in “Automix” functions
- An Event List allows you to edit the time position, value and channel status of individual automix events
- Up to sixteen automixes can be written to special memory areas within a song and recalled at any time. Data for the sixteen automixes in memory are saved on the hard disk as part of the current song

Synchronization

- Can act as either master or slave for synchronization via MTC or MIDI CLOCK for virtually unlimited system expansion and is also MMC compatible, allowing remote transport control via MIDI control change and parameter change messages

Optional Internal CD-RW Drive

- The CD-RW drive option allows you to use inexpensive, reliable CD-R or CD-RW media to take your projects from initial recording right through to creating audio CDs without leaving the digital domain as well as back up and retrieve valuable audio and mix data. You can also use the CD-RW drive to play back audio CDs or load audio and other data directly from CD-ROMs

Peripheral Connectivity

- A 50-pin SCSI-2 interface is provided for direct connection to a wide range of fixed and removable data storage devices including CD-RW drives
- MIDI In/Out/Thru jacks and a TO HOST interface provide direct connection to MIDI-capable computers and MIDI devices allowing you to automate mixer functions from an external MIDI controller or sequencer. MIDI controller data can be recorded as automix data for vastly enhanced automix versatility
- A footswitch jack makes it easy to punch in or out while playing an instrument

Built-In Multi-Effects

- Two stereo effect processors, utilizing custom Yamaha DSP LSIs, provide a wide range of ambience, delay, modulation, distortion, and amp simulator effects
- Effects can be used in an Aux send/return configuration or inserted directly into an input channel or the stereo bus
- A preset “Effect Library” provides a broad range of setups that can be used as is or as a starting point for your own variations
- Edited effects programs can be stored in any of the effects libraries 86 user locations



Effect Edit View

Reverb Effects:	Reverb Hall, Reverb Room, Reverb Stage, Reverb Plate, Early Ref, Gate Reverb, Reverse Gate
Delay Effects:	Mono Delay, Stereo Delay, Modulation Delay, Delay LCR, Echo
Modulation Effects:	Chorus, Flanger, Symphonic, Phaser, Tremolo, Auto Pan, HQ. Pitch, Dual Pitch, Rotary, Ring Mod, Mod. Filter
Guitar Effects:	Distortion, Amp simulate
Dynamic Effects:	Dyna. Filter, Dyna. Flange, Dyna Phaser
Combination Effects:	Rev+Chorus, Rev>Chorus, Rev+Flange, Rev>Flange, Rev+Sympo, Rev>Sympho, Rev>Pan, Delay+ER, Delay>ER, Delay+Rev, Delay>Rev, Dist>Delay
Other Effects:	Multi.filter

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YAMAHA

AW2816 / AW4416

EQ and Dynamics On All Channels

- 4-band fully parametric EQ and dynamics processing are provided for all input channels and the main stereo outputs. The stereo effects returns offer 4-band parametric EQ
- Separate EQ and Dynamics Libraries are provided, each containing 128 presets
  - ➢ Presets 001 - 040 have been professionally configured for specific recording tasks
  - ➢ Presets 041 to 128 are user presets where you can save your own setups

4-Band Fully Parametric EQ

- Each of the four bands features–

  Frequency Range .....21 Hz–20.1 kHz

  Boost/Cut Range .....±18 dB

  Q Range Bandwidth.....10 – 0.10
- The HIGH and LOW EQ bands can be switched to shelving EQs by turning the Q knob fully clockwise or a Low or High Pass Filter by turning the Q knob counterclockwise

Dynamics

Compressor & Expander Parameters

Parameter	*Value*
Threshold (dB)	–54 to 0 (55 points)
Ratio	1.0:1 to ∞:1 (16 points)
Attack (ms)	0 to 120 (121 points)
Outgain (dB)	0 to +18 (36 points)
Knee	hard - 5 (6 points)
Release (ms)	6 ms to 46.0 sec (160 points)•

Compander Parameters

Parameter	*Value*
Threshold (dB)	–54 to 0 (55 points)
Ratio	1.0:1 to 20:1 (15 points)
Attack (ms)	0 to 120 (121 points)
Outgain (dB)	–18 to 0 (36 points)
Width (dB)	1 to 90 (90 points)
Release (ms)	6 ms to 46.0 sec (160 points)•

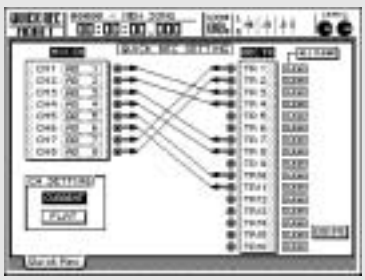
Gate and Ducking Parameters

Parameter	*Value*
Threshold (dB)	–54 to 0 (55 points)
Range (dB)	–70 to 0 (71 points)
Attack (ms)	0 to 120 (121 points)
Hold (ms)	0.02 ms to 2.13 (216 points)
Decay (ms)	6 ms to 46.0 sec (160 points)*

**@ 44.1kHz*



Digital Patch (In) View



Quick Record View

Digital Patching

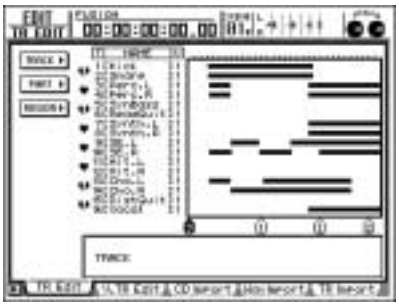
- Routing Inputs to the 16 hard disk recorder tracks and/or 18 mix buses is done entirely within the digital domain using an intuitive graphic interface. This provides an easy and efficient way of configuring seemingly complex record/mix setups while eliminating the signal loss inherent in analog patching systems
- Up to 20 different patch configurations can be saved in the internal "Patch Library" memory for instant recall
- Signal routing is aided by an graphic interface to ensure that the patching process is easy and intuitive

Quick Record

- The "Quick Record" function automatically sets up 8 record tracks (or up to 16 when an optional I/O card is installed) when starting work on a new song

Non-Destructive Editing

- All editing is non-destructive, meaning the original audio data is left untouched
- As many as 15 undo/redo levels are available, leaving plenty of opportunity to change your mind – even after several edits
- Basic operations such as copy and move
- Advanced editing functions including time compression/expansion (50% - 200%) and pitch change (±1 octaves) capabilities that give you extraordinary creative control



Song:
Name, Comment, Delete, Copy, Optimize, Protect

Track:
Name, Erase, Copy, Exchange, Slip, Append, Time Comp/Expand, Pitch Change, Import, Export

Part:
Erase, Delete, Copy, Move, Insert, Append, Time Comp/Expand, Pitch Change, Export Region Name, Erase, Delete, Copy, Move, Divide, Trim In/Out, Time Comp/Expand, Pitch Change, Import, Export

Locate & Auto-Punch Functions

- Intelligent location functions allow you to find locate points fast and easy —
  - ➢ Direct key access to START, END RTZ, A, B, LAST REC IN, LAST REC OUT, and ROLL BACK points
  - ➢ Up to 99 assignable locate points, which can be specified in time or measure numbers, are available per song
- A precision Wave Display allows you to search for locate or edit points by viewing the audio waveform
- A tempo-map coupled metronome provides a guide for creating precise auto punch-in/out points

AW2816 / AW4416

AW4416 Additional Step-Up Features



Inputs and Outputs

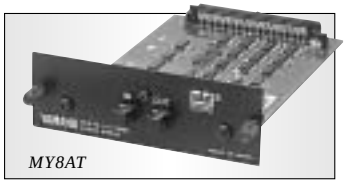
- Channels 1 and 2 offer 1/4″ TRS insert I/O that allow you to patch external analog signal processing devices
- Three Option I/O slots are provided on the rear panel allowing you to add 24 additional inputs and outputs to the system in a variety of analog and digital formats

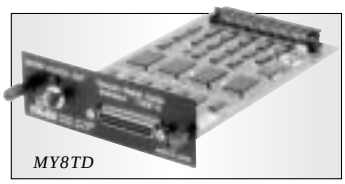


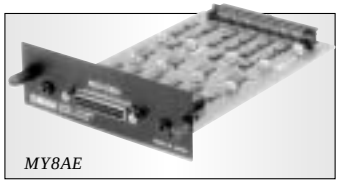
Sampling Pads

- 8 sample pads are provided allowing you to trigger drum loops, incidental hits and sound effects or fly in vocal choruses without taking up valuable hard disk playback tracks
- Samples can be assigned to the pads in a number of ways —
  - ➢ Recorded directly from a mixer input channel
  - ➢ Loaded from hard disk or other external SCSI device
  - ➢ Imported from an audio CD or in WAV format from CD-ROM
- Up to sixteen sounds can be assigned to the eight pads by switching between the two available pad banks and each sample pad can be individually assigned to any of the 24 mixer input channels
- The sample pad section offers 8-notes of polyphony and the ability to load up to 90 seconds (approx. 8MB) worth of samples into the available RAM
- Sounds are sampled into the 8 pads at the same bit depth (16 bit or 24 bit) and sample rate (44.1 kHz or 48 kHz) as the current song
- Each sample pad can trigger a mono sample — stereo samples can be utilized by assigning the left and right audio regions to adjacent pads
- The playback start and end locations of a sample can be trimmed allowing you to eliminate unwanted portions of a sample
- A simple sequencer is provided that allows you to record and edit, with frame accuracy, the timing of trigger events that occur when you strike a pad
- Events can be moved, copied repeatedly as a pattern, or erased

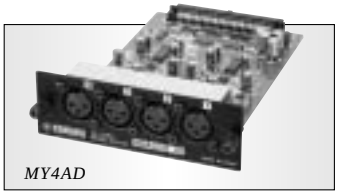
Interface Cards

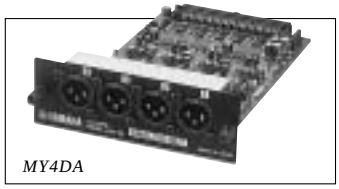

MY8AT

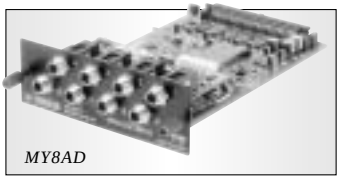

MY8TD


MY8AE


MY8mLAN


MY4AD


MY4DA


MY8AD

Card	Function	# of Inputs & Outputs	Format	Resolution / Sample Rate	Connector
MY8AT	Digital I/O	8/8	ADAT Lightpipe	24-Bit, 44.1/48 kHz	(2) Toslink
MY8TD	Digital I/O / Word Clock Out	8/8	T/DIF	24-Bit, 44.1/48 kHz	D-sub 25pin / BNC
MY8AE	Digital I/O	8/8	AES/EBU	24-Bit, 44.1/48 kHz	D-sub 25pin
MY8mLAN	Digital I/O / MIDI	8/8 (audio) 16/16 (MIDI)	mLAN	24-Bit, 44.1/48 kHz	IEEE 1394 (FireWire)
MY4AD	Analog Inputs	4 Inputs	Balanced	24-Bit	XLR
MY4DA	Analog Outputs	4 Outputs	Balanced	20-Bit	XLR
MY8AD	Analog Inputs	8 Inputs	Balanced	20-Bit	1/4″ TRS

PS-02



Self-Contained Palmtop Recording Studio

The Zoom PS-02 Palmtop Studio combines a "smart media" multi-track recorder, drum and bass machine, guitar multi-effect processor and amp modeler in a super compact device small enough to fit in your jacket pocket. As it is portable and self-contained, you can easily create a digital multitrack demo anytime, anywhere. This new generation multi-track recorder uses a SmartMedia card and offers 3 audio tracks, a stereo drum track and a bass track for a combined 6 tracks of creative power. Even better, up to ten virtual takes per audio track let you hone a song until you get it just right. And digital technology ensures that the sound stays crisp even after multiple passes.

The PS-02 incorporates more than 200 rhythm patterns using realistic drum and bass sounds. Various patterns for intros, fill-ins, and endings let you build your own song configurations. You can program drum patterns and bass chord progression for up to 100 songs. 50 template songs are already contained in the unit, which can be used for practice or as stepping-stones for your own creations.

The PS-02 features 50 built-in effects for all sorts of instruments, with the capability to use 6 modules and 8 types simultaneously. The patch memory with 60 preset and 60 user patches lets you create and store your own variations. For guitars, the Variable Architecture Modeling System developed for the GFX-8 Guitar Effects Processor is used, letting you record with genuine amp modeling sounds. A built-in microphone allows convenient vocal recordings, while a high-quality Mic-preamp with a direct connection of a dynamic microphone, allows high class recordings of vocals and any kind of instrument.

FEATURES

- Three available audio tracks are recorded directly onto a Smart Media card.
- Record and playback up to 3 separate audio tracks with the ability to digitally "bounce" tracks together.
- Each audio track offers 10 additional virtual tracks or "takes" to choose from.
- A bounce function available with panning capability allows you to overdub or mix audio tracks onto a buffer track for an endless number of audio tracks.
- Virtual Track function lets you record up to ten takes on each of the audio tracks. One of these takes can be selected and played back.
- You can get approximately 17 minutes of recording time in Stereo "Hi-Fi" mode with a 64MB memory card (32kHz) or over 2 hours in Mono "Long" mode with a 128MB memory card.
- A punch in/out function is also available.
- Stereo recording of a CD or MD source input via the AUX IN jack is also possible.

Inputs and Outputs

- 20-bit oversampling A-to-D and D-to-A converters
- Unbalanced 1/4″ input with switchable gain settings for accommodating both guitar and mic level signals.
- Built-in dynamic microphone and Auto-chromatic tuner
- 1/4″ stereo output jack and 1/8″ mini stereo headphone output.
- 1/8″ mini stereo aux input

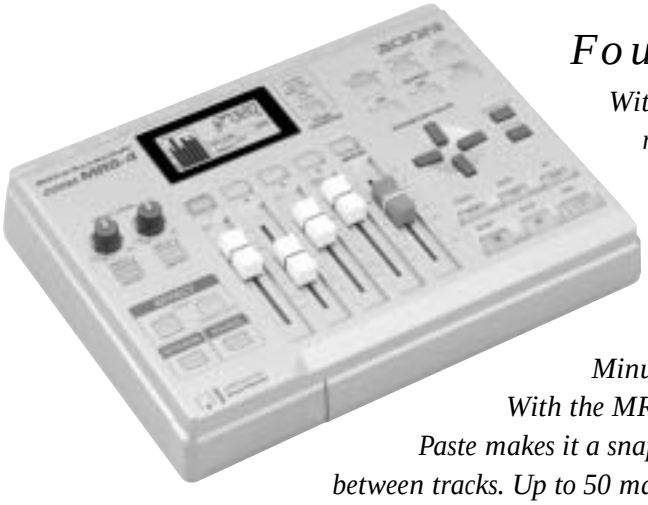
Rhythm Track

- 2 internal tracks of drums and percussion and 1 internal track of bass sounds each with 256 preset patterns
- A selection of High-quality stereo drum patterns, drum kits and bass sounds can be sequenced as a basis for your recording.
- Incorporates more than 200 rhythm patterns using realistic drum sounds. Combine patterns and chords to create up to 100 songs – ideal for composition or practice.

24-bit Multi-Effects and Guitar Amp Modeling

- A variety of Zoom guitar-based multi-effects are provided as well as a number of guitar and bass modeling sounds using proprietary VAMS technology.
- A dedicated Vocal effects can be used with the PS-02's built-in dynamic microphone.
- Additional effects are available at mix-down and for mastering.
- 60 preset and 60 user effects programs are provided based on a total of 50 effects – up to 6 effects can be used per program.
- Back-lit LCD display
- Runs on the included power supply or up to 4 hours using 4 AAA batteries.
- Extremely compact size – 3.58 x 3.26 x 1.32″ (HWD).
- Includes an 8 MB Smart Media card, power supply unit and a belt clip.

MRS-4



Four-Track Digital Recorder

With it's simple 4 track cassette-like interface, the MRS-4 is truly digital recording made easy. Just turn the power on and start recording. Saving recorded audio to smart media cards, the MRS-4 can bounce tracks without degradation, giving you a virtually unlimited number of tracks. The custom back-lit LCD screen always displays all functions along with 8 points x 8 level meters (input 1 & 2, track 1-4, master L/R), and location in Minutes/Seconds/m-Seconds plus optionally in Bars and Measures. With the MRS-4, extensive digital editing tools are at your fingertips. Copy & Paste makes it a snap to create loops, erase any unnecessary parts or exchange data between tracks. Up to 50 marker points can be inserted anywhere in a song, A-B repeat from any point or block is possible, and manual and auto punch in/out capabilities are included.

Up to 2 tracks can be recorded simultaneously. With 8 virtual takes for each 4 physical tracks, you get up to 32 takes of recording. Save your work to the supplied 32MB Smart Media card, and get 17 track-minutes in Hi-Fi mode and 34 track-minutes in Long mode. With an optional 128MB card, you can record up to 135 track-minutes in Long mode. Stereo-Link function enables stereo sources (like synthesizer, sampler, drum machine) to maintain their surround-sound. Stereo bouncing is possible onto virtual takes even when the 4 playback tracks are full.

The MRS-4 is light-weight and compact, has a 2 way power supply as Battery and PSU, and will operate for over 5 hours with 4 AA batteries, allowing for flexible song writing anytime, anywhere.

FEATURES

- 4 tracks simultaneous playback tracks and 2 tracks simultaneous record tracks.
- 8 virtual takes per track, a total of 32 tracks can be recorded.
- The digital bounce feature allows you to internally mix the four playback tracks down to two tracks.
- 17 track-minutes of recording time in Hi-Fi mode (32kHz) using the supplied 32 MB Smart Media card, 34 track-minutes at Lo-Fi (16kHz)
- Five 45mm faders (four channel and one master)
- Two-band parametric EQ, pan and an effect send are provided for each channel.

Inputs and Outputs

- Two 1/4″ inputs with continuously variable input gain to accommodate a wide variety of audio sources.
- Two 1/4″ Aux inputs are provided also for accommodating line level signals
- Unbalanced RCA master outputs and a 1/4″ stereo headphone output.

Editing

- Digital editing tools include Copy and Paste, Erase and Exchange.
- Manual and Auto Punch In/Out
- 50 marker points for transport position and A-B Repeat, from any point or any block.

Effects

- Insert effects include dynamic effects such as compression and limiting as well as guitar-based effects like cabinet Simulation and a mic preamp can be easily added to the tracks.
- Send effects such as Hall, Room, Doubling, Delay (Short and Long), Chorus (Mono and Stereo) are adjustable using the Track Parameter function.
- The Power, Boost and Vocal Mix Down effects can be used simultaneously with the send effects for a perfectly finished mix.

Additional Features

- Large back-lit LCD displays two input levels, tracks 1-4 and Master L/R output levels.
- Location can be displayed either as Measure/Beat or Min/Sec/mSec.
- Built in metronome and chromatic tuner.
- Free downloadable "MRS-4 Card Manager" software can convert audio files for each track to/from WAV files and can be used with your desktop DAW software.
- Links with the Zoom MRT-3 RhythmTrak. The MRS-4 receives signal directly from the outputs of the MRT-3 and is synchronizes via MIDI clock.
- Runs on 4 "AA" batteries or an optional AC power supply.

ZOOM

MRS-1266 & MRS-1266CD

Digital Multitrack Hard Disk Recorders



The MRS-1266 and MRS-1266CD represent Zoom's most advanced digital audio workstations. Both units feature ten tracks of 16-bit / 44.1kHz playback, 24-bit effects, plus two independent drum and bass tracks using the built in drum machine and bass synth. Both units accommodate six simultaneous record tracks from a variety of input sources including: two XLR mic inputs with switchable phantom power; two 1/4˝ instrument inputs; and six 1/4˝ line inputs. That's enough inputs and tracks to capture your whole band live. The MRS-1266CD includes an internal CD-R/W drive (optional on the MRS-1266) that allows you to back-up data from the internal 20GB hard drive as well as make audio CDs from your final mixes.

FEATURES

- Six track simultaneous record tracks and ten playback tracks: 8 mono and one stereo. A stereo master track is also provided for mixdown.
- Ten virtual tracks are provided for each of the ten audio tracks and the stereo master track. This allows the you to record multiple takes and pick the best performance. With a 20 GB hard drive, you can record up to 60 hours of music.
- The bounce function allows you to mix down the 10 ten audio tracks along with the drum and bass tracks into one or two new audio tracks. The target for the bounced tracks can be V-tracks or the dedicated master track.
- Audio is recorded using CD quality (uncompressed) 16-bit 44.1kHz to the included 20GB internal hard drive.
- All internal processing as well as A-to-D and D-to-A conversion is done at 24-bit.
- The optional internal CD-R/RW drive (included in the MSR-1266CD) allows you to create audio CDs of your mixes and back-up data.
- You can store up to 999 songs with 100 marker points per song and 100 sequence and manual scenes per song.

Mixer

- Twelve faders: channels 1 -8 are mono channels, 9/10 is a stereo channel, drum, bass, stereo master
- Each channel features a dual band EQ with sweepable frequency ±12dB boost and cut; two effect sends; pan and solo
- Besides the 10 playback tracks, there are six simultaneous inputs for accessing external synths and processors, and a stereo master channel for mixdown.

Inputs and Outputs

- Two XLR balanced mic inputs with switchable phantom power as well as two 1/4˝ unbalanced Hi-z instrument inputs and six 1/4˝ unbalanced line inputs.
- Stereo RCA main outputs and an optical S/PDIF digital output
- 1/4˝ stereo headphone output plus a 1/4˝ stereo sub out (each with dedicated level controls) that can be used for a second set of phones or for carrying a submix
- MIDI In and Out connectors will allow you to use an external MIDI controller and/or sequencer to program the 1266's drum and bass patterns, or you can use the MIDI out to control external devices.

Rhythm Section

- Built-in bass and drum Machine with 12 touch sensitive pads (a full chromatic scale), 30 note polyphony and a 20,000 note capacity.
- There are over 400 pre-programmed patterns included plus you can create over 500 of your own patterns per project.
- You can create your own original patterns using either step or real-time recording.
- 127 different sounding drum kits with 36 sounds per kit (12 pads x 3 banks) featuring the best of Zoom's critically acclaimed drum sounds. 26 high quality bass sounds, each with a 5 oct. range are also provided.
- Completed rhythm patterns can be processed with the insert and Send/Return effects and bounced to an audio track.

Phrase Import

- Pre-recorded drum beats, samples and phrases can be imported using the optional CD-R/RW drive.
- Up to 100 samples can be stored per project, and by cutting and pasting, loops can be created and dropped in and out or songs quickly and easily.

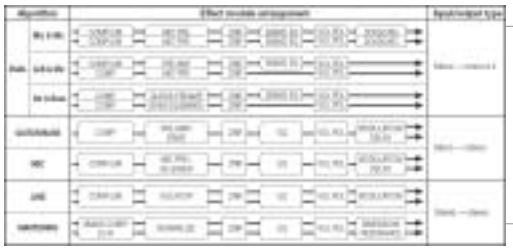
MRS-1266 & MRS-1266CD

24-bit Digital Multi-Effects

- There are two effects types available: an insert effect and two Send/Return Effects
- The **Insert effect** can be inserted into any input channel, track channel or on the master fader channel for mixdown.
- There are 94 different insert effects available based on five different algorithms including VAMS (Variable Architecture Modeling System) Guitar/Bass Amp Modeling, Microphone, Line, Dual input (guitar and vocal mic) and Mastering. Each algorithm is built from four major effect modules (compressor, pre-amp/drive, equalizer and modulation/delay).
- A total of six effect types can be chained together, and saved into any of the 280 patch locations available for each project.
- The two Send/Return effects allow you to add chorus/delay and reverb separately on each track via the send control available on each channel. There are 20 chorus/delay patches and 20 reverb patches available per project.

VAMS

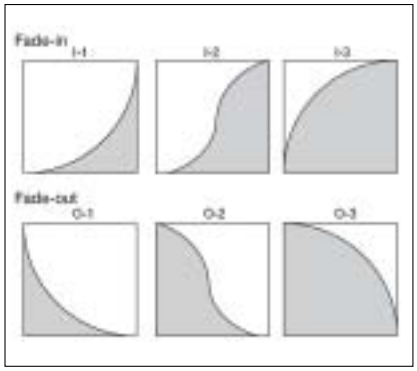
(Variable Architecture Modeling System) refers to ZOOM's amplifier modeling system. It accurately simulates the sound of classic amplifiers and cabinets. This hybrid analog / digital processing allows you to plug in a guitar, bass or any other instrument direct, and get a rich, accurate amp tone without having to mic a real cabinet.



This chart shows the structure of the various Insert effect algorithms including the six effects and the I/O routing available per algorithm. Each effect within each algorithms is fully editable.

Track Editing

- Auto and manual punch in and out recording. You can specify precise start and end points using the Scrub and Preview functions and practice the part using the A/B repeat function.
- Additional track editing features include: the ability to Trim the beginning or end of a specified track; combine usable portions of a different takes, Copy/ Erase/ Exchange whole tracks, and Copy/ Move/ Erase/ Reverse just a specified part of track.
- The time compression and expansion algorithms allows you to to change the length of a specified track without affecting its pitch. You are given the choice of overwriting the existing audio track or pasting the modified track data to a new track.
- You can also set the Fade In or Out time of a track using any one of three fade curves.



Scene Memory Automation

- You can save fader positions, effects patches, send-return effects and key status as a "Scene".
- Up to 100 scene setting locations are available per project.
- You can automate level and effect changes by assigning specific scenes to different location markers within a song.

Logical User Interface



- Designed with ergonomics in mind with all of the functions and buttons logically positioned for easy, intuitive access. Level meters and tracking info is displayed on 96 x 63mm back-lit display. The ins and outs you use the most-guitar/bass, headphones, sub-out, expression pedal, and foot switch--are all located on the front of the chassis.

Additional Features

- An optional, user-installable USB interface will allow you to share audio data with a Windows or Macintosh computer and your favorite DAW software.
- Built-in auto chromatic tuner
- A footswitch jack can be used to start and stop the transport or can be switched to control punching in and out
- An expression pedal input can be used with the optional Zoom model FP02 expression pedal for controlling effects parameters.

Options

- **ZUSB-01** is a USB option card that allows you to connect the MRS-1266 or MRS-1266CD to your PC or Mac.
- **CD-01** - Optional (User Installable) CD-R/RW Drive Kit for the MRS-1266 allows you to master and burn your own CDs, backup and restore project data as well as import WAV/AIFF and Audio CD data.
- **FP02** – Expression pedal for controlling effects parameters.



- **FS01** footswitch for controlling start/stop and punch I/O.


CROWN


ASHLY


YAMAHA


crown K2
crown K2

B&H

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The Professional Audio SourceBook

Section 6

Power Amplifiers

Alesis **548**

Ashly **549-551**

Crest Audio **552-555**

Crown **556-559**

Hafler **560-563**

Mackie **564-565**

QSC **566-569**

Rane **570**

Samson **571-573**

Yamaha **574-575**

RA SERIES

150/300/500 Watt Studio Reference Amplifiers

With their rugged construction, striking good looks, and a long list of features, the RA series will quickly find a home in project and commercial recording studios, broadcast and post-production control rooms around the world. Whether used for studio reference monitoring, live sound reinforcement, or instrument amplification, the RA-Series will provide clean, quiet, reliable power with the high quality found in all Alesis products. They are designed to provide more power and better performance at a lower overall cost per watt than any other competing amp. As a result, they can better match the power needs of virtually any passive speaker or studio monitor, offering a significantly better fit for an exceptionally wide range of budgets, applications and customers. They also provide a crisp, clean signal for outstanding performance and extended speaker life.



RA150



RA300



RA500

Designed for studio use, each RA series amp incorporates advanced convection cooling for heat dissipation. The result is long, stable operation unaffected by heat-related complications and without the added distraction and annoyance of fan cooling. Also helps eliminate the danger of losing hard-to-hear details due to noisy amp performance.

FEATURES

- DC coupled, fully complementary discrete amplifier topology
- Actively biased, dual differential inputs
- Wide bandwidth, low distortion design
- Fully protected from all fault conditions
- Front panel level controls
- LED output level metering (RA300 and 500 only)
- Overload/fault indicators
- Bridged mono operation
- Relay-controlled turn on/off
- Silent convection-cooled design
- Heavy-duty steel chassis
- RCA inputs (-10dB unbalanced)
- 1/4″ TRS inputs (+4dB balanced)
- XLR input connectors (RA 300, RA 500)
- Heavy-duty, dual binding post output connectors

RA SERIES SPECIFICATIONS

	RA150	RA300	RA500
Height	2 RU	2RU	3U
Rated Output, 4Ω	75w RMS per channel	150w RMS per channel	250w RMS per channel
Rated Output, 8Ω	45w RMS per channel	90w RMS per channel	150 RMS per channel
Rated Output, 8Ω	150w RMS Bridged Mode	300w RMS Bridged Mode	500 RMS Bridged Mode
S/N Ratio	>105dB	>105dB	>105dB
Frequency Response	10Hz - 70kHz +0, -3dB	10Hz - 70kHz +0, -3dB	10Hz - 70kHz +0, -3dB
Distortion	< 0.02% @ 4Ω, 20Hz - 20kHz	< 0.02% @ 4Ω, 20Hz - 20kHz	< 0.02% @ 4Ω, 20Hz - 20kHz
Input Sensitivity	+4dB for rated output	+4dB for rated output	+4dB for rated output
Crosstalk	-85dB, 1kHz	-85dB, 1kHz	-85dB, 1kHz
Output Offset	±50 mV, Servo controlled	±50 mV, Servo controlled	±50 mV, Servo controlled

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POWERFLEX SERIES

Four- and Six-Channel Power Amplifiers

Using a state of the art, high-speed MosFet switching design with spread spectrum switching technology, the Powerflex 4400 and 6250 provide uncompromised power with exceptional efficiency greatly reducing wasted power (heat). The 4400 delivers 275w RMS @ 8Ω or 400W RMS @ 4Ω of superior audio fidelity through four independent channels, while the Powerflex 6250—five times as efficient as conventional Class A or AB linear power amplifiers—delivers 150W RMS @ 8Ω or 250W RMS @ 4Ω through six independent channels.



▲*Powerflex 4400*

▼*Powerflex 6250*



- Their four or six channels may be used as individual, independent power amps with more than enough power to satisfy nominal audio applications. Additionally, three and two channel configurations of the Powerflex 4400 are possible for even greater flexibility. Channels may be bridged in pairs delivering 800W RMS @ 8 ohms.
- On the Powerflex 6250, five, four and three channel configurations are possible. Channels may be bridged in pairs providing for up to three higher power (500W/Ch @ 8 ohms) or 70-volt applications. The Powerflex 6250 can drive as many as six independent 25v or three independent 70v amps without the need of external transformers.
- The 4400 features bridgeable channels and switchable high pass filter on each pair of inputs. The 6250 offers a switchable high pass filter on each input. Both feature level controls located on the rear panel, XLR-1/4″ and euroblock input connections, 5-way binding post output connections and a quiet three-speed fan for on-demand cooling.
- Applications include any multichannel power amp application, monitor amp, small FOH system amp, paging systems, cinema systems, boardroom audio applications, multiple zone restaurant, bar, theme park or auditorium. The Powerflex 6250 can also be used for any 25 and 70V system, surround sound, recording studios, PA and DJ applications.



SRA-120 front and rear view



SRA-120

Stereo Amp

The SRA-120 is a professional amp with superior sonic quality, rugged construction, and many useful features in a single rack space chassis. Very compact (only 10″ deep and under 20 lbs.), the SRA-120 is suitable for full-range applications, such as small control room monitor systems or as an ideal headphone distribution amp. With its excellent bandwidth, the SRA-120 is also well-suited for driving the high end of a bi-amplification sound system setup.

- The amplifier will deliver 60 watts per channel into 4 ohms stereo, 45 watts per channel into 8 ohms stereo, or 120 watts total into 8 ohms mono-bridged.
- Amplifier design is based on Class A voltage amplifier stages with a complementary bipolar output section for low distortion and excellent overload behavior.
- Turn-on delay circuitry and instantaneous turn-off to eliminate any transients to the speaker. Each channel will also independently turn off its output if an overheated condition occurs.
- Rear panel switches let you select between two input sensitivities, stereo or mono operation, or normal or bridged mode.
- Input connections can be made via 1/4″ balanced phone jacks or barrier strips with ground lift provision.
- Level attenuators for each channel are provided on the front panel, along with a stereo headphone jack.
- Status LEDs on each channel include signal present, clip alert, and “protect-mode”

FTX-1501/FTX-2001

Stereo Amplifiers



The FTX-1501 and FTX-2001 utilize ASHLY's proven output stage design, consisting of the latest generation of paralleled power MOS-FETs for current gain. These devices have smooth transconductance curves and run at a relatively high idle current, thus preventing crossover notch distortion. FTX Series III amplifiers require no dissipation limiting protective circuitry and provide virtually infinite power gain, keeping load reflections from the driver stage promoting stability and extremely low distortion.

- The FTX-1501 puts out 185 watts at 8Ω, 275 watts at 4Ω and 275 watts at 2Ω
- The FTX-2001 delivers a full 300 watts per channel at 8Ω, 475 watts into four, and 550 watts when confronted with a true two ohm load
- Both can be mono-bridged at a power rating twice their 4Ω per channel capability
- The FTX-2001(only) can be used to drive 70V systems when configured in bridged mono mode without the use of external transformers
- Superior sonic performance with MOSFET outputs and class-A driver. Fan cooled for continuous high-power operation
- Both amps are Ashly PowerCard compatible for other input option possibilities
- They are also THX approved for use in cinema and home theater applications. To ensure compliance with the specifications of commercial installations, they are U.L. approved
- Modular construction for reliability and serviceability. Both also include Ashly's exclusive 5-year worry-free warranty



MFA-8000

High-Power Stereo Amplifier

The MFA-8000 is a very high power amplifier for the true sound professional. A three-level MOSFET power supply design is employed for maximum power efficiency and minimum AC power requirements. All power and gain stages use discrete, full complimentary transistors for very low distortion and clean overload behavior. The MFA amp utilizes a dual-monaural design, with each channel electrically and physically identical. This improves audio performance by minimizing power supply interaction, while also increasing reliability. Any necessary maintenance is also very easy to perform. The MFA-8000 puts out 800 watts continuous into 8 Ω, 1200 watts into 4, and a thundering 1500 watts into 2 Ω operating conditions.

Employs Sophisticated Monitoring Circuits to Detect Abnormal Conditions

—A highly accurate detector illuminates the true CLIP LED at the onset of clipping. Further increases in signal level will activate a limiter circuit to prevent severe overload distortion from occurring, while an illuminated LIMIT LED lets you know that the function has been activated.

—Advanced thermal management circuitry senses temperature at four different locations in the amp and adjusts fan speed accordingly.

—A Thermal LED on each channel indicates above normal operating temperatures, while protective circuitry automatically attenuates the input signal until the amplifier returns to within a normal operating range.

—Output is automatically shut-down in the event of severe overheating, shorted output, DC voltage, or ultra high frequencies at the output. Each channel's limiting, thermal, and protection management is completely independent of the other channel.

- Each channel features a ten-segment LED level meter on the front panel
- Inputs include balanced XLR (with a transformer available as an option), 1/4″ TRS and barrier strip terminals with ground lift provision. Outputs include Speakon and five-way binding posts.
- High power output with efficient power supply and output stage
- PowerCard compatible
- Modular construction for reliability and serviceability

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FTX / MFA OPTION CARDS

PowerCard Input Options

Four option cards are available to replace the standard input card on the FTX and MFA amplifiers. Replacement is fast and simple, simply remove the standard input module, disconnect the signal connector, and reconnect the separate power and signal headers to the new input option card.



TR-2



XR-1



MM-6



CL-2

TR-2 Transformer Isolated Input

The TR-2 Transformer Isolated Input features transformer isolation of inputs, stereo/mono/bridged mono switch, 1/4″ TRS, XLR, and barrier strip connectors, and individual gain potentiometers. This card is identical to the 'stock' input card shipped with every FTX and MFA amplifier except the outputs are transformer balanced.

XR-1 Three-Way Crossover

The XR-1 Three-Way Crossover features 24dB/octave filters, 20 Hz Low-cut filter, 360 degree phase control, internally adjustable frequency, CD Horn EQ switching, auxiliary outputs, and a mode switch for biamped or low only configuration.

MM-6 Mic/Line Mixer

The MM-6 Mic/Line Mixer features two stereo or four mono line inputs, two mic inputs with low-cut filter, switchable phantom power, stereo-mono bridged mono switching, and insertion points on both channels.

CL-2 Compressor/Limiter

The CL-2 Compressor/Limiter features adjustable threshold (peak detect) with indicator LED, 10:1 ratio for driver protection, stereo mono-bridged mono operation, and tamper-proof recessed controls.

All amplifiers are covered under Ashly's exclusive 5-year worry-free warranty.

ASHLY AMPLIFIER SPECIFICATIONS

	Powerflex 4400	Powerflex 6250	SRA-120	FTX-1501	FTX-2001	MFA-8000
Power output per channel, all channels driven						
8 ohm	275 Watts RMS	250 Watts RMS	45 Watts RMS	185 Watts RMS	300 Watts RMS	750 Watts RMS
4 ohm	400 Watts RMS	125 Watts RMS	60 Watts RMS	275 Watts RMS	475 Watts RMS	1200 Watts RMS
2 ohm	—	—	—	275 Watts RMS	550 Watts RMS	1500 Watts RMS
25V	150 Watts RMS	—	—	—	—	—
Bridged Mono Mode						
8 ohm	800 Watts RMS	500 Watts RMS	120 Watts RMS	550 Watts RMS	950 Watts RMS	2400 Watts RMS
4 ohm	—	—	—	550 Watts RMS	1100 Watts RMS	3000 Watts RMS
70V	—	500 Watts RMS	—	—	—	—
Power output per channel, from 20Hz-20kHz <.1% SMPTE IMD						
8 ohm	—	—	40 Watts RMS	190 Watts RMS	250 Watts RMS	720 Watts RMS
4 ohm	—	—	55 Watts RMS	280 Watts RMS	400 Watts RMS	1100 Watts RMS
Weight	45 lbs.	45 lbs.	20 lb.	46 lbs.	60 lbs.	68 lbs.
Dimensions	19 x 5.25 x 16.5″	19 x 5.25 x 16.5″	19 x 1.75 x 10″	19 x 3.5 x 16.5″	19 x 5.25 x 16.5″	19 x 5.25 x 16.5″

CA SERIES

The CA series is designed to achieve unsurpassed sonic performance and long-term reliability—even when operating under extreme stress—in touring or fixed installation applications. Absolute sonic accuracy is the hallmark of every Crest amplifier. Bass is solid and defined to the limits of audibility, with ample current reserves delivered by an "over-engineered" power supply and advanced Class AB output circuit design. Wide-bandwidth output devices assure detailed, transparent high frequency response. And, thanks to Crest's exclusive IGM circuit, they (except the CA-2) will drive 2-ohm loads safely without compromise in performance. CAs form the backbone of discriminating rental systems and high-power club installations world-wide. Auditoriums, churches, regional tour sound, and mobile DJs are all ideal candidates for the CA series´ combination of value and sonic quality.



FEATURES

- Crest's legendary "overbuilt" power supply
- Toroidal power transformer
- High-speed, wide-bandwidth output devices
- Twin tunnel cooling with back-to-front air flow
- Input sensitivity selection
- Recessed, stepped attenuators
- Dual, variable speed DC fans
- Massive, extruded aluminum heat sinks
- Balanced XLR and 1/4″ (TRS) inputs
- 5-Way binding post outputs or Speakon connectors (market dependent)
- Stereo/parallel/bridged mono mode selector switch
- Ground lift switch
- Front panel circuit breaker switch
- Modular construction
- Five-year warranty

Construction and Quality Control



The CA series is built exclusively in Crest's own USA manufacturing facility, with internal components selected for premium quality and proven durability. Each modular subassembly is pre-tested, and the assembled CA amp receives a rigorous "hot room" burn-in before thorough final checkout on precision test equipment.

Tour Class Protection

CA series is designed to operate at full-rated output under conditions that are far from "normal." Because human error and external circumstances can create extreme conditions that no amp can safely handle, all the amps (and connected loads) are safeguarded by a comprehensive array of protection circuits:

- ACL (Active Clip Limiting) automatically reduces gain in any channel that is continuously clipping, protecting speakers and output stages from uncontrolled feedback and extreme system gain settings. ACL is removed from the circuit and totally transparent when normal conditions are restored.
- IGM (Instantaneous Gain Modulation) monitors connected loads to detect conditions that may overstress output devices, allowing safe operating into nominal 2 ohm speaker loads.
- AutoRamp gradually increases gain to attenuator setting when amp is turned on.
- TourClass Protection also includes: comprehensive thermal management, short circuit protection, DC voltage, turn on/off transient, current in-rush and sub/ultrasonic input protection.

PRO 200 SERIES

CA SERIES POWER SPECIFICATIONS AT 20Hz - 20kHz, 0.1% THD+N					
	8Ω Stereo	**4Ω Stereo**	**2Ω Stereo**	**8Ω Bridged**	**4Ω Bridged**
CA2	150W	200W		450W	
CA4	245W	400W	450W	800W	900W
CA6	350W	500W	600W	1000W	1200W
CA9	550W	800W	900W	1600W	1800W
CA12	650W	1100W	1100W	2200W	2200W
CA18	950W	1700W	2400W	3400W	4800W

PRO 200 Series

Crest's latest series of professional power amplifiers– the Pro 200 Series– offer increased power, lower weight, and a smaller enclosure at a reduced cost. The Pro 8200 offers 1450 watts per channel (4500 watts bridge mode) into 4 ohms, Pro 7200 provides 1000 watts per channel (3300 watts bridge), and Pro 5200 has 525 watts per channel (1800 watts bridge).

Apart from the differences in power, the three amplifiers have very similar specifications, including a frequency response within +0, -3dB from 10 to 100kHz, with hum and noise better than -110dB, "A" weighted. The input impedance is 15k ohms, balanced. Each occupies 2RU and weigh 25 lbs.



- The Pro 200 Series amps feature Automatic Clip Limiting (ACL) to protect connected drivers, while IGM Impedance Sensing automatically modifies the gain to suit whatever impedance outputs are connected, allowing more efficient operation and loads as low as 2 ohms, as well as short-circuit protection.
- Tunnel-cooled high-efficiency heat sinks and variable speed DC fans limit the operating temperature and thus extend the life of the amplifiers. Automatic temperature sensing systems monitor the temperature of the air as it is funneled from the rear-mounted cooling fans to the front panel exhaust, and shut down a channel if temperatures exceed operating limits. AutoRamp circuitry minimizes the likelihood of power-on "thumps".
- Input connections are via XLR on the rear panel, and are actively balanced. Both Speak-On and binding post output connections are provided per channel for loudspeaker connection.
- Front panel controls include input attenuators and a combination AC power switch and circuit breaker. Five front panel LEDs per channel indicate ACL, signal, temperature protect, DC protect (which lights when a DC voltage is sensed), and Active (output relay closed and operational).
- On the rear panel, a mode switch determines stereo, parallel, or bridged mono mode, and a gain select switch determines whether the overall amplifier gain structure is x20 or x40.

PRO 200 SERIES SPECIFICATIONS		
	4Ω	**4Ω Bridged**
Pro 8200	1450W	4500W
Pro 7200	1000W	3300W
Pro 5200	525W	1800W

CPX SERIES



Designed for mobile sound systems, fixed installations and a wide variety of audio applications, the CPX Series delivers superlative performance with Crest's legendary reliability and sound quality. Series includes three models - the CPX900, CPX1500 and CPX2600. They offer a comprehensive feature set that includes built-in defeatable GCL clip limiting, low pass filters (18dB per octave @ 80 Hz), and electronic crossovers with direct outputs. In addition, all CPX Series amplifiers feature extensive protection circuitry heretofore not found in amps at this price point. These include: GCL Gain comparator (monitors input/output gain), Thermal Protection, DC Triac Crowbar (prevents DC in/out), and Turn On/Off Muting. In addition, all CPX amps utilize a unique heat sink configuration which creates a "venturi effect" that greatly enhances the cooling capabilities of these amplifiers.

FEATURES

- They feature a built-in crossover (150 Hz, 24dB per octave tuned for subwoofers), with individual in/out switches for Channels A & B.
- Barrier strip and female XLR 1/4-inch TRS input connectors; speakon output connectors for channel A & B, and bridged mono binding posts for bridged mode.
- Amps are 2U tall, have a mode selector switch for stereo and bridged-mono operation, and have a two-speed DC cooling fan.

Protection Circuitry

GCL (Gain Comparator Limiting): Monitors input/output gain and reduces signal when distortion or excessive high current conditions occur. Protects speakers from dangerous transients and unexpected level changes.Transparent in operation, yet very effective for its designed purpose.

Thermal: Amp monitors internal heat sink temperature and temporarily shuts down during extreme thermal conditions.

DC Triac Crowbar: Prevents amps from putting out or receiving dangerous DC over the speaker lines.

Turn On/Off Muting: Outputs are muted during turn on and turn off to prevent audible pop through speakers.

CPX SERIES POWER SPECIFICATIONS

	8Ω Stereo	4Ω Stereo	2Ω Stereo	8Ω Bridged	4Ω Bridged
CPX 2600	500w	750w	1100w	1550w	2200w
CPX 1500	300w	500w	750w	1000w	1500w
CPX 900	180w	300w	450w	600w	900w

1kHz, 0.1% THD+N (2Ω stereo and 4Ω bridged have 1% THD)

LQ Series Speakers

Designed for a wide-variety of mobile and fixed sound reinforcement applications, the hallmark of the injection-molded LQ series is incredibly high power handling capability with the ability to maintain clarity, even at extreme power levels. All LQ enclosures are made from high-impact, injection-molded polypropylene in a trapezoidal shape for easy stacking or cluster formation, and feature extensive ribbing and bracing. The grille utilizes perforated metal protected by a high-quality vinyl coating.



Power Handling	Continuous	Peak
LQ10	250W	1000W
LQ12	500W	2000W
LQ15	500W	2000W

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XR-20/XR-24

20- and 24-Input Rackmount Mixers

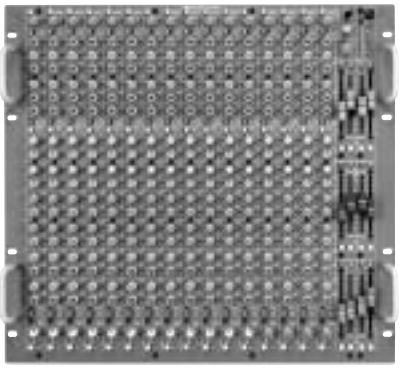


The XR-20 and XR-24 address a wide range of applications, including project studios, houses of worship, broadcasting, and all facets of live sound reinforcement. Crest's engineers group drew from decades of large and medium-format console design to create a compact, feature-filled rack mount mixer that exceeds all others in terms of quality, reliability and flexibility, regardless of price. AES-suggested grounding procedures are followed, resulting in excellent rejection of RF interference and ground-related noise. X-Rack mixers exhibit less than ± 30° of input to output phase shift, which is significantly better than most other consoles. XLR connectors are used on all primary inputs and outputs, while 1/4˝ TRS connectors are used for inserts, making it easy to interface with most professional audio gear.

FEATURES

- The XR-20 has 12 mono + 4 stereo inputs (20 inputs total); the XR-24 has 8 mono and 8 stereo inputs (24 inputs total). All inputs have mic pre-amps and individual 48volt phantom power switches.
- 100mm faders on all inputs and Left, Right, and Mono (center) outputs. 60mm faders on the 4 subgroups.
- Four-band EQ on all inputs
- 18dB per octave high pass filters on mono input channels
- Insert on all inputs, subgroups, aux outs and L, R & M outputs
- 6 Aux buses, selectable pre or post fader in pairs
- Auxes 1 and 2 are configured as Level/Pan for stereo operation
- XLR L, R & Mono outputs switchable between Line and Mic level
- Internal universal-voltage power supply
- 1/4˝ TRS and RCA connectors on Alt Output, Monitor Output and Tape Input
- Five-Year Crest Audio warranty

XR-M



The XR-M is a compact and versatile rack-mount, monitor mixer designed to provide professional, desktop quality mixing performance in control applications, fixed installation systems and more. Housed in a compact 10-rack-space package, the XR-M provides up to 12 independent mono mixes, or up to six stereo mixes for stereo "in-ear" monitoring systems. Taking up minimal space in a sound control booth, the XR-M is small enough to travel with musicians who take their own self contained in-ear system with them on the road. Highly versatile, it can be used as a mono/stereo matrix mixer with 12 mono and four stereo mic/line input channels, all with mic preamps and individual 48V phantom power switches.

- Optimized for "In Ear" monitor systems, capable of generating up to 6 stereo mixes (each input channel with individual level and pan controls) or up to 12 mono mixes.
- May be used as a stereo or mono monitor mixer or a combination of both, or a matrix mixer or for front-of-house applications.
- 12 Mono inputs plus 4 stereo mic/line inputs (20 total), all with Mic pre amps and individual (per channel) 48volt Phantom power.
- Splitter system on all mic/line inputs with ground lift switches on each input channel.
- Additional common stereo inputs on XLR and 1/4˝ connectors for use with click or Reference track sends to artists
- Four-band mid sweep EQ on mono and stereo input channels.
- 18dB per octave high pass filter on mono input channels
- 12 balanced XLR inputs and outputs

CROWN

CE 1000/CE 2000

Power Amplifiers

Crown's CE-Series amplifiers are professional tools designed and built for professional use. They feature front-panel detented level controls, useful function indicators, proportional fan-assisted cooling, short- circuit protection and more.

For superior flexibility, they can work with a range of Crown SST (System Solution Topologies) modules which provide true professional features like fixed-point crossover and summed bass output. With a powerful, reliable performance, CE-Series amps easily handle real 2-ohm loads and are capable of chest-thumping lows. Backed with a three-year, no-fault warranty that covers everything.



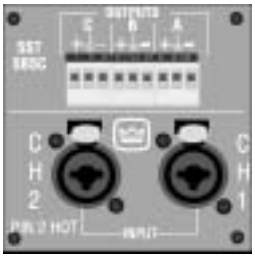
FEATURES

- Accurate, uncolored sound with very low distortion for the best in music and voice reproduction
- Bridge-mono/stereo mode switch allows you to set up your amps/speakers in the configuration that best suits your needs
- Extremely versatile they can handle a wide range of speaker impedances and outputs
- Switchable input sensitivity
- Proportional speed fan optimizes cooling efficiency
- Removable front-panel level control knobs
- They can be mounted in 19″ or shallow 14″ racks or stacked on top of each other
- Choice of Speakon (comes standard), binding post, or barrier strip outputs
- Choice of balanced 1/4″ (6.35-mm), XLR, or barrier strip inputs
- Advanced protection circuitry guards against shorted, open or mismatched loads; over- loaded power supplies; excessive temperature; chain destruction phenomena; input overload damage; and high-frequency blow-ups
- They also protect loudspeakers from input/output DC, large or dangerous DC offsets and turn-on/turn-off transients
- Three-year No-Fault (fully transferable) warranty even covers round-trip ground shipping in the U.S. for all warranty work

SST Modules

For superior flexibility, Crown CE-Series amplifiers are compatible with a range of Crown SST (System Solution Topologies) modules to tailor the amp to your application. The modules provide true professional features like fixed-point crossover and summed bass output and install easily into the back of the amplifier to save rack space and simplify system wiring.

The SST-MX module is a fourth-order Linkwitz-Riley type crossover with a fixed crossover point set at 100 Hz. Truly a professional crossover, the SST-MX uses hard-wired precision components that stay at the frequencies you have selected for consistent bandwidth control.



- Stereo biamp
- 24-dB/octave Linkwitz-Riley crossover fixed at 100-Hz
- Stereo sub outputs
- Neutrik Combo input jacks
- Barrier block balanced outputs
- Optional high-pass filter bypass on amplified outputs adapts system for full-range use

The SST-SX module is a fourth-order Linkwitz-Riley type crossover with the crossover point switchable to either 80 or 120 Hz. High-frequency (HF) signals are routed to the internal amplifier channels, and low-frequency (LF) signals are summed and routed the mono output connector. Truly a professional crossover, the SST-SX uses hard-wired precision components that stay at the frequencies you have selected for consistent bandwidth control.

- Stereo biamp
- 24-dB/octave Linkwitz-Riley tuned filters
- Crossover switchable to either 80 or 120 Hz
- Neutrik Combo (1/4″ (6.35-mm) phone and 3-pin XLR) input jacks
- Male 3-pin XLR Sub Output
- SST modules install into the back of your amp, saving rack space & simplifying system wiring

SST-MX: 100-Hz Linkwitz-Riley crossover with stereo sub-bass outputs

SST-SX: 80-/120-Hz switchable Linkwitz-Riley crossover with mono summed sub-bass output

SST-SBSC: variable Linkwitz-Riley stereo cross-over with mono-summed sub-bass output

CE 4000



Engineered from the bottom up for top performance and unmatched reliability, the CE 4000 was designed using Crown's award-winning BCA (Balanced Current Amplifier) engineering to provide superior power output, increased efficiency, legendary Crown sound, and extraordinary reliability. With typical output power at 1,200 watts per channel, and Burst Power up to 1,800 watts per channel, the CE 4000 not only handles but excels at handling 2-ohm loads. In repeated stress tests, the CE 4000 continued to perform at levels 12 dB into clip, long after others had shut down. Flexibility is also unsurpassed with features like selectable on-board high- and low-filter sets, optional signal control modules, and dual output connectors. So whether you're a touring band, DJ, A/V rental company or looking for a fixed install amp, the CE 4000's lightweight, modern design and lowest cost per watt of any amp in its class, make it the obvious choice.

FEATURES

- Patented BCA (Balanced Current Amp) technology delivers high efficiency and superb sound. Generates more power and less heat than traditional linear amplifiers
- Delivers 1800 watts per channel (both driven) into 2 ohms, 1200 watts into 4 ohms, and 600 watts into 8 ohms
- Compatible with SST (System Solution Topologies) input modules for additional signal control. SST modules offer a variety of active crossover configurations
- 31-step detented rotary level control for each channel
- 3-way sensitivity switch is switchable among 0.775, 1.4 or 3.46 volts for full output into 8-ohm load
- Protection from shorted, open or mismatched loads, input and high-frequency overloads. Loudspeakers protected from in/out DC, and power on/off transients
- Amplifier is automatically muted during power up to avoid turn-on transients
- Designed for the utmost in flexibility ,the CE 4000 features sub-bass output via integral 4-position high- pass and 3-position low-pass filter sets for each channel

— 4-position, selectable High-Pass filter switch provides settings for Flat, 30, 40 and 50Hz. Filter rolloff is 18 dB per octave

— 3-position, selectable Low-Pass filter switch provides settings for Flat, 80 and 100 Hz. Filter roll-off is 24 dB per octave

- Clip LED indicators for each channel
- 5-Way Fault Indicator:
  1) blinks during power-up;
  2) excessive heatsink temperature;
  3) transformer thermal protection;
  4) short at amplifier output;
  5) output stage non-operational
- The CE 4000 is extremely energy efficient—more than twice conventional switching amps. It requires much less energy to deliver its massive power into virtually any load, no matter how difficult. This efficiency means that you realize significant savings over the life of the amp. It also allows more CE 4000's to be plugged into a single AC circuit, reducing installation costs
- Choice of dual output connectors:

—5-way binding post plus Neutrik Speakon

—5-way binding post plus barrier strip

—Dual Neutrik Speakon

- Enhanced switch-mode power supply with PFC (Power Factor Correction) allows you to plug the amp in anywhere in the world with any line voltage.
- Easy to transport and set up (only 34 lbs.)
- Extremely reliable keeps running under the harshest of conditions. However, just in case, it includes a 3-Year "No-Fault" full warranty (even covers round-trip ground shipping in the U.S.for all warranty work)

CROWN AMPLIFIER SPECIFICATIONS

	CE1000	CE2000	CE4000
2 ohm Dual*	560w	975w	1800w
4 ohm Dual*	450w	660w	1200w
8 ohm Dual*	275w	400w	600w
4 ohm Bridge-Mono	1100w	1950w	3600w
8 ohm Bridge-Mono	900w	1320w	2400w
Frequency Response	±0.2dB from 20Hz to 25 kHz @1 watt	±0.2dB from 20Hz to 25 kHz @1 watt	±0.25dB from 20Hz to 25 kHz @1 watt
S/N Ratio A-Weighted	>105dB**	>105dB**	>102dB**
S/N Ratio 20-20k Hz	>100dB**	>100dB**	>102dB**
Dimensions	19 x 5.25 x 12.25″	19 x 5.25 x 12.25″	19 x 5.25 x 16.25″
Weight	32.6 lbs.	40.3 lbs.	33.3 lbs.

** per channel, maximum power @ 1 kHz with 0.5% THD ** below rated 1kHz power*

CROWN

D-45/D-75A

With their long history of reliability, D-45 and D-75A amps are the choice of professionals for use on the road, at home, in recording studios, laboratories and in public facilities. Very compact (1RU), the convection cooled D-45 and D-75A provide ultra-low distortion for medium power applications making them perfect for moderate power applications such as recording or broadcast studio near-field monitoring, video suite audio monitoring, a recording/broadcast headphone amp or a small paging system. Designed, built and carefully checked to ensure reliable operation with a wide variety of loads, Crown's AB+B circuitry ensures efficient use of output transistors while incorporating protection against shorted, open, mismatched or low-impedance loads. Both include the industry's only 3-year, no-fault, fully transferable warranty.

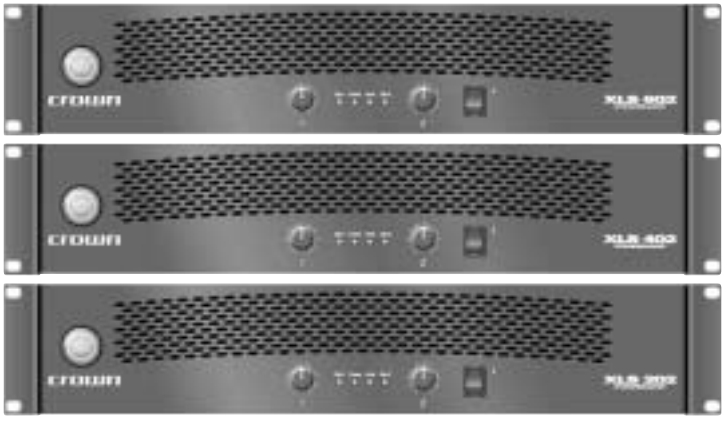


- Powerful AB+B class circuitry yields maximum efficiency with minimum crossover "notch" distortion
- IOC (Input/Output Comparator) alerts of any distortion that exceeds 0.05% to provide proof of distortion-free performance
- Detented level controls for precise repeatability
- Ultra-low harmonic and intermodulation distortion result in the best dynamic transfer function in the industry
- Very low noise and wide dynamic range
- High damping factor provides exceptional loudspeaker motion control
- Signal presence indicators verify the presence of amplifier output
- Convection cooling dissipates heat through the heat sinks and chassis for optimal cooling and maintenance-free operation
- Mounts in a standard 19″ rack
- 3-Year, no-fault, transferable warranty

POWER OUTPUT @ 1 kHz with 0.5% THD

	D-45	D-75A
4 ohm Dual*	35w	55w
8 ohm Dual*	25w	40w
16 ohm Dual*	20w	25w
8 ohm Bridge-Mono	70w	110w
16 ohm Bridge-Mono	50w	80w

** maximum power @ 1 kHz with 0.5% THD*



XLS SERIES

XLS202, XLS402, XLS602

The XLS Series represents a new era in affordable, quality power amplification. Consisting of three models in a uniform, rugged chassis, the XLS Series incorporate the best of tried-and-true design principles and innovative features to meet demanding audio requirements—reliably and within budget. Plus, they are backed by Crown's unequaled 3-year warranty that covers everything.

POWER OUTPUT @ 1 kHz with 0.5% THD

	XLS202	XLS402	XLS602
2 ohm Stereo*	250w	570w	240w
4 ohm Stereo*	200w	400w	600w
8 ohm Stereo*	145w	260w	370w
8 ohm Bridge-Mono	—	800w	1200w

** per channel*

- Housed in a rugged, all-steel 3U chassis
- Selectable high-pass filter on each channel enables amplifier to work more efficiently when full band-width is not required
- Precision detented level controls, power switch, and four LEDs, which indicate clip for each channel, power and fault conditions
- Pair of linear optocoupler clip limiters protects loudspeakers from being overdriven
- Efficient forced-air fan prevents excessive thermal buildup
- Electronically balanced XLR inputs and touch-proof binding post outputs
- 3-year, fully-transferable warranty

B&H PHOTO - VIDEO - PRO AUDIO *EQUIPMENT LEASING AVAILABLE*

MACRO-TECH 02 SERIES



Crown's Macro-Tech Series (MA-602, 1202 and 2402) deliver superior sonic accuracy and enormous amounts of power from a low-profile design—perfect for touring and fixed installations. They offer a combination of features and flexibility not available from any other amplifier, including three separate power supplies, PIP2 compatibility, XLR connectors, detented level controls and more. Patented ODEP protection and Grounded Bridge output circuitry provide incredible reliability while at the same time virtually eliminating distortion. Finally, dual bridge-mono or parallel-mono modes allow you to optimize power to your loads, while IOC (Input/Output Comparator) circuitry acts as a supersensitive distortion meter, providing you with proof of distortion-free performance.

- Crown's Grounded-Bridge design delivers large voltage swings without using easily stressed output-transistor configurations like conventional amplifiers. The results are lower distortion and superior reliability.
- Patented ODEP (Output Device Emulation Protection) circuitry compensates for overheating and overload to keep the amps working when others would fail.Front panel ODEP indicators show the reserve thermodynamic headroom for each channel.
- Complete protection against shorted, open, and mismatched loads, overheating, DC, high-frequency overload, and full internal fault protection provided by "Quad-Mute."
- Dedicated power supply transformers isolate channels in stereo mode for superb crosstalk characteristics and reliability —each channel is virtually a separate amplifier
- High damping factor provides superior control over low-frequency drivers for a clean, accurate low end
- PIP (Programmable Input Processor) connector accepts accessories that tailor the amplifier to suit specific applications
- Enhanced PIP2 connector accepts new accessory modules that further tailor the amp to suit specific applications, including wideband load current monitoring
- Two mono modes (bridge and parallel) for driving a wide range of load impedances
- 3-year, no-fault, fully transferable warranty



K2 Series

Housed in a durable, maintenance-free package that's just 2RU high, K2 Series amps incorporate Crown's exclusive Balanced Current Amplifier (BCA) circuitry to provide massive amounts of power while generating just one-tenth the heat of conventional amplifiers. In fact, the powerful K2 can deliver 1250 watts per channel into 2 ohms. And since they are incredibly efficiency, they require no internal cooling fan and feature a closed chassis. No fan means no fan noise, while the closed chassis makes them virtually immune to dust, cigarette smoke, stage fog and spilled liquids. In addition, each amp is equipped with a "green" circuit that further reduces energy use during idle periods. Energy draw is so low—only 12 watts or less—the amps may be kept in a continual ready state at minimal cost.

- Accurate,uncolored sound with very low distortion for the best in music and voice reproduction
- High damping factor for tight, clean bass response
- Mono mode switches let you set up your amps/speakers in the configuration that best suits your needs —with combined amp inputs, combined amp outputs, or both
- Advanced protection circuitry guards against shorted outputs, open circuits, DC, mismatched loads, overheating, high-frequency overloads and internal faults
- Extremely versatile, handling a wide range of speaker impedances and outputs
- BCA (Balanced Current Amplifier) power design provides super efficiency for quiet, fan-free operation
- Switchable input sensitivity
- "Soft start" prevents the amps from drawing large currents when first turned on
- Uniquely molded, cast-aluminum front panel provides exceptional air circulation
- They mount in a 19″ rack
- 3 year "No-Fault " full warranty

POWER AMPLIFIERS

Hafler designs audio components for one purpose—to help you do better work. Employing some of the best engineers in the world, they have developed patented, cutting-edge technologies such as trans•nova, trans•ana and DIAMOND. Hafler power amps are meticulously assembled using patented surface mount technology (SMT), upon which intricate macro-components are affixed by computerized machinery. Products manufactured using surface mount technology offer advantages that can be heard as well as measured, including low level components that operate at nearly identical temperatures, which is critical at low signals. SMT results in precise, hairline tolerances between components for optimum performance, the end product being fewer incidences of human error in placing parts, reduced hum and noise, and more efficient use of space. The bottom line: the only thing you hear through Hafler equipment, is music.

P1000 *100-watt Trans•ana Power Amplifier*



A high-quality amplifier with transparent and detailed sound, the P1000 is designed for broadcast studio monitoring, recording or critical listening, headphone system amplification, surround sound applications, paging systems, balanced or unbalanced use, and 115V/230V requirements. The P1000's unique Trans•ana circuitry is based on TRANSconductance Active Nodal Amplifier topology, which operates the output stage with its full voltage gain, allowing the input stage to operate from a low voltage regulated supply. The signal is then shifted up in level to the high voltage section by the driver stage, which forms an active node at ultrasonic frequencies. This results in very stable, highly linear operation producing a natural and accurate soundstage with exceptional image focus.

Amplifier Technology

In older designs, an engineer had to choose whether to amplify voltage or current at any given gain stage, usually requiring extensive circuitry to drive an amplifier's output devices. But since every stage of amplification adds noise, crosstalk and distortion to the signal, the music often gets lost in the process. Hafler's Trans•nova circuitry configures output MOSFETs to provide voltage and current gain - what they call "power gain," with fewer gain stages required in amplifier front ends. Noise and crosstalk are also greatly reduced because the signal travels through the front end at a low level. Trans•ana design achieves the same end result in a simpler, lower cost circuit. DIAMOND works in conjunction with the amplifier front end to combine the linearity of a Class A amplifier stage with the headroom of Class B. Providing up to 14dB additional dynamic range in the amplifier front end, this flawless simplicity in the signal path keeps your music perfectly intact.

- Offers an amazing 100 watts of Hafler power from a single rack space. Combines superior sonic quality, weight and power consumption in a very compact size
- Trans•ana circuitry using patented MOSFET output stage configured for "power gain" allows simpler front end circuitry, and dramatically lower cross talk and noise. Provides a very open three dimensional sound quality at a more affordable price.
- Proprietary NOMAD (NOn-Multiplying Advanced Decision) system provides safe operating area. Smart protection circuitry, NOMAD accurately computes the allowable device current for the device voltage and clamps the gate drive when the actual current exceeds this value.
- Thermal sensing network continuously monitors the heatsink temperature and shuts down the amp to protect it from excessive operating heat. Soft start circuit prevents sending of potentially destructive turn-on/off transients to the speakers.
- Front mounted rotary level controls for each channel
- Front mounted headphone jack lets you easily monitor your work.
- Balanced and unbalanced inputs offer versatile connections. Each of the two channels has dual-function (XLR or 1/4″ TRS) Neutrik combo connectors, as well RCA jacks for unbalanced use.
- Front panel LED indicators (signal, clipping, and active thermal protection) for monitoring each channel.
- Ground-lift switch lets you fit the amp in a metal rack without worrying about hum
- The P1000 has no cooling fans, and its rackmounting case is only 1U high.
- 12″ wide central portion houses the circuitry and mains transformer, and each of the rack ears is incorporated into a pair of large finned heatsinks that make up the remainder of the rack width. This provide plenty of air space for better air circulation.

B&H PHOTO - VIDEO - PRO AUDIO

P1500/P3000

200-and 400-watt Trans•nova Power Amplifiers



Designed for the beginning studio professional or musician for use in studio monitoring, touring sound and fixed installations the P1500 delivers nearly 200 watts of power from a 2RU high chassis and features James Strickland's radical Trans•nova topology (TRANS Conductance NOdal Voltage Amplifier) as well as high-quality MOSFET outputs. Same features but twice as powerful, the convection-cooled P3000 is the ultimate studio and touring amp, delivering an unbelievable 400 watts of pure power out the same size chassis.

Unlike other amplifier designs that use "buffers" following voltage gain stages, thus killing current gain, and requiring more parts, resulting in more noise and distortion, Trans•nova provides voltage and current gain at the output stage (power gain), requiring less componentry. The simple nature of Trans•nova's circuitry provides transparent, dynamic and three-dimensional sound.

FEATURES

MOFSET Outputs

- They incorporate MOSFET (Metal Oxide Semiconductor Field Effect Transistor) outputs for high output current, rugged reliability and superb sound quality. This sturdiness enables the amp to drive reactive speaker loads without the performance and sound penalties imposed by elaborate protection schemes. Very expensive to manufacture, the advantages of high-quality MOFSET outputs include:

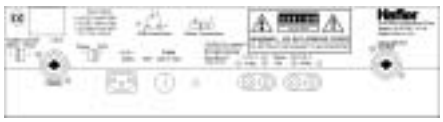
–Extremely fast switching speed, audio band is easy!

–Ultrawide bandwidth for effortless audio performance

–Very rugged, can pass huge bolts of current into a loudspeaker

–Do not "thermally run away" like bi-polar types, needing less feedback control

–Have "soft saturation"-much like tubes



Rear view of the P1500 and P3000

Trans•nova Circuitry

- They incorporate the latest refinement of the Trans•nova (TRANSconductance NOdal Voltage Amplifier) circuit. Proven to offer sound quality to satisfy the most analytic audiophile or the most demanding professional, their natural sound and realistic reproduction have made trans•nova amplifiers preferred in many critical installations.
- Each channel of these amps is built as a self-contained module which only requires mounting the heatsink and connection to the chassis-mounted transformer and binding posts to be fully functional. The circuit board contains all the operational components. This simplifies construction and improves service accessibility.

Input Sensitivity

- Input sensitivity for each channel can be adjusted individually via the front-panel level controls. Gain control on an amplifier is usually fully advanced to its maximum (rated) sensitivity. However, in public systems where it is necessary to match levels, the knobs can be removed and the controls adjusted with your fingers or a screwdriver.

LED Status

- In addition to 'Signal', 'Clip' and 'Thermal' LED indicators, a 'Short' LED lights up when a potentially damaging short has been detected. (Due to the self-protecting properties of the output power MOSFETs, there is no need for sonically degrading voltage and current limiting circuits. However in case of a short in the speaker load or cables, a speaker detection circuit shuts down that channel and lights the SHORT indicator.)

Additional Features

- 3.5″rack mount (2-rack spaces)
- 1dB increment gain controls
- Stereo/bridged mono
- Chassis/float ground switch
- Level control security covers
- Serviceable channel modules
- XLR or 1/4″ balanced inputs
- Gold-plated 5-way binding posts for output connectors. These posts directly accept 12 AWG wire or banana plugs and are spaced on 3/4″ centers to accept dual banana plugs.
- 5-year warranty

HAFLER

P4000

400-watt Trans•nova DIAMOND Power Amplifier



Featuring 400-watts of power in a 3RU high chassis and the trans•nova circuit with the added brilliance of DIAMOND (Dynamically Invariant AMplification Optimized Nodal Drive), the P4000 is perfect for studio monitoring, touring sound and fixed installations. The DIAMOND transconductance driver stage combines the linearity of Class A operation with the current headroom of a Class B system providing up to 14dB additional dynamic range in front end. The sonic result of all this unique technology is very easily heard. The P4000 presents a deep, wide sound stage with incredible musical transparency and detail.

P7000 *700-watt Trans•nova DIAMOND Power Amplifier*

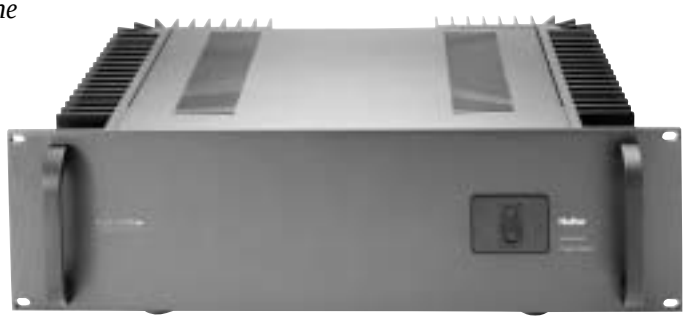


The most powerful and versatile amplifier in the line up, the 2RU high, dual fan-cooled P7000 combines Hafler's best technologies with unusual flexibility like level control, bridging, and plug-in electronic crossovers, making it perfect for studio, cinema, or musical instruments such as bass guitar or keyboards. The sonic result of this unique combination of versatility and power is immediately apparent to the discerning listener. Delivers a deep, rich expansive sound stage with incredible transparency and detail.

- Active crossovers are incorporated at the input of the amplifier. These crossovers are controlled through the use of Hafler's X-Card plug-in modules. Each X-Card can operate as a full-range, high-pass or low-pass filter with a 12dB per octave Butterworth alignment.
- Each channel utilizes two 100Hz X-Cards which in combination can be configured as a 24dB per octave slope or a 12dB per octave bandpass filter. Since the X-Card contains the resistors and capacitors that establish the crossover Q and frequency; specific system requirements can easily be accommodated just by changing the component values.
- Input configuration switches allow the amp to be configured for conventional stereo, two channel mono or single channel bridged use. When the amplifier is run in two-channel mono mode, the level controls and crossovers for each channel are fully functional which allows for using the amplifier as a single channel in a bi-amped system.

P9505 *500-watt Trans•nova DIAMOND Reference Amp*

The standard of sonic excellence for professional applications, the 9505 features the best of Hafler's legendary technology and performance to create the ultimate balanced amplifier—at an affordable price. Utilizing triple-matched J-FET inputs and MOSFET outputs in combination with trans•nova circuitry and the brilliant DIAMOND driver stage the 9505 effortlessly outperforms products costing two or three times as much, delivering crystal clear, true sounding music with less coloration and added dimension.



SR2300/SR2800/GX2800

300 and 700-watt per Channel Amplifiers



The SR2300, SR2800 and GX2800 amplifiers are 2RU high, two-channel, fan-cooled power amps suitable for use in the most demanding sound reinforcement and commercial sound installations. They offer outstanding efficiency by means of three technologies: high-efficiency TRANS•nova Class-G circuitry, high-frequency switching power supplies, and constant power output/load impedance selection switches. The GX2800 has the same power output as the SR2800 plus adds selectable low pass/high pass crossovers, time delay, CD Horn EQ and phase switch. However, it doesn't have daisy chain jacks.



TA1600

The TA1600 (Trans*Amp) is a two-channel, two-rack height, convection-cooled MOSFET power amplifier. It is suitable for various applications such as instrument amplification, DJ, Monitoring, Live Sound Reinforcement, and permanent installations.

The circuitry used in the TA1600 is the latest refinement of the patented Trans*nova circuit. It has been proven to offer sound quality to satisfy the most analytic audiophile or the most demanding professional. The natural sound and realistic reproduction have made Trans*nova amplifiers the choice in critical installations. They have proven extremely fault tolerant even in abusive situations. This ruggedness enables the amplifier to drive reactive speaker loads without the performance and sound penalties imposed by elaborate protection schemes.

Specialized circuits prevent damage to the amp and speakers, without affecting the audio signal. A soft start circuit prevents sending potentially destructive turn-on and turn-off transients to the speakers. A thermal sensing network monitors the heatsink and transformer temperature, and shuts down the amplifier to protect from excessive operating heat. The need for internal fuses has been eliminated. A sensing circuit monitors the output signal and shuts down operation when it detects a short in the output load. In addition, LED indicators give a visual representation of the operating status of each channel. Any type of fault (clipping, short protection, or thermal standby) will be indicated with a red LED.

HAFLER AMPLIFIER SPECIFICATIONS

	P1000	P1500	P3000	P4000	P7000	P9505	TA1600	SR2300	SR2800	GX2800
Power @ 8Ω **Power @ 4Ω**	50W 55W	75W 85W	150W 200W	200W 275W	350W 500W	250W 375W	60W 75W	300W 300W	600W 600W	700W 700W
8Ω Bridged Mono	110W	170W	400W	550W	1000W	750W	—	600W	1200W	1400W
Circuitry	Trans•ana	Trans•nova	Trans•nova	Trans•nova/ Diamond	Trans•nova/ Diamond	Trans•nova/ Diamond	Trans•nova	Trans•nova/ Diamond	Trans•nova/ Diamond	Trans•nova/ Diamond
RU Spaces	1RU	2RU	2RU	3RU	2RU	3RU	2RU	2RU	2RU	2RU
Cooling System	Convection	Convection	Convection	Convection	Dual Fan	Convection	Convection	Dual Fan	Dual Fan	Dual Fan
LEDs	Signal Clip Thermal	Signal Clip Thermal Short	Signal Clip Thermal Short	Signal Clip Thermal Short	Signal Clip Thermal Short	Power On —	Signal Clip Protect	Signal Clip Thermal Short	Signal Clip Thermal Short	Signal Clip Thermal Short
Inputs	XLR • ¼″ balanced	XLR • ¼″ balanced	XLR/¼″ bal RCA unbal	XLR/¼″ balanced	XLR/¼″ balanced	XLR/¼″ balanced	1/4″ balanced	XLR/¼″ bal RCA unbal	XLR/¼″ balanced	XLR/¼″ balanced
Outputs	Barrier Strip	5-way	5-way	5-way	5-way	dual binding	Barrier Strip	5-way Binding	5-way Binding	5-way
Warranty	5-Year	5-Year	5-Year	5-Year	7-Year	7-Year	1-Year	5-Year	5-Year	5-Year

FR SERIES

High Current Power Amplifiers

Incredibly affordable, the FR (Fast Recovery) Series power amps are traditional "lead sled" designs with massive toroid transformers, huge storage capacitors, state-of-the-art output devices and cooling systems that outperform all amps in their price range. Designed to perform reliably under the most adverse conditions, the amps feature sustained high-current output capability, high voltage and current slew rate, and defeatable clipping eliminator, to ensure that the output of is distortion-free. They also deliver enormous amounts of current into extremely low impedances – with dynamic power reserves to spare. For added convenience and versatility, they offer both XLR and TRS inputs, extra XLR signal pass-through, integral servo-controlled limiters, elaborate but normally inaudible protection circuits, automatic turn-on delay, front panel LED ladder displays and detented level controls calibrated in both volts and dB. So whether you want to run your whole sound reinforcement system with a single amp, or need rack upon rack of them back stage, there's an FR Series amp tailored for your application.



M800



M1400



M1400i



M2600

FEATURES

Fast Recovery

- FR series amps sound better than their competition when driven into clipping (overload). When an amp clips, you start hearing harmonic distortion. Most amps use negative feedback to help control clipping distortion. The output section of the amp sends a signal back to earlier stages to "throttle the system down. However, this causes its own problem: During clipping, when the amp relies on negative feedback, it electronically "confuses" earlier amp stages, causing them to latch in a clipping mode instead of recovering quickly. Latching causes an amp to sound terrible when pushed too hard, which it inevitably will be.
- To effectively deal with clipping, an amp must be able to recoup almost instantaneously. That's what Mackie means by Fast Recovery. FR Series amps use very sparing amounts of negative feedback. Then they use complementary Baker Clamp circuits on the positive and negative voltage amp stages, which prevent the stages from saturating (and latching) during periods of overdrive. An additional transistor senses when the Baker Clamp is active and then activates the amplifiers' internal limiting circuits. The result is no latching, instant recovery from overdriving the amp — and way better sound.

Designed for professionals and amateurs alike, the FR Series amps are great for live sound reinforcement applications, as well as studio or broadcast control rooms. With the ability to deliver massive amounts of current instantly, they can handle power-sucking subwoofers, yet remain discriminating and responsive when driving a bank of delicate tweeters.

They are designed to drive low impedance loads effortlessly and reliably. Most power amps have difficulty driving anything lower than 4 ohms, but the FR Series can easily drive 2 ohms all day. And like Mackie's Audio Mixers which are well-known for their ability to withstand the abuses of the road, the FR Series amplifiers are just as durable.

Better Cooling

- The more heat you can conduct away from an amp's output devices (transistors) the longer they'll last. The better physical contact that can be made between a transistor and its heat exchanger, the more heat can be pulled away and dissipated. FR amps have a mirror-polished heat exchanger that maximizes thermal transfer.
- The exchanger itself is transverse mounted with its air inlet in the middle. Cool room air enters the front of the amp, then travels down two short cooling tunnels and exits on the amplifier's sides. This T-Tunnel design provides a more constant thermal gradient than one long cooling tunnel — in other words, two short air passages keep all transistors closer to the same temperature instead of getting progressively hotter as the air temperature increases down one long exchanger.
- A dual-speed fan in the center directs air from the front of the amps through a large intake manifold into the cooling tunnel. The cool air is evenly distributed from the middle of the tunnel to each end, where the warm air exits the amp on either side.

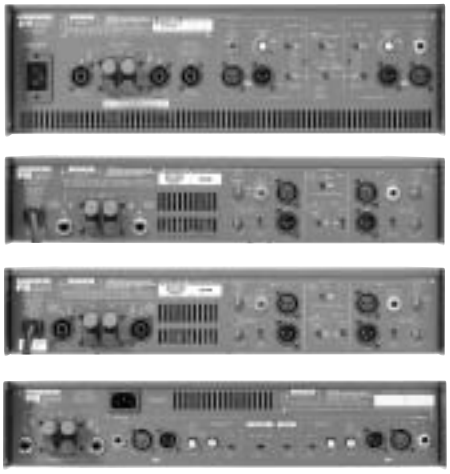
B&H PHOTO - VIDEO - PRO AUDIO

ONE HOUR FREE PARKING
AT 349 W. 34th STREET (with purchase of $100 or more)

FR SERIES

Sound Enhancing Features

- All PA cabinets only reproduce bass down to a certain point, called the tuned frequency. Below that point, you get audio sludge and potential woofer damage. FR Series variable low cut filters let you feed your system only the frequencies it can handle. You can "dial in" any tuned frequency from 5Hz to 170Hz. Each amp also includes an infrasonic stabilizer circuit that cuts the really low frequencies that cause visible woofer cone flutter.
- To improve high frequency reproduction, many systems use constant directivity (CD horns). They improve treble dispersion by more evenly distributing high frequencies. But in doing so, they actually create a frequency "dip" that reduces important frequencies anywhere from 2.5kHz to 5kHz. The old way to compensate for this was a very expensive crossover module or a graphic equalizer. The M•800, M•1400 and M•1400i include separate left and right CD horn EQ adjustments. This 6dB/octave EQ's "knee" position is sweepable from 2kHz to 6kHz -- Mackie extended the high end boost so you can add "Air" EQ even if you're not using CD horns.
- The M•1400/M•1400i feature 3rd order, 18dB/octave, Bessel electronic crossovers with 2 selectable frequencies. The M•2600 features 4th order, 24dB/octave, Linkwitz-Riley electronic subwoofer crossovers with 3 selectable frequencies. All have uniform time-delay low-pass filters. Just select the crossover frequency, connect the subwoofer output to another amp – and drive that subwoofer all over the map.



Additional Features

- Amps convert some of the raw power from a wall socket into modulated musical power for your speakers. The rest is turned into heat. FR amps achieve 74.5% efficiency, just shy of 78%— the theoretical maximum.
- It's easy for an amplifier to put out high frequencies. But low frequencies demand awesome amounts of power. In order to sound good, a power amplifier has to be able to deliver flat response over an extremely wide range of frequencies. The outputs of an FR Series amplifier are very flat and wide: +0/-3dB from 10 Hz to 80 kHz and full power capability to 100 kHz.
- An amp must be able to reproduce brief, intense bursts of energy–transient peaks–at the same rate they are input to the amp. The ability to do this is measured by its voltage slew rate. The slew rate of an FR Series amp exceeds 50V/µsec. That's one of the fastest slew rates of any pro audio amp.
- The can operate on as little as 63% of their rated voltage (a big plus during brownouts or at the end of a long extension cord)
- Heavy-duty 5-way binding post output
- Rack-mountable (2U) with rear support rails for extra support
- Separate left and right detented gain controls, calibrated in both dB (off to 30) and volts (with indications of 1v, 2v and 3v)
- Balanced XLR and 1/4˝TRS input jacks plus extra male XLR signal pass-through
- 6 discrete LEDs including signal present and overload indicate signal levels
- Switch for Stereo, Mono or Bridge modes
- Limiter eliminates clipping in both channels
- Subwoofer mode switch with dual-frequency select (63Hz/125Hz)
- Separate L/R Protection Circuits:

— Automatic 3-second turn-on delay prevents damaging thumps from reaching the speakers

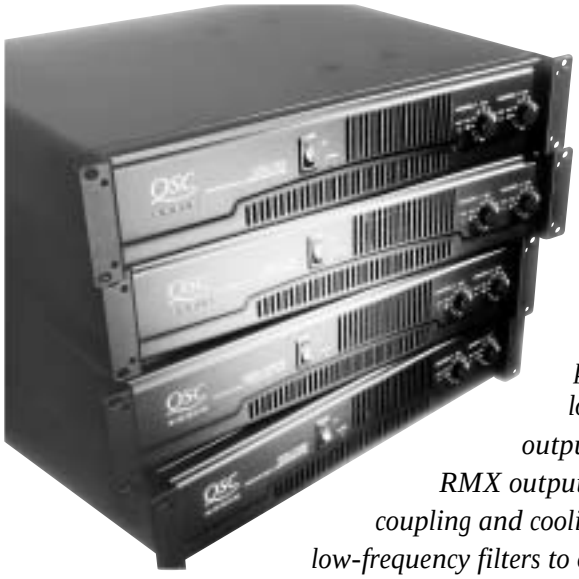
— Short-circuit protect with separate L/R indicating LEDs

— Thermal protect with combined CH1 and 2 Cold and Hot indicating LEDs

	M•800	M•1400	M•1400i	M•2600
4-ohm bridge	800W	1400W	1400W	2600W
8-ohm bridge	550W	1000W	1000W	1700W
2-ohm load	400+400W	700+700W	700+700W	1300+1300W
4-ohm load	275+275W	500+500W	500+500W	850+850W
Low cut filters	yes	yes	yes	yes
Limiter circuit	yes	yes	yes	yes
CD Horn EQ	yes	yes	yes	—
Subwoofer crossover	—	2-position	2-position	3-position
XLR pass-through	yes	yes	yes	yes
5-way binding posts	yes	yes	yes	yes
Neutrik Speakons	—	yes	—	yes
1/4" TS outputs	yes	—	yes	—
T-Design cooling	yes	yes	yes	yes
Demand-sensitive fan	yes	yes	yes	yes
Detented level controls	yes	yes	yes	yes
LED output display	yes	yes	yes	yes
Signal present, OL and Channel Status LEDS	yes	yes	yes	yes
Temp & Short Circuit LEDs	yes*	yes	yes	yes

**(Temp LEDs only)*

RMX SERIES



Professional Power Amplifiers

The RMX Series amplifiers offer clean, dynamic power with legendary QSC sound quality and reliability for unmatched performance. Providing professional quality at affordable prices, they range in power from 430 watts to 2000 per channel at 2 ohms, all in a compact chassis only 2 rack spaces high and less than 16″ deep. They use a powerful high-current toroidal transformer and ample filter capacitance to ensure generous energy reserves. This makes a critical difference in sound quality, because it allows the amps to effortlessly punch out high level transients, even when driving abusive 2-ohm speaker loads. A full complementary output circuit using the highest-grade linear output devices delivers ultra-low distortion and a ruler-flat frequency response. RMX output devices are direct-mounted to the heat sink for optimum thermal coupling and cooling. Other features include user-defeatable clip limiters and selectable low-frequency filters to optimize sonic performance. They are equipped with XLR and 1/4″ balanced inputs, and Speakon and binding post outputs.

FEATURES

RMX Series Features

- High-current toroidal transformers for greater 2-ohm power and low noise
- Barrier strip, XLR and 1/4″ balanced inputs plus Speakon and binding post outputs provide easy connections.
- Low-noise variable speed fans with rear-to-front air flow keep amps and racks cool
- Front mounted gain controls for easy access
- Signal and Clip LED indicators help monitor performance
- Amplifier protection includes full short circuit, open circuit, thermal, ultrasonic, and RF protection. Stable into reactive or mismatched loads.
- Independent DC and thermal overload protection on each channel automatically protects amp and speaker
- 3-year warranty plus optional 3-year extension available

RMX "HD" Step-up Feature

- RMX 1850HD and 4050HD "Heavy Duty" provide improved thermal performance to give you higher continuous power in 2-ohms stereo or 4-ohms bridged applications.
- Independent defeatable clip limiters reduce distortion without sacrificing peak performance. When an RMX is pushed to extremes, the clip limiter automatically finds the exact point of overload and keeps the amp in its "Peak Zone", preventing severe distortion without reducing performance.
- Prevent low-frequency muddiness with selectable 30 or 50Hz low-frequency filters and boost your system's response by matching the amp's range to the speakers. Each channel is separately adjustable: 50Hz for compact full-range speakers; 30Hz for subwoofers and large full-range systems (OFF for studio monitoring).
- The clip limiter and low-frequency filter are independently adjustable for each channel

	RMX 850	RMX 1450	RMX 1850HD	RMX 2450	RMX 4050
STEREO MODE, EIA	**Both Channels Driven Continuous Average Output Power Per Channel**				
8Ω EIA 1 kHz, 0.1% THD	**200W**	**280W**	**360W**	**500W**	**850W**
4Ω EIA 1 kHz, 0.1% THD	**300W**	**450W**	**600W**	**750W**	**1400W**
2Ω EIA 1 kHz, 1% THD	**430W**	**700W**	**900W**	**1200W**	**2000W**
BRIDGE MONO MODE	**Bridge Mono Mode Operation**				
8Ω EIA 1 kHz, 0.1% THD	**600W**	**900W**	**1150W**	**1500W**	**2800W**
4Ω EIA 1 kHz, 1% THD	**830W**	**1400W**	**1800W**	**2400W**	**4000W**

Thermal Management System provides continuously variable speed fan for quiet operation. Rear-to-front air flow keeps amps and racks cool.



PLX SERIES

Professional Power Amplifiers

The PLX Series define high impact. Up to 3,400 watts in a 2 rack-unit chassis that's only 13˝ deep and 21 lbs. They also include QSC's exclusive PowerWave Technology used in the award-winning PowerLight Series for chest-pounding bass and crystal-clear highs. They also feature hum-free noise floor and ultra-low distortion of 0.03% THD. And to keep them running under the most demanding conditions, they incorporate QSC's Advanced Thermal Management System.



FEATURES

PLX Series Features

- Exclusive PowerWave power supply for superior bass and lighter weight
- Stereo, parallel-input, or bridged mono modes
- Ultra-low distortion (0.03% THD at 8 ohms) and studio-quality noise floor (-107 dB)
- Detented gain controls
- Front panel LED indicators for power, signal, -20 and -10 dB, clip/ protect, parallel inputs, and bridged mono mode
- Class H output reduces AC power draw and heat by 40% (PLX 2402, 3002, 3402)
- Selectable clip limiters and subsonic filters (30 or 50 Hz roll- off) reduce distortion and protect speakers
- Variable-speed fan for low noise and solid 2 ohm performance
- Balanced 1/4˝ TRS & XLR inputs, Speakon and shock-proof binding post outputs
- Amplifier protection includes full short circuit, open circuit, thermal, ultrasonic, and RF protection. Stable into reactive or mismatched loads.
- Advanced Thermal Management (ATM) system ensures PLXs run cool and reliable even under harsh conditions. Using a high-output variable speed fan, ATM pulls cool air from the rear of the amp and propels it through a heat shroud precision-fitted to flow air over the heat sinks and out of the amplifier.
- Compact 2 RU high, 21 lbs. and 13˝ deep chassis fits into any rack
- 3-year warranty plus optional 3-year extension available

PowerWave Technology

With PowerWave technology, the PLX series make your music to a whole new level. Not only does it deliver bigger bass and cleaner highs, PowerWave also cuts wasted heat and boosts reliability. PowerWave makes the amps better sounding and more compact—so you don't have to settle for conventional "lead sled" designs with hum, sagging supplies, and backbreaking weight. Provides heavyweight audio performance in less than half the size and weight of typical amps.

PowerWave is a switching power supply that provides ample current to the audio power circuitry by charging the supply rails 230,000 times a second through an ultra-low impedance circuit. So unlike other amps with conventional supplies, the audio signal is never starved prematurely and remains crisp and clean. It virtually eliminates hum and greatly reduces noise, providing a vast dynamic range that can handle any music without running out of headroom.

—Conventional power supplies draw 60 Hz AC from the wall directly into the power transformer. This low frequency requires a massive iron core and hundreds of windings. In fact, a conventional 3000-watt amp needs a transformer that weighs at least 35 lbs. as much as two-thirds of the amp's total weight.

—Worried about hum? PowerWave gets rid of it completely. The 115-kHz PowerWave supply eliminates 60 Hz fields that can couple into internal or external audio circuitry.

—Bass A PowerWave transformer has lower impedance and greater efficiency because its copper windings are short and thick. In essence, it provides a "bigger pipe" to get electrical energy to the amp's output circuitry. Helps the amps deliver chest-pounding bass.

—PowerWave supply charges the rails 230,000 times per second—a vast improvement over 100–120 times per second in conventional supplies. This high recharge rate minimizes AC ripple that can degrade sonic quality

	PLX1202	PLX1602	PLX2402	PLX3002	PLX3402
8-ohms (stereo)	200	300	430	830	200
4-ohms (stereo)	280	450	700	1400	200
2-ohms (stereo)	360	600	900	1800	200
16-ohms (bridged mono)	500	750	1200	2400	200
8-ohms (bridged mono)	200	300	430	830	200
4-ohms (bridged mono)	200	300	430	830	200

ISA SERIES

Professional Power Amplifiers



Designed as a cost-effective solution specifically for sound contractors, the ISA Series feature three low impedance models (ISA 280, ISA 450, ISA 750) rated for 2-ohm operation, and three "T" versions with 25, 70 and 100 volt outputs for distributed audio systems (ISA 300T, ISA 500T, ISA 800T). Versatile loading options and a comprehensive feature set make the ISA Series a rugged, cost-effective power solution for any permanently installed sound system.

Unique among distributed sound power amplifiers, the "T" versions will drive 8- or 4-ohm loads and a distributed system on the same channel simultaneously. This enables a contractor to reduce the number of required amplifier channels by allowing a limited number of distributed speakers to be attached to the same amp as is powering the main sound system.

Housed in a rugged 3RU chassis, all amps feature rear panel gain controls for tamper resistant operation with 2 dB detents for quick and repeatable settings. Inputs include XLR and detachable Euro-style connectors. Outputs feature covered barrier strip connectors for safety agency compliance. For added versatility, a DataPort permits use with compatible signal processing accessories such as the XC-3 Crossover, SF-3 Subwoofer Filter, LF-3 LF Filter and the DSP-4 with external power supply.

FEATURES

- Rear panel gain controls for tamper resistant operation with 2-dB detents for quick and repeatable settings
- XLR and detachable Euro style input connectors
- DataPort V2 for use with compatible signal processing accessories (XC-3, SF-3, LF-3) and DSP-3 (with external power supply)
- Independent defeatable clip limiters for reduced distortion
- Selectable high-pass filters protect speakers and prevent speaker transformer saturation with minimal effect on program material (30Hz or 70Hz on non-"T" models, 50Hz or 75Hz on "T" models)
- Class H complementary bipolar output circuitry reduces AC power draw and waste heat by 40% (ISA 750 and 800T)
- Front panel indicators include power, signal, and clip
- Covered barrier strip output connectors for safety agency compliance
- Automatic variable-speed fan for quiet normal operation with maximum cooling on demand
- Rear-to-front air flow keeps equipment racks cool
- Stereo, bridge, or parallel operating modes
- Comprehensive protection circuitry including DC, infrasonic, thermal overload, and short circuit protection
- 3-year warranty plus optional 3-year extended service contract

"T" Models

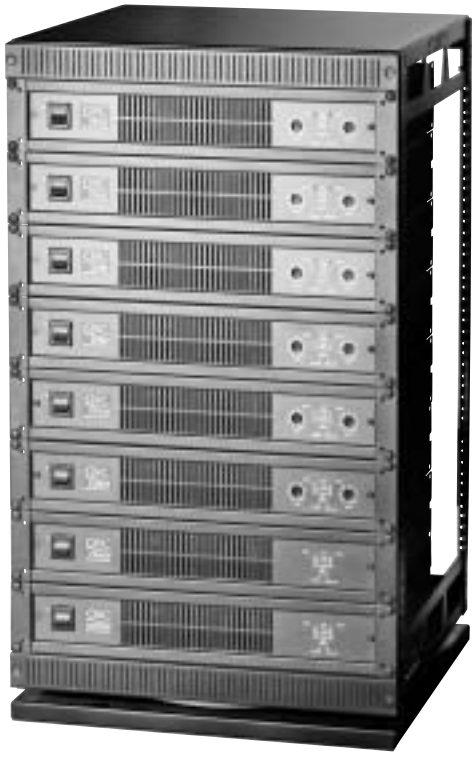
- 25-, 70-, 100-volt, and low-impedance outputs available simultaneously
- Simple barrier strip connection determines output mode

	ISA280	ISA450	ISA750	ISA300T	ISA500T	ISA800T
STEREO MODE, Both Channels Driven Continuous Average Output Power Per Channel						
8Ω 20-20k Hz 0.1% THD	185W	260W	450W	185W	260W	450W
4Ω 20-20k Hz 0.1% THD	280W	425W	650W	280W	425W	650W
2Ω 1 kHz 1% THD	430W	700W	1200W	430W	700W	1200W
XF OUTPUT POWER, STEREO MODE, 50 Hz-16 kHz 0.5% THD						
70V or 100V	-	-	-	300W	500W	800W
25V	-	-	-	240W	400W	640W
BRIDGE MONO MODE Continuous Average Output Power						
8Ω 1 kHz 0.1% THD	600W	900W	1500W	600W	900W	1500W
4Ω 1 kHz 1% THD	830W	1400W	2400W	830W	1400W	2400W
XF OUTPUT POWER, BRIDGE MONO MODE, 50 Hz-16 kHz 0.5% THD						
140V or 200V	-	-	-	600W	1000W	1600W
50V	-	-	-	480W	800W	1280W

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CX SERIES

The CX Series is designed to meet the specialized needs of sound contractors. Eight 2-channel, three 4-channel and one 8-channel including 70-volt models have been designed from the ground up, combining QSC's exclusive PowerWave technology with specific features to meet the requirements of fixed installations. With high output power, versatile loading options, high thermal capacity and reliability, the CX Series is the ideal solution to any permanently installed sound system.



CX 2-Channel Amps

- 8 models to meet your exact power requirements (five low impedance models, three 70-volt direct models)
- Exclusive PowerWave switch-mode power supply technology for high performance and compact size
- Custom integrated security cover for tamper proof installations
- Variable speed fan for low noise
- 1dB detented gain controls for fast and accurate gain settings
- Active In-rush Limiting eliminates AC inrush current, removing the need for expensive power sequencers
- XLR and detachable Euro style input connectors
- Comprehensive front panel indicators include signal, clip, protect and QSC's bridge-mono and parallel-input LEDs
- Dip switch control for clip limiters, high pass filters, bridge-mono and parallel operation
- HD15 DataPort connector for QSControl computer control or QSC signal processing accessories
- Selectable high pass filters protect speakers and prevent speaker transformer saturation with minimal effect on program material (33 Hz or 75 Hz on non-V models, 50 Hz or 75 Hz on V models)
- Barrier strip output connector
- Comprehensive protection circuitry including DC, infrasonic, thermal overload and short circuit protection
- Class H complementary bipolar output circuitry for high efficiency (CX702, CX902, CX1102 & CX1202V)
- Optional external transformer pack for isolated 70- and 100-volt outputs (converts CX302 to 400 w/ch isolated output)
- Amps are 2 RU high, 14″ deep and weigh only 21 lbs. for easier racking & shipping
- 3-year warranty plus optional 3-year extended service contract

CX-4 Channel Amps

Same as 2-channel models except:

- Channel 1&2 and 3&4 bridgeable for maximum flexibility
- High-performance Class AB+B complementary bipolar output circuitry
- Detachable Euro style input connectors
- 3 models to choose from (two low impedance one 70-volt direct)

CX 8-Channel Amp

Same as 4-channel models except:

- Channel pairs bridgeable for maximum flexibility (1&2, 3&4, 5&6, 7&8)
- Four HD15 DataPorts (one per channel pair) for QSControl computer control or signal processing accessories
- Detachable Euro style input and output connectors
- 1dB recessed detented gain controls

CX-SERIES SPECIFICATIONS

Stereo Mode, All Channels Driven

	70v[1]	8-ohms[2]	4-ohms[2]	2-ohms[3]
CX302V	250w	—	—	—
CX602V	440w	550w	—	—
CX1202V	1000w	700w	1100w	—
CX302	—	200w	325w	600w
CX502	—	300w	500w	800w
CX702	—	425w	700w	1200w
CX902	—	550w	900w	1500w
CX1102	—	700w	1100w	1700w
CX204V	200w	—	—	—
CX254	—	170w	250w	450w
CX404	—	250w	400w[4]	not recommended
CX168	—	90w	130[5]	—

[1]1kHz, 0.05% THD [2]20-20k Hz, 0.05% THD [3]1kHz, 1% THD [4](0.1% THD 1 kHz) [5]0.1% THD 20 Hz–20 kHz

MA3

Multichannel Amplifier



The MA 3 is a three-channel amplifier specifically designed for use in paging, foreground music, and background music distribution. Each channel of the MA 3 delivers 40 watts of continuous average power into 8 ohms and 60 watts into 4 ohms. Up to three optional 40 watt, 70.7 volt or 100-volt constant-voltage audio transformers may be installed inside the MA 3, eliminating the need for external wiring or transformer mounting. The combination of a solid, conservative power supply and forced air cooling allows the MA 3 to simultaneously deliver 60 watts of continuous-average-power to all three channels.

- SPiKe dynamic protection circuitry completely safe-guards each channel against over-voltage, under-voltage, overloads, transients from inductive loads, thermal runaway and instantaneous temperature peaks. Biasing is not allowed to occur when an under-voltage condition exists, reducing turn on and turn off transients.
- Fast-response limiters allow the MA 3 to tolerate up to 20 dB of overdrive into 8 and 4 ohm loads while holding THD below 1%. This means no loss of speech intelligibility or harsh clipping. Greatly increases the dynamic range of the system without external limiters. Peak-responding, load-adaptive meters accurately indicate the remaining headroom, and are helpful in setting system levels and indicating signal compression.
- Balanced inputs with Euroblock connectors are provided. Euroblock output connectors accept up to 12 gauge wire. Rear panel level controls allow amplifier sensitivity adjustment.
- Internally selectable 80 Hz highpass filters offer protection against over excursion of small bookshelf speakers and saturation of distribution transformers at low frequencies.

MA6S *Multichannel Amplifier*



The MA 6S supplies six completely independent amplifier channels in one 3-space rack-mount unit. Each channel delivers 100 watts RMS into an 8-ohm load, or 150-watts each into a 4-ohm load. Adjacent pairs of Channels (1 & 2, 3 & 4, 5 & 6) operate bridged to provide 300-watts into an 8-ohm load. Each channel of the MA 6S includes an input Voltage Controlled Attenuator (VCA). This carefully designed VCA prevents clipping at the output of the amplifier for any load for up to 15 dB of input overdrive. This VCA provides additional headroom to each Channel, making the 100 watt ratings effectively much higher, and the results much louder.

- Unprecedented degree of flexibility:
  - Use it as a monitor amplifier for six separate Channels of stage monitor mixing;
  - Drive a pair of two-way speakers bi-amped with two bridged pairs at 300w for the low end and two 100w channels for the high end, each with limiters for driver protection
  - Run three bi-amped monitors at 100w each
  - Drive three passive two-way monitors at 300 watts each
- All channels of the MA 6S are completely independent. Each is contained on its own circuit board making it immune to the actions of any other channel.
- Power supply is designed so that signal won't back-feed from a heavily loaded channel to any other channel. This makes power supply cross talk between channels inaudible.
- Euroblock quick disconnects allow fast and inexpensive connection of input and output signals. Up to 12 AWG wire can be used.
- Individual indicators for each channel indicate the status of the Mute Relay, Clipping Protection, and Safe Operating Area (SOA) Protection circuits.
- Heavy gauge all-steel construction offers a high level of mechanical integrity, even when the amp is rack-mounted and unsupported in the rear. As an added benefit, mounting holes are provided on the side rails so the amplifier may be supported from the back.

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SERVO 120



Ideal for powering near field monitors or a headphone amplifier, the Servo 120 gives you "reference-class" audio performance and reliability in a compact package. Ideal for recording, live sound and home theater situations, the 120's bipolar design and toroidal transformer power supply ensure reliability and quiet performance. A headphone jack with a speaker-disable switch lets you work through late night mixing sessions without disturbing anyone.

- 1RU servo-controlled stereo power amp
- 60w per channel into 4 ohms (stereo), 120w watts into 8 ohms (bridged mono mode)
- Wide, flat frequency response of less than 10Hz to plus 100kHz for high-quality audio
- Relay-controlled power-on circuitry prevents speaker "thumps"
- Individual left and right Level controls with 5-segment, 3-color LED meters
- Push-spring terminal outputs
- Convection-cooled design eliminates fan noise and maintenance problems
- Bipolar design and toroidal transformer power supply
- Overheating and over-current protection circuitry
- Front panel headphone jack with speaker-disable switch
- Balanced 1/4″ TRS inputs and unbalanced RCA inputs on gold-plated connectors for improved conductivity

DS70 *70-Volt Speaker Distribution System*



Rugged and reliable, the 2RU high DS70 transforms the 4 ohm output of a standard power amp to a 25 or 70-volt level for use in multiple ceiling-mounted speaker systems. 4 independent channels allow for 4 mono or 2 separate stereo zones.

- Transforms conventional 4 ohm amplifier output to a 70 or 25 volt level for sound distribution in commercial installations
- 4 independent signal paths
- Binding post inputs and terminal strip outputs
- Maximum power for each output: 120 watts (240 watts in bridged mode)
- 4 high-quality toroidal transformers for full-range audio with extended high frequency reproduction

F800/F1200



The 3RU high F800 and F1200 are rugged amps ideal for sound reinforcement, DJ use, commercial installations and PA systems. The F800's amp section produces 400 watts per side into four ohms and 800 watts into eight ohms in the bridged mono mode. The F1200's amp section produces 600 watts per side into four ohms and 1200 watts into eight ohms in the bridged mono mode. Housed in a three rack space chassis, they both feature a bipolar design to ensure stable operation and accurate audio reproduction. Their front panel includes a Power On/Off switch along with Power, Bridge and Peak LEDs. Other practical features include 1/4″ balanced and RCA input connectors and Binding Post output connectors.

- F800 produces 400w per channel into 4 ohms and 800w into 8 ohms (bridged mono mode), the F1200 produces 600 watts per channel into 4 ohms and 1200 watts into 8 ohms (bridged mono mode)
- Dual temperature-sensitive, speed-controlled fans for maximum efficiency and reliability
- Stable bipolar design for long life and enhanced audio reproduction
- Dual protection, power and bridging LEDs
- Balanced 1/4″ and RCA input connectors
- Binding post output connectors
- Resettable in-line fuse on rear panel

SERVO 170/260/550

Power Amplifiers



A versatile performer in a compact package, the Servo 170 is excellent for near field monitoring, stereo midrange/high frequency bi-amping applications and low power distribution systems. It also works effectively for powering vocals and acoustic instrument setups in small clubs, restaurants and bars. Like all Samson Servo amps, the 170 is designed to ensure quiet, reliable performance every time you use it.



A mainstay of the line, the Servo 260 can be used to power near field monitors, passive headphone distribution systems in the studio and small live sound reinforcement setups. It is also a great choice for stereo keyboard or guitar amp setups with external mixer or preamps, or amplifying a large screen TV or video system. The Servo 260 is dedicated to transparent audio, consistent performance and the rugged construction you need from a professional amplifier.



Using the Servo 550 in the bridged mono mode is a great way to power a separate subwoofer system as an enhancement to your overall sound. The 550 also shines in other live sound reinforcement uses, monitoring and dedicated recording applications like powering near and midfield monitors and passive headphone distribution systems. A/V installers appreciate the 550's versatility, quiet performance and reliability for distributed sound, clubs, restaurants and home theater settings.

Servo 170

- Servo-controlled stereo power amps in two rack spaces
- 85 watts per channel into 4 ohms
- Wide, linear 20Hz to 50kHz frequency response for superior audio performance
- Relay-controlled power-on circuitry prevents speaker "thumps"
- Front-panel Clip and Idle LEDs for both channels
- 1/4″ and push-spring terminal outputs
- Balanced 1/4″ TRS and unbalanced RCA inputs
- Circuitry protection against overheating and over-current
- Independent left/right input level controls with 41 detents
- Bipolar output stage for quiet audio performance you can always count on
- Convection-cooled design eliminates fan noise for quiet performance

Servo 260 Step-up Features

- Produces 130 watts per channel into 4 ohms

Servo 550 Step-up Features

- Servo-controlled, 3-space rack-mount stereo and mono-bridgeable power amp
- 275w per channel (stereo) into 4 ohms and 550w into 8 ohms (bridged mono mode)
- Fully balanced 1/4″ inputs
- Balanced 1/4″ and binding post outputs
- Front-panel Clip, Idle, Protection and Power LEDs

	Servo 170	Servo 260	Servo 550
4Ω	85w	130w	275w
8Ω	60w	90w	235w
8Ω Bridged	—	—	550w
Frequency Response (-3dB)	<20 Hz to 50 kHz	20 Hz to 50 kHz	<10 Hz to 85 kHz
Dimensions (WHD)	3.5 x 19 x 9.5″	3.5 x 19 x 9.5″	19 x 5.2 x 9.2″
Weight	13.7 lb.	17.6 lb.	32.4 lb.

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S500/S700/S1000/S1500/S2000



Sturdy and reliable, the S500 is perfect for live sound or fixed installations where 500 watts is the right amount of power.

Exceptionally rugged and reliable, the S700 fits seamlessly into a variety of sound reinforcement applications and fixed installations. It is capable of producing 350 watts a side and 700 watts (mono bridged) from a compact rack-mountable format. Two temperature-sensitive, speed-controlled fans and a bipolar design as well as comprehensive control and protection capability ensure stable performance over the long haul.

A smart choice for powering all kinds of live sound projects, commercial installations and PA systems, the rugged S1000 provides plenty of power, reliability and protection. Dedicated to transparent audio and long term reliability, the S1000 is built tough and provides exceptional control and interfacing capability.

Efficient and highly reliable, the S1500 provides plenty of power on the road, in commercial installations and for PA use. It produces 750 watts per channel into 4 ohms (stereo mode) and 1500 watts into 8 ohms (bridged mono mode). Its flat frequency response ensures great audio. Other features include full front-panel control, metering and protection capability and a wide range of interconnect options.

The rugged S2000 delivers plenty of power, reliability and protection in live sound situations, commercial installations and PA situations. It produces 1000 watts per channel into 4 ohms (stereo mode) and 2000 watts into 8 ohms (bridged mono). It offers a wide frequency response; dual temperature-sensitive, speed-controlled fans, Bipolar design for consistent performance and full front-panel control, metering and protection capability along with a wide range of interconnect options.

S500 Features

- Heavy-duty dual rack space stereo power amplifier
- 250 watts per channel into 4 ohms, 150 watts into 8 ohms (stereo) and 500 watts into 8 ohms (bridged mono)
- Temperature-sensitive, speed-controlled fan
- XLR and locking TRS inputs; banana jack outputs with Speakon connectors
- Parallel outputs allow several amps to be "daisy-chained" together
- Front panel input level controls with 41 detents and 3-segment output LED meters
- Overheating and over-current protection circuitry with LED monitoring

S700 Step-up Features

- Produces 350 watts a side into 4 ohms (stereo mode) and 700 watts into 8 ohms (bridged mono mode)
- Dual temperature-sensitive, speed-controlled fans
- Bipolar design for greater stability
- Relay-controlled outputs linked to the protection LEDs

S1000 Step-up Features

- 500w per channel into 4 ohms (stereo) and 1000w into 8 ohms (bridged mono mode)
- Linear frequency response for exceptional audio performance

S1500 Step-up Features

- 750 watts per channel into 4 ohms (stereo mode) and 1500 watts into 8 ohms (bridged mono mode)
- Flat frequency response for superior audio performance
- AC mains: IEC and Neutrik Powercon connector included

S2000 Step-up Features

- 1000 watts per channel into 4 ohms (stereo mode) and 2000 watts into 8 ohms (bridged mono)
- Linear frequency response for exceptional audio performance

YAMAHA

P1600/P3200/P4500



"P" Series Amps with Exceptional Efficiency

Remarkably efficient, the "P" series also offers excellent cost per watt value. Utilizing Yamaha's EEEngine technology, these amplifiers deliver outstanding sound quality with remarkable efficiency. All of this power and performance comes in a compact 2-space rack-mount size that makes installation and transport easy. If you need transparent amplification with excellent reliability and superior performance, give the new "P" series a close look.

FEATURES

Power Efficient

Yamaha's EEEngine (see box) consumes power efficiently, which allows the amplifier to produce the same amount of output power with about half the input power required. The EEEngine outperforms other efficiency-enhancing schemes like drive voltage switching, pulse-width modulation and switching series regulators - without degrading sound quality.

Quality Sound

Transparent amplification without sound coloration is supplied with a frequency response of 10Hz to 50 kHz. A high damping factor of >200 provides excellent bass response and total harmonic distortion is <0.05% — perfect for today's digital sources and musical instruments.

Input/Output

Input attenuators are 31-position, dB-calibrated and allow independent L/R adjustment with smooth, noise-free control. Balanced XLR jacks, a barrier strip and balanced 1/4″ phone jacks accommodate input connections. Five-way binding posts provide solid, reliable speaker connections.

Three Output Modes

— In stereo mode, separate input signals are routed to each channel with independent L/R signal control.

— An extra margin of flexibility is provided in parallel mode, which feeds a single input signal to both channels while maintaining independent L/R signal control.

— Bridged mono mode can be used for extra high power.

Comprehensive Protection

◆ They offer comprehensive protection designed to prevent accidental damage to the amplifier itself and any connected speakers.

— Power ON muting suppresses the outputs until the amplifier circuitry has stabilized.

— To keep the operating temperature stable, they are equipped with ultra-quiet variable speed cooling fans.

— Added protection is supplied with heat oversink protection, noise filtering, a PC limiter for short circuit protection and an attenuator security cover to protect level settings.



Yamaha's EEEengine

By means of switching drive system, Yamaha's EEEngine technology reduces AC power requirements and heat loss while maintaining output power and sound quality. Input power is supplied through a high-efficiency current buffer. During low requirement periods, it transparently switches the input power on and off automatically as needed. As the signal becomes large, an independently responding auxiliary power line supplies the additional power, as required. The output isn't compromised because the auxiliary power line is driven by the power supply voltage, which can maintain the maximum output to the speaker load. The amplifiers make more efficient use of the AC power than conventional amplifier technology. With EEEngine technology, the electric company will wonder how you're doing it!

Power Rating- (20Hz - 20kHz)

P1600:
160W @ 8Ω stereo, 200W @ 4Ω stereo, 400W bridged mono

P3200:
340W @ 8Ω stereo, 440W @ 4Ω stereo, 880W bridged mono

P4500:
460W @ 8Ω stereo, 620W @ 4Ω stereo, 1240W bridged mono

P2500/P3500/P5000/P7000S

"P" Series II Amps



Yamaha's engineers have designed these new P-Series power amplifiers as perfect mates to the popular Club Series speakers. To achieve this, they provided them with big power output that matches the power handling capabilities of the speakers. Each amp is also equipped with the specially designed YS (Yamaha Speaker) Processing circuit to deliver a signal that is tuned to the specific response of the Club Series speakers for superior audio quality and performance.

The amps features both XLR and 1/4˝ TRS inputs and Neutrik Speakon, phone plug, and 5-way binding post outputs to make connections quick and easy. Other features include Yamaha's exclusive EEEngine technology which delivers high performance with exceptional efficiency, sweepable high and low pass filters for optimizing output to any loudspeakers, compact and durable 2U chassis, and variable speed cooling fans.

FEATURES

- They deliver a robust amount of professional quality power in a wide range of configurations. In a 4- ohm bridged application, the P7000S delivers 3200 watts, the P5000S delivers 2600 watts, the P3500S 2000 watts, and the P2500S 1300 watts.
- Each is equipped with independent sweepable high and low pass filters on each channel so you can optimize output for subwoofer or full range systems to provide peak audio quality and performance from any speaker system you may be using.
- The P2500S and P3500S are respectably light in weight making transport and setup less strenuous. What's amazing though, is the comparison in weight per watt when you look at the P5000S and P7000S. The use of a switching power supply makes these two models very lightweight while maintaining superior audio quality.
- To provide the most flexible connectivity possible, all four are equipped Neutrik Speakon output jacks, and 1/4-inch output jacks for each channel in addition to 5-way binding post. Channel inputs consist of both XLR and 1/4-inch TRS jacks.
- Continuously variable-speed fans mounted internally on either side of the front panel offer quiet, efficient cooling.
- Their industrial design delivers a refined and impressive look with deep blue faceplates bordered in silver trim, handles, and mounts. The chassis is specially designed to provide added durability and strength.
- EEEngine Technology makes more efficient use of AC power by reducing power consumption and heat generation without degrading output power or sound quality. See box on previous page.
- Comprehensive protection consists of power on/off muting, DC detection, thermal protection, current limiting, and a protective cover for the attenuators.
- With YS (Yamaha Speaker) Processing, you get direct compatibility with Yamaha's Club Series speaker. This circuit optimizes output from the amps to match the characteristics of Club Series delivering a signal that is precisely what the speaker requires. With the YS Processing system, extra-smooth highs and enhanced low frequency from your Club Series loudspeaker system is assured.

Models	P7000S	P5000S	P3500S	P2500S
1kHz, THD+N=1%				
8Ω/Stereo	750W+750W	525W+525W	390W+390W	275W+275W
4Ω/Stereo	1100W+1100W	750W+750W	590W+590W	390W+390W
8Ω/Bridge	2000W	1500W	1180W	780W
20-20kHz, THD+N=0.1%				
8Ω/Stereo	700W+700W	500W+500W	350W+350W	250W+250W
4Ω/Stereo	950W+950W	700W+700W	450W+450W	310W+310W
8Ω/Bridge	1900W	1400W	900W	620W
1KHz, 20mS nonclip				
2Ω/Stereo	1600W+1600W	1300W+1300W	1000W+1000W	650W+650W
4Ω/Bridge	3200W	2600W	2000W	1300W









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The Professional Audio SourceBook

Section 7

Studio Monitors

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M1 ACTIVE MK2

The M1 Active MK2 combines Alesis' expertise in reference monitor design and studio amplification to provide professional accuracy, clarity and performance in an affordable powered monitor. A bi-amplified reference monitor, the M1 Active MK2 is incredibly accurate with a wide sweet spot, high power handling and detailed high- and mid-frequency response that translates perfectly to any play-back system. Ideal for desktop video and post production facilities as well as home and project recording studios.



- Attractive mirror-imaged, charcoal- toned cabinet and shielded drivers allow placement next to computer or video monitors, while blending beautifully into your studio's decor without being obtrusive.
- Can be mounted either horizontally or vertically to fit your studio's requirements.
- The woofer and tweeter were designed for the most linear, low distortion response. The 6.5″ woofer cone is formulated from non-woven carbon fiber. Used in the skins of combat aircraft wings, carbon fiber has more tensile strength and is 25% lighter than polypropylene (the most common driver material). The result is quicker transient response in the low frequency range for superb bass response as well as a vast improvement in the detail of critical upper-mid frequencies.
- Internal active crossover and custom dual amps ensure consistent performance under different studio conditions and program material. The result is a broad, extremely flat frequency response, clear imaging with a wide sweet spot, high power handling and detailed high- and mid-frequency response.
- Pure silk 1″ tweeter dome's natural response prevents ear fatigue during long mix sessions. It also features an internal pole piece mounted phasing plug and is ferrofluid cooled to maintain the best balance of transient response to power handling.
- Offers a audiophile-quality 75w woofer and 25w tweeter amp along with 8th order high- and low-pass electronic crossover filters. The crossover point is a low 1500Hz, and produces an extremely wide dispersion zone as well as low midrange coloration.
- Time alignment circuitry in the tweeter section of the crossover synchronizes frequency output from the high and low drivers avoiding the time smear that's typical in less advanced powered speakers.
- Dual front-mounted ports provide fast, coherent and extended low frequency augmentation. Also, the unique offset design of the high frequency driver allows for the flattest frequency and power response in a speaker that offers such wide dispersion.



Monitor One MK2

An award winning studio monitor, the Monitor One has an extended dynamic range, excellent imaging and transient reproduction, powerful bass and smooth, extended high frequency detail. Ideal for high-quality monitoring in project studios and commercial recording facilities. Exclusive SuperPort speaker venting technology extends low frequency response while eliminating the "choking" effect of port turbulence.

- 6.5″ polypropylene low-frequency driver with tight, accurate bass response is matched with a 1″ pure silk-dome, high-frequency transducer for smooth, accurate detail and response, and a broad sound stage
- Ferrofluid-cooled, high-frequency driver prevents ear fatigue during long mixing sessions
- 45 Hz – 20 kHz +/- 3 dB frequency
- 2kHz crossover point provides a true, flat frequency response for accurate mixes that translate to other playback systems
- The cabinet has an attractive professional finish, and is a mirror image; left/right pair for symmetrical speaker placement
- Magnetic shielding allows use next to video and computer monitors
- Radiused edges for reduced edge diffraction, along with improved porting. Thanks to exclusive SuperPort venting technology, the monitor eliminates the "choking" effect of smaller, shorter ports. The result is a low-frequency response which extends well below comparable near field monitors.

PROLINEAR 720DSP

Powered Studio Reference Monitor with DSP Control



Building on Alesis' award winning speaker designs, the ProLinear 720DSP delivers unprecedented accuracy, control, and flexibility for any studio monitor application. The ProLinear 720DSP is a bi-amplified two-way reference monitor with a 7˝ kevlar woofer and a 1˝ silk-dome tweeter, with several distinct advantages over traditional studio monitors:

By integrating proprietary 28-bit DSP, the ProLinear 720DSP implements the crossover entirely in the digital domain to provide a precisely controlled response. The ProLinear 720DP further leverages its digital control to enable users to set and store multiple EQ curves for individual speakers, or for every speaker in the studio. Settings can also be adjusted via the built-in 4-band fully parametric EQ. Additionally, by using the PC interface, users can adjust and set EQ settings remotely from a "sweet spot" listening position. No other monitor offers this degree of control, flexibility, or total performance.

- Built-in digital crossover and 4-band parametric EQ with 8 pre-sets and 8 user defined settings
- 24-bit/48kHz processing
- 80 watts (LF) / 40 watts (HF)
- Serial interface allows PC control of up to 16 speakers



PROACTIVE 5.1

THX-Certified Surround Sound Speaker System

The ProActive 5.1 is equipped with Dolby Digital and DTS hardware decoding for true 5.1 digital sound. The first system to deliver THX-certified surround sound to the professional audio and music industry, the ProActive 5.1 is the ideal solution for the home recording studio, with simultaneous connection to up to four audio sources including PCs, home audio and recording equipment, and other sources such as DVD players and portable audio players.

THX certification involves rigorous testing of sound performance, user interface and construction. All speakers that receive THX certification must meet stringent performance parameters, including: bandwidth, frequency balance, peak sound pressure levels and the ability to play up to their rated output level.

- THX-certified, 5.1-surround sound system with 450 watts of RMS power.
- 8˝ long throw subwoofer driver with a flared bass port for powerful, distortion-free bass.
- Dolby Pro Logic II for creating realistic 5.1 surround sound from stereo music, movies and games.
- Digital optical and coaxial inputs, as well as six channel analog inputs, for simple connection to nearly any audio device.
- System is controlled by a digital console with a wireless remote, for easy adjustment of the speaker controls from anywhere in the room.

DESKTOP SPEAKERS



DM-5
Bi-amp Monitors

Durable, dependable and affordable, expect superior performance from this near-field monitor. 120mm woofer is powered by a 30W amplifier, the 19mm tweeter by its own 20W amp and both have their own level control. In addition to analog input (Speakon, XLR and 1/4″ in a single connection), accepts S/PDIF, coaxial and optical digital input and supports 24bit/96kHz data.

- 120mm LF driver with 30w amp, 19mm HF driver with 20w amp
- Bass-reflex ducts for rich, bass-range reproduction
- Digital in and power indicators on the front panel
- S/PDIF digital inputs (coaxial and optical) support 96 kHz sampling rate and 24-bit D/A conversion

MA-5A/MA-5D
Analog & Digital Stereo Micro Monitors



Bass-reflex system powered speakers in a compact and stylish design, the MA-5A provides a wide range of dynamic sound. The MA-5D is identical but adds a digital input on the front.

- 5w x 5w, 70 mm full range speakers
- Equipped with 2 inputs (RCA/stereo mini); you can also mix 2 analog input sources
- Plug-in a variety of audio devices, such as CD/MD/DVD player and electronic musical instruments
- Bass, Treble control knobs on front panel

MA-10A/MA-10D
Active Analog & Digital Desktop Monitors



Available in Blue or Black

The MA-10A is a compact active monitor featuring some remarkable specifications as well as a striking contemporary blue or black natural wood finish. Fine wooden cabinets ensure a more defined sound quality. They are magnetically shielded and provide 20 watts of genuine power (RMS) with bass-reflex enclosure system. It is so important to monitor your audio with a good pair of speakers that can guarantee good dynamic range and frequency response.
The MA-10D offers the same great sound as the MA-10A—plus adds digital input capacity. Listen to any source—analog or digital. Connect digital sources such as computers, MP3 players, MiniDiscs, DAT machines, CD players and cassette decks. High resolution 24-bit/96 kHz D/A converters provide great detail and impressive dynamics. Impressive performance from a very small package.

- 10w + 10w stereo amplifier powers both monitors
- Mini-stereo Line 1, RCA L/R Line 2
- 2-way bass-reflex enclosure, wood grain cabinet for better sound response (frequency response is 45Hz to 35kHz
- Bass, Treble controls and 2 independent line volume

MA-20D
Digital Stereo Near Field Monitors



Designed for the higher-end consumer and hobbyist, the MA-20D speakers are attractive Near-field Monitors with impressive sound capabilities - two 20-Watt amplifiers and a stylish black & silver design. Building on the widely popular, MA-10 series, the MA-20D's have twice the power and all the style of the MA-10's. They include the same functionality of the MA-10D and MA-10DBK speakers; including Treble & Bass controls, two independent Volume controls, S/PDIF digital input, and a headphone out. Each speaker in a MA-20D pair is individually powered with its own amplifier and have a bigger woofer for a better low-end response.

- Superb Response for unparalleled sound quality in its price range
- 20W x 20W power in a 2-way bass-reflex design
- Power amp in each speaker for uniform stereo sound
- 4¾″ woofer for good punch and excellent Bass Response
- 24-bit/96 kHz capable S/PDIF connections

20/20

Precision Direct Field Monitors



The 20/20 Direct Field Monitor delivers wonderfully clean, accurate, pleasing sound at a price most musicians would consider eminently affordable. Exceptionally reliable, attention to detail permeates every aspect of the design. Custom asymmetrical second-order passive crossover ensures smooth sonic transition from the woofer to the tweeter at the crossover point. The tweeter is recessed to provide flatter frequency response. The diecast aluminum trim ring which minimize acoustic discontinuity between the driver and the cabinet, is also a unique, self-aligning four-point mounting system for the woofer.

Tweeter

- Magnetically shielded, 25mm natural silk dome provides accurate, detailed high frequency reproduction.
- Internal high frequency pole piece damping element aids in smoothing out the signal, allowing for extended listening periods.
- For improved reliability, there is a ferrofluid-cooled aluminum voice coil support that reduces heat; a special flexible lead wire minimizes wire fatigue as well as insures consistent performance.

Woofer

- The woofer provides smooth, consistent extended-range frequency response without the need of added EQ
- The driver itself is a magnetically shielded 8″ mineral-impregnated polypropylene cone with a highly-damped linear rubber surround
- It also boasts a 1½″ high-temperature voice coil for reliable operation at high power levels

Cabinet

- 5/8″ laminated MDF cabinet filled with acoustical damping material reduces internal standing waves—a perfect acoustic complement to the transducer components.
- Front-mounted large diameter low air restriction bass port delivers direct, uncolored low frequency reproduction, even at very high volumes.
- The port is front-mounted, allowing the speakers to be placed close to walls without compromising the low end response.

20/20BAS

Biamplified Precision Direct Field Monitors

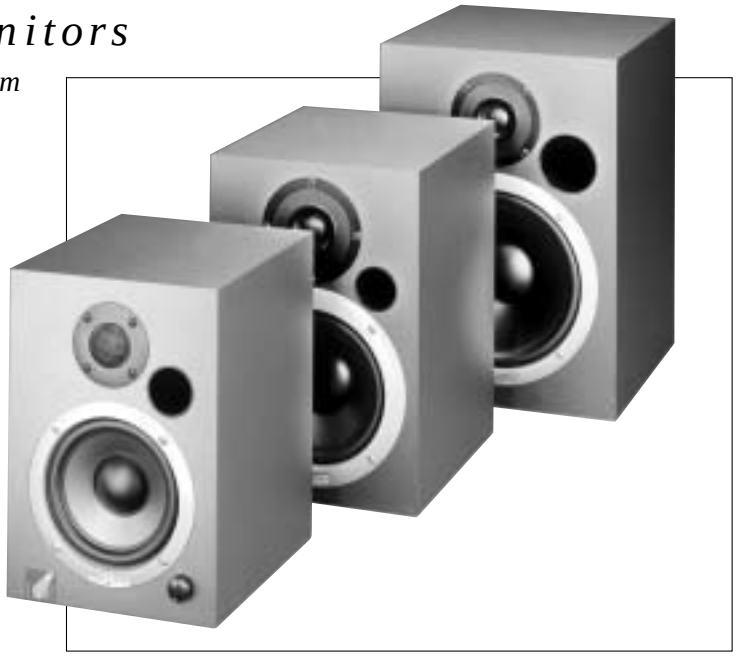


The 20/20BAS puts world-class reference monitoring into the hands of musicians previously shut out by the high cost of those systems. Based on the 20/20 design, which in itself offers exceptional sound, the 20/20BAS takes it to the next level with an active asymmetrical fourth-order crossover—with phase response so accurate that the resulting stereo imaging simply must be heard (make that seen) to be believed. Add the custom 130w amplifier designed specifically for the woofer, followed by the 70w tweeter amp—that's 200 watts of ultra clean power per side, and then independent trim controls for each amplifier, making it easy to adjust the speakers' response to your particular monitoring environment. Simply put, the 20/20 delivers exactly what you want in a reference monitor system: superbly clear, detailed, sound. Flat, uncolored, accurate frequency response. Precise imaging. Non-fatiguing to the ears.

EVENT ELECTRONICS

PROJECT STUDIO PS5/PS6/PS8

Bi-Amplified Direct Field Monitors



It's about confidence. Confidence that your monitoring system is accurate. Confidence that what you're hearing from your speakers isn't being colored. Confidence that your mixes will sound the way you meant them to sound when they get played in the car, or in your living room – or in the office of a record company executive. The Project Studio monitors inspire that confidence, giving you the same precise, accurate sound as the award-winning 20/20 series. Not everyone has the space for full-size direct field monitors. But everyone does have the need for speakers that provide pure, accurate, detailed sound. And that's exactly what you get with the Project Studio series:

Full-range, non-fatiguing monitoring in a small profile format. More importantly, PS speakers provide you with monitoring you can trust, so you know that your mixes will sound the same in the outside world as they do in your studio. This is accomplished through the use of custom driver components, as well as amplifiers and electronic crossovers designed specially for those components. The result: Increased dynamic range. Higher SPL. Greater transient response. Improved damping. Smoother phase response. Lower intermodulation distortion. Oh, yes, and great sound.

TR5/TR8

Tuned Reference Bi-amplified Direct Field Monitor Systems



Like their names intimate, the TR (Tuned Reference) 5 and 8 monitors are precision tuned to provide flat frequency response and uncolored sound, giving you a true sonic reference for mixing. So when you mix on a Tuned Reference monitor, you know your mixes will translate accurately to other playback systems.

However, you might think that this level of mix precision comes at a high price. But thanks to a powerful new ultra-efficient amplifier (actually two of them, since these are biamplified systems), they are able to bring you high definition, non-fatiguing monitoring at prices so low they're downright silly. But don't let the cost fool you: The TR series monitors are filled with time-tested Event technology, and designed by the same engineering team behind their full line of critically-acclaimed monitors. So pick up a pair of TR monitors today. Your mixes will thank you.

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ONE HOUR FREE PARKING
AT 349 W. 34th STREET (with purchase of $100 or more)

	PS5	PS6	PS8	TR5	TR8
LF Driver	5¼″ magnetically shielded mineral-filled polypropylene cone with high-temperature voice coil and damped rubber surround	6½″ magnetically	8″ magnetically	5¼″ magnetically	8″ magnetically
HF Driver	1″ magnetically shielded natural silk dome with ferrofluid voice coil coolant	1″ magnetically shielded natural silk dome with ferrofluid voice coil coolant	1″ magnetically shielded natural silk dome with ferrofluid voice coil coolant	25mm magnetically shielded natural soft dome neodymium with ferrofluid-cooled voice coil	1″ magnetically shielded natural silk dome with ferrofluid-cooled voice coil.
Frequency Response	45-39k Hz, 53-19k Hz, ±3dB, Ref 500Hz	45 – 20k Hz, ±3dB, Ref 500Hz	35 – 20k Hz, ±3dB, Ref 500Hz	45-39k Hz; 53-19k Hz, ±3dB, Ref 500Hz	35Hz – 20kHz, ±3dB, Ref 500Hz
Crossover	2.6kHz active 2nd order asymmetrical	2.6kHz active 4nd order asymmetrical	2.2kHz active 4nd order asymmetrical	2.6kHz active 2nd order asymmetrical	2.6kHz active 4nd order asymmetrical
LF Amplifier Power	70W	70W	70W		
HF Amplifier Power	30W	30W	30W		
Noise	>100dB below full output, 20kHz bandwidth, unweighted				
Input Connectors	XLR and 1/4″ connector accepts balanced or unbalanced sources			RCA connector; accepts unbalanced sources	
Polarity	Positive signal at + input produces outward LF cone displacement				
Input Impedance	40kΩ balanced, 20kΩ unbalanced				
Input Sensitivity	0.9V input produces full output with Input Sensitivity control at maximum			0.5V RMS input at the balanced ins or 0.25V RMS input at the unbalanced in produces output with Input Sensitivity control at maximum	
Input Level Control Range	20dB	20dB	20dB	20dB	20dB
Dimensions (WHD)	7½ x 10½ x 9″	8¼ x 12½ x 10″	10¼ x 14¾ x 11¾″	7.5 x 10.5 x 9″	10¼ x 14¾ x 11¾″
Weight	16½ lbs. each	23 lbs. each	26½ lbs. each	12.6 lbs. each	24.7 lbs. each

S250 System Subwoofer

The S250 Subwoofer is the core foundation of a professional full bandwidth surround sound monitoring system. With a massive 15″ driver and 250W of clean, linear power, the S250 effortlessly provides the high SPL low frequency output necessary for multi-channel production work. Why a subwoofer with six inputs (discrete, active, balanced inputs, at that)? The first input is used for monitoring the Dolby Surround LFE (Low Frequency Effects) channel; signals appearing at the LFE input are fed directly to the subwoofer amplifier through a dedicated 120Hz crossover. The other five inputs are for the five mains channels. These signals are summed into an additional bass channel, used to extend the main monitors' frequency response well below their natural roll-off.



- The dual input, dual crossover design features three operational modes, allowing you to monitor the Dolby LFE channel, extend the low frequency performance of the mains monitor channels, or do both functions simultaneously.
- Includes a variable Monitor Blend crossover control, fed by the summed mains signal; use the control to precisely and seamlessly match the subwoofer's high end with the natural low frequency roll-off of the mains monitors.
- The Variable Phase Control, also fed by the summed signal, allows you to compensate for phase differences that arise from sub /mains-to-listener placement. The result is system-wide low frequency response that is accurate, defined, and completely in your control.
- This flexibility, along with low frequency reproduction that is simply unrivaled for clarity, definition, and pure, thundering power, makes the S250 the perfect complement to today's direct field monitoring systems--including the 20/20bas, 20/20, and Project Studio models.

PM-1



Powered Studio Monitors

Listening correctly doesn't have to be expensive. With 120 watts of biamped power, a bass response that's full and dynamic, and a high end that sizzles, the PM-1 not only delivers astonishing sound—it does so at an even more astonishing price. Consisting of a 6.5″ LF driver and a 1″ soft dome tweeter, the PM-1 offers an impressive 50Hz to 20kHz frequency response and a biamped output power of 75w+45w @ 8 ohms. The amplifiers are precisely calibrated to match the performance of the drivers

High Frequency Driver

The voice coil is made of super high purity copper wire to achieve very low distortion. The 1-inch soft dome tweeter employs Fostex's UFLC technology (poly urethane film laminated cloth) to achieve lightweight and high stability performance.

Low Frequency Driver

The 6.5″ LF driver has been developed to output extremely pure music reproduction by employing the latest technology in cone material. The LF cone uses a mixture of cut and milled fibers that are made from aromatic polyamide. It is then impregnated with resins to simultaneously achieve high rigidity and optimum damping. The cone has an olefin film thermally adhered to its surface to control frequency response and to establish long term reliability. The center cap is made of non-wood cellulose material and is also impregnated with resin.

PM-1 SPEAKER SPECIFICATIONS

CABINET DIMENSIONS
8.5 x 15 x 11.25″ (WxHxD) - includes heat sink

WEIGHT 22.9 lbs.

FREQUENCY RESPONSE
Free-Field Frequency Response:
50Hz ~ 20KHz ± 2dB
Low Frequency Cutoff: 38Hz (-10dB)
High Frequency Cutoff: 23.5Hz (-10dB)

ACOUSTIC OUTPUT
Maximum Short Term SPL:
80Hz ~ 3kHz > 105dB SPL @1m

AMPLIFIER POWER
High Frequency 45w, Low Frequency 75w

AMPLIFIER DISTORTION
<0.016% THD @ 30 watts/8Ω 20Hz-20kHz

AMPLIFIER S/N RATIO
>86dB; 20Hz-30kHz not weighted

POWER REQUIREMENTS
120v AC , 60Hz

POWER CONSUMPTION
100 watts

PS-3.1 *Powered Monitor System*

The PS-3.1 offers a complete 2-way monitoring solution, combining one 5¼″ bass reflex subwoofer and a pair of matching shielded satellite speakers with 3″ full-range drivers in a compact, affordable package. The subwoofer contains its own 15-watt power amplifier, a crossover network, and two 5-watt power amplifiers for powering the satellite speakers—making the PS-3.1 a self-contained system that fits nearly any studio application without the need for external amplification. System response is 80Hz-20kHz.



- Satellite dimensions are 7¼ x 6⅜ x 4¹¹⁄₁₆″ (HDW) and weigh 3.7 lbs each.
- Subwoofer is 10 x 11¼ x 8¼″ (HDW) and weighs 14 lbs.

P95 Right Angle Mic Stand Adapter.....**14.95**

P96 Right Angle Mic Stand/Swivel.....**28.95**

P97 Wall Mount Bracket.....**29.95**

B&H PHOTO - VIDEO - PRO AUDIO *EQUIPMENT LEASING AVAILABLE*

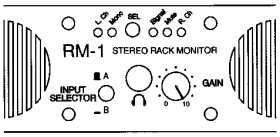
RM-1

1 RU Stereo Monitor



The ideal solution for monitoring in tight industrial environments, the RM-1 is the perfect speaker for machine rooms, VCR monitoring, surveillance, mobile and stationary control rooms, theme park applications or any other situation where monitoring is required and space is tight. The unit is a 1 rack space high industrial design, utilizing robust construction for around the clock operation. The aluminum front panel has a multitude of user options so that the audio output may be configured for the application at hand.

- A/B input select switch selects the A or B stereo group to output to the drivers or headphone jack. Rear panel, stereo left and right inputs let you monitor two different stereo sources from one unit.



- Dual Concentric Volume control can be used to adjust the left or right inputs independently. Front portion of the control adjusts the left channel; the rear portion of the control adjusts the right channel input volume.
- Two stereo balanced +4 dB inputs with four XLR and 1/4″ Neutrik combo connectors
- Front panel stereo Gain control
- On-board electronic matrix switching for: L Channel only, R Channel only, Mono, Stereo and Mute
- High intensity multi-colored LEDs for mode tallies
- Front panel headphone jack with gain control; mutes speakers when used with phones
- Signal presence indicators change with intensity of audio input
- Multi-line power conversion with on-board input transformer
- Two 15-watt in 8 ohms amplifiers
- Fully shielded to prevent interference from video monitors



6301 BEAV

A small, portable, versatile monitor that delivers impressive sound, the 6301 BEAV can be used in pairs or alone. Ideal for use with instruments in project studios, on stage, in remote recording vehicles, video productions and multimedia presentations. The 6301 BEAV accepts any line level input and has a built-in 10-watt amplifier (8 ohm load) that is perfectly matched to a magnetically-shielded 10cm full range driver to deliver big sound, full of clarity from such a small monitor. Manufactured from cast aluminum, these little monitors have earned a reputation for being rugged and road worthy.

- XLR and 1/4″ inputs
- Rotary level control
- EXT SP terminal
- Automated protection circuit
- Optional 9610 Bracket (vertical/horizontal)

SPA-11



The powered SPA-11s offer superb sound in compact, portable cabinets. Each incorporates a 100-watt internal amplifier with two 10cm full-range speakers, mic and line inputs as well as a level control, and they can be daisy-chained and stacked for multiple speaker coverage. A number of mounting and stacking accessories are available to facilitate installation - both temporary and permanent. Constructed of resin-impregnated plastic for durability and superior acoustics.

OPTIONAL MOUNTING ACCESSORIES FOR SPA-11

P60 Vertical U-bracket **59.95**
P61 Horizontal U-bracket.....**59.95**
P63 Vertical Stacking Kit.....**29.95**
P64 Horizontal Stacking Kit **19.95**

GENELEC

1029APM

Compact 2-Way Active Nearfield Monitor



With performance comparable to much larger systems, the 1029A's excellent dispersion and precise imaging, together with its compact size, make it ideal for near field monitoring, mobile vans, home studios, multimedia and home theaters.
The amp unit includes an active electronic crossover, over-load protection circuitry and two power amplifiers, one for each driver. The 1029A has a LF extension (-3dB at 68Hz) sufficient for most monitoring applications. (If a lower cutoff frequency is required, it can be complemented with the 7050A subwoofer). Genelec's unique Directivity Control Waveguide (DCW) technology provides excellent stereo imaging and frequency balance, even in difficult acoustic environments. Versatile tone controls allow further matching of the system to its surroundings. A pair of 1029A's can produce peak acoustic levels of over 110 dB SPL at 1m. The speakers can be used vertically or horizontally, and are easy to set up and use. The 1029A's integrated design allows the amplifiers and the drivers to be calibrated as a single unit ensuring consistent quality. The rugged cast aluminum cabinet has rounded corners and a hard painted outer surface. Available in black, gray, or white cabinets. (The protective grilles and volume and power knobs are black regardless of cabinet color).

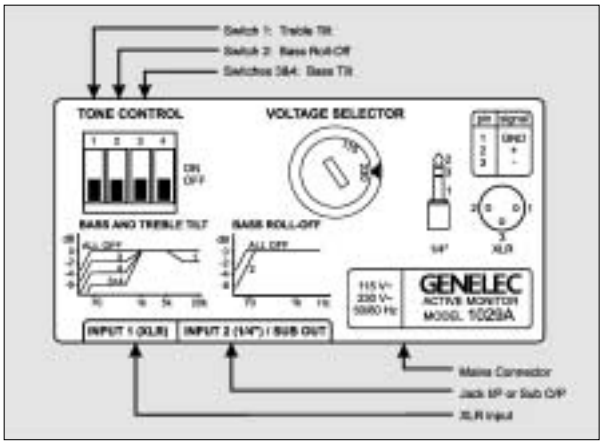
FEATURES

Crossover Filters

Although more commonly used in large and expensive control room monitors, an active crossover is the ideal method for dividing the input signal between the driver units. To maintain uniform frequency balance in differing acoustic environments, special calibrated controls are included in the active crossover network. These controls include treble 'tilt', bass 'tilt' and bass 'roll-off' switches.

Amplifiers

The bass and treble amplifiers produce 40 watts of output power each, with very low THD and IM distortion values. The amps are designed to ensure the highest subjective sound quality possible. The amplifier unit also contains a protection circuit that monitors the output levels and prevents any damage to the drivers. This makes the system immune to overloads and spurious signals.



Two input connectors (balanced XLR and 1/4˝ TRS) offer great flexibility as they can be used in parallel with two sources connected to the monitor at the same time. An additional configuration is using a single 1029A to monitor a stereo output. The volume control is located on the front panel. This allows easy level matching with other audio equipment.

Drivers

- A 3/4˝ metal dome tweeter is loaded by a DCW, and is used to reproduce the high frequencies. The DCW is integrated into the one piece cabinet front baffle.
- The 5˝ woofer is a bass cone driver mounted in a 4.5 litre vented cabinet. The -3 dB frequency is 68 Hz and the LF response extends down to 65 Hz (-6dB).
- Protective grills are placed in front of both drivers, and both are magnetically shielded, allowing use near video monitors.

Mounting Options

- On the base of the monitor is a 3/8˝ UNC threaded hole which can accommodate a standard microphone stand.
- There is a provision for an Omni-mount size 50 bracket, for which two M6x10mm screws are required.
- Alternatively the speaker can be hung on M4 screws with suitable heads by one of the three keyhole slots on the backpanel.
- The speaker can be hung in a horizontal or vertical position. Friction pads are provided for placement on a shelf or a stand.

B&H PHOTO - VIDEO - PRO AUDIO

1030A

6½″ Bi-amplified Active Monitor



The Genelec 1030A is a very compact bi-amplified active monitor system, which has performance comparable to much larger systems. The vented speaker enclosure has an amplifier unit set into the back. This unit contains an active electronic crossover, over-load protection circuitry and two power amplifiers: one for each driver. The system's excellent dispersion and precise imaging together with its compact size make it ideal for near field monitoring, broadcast and TV control rooms, mobile vans, home studios and travelling engineers. Genelec's unique Directivity Control Waveguide (DCW) technology is used to provide excellent stereo imaging and frequency balance, even in difficult acoustic environ-ments and the versatile crossover controls allow further matching of the system to its surroundings. A pair of 1030As can produce peak acoustic levels of over 115 dB SPL at 1m. The speakers may be used in vertical or horizontal orientation.

Same features as the 1029APM EXCEPT—

- Bass and treble amps produce 80 and 50 watts of power each, with low THD and IM distortion values
- 3 /4″ metal dome driver, loaded by DCW, is used to reproduce the high frequencies
- The bass driver is a high efficiency 6-1/2″ polymer composite cone driver in a 6.5 liter vented cabinet. The -3 dB frequency is 52Hz and the low frequency response extends down to 47 Hz. (-6dB)
- Like the 1029AMP there are three special calibrated controls in the active crossover network. However, in the 1030A the treble and bass 'roll-off' switches are adjustable in 2dB steps:

1029AMP

–Treble tilt control operating range from 0 to -2dB @15 kHz

–Bass roll-off control operating in a -6dB step @85 kHz

–Bass tilt control operating range in 2dB steps: from 0 to -6dB @150Hz

1030A

–Treble tilt control operating range in 2dB steps: from +2 to -4dB and MUTE

–Bass roll-off control operating range in 2dB steps: from 0 to -8dB @50 Hz

–Bass tilt control operating range in 2dB steps: from 0 to -6dB @100Hz and MUTE

Accessories:

Vertical Wall Mount Kit.....**259.95**
Vertical Floor Stand Kit **364.95**
Green Carry Bag for 1030 pair **169.95**
Green Carry Bag for 1029 or 2029 pair....**84.95**

DCW (Directivity Control Waveguide) Technology

Incorporated in the 1029A and 1030 monitors, Directivity Control Waveguide (DCW) technology is a means of improving the performance of a direct radiating multi-way loudspeaker under normal listening conditions. One of the basic aims is to match the performance of the drivers in terms of both frequency response and directivity. This results in a smoother overall frequency response on and off axis. In addition, the improved directivity control causes more direct sound and less reflected sound to be received at the listening position. This provides improved stereo imaging and ensures that the system is less sensitive to differing control room acoustics than conventional direct radiator design. DCW Technology improves the drive unit sensitivity by +2 to +6 dB (depending on the particular application), thus also increasing the available system maximum sound pressure level.

	1029AMP	1030A
SPL		
Short term RMS at 1m	100dB SPL	105 dB SPL
Peak with music @1m	110 dB SPL	110 dB SPL
Drivers		
Bass	5″	6½″
Treble	3/4″ metal dome + DCW	
Crossover Frequencies	3.3 kHz	3.5 kHz
Frequency Response	70-18k Hz	55-18k Hz
Amplifier Power (1-ch)		
Bass	40 W	80 W
Treble	40 W	50 W
Dimensions (HWD)	9¾ x 5 15/16 x 7½″	12¼ x 7⅞ x 9½″
Weight	13.2 lb.	17 lb.

GENELEC

2029A/2029B

Digital Monitoring Speakers



The 2029A and 2029B are near field monitors that combine digital audio and analog audio inputs in a single speaker system. They are ideal for working with a digital audio workstation or if you are processing audio in a modern studio where your signal is digital. The 2029A/B support all the same modes of operation as the analog 1029A. You can use them with a subwoofer. You can use them in surround audio systems. Due to their compact size, integrated construction, excellent dispersion and precise stereo imaging, they are ideal for near field monitoring, mobile vans, digital audio workstations, broadcast and TV control rooms, surround sound systems, home studios, multimedia applications and also for use with computer sound-cards. They also incorporate Genelec's Directivity Control Waveguide (DCW) technology for excellent frequency balance even in difficult acoustic environments.

FEATURES

Digital Interface

- The 2029A is precision-aligned and balanced—from the single stereo 24-bit S/PDIF digital input—to the highly efficient 110dB/SPL matched drivers. The 2029A is 48kHz compatible. The 2029B offers the same precision alignment with an AES/EBU digital interface on a digital XLR-type input. The 2029B is 9kHz compatible. Both monitors incorporate extremely linear, integrated D-to-A converter circuitry to offers a precision-matched electrical interface to the active electronics and amplifiers.

Drivers

- The bass frequencies are reproduced by a 130 mm (5″) bass driver mounted in a 4.5 litre vented cabinet. The -3 dB point lies at 68 Hz and the frequency response extends down to 65 Hz (-6 dB).
- The high frequency driver is a 3/4″ metal dome. Uniform dispersion control is achieved with the revolutionary DCW Technology pioneered by Genelec.
- Magnetic shielding is standard on the 2029A/2029B. Shielding is vital for applications such as video post production, where stray magnetic fields must be minimized.

Digital Audio

- The quality of a digital audio signal is defined by two parameters: word length and sampling rate. The word length defines how precisely the audio signal is represented. Longer word length leads to smaller noise and distortion level. The typical word length in CD records is 16-bits. Studio recording systems use word lengths of 20-bits and above. The sampling rate determines what frequencies can be represented in the digital audio signal. A higher sampling rate allows higher frequencies to be recorded.
- Turning the digital presentation to an analog signal using a DA converter involves significant sources of error. Your digital-to-analog converter may have inferior performance. It may be misaligned with your amplifiers. The interface between the converter and the amplifier may distort the signal or it may change the frequency balance. Your monitoring volume level may need to be adjusted in the digital domain instead of analog. Genelec 2029A Digital allows you to solve all of these problems. The alignment of the whole system from the digital input connector is carefully balanced, to make sure that you hear the whole digital truth, and nothing but the truth. All you have to do is to supply the digital signal, and adjust for the volume you desire.

Integrated Construction

- As the digital interface and amplifiers are built into the speaker enclosure, the only connections required are the mains supply and the digital input signal, making them very easy to set up and use.

Crossover

- Their active crossover network is acoustically complementary and the slopes are 24 - 32 dB/octave. The crossover frequency is 3.3 kHz. The room response controls ('treble tilt', 'bass tilt' and 'bass roll-off') allow exact match to any installation.

Amplifiers

- The amplifier unit is built inside the speaker enclosure. The bass and treble amplifiers both produce 40 W of output power. The fast, low distortion amplifiers are capable of driving a stereo pair to peak output sound pressure levels in excess of 110 dB at 1 m. The unit incorporates special circuitry for driver overload protection.

Tone Controls

- The response of the system usually has to be adjusted to match the acoustic environment. The adjustment is done by setting the tone control switches on the rear panel. The tone control has four switches and can adjust 'treble tilt', 'bass tilt' and 'bass roll-off.' The factory settings for these are 'ALL OFF' to give a flat anechoic response.

SUBWOOFERS

	2029A	2029B
Input Word Length	16 - 24 bits	16 - 24 bits
Input Format	IEC958, S/P-DIF, EIJAC CP-340	AES/EBU
Digital Input	RCA 75Ω	XLR 110Ω
Input Sampling Rate		
No De-emphasis	25 - 55 kHz	29 - 100 kHz
Using de-emphasis	44.1 and 48 kHz	44.1 kHz
SPL		
Short term RMS @1m	100 dB SPL	100 dB SPL
Peak per pair with music material	110 dB SPL @ 1m	110 dB SPL @ 1m
Drivers		
Bass	5″	5″
Treble	3/4″ metal dome + DCW	
Crossover Freq.	3.3 kHz	3.3 kHz
Free Field Freq. Response ± 2.5 dB	70 - 18k Hz	70 - 18k Hz
Amplifier power /ch		
Bass	40 W	40 W
Treble	40 W	40 W
Dimensions (H x W x D)	9¾ x 5¹⁵⁄₁₆ x 7½″	9¾ x 5¹⁵⁄₁₆ x 7½″
Weight	13.2 lb.	13.2 lb.

Analog and Digital Cabling

Special interconnect cable connects between the XLR connectors of the two 2029A Digital monitors. The IEC958 interface cable from your digital audio source connects to one of the 2029A Digital units. In the digital audio reproduction mode the output level for both speakers pair is controlled with the right unit. The balance is automatically calibrated correctly with the proposed output level setting on the left unit.

Mounting Options

There are several possibilities for mounting the 2029A/2029B. On the base of the monitor is a 3/8″ UNC threaded hole which can accommodate a standard microphone stand. There is a provision for an Omnimount size 50 bracket, for which two M6x10mm screws are required. Alternatively the speakers can be hung on M4 screws with suitable heads by one of the three keyhole slots on the back-panel. They can be hung in a horizontal or vertical position. Friction pads are provided for placement on a shelf or a stand.

7050A *8″ Active Dual-Input Subwoofer*



The 7050A is a very compact subwoofer incorporating all the amplifier and crossover electronics needed to combine it with a Genelec 1029 or 2029 series speaker. Adding the 7050A to a 1029/2929 system creates a compact nearfield monitoring system capable of a flat frequency response from 38 Hz to 20 kHz (± 2.5 dB).

- Summed, balanced inputs on two XLR connectors
- Sensitivity can be attenuated by from 0dB to -18 dB for easy level matching with the main speakers for varying positions.
- Produces 70 W of output power, with very low THD and IM distortion.
- 8″ magnetically shielded long throw cone driver is capable of producing SPLs up to 105 dB in half space.
- To make the system immune to overloads and spurious signals, the circuitry includes driver overload protection and power-on signal muting. Also incorporates thermal overload and short circuit protection.
- Like the larger 7000 series subwoofers, the 7050A features Genelec's Laminar Spiral Enclosure (LSE) cabinet construction resulting in a robust and reliable system. The handsome cast-aluminum grille protects the magnetically shielded 8″ driver.

7060A/7070A *10″ and 12″ Active Multi-Channel Subwoofers*

The 7060A and 7070A are powerful and precise bass monitoring tools for today's 5.1 or 6.1-channel surround sound or traditional stereo systems. With their 19Hz (7070A) or 29Hz (7060A) lower cutoff frequency, high sound pressure output capability and versatile bass management systems, they can be adapted to all low frequency monitoring situations.

- The built-in bass management unit has six signal input and output channels (L/C/R Front and L/C/R Rear), LFE input and summed signal output connectors, providing great flexibility and easy connection.
- The discrete LFE signal input is equipped with a selectable 85/120 Hz low-pass filter and a 0/+10 dB LFE sensitivity switch.
- Adjustable sensitivity, bass roll-off and phase matching controls let you tailor the response of the subs to the environment.
- Laminar Spiral Enclosure (LSE) cabinet construction ensures reliability
- Integrated 85 Hz test tone generator for accurate crossover phase alignment.

	7050A	7060A	7070A
SPL (Short Term RMS @1m)	100dB SPL	108dB SPL	112dB SPL
Input Channels	2 (for stereo only)	6	6
Sum Out Outputs	0	1	1
Driver	8″	10″	12″
Bass Level	0...-18 dB (variable)	+12 to -6 dB (variable)	+12 to -6 dB (variable)
Bass Roll-Off	n/a	0 to -6 dB (2 dB/step)	0 to -6 dB (2 dB/step)
Phase Adjustment	n/a	0 to 270° (90 °/step) @85 Hz	
Frequency Response	38 Hz - 85 Hz ±3 dB	29 Hz - 85 Hz	19 - 85 (120) Hz ±3 dB
Amplifier Power (1-ch.)	70 W	120 W	250 W
Dimensions (HWD)	410 x 350 x 314 mm	527 x 462 x 360 mm	625 x 555 x 490 mm
Weight	33 lb.	57 lb.	110 lb.

HAFLER

TRM6.1/TRM8.1

6- and 8-inch 2-way Professional Active Monitors



You have one goal when choosing a reference monitor. Music. Accurate music. Nothing else matters. Hafler has the same goal. They don't design for price points. They design for application and performance. However, Hafler did challenge their best engineers to design the finest sounding active near-field monitor available at an affordable price. The result was the TRM6.1 Trans•ana and TRM8.1 Trans•nova Reference Monitors – heralded by some as the finest powered monitors available.

Every component is designed and matched to deliver the most accurate music possible. They provide honest, consistent sound from top to bottom with sonic clarity usually found in much more expensive speakers. They feature built-in amplification, active crossovers, and Hafler's patented power amp circuitry – known for its accurate soundfield, in width, height and depth.

TRM6

Following in the footsteps of the TRM8, the TRM6 incorporates Trans•ana amplifier technology at an even more affordable price. Whether tracking, mixing, or mastering, the TRM6 delivers the consistency and accuracy that are a must in today's competitive environment.

- Discrete Hafler Class G Trans•ana mosfet amplifiers (55 watt for the woofer, 33 watt for the tweeter
- Balanced XLR, unbalanced RCA inputs

They Both Feature

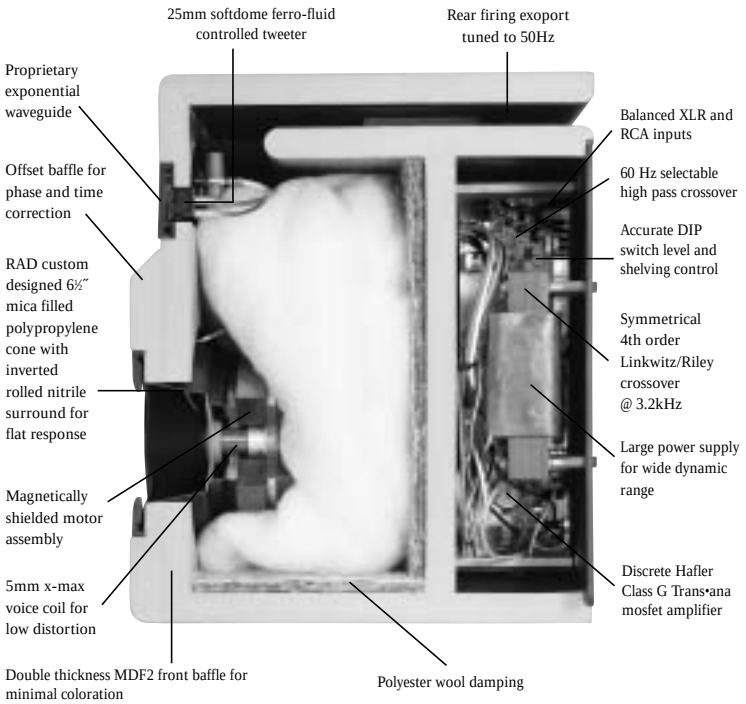
- Patented Trans•ana (TRM-6) or Trans•nova (TRM-8) output stage configured for "power gain" allows simpler front end circuitry, dramatically lower cross talk and noise
- 1″ softdome ferro-fluid controlled tweeter for more open, natural sounding high frequencies.
- MOSFET outputs deliver higher output current, rugged reliability, superb sound quality
- Symmetrical 4th order Linkwitz/Riley crossover
- +/-4dB Bass and Treble shelving adjustment to match the environment
- Shielded woofer allows use near computer monitors
- Wide dynamic range at all levels (no compression)
- 3-Year Warranty

TRM8

- DIAMOND driver stage provides up to 14dB additional dynamic range in amplifier front end.
- Balanced XLR or 1/4″, unbalanced RCA inputs
- Pre-drilled for Omni Mount

THE REFERENCE

Hafler designs monitors for application and performance. Every component of their TRM Series monitors is designed and matched to deliver the most accurate music possible. To Hafler's engineers, nothing else matters.



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ONE HOUR FREE PARKING
AT 349 W. 34th STREET (with purchase of $100 or more)

TRM10.1/TRM12.1

M5 MONITOR

The M5 monitor is yet another example of Hafler's brilliant loudspeaker design and manufacturing capabilities. The M-5 is an unparalleled solution for broadcast and project studio environments. Portable and lightweight, it offers all the qualities of the TRM6 in a more compact, non-amplified package.

- 70Hz- 21kHz frequency response ±3dB
- TRM6 tweeter/waveguide for smooth response on and off axis
- Control of woofer at high output levels
- Shielded woofer magnet allows use near computer monitors
- Tweeter level control is user selectable from the front panel (3dB L-Pad)
- 4th order Linkwitz-Riley crossover at 3.2kHz
- Zobels, tweeter overload protection
- 5-way binding posts
- 1 Year Warranty

Downfiring Trans•ana Class-G Active 10- and 12˝ Subwoofers

A perfect match for the TRM6.1, the TRM10.1 Subwoofer delivers a lot of punch from a small package, and offers unusual flexibility and adjustability at a great value. Extends your system's frequency range to well below 30Hz. However, when your program material demands ultra-low distortion energy at rock bottom low frequencies the TRM12.1 is a powerhouse, delivering true full range monitoring.



- ◆ Patented Trans•ana output stage configured for "power gain". Allows simple front end circuitry, dramatically lower cross talk and noise.
- ◆ Adjustable symmetrical 4th order Linkwitz-Riley crossover from 40 to 110Hz
- ◆ CLASS G: Provides higher efficiency amplification at all listening levels
- ◆ Balanced XLR, unbalanced RCA inputs
- ◆ Adjustable input level sensitivity
- ◆ 3-year warranty

	TRM6.1	TRM8.1	TRM10.1	TRM12.1
Frequency Response	55Hz-21kHz ±2dB	45Hz-21kHz ± 2dB	30Hz-110z ±2dB	25Hz-110z ±2dB
Peak Acoustic Output (per pair @ 1m)	>118dB w/music	>123dB w/music	>118dB w/music	>121dB w/music
High Frequency Driver	1˝ (25mm) soft dome	1˝ (25mm) soft dome		
Low Frequency Driver	6˝ polypropylene cone	8˝ polypropylene cone	10˝ Cellulose Fibre cone	12˝ Cellulose Fibre cone
Amp Power Rating FTC	33 watts @ 6Ω (high frequency) 50 watts @ 4Ω (low frequency)	75 watts @ 6Ω (high frequency) 150 watts @ 4Ω (low frequency)	200 watts @ 4 Ohms; Trans•ana Class-G	200 watts @ 4 Ohms; Trans•ana Class-G
Signal-to-Noise	>100dB	>100dB	>100dB	>100dB
Slew Rate	100V/µs	100V/µs	100V/µs	100V/µs
CMRR	70dB typical @ 1kHz	70dB typical @ 1kHz	70dB typical @ 1kHz	70dB typical @ 1kHz
Input Impedance		47kOhms per phase balanced		
Input Sensitivity Range				
Unbalanced:	500mV to 3V	500mV to 3V	500mV to 3V	500mV to 3V
Per Phase Balanced:	275mV to 1.5V	275mV to 1.5V	275mV to 1.5V	275mV to 1.5V
Filter Type	24dB/octave Linkwitz-Riley @ 3.2kHz	24dB/octave Linkwitz-Riley @ 2.5kHz		
Power Consumption (Full Power)	405W /4.75A @ 120 VAC	405W /4.75A @ 120 VAC	405W /4.75A @ 120 VAC	405W /4.75A @ 120 VAC
Subsonic Filter	30Hz @ 12dB/octave	30Hz @ 12dB/octave		
Bass Shelving	48Hz to 200Hz, ± 4dB (-4, -2, 0, +2, +4dB)	40Hz to 200Hz, ± 4dB (-4, -2, 0, +2, +4dB)		
Treble Shelving	3.5kHz to 20kHz, ± 4dB (-4, -2, 0, +2, +4dB)	3kHz to 20kHz, ± 4dB (-4, -2, 0, +2, +4dB)		
Dimensions (WHD)	8.875 x 13.25 x 11.5˝	10.25 x 15.875 x 13˝	16 x 14.75 x 16˝	19.5 x 18 19.25˝
Net Weight	25 lbs.	35 lbs.	60 lbs.	96 lbs.

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CIRCLE 3 / CIRCLE 3 ACTIVE

Passive and Active 4½˝ Nearfield Monitors



The Circle 3 (passive version) and Circle 3 Active (powered version) are compact monitor speakers designed for nearfield use, or in critical monitoring applications where space is limited. The Circle 3 uses a highly linear, custom-designed 4.5˝ bass driver, built on an acoustically dead chassis to eliminate undesirable resonant effects. High frequencies are reproduced by a ferro-fluid cooled soft dome tweeter. And despite standing just over 10˝ high, the Circle 3 delivers a robust, full range sound with a surprisingly extended low end response.

The Circle 3 Active adds a bi-amp module delivering 60w of power to the bass driver and 30w to the tweeter. And unlike most compact active monitor designs, the Circle 3 Active's LF amplifier is not limited, as this unbalances the sound at high listening levels. Rather, the bass drivers are engineered to handle the full amp output. And controlled order crossover management makes the Circle 3's particularly easy to listen to, even on long sessions. Magnetic shielding on both versions, makes them ideal for use in any application where close proximity to a CRT monitor is required.

CIRCLE 1

12˝ Subwoofer



Much more than just a subwoofer, the Circle 1 is a complete speaker management system, forming the heart of a high performance, professional surround sound monitoring system. In addition to a 12˝ low moving mass bass driver and 160-watt amp module, the Circle 1 also features 5 channel active filtering with a mode switch making it easy to configure the system for a variety of stereo and surround monitoring applications.

- Low frequency boost control is provided to increase the level of the bass component in low level monitoring applications
- Phase control can be used to achieve optimum sub bass performance in differing room acoustics.
- XLR connectors on all inputs and outputs
- Design of the cabinet porting assists in producing a frequency response that extends down to 33Hz.
- Mode switch for different surround decoders and stereo setups

Connection to the Circle 1 is via six balanced XLR inputs and five XLR outputs. Although the Circle 1 is designed primarily for use in a surround sound system, there are many people who would like to increase the bass frequencies or overall level available from their speakers. So when two or more speakers are connected to the input channels, the Circle 1 sums the signals from the front left and right speakers and filters out their bass frequencies for reproduction from the sub-woofer, leaving the satellite channels to reproduce the mid and high-frequencies.

A switch on the back of the Circle 1 sets it for 5.1 surround sound operation where, instead of reproducing the filtered frequencies from the satellite speakers it is driven from a dedicated sub input. Additional rotary controls are provided for input sensitivity, LF Boost and Phase. With its outputs connected to five active Circle 5's, the result is a powerful, detailed and truly universal 5.1 monitoring system.



CIRCLE 5 / CIRCLE 5 ACTIVE

Passive and Active 8˝ Midfield Monitors



Accurate and revealing monitors ideal for use in midfield stereo or surround sound monitoring applications, both the passive (Circle 5) and powered (Circle 5 Active) share a custom-designed 8˝ bass driver that is injection-molded from a specially formulated plastic compound. The rolled sheets used by other manufacturers retain a grain pattern in the molded cone, introducing unwanted compressions both with and across the grain and ultimately degrading performance. The random grain pattern in the HHB cone suffers none of these problems and resonances are minimized yet further by varying the thickness of the cone across its surface.

Like the Circle 3, the Circle 5 uses controlled order crossover management to achieve an accurate sound that never tires the listener. And like the Circle 3 Active, there is no amp limiting so the full bass component of sound is present at all volume levels.

Circle 3

- Custom 4.5˝ (110mm) bass driver with acoustically dead chassis reduces any unwanted resonant effects, producing a perfectly damped impulse response. This results in both the active and passive speakers being incredibly easy to listen to, even over long periods of time.
- Ferro-fluid cooled soft dome tweeter
- Active biamp module: Bass (60 watts), Tweeter (30 watts)
- Circle 3 uses binding posts for connections, the Circle 3 Active uses XLR and RCA connectors
- No limiting on LF amplifier so the full bass component of sound is present at all volume levels—sound is always balanced
- Controlled order crossover management controls the rate at which the LF ramps down and the HF ramps up at the crossover point, resulting in an accurate, untiring sound —even on long sessions.
- Magnetically screened for use near video and computer monitors

Circle 5

- Custom 8˝ (200mm) bass driver with injection molded cone
- Ferro-fluid cooled soft dome tweeter with Polyswitch protection. Rather than attenuating HF power and unbalancing the sound in the event of overload, Polyswitch cuts the power completely until the tweeter has cooled, illuminating a warning LED in the process.
- Active biamp module: Bass (140 watts), Tweeter (70 watts)
- Circle 5 uses binding posts for connections, the Circle 5 Active uses XLR and RCA connectors
- No limiting on LF amplifier so the full bass component of sound is present at all volume levels—sound is always balanced
- Controlled order crossover management controls the rate at which the LF ramps down and the HF ramps up at the crossover point, resulting in an accurate, untiring sound —even on long sessions.
- Anti-magnet cancels magnetic fields that can distort video monitors

	Circle 3 Passive	Circle 3 Active	Circle 5 Passive	Circle 3 Active
Transducer System	2 way monitor loudspeaker	2 way monitor loudspeaker	Ported 2 way monitor loudspeaker	Ported 2 way monitor loudspeaker
Drive Units LF:	Custom 4.5˝ NRSC cone, 1˝ voice coil, high linearity low-loss suspension polymer chassis and shielding antimagnet		Custom 8˝ polymer cone, high compliance rigid self-sealing long throw rubber surround. 1˝ high temperature aluminum voice coil and antimagnet.	
Drive Units HF:	1˝ ferro-fluid cooled soft dome, high efficiency Neodymium magnet, highly focused magnetic field 1˝ aluminum voice coil		Custom 1.1˝ ferro-fluid cooled soft dome with rear damping chamber. 1.1˝ high temperature aluminum voice coil and antimagnet.	
Frequency Response	70Hz - 20kHz ±3dB	70Hz - 20kHz ±3dB	48Hz - 20kHz, +/- 3dB	48Hz - 20kHz, +/- 3dB
Sensitivity	83dB 1w/1m.	1V RMS	87dB 1W/1m	1V RMS
Suggested Amplifier Power	30 - 100w	30 - 100w	50 - 200w	50 - 200w
Impedance	8Ω nominal	8Ω nominal	8Ω nominal	8Ω nominal
Power Handling (Program)	60w	60w bass, 30w tweeter	120w	140w bass, 70w tweeter
Crossover	3.5kHz	3.5kHz	2.6kHz	2.6kHz
Dimensions (HWD)	10.6 x 6.9 x 7.8˝	10.6 x 6.9 x 8.7˝	16.5 x 10 x 11.8˝	16.5 x 10 x 11.8˝
Weight (each)	8.1 lbs.	11 lbs.	22 lbs.	27.7 lbs.
Finish	Durable black paint	Durable black paint	Durable black paint	Durable black paint

CONTROL SERIES

The Control Series combine high performance and smooth, powerful, wide range response with a compact design and rugged construction for unprecedented versatility. Also one of the easiest systems to install, the Control Series are perfect for applications where tight corners and tough angles are all too common. A wide variety of available mounting systems allows positioning for optimum performance.



Control 5

Control 1 (black finish)

Control 1 (white finish)

Control 1

The Control 1 is a 150-watt, two-way miniature monitor with well balanced sound and exceptional power handling. Ideal for any installation requiring professional control monitor performance from a compact source including recording studios, mobile audio video control rooms and broadcast studios. It is also highly suitable for foreground and background music use in restaurants, discos, and A/V applications.

- Incorporating a 5¼″ low frequency loudspeaker, 3/4″ high frequency radiator and high performance dividing network, the Control 1 provides full-range, low distortion reproduction in a variety of applications
- High and low-frequency transducers are magnetically shielded, allowing use near video monitors
- Mounting versatility is enhanced by a complete line of installation accessories

Control 5

The Control 5 is a high-performance 175 watt, two-way wide range control monitor suitable for use as the primary sound source in a variety of applications. Smooth, extended frequency response combines with wide dynamic capabilities to provide acoustic performance that makes the Control 5 ideal for recording studios, audio-video control rooms, remote trucks and broadcast studios. Clean, functional visual design also makes ideal for foreground music systems, moderate level sound reinforcement, discos and music playback systems.

- 6½″ low frequency driver provides solid, powerful bass response to 50 Hz
- High frequency response to 20 kHz is handled by a pure titanium 1″ dome
- HF (high frequency) and LF (low-frequency transducers are magnetically shielded, permitting use in close proximity to video monitors
- Molded of dense polypropylene foam, the Control 5 enclosure offers a pleasing shape that fits easily into any environment
- Wall, ceiling, rack and tripod mounting systems allow positioning in exactly the right spot for optimum performance
- Available in black, gray or white finish
- The dividing network incorporates protection circuitry to prevent system damage and utilizes high quality components including bypass capacitors for outstanding transient accuracy

	CONTROL 1	CONTROL 5
Frequency Response	120Hz - 20 kHz (±3db)	75Hz - 20 kHz (±3db)
Power Capacity[1]	150w	175w
Sensitivity: 1 W, 1 m	87dB SPL	89dB SPL
Directivity Factor (Q)	2.8	2.8
Directivity Index (DI)	4.5	4.5
Nominal Impedance	4Ω	4Ω
LF Components	5¼″	6½″
HF Components	3/4″	1″
Finish	Black or white	Black or white
Dimensions	9.25 x 6.25 x 5.6″	15.25 x 9.8 x 9″
Net Weight (each)	4 lbs.	10 lbs.

[1] *IEC filtered random noise (50Hz-5kHz) with a crest factor (peak to average ratio) of 6dB*

MOUNTING ACCESSORIES



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STUDIO MONITORS

4200 SERIES



The standard for over a decade, the 4206 (6.5˝) and 4208 (8˝) are console-top monitors designed specifically for near field use in multi-purpose studio environments. Starting with the unique Multi-Radial sculptured baffle, the monitors direct the axial output of the individual components for optimum summing at the most common listening distance (approx. 3 to 5´). To deliver maximum stereo imaging while reducing the potential for listener fatigue, the pure titanium dome tweeter and cone transducer are aligned to deliver both high and low-frequency information to the listening position at precisely the same instant. They monitors also deliver smooth, extended bass output with little distortion and power compression. The 4206 and 4208 are rated at 8 Ω and are magnetically shielded.

- Multi-Radial baffle positions the drivers to achieve alignment of their acoustic centers so that low, mid and high frequency information reach the ears at the same point in time—resulting in superb imaging and greatly reduced phase distortion.
- 1˝ titanium HF transducers deliver smooth, extended response. The tweeter features a uniquely shaped "lens" which acts as a mechanical filter to balance energy output to within ±2 dB to 20 kHz.
- The long linear excursion design of the low frequency transducers results in smooth extended bass output with less distortion and power compression.
- Smooth transition from low to high frequency components is achieved through carefully engineered high complexity crossover networks. Tight tolerances are maintained to achieve seamless summing and minimal distortion.

4400 SERIES

For years recording, broadcast, movie and TV studios worldwide have depended on the 4400 Series as their critical listening source. Countless music recordings and movie soundtracks were created with 4400 Series monitors, while broadcast studios rely on them to deliver faithful sound reproduction, typically 24 hours a day, seven days a week. Available in two-way and three-way systems, they all utilize JBL transducer technology with SFG magnet structures, large diameter voice coils and titanium dome tweeter.



Low frequency drivers employ Symmetrical Field Geometry (SFG) magnet structures to minimize harmonic distortion, resulting in maximum definition of bass and low midrange information. Large voice coils provide excellent transient response characteristics and outstanding power handling.

Titanium dome tweeter further minimizes distortion levels, delivering even smoother, clearer high frequency information. And the tweeters are oriented to the create "Left" and "Right" models, achieving mirror-imaged pairs for excellent imaging.

With a tight crossover network design, the 4400 Series provides absolutely smooth transitions between transducers for perfect imaging and unparalleled power response.

4408A 2-way 8˝ Studio Monitor

Two-way compact monitor system ideal for smaller recording studios or for broadcast studio monitor control rooms.

4410A 3-way 10˝ Studio Monitor

A three-way monitor loudspeaker system designed as a vertical line array. This system delivers incredibly fine transient response characteristics and spatial detail.

4412A 3-way 12˝ Studio Monitor

For applications requiring maximum low frequency output from a bookshelf-sized monitor, this 3-way system tightly clusters its transducer complement for accurate close proximity listening.

LSR SERIES

THX-Approved Studio Monitors

The Linear Spatial Reference (LSR) philosophy is based on a set of design goals that carefully control the overall performance of the system ina variety of acoustic spaces. Instead of focusing on a simple measure such as on axis frequency response, LSR designs require much better control over dispersion via transducer design and crossover frequency selection. Critical decisions of image placement, EQ, balance and timbre are typically made within ±15° vertically and ±30° horizontally. This workspace is where the engineer, producer and artist make critical mixing decisions. By incorporating LSR into the system design requirements, placement rules are relaxed, a more stable stereo image is maintained and off axis coloration is minimized.

LSR25P Compact Bi-Amplified Monitor



The LSR25P is sized to provide an extremely accurate reference for workstations, edit suites and small control rooms.

- Combines 100 watts of low frequency power and 50 watts for the high frequency with an active crossover system
- 5.25″ woofer incorporates Symmetrical Field Geometry (SFG) with integral shorting ring for maximum linearity.
- Unique Workstation Boundary Compensation control adjusts the bass and mid-bass output for optimum performance. Precisely maintains correct spatial response when used next to video monitors in typical workstation environments.
- User adjustments include level calibration and fine tuning of levels with a variable level control. XLR and RCA inputs accommodate balanced and unbalanced signals.
- Frequency response can be tailored to compensate for placement and room absorption characteristics. Select between two high frequency response contours. An 80Hz low frequency filter can be added for use with subwoofers that have their own low pass filtering.
- Integrated mounting points for horizontal or vertical suspension. Shielded for use near video monitors.

LSR28P Bi-Amplified Monitor



- Combines 250 watts of low frequency power and 120 watts for the high frequency. with an active crossover system
- 8″ woofer is based on JBL's patented Differential Drive technology. With dual drive coils, power compression is kept to a minimum to reduce the fatigue of spectral shift as power levels increase. An added third coil between the drive coils acts as a dynamic brake to limit excess excursion and reduce audible distortion at the highest levels.
- Adjustments include level calibration and fine tuning of levels with a variable level control. XLR and RCA inputs accommodate balanced and unbalanced signals. The frequency response of bass and treble amplitudes can also be adjusted to compensate for placement and room absorption characteristics.

LSR32 3-Way Studio Monitor



- 12″ woofer based on JBL's patented Differential Drive technology
- The midrange is a 2″ neodymium magnet structure with a woven 5″ Kevlar cone. The crossover points are located at 250 Hz and 2.2 kHz. to match the directivity characteristics of the three transducers.
- The high frequency transducer of all three monitors is 1″ composite diaphragm integrated with an Elliptical Oblate Spheroidal (EOS) Waveguide with 60 x 100° dispersion which is critical to the smooth spatial response required in today's working environments.

LSR Measurement Techniques

LSR is the underlying design philosophy that explains why speakers that measure the same, sound different. By going beyond simple on-axis frequency measurement, LSR techniques define the ultimate performance specifications of JBL's monitoring technology—what it will sound like in your room. They go beyond the performance of an on-axis frequency response at one point in space, which other manufacturers use.

LSR uses a technique of measuring a monitor over a sphere that encompasses all energy radiated into the listening room in every direction. This data reflects 1296 times the information of a single on-axis frequency response curve. Using psychoacoustic principles allows the calculation and optimization of the entire sound field heard by the listener - this includes the direct sound field, the reflected sound field and the reverberant sound field. In place of spectral smoothing, which actually conceals data, LSR techniques expose flaws in systems such as resonances, poor dispersion and other off-axis colorations.

LSR12P Powered Subwoofer



The LSR12P is the ideal companion to the LSR25P, LSR28P and LSR32. It is multi-channel compatible for AC-3, DTS and other surround sound formats. It features a 12″ Neodymium woofer based on JBL's patented Differential Drive technology. The cone is made of carbon fiber composite for low cabinet resonance and stable inertial ground. The integrated 250-watt power amp provides correct drive levels for optimized frequency response. Bass management for multiple formats is provided with separate left, center and right crossover functions as well as a discrete input.

- Subwoofer signals are fed by the Left, Center, Right and Discrete inputs. This unique arrangement provides excellent bass management for a wide variety of formats. The subwoofer signals can be derived from the front channels or from the discrete input with a simple remote switch contact. Alternatively, the subwoofer can be muted with full range signals routed to the front channels for comparison with or without subwoofer support. Calibrated input levels for -10 dBV and +4 dBu inputs are included.
- In addition, an input attenuator can be inserted into the circuit for continuously variable level control to fine tune subwoofer level matching. Subwoofer low pass filtering uses steep 5th order low pass filtering to minimize the possibility of localization of the subwoofer acoustic output. High pass filtering for the Left, Center and Right outputs are at 12 dB/Octave.
- There are six dip switches that allow you to optimize the LSR12P's performance. The first three switch settings adjust the calibration level of the Left, Center and Right inputs. Switch four inverts the signal polarity of the subwoofer feed. This is used to correct for placement mismatch between the subwoofer and satellite spacing. Switches five and six adjust the low frequency spectrum to compensate for boundary effects when a subwoofer is placed against walls or corners.
- A 1/4″ jack is included on the back panel which allows remote control bypass of the subwoofer and selection of the discrete input. Shorting the Tip and Sleeve of the jack will remove the high pass filtering from the Left, Center and Right outputs and sources the subwoofer feed from the discrete input.

4200 and 4400 SERIES SPECIFICATIONS

	4206	4208	4408A	4410A	4412A
Frequency Response:	65Hz-20kHz (±2dB)	65Hz-20kHz (±2dB)	50 Hz-20 kHz (±2dB)	45Hz-20kHz (±2dB)	45Hz-20kHz (±2dB)
Frequency Range:	42 Hz - 21kHz (-10dB)	38 Hz - 21kHz (-10dB)	35 Hz - 30 kHz (-10dB)	33 Hz - 30kHz (-10dB)	30 Hz - 30kHz (-10dB)
Sensitivity (1w, 1m):	87 dB SPL	89 dB SPL	89 dB SPL	90 dB SPL	89 dB SPL
Nominal Impedance:	8Ω	8Ω	8Ω	8Ω	8Ω
Crossover Frequency:	2.8 kHz	2.6 kHz	2.5 kHz	900Hz, 4.0 kHz	850 Hz, 4.0 kHz
Transducer: LF:	6.5″	8″	8″	10″	12″
MF:	—	—	—	5″ cone	5″ cone
HF:	1″	1″	1″	1″	1″
Net Weight:	15 lb.	9.3 lb.	26 lb.	43 lbs.	47 lbs.
Dimensions (HxWxD):	15¾ x 9 x 9½″	15¾ x 9 x 11″	17¼ x 12 x 11⅝″	23½ x 14¼ x 11¼″	14¼ x 23½ x 11.76″

LSR SERIES SPECIFICATIONS

	LSR25P	LSR28P	LSR32	LSR12P
Frequency Response (-6dB):	56Hz-22kHz	37Hz-22kHz	40Hz-22kHz	25Hz-110kHz
Power:	LF: 100w HF: 50w	LF: 200w HF: 70w	200w IEC	250w
Sensitivity:	+4 dBu /-10dBV	+4 dBu /-10dBV	90 dBu (1 W / 1m)	+4 dBu /-10dBV
Distortion (96dB):	<0.5% (150Hz-20kHz)	<0.5% (120Hz-22kHz)	<0.5% (120Hz-22kHz)	<1.0%
Crossover Frequency:	2.8 kHz	2.6 kHz	2.5 kHz	900Hz, 4.0 kHz
Net Weight:	17 lb.	45 lb.	47 lb.	50 lbs.
Dimensions (HxWxD):	6.8 x 10.6 x 9.5″	16 x 13 x 12.75″	25 x 15.5 x 11.5″	25 x 15.5 x 12″

V-SERIES STUDIO MONITORS

Bi-Amplified Reference Monitors



The V-Series are bi-amplified near-field reference monitors capable of satisfying the most critical listening requirements. Separate built-in power amplifiers for the woofer and tweeter and a true electronic crossover that tailors the power and frequency response specifically for each transducer, combine to deliver the dynamic range and performance required for everything from digital production to 5.1 surround sound. KRK's goal for the V-Series was to create monitors capable of accurately reproducing sound with unsurpassed clarity and accuracy. Other monitors tend to modify, extend, or "tilt" their response above 1 kHz to sound more "spatial, exciting and impressive". What is often overlooked is this practice tends to make you incorrectly balance your mixes to compensate for this effect, leading to poor "translation". The V-Series was designed to reproduce the signal with a minimal of frequency enhancement. Rather than attempting to defy physics, they deliver unaltered high frequencies and respectable low frequency response with exceptional accuracy from their woofer in a reasonably small cabinet.

Ability to Translate -

A mix made on a V-Series monitor will translate to other environments and other monitors with no surprises—saving you time and eliminating extra work. They monitors will perform flawlessly for years in professional and non-professional environments alike.

Balanced Power –

KRK's amplifier construction is proven, durable and straightforward. Output wattage is balanced to optimize LF and HF driver performance. V-Series monitors use a toroidal power transformer for the lowest possible hum and minimum noise artifacts

Flexibility -

They contain an active input circuit specifically designed for balanced or unbalanced input signals. They accept XLR and 1/4″ phone or TRS connectors.

Active Filter Crossovers -

Each monitor cabinet contains three active filters (subsonic, low-pass and high-pass filter). These three filters work together to deliver superior linear frequency response. Active filter crossovers generate less heat resulting in no signal drift or component variations to provide a more stable image.

No Compression -

V-Series monitors contain no signal compression which artificially limits the dynamic range of the program source. Although some monitors use signal compression to make them sound louder, fuller, and to control unwanted characteristics of the raw drivers, the result is a compromised mix. V-Series design is uncomplicated without extras between you and your mix.

Furniture-Grade Construction -

The cabinetry of a V-Series monitor is constructed from 3/4 to 1″ medium density fiberboard (MDF). They are specifically designed to minimize unwanted cabinet resonance. All edges and port openings are heavily radiused to reduce edge diffractions and port flutter. Finish is a neutral fingerprint proof gray Zolatone

Custom Made Drivers -

All except the V6 (uses a specially designed polyvinyl LF driver) use a woven Kevlar LF driver custom-designed for their individual cabinet volume with one goal in mind—minimal compromises, no surprises. Kevlar is one of the strongest, lightest, most rigid materials that can be used in modern speaker cone construction.

V4 ORBITAL PACK

Surround yourself in quality Hi-Fi sound at an easily affordable price. Get a 5.1 linear system, which includes five V4 monitors and one S8 subwoofer, in one complete package. The V4 has separate amplifiers for both the tweeter (15w) and the woofer (30w), which makes the sound extremely clean and virtually eliminates any distortion. The S8 subwoofer is powered with 100 watts, and includes an 8″ Kevlar woofer, variable low-pass filter, phase correction switch and variable input sensitivity.

The V4 Orbital Pack is living proof that good things come in small packages!



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AT 349 W. 34th STREET (with purchase of $100 or more)

V-SERIES STUDIO MONITORS

V4



There's big sound coming out of the V4 mini monitor. In KRK fashion, the V4 has separate amplifiers for both woofer and tweeter, which make it extremely clean and virtually eliminating any distortion. Video shielding is standard, making it an optimum choice for small audio workstations. A Neutrik Combo connector gives it ultimate flexibility for both balanced and unbalanced signals, it also gives you the option of using 3 different connectors (XLR, 1⁄4″ TRS, or 1⁄4″ Instrument). Paired with the S8 subwoofer, it just goes to show that good things do come in small packages!

V6



The exciting V6 offers the best sound per dollar in a small, active near field monitor. Featuring separate amplifiers for the woofer and tweeter, an active crossover, and video shielding as standard, the V6 is the perfect cost-effective active monitor solution for digital audio workstations. Using the Neutrik Combo connector, the V6 can handle both XLR and .25″ phone plug inputs. Teamed up with the new S10 powered subwoofer, five V6's make a great 5.1 surround sound mixdown system.

V8



For the professional who needs an active near field monitor able to satisfy the most critical listening requirements, yet be portable enough to take on location, the new V8 studio reference monitor is the answer. The V8 uses a separate power amplifier for the woofer and tweeter with an electronic crossover so that the power and frequency response is tailored to each transducer. The V8's Kevlar woofer is extremely fast and does not have audio band resonances that color the sound. Using the Neutrik Combo connector, the V8 can handle both XLR and .25″ phone plug inputs, and video shielding is standard.

V88



Want more bass? More SPL? The V88 has two 8″ woofers that will definitely shake up your studio. The V88 is actually bi-amped, with a 160 watt amplifier for the low frequencies and an additional 60 watt amplifier for the tweeter.There is no other monitor with this much output in this form factor. And of course it is magnetically shielded.

	V4	V6	V8	V88
Video Shielding	Standard	Standard	Standard	Standard
Drivers	HF 1″ Fabric Dome LF 4″ Woven Kevlar	HF 1″ Silk Dome LF 6″ Polyvinyl	HF 1″ Silk Dome LF 8″ Woven Kevlar	HF 1¼″ Polymer Dome LF2 x 8″ Woven Kevlar
Input	XLR / 1/4″ TRS Combo, 10k ohm Balanced; Pin 1+Sleeve=Ground, Pin 2 + Tip = High, Pin 3 + Ring = Low	XLR / 1/4″ TRS Combo, 10k ohm Balanced; Pin 1+Sleeve=Ground, Pin 2 + Tip = High, Pin 3 + Ring = Low	XLR / 1/4″ TRS Combo, 10k ohm Balanced; Pin 1+Sleeve=Ground, Pin 2 + Tip = High, Pin 3 + Ring = Low	XLR / 1/4″ TRS Combo, 10k ohm Balanced; Pin 1+Sleeve=Ground, Pin 2 + Tip = High, Pin 3 + Ring = Low
Controls				
Variable Input Sensitivity	+6dB to -30dB	+6dB to -30dB	+6dB to -30dB	+6dB to -30dB
3 Position HF Adjust	—	—	+1dB, Flat, -1dB	+1dB, Flat, -1dB
3 Position LF Adjust	—	—	-3dB @45Hz, @50Hz, @65Hz	-3dB @39Hz, @47Hz, @55Hz
Amplification	HF 15 Watts, LF 30 Watts	HF 30 Watts, LF 60 Watts	HF 60 Watts, LF 120 Watts	HF 60 Watts, LF 160 Watts
Frequency Response	65Hz - 20kHz ±2dB	58Hz - 20kHz ±2dB	47Hz -20kHz ±2dB	39Hz - 23kHz ±2dB
Maximum SPL @ 1m	101dB Music 104dB Peak	102 dB Music, 105 dB Peak	108dB Music, 111 dB Peak	112 dB Music, 115 dB Peak
Dimensions (HWD)	9¼ x 6 x 7¾″	13⅜ x 9⅞ x 10″	15⅞ x 11 x 12″	13⅜ x 20 x 16¾″
Shipping Weight	11 lbs.	59 lbs. per pair	38 lbs. each	52 lbs. each

CLOSE-FIELD PASSIVE MONITORS

RoKit II



The same cutting-edge technology and smooth, musical sound as the K•RoK II in a compact package. KRK's most affordable monitor, the RoKit II brings KRK sound quality to desktop workstations, offline editing suites, or any environment and budget where great audio isn't just a luxury, but an absolute necessity. The RoKit II monitor features 7/8" MDF (medium-density fiberboard) construction to control sound reflection and standing waves within the monitor. Double-layer anodized aluminum cone offer extremely low distortion. Carefully molded woofer beauty rings improve midrange frequencies, while advanced dual-port tube designs create improved air flow for rich, solid bass response. Standard magnetic shielding makes placement easy in any studio environment.

6000



The 6000 is a high quality compact monitor prescribed for all close-field applications. Weighing only 18 lbs. each (shielded versions are 4 lbs. more), the portable 6000 combines extremely smooth frequency response with very low distortion. Modestly priced, it derives its sonic attributes from the 7000B and shares the same special design inverted dome 1″ Kevlar tweeter. The 6000 is also available fully shielded for video applications (6000/S.)

K•RoKII



The next generation of K•RoK monitors take passive near-field to a higher level. The attractive new-look cabinet offers an 8″ aluminum tweeter and 1″ silk dome tweeter that offers exceptional, accurate, smooth response and higher power handling without ear fatigue. The K•RoK II monitor features 7/8″ medium-density fiberboard construction to control sound reflection and standing waves within the monitor. Double-layer anodized aluminum cone offer extremely low distortion. Carefully molded woofer beauty rings improve midrange frequencies, while advanced dual-port tube designs create improved air flow for rich, solid bass response. Standard magnetic shielding makes placement easy in any studio environment.

7000B



The 7000B close-field monitor is the industry standard for top-of-console placement. It offers improved frequency response and lower distortion, a Kevlar woofer and tweeter, and features KRK's phase alignment design. Also very portable at 25 lbs. (each), and compact enough (14.25 x 11 x 11″) to fit neatly into facilities with limited space. With a larger 7″ woofer and larger cabinet, the bass response of the 7000B extends lower than the 6000. Available fully shielded for video applications (7000B/S).

KRK PASSIVE MONITOR SPECIFICATIONS

	RoKit II	K•RoK II	6000	7000B
Video Shielding:	Standard	Standard	Optional	Optional
Drivers:HF:	1″ Silk Dome Tweeter	1″ Silk Dome Tweeter	1″ Inverted Dome Kevlar	1″ Inverted Dome Kevlar
LF:	6.5″ DLDA aluminum woofer	8″ DLDA aluminum woofer	6″ Polyglass	7″ Polyglass
Frequency Response:	62Hz-20kHz ±2dB	52Hz-20kHz ±2dB	62Hz-20kHz ±3dB	50Hz-20kHz ±3dB
Sensitivity 1W @ 1m:	89dB	90dB	89dB	91dB
Maximum SPL @ 1m:	104dB	106dB	106dB	110dB
Nominal Impedance:	8 Ω	8 Ω	8 Ω	8 Ω
Maximum Power Handling:	120 Watts	120 Watts	75 Watts	150 Watts
Dimensions (WHD):	13 x 8 x 9″	15 x 9 x 10″	13 x 9 x 10″	14¾ x 11 x 11″
Shipping Weight:	21 lbs. each	26 lbs. each	36 lbs. per pair	50 lbs. per pair

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EQUIPMENT LEASING AVAILABLE

EXPOSÉ E8T

For those seeking perfection, the Exposé E8T is by far the most advanced studio reference monitor available. From the unusual shape to its cutting-edge components, the E8T was designed as the ultimate near-field monitor. Extremely linear frequency response and beautiful stereo imaging starts with two symmetric discrete power amplifiers and newly designed crossover circuitry. A double-woven Kevlar woofer provides commanding bass. The 1˝ Tioxid 5 tweeter matches titanium for mass and stiffness, but offers superior damping qualities for highs you can reach out and touch. Available in custom colors and finishes. Video shielding is standard.



S SERIES SUBWOOFERS



S8

The S8 is ideal for small areas. Construction consists of 3⁄4˝ MDF, which reduces resonance diminishing coloration to provide a more even sound. Like it's big brothers, the S12 and S10, the S8 has radiused corners and edges, and a slotted port to give you tighter bass. One of the features of the S8 is a low pass variable that enables you to tailor the crossover point to match satellite monitors. The phase switch also gives you flexibility in room placement. Matched with five V4 powered speakers, it's the perfect surround solution for limited space.

S10



The S10 features a cast frame 10˝ woofer with a special Kevlar cone. Created wide instead of tall, it fits easily under your mixing console. The built-in active crossover is designed for both LCR and 5.1 systems. Look inside and you'll see incredible attention to detail; the S10 has both vertical and horizontal internal braces, and even the ports are covered with sound deadening material. These are the kind of details that add up to make the S10 the best sounding active subwoofer in its price range.



S12

Virtually the same as the S10, the S12 features a cast frame 12˝ woofer with a special Kevlar cone. It has both vertical and horizontal internal braces, and the ports are covered with sound deadening material. Compared to the S10, the S12 has a better low frequency limit (31 Hz) and is 3 dB louder.

	S8	S10	S10
Drivers	LF 8˝ Woven Kevlar	LF 10˝ Woven Kevlar	LF 12˝ Woven Kevlar
Input Connections	2 channel XLR 10K Ω Balanced	2 channel XLR 10K Ω Balanced	2 channel XLR 10K Ω Balanced
Output Connections	2 channel XLR	2 channel XLR	2 channel XLR
Controls	System Gain +6dB to -30dB Variable; Low Pass 50Hz-130Hz Variable; Phase Adjust Switch 0° or 180°		
Line Out HP Filter	80Hz fixed	80Hz fixed	80Hz fixed
Amplification	100 Watts	125 Watts	160 Watts
Frequency Response	43Hz to 50Hz-130Hz Variable	38Hz to 50Hz-130Hz Variable ± 2dB	31Hz to 50Hz-130Hz Variable ± 2dB
Dimensions (HWD)	10½ x 14¾ x 12⅝˝	13 x 18¾ x 17˝	15 x 22 x 20˝
Shipping Weight	34 lbs. each	53 lbs. each	72 lbs. each

MACKIE

HR624/HR824

6.7″ and 8.75″ 2-Way Active Near-Field Monitors



One of the most accurate near-field monitors ever built, thousands of recording studios and post-production facilities rely on the HR824 to reveal details that comparable monitors simply cannot resolve. Frequency response is an astonishing 38Hz to 20kHz ±1.5dB. To achieve such a sweet spot and smooth response, the HR824 uses technology not found in typical vented-box near field monitors. A cast zinc exponential waveguide disperses sound evenly and integrates the output of the HF and LF drivers. Its servo-coupled LF transducer works in conjunction with a rear-firing passive transducer to extend low octaves and eliminate vent noise. The HR824 uses two separate high-current amplifiers with discrete circuitry. And it is fully stuffed with dense foam to fully damp internal vibrations before they can escape through the LF transducer.

Employing the same acoustic technology as the HR824, but in a compact version, the HR624 is deal for rear surround sound channels in existing HR824 front left/right systems or as a main monitor system in control rooms and editing suites where space is at a premium. When using the HR624 as the primary monitor in project rooms or off-line editing suites, mixes will flawlessly transfer to an "A" room that employs HR824's. The HR624 is flat ±1dB from 55Hz to 20kHz.



HR626

Dual 6.7″ Active Studio Monitor

The latest addition to the HR series, the 140-watt HR626 employs a D'Appolito woofer-tweeter-woofer design (one 6.7″ woofer on each end of the cabinet with the dome tweeter in the center) to provide a bottom end coupled with low-level accuracy of the mid range—the essential component for establishing vocal placement within a mix. Placing the HR626 in a vertical position provides extremely accurate reference monitoring for mid-to large-sized studios. Additionally, with its symmetrical left-right sound field, the HR626 can be placed horizontally making it a perfect center channel for 5.1 and 7.1 surround sound studio or home theater systems.

THX PM3 certified, the HR626 uses a high-frequency Waveguide-loaded 1″ dome tweeter for crisp, evenly dispersed highs along with dual 6.7″ low frequency drivers that are able to provide extended low-frequency response while preserving the speed and accuracy of a single 6″ design.

Like the HR624, the HR626 incorporates dual FR Series amps (100w to the woofer, 40w to the tweeter), 1/4″ XLR balanced and RCA unbalanced inputs, sensitivity control, 80Hz high-pass filter, ±2dB high frequency shelving filter, Acoustic Space control and Auto On switch. Front panel provides a Standby/Mute switch along with power and overload LEDs. 5/8″ internally braced MDF cabinet is extremely rigid and non-resonant.

HR624/HR824 Features

- Wave guide technology matches the high frequency dispersion pattern to the bass transducer for a seamless midrange transition. Furthermore, the high frequency transducer is recessed, providing acoustic alignment of high and low frequencies.
- 1″ high-frequency aluminum dome driver with ferrofluid-cooled voice coil.
- High frequency shelving control adds or attenuates frequencies above 10kHz by 2dB
- 6.7″ (HR624) or 8.75″ (HR824) RCF Precision low frequency transducer with mineral-filled polypropylene cone, die-cast magnesium frame and extra-long-throw voice coil are servo-coupled with a dedicated 100-watt or 150-watt amplifier
- 40-watt (HR624), 100-watt (HR875) high frequency amplifier. Proprietary Fast Recovery circuitry prevents latching when the amps are driven at their maximum.
- Low frequency switch with 49Hz (full range) and 80Hz roll-off settings (HR624); 37Hz (full range) 47Hz and 80Hz roll-off settings (HR824)

HR624/HR824

- Magnetic shielding
- Input sensitivity control, ideal for dealing with sensitive artists or insensitive clients
- Overload LED indicates when protection circuitry kicks in
- Acoustic space switch allows bass optimization when placed freestanding (full space), against back walls (half space), and corner placement (quarter space)
- 1″ MDF (Medium Density Fiberboard) front board. Internal cavity is filled with high-density foam to absorb the internally-projected midrange before it can exit through the low frequency transducer cone
- Three mode power switch: On, Auto On (with front panel clipping LED and a signal present/power indicator), and standby
- XLR and TRS jacks accept virtually any line-level input from a mixing console
- Power cord and all inputs (balanced XLR and 1/4″ TRS, unbalanced RCA) are mounted on a downward-facing, vertical panel. This allows speaker placement close to walls and eliminates strain on the connectors.

HRS120/HRS150

12″ 500-watt and 15″ 950-watt Active Subwoofers



The HRS120 is not only designed to work perfectly with the HR624/824 monitors, but with any active system where fast, precise low-frequency reproduction and flexible monitoring and control options are required. Rather than creating boomy "home theater"-style bass, the HRS120 delivers extremely accurate low end (flat to 21 Hz and just 3dB down at 19Hz). If your mixes are heavy in bass and beats, such as rap, hip hop and techno, the HRS150 with 950 watts of power driving the 15″ downward firing woofer and dual 12″ passive radiators is the perfect choice.

- The subs can be defeated while keeping the satellite speakers in high-pass-only mode, or switch out the subwoofer and open up the satellite speakers to their full frequency range
- A 110/Hz elliptical filter for Dolby AC-3, THX Crossover Defeat switch, and 0dB/+10dB Output Level switch for AC-3/THX Dot1 operation make the HRS120 a serious tool for sound production
- The HRS150 provides simple switches for THX low cut filter and LFE gain allowing accurate THX mixing and monitoring. Additional filter options allow for AC-3 and DTS applications.
- Left and right inputs are provided for a 24dB per octave variable crossover ranging from 55-110 Hz allowing the subs to match crossover points with any monitors.

	HR824	HR624	HR626
Frequency Response	39Hz - 20kHz ±1.5dB	52Hz - 20kHz ±1.5dB	42Hz - 20kHz ±1.5dB
Maximum SPL @ 1m	120dB	115dB	118dB
LF Transducers (Low Frequency)	8.75″ with die-cast magnesium frame polypropylene cone	6.7″ with die-cast magnesium frame polypropylene cone	6.7″ with die-cast magnesium frame polypropylene cone
HF Transducers (High Frequency)	1″ viscous edge-damped aluminum dome with ferrofluid-cooled voice coil	1″ viscous edge-damped aluminum dome with ferrofluid-cooled voice coil	1″ viscous edge-damped aluminum dome with ferrofluid-cooled voice coil
Transducers (Passive)	6x12″ elliptical flat piston with variable thickness filleted rubber surround	6x9″ elliptical flat piston with variable thickness filleted rubber surround	6x9″ elliptical flat piston with variable thickness filleted rubber surround
Amplifier (LF Power)	150 watts, 4Ω load, 350w peak	100 watts, 4Ω load	100 watts, 4Ω load
Amplifier (HF Power)	100 watts, 6Ω load, 210w peak	40 watts, 8Ω load	40 watt, 8Ω load
S/N Ratio (LF)	> 102 dB, referenced to 150 watts into 4 Ω	>101dB, referenced to 100w into 4Ω	>101dB, referenced to 100w into 4Ω
S/N Ratio (HF)	> 102 dB, referenced to 100 watts into 6 Ω	>93dB, referenced to 40w into 8Ω	>93dB, referenced to 40w into 8Ω
Crossover Type	Modified Linkwitz-Riley 24dB/oct @ 2kHz	Modified Linkwitz-Riley 24dB/oct @ 3kHz	Modified Butterworth 18dB/oct @ 1.8kHz
Enclosure	3/4″ MDF with 1″ MDF front panel	5/8″ MDF with 1″ MDF front panel	5/8″ MDF with 1″ MDF front panel
Controls	Acoustic Space EQ: flat/-2dB/-4dB @40Hz High Frequency EQ: flat/±2dB@10kHz LF Roll-off EQ: flat/47Hz (4th-order Chebyshev filter)/ 80 Hz (4th-order Butterworth filter)		Acoustic Space EQ: flat/–2 dB/–4 dB @ 40 Hz High-Frequency EQ: flat/±2 dB @ 10 kHz; LF Roll-off EQ: flat/80 Hz (2nd-order Butterworth filter)
Dimensions (HxWxD)	15.75 x 10 x 12.2″	13 x 8.25 x 12.5″	8.8 x 19.6 x 12.4″
Weight	33 lbs.	25 lbs.	31 lbs.

SP-5B/SP-8B

Active Near-Field and Reference Studio Monitors



If you have limited space or budget, the sound of the SP-5B near-field monitors will amaze you. You get the same engineering criteria and component quality that goes into the SP-8B reference monitors: two-way bi-amped power, magnetic shielding, swivel-mounted tweeter, high-quality crossover network electronics and Sub-Frequency Port. The 5″ mineral-filled polypropylene woofer and 3/4″ silk tweeter accommodate a smaller cabinet design that allows placement just about anywhere—all with a frequency response of 48Hz to 22kHz. Backed by 75 watts of power, these babies scream.



The SP-8B offers two-way bi-amped near-field technology delivering 100 watts of professional sound at a fraction of the cost of similar-sized monitors. The 8″ mineral-filled polypropylene woofers and 1″ silk tweeters work in conjunction with 4th-order Linkwitz Riley crossover networks and custom cabinet designs to yield smooth frequency response across the audio spectrum. The Sub-Frequency Port even pumps out frequencies down to 33Hz. Magnetic shielding minimizes interference with computer monitors and the unique swivel-mounted tweeter design even allows you to fine-tune your sweet spot.

Why Studiophile Monitors?

You can have all the best gear in the world and still turn out inferior mixes unless you have monitors that tell your ears what's really happening. With that in mind, M-Audio designed the Studiophile line—world-class reference monitors that deliver exceptional quality at amazingly affordable prices. Choose from the SP-8B with 8″ woofer or the bookshelf-sized SP-5B with 5″ woofer. Add the optional SP-8S subwoofer to either and you've got all the bass you need to rattle the rafters.

The key to the Studiophile line's accurate sound is the synergistic design and integration of all elements, thus providing superior performance and fidelity over component systems. Specially designed network and power amplifiers properly distribute low, mid, and high frequencies to reduce distortion and power loss. M-Audio further optimized the active circuitry to deliver flat frequency response, smooth transition in the crossover region, and maximized power handling.

The silk tweeters are swivel-mounted, allowing you to adjust their position for ultimate control over imaging and sweet spots. The tweeters also incorporate magnetic shielding to minimize reaction time, as well as ferro-fluid and internal damping technologies to minimize resonance. The woofers feature a magnetically shielded mineral-filled polypropylene cone with a high-temperature tolerant voice coil and damped rubber surround. And the cabinet designs produce amazing bass for these sized enclosures thanks to a Sub-Frequency Port designed to discharge extreme low frequencies.

Studiophile Standard Features

- Active design with integral crossover and amplification
- Silk high-frequency drivers incorporate ferrofluid cooling and internal damping technologies to minimize resonance
- Low-frequency drivers feature mineral-filled polypropylene cones with high-temperature tolerant voice coils and damped rubber surrounds
- Electronic time alignment of drivers delivers controlled dispersion for superior three-dimensional imaging
- All drivers are magnetically shielded for desktop music production
- Protected from RF interference, over temperature, turn on/off transients
- The tweeter is swivel mounted so that you can adjust the position and direction. This unique feature accommodates various loudspeaker-mounting positions, allowing great control over imaging.
- The Sub-Frequency Port discharges extreme low frequencies under 30Hz. The monitors take advantage of this port to generate extreme low frequencies despite the small footprint of the enclosures.

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BX5/BX8

Reference Monitors



The latest in the popular Studiophile series , the BX5 and BX8 deliver exceptional quality and professional sound. They offer 5.25˝ (BX5) and 8˝ low-frequency drivers (BX8), 1˝ high-frequency drivers, and 75 or 130 watts of bi-amped near-field technology at a fraction of the cost of other monitors of this quality. The 1˝ natural silk high-frequency drivers employ a flush design and electronic time alignment with the low-frequency driver, resulting in controlled dispersion for superior three-dimensional imaging. The tooled rear Sub-Frequency Ports reduce low-frequency turbulence and optimize driver excursion for amazing bass in a small footprint.

Monitors that adapt to your needs

Your monitors should be able to adapt to any changes in placement or work environment. That's just what the BX5 and BX8 do. Since a monitor's bass response can change depending upon proximity to walls, their Acoustic Space control section allows you to optimize their performance—no matter whether you need to put them on stands, on a bookshelf or in the corners. You can also adjust the high-frequency response to compensate for reflective surfaces like glass. The presence control gives you an automatic mid-range boost for added flexibility. There's even a variable low-frequency roll-off to optimize response at crossover points when adding a subwoofer like the SP-8S.



Acoustic Space Control Section

	SP5B	SP8B	BX8
Type	2-way near field	Two-way near-field	Two-way studio reference
Frequency Response	33Hz - 22kHz	33Hz - 22kHz	37Hz - 20kHz
Crossover Frequency	2.7kHz	1.8kHz	1.8kHz
Power	LF: 42w; HF: 33w	LF: 55w; HF: 45w	LF: 65w; HF: 65w
S/N Ratio	>100dB below full output, 20kHz bandwidth	>100dB below full output, 20kHz bandwidth	>100dB below full output, 20kHz bandwidth
Input Impedance	30k Ω balanced, 15k Ω unbalanced	20k Ω balanced, 10k Ω unbalanced	20k Ω balanced, 10k Ω unbalanced
Dimensions (HWD)	9.8 x 6.2 x 7.9˝	15 x 9.8 x 11.8˝	15 x 9.8 x 11.8˝
Weight (per unit):	11 lbs.	20.3 lb.	20.3 lb.

SP8S *Subwoofer*



While the SP-8B and SP-5B offer as much or more bass response as you'll find in any monitors of similar size, the addition of an SP-8S subwoofer puts 120 watts of truly bone-shaking bass in your face. In fact, so many consumers have subwoofers that you're not really mixing accurately without having a sub yourself. M-Audio's unique Stereo Bass Management System splits the signal at the variable crossover frequency, routing everything below to the sub and everything above to the mains. The SP-8S is so affordable, it makes an excellent companion to other monitors as well.

What's Stereo Bass Management?

When subwoofers are added to many systems, all frequencies are routed to both the main monitors and the sub. The sub typically has a low-pass filter to roll off the highs-as does the SP-8S. The problem is that both the main monitors and the sub are still trying to put out the same low frequencies, thereby compromising fidelity. Our Stereo Bass Management System solves that and delivers optimal quality. The SP-8S accommodates the stereo outputs from your mixer, and your main monitors then connect to outputs on the sub. The SP-8S's internal crossover network splits the signal at a user-defined crossover frequency (50-180Hz), routing everything below it to the internal sub and everything above it the main outputs. This significantly improves fidelity because the sub and mains aren't tripping over each other to reproduce the same frequency-and each component only has to reproduce frequencies to which it is ideally suited.

NHT PRO

A-20/C-20

Powered Studio Monitors

NHTPro believes a studio monitor shouldn't deliver "good" sound, but complete sound. Like a fine sports car, the goal isn't comfort, but total feedback about road and driving conditions. When all is said and done, your studio monitor is what links flesh and blood into the recording console, enabling you to make the expert decisions that mean exceptional product.
Through it all - miking decisions, mixing decisions, processing decisions - your monitors must remain unflinchingly accurate and unerringly consistent. There is little point in developing new monitors and then adhering to traditional design decisions. This is why you'll find that many important aspects of NHTPro's modular monitoring systems, while sharing basic functional elements with other speakers, are unconventional.



Modular Amplification

The first element that distinguishes NHT monitors from other powered monitors is their use of "external" amplifier/crossover modules. The benefits of powered systems are widely recognized, buy it often goes unnoticed that integrating the crossover and amplifier into the speaker enclosure can produce performance and convenience penalties. NHT's external approach maintains the advantages of a unified system, while overcoming the limitations of single box designs. Dynamic headroom and sustained power output capabilities are on par with the finest rack-mounted amplifiers, with response tailored exactly to the monitors. Controls are easy to reach, logical to use, and designed for positive, repeatable setting.

Crossovers

They use passive components that are small, electrically well-behaved and can be made to naturally "track" changes in the driver frequency for consistent response at high listening levels. Furthermore, single amp systems can be designed with a lower residual noise floor than bi-amped ones, due to a much simpler signal path, and to the introduction of bandwidth filtering just prior to the driver terminals. Each modular systems comes with matched, controlled-parameter, low impedance XLR speaker cables.

A-20 System

When your work demands the absolute in resolution, the A-20 is your ally in excellence. By providing both frequency and time domain performance that sets new standards of accuracy, the A-20 reveals the minute differences in sound character that distinguish the very finest recordings. Details of microphone selection and placement, processing and equalization, reverberation and acoustic signature,s signal purity and integrity, are revealed without emphasis or disguise.



C-20 amplifier

C-20 System

The C-20 is a powered center channel monitor designed for surround recording. Combined with the A-20 system and B-20 powered subwoofer, the C-20 forms the basis of a matched, balanced reference-quality 5.1 monitoring solution. The C-20 is an exact duplicate of A-20 stereo monitoring system in a monaural configuration. The 250-watt amplifier is housed in a space saving 1RU chassis with the same input and control set. It employs the same A-20 driver configuration in an acoustic suspension design. However, the enclosure is rectangular for proper center channel dispersion.

Features

- Dedicated 250-watt/channel, fully discrete mono (C-20), stereo (A-20) amplifier
- 6.5″ high excursion, treated paper cone woofer
- 1″ metal-dome, fluid-cooled tweeter
- Magnetic shielding (partial on the A-20)
- Power, clipping indicator
- Matched impedance cable(s) included
- Angled speaker cabinets improve detail imaging (A-20 only)
- Headphone output (A-20 only)

Controls

- Inputs: XLR and 1/4″ TRS (parallel)
- Outputs: XLR
- 5-position input sensitivity: (+11, +4, -3, -10dB, mute)
- 5-position LF compensation for boundaries
- 5-position HF compensation for near, mid, and far field
- Power: On/Off
- Diagnostic display: SPL, line VAV, heat sink temperature

B-20

Powered Subwoofer System

Designed to integrate seamlessly with NHTPro's flagship A-20 monitors or any larger powered monitoring system, the B-20 produces remarkable, musical bass from a compact and flexible design. With output >100dB at 30Hz, the B-20 adds full frequency response and dynamics to near, mid and far-field monitoring environments. Offering a fully featured set of controls, the system can be seamlessly integrated with most monitors and most rooms. And the remote control included in the B-20 system integrates it into any sort of mix or production, offering on the fly stereo/mono switching and system in/system bypass, which also engages or defeats the high pass filter.



FEATURES

- The B-20 is a 3-piece system consisting of two acoustic suspension subwoofers mated to a dedicated external stereo amplifier. The system maintains nearly perfect transient response, and whether in stereo or mono mode delivers flat bass response with minimal room interaction and remarkable system coherence.
- Employs two custom designed, extremely long throw paper cone woofers with massive surrounds. Excursion for the woofer is 2″ peak to peak and 1.6″ linear allowing high output with low distortion.
- Each 14″ square B-20 cabinet is constructed of 1″ MDF bonded to high-pressure laminate both internally and externally creating a rigid, sonically-neutral environment for each driver. Cones are provided to decouple the woofer from the floor if required.
- The system's control amplifier is similar to other NHT Pro modular designs. This approach offers superior heat dissipation and eliminates potential vibration induced problems allowing higher power levels and improved headroom. As a result, the B-20 Control Amplifier, featuring dual-mono construction with discrete output devices, delivers 250W RMS/CH (400W peak), while taking up only two rack spaces.
- Ergonomically, this modular design puts the B-20's user controls, all of which are located on the front of the amplifier, within reach of the listening position providing ease of setup and fine adjustments.

Controls

- A five-position low frequency compensation circuit is provided for adjusting the system's bass response for corner placement (three boundaries), mid-room room placement (zero boundaries), or anywhere in between. Lets you maintain a consistent, flat power response in a variety of positions or differing locations.
- High and Low Pass filters provide the ability to seamlessly match the subwoofers to the monitors and add improved system dynamics. High pass settings are 35Hz, 60Hz, 85Hz and 100Hz. The low pass settings are 70Hz, 85Hz, 95Hz and 105Hz. In addition each filter circuit includes a Bypass setting.
- Five-position input sensitivity control (+11dB, +4dB, -3dB, -10dB, and Mute) provides for noise free gain matching with virtually any output device, but also allows the listener to easily turn off the monitors when other playback systems are desired.
- Five position (-180, -90, 0, 90, 180°) phase compensation control for such variables as room placement, listening position and the types of monitors being used. Phase positions are
- Subwoofer gain control matches the B-20 output level to that of the monitors. The B-20 gain circuit offers a 20dB range of adjustment.

	B-20	S-00
Type	Modular, active stereo subwoofer system	Integrated, Self-powered Subwoofer
Configuration	Acoustic suspension	Acoustic suspension
Woofer	Two, 10″ ultra-long excursion (1″ linear, peak-to-peak), treated paper	8″ treated paper cone
Amplifier Power	250w RMS/channel), 400w peak	125W (continuous rms/ch), 250W (500ms peak)
Response	29Hz - 100Hz @1M	39Hz - 110Hz @ 1M ± 3dB
-6dB LF Cutoff	25Hz (in-room response)	
Peak Acoustic Output	110dB SPL @ 40Hz	108dB SPL @ 60 Hz
Control Amp Dimensions / Weight	3.5 x 19 x 13.375″ (HWD) / 37lbs.	
Subwoofer Dimensions / Weight:	14 x 14 x 16″ (HWD, including grill) / 38lbs (each)	13.25 x 10.25 x 13″ (HWD) / 30 lbs.
Monitor Enclosure Materials	1″ mdf w/ HP laminate (internal + external)	

NHT PRO

M-00

Powered Mini-Monitor



Versatile, powerful, compact and portable, the M-00 combines the refined sonics of a high-end mini-monitor with rugged construction, high output, logical controls and a wide range of input options, making it ideal for location work, multimedia, the control room, editing suite, surround monitoring and desktop audio production. The M-00 packs substantial output, surprising bass response and startling accuracy in a remarkably small monitor. The fully magnetically shielded M-00's are sold individually, providing application flexibility in mono monitoring, two-channel, 5.1 or daisy-chained configurations for use in "light" commercial installations.

- Paper cone 4.5″ woofer with a two-layer voice coil allows sustained high temperature operation, while the 1″ ferro-fluid cooled, fabric dome tweeter employs a sophisticated "under-hung" motor design for improved transients and lower distortion.
- Cast aluminum-zinc alloy enclosure is structurally rigid, minimizing audible resonance and sonic coloration. The enclosure also acts as a massive heatsink, allowing the use of a high voltage, discrete power amplifer.
- Fully discrete power amp provides 75W RMS continuous (150W peak) output. Magnetically shielded as well to allow use in direct proximity to computer monitors.
- XLR, TRS and phono input jacks, allow easy direct connection to a variety of devices while minimizing the need for adapters.
- Also provides noise free connection to a wide variety of devices, via its +4dB/-10dB input sensitivity switch.
- XLR and TRS inputs are paralleled so that multiple M-00's can be daisy chained on a single channel
- Capable of near-field and mid-field listening positions depending on your listening proximity, with a simple flip of the switch.
- The M-00 has mounting bosses and threaded holes for use with the popular OmniMount 50/53 Series bracket systems. May also be wall or ceiling mounted .

	A-20	M-00
Type	Modular, self-powered near/mid/far-field monitor	Integrated, self-powered near/mid-field monitor
Woofer / Tweeter	6.5″ treated paper / 1″ aluminum dome	4.5″ treated paper / 1″ soft fabric dome
Amplifier Power	250w (continuous rms/ch), 400w (100ms peak)	75w (continuous rms/ch), 150w (100ms peak)
Peak Acoustic Output	117dB SPL (100ms pink noise @ 1M)	111dB SPL (100ms pink noise @ 1M)
THD @ 90dB SPL	< 0.4% (100Hz - 10KHz @ 1M)	< 1.0% (100Hz - 10KHz @ 1M)
Dimensions (HWD)	14 x 7.5 x 11.9″	9 x 5.7 x 7.3″
Weight	17 lbs.	14 lbs.

S-00 Powered Subwoofer



For those wanting a full-range compact surround monitoring system, the S-00 is a compact, subwoofer designed for use with the M-00 or other small powered monitors. The S-00 features a 125-watt high voltage built-in amplifier incorporating low and high pass filters and a footswitch for bypass. Its driver is a long-throw treated paper 8″ woofer providing output down to 39Hz. Less than one cubic foot in volume, the S-00's cabinet is constructed from 3/4″ MDF, providing a structurally sound and fully acoustically neutral cabinet. A durable powdercoat finish protects it in portable uses.

- Switchable high pass filter, variable low pass filter
- Detented variable subwoofer output control
- Defeatable Auto Power-On sensing circuit with front LED indicator
- Two-position input sensitivity (+4, -10dB) control
- Subwoofer + highpass filter defeat via footswitch
- Balanced TRS outputs; parallel balanced TRS, XLR, and unbalanced phono inputs

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EQUIPMENT LEASING AVAILABLE

RESOLV 50A



Active Reference Monitors

The Resolv 50a has a compact cabinet that will fit anywhere. It features 70 watts of bi-amped power; with 50 watts driving a 5.25″ midrange driver, and 20 watts driving the 1″ ferro-fluid titanium high frequency driver. It has an active crossover, 1/4″ and RCA inputs and the compact cabinet is ported, carefully tuned and wall mountable (with optional bracket).

RESOLV 65/RESOLV 65a

Passive/Active Reference Monitors



Ideal for professional, project and personal recording, as well as multimedia setups, the Resolv 65 (passive) and 65A (powered) sound full and balanced with an uncompromised transparent midrange, leaving room to accurately mix vocals, snare hits and reverb trails.

- They feature 6.5″ copolymer butyl surround Woofer, a 1″ titanium diaphragm, high-frequency driver and gold plated binding posts all mounted in a ported tuned cabinet.
- The Resolv 65 and 65a utilize a midrange contour control. In the nominal position, the monitors are clinically accurate, but by dialing in the control, you can tailor the frequency response from Hi-Fi to an aggressive midrange, emulating classic near field monitors.
- Otherwise identical, the Resolv 65a is powered by a dual power amp, (75 watts low and 25 watts high) perfectly matched to the drivers. The active crossover allows precise control of the crossover frequency.



RESOLV 80a

Ideal for mid- to large-sized control rooms where high volume and serious bass response is required. Bi-amped, with 75 watts driving an 8″ woofer and 25 watts driving a 1″ ferro-fluid filled titanium tweeter. The Resolv 80a has 1/4″, RCA and XLR inputs. The 80a features a four-position adjustable midrange control.

RESOLV 120a *Active Subwoofer*



Taking care of the low end is the Resolv120a powered subwoofer. A powerful 120 watt low frequency amplifier drives a heavy-duty 10″ transducer, reproducing tones between 40–180 Hz. The unit has a built-in active crossover, a phase switch, an auto sleep mode and a convenient mute switch jack allowing the user to easily switch on and off the subwoofer on the fly.

	Resolv 50a	Resolv 65a	Resolv 80a	Resolv 120a
LF Amp	50 watts @1k into 4Ω	75 watts @1k into 4Ω	75 watts @1k into 4Ω	120 watts into 4Ω
HF Amp	20 watts @10k into 4Ω	30 watts @10k into 4Ω	25 watts @10k into 4Ω	—
Frequency Response	60 Hz to 30 kHz	40 Hz to 20 kHz	40 Hz to 20 kHz	30 Hz to 150 Hz
Dimensions (HWD) / Weight	12 x 8 x 9.5″ / 14 lbs.	13⅜ x 8.5 x 9⅞″ / 18 lbs.	16 x 11¼ x 13″ / 21 lbs.	14.5 x 14.5 x 17.5″ / 39 lbs.

TANNOY

SYSTEM 600/800

6.5″ and 8″ Dual Cocentric Near-Field Monitors



Compact, professional near-field reference monitors, the System 600 and System 800 offer the advantages of point source operation in a very cost-effective design. Their advanced 6.5- and 8″ Dual Concentric drivers incorporate a high performance tweeter positioned immediately behind the center of the bass cone. This design produces all frequencies from a single point, mirroring the way sound is produced in real life. The accurate phase response achieved delivers a realistic soundstage along with a smooth and natural sound which is unfatiguing over long listening periods. The speakers are also capable of high volume levels without compression and deliver an excellent transient response. In addition, the attractive octagonal shape of their cabinet with their original rounded edge corners front panel contributes to the excellent acoustic performance of the cabinet. And though they are designed to be located on top of the mixing console, in a landscape format, they can be used in portrait mode without any drawback, if the situation calls for it.

Capable of producing accurate, high quality sound for a wide range of applications, the System 600 is the ideal choice for those who want consistent performance in a compact enclosure at a very sensible price. A slightly larger version, the System 800 uses an 8″ Dual Concentric driver for extended bass and higher monitoring levels. Finally, for those who need to use the near-fields near video monitors, shielded versions of the System 600 and System 800 are available.

Proto-J



The Proto-J is the latest playback monitor from Tannoy. It has an extremely detailed, dynamic sound with a wide, flat frequency response - all of which are essential for monitoring with reliable accuracy. The 3/4″ soft dome HF unit is seamlessly matched to a 6.5″ long throw bass unit by the precision low-loss crossover network. The drive units are discretely mounted on a 3/4″ MDF baffle. The front edges of the vented enclosure are chamfered to reduce high frequency diffraction. The back panel features two 5 way binding post inputs and integral inserts to accommodate OmniMount 50 Series wall brackets that allows wall mounting for easy and flexible 5.1 surround sound installs. Proto-J is designed to compliment the Tannoy PS 110B 5.1 Subwoofer as a complete package.

Dual Concentric Design

Conventional monitor designs lock you into a tiny ‘sweet spot’ monitoring position. Move to adjust a fader and the sound changes. Not with Tannoy’s Dual Concentric monitors. The patented Point Source design creates a coherent wavefront that is consistent both on and off axis. The result is a high performance reference monitor which you can trust, providing consistent accurate sound over a wide mix position.

The System 600 and 800 monitors use the Dual Concentric design, in which the woofer and tweeter are mounted on the same axis. Their 6.5- or 8″ polypropylene woofer crosses over at 1.6 kHz to the tweeter centered in the woofer cone. This design forms a point source, which results in sharp stereo imaging. It also prevents the lobing often seen with staggered drivers. If you sit anywhere off-axis — up, down, left or right — the response is consistent. The speakers can be placed horizontally or vertically with the same response. The shape of the woofer cone acts as a hyperbolic waveguide for the tweeter sound waves, producing a spherical wavefront. And since the low-and high-frequency signals are aligned in time, the phase response is uniform and the transient response is tight.

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EQUIPMENT LEASING AVAILABLE

SYSTEM 600A/800A

Active 6.5″ and 8″ Dual Cocentric Near-Field Monitors

System 600/800 Features

- Ferrofluid cooling in the tweeter lets it handle high power. Diaphragm is made of aluminum/magnesium for stiffness and low mass. Around the diaphragm is a nitrile rubber surround with a narrow roll that eliminates resonances below 25 kHz, and cannot be destroyed by fatigue.
- Signal input is via a Neutrik combi connector, which combines a three-pin XLR-type with a TRS locking 1/4" phone jack.
- Made of 36mm thick rigid MDF, damped to prevent breakup. The cabinet is covered in black vinyl wrap, while the front baffle is painted midnight blue.
- Cabinet is shaped like an elongated octagon, with beveled edges that reduce diffraction.
- Two panels in the cabinet are fitted with magnetic shielding plates
- Each speaker weighs 23.1 lb. and measures a compact 9 x 14 x 11″ (HxWxD)

System 600A/800A Step-up Features

- Two EQ switches on the back control the bass and treble response (LF /HF contour)
- The LF settings are free-field and half-space. If your speakers are placed relatively far from nearby surfaces, use the free-field setting. If the speakers are near a wall or console, use the half-space setting
- HF adjustments are +2/0/-2 dB shelving
- Selectable input level (-10 dBV and +4 dBu)
- On the front of the cabinet are two bass-reflex ports and a power-on LED



Combining the proven quality of the System 600 and 800 with two amplifiers and fully optimized electronic crossover in a single compact package, the System 600A and System 800A provide the complete solution to active monitoring, yet are in the same price range as some passive speakers. The System 600A and System 800A offer all the advantages of point source operation in a cost-effective design, combined with the benefits of active speaker technology: a highly optimized active crossover section; ideal matching between amplifiers and transducers; higher output for lower distortion. Active user controls are also provided to allow further bass extension (LF contour) and greater flexibility.

System 600A

Active reference monitor, with two 70W amplifiers and electronic crossover. Unique cabinet shaped for optimum acoustic performance. Ideal for left-center-right in a project studio surround system.

System 800A

8″ Dual Cocentric drive unit, with two 90W amplifiers and electronic crossover. A complete monitoring package with the ability to exploit the full performance of today's digital recording equipment.

	System 600	System 800	System 600A	System 800A
Description	Nearfield reference monitor 6.5″ Dual Concentric unit	Nearfield reference monitor 8″ Dual Concentric unit	Active nearfield reference monitor 6.5″ DC unit. 70w + 70w Active crossover. Shielding option	Nearfield reference monitor 8″ DC unit. 90w + 90w Active crossover. Shielding option
Frequency Response	52Hz - 20kHz	47Hz - 20kHz	44Hz - 20kHz	44Hz - 20kHz
Power Handling RMS/Program	80w/160w	90w/180w	—	—
Built-in Amp power RMS	—	—	HF 70w + 70w	HF 90w + 90w
Input Sensitivity	—	—	-10dBu or + 4dBU	-10dBu or + 4dBU
Sensitivity anechoic/half space	90dB/93dB	92dB/95dB	—	—
Impedance	8Ω	8Ω	10kΩ balanced	10kΩ balanced
Dispersion	90° conical	90° conical	90° conical	90° conical
Dimensions	8.7 x 14.2 x 10.5″	10.8 x 17.7 x 10.5″	8.7 x 14.2 x 11.4″	10.8 x 17.7 x 11.4″
Weight	16.5 lbs.	23 lbs.	20.9 lbs.	28.6 lbs.

REVEAL/REVEAL ACTIVE

6.5″ Two-Way Nearfield Studio Monitors

The Reveal combines a number of speaker cabinet innovations with a distinctive speaker enclosure. Ideal for multimedia, project studios, an editing desk, or a broadcast production room, the Reveal has an extremely detailed, dynamic sound with a wide, flat frequency response – all essential for monitoring with reliable accuracy. The 1″ soft dome HF unit is seamlessly matched to a 6.5″ long throw bass unit by the precision low-loss crossover network. The drive units are mounted on a massive 1.5″ thick baffle, curved to minimize diffraction and creating the loudspeakers distinctive styling. Both drive units are magnetically shielded to allow the system to be used close to video monitors. Housed in an attractive red/gray finish, the Reveal's compact dimensions (13.75 x 8.75 x 10.75″ HWD) and shielding make them ideal for studios where space is at a premium.

The same monitor as the plain Reveal, the Reveal Active adds two 50-watt amplifiers and fully optimized electronic crossover in exactly the same sized cabinet. By integrating power in the cabinet, the amps are ideally matched to the drivers and distortion is lower at the same volume level compared to passive speakers. Like the Reveal, the Reveal Active is compact (same dimensions) and shielded. However they are a different color (gray with an azure blue front) and slightly heavier (each speaker is 18.7 lbs. vs. each 15.4 lb. plain Reveal speaker).

	Reveal	Reveal Active
Description	Playback monitor; 6.5″ LF; 1″ softdome tweeter	Active nearfield monitor; 6.5″ LF shielded; 1″ softdome tweeter; 50w + 50w Active Crossover
Frequency Response	65Hz - 20kHz	62Hz - 20kHz
Power Handling RMS/Program	50w/100w	—
Built-in Amp power RMS	—	HF 50w + LF 50w
Input Sensitivity	—	0.775Vrms (0dBU)
Sensitivity anechoic/half space	87dB/90dB	—
Impedance	32kΩ balanced	6Ω
Dispersion	90°	90°
Dimensions	13.4 x 8.3 x 10.2″	13.4 x 8.3 x 10.2″
Weight	15.4 lbs.	18.7 lbs.



PS110B

10″ Active Subwoofer

The PS110B delivers extended bass to small nearfield monitors and provides LF (low frequency) impact to create either subtle or exciting effects. Use with passive and active nearfield monitors such as the Reveal or System 600, where the high pass outputs from the sub allow greater performance from the monitors—great for studios where space is at a premium. In multi-channel environments, the PS110B is ideal for the LFE (low frequency effects) channel in surround sound environment. A single PS110B can be used in small to medium-sized rooms, while multiples can be used in larger environments.



PS350B

10″ Active Subwoofer

Featuring Tannoy's unique "True Bass Management" the PS350B allows referencing of mixes in full-range stereo or 5.1 surround at the the top of a foot switch. The unit offers balanced Left, Center, Right, and LFE (sub in) XLR inputs, with Left, Center and Right balanced XLR outputs. Controls include phase switching, master gain, bass boost, LFE all pass/LFE X-over low-pass, and variable crossover adjustment. With powerful and articulate low frequency performance, the PS350B is the perfect tool for today's modern format studio.



	PS110B	PS350B
Description	Active sub-bass system. 10″ long excursion driver	
Frequency Response	31Hz - 150Hz	29Hz - 300Hz +/-3dB
Built-in Amp power RMS	110w	350w instantaneous
Input Sensitivity	Continuously variable	Continuously variable
Input Impedance	10Ω, 2 channel	10Ω, 2 channel
Dispersion	omni-directional	omni-directional
Dimensions	17.5 x 11.2 x 15.9″	20 x 18.06 x 20″
Weight	33 lbs.	51 lbs.

SYSTEM 1000/1200

10″ and 12″ Dual Concentric Mid-Field Monitors

Incorporating 10″ and 12″ Dual Concentric drivers in the same unique cabinet design as the System 600/800, the System 1000 and System 1200 are professional systems designed for a wide range of midfield monitoring applications from music/project studios to broadcast and multimedia. They are capable of high volume levels without compression and deliver excellent transient response. Their unique Dual Concentric design produces all the frequencies from a single point, mirroring the way sound is produced in real life. The accurate phase response achieved delivers realistic soundstage, along with a smooth and natural sound which is unfatiguing over long listening periods. The System 1000 can be used as a near or mid-field reference monitor, the System 2000 as a midfield or for main monitoring in smaller studios. Both are also available with shielding option.





SYSTEM 15 DMT II/ 215 DMT II

15″ Dual Concentric Main Monitors

The System 15 DMT II is a main monitor system for recording and post-production studios where the requirement is for high performance, natural sound reproduction. With a single coherent wavefront thanks to the 15″ Dual Concentric point source driver, sound presented to the engineers is remarkably distortion free. And with the enlarged monitoring position provided by this remarkable drivers wide dispersion, monitoring with the System 15 DMT II is particularly fatigue free.

The System 215 DMT is the ultimate realization of the Tannoy Dual Concentric principles in a studio monitor. The 15″ Dual Concentric driver, with its wide dispersion characteristics, provides an enlarged 'sweet spot' monitoring position, while the complementary 15″ bass driver reproduces the true impact of low frequencies. A superb, natural sounding monitoring system for those studios where only the best is good enough.

	System 1000	System 1200	System 15 DMT II	System 215 DMT II
Description	Midfield reference monitor. 10″ Dual Concentric unit	Main reference monitor. 12″ Dual Concentric unit	Midfield reference monitor. 15″ Dual Concentric unit	Midfield reference monitor. 12″ Dual Concentric unit, 380mm, 15″ LF
Frequency Response	45Hz - 20kHz	40Hz - 20kHz	38Hz - 25kHz	35Hz - 25kHz
Power Handling RMS/Program	100w/200w	180w/350w	200w/400w	250w/500w
Sensitivity anechoic/half space	94dB/97dB	95dB/98dB	98dB/101dB	101dB/104dB
Nominal Impedance	8Ω	8Ω	8Ω	4/8Ω
Dispersion	90° conical	90° conical	90° conical	90° conical
Dimensions	13 x 21.3 x 11.7″	15.6 x 25.6x 16.1″	33.1 x 21.7 x 17.3″	30.9 x 35.7 x 22.8″
Weight	36.4 lbs.	59.5 lbs.	99.2 lbs.	187 lbs.

ST50/ST100

SuperTweeters

Tannoy's SuperTweeters were designed to provide the extended high frequency response demanded by modern program material and sources such as SACD (Super Audio CD), DVD-Audio and DTS which have driven the requirement for loudspeakers with extended frequency bandwidth performance. By allowing the listener to experience a far wider range of bandwidth information of instruments than is currently possible with conventional loudspeakers, the ST50 and ST100 complete the musical picture. They not only have the ability to resolve fine detail of high frequency notes but also effectively enhance the listening experience even at lower frequencies.



FEATURES

Music contains transient information and rich harmonics beyond the range of human hearing for pure tones. Even bass notes have leading edge transients reaching 30kHz with other instrumentation extending yet further.

Operating between the roll-off point of the high frequency unit of your existing loudspeakers and 54 kHz, the SuperTweeter will accurately reproduce the leading edge of individual notes allowing the listener to experience the entire bandwidth information of instruments.

Extending the frequency response by two octaves, corrects time and phase response within the bandwidth of normal human hearing. Taking these acoustical phase anomalies beyond the range of human hearing adds realism to the soundstage by improving imaging and placement of instruments. The SuperTweeter provides an increased immediacy, airiness and impact - making music sound more natural and true to life.

With the universal nature of the crossover design the SuperTweeter concept has been extended to match the majority of quality loudspeakers, whether they are from Tannoy or from other manufacturers. Adjustments are provided for crossover frequency and sensitivity level to allow system matching.

Discrete drivers, widely used in loudspeaker design will benefit greatly from the addition of a SuperTweeter. The spatial imaging and detail accuracy will be enhanced without the impression that another information source has been added—such is the integration of the SuperTweeter into the system.

The ST50 is constructed from solid aluminum die-castings coated with a soft feel, non-reflecting finish. Not only are they designed to visually blend in with all types of speakers, but they also provide an instant acoustic upgrade that is unavailable in any other way.

The ST100 comes in solid black ash and with a Performance Platform. Includes specific crossover and frequency control settings suited to all types of speakers, regardless of manufacturer. In addition the ST100 is optimized for use with older Tannoy Dual Concentric models, many going back over 40 years to the original 15˝ Monitor Red derivatives.

- High stiffness titanium dome minimizes effects of dome break-up and extends frequency response.
- Neodymium magnet system also extends frequency response by minimizing eddy current losses in voice coil.
- Copper clad aluminum voice coil on aluminum former for high power handling.
- Provision for driver magnet system earthing, to avoid interference and information masking caused by radio interference.
- Gold plated terminals for optimum signal transparency.
- Radio frequency quality inductors in crossover for good high frequency characteristics.
- Non-inductive resistors to avoid losses associated with normal wire wound types.
- High quality polypropylene film capacitors, selected for high frequency use.
- Unique Tannoy Performance Platform gives stable support and variable adjustment for the ideal listening position.

	ST-50	ST-100
Recommended Amplifier Power	Up to 200 Watts	Up to 250 Watts
Power Rating to DIN IEC 268	110 Watts RMS 450 Watts Peak	135 Watts RMS 550 Watts Peak
Max. Sensitivity (2.83V @ 1m)	93 dB	95 dB
Nominal Impedance	8 Ohms	8 Ohms
Frequency Response (-6 dB)	To 54 kHz, usable output (-18 dB) to 100 kHz	
Driver Type	25mm 25 micron titanium dome, 24-carat vapor deposited gold finish with neodymium magnet	
Crossover Frequency	14, 16 or 18 kHz adjustable	
Level Adjustment	85dB, 89dB, 93dB	87dB, 89dB, 91dB, 93dB, 95dB
Finish	Black with gold anodized Performance Platform	Black Ash

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MSP10 STUDIO

Powered 8˝ Studio Monitors

From the classic NS1000 through the ever-popular NS10M series, thousands of pros worldwide have chosen Yamaha speaker systems for critical monitoring applications because of their exceptionally accurate, natural reproduction and reliability. With the MSP10 Studio, Yamaha takes monitor performance to new levels.

The monitor features an 8˝ long-throw woofer housed in a compact bass reflex design cabinet to provide deep and tight low end reproduction. The 1˝ pure titanium dome tweeter with wide dispersion waveguide horn provides a smooth, high frequency response well past 40kHz— perfect for 24-bit/96kHz masters and a well-articulated stereo image. Other features include trim switches for easy room matching, XLR inputs and magnetic shielding.

Designed for post-production, broadcast, professional recording and project studios, the MSP10 Studio has a 120-watt power amp for the low/mid driver and a 60-watt amp for the tweeter. It also adds a low-cut filter for optimum matching with high-performance subwoofers like the SW10.



- Custom-designed 8-inch woofer and 1-inch titanium-dome tweeter feature advanced magnetic structures that achieve exceptionally low distortion.
- Tweeter operates in conjunction with a waveguide horn that achieves broad, uniform high-frequency dispersion for optimum balance regardless of listening position. Advanced driver and enclosure design also ensures smooth, uniform dispersion across the system's full reproduction range.
- 120-watt power amplifier for the low/mid driver and a 60-watt power amplifier for the tweeter (total power of 180 watts).
- Building the amplifiers into the speaker cabinet allows for the best possible damping, for tight, controlled bass and fast transient response for accurate high frequency reproduction. The overall result is exceptionally smooth, natural response over the crossover range with an absolute minimum of distortion at all frequencies.
- 3-position LOW (0, -1.5, -3 dB at 50 Hz) and 3-position HIGH (+1.5, 0, -1.5dB at 10 kHz) frequency TRIM switches enable adjustment of the bass and treble, letting you optimize system response in a wide range of acoustic environments.
- Balanced XLR input for compatibility with professional equipment. Also allows the speakers to be placed at the end of long cable runs without being susceptible to hum and induced noise.
- Sensitivity control dial lets you adjust the volume according to the output sensitivity of the connected device.
- Magnetic shielding allows placement near all types of video and computer equipment
- High-pass filter cuts frequency ranges below 80 Hz for optimum matching with a high-performance subwoofer system.
- Clip indicator if the output level is too high
- Wall-mounted with optional wall brackets

SW10 Powered Subwoofer



If you need the kind of bass response normally associated with a large speaker system, without giving up the compact convenience and positioning ease of the MSP10 Studio, simply add an SW10 Subwoofer for solid, accurate bass response down to well below the audible limit. The SW10 features a long-stroke 10˝ woofer which achieves exceptionally smooth, low-frequency reproduction, while a built-in amplifier delivers 180-watts of power. A variable low pass filter (40 to 120 Hz) makes it easy to achieve optimum crossover with just about any main speaker system, and a built-in phase switch allows instantaneous phase reversal without having to modify cables or connections. The SW10 features three balanced XLR-type inputs and outputs (the outputs feed the audio signal through to the main speakers) for direct connection to professional equipment. Finally, heavy wood construction ensures that the engineer hears the audio signal, not the cabinet.

MSP3



Powered Monitor Speakers

Engineered for today's multimedia environments, the compact, two-way powered MSP3 provides high quality, cost-effective monitoring for home studio, computer-based recording and educational applications as well as personal computers, gaming systems and music keyboards without integrated speakers.

Utilizing many of the same technologies found in the MSP5 and MSP10 Studio professional level monitors, the internally powered MSP3 provides 20W of output power to a 4″ woofer that reproduces a surprisingly tight low end, and a 3/4″ dome tweeter with a smooth, high frequency response to 22 kHz. For added flexibility, there are balanced XLR and 1/4″ phone jacks and unbalanced RCA connectors on each speaker. In addition, each MSP3 has trim switches that allow both the low and high frequency response to be custom-tailored, allowing installation in a wide range of environments. Full magnetic shielding allows positioning near sensitive recording equipment and media.

- Designed for use in personal music production systems, a built-in 20-watt amplifier drives both the 4″ cone and 3/4″ tweeter
- Magnetic shielding and compact size allow placement near video monitors
- Balanced XLR, 1/4″ and unbalanced RCA inputs for versatile connectivity
- Separate volume controls for Line 1 (RCA) and Line 2 (XLR/phone) inputs plus High and Low tone controls for sound tailoring
- Incorporates Yamaha's Waveguide Technology for uniform dispersion of the tweeter's high frequencies
- Mic stand mountable with optional BMS10A Microphone Stand Adapter



MSP5 Speakers shown with YST-SW305 Subwoofer

MSP5

Powered Monitor Speakers

High quality, powered monitors have traditionally demanded premium prices. The bi-amplified MSP5 monitor speakers deliver extraordinary sound quality, at an equally attractive price! The ideal choice for a wide range of professional recording and post production applications including 5.1 surround mixing. Ultra-compact enclosures allow easy placement. Small face print makes them very accurate for "point source" listening. Hookup is quick and easy with both XLR and 1/4" inputs. 67 combined watts of audio power in a monitor this size, let alone near this price, was previously unheard of -- until now. Titanium offers excellent response, separation and durability. The MSP5's dome tweeter, surrounded by a wave guide horn offers a broad coverage area of 120°. Upgrading to studio level monitoring has never been easier.

- With two separate amplifiers, incredibly lower distortion is realized. 67 watts (40 to the woofer and 27 to the tweeter) generates a much larger sound field than would normally be expected from a system of this size.
- Flat frequency response out to 40kHz makes the MSP5 great for mastering, where high sample and bit rates are the norm.
- XLR-balanced and 1/4″ unbalanced inputs
- The smooth circular horn surrounding the tweeter delivers uniform dispersion without distortion or coloration.
- Complete magnetic shielding and compact size permits easy placement near all types of audio, video and computer equipment.
- 4-position low frequency (+1.5, 0, -1.5, -3 dB at 60 Hz) and 3-position high frequency (+1.5, 0, -1.5dB at 15 kHz) frequency TRIM switches enable adjustment of the bass and treble, letting you optimize system response in a wide range of acoustic environments.

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YST-SW005

Advanced Active Servo Processing Subwoofer

A compact, high performer, this affordable subwoofer can be purchased separately or as part of a monitor system. The YST-SW005 features Advanced YST for dynamic bass response, 55W high power output, switchable high cut filter, auto standby with sensitivity selector and two input connections to deliver powerful, high quality bass with superb efficiency and convenient operation.
Available in a complete system, the MSP3 Speaker Bundle includes two MSP3 powered monitors and one YST-SW005 complete with cabling and instructions to make hook up fast and easy.



YST-SW005 Subwoofer shown with MSP3 Speakers—MSP3 Speaker Bundle

- Advanced Yamaha Active Servo Technology (Advanced YST) for powerful bass response
- 55W high output power
- 6-1/2″ multi-range driver
- Magnetic shielding
- Switchable high-cut filter (high or low)
- Auto standby with sensitivity selector (high, low or off) and LED indicator
- Two input connections: Speaker Level or Line Level (pin jack)
- Low 0.8W standby power consumption for exclusive power transformer .

	MSP3	YST-SW005	MSP5	YST-SW305
GENERAL				
Type	Amplified, 2-way, bass-reflex powered speaker	Amplified, bass-reflex subwoofer system	Biamplified, 2-way, bass-reflex powered speaker	Amplified, bass-reflex subwoofer system
Crossover Frequency	4kHz	100Hz or 200Hz switchable LPF	2.5kHz	40Hz to 140Hz continuously variable LPF
Overall Frequency Response	65Hz-22kHz (-10dB)	30Hz-200Hz (-10dB)	50Hz~40kHz (-10dB)	20Hz-160Hz (-10dB)
Dimensions (W x H x D)	5 1/16 x 9 5/16 x 6 5/16″	7⅛ x 14⅜ x 14¾″	6⅝ x 11 x 8¾″	9 5/16 x 14⅜ x 12 5/16″
Weight	9.70 lbs.	18 lbs.	16.5 lbs.	44 lbs.
SPEAKER SECTION				
Components	LF: 4″ cone; HF: 7/8″ Dome	6.5″ long throw woofer	LF: 4½″ cone; HF: 1″ titanium dome	2x 8″ long throw, high compliance
Enclosure	Bass reflex design; Magnetic shielding construction	Bass reflex design Magnetic shielding construction	Bass reflex design; Magnetic shielding construction	Bass reflex design; Magnetic shielding construction
AMPLIFIER SECTION				
Output Power	20W at 1kHz, RL=4Ω	55W at 100Hz, RL+5Ω	Biamplified system LF: 40W at 400Hz. RL=4Ω HF: 27W at 10kHz. RL=6Ω	200W at 100Hz, RL+5Ω
Input Sensitivity	Line 1: -10dB Line 2: +4dBdB	30 mV	Line 1: -10dB Line 2: +4dBdB	30 mV
Input Impedance	Line 1 and Line 2: 10kΩ	12kΩ	Line 1 and Line 2: 10kΩ	12kΩ
Controls	Level: Line 1 & Line 2 Tone Control: Low & High	Level control, high cut filter (low or hi), Auto standby switch, Power ON/OFF	Level: Line 1 & Line 2 Tone Control: Low & High	Level control, high cut filter (continuously variable), Auto standby switch, Power ON/OFF
Power Indicator	Green LED: Power ON	Green LED: Power ON; Red LED: Power OFF	Green LED: Power ON	Green LED: Power ON; Red LED: Power OFF
Power Requirement	AC120V 60Hz	AC120V 60Hz	AC120V 60Hz	AC120V 60Hz
Power Consumption	30W	60W	60W	80W

MS101II / MS202II

10- and 20-Watt Powered Speaker Systems

Designed for home and portable recording, multimedia, MIDI studios and PA applications, the MS101II and MS202II pack lots of power and outstanding response in small packages. They feature 10 watts of power with a single 4″ full-range driver (MS101II) or 20 watts of power with two 4″ full-range drivers (MS202II) in a bass-reflex enclosure. Frequency response is 70Hz to 18 kHz. Each has multiple inputs, and offers comprehensive Low, High EQ and volume controls. Fully shielded, they can be used in close proximity to computer or video monitors without affecting the picture.





- Independent Low and High EQ controls allows sound to be tailored to the listener's requirements and the acoustic environment. The low end is enhanced by an internal EQ circuit that boosts the bass without muddying the sound.
- Full magnetic shielding allows use in close proximity to video or computer monitors
- The MS101*II* has two line inputs (RCA in the rear, 1/4″ mono phone jack in the front) as well as a 1/4″ mono output in the rear
- The MS202*II* has a 1/4″ mic input plus three line inputs (RCA in the rear, two 1/4″ mono phone jacks in the front) as well as a 1/4″ mono output in the front

MS60S

60w Powered Speaker System



Designed for a wide range of applications, including small sound reinforcement, and on-stage or studio monitoring, the 60-watt MS60S is the ideal "small-venue" portable amplification system. Multiple input configuration and controls permit direct and trouble-free connection with any setup.

- Completely portable, the MS60S has a built-in carrying handle and weighs only 22 lbs. It can also be placed horizontally or vertically.
- 60 watts of power with an 8″ low-frequency driver and horn-type high frequency driver. A sturdy metal grille protects the drivers.
- The amplifier comes with Active Servo Technology for a truly BIG sound from a small box
- Three inputs (balanced XLR mic/line level, and two unbalanced 1/4″ phone inputs) allow use with a mixer/line level source, guitar/bass/keyboard source, and Mic source... all at the same time
- Sound can be tailored using the two-band EQ circuit, giving you 5dB of cut /boost at the low end, and 8dB of cut/boost in the treble
- Volume levels can be adjusted for proper balance between all three sources using the Input 1, Input 2, and Master Volume Level controls

MSR100

8″ 100-watt Powered Speaker



A mixer, power amplifier, and speaker system combined in a single, compact unit, the MSR100 is perfect for small events and club gigs. Compact in size but big in sound and versatile features, the MSR100 packs a powerful sonic punch — making it ideal as a personal stage monitor; a utility speaker for electronic drums, keyboards, guitars, and vocals; or a small-scale FOH PA system.

- An 8″ custom woofer handles the low end, while a 1" pure titanium diaphragm compression driver delivers the high frequencies. Horn provides a wide 90° x 40° sound dispersion for optimum coverage.
- Built-in mixer provides 3 inputs (balanced XLR mic/line level with a -50dB/+4dB pad switch, and two unbalanced 1/4″ phone inputs)
- Each input has its own level control, with a Master Level control for overall output. Two-band EQ gives further control over the final sound. A CLIP indicator alerts you when the input signal is too high.
- Stand upright for conventional sound reinforcement, or lay it on its angled side for use as a floor monitor. A variety of mounting options let you place it on a pole, suspend it from the ceiling, attach it to a wall, or even safely stack multiple units on the floor.

SPEAKER ACCESSORIES

BAS10 Free Angle Clamp

The BAS10 frees up desk or counter space by elevating the speakers up to 14″ above the work surface. Its clamp allows horizontal mounting or it can be rotated 90° clamping to an upright on the desk or table. The ball and socket then positions the speaker at most any angle. The BAS10 Free angle clamps are sold as pairs, and include screws to attach them to Yamaha speaker systems.....**64.95**

BMS10A Mic Stand Adaptor

The BMS10A adapter mounts speakers of up to 11 lbs to a microphone stand. It fits the 5/8″-27 thread standard and comes with a 3/8″-16 European thread adapter. The BMS10A comes with two 5mm screws to attach Yamaha speakers equipped for mounting and is sold as a single unit.....**26.95**

BWS20 Series Wall/Ceiling Mounting Brackets

The BWS20 Series allow mounting of speakers weighing up to 15.4 lbs. Designed for use with the Yamaha MS10II, MS202II, MS20S, and MSP3, these brackets will also work with other speaker systems. Hardware is included.

Designed for maximum safety, all adjustment points are connected in a way that keeps them from coming apart, even if all the adjustment screws are missing! Screws to attach these brackets to the speaker are included. Additional hardware to attach these brackets to the wall or ceiling is required. Thick steel flanges on these brackets are both welded AND riveted to assure their strength.

BWS20-120: 4⅝″ **69.95**

BWS20-190: 7 1⁄16″ **72.95**

BWS50 Series Wall Mount Brackets

The BWS50 Series allow mounting of speakers up to 44 lbs. Designed for the Yamaha MS60 and NS10MC speakers, these mounting brackets can easily adapt to other speaker systems as well. Collars on both ends of the bracket allow 360° rotation. Additionally, the joint in the shaft of the bracket swivels over 200° (in 6-degree increments). Flexibility like this allows an endless possibility of mounting positions, in order to attain the desired coverage.

BWS50-190:
Wall Mount Bracket for MS60S (pair).....**124.95**

BWS50-260:
Wall Mount Bracket for MS60S (pair).....**139.95**



SM SPKR BRKT Multi-Angle Mount

This bracket provides a low cost solution for mounting smaller (up to 15 lbs.) speaker systems. An ingenious radiused mounting plate allows up to a 140-degree range of rotation. The wall and speaker components separate for ease in mounting to their respective surfaces before being joined. Two 5mm screws attach to Yamaha speakers equipped for mounting are included. Other speakers can also be accommodated. Includes steel cables for extra safety.....**49.95**

Varistand Height Adjustable Speaker Stands

The Varistand all steel design gives solid support for small monitor speaker systems. While designed to support the MSP5 monitors (with the AP5 adapter plate); the mounting platform of the Varistand accepts many different bolt patterns.

The base of the stand is hollow, allowing the addition of sand or lead ballast to increase stability. The ingenious T-shaped base has a very small footprint,allowing the user to place them into tight spots behind desk and consoles.

Stands are height adjustable with one knob, and extend from 27″ to 42″. Maximum recommended weight is 20 lbs. per stand **139.95**

STS-10 Speaker Stand

Supports loudspeakers on any flat surface. Unique ball and socket design allows the speaker to be secured at any angle. Two 5mm screws are provided for attachment of the MS101II, MS202II, MS20S and MSP3 .....**Call**

AP5 Steel Adapter Plate: All-steel plate permanently mounts the MSP5s (pair) to SM SPKR BRKT or the VARISTAND.....**24.95**

BFS60: MS60S Foldback Adapter **35.95**

BWS251-300:
Wall Mount Bracket for MSP10 (pair).....**129.95**

STS2:
Tilt Back Stand for MS101II and MS202II.....**24.95**

RK60: Rack Mounting Kit for MS60S. Permanently mount your MS60S in a standard 10″ equipment rack. Constructed completely of heavy gauge steel, it is designed to attach easily to the MS60S with screws that are supplied with the speaker **49.95**

UHF EarAmp

Wireless Personal Monitoring System



The UHF EarAmp brings professional in-ear-monitoring to a wide range of users. Offering superlative audio quality and extensive control, the EarAmp system includes individual volume and balance controls for the best possible mix, and separate bass and treble controls to fine-tune the signal and a limiter control to set the audio level and protect your ears.

Ideal for keyboard players, drummers and other more stationary performers who want the advantages implicit in personal monitoring, the EarAmp can be operated in stereo mode with all controls operating as labeled, or in the Mix mode with two independent mono signals with the balance control adjusting the relative volume of source 1 to source 2. In Mix mode, the mixed signal is presented to the headphones in mono. Also includes a 15 dB pad control to attenuates "hot" inputs, a headphone jack and all required cabling.

By eliminating the need for loud onstage "wedge" monitors, the EarAmp allows you to hear yourself better, greatly reduces feedback problems, and makes the overall sound coming from the stage significantly cleaner.

EarAmp is one of the most unique audio products to ever come to the stage. The system brings unparalled control and mobility to stage monitoring. Wedge monitors require a separate mixer and engineer. With EarAmp, artists can hear themselves better and fine-tune their mixes on the spot.

Overall stage volume and ambient noise are greatly reduced, eliminating feedback problems and promoting a cleaner front-of-house mix. EarAmp delivers a true stereo mix with discrete L/R replacement of channels. The resulting sound is more intelligible so performers can focus on the show—with a sense of intimacy never before possible. The ideal environment for performing.

EarAmp uses state-of-the-art PLL synthesized technology to optimize the desired signal while filtering out spurious emissions and noise. This highly developed circuitry ensures stable performance and reliability. Unparalled frequency agility allows each user to instantly select from over 128 channels, for flexibility with unquestioned performance—even in the most RF-hostile environments.

Diversity Wireless Monitor System

EarAmp's freedom of movement and reliability come in part from its diversity design. If you already use a wireless mic or instrument system you know that diversity reception is essential for ensuring signal reliability.

With its microprocessor controlled diversity design, UHF EarAmp's range is maximized. The user experiences fewer drop-outs, increased rejection of unwanted signals and crystal-clear audio fidelity.

UHF EarAmp offers an extraordinary level of frequency and agility with 8 groups of 16 channels each—128 total. Frequencies can be selected by group and channel, or by direct selection in 25 kHz increments. And a single transmitter will broadcast to an unlimited number of receivers.

Wireless EarAmp Channel Selection Chart (U.S. Frequencies)

Channel	Group 38	Group 39	Group 40	Group 41	Group 42	Group 43	Group 44	Group 45
1	614.35	620.35	626.35	632.35	638.35	644.35	650.35	656.35
2	616.10	622.10	628.10	634.10	640.10	646.10	652.10	658.10
3	616.50	622.50	628.50	634.50	640.50	646.50	652.50	658.50
4	617.40	623.40	629.40	635.40	641.40	647.40	653.40	659.40
5	619.30	625.30	631.30	637.30	643.30	649.30	655.30	661.30
6	620.00	626.00	632.00	638.00	644.00	650.00	656.00	662.00
7	614.70	620.70	626.70	632.70	638.70	644.70	650.70	656.70
8	615.05	621.05	627.05	633.05	639.05	645.05	651.05	657.05
9	615.40	621.40	627.40	633.40	639.40	645.40	651.40	657.40
10	615.75	621.75	627.75	633.75	639.75	645.75	651.75	657.75
11	617.00	623.00	629.00	635.00	641.00	647.00	653.00	659.00
12	617.80	623.80	629.80	635.80	641.80	647.80	653.80	659.80
13	618.15	624.15	630.15	636.15	642.15	648.15	654.15	660.15
14	618.50	624.50	630.50	636.50	642.50	648.50	654.50	660.50
15	618.85	624.85	630.85	636.85	642.85	648.85	654.85	660.85
16	619.65	625.65	631.65	637.65	643.65	649.65	655.65	661.65

UHF EarAmp

Receiver



The system operates in both Dual Mono and Stereo modes. This gives performers the choice to listen the way they want. The unique Dual Mono mode allows for blending of two discrete input signals, which are routed to both ears using the balance control to mix them. The dedicated volume control adjusts the overall level to suit the listener's needs. A center detent lets performers feel the setting if changes are required during the show. An adjustable Limiter manages signal levels, prevents overload and protects the listener's ears from aggressive transients.

- Independent Treble and Bass controls allow performers to achieve their own personal in-ear mix. There's even a Loudness switch for enhanced articulation when the unit is operating at low levels.
- Frequency selection is fast and easy. An exclusive feature allows the receiver to automatically reset to the correct frequency for precise channel-matching at the flip of a switch and the push of a button.
- The UHF EarAmp receiver runs for approximately 8 hours on a single 9v battery. An LED on the receiver alerts you to low battery levels.

Transmitter

UHF EarAmp's transmitter features dual inputs for a choice of stereo or an additional mono signal in the performer's mix. The level is independently adjustable for each input. A 5-segment LED meter tracks the input level for optimum gain staging. In addition, an easy to read backlit LCD displays the selected Group, Channel and Frequency.

- The transmitter provides monitor engineers with the flexibility they need. Extensive input/output capabilities allow for inserting custom sub-mixes, adding effect returns or daisy-chaining transmitters.
- Neutrik Como (XLR or 1/4˝) inputs also provide for a range of signal levels and sources. A switchable 15dB pad is provided on both the main and auxiliary inputs.
- The transmitter is portable as well as rack mountable

Buds

Samson's high-quality flesh-colored ear buds provide the perfect match for UHF EarAmp's performance, allowing a slight amount of stage ambience into the mix. However, the system is universal and allows for the performer's choice of any standard third-party earpiece for total freedom of choice.

SYSTEM

Frequency Coverage	774.1 to 864.5 MHz
Channels Available	182 in 13 groups, country dependent. Direct frequency selection to nearest 25 kHz
RF Range:	Approximately 300´
System Modulation:	FM, Multiplex Stereo. ±50 kHz deviation of main carrier
Noise Reduction:	Samson compandor system
Audio Frequency Response:	40 Hz to 15 kHz
Audio Distortion:	0.8% nominal
S/N Ratio:	>80dB (>50uV at receiver, stereo or >15uV at receiver, mono)
Stereo Channel Separation:	40 dB nominal

TRANSMITTER

RF Output	10 mW ERP
Main Audio Inputs (L & R)	
Connectors	Neutrik Combo (XLR or 1/4˝ phone) jacks
Impedance	40K Ohms, balanced or 20K Ohms, unbalanced
Attenuator	Switch selected, 0 or – 15 dB
Level	- 10 dBu nominal, + 11 dBu maximum (attenuator out) + 4 dBu nominal, + 25 dBu maximum (attenuator in)
Daisy-Chain Out (L & R)	
Connectors	1/4˝ phone jacks
Electrical	Wired in parallel with main audio inputs
AUX Audio Inputs (L & R)	
Connectors	1/4˝ phone jacks
Impedance	40K Ohms, balanced or 20K Ohms, unbalanced
Attenuator	Switch selected, 0 or –15 dB – dedicated to the AUX inputs
Level	- 10 dBu nominal, + 11 dBu maximum (AUX attenuator out) + 4 dBu nominal, + 25 dBu maximum (AUX attenuator in)
Limiter	Limits maximum modulation to comply with regulatory limits Permits overdrive up to 20 dB with low distortion
Power	12 VDC, 500 mA from supplied AC adapter

RECEIVER

Audio Output Power	200mW/CH, 1% THD into 32Ω load, 9 VDC 120 mW / CH, 1% THD into 32 Ohm load, 7 VDC 70 mW / CH, 1% THD into 32 Ohm load, 6 VDC
Audio Output Limiter	Adjustable from >200 mW to <50 mW into 32Ω
Low Frequency EQ	±10 dB @ 150 Hz
High Frequency EQ	±10 dB @ 5000 Hz
Loudness Switch	Flat or Loudness contour, selectable with slide switch
Battery Requirement	7 VDC to 9 VDC for full spec operation up to 120 mW audio output. 6 VDC to 7 VDC for reduced spec operation. 9 V Alkaline or Alkaline Plus (Duracell MN1604 – Ultra is suggested)
Low Battery LED	Illuminates when voltage drops below 7 VDC
Battery Life	4 hours typical – volume level dependent

PSM SERIES

Personal Monitors for Installed Sound

Since their introduction, Shure PSM systems have become known as the best wired/wireless monitors available for sound quality, flexibility and affordability. What's more, they are simple to use, conserve space on stage, and are easy to transport. They deliver a full frequency range and great stereo separation. An impressive signal-to-noise ratio minimizes ear fatigue while increasing fidelity. This allows performers to concentrate better on their performance because they can now hear their mix more clearly at lower volumes. The audience will hear a better performance, too, because no wedges means no feedback can occur when users (intentionally or unintentionally) alter the position of the floor monitors relative to the microphones, and a cleaner mix for the room.

Unlike conventional floor monitors, Shure's PSM systems function as a closed system. The mix is sent directly to the performer's ear, so there is no chance of the monitors coloring or competing with the front-of-house speakers. The difference in perceptible sound quality is truly amazing. All PSM systems feature mono, stereo and MixMode operation. MixMode gives performers greater control over their monitor mix by turning the receiver into a 2-channel mixer.

MIXMODE

In MixMode, the performer receives two discrete channels of audio, just like in stereo. The difference is that each channel is sent to both the left and right earphone. The performer hears the two signals mixed together in each ear.

By adjusting the balance knob on the body-pack, performers can control the relative level of each signal, making one softer and the other louder in both ears. For example, a performer may choose to have their vocals (or instruments) sent through one channel and the rest of the band sent through the other. Using the balance knob on the receiver, the performer can determine exactly how much of themselves they want to hear in the mix. Musicians, in particular, enjoy the flexibility of MixMode because it gives them the ability to adjust their mix from one song to the next without having to signal the engineer at the soundboard. This means they can have exactly the right balance for every song with a minimum of fuss and distraction. Other possible MixMode configurations:

- One channel is lead instruments and the other is the rhythm section
- One channel is the bands monitor mix and the other is crowd noise
- One channel is the musician's mix and the other is a click track

To meet the budget requirements for a wide range of installations, Shure offers a full line of PSM systems in both wireless and hardwired configurations:

PSM 200

Affordable Monitoring - Get In Ear From Here

Affordable, full featured entry to personal monitoring. Whether you're playing live or rehearing, you need to hear yourself to be your best. The PSM 200 is for performers who want to start out In Ear and stay there.

PSM 400

Versatile Systems for Working Musicians

Clear, consistent monitor mixes. Every Day. Every Night. This addition to the PSM family offers user-friendly features such as LCD controls and a battery fuel gauge. The wireless system is frequency agile with 16 UHF frequencies (8 compatible).

PSM 600

Professional Wired and Wireless Systems

Wired and wireless personal monitoring for professional installations and touring sound. The wireless PSM 600 system offers two-channel selectable UHF operation with th ability to use up to 10 systems simultaneously. A hardwired unit is also available for applications where wireless mobility is not a consideration.

PSM 700

Advanced Wireless Touring Gear

Unsurpassed wireless personal monitoring for professional installations and touring sound. With two groups of 16 compatible UHF frequencies, the PSM 700 gives system engineers the power to create up to 16 stereo mixes simultaneously - an important feature for larger installations with many performers who want their own mix.

PSM200

Personal Monitor Systems



The PSM200 is a frequency-agile, in-ear monitor system for use in a wide variety of applications. Its many features help overcome the difficulties associated with stage monitoring:
The P2R Receiver is the world's first bodypack receiver to function as a wired receiver, a wireless receiver, or both at the same time.
The P2T TransMixer is another breakthrough design for in-ear monitoring. It allows you to connect two mic or line-level inputs directly into the transmitter. The two signals can then be mixed independently. You can connect microphones or instruments directly to the inputs, use line outputs from a mixing console, or connect it to Shure's Aux-Pander. No matter what the output device, you can probably plug it into the P2T TransMixer.
The complete PSM200 System includes the E2 Earphones, which feature a dynamic driver that produces crisp, full-spectrum sound. They include a variety of sleeve options to provide the best fit and isolation for your ears. The E2, like all Shure earphones, work well with CD and MP3 players, and anything else with a 1/8˝ (3.5mm) stereo output.

System Features

The PSM 200 makes the benefits of in ear monitoring accessible to a wide array of cost-conscious musicians and houses of worship with exciting features including the built-in ability to upgrade the receiver from wired to wireless, a transmitter that is also a mixer, and a high fidelity earphone. Wired or wireless systems feature integrated mix control with two mic/line XLR-1/4˝ combo input channels plus XLR split outputs that connect signals to a mixing console or other device without affecting original settings. The frequency-agile wireless transmitter features eight channels per system (up to four simultaneous compatible systems).

P2R Hybrid Receiver

For those that can't initially afford wireless, the P2R Hybrid Receiver protects their investment by allowing them to upgrade at a later date. The hybrid receiver starts as a wired bodypack and with the addition of the Transmixer, operates fully as a wireless receiver.

- Dual-function receiver operating as a wired and wireless bodypack
- Volume control, and a built-in limiter that can't be defeated.
- LED indicators for power, RF signal strength, frequency, limiter, and battery level.
- Integrated cable management system keeps cables under control
- A 1/4" line input jack for convenient, integrated line input to connect monitor mixes, click tracks, or effects
- Up to 6 hours, volume dependent, on a standard 9V battery.

P2T TransMixer

By combining frequency-agile wireless transmitter functionality and a two-channel mixer, the P2T TransMixer offers you complete control of the in ear mix right on the stage

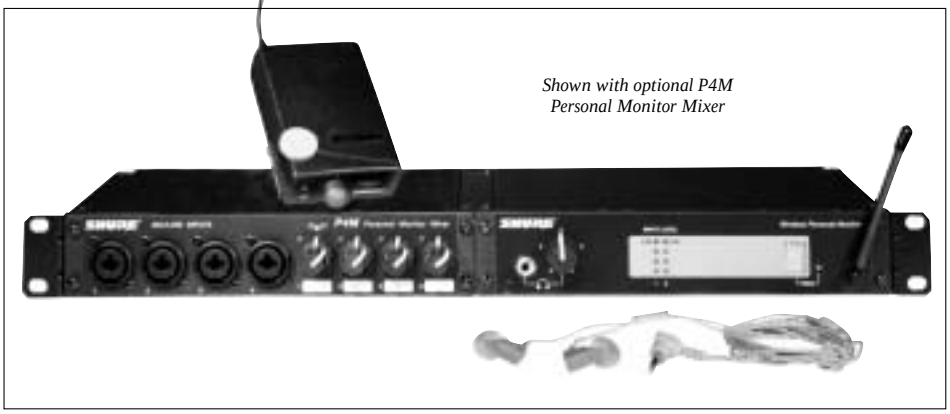
- A frequency-agile transmitter with eight channels per system.
- Up to four simultaneous compatible systems (country dependent).
- A range of 100 m (300 ft.) under optimal conditions.
- LED indicators for input level and frequency.
- Volume controls for mic/line inputs helps you create your own mix.
- Integrated mix control with two mic/line XLR-1/4˝ combo input channels for personal control.
- XLR split outputs that connect signals to a mixing console or other device without affecting original settings.

E2 Earphone

- In ear, high fidelity sound Isolation to help you keep what you hear on stage under your control Interchangeable flex and foam sleeves to give you the perfect fit.

PSM400

Versatile Personal Monitor Systems



Shown with optional P4M Personal Monitor Mixer

The PSM 400 makes professional quality, personal monitoring available to more performers than ever before. Designed for performers who are tired of the limits of floor wedges, the PSM400 offers hands-on control and freedom to musicians who have a limited budget for gear. With a wide range of available options - from complete wireless and wired systems to individual components - there is a PSM 400 that is right, rehearsal through performance, for almost anyone who performs live music. The PSM400 offers a personal mix free of unwanted stage sounds; individual volume and balance controls; a freedom of movement that releases performers from standing in only one spot on stage; and an ease of portability far greater than bulky floor wedge monitors. Furthermore, the PSM series, when used properly, can aid performers in hearing conservation. These many benefits are further enhanced by the P4M mixer, which brings complete control of the in-ear mix right to the stage. Setting the standard for price and performance, PSM400 systems are available in both wired and wireless versions and offer user-friendly features such as LCD controls and a battery fuel gauge. The wireless system is frequency agile with 16 UHF frequencies (8 compatible).

P4HW

Wired Bodypack Receiver

- Attached input cable
- High-impact plastic chassis
- Operating controls include operating mode (Stereo, Mono, MixMode), volume and balance controls, defeatable limiter, high-frequency boost , -15 db Pad
- Backlit LCD display shows battery life, input levels and operating mode

P4R

Wireless Bodypack Receiver

- High-impact plastic chassis
- Frequency locator
- Operating controls include operating mode (Stereo, Mono, MixMode), volume and balance controls, defeatable limiter, high-frequency boost, and frequency select
- Backlit LCD display shows RF reception, battery life, operating mode and frequency

P4T

Wireless Transmitter



- Detachable front-mounted antenna
- 1/2 rack space
- Built-in audio limiter
- Controls include operating mode (Stereo, Mono, MixMode), earphone volume and frequency select
- 4-segment input level and transmission frequency LED indicators
- Two 1/4″ TRS line level inputs; two 1/4″ TRS loop outs and local earphone monitor
- External power supply

PSM 400 SYSTEM CONFIGURATIONS

P4MHWE1
Wired Personal Monitor System with Onstage Mix Control

P4M
Personal Monitor Mixer

P4HW
Wired Bodypack

E1
Single-Driver Earphones

P4TRE1
Wired Personal Monitor System

P4T
Wireless Transmitter

P4R
Wireless Bodypack Receiver

E1
Single-Driver Earphones

P4MTRE1
Wired Personal Monitor System with Onstage Mix Control

P4M Personal Monitor Mixer

P4T Wired Bodypack

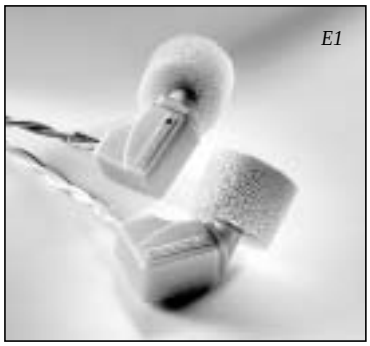
P4R
Wireless Bodypack Receiver

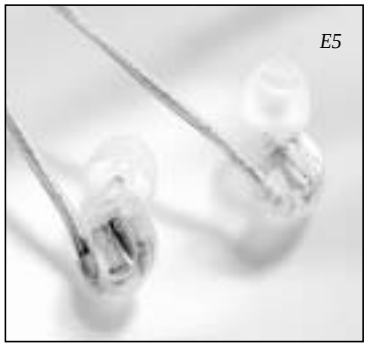
E1 Single-Driver Earphones

PSM400

E1 and E5 Earphones—In Ear Comfort and Clarity

Unobtrusive, low mass/high energy earphones, the E1 and E5 are designed specifically for use with Shure PSM systems and deliver the incredible sound and performance ear buds and other headphones can't. Don't go In-Ear without them. The E1 and E5 earphones provide numerous advantages over earbuds. Along with their high fidelity, full-bandwidth transducers, the natural design of these stereo earphones allows them to rest comfortably in the ear canal, providing a tight seal for sonic isolation and a snug fit. Sound quality and isolation are the two most important things to consider when choosing an earphone. The E1 and E5 provide superior sound quality due to their high fidelity, full-bandwidth transducer design. Virtually the same, the E5 is a special dual-driver earphone that delivers an extended frequency response resulting in truly incredible sound. They have a universal 1/8˝ plug that connects to Shure's PSM body pack receivers, as well as many other personal audio applications, and they can be used with foam, flex or triple flange accessory sleeves (provided).





E1 Single-Driver Earphones	
Transducer Type:	low mass/high energy
Sensitivity (at 1 kHz):	113.5dB SPL/mW
Impedance (at 1 kHz):	29 ohms
Output Connector:	Gold-plated stereo, 3.5 mm (1/8˝) phone plug
Cable Length:	60˝

E5 Dual-Driver Earphones	
Transducer Type:	Dual low mass/high energy
Sensitivity (at 1 kHz):	122dB SPL/mW
Impedance (at 1 kHz):	110 ohms
Output Connector:	Gold-plated stereo, 3.5 mm (1/8˝) phone plug
Cable Length:	61˝

Earphone Sleeves

Each choice provides comfort and isolation in a variety of sizes and materials for the right fit.

Reusable Plastic Flex Sleeves (5 pairs):

PA756S - Small ..**15.90** **PA756M - Medium** ..**15.90** **PA756L - Large** ..**15.90**

PA750 Cylindrical Foam Ear Sleeves (5 pairs)..**31.83**

PA755 Triple Flange Ear Sleeves (1 pair) **15.90**

dbx IEM

In-Ear Monitor Processor with Digital Multi-band Compressor/Limiter/Parametric EQ/PeakStop Limiter/Lexicon Reverb



A digital signal processor specifically designed for in-ear monitor applications, the IEM is the ideal tool for fine-tuning mix monitoring for in-ear monitoring devices. Not only does the IEM provide the legendary dbx effects in the digital domain, such as 4-band stereo compression, PeakStop limiting and 5-band parametric EQ, but the IEM also includes stereo adjust, dbx proprietary Type IV conversion system, and for good measure, custom reverb algorithms using industry-standard Lexicon technology.

- In addition to an intuitive user front panel which provides instant access to all effects at the push of a button, and a large custom display which shows all operational information in a logical manner—all parameters and features can be controlled by the bundled software. On-board IEM firmware is updateable via the web and the built-in RS-232 port.

PSM600

The PSM 600 features crystal-controlled analog operation. Ten personal stereo mixes can be created using as many wireless transmitters. Or, by using only PSM wireless transmitter along with several PSM 600 wireless receivers, bands using a single monitor mix can have a full professional monitor system for a fraction of the usual cost. For musicians who maintain a single position throughout the performance (seated performers such as drummers and keyboard players), PSM 600 wired body packs are the perfect choice, allowing a direct patch from the sound board.



P6HW
Wired Bodypack Receiver

- Detachable input cable
- Rugged metal chassis
- Controls include mode (stereo, mono, MixMode), volume and balance, defeatable limiter, high-frequency boost,-15 db Pad
- LEDs show peak input, battery life

P4R
Wireless Bodypack Receiver

- Rugged metal chassis
- Frequency locator
- Controls include mode (stereo, mono, MixMode), volume, balance, defeatable limiter, high-frequency boost, frequency select
- LEDs show RF reception, battery life

P4T Wireless Transmitter

- 1/2 rack space, built-in audio limiter
- Remotable/combinable antennas
- Controls include input knob, mode (Stereo, Mono, MixMode), frequency select
- 8-segment input level and transmission frequency LED indicators
- Two 1/4″ TRS line level outputs, earphone monitor



PSM700

If you need more than 10 personal mixes but don't want to compromise wireless mobility, the PSM 700 is synthesized, giving you greater frequency agility. Plus its sound quality surpasses all other personal monitors on the market. With 32 compatible UHF frequencies, the PSM 700 gives system engineers the power to create up to 16 stereo mixes simultaneously—perfect for larger installations with many performers who want their own mix.

- Frequency Locator Mode reduces set-up time by allowing the installer to locate the optimal frequencies using the receiver as a tool.
- Dynamic Overload Control automatically protects the system from distortion that can occur when a performer gets too close to the transmitter.
- PSM 700 antenna combiners can be linked into one dedicated combiner to minimize the number of antennas used during the performance.

P7R
Wireless Bodypack Receiver

- Rugged metal chassis
- Frequency locator
- Dynamic overload protection
- Controls include mode (stereo, mono, MixMode), volume, balance, defeatable limiter, high-frequency boost, channel selector and group select switch
- LEDs show RF reception, battery life

P7T Wireless Transmitter

- 1/2 rack space
- Remotable/combinable antennas
- Controls include input knob, mode (stereo, mono, MixMode), earphone volume, channel selector and group select switch
- 8-segment input level and frequency group LED indicators
- Two 1/4″ TRS/XLR combo line level inputs; two 1/4″ TRS loop outs, earphone monitor

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PSM SERIES AT A GLANCE

	PSM 700	PSM600	PSM 400	PSM 200
WIRELESS SYSTEM				
Number of Selectable Channels	32	2	16	8
Number of Compatible Channels	16	10	8	4
Output Level	+6 dBV @ 0.9% THD+N	+6 dBV @ 0.9% THD+N	+3 dBV @ 0.9% THD+N	
Signal to Noise Ratio	78 dB	78 dB	80 dB	80 dB
Stereo Separation	35 dB	35 dB	35 dB	
WIRELESS TRANSMITTER				
RF Output Power	100 mW	100 mW	50 mW	30 mW
Frequency Selection Method	Front panel rotary switch with slide switch	Front panel rotary switch	Recessed, momentary button	
Stereo/Mono Selection Method	front panel switch	front panel switch	automatic	
Input Level Control	Front panel knob	Front panel knob	None	Front panel knobs
Audio Metering	8-LED string	8-LED string	4-LED string	Tri-color LEDs
Local Headphone Amplifier	1/4″ and 1/8″ jack	1/4″ and 1/8″ jack	1/8″ jack	
Antenna	Front or rear mounting	Front or rear mounting	Front or rear mounting	Attached, front mounted
Input Connectors	XLR\1/4″ combo (line)	XLR\1/4″ combo (line)	1/4″ TRS combo (line)	XLR\1/4″ combo (mic/line)
Loop Thru Outputs	1/4" TRS	1/4" TRS	1/4" TRS	XLR split outs
Limiting	Built-in	Built-in	Built-in	Built-in
Power Supply	110/220V 3-prong IEC	110/220V 3-prong IEC	Locking, external (PS40)	External (PS20)
Rack Mounting	1/2 rack (hardware included)	1/2 rack (hardware included)	1/2 rack (hardware included)	1/2 rack (hardware included)
Input Pad	Yes	Yes	No	
WIRELESS RECEIVER				
Power Supply/Battery Life	9V/4-6 hours	9V/4-6 hours	9V/4-6 hours	9V/4 hours
Antenna	Detachable whip Lemo threaded	Detachable whip Lemo threaded	Attached whip bottom exit	Attached whip
Battery Status Indicator	Low battery LED	Low battery LED	Fuel Gauge & LEDs in LCD	Battery level LED
Volume Control				
Limiting	Built-in	Built-in	Built-in	Built-in (non-detachable)
Equalization	10 kHz shelf	10 kHz shelf	10 kHz shelf	None
Frequency Selection Method	DIP switch & rotary switch in battery compartment	DIP switch in battery compartment	Push button	
Personal Mix Control	MixMode and panning/balance knob	MixMode and panning/balance knob	MixMode and panning/balance knob	
RF Indicator	LED	LED	LED	
EARPHONES				
Model Number	E5	E1	E1	
Drivers (per ear & type)	2 balanced armatures	1 balanced armature	1 balanced armature	
Universal Sleeves	foams, triple flanges, flex tips			
Custom Sleeves	custom sleeves available			
ACCESSORIES	4x1 antenna combiner, unidirectional antenna for remote mounting	4x1 antenna combiner, unidirectional antenna for remote mounting		

P4M

Four-Channel Personal Monitor Mixer



A highly versatile four-channel, two-bus stereo mic/line mixer designed to provide greater control and flexibility in monitoring situations, the P4M interfaces with both personal monitor and floor monitor systems. For the performer, the P4M offers the ability to make quick adjustments without having to go back to the main mixer. For the installer, the P4M's routing capabilities make it a great solution not only for monitoring, but for any application that can benefit from a convenient two-bus mixer. The P4M features four balanced mic/line inputs, each with level and panning controls. Corresponding XLR split outputs allow input signals to pass through the unit unaffected, while providing an independent mix.

Personal Stereo Monitors

When coupled with personal monitors, the P4M gives performers the ability to adjust personal mix levels onstage - even during a performance - without affecting the front-of-house system. This enables users to create and maintain consistent monitor mixes, regardless of the venue.

Installed Sound Systems

The P4M can also be used as a small "piggy-back" mixer in applications such as houses of worship, or courtrooms. The unit acts as a console bypass when a permanent installation of a large, operator-controlled mixing console may require just a few microphones for smaller events.

Given its dual-bus output capabilities, the P4M can also serve as a "two zone" mixer, routing any combination of mic or line-level signals to two different zones.

Each mic/line input splits internally, so you can "tap into" existing signals, control them the way you want, and let the original signals pass through uninterrupted to their intended destinations.

The P4M has a pair of 1/4″ auxiliary inputs that can be used to add channels from other sources. For example, multiple P4M mixers can be linked together to increase the number of inputs. In addition, sources not present in the original mix - such as a click track for live percussion - may be included via the auxiliary inputs.

Applications

Created to provide greater control and flexibility in monitoring situations, the P4M's routing capabilities also make it a great solution for any application that can benefit from a convenient 2-bus mixer:

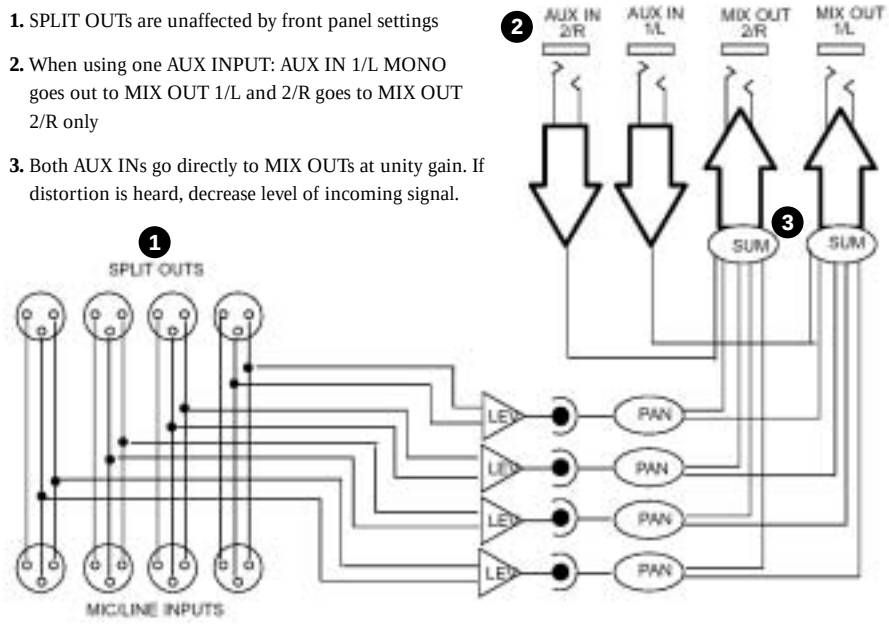
- Capturing a secondary mix for recording or broadcast feeds
- Sending and controlling mic/line level inputs to an existing mix
- Sending ambience or other signals to a remote location such as a "cry" room
- Creating a small "piggyback" sub mixer for activities where a live engineer is not viable
- Enabling director's cues and interruptible foldback

AUDIO SIGNAL PATH FOR THE P4M MIXER

1. SPLIT OUTs are unaffected by front panel settings

2. When using one AUX INPUT: AUX IN 1/L MONO goes out to MIX OUT 1/L and 2/R goes to MIX OUT 2/R only

3. Both AUX INs go directly to MIX OUTs at unity gain. If distortion is heard, decrease level of incoming signal.



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ONE HOUR FREE PARKING
AT 349 W. 34th STREET (with purchase of $100 or more)

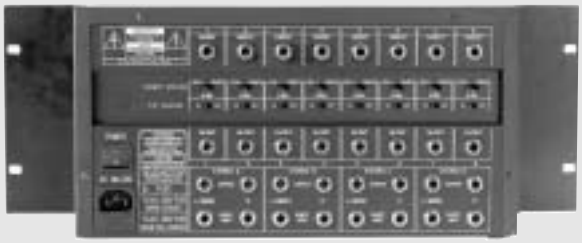
AUXPANDER

Auxiliary Expander: 8x8 Insertable Matrix Mixer



Traditionally, the number of auxiliary sends available on mixing consoles has been sufficient and proportional to the number of inputs needed for a given application. However, with Personal Monitors (PSM) rapidly becoming the monitoring solution of choice for live performers, the demand for auxiliary outputs to serve those systems can quickly outstrip the supply on most small to medium-size mixing boards. The AuxPander is an auxiliary expander designed as a flexible and cost-effective way to address this issue. With an AuxPander, you can boost the output abilities of existing mixing consoles to eight busses without investing in a dedicated monitor console or bulky, complex large frame mixing board. An easy-to-use, 8 x 8 analog mixing matrix that will grow and adapt with your system, the compact and transportable AuxPander is capable of mixing eight input signals and distributing them to any of the eight available outputs, in any combination, and at any volume level.

- Offers endless options for mix and signal path creativity. Craft personal monitor mixes for an individual or an entire band. Quickly and easily add users and resources to your mix—night after night, gig to gig.
- Strengthens every link in the production chain. House sound or recording engineers can easily accommodate special recording and monitoring requests. Bands traveling with an AuxPander can ensure a uniform monitoring environment from a rehearsal to a gig. No matter what mixing equipment awaits, just plug and play.
- Use in conjunction with modular digital multitrack recorders (MDMs) for live and studio recordings.
- Combine input signals from multiple sources and route them to a single output.
- Inserted between a MDM and a mixing console, the AuxPander can mix to individual stereo cue mixes without interrupting the signal flow to the mixing console.
- Create stereo auxiliary sends to incorporate the stereo inputs of effects processors.
- Mix to eliminate feedback in certain distributed installations where each monitor receives a different mix by subtracting the source closest to each monitor.
- AuxPander is expandable both horizontally (16 x 8) and/or vertically (8 x 16) if more output mixes are required.



AuxPander taps up to eight input signals from a standard mixing console through its insert points. it then provides up to four stereo or eight mono auxiliary mixes and a new insert point without disrupting the levels established at the front of house soundboard.

- Balance controls let you pan the input signals between the left and right channels of a stereo output
- Level controls let you increase or decrease the volume of each of the eight input signals. Route signals to any of the eight outputs at any level and in any combination.
- With direct input jacks you can expand input channels (daisy-chaining) while keeping the same number of mixed outputs. For example, connect one AuxPander to another to create a 16x8 matrix.
- Four pairs of balanced, line-level 1/4˝ output jacks carry the mix created
- Insert jack lets you integrate processors (EQ, compressors, gates, etc.) in a particular signal path. Once a processor is connected, it can affect the AuxPander inputs, the console inputs, or both via the Insert Assign Switch.
- In Global Mode both the AuxPander and the mixing board receive the processed signal. In Local Mode only the AuxPander receives it and the console's inputs are unaffected. In Remote Mode only the mixing board receives it while the AuxPander inputs are affected.
- A single jack, rather than dedicated "in" and "out" jacks, receives and returns signals from the mixer, eliminating the need for Y-cables. One conductor on a 1/4˝ TRS cable jack receives signals. The other returns them. So connections of a console's insert points can easily be made with a TRS-TRS cable.
- When using 1/4˝ TRS cables, some consoles use the tip conductor as the send (tip-send) and others use the ring conductor (ring-send). AuxPander in/out jacks are compatible with both via the Tip Assign Switch.








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The Professional Audio SourceBook

Section 8
Headphones

Headphones

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HEADPHONES

K 55



The K 55 stand out from "good value" headphones by their truly exceptional price/performance ratio. These big, elegant headphones provide both excellent comfort and superior sound. Large 1.6˝ "Turbo Bass" transducers deliver an uncommonly powerful bass range and a clean high end for unparalleled listening pleasure. The closed, circumaural earphone design adds to low-end power. The unique AKG styling makes the K 55 an eye-catcher while low weight, soft circumaural leatherette ear pads, a self-adjusting headband, and flexibly suspended earphones ensure a secure, comfortable fit.

K 66



Elegant, genuine Hi-Fi stereo headphones, the K 66 combines excellent wearing comfort with superior sound and optimum price/performance. With the proven AKG self-adjusting headband, soft circumaural leatherette ear pads, low weight, and flexibly suspended earphones, the K66 conforms perfectly to your head for a comfortably fit. Unique styling makes the K 66 an eye-catcher and the semi-open earphones provide an airy sound. AKG "Natural Sound" technology uses large, high performance 1.6˝ transducers to ensure superior sound quality for a perfect listening enjoyment.

K 70



The K 70 is a lightweight monitor quality headphone that offers many of the features found in AKG's professional format headphones. Modern lightweight open back headphone with full bandwidth reproduction, self adjusting headband, replaceable and washable ear pads—all in a semi-professional cost effective format. The K70 offers premium audio quality at a price anyone can afford.

K 100



A brilliant sound performance, recyclable material construction, comfortable self-adjusting headband, changeable and easy-to-clean ear pads, changeable ASD filter, and an exclusive inner leather headband make the K100 an outstanding headphone.

K 270S



Closed design, excellent isolation and low sound leakage. The patented Parabolic Transducer Array in the K 270S has two optimized transducers in each earcup, both precisely configured in a tuned enclosure to focus sound waves at the entrance to the ear. The result is a dramatically increased perception of spaciousness, extremely accurate response, and no audible distortion at output levels far beyond the capability of single transducer headphones. This, combined with the enclosed design, makes the K 270S an ideal choice in situations where quality monitoring is required in competition with high ambient levels, such as for live or studio drummers. A micro-switch built into the headband disconnects the audio circuit whenever the headphones are removed, preventing potentially damaging feedback or leakage when "hot" headphones are suddenly exposed to open microphones.

HEADPHONES

K 301

In the K 301, the oscillatory behavior of the diaphragm has been completely redesigned, with high internal attenuation suppressing unwanted partial oscillations. This has been made possible by a very finely structured diaphragm sound zone. Music lovers of any style will enjoy the highly accurate, balanced tonal response. Self-adjusting headband.

K 401

The K 401 provides everything rock and pop music fans want to hear: tight, powerful bass; crystal-clear sound quality; and impressive power. The low frequency performance is enhanced by the Varimotion System technology, resulting in a powerful overall sound. Self-adjusting headband.

K 501

The K 501s are concert hall headphones designed for lovers of classical music. The unique Varimotion System technology provides perfect spatial separation and an outstanding sound resolution, resulting in a superbly defined sound. Self-adjusting headband.



	K55	K66	K70	K100	K270S	K301	K401	K501	K1000
Type	Sealed, Circumaural, Dynamic	Sealed, Circumaural, Dynamic	Semi open, Supra aural	Semi open, Supra aural	Sealed, Circumaural,	Open-air, Circumaural Parabolic	Open-air, Circumaural	Open-air, Circumaural Dynamic	Dynamic Transducer w/ VLD magnet
Frequency Range	16 - 20k Hz	18 - 22k Hz	20 - 20k Hz	20 - 28k Hz	20 - 28k Hz	20 - 25k Hz	18 - 28k Hz	16 - 30k Hz	
Sensitivity	96 dB/mW	96 dB/mW	105 dB/mW	103 dB/mW	92 dB/mW	94dB/mW	94dB/mW	94dB/mW	74 db/mW (free field)
Rated Impedance	32 Ω	32 Ω	100Ω/system	100Ω	75Ω	120Ω	120Ω	120Ω	120Ω
THD:	<1%	<1%	<1%	<1%	<0.4%	<1%	<1%	<1%	
Weight (without cable)	7.4 oz.	7.4 oz.	7.8 oz.	6.7 oz.	9.5 oz.	8.1 oz.	8.3 oz.	8.3 oz.	9.5 oz

K 1000 *Automated Reference Listening System*



A standard for music lovers and sound purists throughout the world

Our sense of hearing detects depth and directional information in sound via a complex series of comparisons of what is heard by the two ears. Differences in sounds arrival time, as well as phase and frequency response, are used to determine where the sound is coming from. A stereo pair of loudspeakers provide this directional information for the listener. Conventional headphones make this process impossible; sounds appear on each side, but with no sense of space or depth. The K1000 was the first commercial headphone to produce the experience of proper depth and spatial orientation.

Highly accurate, close field reference monitor, the K1000 features a unique open air, radial magnet transducer to produce total acoustic transparency. Speakers are angled in parallel with the ears and do not touch them. Use as confidence monitors in acoustically inferior edit rooms. Standing waves, comb filtering, and reverberant fields are deleted from the mix perspective. Its open binaural design will not produce “in head localization”. Natural time delay from the speaker is heard in each ear, providing proper depth and spatial orientation.

STUDIO HEADPHONES

AKG introduced the K 140 in 1974 and the K 240 Monitor in 1975, which became the most widely used headphones in recording studios and broadcast facilities worldwide. Generations of musicians and audio engineers use these AKG products to create great mixes,hit songs and albums cherished by millions around the world.
The new AKG Professional Studio headphones carry on the tradition of these audio tools and are a sound investment that will pay handsome returns with years of flawless sonic performance for professional recording engineers and demanding users of high quality home or portable audio equipment.

New XXL Transducers with Varimotion Diaphragm



The new XXL transducers use AKG's patented Varimotion diaphragm to ensure extremely accurate response at any listening level. This technology creates a diaphragm that is thinner, more elastic toward the outside edges for more powerful bass response and thicker, less resonant in the center for pure, accurate mids and clear, detailed highs. The large size of the transducers makes the overall sound more lifelike and three-dimensional to capture the emotional and sonic content of your soundtracks.

Gimbal-type Suspension

A gimbal-type suspension makes sure the earcups of both supra aural and circumaural models will conform to the user's ears without pressing on them. Thanks to this dynamic adjustment mechanism, you can wear the headphones for hours without feeling pinched by them.

Detachable Cable

The headphones utilize a detachable input cable for easy replacement if ever necessary. The cable and headphones use mini XLR connectors for perfect contact and secure connection. The cable is made of 99.99% oxygen-free copper and has gold-plated plugs on both ends including the screw-on 1/8″ to 1/4″ jack adapter for maximum signal transfer.

Self-Adjusting Headbands

To ensure your listening comfort, the Professional Studio headphones utilize a self-adjusting headband with a flexible steel structure to deliver optimum fit for your headphones without any fiddling. The structure is flexible, yet strong enough to allow bending for single ear use and is durable enough to withstand rough handling. A gimbal-type suspension ensures the earcups of all Professional Studio models will conform to the shape and size of your ears without pressing on them for hours of fatigue-free, comfortable listening.

Rugged construction,a non-twisting headband, and powerful sound allow you to use your headphones on a single ear only or wear them in any other way you may prefer. AKG Professional Studio headphones are ideal for DJ and live sound use because they are durable enough to withstand the roughest kind of handling during transport or on stage.

Perfect for Many Applications

You need good quality audio for all your listening applications– including computer, Minidisc, portable CD or DAT player – as well as in the studio. Owing to their high sensitivity and wide dynamic range, the Professional Series headphones are efficient enough to be used with these lower level output sources and still deliver the high definition sound you need when working in the studio or on stage.

For applications that require the performance and 600-ohm impedance of the original models,the famous K 141 and K 240 Monitors are still available with their fixed cables.

K 141 Monitor

A proven "workhorse" throughout the recording industry, the K 141M serves as the reference standard for supra-aural (on-the-ear) monitor headphone designs. A small coupling volume in the ear cup, combined with an "integrated semi-open" venting system behind the transducer, gives the K 141 M its trademark sound: punchy, accurate and crisply defined. The K 141 M will deliver high sound pressure levels without distortion and it is solidly built to withstand around-the-clock session use.

K 240 Monitor

A classic design which has become a standard in monitoring. The "integrated semi-open air" design incorporates a highly accurate dynamic transducer, and an acoustically tuned venting structure behind the element that produces a naturally open sound quality. The circumaural (around-the-ear) pads are designed for hours of wearing comfort, while the steel-cable, self-adjusting headband construction and single-sided pro quality audio interconnect ensure extended reliability in demanding studio applications.



STUDIO HEADPHONES


K 141

K 141 Studio

The K 141 Studio is an updated version of the classic AKG K 141 Monitor, one of the world 's most popular models. Lightweight and comfortable, easy to use and listen to, the K 141 Studio features new XXL transducers with lower impedance for higher sensitivity and a more powerful sound. Supra-aural (on-the-ear) design combined with a semi-open vented system gives the K 141 Studio its trademark sound: punchy, accurate and crisply defined.

K 171 Studio

Designed specifically for monitoring in noisy environments, the K 171 Studio combines the benefits of a closed-back design with the lightweight and comfort of supra-aural headphones—an excellent choice for DJ and broadcast applications where no sound can bleed from the headphones into live microphones. Closed-back, loud and rugged design gives the K 171 a different low-frequency character than the K 141, yet maintains its comfort and flexibility.


K 240


K 171

K 240 Studio

An update of the "classic " AKG K 240 Monitor, the new K 240 Studio delivers even wider dynamic range and higher sensitivity than its well-known and widely used predecessor. The secret to the semi-open K 240 Studio 's sound quality is the new XXL transducers in acoustically tuned vented enclosures with circumaural (around the ear)pads. The circumaural pads envelope your ears nicely and reward you with smooth,solid bass and well-balanced, articulate mids and highs.

K 271 Studio

The K 271 Studio combines the benefits of AKG 's circumaural design for extreme comfort and a closed-back design for maximum isolation from ambient noise. Thanks to its XXL transducers it sounds clean, smooth and very rich. The K 271 Studio is the perfect choice for any application where sonic bleed could cause problems such as broadcast work. It also has a switch in the headband that mutes the audio just as soon as the headphones are taken off.


K 271

	K 141 Studio	K 171 Studio	K 240 Studio	K 271 Studio
Type	Dynamic	Dynamic	Dynamic	Dynamic
Earphones	Semi-open, supraural	Closed-back, supraural	Semi-open, circumaural	Closed-back, circumaural
Frequency Range	18 to 24,000 Hz	18 to 26,000 Hz	15 to 25,000 Hz	16 to 28,000 Hz
Sensitivity	101 dB SPL/mW	94 dB SPL/mW	91 dB SPL/mW	91 dB SPL/mW
Max. Input Power	200 mW	200 mW	200 mW	200 mW
Rated Impedance	55 Ω	55 Ω	55 Ω	55 Ω
Weight (without cable)	7.9 oz	7.1 o	8.5 oz	8.5 oz
Detachable Cable	Plug-on single cable, approx. 10′	Plug-on single cable, approx. 10′	Plug-on single cable, approx. 10′	Plug-on single cable, approx. 10′
Screw-on Stereo Adapter	Gold plated mini to 1/4″ jack	Gold plated mini to 1/4″ jack	Gold plated mini to 1/4″ jack	Gold plated mini to 1/4″ jack

HEARO 999 AUDIOSPHERE

Digital Surround System

The HEARO 999 is a new reference for headphones monitoring. Many sound engineers, producers, and musicians use the same monitors for years so they become their definitive reference for assessing a mix. The HEARO 999 Processor allows you to take a familiar reference sound wherever you go so you can evaluate any mix meaningfully, no matter what monitors may be available on location. The beauty of this system is you can always tell whether a mix actually meets your own standards or not because the HEARO 999 Processor gives you the same familiar audio wherever you are, anytime. At last, you can carry your own proven reference monitor system with you that enables you to decide on your own whether a mix sounds good enough or not. The perfect choice for professional mixing, editing, overdubbing, etc. in the studio or on location.



Thinking of Buying a Headphone?

An indispensible tool for audio and video professionals, headphones provide a cheap, but excellent reference environment– the standard against which open-air room listening is judged. Audio reference monitoring is the most important tool producers have for deciding what changes to make to a track and for insuring consistency throughout a production in which narration, music and background audio may be recorded at different locations. By design, headphones eliminate the acoustic interaction with a room that can affect the recording of audio signals, and they reproduce exactly the sound that the audio engineer generated during the final mix-down of a recording or that the cameraman captured during the interview. Of course they offer the ultimate in portability. With a good pair of headphones, your sound will meet your standards wherever you go.

Excellent Acoustics and Great Value

Headphones reproduce sound so well because they have less air to move than do open-air speakers, and because they do it in a tightly controlled environment. Their sound quality can be five times better than that of speaker systems selling for the same price, or better yet a $200 set of headphones can have the same specifications for fidelity (frequency response, nominal impedance, sensitivity and total harmonic distortion) as room speakers costing $1000. In addition, headphones are a great value. They are basically small, low-power speakers, and they don't contain expensive parts that can heat up and blow the way speakers do. Ultra high reliability is also the result of new materials and new manufacturing techniques.

The Makings of a Good Headphone

Many headphones, especially consumer ones are poorly engineered and use the cheap circuitry. Headphones are miniature speakers and miniature speakers require miniature power amps. For professional applications, the headphone amp should be a separate part of the amplifier. That's because headphone amps and speaker amps require completely different specs for output power, damping factor, output impedance, slew rate and other variables. If the amplifier is not optimized for headphones, you will soon suffer from common problems like inner-ear fatigue and poor sound imaging. Poorly imaged audio will have no feeling of continuity from left to right, and no sense of depth or space.

Headphone Pad Design

Another headphone-related ailment is outer-ear fatigue. The outer ear is an amazingly delicate part of the body, and few headphones are designed with the long-term listener in mind. Even open-air foam pads can be annoying and painful when pressed against your ears for a while. The design of the headphone pads and the level of comfort they provide should be a deciding factor in your choice. Headphone ear cups generally come in three designs: "supra-aural" (the against-the ear or "open" design), "circumaural" (the around the ear or "closed" design) and inside-the-ear, or Walkman-style. Inside-the -ear headphones can be brutally uncomfortable for extended listening and are certainly a last resort for professional applications. They also feature some of the worst imaging and specs available.

Open-air supra-aural headphones don't block out sound from the outside world and are a good choice if you need to listen for the phone while working. If you don't want interruptions, buy close-ear, or "sealed", circumaural headphones—the type worn by most audio professionals for critical listening applications. Circumaural headphones also prevent audio "leakage", or the bleeding of sound into the mic while the headphones feed the listener. Enclosed headphones are ideal for while doing voice-overs for video work, because they prevent the original audio track from bleeding into the voice-over. The voice-over takes up a mixer track whose volume has to be separate from that of the background tracks captured with the video camera's mic.

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HEADPHONES

ATH-M2X / ATH-M3X

Mid-Size Dynamic Headphones



If listening to CDs through your headphones leaves something to be desired—like richness, accuracy and comfort—plug in the ATH-M2X or ATH-M3X. Designed to reproduce the wide dynamic range found in digital program materials, their high-performance drivers and low-mass diaphragms provide maximized power handling capability with improved frequency and transient response. The open-back ATH-M2X offers a clean, linear low-end, while the closed-back ATH-M3X is better for isolation and powerful bass. Both have soft cushioned earpads and an adjustable headband for hours of listening comfort. The M2X includes an 11.5″ cable while the M3X includes a 9.8″ cable with 3.5mm stereo mini-plug and 1/4″ phone plug adapter

ATH-COM1/ATH-COM2



Ideal for hands-free voice-activated computer, multimedia and DJ applications, AT's COM Series headsets offer excellent sound quality for monitoring and speaking or singing with a combination headphone plus dynamic microphone. Choose the ATH-COM1 for one-ear monitoring, or the ATH-COM2 with dual-earphone stereo sound and enhanced operator isolation. Both have neodymium dynamic microphones for crisp, accurate vocal pickup. The flexible mic boom, cushioned earpads and overall lightness provide added convenience and prolonged listening comfort.

- Flexible mic boom, cushioned ear pads and overall lightness provide added convenience and comfort
- They include durable 4.9′ cable with two 3.5mm input/output mini-plugs and two 1/4″ phone plug adapters

ATH-M30

Full-Size Dynamic Headphones



Embodying AT's latest transducer technology, the ATH-M30 offers sonic accuracy in a design engineered to provide maximum comfort for hours of listening. The ATH-M30 features a low profile, closed back design providing exceptional clarity and superior isolation in high volume listening situations. Extremely accurate, the headphone features 40mm drivers, neodymium magnets, CCAW voice coils and has a frequency response of 20Hz-20kHz. The ATH-M30 has a lightweight and adjustable cushioned headband for maximum listening comfort. In features a gold-plated stereo 1/8″ (3.5mm) connector with strain relief and professional screw-on 1/4″ adapter, and a sturdy, ultra-flexible 11″ cable with single-side exit.

ATH-D40FS / ATH-M40FS

Precision Studiophones



AT's Precision Studiophones deliver premium performance, long-lasting reliability and comfort for specialized uses. Built for power and precision with large 40mm diameter drivers, neodymium magnets and CCAW voice coils, they deliver 1600 mW of power handling and very high SPL capability while maintaining exceptional clarity and accuracy. They have adjustable padded headbands and sturdy, ultra-flexible strain-relief cables made of OFC litz wire, with convenient one-side exit. Circumaural earpieces provide maximum isolation, and can be rotated 180° to permit one-ear monitoring. The ATH-D40FS features an enhanced bottom-end response (from about 20Hz to 500Hz), making it ideal for use with predominantly low-frequency sound sources such as bass instruments and drums. The ATH-M40FS is designed specifically for professional monitoring/mixing, with a flat, extended low-frequency response to 5Hz that ensures accurate sound reproduction.

HEADPHONES WITH MICS

DT-108 / DT-109

Single- and Dual Sided Headset with Noise Cancelling Unbalanced Mic



The DT-108 (single-muff) and DT-109 (dual muff) headsets are designed for live and remote broadcasting in studio, film and TV-applications. One side is open on the DT-108 to monitor ambient sound. The headphone transducers feature extended frequency response. The protective ear cap attenuates up to 20 dB of ambient noise. Adjustable, noise-cancelling microphone provides clear, audible speech transmission.

- Highly sensitive headphone and microphone transducers
- Excellent ambient noise attenuation
- Capable of handling very high SPL
- Rugged and field serviceable construction
- Excellent comfort for long-term wearing
- Modular construction for easy servicing

DT-190

Dual-Sided Headset with Noise Cancelling Balanced Microphone



A closed headset with the addition of the DM-190 dynamic, hypercardioid microphone, the DT-190 is designed for broadcasting, monitoring and communications. Use it for commentating at sports events or for intercom systems at live concerts and other situations which require precise audio performance. The DM-190 is a balanced, noise cancelling microphone with high SPL capability. It is mounted on a fully adjustable boom allowing accurate positioning for optimum sound and minimal explosive “p” popping.

- Noise cancelling microphone and high SPL capability (97dB)
- Excellent isolation from ambient noise
- Modular design for easy servicing
- Ruggedly constructed

DT-280/DT-290

Single- and Dual Sided Headset with Hypercardioid Dynamic Microphone



The DT-280 and DT-290 are the benchmark for low-profile, studio quality, professional on-air headsets. Lightweight with fully modular construction for easy field servicing, these closed headsets are ideal for announcement or command purposes in broadcasting applications. They feature an efficient neodymium magnet system, precision reproduction and a balanced sound. They also have a wide 10Hz-30kHz transmission range and come with 80-ohm systems.

- Soft circumaural earcups and the fully adjustable padded headband offer a high degree of wearing comfort.
- Dynamic microphone can be optimally positioned with the flexible gooseneck.
- Hypercardioid polar pattern guarantees high-gain-before feedback and noise cancellation.
- If necessary the assembly of the microphone holder can be changed so that the mic can be worn on the left or right side. The cables to the earcups and the microphone run safely in the headband.
- The connecting cable is single-sided, plug-in type with a golden combi 1/4″/mini jack plug.
- All parts such as earpads, headband and gooseneck microphone are easy to replace.

DT-297 PV

Dual Sided Headset with Hypercardioid Dynamic Microphone



The lightweight, closed DT 297 PV features an efficient neodymium magnet system, high reproduction precision and an optimized speech reproduction for an excellent intelligibility. Has s wide 10Hz-30kHz transmission range

- Mic can be optimally positioned with the flexible gooseneck.
- Soft circumaural earcaps and the fully adjustable padded headband offer a high wearing comfort.
- The cables to the earcaps run safely in the headband

HEADPHONES

DT-150
Dynamic Studio Headphone



The DT 150 is a dynamic studio headphone for use in broadcasting, film and recording studios and other audio monitoring applications where size, comfort and performance are primarily requirements. Due to its closed design it features excellent ambient noise isolation for reference monitoring. The padded headband can be adjusted for any head size.

- High long-term wearing comfort
- Rugged construction
- Extended bass response
- Modular construction for easy servicing

DT-250
Dynamic Studio Headphone



The DT 250 is a stylish, closed dynamic headphone offering excellent ambient noise isolation and a linear response designed for reference monitoring. It is suitable for all kinds of professional environments including film, sound, ENG/EFP, and other audio monitoring applications where size, comfort and performance are the primary requirements. It features an efficient neodymium magnet system, reproduction precision and a balanced sound. The soft circumaural earcups offer a high degree of comfort and excellent ambient noise insulation. The padded headband is fully adjustable and can be adapted to each head size. The main connecting cable is single-sided with combination 1/4″/mini jack plug.

DT-770 PRO
Stereo Studio Headphone



The DT770 PRO is a closed, dynamic headphone designed for professional studio applications. The rugged headband construction gives the headphone a secure fit and the reliability that is required for daily use. Features a wide 5 Hz to 35kHz frequency response and 250-ohm supplied gold-vaporized 1/8″ stereo plug with 1/4″ adapter.

- Closed studio headphone, equalized to meet diffuse field EQ requirements
- Innovative bass-reflex-system
- Robust spring steel headband
- Single-sided cable
- Soft head-pad
- Gold plated mini stereo jack (3.5mm) and 1/4″ adapter

DT-831
Stereo Hi-Fi Headphone



Closed back Hi-Fi headphones, the DT831 features an extremely strong neodymium magnet system with newly developed magnetic circuit of super-refined steel for natural dynamics. The neutral but direct sound characteristic is an essential feature that you normally would only expect from a headphone in open design.

- Improved acoustical figures due to new computer calculated ear cups
- Acoustically patented ear pads with viscose
- Excellent diaphragm construction eliminates unwanted resonance
- Powerful, distortion-free bass response even at high levels
- Coiled cable with gold-plated stereo mini jack and 1/4″ adapter
- Easy servicing, all parts replaceable.

HEADPHONES

DT-48E



Designed for use in ENG/EFP applications, the DT-48 E, with single-sided cable is especially recommended for reporting applications. Long term comfort is achieved by the soft-feel headband and ear cushions which are filled with foam material while the fully adjustable headband ensures a good fit even on the most demanding locations. The fully enclosed ear cup eliminates ambient noise, essential for accurate monitoring.

- Suitable for live broadcast and studio use
- Highly efficient transducers
- For accurate monitoring of all kinds of recording
- Extremely comfortable

DT-100 / DT-102



The DT-100 headphone is widely used in audio recording and monitoring applications in studios worldwide. It offers a high degree of comfort and can be worn for hours without listener fatigue. The soft ear cushions effectively seal out ambient noise yet apply minimal pressure against the ear. The DT-102 is the identical headphone with the exception that it is a single-ear headphone. The DT-102 is used in broadcast work, recording, TV studio and theater applications where the monitoring of outside ambient information is necessary.



FOSTEX THEATERPHONE HSM6240

Surround Sound for Headphones

With TheaterPhone, monitoring a surround sound mix doesn't have to be restricted to studios with a 5.1 speaker setup. When access to loudspeakers is limited, you can still monitor in the "sweet spot" of a full multi-dimensional surround mix with sounds in front of and behind your head. The experience is as if speakers were mounted up on the walls and around you. In fact, you'll take off the headphones the first time you hear it, just to convince yourself it's not coming from the room.

TheaterPhone HSM6240 features three types of virtual listening spaces approved by Dolby Laboratories: DH1 (reference control room), DH2 (livelier dubbing size room), and DH3 (larger theater-size room). Each room was meticulously modeled with impulse response measurements of real acoustic environments and combined with Lake Technology's extensive experience with Binaural HRTF Virtual Reality research. The result is an innovative technology call Dolby Headphones. Using Dolby Headphone processing, the TheaterPhone HSM6240 delivers a true 6-channel spatial surround sound experience over stereo headphones. TheaterPhone HSM6240 can be inserted into any 5.1 mix bus or stem using the six discrete balanced TRS jacks and six pass through connections provided. Post-processed stereo line outputs are also provided for additional headphone distribution amplifiers. Tri-color channel active LEDs provide a tactile look at each signal input level. When monitoring a 5.1 mix with speakers is inconvenient, inefficient, or impossible, use the TheaterPhone HSM6240 for a headphone experience that is truly three-dimensional.

- Incorporates Dolby Headphone technology—a revolutionary signal processing system that can take up to six channels of audio from any source and make it sound like it's coming from six speakers placed in a real listening room. The "in your head" effect of stereo headphone listening is replaced by a spacious, natural 5.1 soundfield.
- Realistic surround sound simulation on stereo headphones
- Supports stereo and 5.1 formats
- Choice of three listening environments
- 6 discrete input channels for five speakers plus subwoofer
- Post-processed stereo line outputs for headphone distribution amps
- 24-bit/48kHz converters
- Applications include surround sound monitoring, music production, audio post production, project studio surround, machine room monitoring, DVD quality checking, home theater and much more.

T20RP/T40RP/T50RP

RP Series Professional Headphones

True and accurate headphone monitoring is crucial for recording studios, video post houses, remote broadcast facilities, project studios and DJ's the world over. Everyone wants headphones which faithfully reproduce all aspects of the program material, free from any sonic coloration and distortion. Enter Fostex's superb range of RP Professional headphones. A range of headphones which have been designed from the outset to offer the widest frequency range, to minimize listening fatigue and to allow you to accurately pinpoint and analyze all aspects of your creative output.



Unique RP Technology

Key to the excellent and superbly transparent sound reproduction is Regulated Phase Technology, a proprietary transducer and diaphragm design that has earned Fostex more than twenty international patents. This unique technology has been applied successfully to microphones, headphones and loudspeakers used and relied upon in professional and commercial sound installations worldwide. And in the case of the T50RP, this technology has been enhanced still further in a new RP Diaphragm which uses copper foil etched polyimide film to provide superb resistance to high level input peaks of up to 3000mW.

Clarity of Sound

The headphones deliver supreme uncolored reproduction and a natural clear sound across the full frequency range. The highs sparkle, the mids are crystal clear and nothing is missing from the low-end. But it's the sonic quality at high sound pressure levels (SPLs) which really sets these headphones apart. The bass is always powerful and distortion-free even at high levels. Essential when headphone monitoring in noisy environments such as ENG recording, DJ work, and where you need to concentrate on the program material. The leatherette padded ear cushions provide excellent sound isolation and superb deep bass reproduction

Rugged Construction

Manufactured to the most exacting standards with durable, tough and rugged materials and components, RP Series headphones are built to last. They're also built for comfort. Those padded ear cushions and wide, self adjusting headbands ensure maximum comfort and fit during extended wearing periods. A nice touch is the lockable, single entry jack which minimizes cord tangles and offers excellent serviceability. Whether you're monitoring analog or digital audio sources, work in a professional or project studio, mixing live, or need a great pair of headphones for DJ applications, RP Series headphones simply do the job.

The T50RP—Why Is It So Special?

The T50RP utilizes a new development called an 'RP Diaphragm'. This unique Fostex technology employs a copper foil etched polyimide film to provide resistance to high level input peaks of up to 3000mW combined with a neodymium magnet for high sensitivity and excellent transient handling. This combination lifts quality of the T50RP's audio reproduction above that of the already superb T40RP and T20RP. In short, the T50RP offers perhaps the most natural and uncolored sound yet heard from a pair of studio headphones.

T20RP

Semi-open style—smooth vocal reproduction along with punchy low and well balanced mid/high frequencies

- Frequency Response: 20Hz - 30kHz
- Light weight for extra comfort
- Excellent transient characteristics
- Includes lockable 3m headphone cord
- Headpad supplied for extra comfort
- Ideal for professional use in commercial recording studios and DJ work

T40RP

Enclosed style for maximum feedback prevention and clear transparent reproduction of all audio frequencies

- Frequency Response: 20Hz - 25kHz
- Prevents feedback into the mic at recording sessions
- Includes lockable 3m headphone cord
- Headpad supplied for extra comfort
- Ideal for professional use in commercial recording studios and DJ work

T50RP

Damped semi-open style and unique RP technology result in extended low-frequency range plus crystal-clear mid and high frequency reproduction.

- Frequency Response: 15Hz - 35kHz
- Neodymium magnet for high sensitivity and excellent transient handling.
- Includes lockable 3m headphone cord
- Superb quality and perfect for commercial recording use

SAMSON

STEREO REFERENCE HEADPHONES

PH60



Designed to offer great sound in a lightweight, high style "see-through blue" design for personal stereo listening on the go, the Samson P60s use a special 34mm Mylar driver with an isotropic Ferrite magnet for a surprisingly big, open sound not typically found in these kinds of headphones. Their distinctive clear blue appearance also sets them apart.

RH100



Combining professional specifications, exceptional comfort and high quality audio at a very accessible price, the RH100 headphones from Samson also use a newly designed high performance transducer along with a high output neodymium magnet and leading-edge headphone circuitry. Engineered for comfortable listening over extended periods of time and natural, uncolored audio, the RH100s sound great in any kind of studio and/or home music playback situation.

RH300



Offering unparalleled audio quality and a comfortable listening experience, Samson's mid-line RH300 headphones also benefit from the latest transducer technology along with a special driver and neodymium magnet. An open-ear design and specially designed acoustic mesh fabric provide an enhanced ambient listening experience while reducing listener fatigue. Endowed with a wide dynamic range and linear frequency response of 20Hz to 22kHz, these phones are ideal for any number of playback situations.

RH600s



The top-line model RH600s use a specially designed, high performance driver with a thin, responsive 40mm diaphragm coupled with a high output neodymium magnet and sophisticated circuitry to provide true "reference quality" audio. Its ambient open-ear design with special acoustic mesh fabric ensures clear, transparent sound and a comfortable listening experience. The RH600s also feature a very wide dynamic range and a linear frequency response from 20Hz to 22kHz for accurate, balanced audio reproduction.

	PH60	RH100	RH300	RH600S	CH70	CH700
Type	Supra-aural open-back	Circumaural, open-back	Circumaural, open-back	Circumaural, open-back	Closed-back design	Closed-back design
Frequency Range	20Hz to 18kHz	20Hz to 22kHz	20Hz to 22kHz	20Hz to 22kHz	20Hz to 22kHz	20Hz to 22kHz
Sensitivity	101dB	106dB	106dB	115dB	108dB	103dB
Rated Impedance	32 Ω	64Ω	32Ω	40Ω	64Ω	32Ω
Connectors	1/8″ stereo mini-jack with 1/8″ to 1/4″ TRS phono adapter	1/8″ stereo mini-jack with 1/8″ to 1/4″ TRS phono adapter	1/8″ stereo mini-jack with 1/8″ to 1/4″ TRS phono adapter	1/8″ stereo gold plated mini-jack with 1/8″ to 1/4″ TRS phono adapter	Gold plated mini stereo plug and 1/4″ phone adapter	Gold plated mini stereo plug and 1/4″ phone adapter

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STUDIO REFERENCE HEADPHONES

Samson offers two different studio headphones for recording, monitoring and music listening applications: the CH700 and CH70 studio reference headphones. They feature a closed-ear design with a special mesh for clear, transparent sound. The CH700 features sophisticated, high-quality ultra-thin and responsive 40mm Mylar diaphragm drivers with high-efficiency neodymium magnets for high output and superior audio quality. The CH70 also employs 40mm Mylar diaphragm drivers with high-output neodymium magnets. Both headphones employ a wide dynamic range and linear frequency response from 20Hz to 22kHz for exceptional reproduction and a balanced, articulate sound.

CH-70



CH70 headphones are ideal for any listening and monitor application where critical sound reproduction is needed. The closed-ear design provides maximum isolation reducing unwanted bleed that can be picked up on live microphones. The CH70 employs exceptional 40mm drivers with ultra thin Mylar diaphragms and high-output neodymium magnets. The result is a wide dynamic range and linear frequency response from 20Hz to 22kHz for exceptional reproduction and a balanced sound making them a perfect choice for monitoring digital audio. The adjustable headband and lightweight design provide comfort and easy listening through the longest sessions.

CH-700



CH700 headphones are ideal for any listening and monitor application where critical sound reproduction is needed. The closed-ear design provides maximum isolation reducing unwanted bleed that can be picked up on live microphones. The CH700 employs exceptional 40mm drivers with ultra thin Mylar diaphragms and high-output neodymium magnets. The result is a wide dynamic range and linear frequency response from 20Hz to 22kHz for exceptional reproduction and a balanced sound making them a perfect choice for monitoring digital audio. The large earpiece further extends the low frequency response and the adjustable headband and lightweight design provide comfort and easy listening through the longest sessions.

Sennheiser HDC-450 / PXC-250 *Noise-Cancelling Headphones*

Experience high fidelity and comfort any time you travel with Sennheiser noise-cancelling headphones. Recommended for aircraft or other mass transit travel, these units feature active noise cancellation and to reduce ambient noise levels, along with Open Aire design to produce excellent sound quality in a high noise environment.

HDC-450



The PXC 250 is a closed, dynamic, supra-aural mini stereo headphone with NoiseGuard active noise compensation (attenuates frequencies below 1,000 Hz by up to –15 dB). The headphones are foldable and come with a convenient belt pouch, making them an ideal travel companion.

- 1.8m Kevlar-reinforced OFC copper cable with 3.5mm jack plug, 1/4″stereo adapter and 3.5mm double mono (for airline audio)
- High-resolution sound due to Sennheiser "Duofol" diaphragms with spiral embossing
- Bass tube for powerful, punchy bass

PXC-250



An ultra-light, open-aire active noise cancellation headphone, the HDC 451 incorporates Sennheiser's exclusive NoiseGuard technology to reduce low frequency cabin noise by up to 10 dB. Allows you to listen at much lower and safer volume levels.

- Incorporates an in-line battery box that can easily be clipped to your clothing, proving ideal for portable applications.

SENNHEISER

EVOLUTION SERIES

Sennheiser's Evolution Headphone Series offers today's recording artists and musicians an alternative in high performance headphones. Combining outstanding clarity and performance with sleek, sophisticated design and a surprisingly affordable price point, the Evolution Series is the ideal choice for up-and-coming performers who have long sought to bring Sennheiser's technical excellence and sound quality to their own home and studio listening environments. Consisting of three dynamic headphone units, the EH1430, EH2200 and EH2270 all offer durability, versatility and rich, full sound. All incorporate a 3.5mm stereo jack with a 6.3mm (1/4˝) adapter.

- *Bionetic Design– designed to fit the shape of the ear and head, the result is comfort for extended use*
- *Parts can be replaced easily*
- *Duofol Diaphragms for clearer highs and more solid bass response than standard diaphragms. Duofol diaphragms also offer lower distortion*

EH1430

The Evolution EH1430 is an open-air, stereo headphone that can be connected to both Hi-Fi systems and personal stereos (CDs, DATs, DCCs, etc.). Ideal for rock and pop music listening, the EH1430 delivers 22Hz-20kHz frequency response and excellent bass response in a supraural design.

- Open-air, dynamic stereo headphones
- Matte black finish
- 9´ highly conductive OFC cable with 1/8˝ stereo jack and 1/4˝ adapter
- Replaceable ear cushions
- 2-year warranty



EH2200

The Evolution EH 2200 is a closed, hi-fi stereo headphone that can be connected to both hi-fi systems and personal stereos (CDs, DATs, DCCs, etc.). Ideal for DJ monitoring, the EH2200 creates superb sound reproduction with powerful bass in a circumaural design.

- Closed, dynamic stereo headphones
- Very lightweight for extended use
- Award-winning BioNetic design offers an ergonomically correct fit for maximum comfort
- DuoFol technology which fuses two diaphragm materials to virtually eliminate standing waves in the capsule
- Detailed sound reproduction with excellent bass response
- Significant reduction in ambient noise levels, decreasing sensitivity to feedback
- Single-sided, detachable, kevlar-reinforced OFC signal cable
- Fully modular design with replaceable parts for long life



EH2270

The top of the Evolution Headphone Series, the EH2270 is designed for playback in recording applications. It's closed-cup, circumaural ear couplings provide total isolation, while its frequency response of 12-22,000 Hz makes it a great choice for performers and recording engineers in critical listening situations.

- Closed, dynamic stereo headphones
- Award-winning BioNetic design offers an ergonomically correct fit for maximum comfort
- Neodymium magnets for powerful dynamics and high sound pressure levels
- DuoFol technology which fuses two diaphragm materials to virtually eliminate standing waves in the capsule
- Revolutionary damping element for controlled diaphragm oscillation
- Single-sided, detachable, kevlar-reinforced OFC signal cable
- Fully modular design with replaceable parts for long life
- 2-year warranty
- Impressive, full-bodied, detailed sound reproduction with excellent bass response

400 SERIES

Open-Aire Dynamic Hi-Fi Stereo Headphones

Employing Sennheiser's famous Open-Aire construction and featuring high performance capsules, each 400 Series model is an especially good value in a full-sized headphone.



HD 457

Enjoy music at home or on the road - the HD 457 deliver wide dynamics, excellent stereo sound and warm bass for fusion, soul, funk and percussion. Convenient wind-up clip for on-the-go use.

- Open-Aire, dynamic, supraural hi-fi stereo headphones
- Ideal for home mini hi-fi systems and mobile audio sources such as CD, MiniDisc and MP3 players
- Very comfortable to wear due to ultra-lightweight design
- 3 m highly conductive OFC copper cable
- Replaceable fabric-covered ear cushions
- 2 -year warranty
- Clear sound and rich warm bass makes the HD457 ideal for soul, funk, percussion and fusion
- Rugged outdoor design with extremely flexible headband

HD 477

These open, dynamic hi-fi stereo headphones with clear, transparent sound and powerful bass reproduction are ideal for rock and pop music. Mix a trend-setting design with superlative audio and the result is the HD 477. Impress even the most discerning ear with clear, well-defined sound and a wide dynamic range. Lightweight and comfortable, you may never want to take them off! Robust design with detachable signal cable - a durable companion for any application.

- Completely new ergonomic design
- Extremely lightweight and comfortable
- Clear dynamic sound with enhanced bass response
- High-efficiency drivers - great for use with portables
- Detachable OFC copper cable
- Two year warranty

HD 490

This open, supra-aural, dynamic hi-fi stereo headphone is ideal for hi-fi and portable systems. It's acoustic properties create superb sound reproduction with powerful bass. With its ultra-light BioNetic design (headphones adapt to the shape of your head) and a dampening technique (controls diaphragm oscillation), the HD490 is an ideal choice for the most discerning music lover.

- Detailed sound reproduction with powerful bass
- Detachable, single-sided signal cable made from highly conductive OFC copper
- Duofol diaphragms to reduce standing waves
- Suitable for all types of hi-fi equipment, especially digital sound sources



HD 497

The HD 497 open, dynamic hi-fi stereo headphones deliver crystal-clear, well-defined trebles and absolute precision across all audio frequencies.Delivers crystal-clear, well defined highs and absolute precision across all frequencies. With a warm, natural, detailed sound image they not only give a brilliant performance on Hi-Fi systems, they will also bring your portable player to life! Ideally suited to all types of music.

- Completely new ergonomic design
- Extremely lightweight and comfortable
- Extended frequency response and warm, natural sound reproduction
- High-efficiency drivers - great for use with portables
- Detachable OFC copper cable
- Two year warranty

SENNHEISER

500/600 SERIES

Designed for professional and critical listening situations, the 500 and 600 Series headphones exhibit full frequency response and exceptionally transparent sound reproduction. They all feature a computer-optimized magnet system that minimizes harmonic and intermodulation distortion, exceptionally natural, spatial, and accurate sound, and detachable Kevlar reinforced OFC copper cable.



HD 580

A high-end open, dynamic HiFi professional headphone, the HD 580 is ideal for professional recording engineers, specifically those recording classical music, because it offers exceptionally natural, spatial, and accurate sound. The advanced design of the diaphragm avoids resonant frequencies.In every respect, this is a headphone of exceptional quality.

- Extremely lightweight aluminum voice coils ensure excellent dynamic response
- Neodymium ferrous magnet system ensures optimum sensitivity and great dynamic response

HD 590

Exemplifying Sennheiser's BioNetic headphone designs, the HD 590 offers a virtually unsurpassed combination of comfort and performance in an open, dynamic stereo headphone. With a weight of just 9.4 ounces and a sophisticated design the HD590 is an ideal choice for the professional monitoring of classical music recordings.

- Audiophile, Open-Aire, hi-fi stereo headphones
- Award-winning BioNetic design offers maximum comfort
- Duofol technology reduces standing waves
- Dampening element for controlling the diaphragm oscillation, optimally matches the headphones to all sound sources
- Velour ear cushions

HD 600

The HD 600 is an audiophile quality open dynamic hi-fi/professional stereo headphone. The advanced diaphragm design eliminates standing waves in the diaphragm material. The HD 600 can be connected directly to hi-fi systems of the highest quality, in particular DAT, DCC and CD players. This headphone is an ideal choice for the professional recording engineer recording classical music.

- High-quality open metal mesh earpiece covers for an extremely transparent sound
- Extremely lightweight aluminum voice coils ensure excellent transient response
- Neodymium ferrous magnet systems ensure optimum sensitivity and an excellent dynamic response



HD 500 Fusion

Creating a buzz among listeners of jazz, funk, rock and hip hop, the HD 500 fuses improved audio technology with a BioNetic design that offers unsurpassed comfort. Duofol diaphragms and special damping elements ensure a warm, yet transparent sound reproduction.

HD 570 Symphony

Perfect for the demanding classical or jazz aficionado who wants superb sound. The HD 570 has all the refinements: a combination of Duofol diaphragms, lightweight voice coils and innovative damping elements to ensure wide dynamics, fast transient response and a smooth, spatial sound reproduction. The unique BioNetic design ensures many hours of exceptional comfort.

RS SERIES WIRELESS HEADPHONES

900 MHz Wireless Headphones

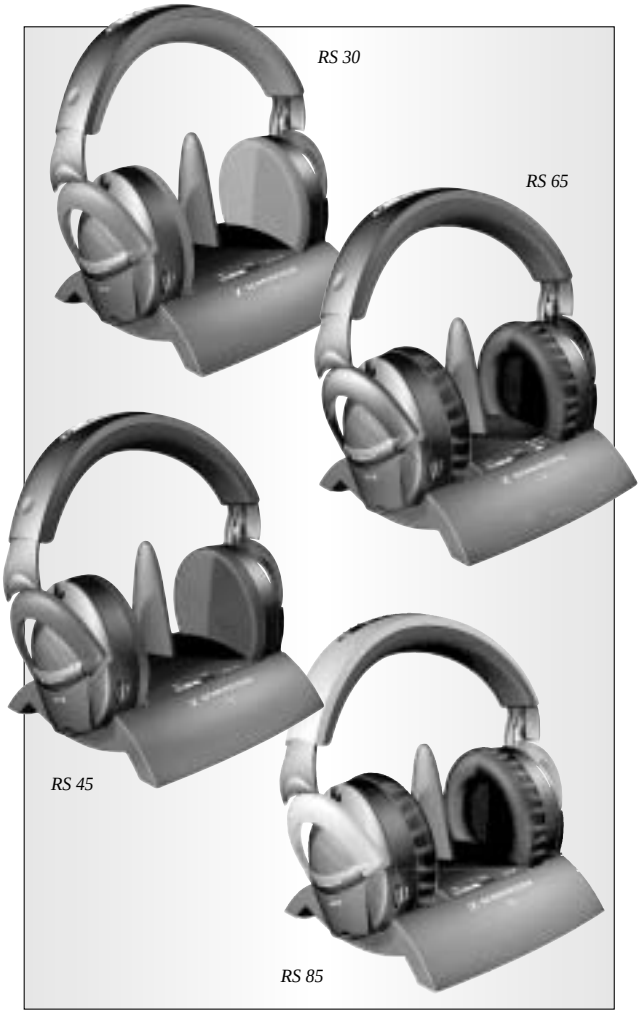
Now musical enjoyment knows no limits: Sennheiser's cordless headphones free you from the restrictions of a headphone cable, delivering quality sound through walls and ceilings. The RS Series not only features dynamic sound reproduction and low noise, but also offers truly innovative design and exceptional comfort. They can be automatically recharged when stored on the transmitter. Each system easily accommodates multi-users, simply by purchasing additional headphones.

RS 30

Transparent, balanced sound makes this system an ideal choice for rock and pop music. The ideal entry into the world of cordless sound. The RS 30 wireless RF system with Open-Aire, supraural headphones offers accurate sound reproduction for "live" effects and a punchy bass response. Its transparent and balanced sound makes this system an ideal choice for rock and pop music. The ideal entry into the cordless world of sound.

RS 45

Wireless RF headphone system featuring Center Lock auto tuning, spatial sound reproduction and solid bass response. The RS 45 wireless RF system with Open-Aire supraural headphones features warm, spatial sound reproduction with powerful bass response. The transmitter automatically recharges the battery when the headphones are placed on the base unit. Center Lock auto tuning ensures hassle-free, comfortable listening.



RS 65

Wireless RF headphone system with Center Lock auto tuning and transparent, balanced sound. A switchable SRS processor turns the RS 65 into a surround sound headphone system— and a TV film into a true home theater experience. The circumaural RS 65 delivers extremely transparent, balanced sound. Velvet-like ear cushions, very light weight and Center Lock auto tuning make listening a pure pleasure. Music recordings will benefit from truly outstanding sound, immersing the listener in a real concert hall atmosphere.

RS 85

Top-of-the-line wireless RF headphone system features "HiDyn plus" noise reduction for ultra-quiet transmission, extended frequency response and unparalleled sound reproduction. Includes Center Lock auto tuning. The ultimate in wireless headphone listening! The RS 85 wireless RF system with Open-Aire, circumaural headphones features "HiDyn plus" noise reduction (used in professional wireless microphone systems) for ultra-quiet and superb sound quality, ensuring a very natural and detailed sound reproduction. Includes Center Lock auto tuning.

	RS30	RS 45	RS 65	RS 85
Frequency Response	30 - 19,500 Hz	20 - 20,000 Hz	20 - 20,000 Hz	20 - 22,000 Hz
THD	< 0.7%	< 0.7%	< 0.5%	< 0.5%
Carrier frequencies (switchable)	925.0 MHz/926.0 MHz/926.5 MHz		925.0 MHz/926.0 MHz/926.5 MHz	
Maximum SPL	96 dB	96 dB	100 dB	101 dB
Signal to Noise Ratio	> 68 dBA	> 68 dBA	> 68 dBA	> 85 dBA
Modulation	FM stereo	FM stereo	FM stereo	FM stereo
Operating time	Approx. 4 hours per battery	Approx. 4 hours per battery	Approx. 4 hours per battery	Approx. 4 hours per battery
Power Supply	BA 151 rechargeable battery	BA 151 rechargeable battery	BA 151 rechargeable battery	BA 151 rechargeable battery
Weight (receiver)	5.6 oz	5.6 oz	5.6 oz	5.6 oz

SENNHEISER

STUDIO MONITOR HEADPHONES

Optimized for reference studio monitoring, these headphones feature closed construction for maximum isolation and incredibly full response. They not only shut out ambient noise, but have also set the standard for natural sound.

HD 202 / HD 212 PRO

The HD 202 and HD 212 Pro are closed, dynamic hi-fi stereo headphones with good insulation against ambient noise and a deep bass response that make them ideal for DJs or anyone who likes to listen to modern, powerful music without disturbing others. They offer high efficiency drivers for maximum performance.

- Earcups can be removed from the headband
- Specially designed damping perforation ensures powerful bass response
- Lightweight diaphragm material with "turbine" embossing for low bass
- High levels and crisp bass for modern rhythm-driven music
- Comfortable, ultra-lightweight design
- Replaceable leatherette ear pads
- 2-year guarantee

HD 202

- Powerful neodymium magnets and lightweight diaphragms for high sound levels
- Clips to the belt: convenient cord take-up for adjusting cable length

HD 212 Pro

- Extra-thin diaphragms for precise trebles
- Well defined mid-band and spatial sound reproduction due to foam damping element

HD 25

Perfect for ENG, sound reinforcement, studio monitoring, and many other applications, the HD 25 is a closed dynamic compact professional monitoring headphone that offers high attenuation of background noise. Capable of handling very high sound pressure levels and featuring extremely robust construction, the HD 25 performs exceptionally well in high noise environments.

- Very good attenuation of background noise
- High sensitivity
- Low weight
- Comfortably fitting adjustable headband
- Detachable single-sided connection cable
- Frequency response: 30-16,000 Hz
- Impedance 85Ω; Sensitivity is 100 dB

HD25SP

A compact, closed back dynamic headphone, the HD25SP offers high attenuation of background noise, extremely robust construction, and is capable of handling very high SPLs (Sound Pressure Levels), making it ideal for use in noisy environments. The HD25SP is a lightweight and comfortable alternative for those who spend a great deal of time wearing headphones and need maximum isolation. While extremely rugged, the HD25SP ensures an accurate response for critical studio recording and mixdown applications. The low impedance, high sensitivity drivers exhibit unparalleled efficiency, providing sufficient volume even in applications that have classically required an additional headphone amplifier.

- 30Hz - 16kHz frequency response
- 85 ohms nominal impedance
- Adjustable headband ensures optimum fit for each user's taste
- Sensitivity is 100dB
- 10′ single sided steel cable with a 1/8″ to 1/4″ stereo phone plug
- Two year warranty on parts and labor



STUDIO MONITOR HEADPHONES

HD250II

A dynamic HiFi/live sound recording headphone, the HD250II is extremely popular with live sound recording engineers and hi-fi enthusiasts alike. The HD 250 II is a closed headphone with the feel of an open design. Their precise sound reproduction, excellent THD and wide dynamics will keep you spell-bound!

- Excellent THD
- Wide dynamics
- Frequency response: 5 – 40k Hz –10 dB; 10 – 25k Hz –3 dB
- Transducer principle dynamic, closed
- Nominal impedance 300 Ω
- Characteristic SPL at 1 kHz 94 dB
- Load rating 200 mW



HD265 Linear

The closed HD265 dynamic stereo headphone is excellent for critical studio listening as well as for professional monitoring applications. The circumaural design combined with the sealed principle deliver a totally quiet listening environment, perfect for professional monitoring applications. The HD265's lightweight, copper-coasted aluminum drive coil in conjunction with the dual layer "Duofol" diaphragm offers amazingly precise sound reproduction with superb bass response. Large, leatherette earpads provide maximum isolation from ambient noise while allowing maximum extended comfort.

- 10Hz- 25kHz frequency response
- 150 ohms nominal impedance
- Sensitivity is 97dB
- Fully adjustable, padded headband.
- Two year warranty on parts and labor
- 10′ OFC (oxygen free copper) cable with a 1/8″ to 1/4″ gold-plated stereo phone plug.

HD 270

The closed, circumaural HD 270 stereo monitoring headphones can be connected to both hi-fi systems and personal stereos (CDs, DATs, DCCs, etc.). Designed for playback in recording applications, the headphone provides total isolation, while its frequency response of 12 - 22k Hz makes it a great choice for critical listening situations.

- Neodymium magnets for powerful dynamics and high sound pressure levels
- Duofol Technology reduces standing waves
- 10′ single-sided, kevlar-reinforced OFC cable
- Fully modular design with replaceable parts for long life

HD 280 Professional

Designed to exceed the demands of the professional environment, the closed, circumaural HD280 boasts extremely robust construction combined with extensive features that meet the requirements of today's most demanding applications. The unique collapsible design combined with swiveling ear cups, offers maximum flexibility in any application.

- Accurate, linear sound reproduction for critical monitoring applications
- Optimum impedance ensures universal compatibility
- Collapsible, rotating earcups
- Up to 32 dB of ambient noise attenuation
- Neodymium magnets for high max. SPL
- Single-sided, coiled cable with 3.5mm mini jack with locking 1/4″ adapter
- Rugged construction with user-replaceable parts
- 2 year warranty

SONY

STUDIO MONITOR HEADPHONES

All Sony Studio Monitor Series Headphones incorporate high-end materials and advanced engineering,to deliver critically clean, exceptionally clear sound for professional and high fidelity applications.

MDR-V600

Designed for high-end studio applications, the MDR-V600 features a circumaural earcup design, 40mm Aura-Nomic driver, Neodymium magnets, CCAW voice coil, reversible earcups, gold-plated plugs, cushioned headband and a folding design.



- Circum-aural earcup design in the natural shape of the human ear positions the driver unit at a correct distance from the ear canal and creates a sealed resonance chamber over the earlobe itself, reducing pressure on the ear for a more comfortable fit, especially during extended listening sessions.
- 40mm Aura-Nomic designed driver is positioned within the earcup at an angle which directly corresponds to the angle of the opening of the ear canal for the most natural listening experience possible.
- 1000 mW power handling for professional applications; use at high output levels
- Uses Neodymium magnets for maximum energy and minimum size
- Gold-plated plugs resist corrosion for high conductivity and low noise
- Reversible earcups enable single-sided monitoring flexibility
- Cushioned headband adds to wearing comfort and a better overall fit. Provides comfortable listening for hours on end.
- Single-sided OFC cable and gold plated Unimatch plug provides both 1/8″ mini-jack and 1/4″ TRS phono connectivity
- Folding design for easy storage.

MDR-V900



Sony's highest-performance studio headphone, the MDR-V900 features a circum-aural earcup design, 50mm Aura-Nomic driver, Neodymium magnets, 3000 mW power handling, and an Amorphous Diamond Evaporated Diaphragm for much more natural sound.

Same features as the MDR-V600 PLUS—

- 50mm Aura-Nomic designed driver
- 3000 mW power handling
- Class 1 oxygen-free copper (LC-OFC) connecting cord for maximum conductivity
- Amorphous diamond evaporated diaphragm stiffen the drivers to maximize their tonal qualities while minimizing additional weight for much more natural sound.
- Amorphous diamond evaporated diaphragm stiffen the drivers to maximize their tonal qualities while minimizing additional weight for much more natural sound

MDR-CD580 Digital Reference Headphones

A high-quality closed-front open-back headphone, the MDR-CD580 features large 40mm drivers and Neodymium magnet for deep bass, wide dynamic range and frequency response. Also offers a double-band headband, UniMatch plug adapter and detachable cable.



- 40mm drivers provide deep bass, low distortion and wide dynamic range. Frequency response extends all the way from a low 10Hz to 25kHz.
- Aura-Nomic design positions the earpads in accordance with the angle of the head.
- Anti-Resistant ABS Resin Baffle Board minimizes resonance to produce purer sound
- Neodymium magnet for maximum energy in a minimum size.
- Detachable OFC cable and gold plated Unimatch plug for /8″ mini-jack and 1/4″ phono connectivity
- Self-adjusting double-band headband provides a customized fit and increased comfort
- Cloth earpads are removable and washable

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STUDIO MONITOR HEADPHONES

MDR-CD780

Digital Reference Headphones



A high-quality closed-front open-back headphone, the MDR-CD780 features large 50mm drivers and Neodymium magnet for superb audio performance. For convenience, it offers 25mm thick Urethane earpads, double-band headphone and a detachable cable.

Same features as the MDR-CD580 PLUS—

- 50mm drivers — larger than many headphones — delivers deep bass, low distortion and wide dynamic range. Frequency response extends all the way from a low 5Hz to 30,000 Hz.
- 25mm thick Urethane earpads distribute pressure evenly around the ear for exceptional comfort over extended listening sessions



MDR-CD2000

Digital Reference Headphones

In addition to large 50mm drivers and Neodymium magnet for superb audio performance, the MDR-CD2000 incorporates a Bio-Cellulose diaphragm blended with Vectran to enhance natural sound reproduction. 25mm thick Urethane earpads ensure exceptional comfort.

Same features as the MDR-CD780 PLUS—

- Bio-Cellulose diaphragm blended with Vectran enhances natural sound reproduction

MDR-CD3000

Digital Reference Headphones



The top-of-the-line, the MDR-CD3000 has a large circumaural design, floating ear cup system, organic diaphragm and earcup housing, and a unique earpad design to deliver performance so superb, it is matched only by the perfection of digital sound itself.

Same features as the MDR-CD2000 PLUS—

- Super-large 50mm drivers with CCAW (Copper Clad Aluminum Wire) construction for increased power handling capacity, light weight and superior frequency response
- Natural fiber Bio-Cellulose diaphragm is organically grown for enhanced sound velocity and natural, delicate music reproduction
- Floating earcup system improves resonance characteristics and sonic performance
- Organic fiber composite earcup housing provides a personal concert hall for the ear with high resonance suppression for pure, natural sound reproduction
- Voice coil with CCAW (Copper Clad Aluminum Wire) construction for high sensitivity
- Litz Wire connector cord made of 6N oxygen-free copper, provides exceptional conductivity for maximum signal transmission and clearer sound reproduction
- Large-sized circumaural design provides an excellent fit for any size head and hours of tireless wear.
- Unique earpad design uses a collagen-like material for a natural feeling of comfort

STUDIO MONITOR HEADPHONES

MDR-7502

Designed for a broad range of applications, for general sound monitoring in the studio or the field. The rugged MDR-7502 is a high quality, low-cost alternative to consumer headphones.

- 60Hz~16kHz frequency response
- 45Ω impedance
- Sensitivity is 102dB
- 30mm driver unit delivers clear, high quality sound reproduction.
- Closed-ear design for comfort and reduction of external noise interference.
- Gold connectors and OFC cord - reliable and stable signal connection and transmission.
- Stereo uni-match plug - 1/4″ and 1/8″ applications.

MDR-7505

The MDR-7505 is a professional stereo headphone designed to meet the needs of professional musicians, broadcasters and DJs on a budget. Their durable and flexible design makes them ideal for recording, control room and live DJ applications.

- Swivel mechanism allows for various wearing positions
- Acoustically sealed ear-cup allows for easier listening in loud environments.
- 40mm driver
- Neodymium magnet for powerful bass and clear highs
- Gold-plated, uni-match connector for universal compatibility with desk top and portable devices.
- Convenient folding design for easier portability
- Extendible, LC-OFC coiled cord for low-loss signal transmission.





MDR-7506

Due to its low impedance, the closed ear MDR-7506 does an outstanding job of cutting down background noise while providing plenty of volume in the studio or field. Ideal for use with MIDI workstations, camcorders or other equipment with less than powerful amplification.

- 15 to 18kHz frequency response
- 63 Ohms impedance
- Sensitivity is 106dB
- Rugged design - proven to be reliable in the toughest situations.
- 40mm driver unit for clear, high quality sound reproduction.
- Folding construction - compactness in storage.
- Closed ear design for comfort and reduction of external noise interference.
- Stereo uni-match plug - 1/4″ and 1/8″ applications.
- Gold connectors and OFC cord - reliable and stable signal connection and transmission.
- Supplied soft case for protective storage.

MDR-7509

The MDR-7509 is a high-quality stereo headphone for critical and demanding professional applications, including studio and broadcast monitoring.

- Circum-aural design covers the entire ear for tight acoustic seal and extreme comfort.
- 5~30kHz wide frequency response
- 50mm drivers for superior bass response and low distortion.
- Neodymium magnet for superior bass response and sensitivity Amorphous diamond evaporated diaphragm provides natural sound reproduction.
- Self-muting mechanism prevents leakage when headphones are taken off.
- Reversible ear-cup design allows single-sided monitoring.
- Gold-plated uni-match plug, with 1/4″ and 1/8″ adapters
- Convenient folding design
- Coiled LC-OFC cord for low signal loss.

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EQUIPMENT LEASING AVAILABLE

NOISE CANCELING HEADPHONES

With these headphones you can actually reduce noise by up to 70% when noise canceling circuit is switched on. Noise canceling circuit actually senses outside noise with built-in microphones and sends an equal-but-opposite canceling signal to the headphones; cancels noise from 50Hz to 1,000Hz, with a maximum of 10dB noise cancellation in the critical range from 100 to 400 Hz --where the human ear is super-sensitive

MDR-NC5

By reducing outside noise by more than 70%, the MDR-NC5 lets you block out noise, listen to music, or do both at once! Ideal for sleeping on planes, trains or buses. Equally great for outdoor use, either walking on a noisy street or gardening in your yard.



- 30mm diameter driver delivers deep bass down to 30 Hz; provides outstanding dynamic range
- Uses Neodymium magnets for maximum energy and minimum size.
- Open-air headphone design provides lightweight comfort, hour after hour
- Up to 46 hours on a single AAA battery
- Folding headband and pivoting earpads allow easy storage in a briefcase or purse
- Noise canceling On/Off switch with LED indicators. Headphones continue to function for music listening even when noise cancelling is off.
- Includes a carrying pouch and airline plug adapter for compatible in-flight audio systems

MDR-NC10



The MDR-NC10 reduces outside noise by more than 70%, letting you block out noise, listen to music, or do both at once! Similar to the MDR-NC5, the MDR-NC10 uses a snug, yet comfortable Fontopia in-the-Ear design eliminating the need for headphones.

- Fontopia in-the-ear design fits snugly, yet comfortably inside the ear - eliminating the headband, so you can enjoy music while you're really moving – while exercising, skating or dancing.
- Hot-line switch to temporarily admit outside sound
- Additional noise reduction thanks to the structure of the headphones, which serve to physically block outside noise.
- Noise canceling On/Off switch with LED indicators.
- Up to 60 hours on a single AAA battery
- Detachable input cord to connect an audio source and listen to music
- Gold-plated mini-plug provides exceptional corrosion resistance, superb conductivity and low noise

MDR-NC20

Incorporating noise canceling circuitry and a closed headphone design, the MDR-NC20 provides even greater isolation from outside sound. Block out noise, listen to music, or do both at once!



- 30mm diameter driver delivers deep bass all the way down to 16Hz; provides outstanding dynamic range
- Closed headphone design provides even greater isolation from outside sound
- Folding headband and pivoting earpads allow storage in a briefcase or purse
- Detachable input cord to connect an audio source and listen to music
- Noise canceling On/Off switch with LED indicators. Headphones continue to function for music listening even when noise cancelling is off.
- Gold-plated mini-plug provides exceptional corrosion resistance, superb conductivity and low noise
- Up to 45 hours on a single AAA battery
- Includes a carrying pouch and airline plug adapter for compatible in-flight audio systems

HEADPHONE AMPLIFIERS

Symetrix 304



This stereo input, four output amplifier drives headphones of any impedance. Used in broadcast, installed sound, and recording applications, high voltage technology provides 6dB volume increase over most competing units. Compact half rack chassis permits 8 channels in one rack space.

- Individual volume controls for each output, master input volume.
- Stereo or mono operation, balanced or unbalanced input.
- Pass Thru jacks permit daisy-chaining of multiple units.

Shure FP-22 Headphone Amplifier



The FP-22 is a battery-powered stereo headphone amplifier that provides headphone feeds from any audio input source. It has the unique capability of monitoring mono, stereo and mixed mono and stereo inputs. It monitors and combines these signals in the headphones without affecting the original signal. With 81dB of gain, it can monitor a wide range of signal sources in many different configurations. Ideal for field production, studio recording and trouble shooting applications.

- Loop through 1/4″ stereo phone jacks (can be internally switched to become separate left and right inputs).
- Overall headphone balance control, overall stereo/mono switch for headphones.
- Independent control of headphone volume as well as loop-through volume
- Balanced loop through XLR connectors are mic/line switchable. A 30dB pad attenuates loud line level sources.
- 1/4″ and 3.5mm stereo jacks
- Powered by 9v battery or external 12-24v DC power supply.

Behringer HA4600 Powerplay Pro

4-Channel Stereo Headphone Amplifier



Developed for the most demanding broadcast, digital production and professional recording applications, the HA4600 is ideal as a reference unit to assess the quality of a final mix, and as a distribution amplifier for the addition of playbacks in studio use. The HA4600 offers premium sound quality even at maximum volume and houses four independent high-power stereo amplifiers each with a stereo aux input, mono/stereo and left/right mute switches, balance and volume controls and more, plus numerous master features, in a single rack space.

- Three outputs for each channel (one front, two rear) enable connection of up to 12 headphones.
- Ultra musical 2-band EQ per channel
- Independent input/aux sections provide up to four stereo mixes.
- Stereo aux inputs for each channel permit the inclusion of additional sound sources.
- Adjustable direct in on the front panel for additional stereo signal source.
- Paralleled main outputs allow cascading of several HA4600's
- 8-digit LED output meters for each channel enable precise level control.
- Servo-balanced, gold-plated XLR and 1⁄4″ TRS inputs and outputs

Fostex PH-50 Headphone Amplifier



The PH-50 is a remarkably quiet headphone DA (distribution amplifier). Each output has its own amplifier as well as its own level control - so individual requirements can be met with ease. Multiple PH-50s can also be cascaded to handle a large number of listeners making it ideal for orchestral, multimedia, education and trade show applications.

The PH-50 is especially handy in the studio environment because an output from the mixer can send a talkback signal to the Aux input (complete with level control) of the PH-50 & be distributed to all headphone outputs. This great cue system makes for more effective communication between engineer/producer & musicians.

- 5 headphone outs - each with its own amplifier
- Each input routes to its direct output, plus Aux 1 source (cue feed)
- Multiple inputs: balanced XLR, phone, unbalanced RCA jacks
- +4dBu or -10dBV switchable for balanced inputs
- Individual level controls with peak reading LED
- Stereo or mono input select
- Cascade inputs & outputs for 'daisy chaining' units
- Aux In with level control for cue feed from main mixer
- Freq. response is 30Hz - 20kHz

HC-4

Four Stereo Headphone Amplifier

Four people want to listen to the same mix under headphones. Simple, just take the HC 4, plug in, turn on, adjust your individual volumes, and groove.



- Master stereo Inputs which can be used to drive the four stereo headphone amplifiers.
- A Mono switch is provided on the front panel to drive both channels of all four amplifiers from a common mono source.
- Delivers up to 200 milliwatts into headphones with rated impedances between 32-600 ohms (depends on the exact impedance of the headphone in use and the number of headphones connected to the unit).
- Under most conditions, the HC 4 produces sound pressure levels in excess of 120 dB. This level is more than adequate for most listeners.

HC-6 *Six Stereo Headphone Amplifier*



Featuring six stereo headphone amplifiers, the HC 6 performs a number of necessary tasks, not all of which are headphone related. The HC 6 provides Master stereo Inputs which can be used to drive any or all of the 6 stereo headphone amplifiers. A Mono switch is provided on the front panel to drive both channels of all 6 amplifiers from a common mono source.

- Six individual unbalanced stereo inputs are available to drive any individual amplifier from a separate stereo source.
- Under most conditions, the HC 6 produces sound pressure levels in excess of 120 dB. More than adequate for most listeners.
- Outputs for all six channels are provided on the front panel and on the rear panel.
- Two pairs of headphones may be driven from each channel.
- Inputs can be converted to mono balanced Inputs through internal jumpers if required.
- Delivers up to 450 milliwatts into headphones with rated impedances between 32-600 ohms. The precise amount of power depends on the exact impedance of the headphone in use and the number of headphones connected to the unit.

SAMSON S•PHONE

Four-Channel Headphone Mixer/Amplifier



The S•Phone is a versatile single rack space four channel headphone amplifier, loaded with advanced features that provide tremendous flexibility for monitor and cue mixing. The Master Input features an input level and LED meter to control and display the overall stereo input. A convenient Stereo Auxiliary input is provided on each channel allowing individual line signals from direct or buss outputs to be mixed with the main stereo input. A Balance control adjusts the ratio of the Main and Auxiliary signal or the left to right balance (based on the setting of the ST/2CH switch) so that each headphone mix can be further tailored for a particular listener's individual needs ("More Me").

- Two master stereo inputs plus stereo link output
- 8-segment LED metering on each channel and master
- 2-band EQ on each of the four channels
- Three headphone outputs per channel (one front and two rear panel) with an overall volume control and individual level meters.
- Auxiliary input with balance control provides four channels of "More Me" mixing
- Mute and Stereo/2-channel mode switches on each channel
- Maximum output power on each channel (regardless of headphone impedance)
- Three-year warranty

HEADPHONE AMPLIFIERS

HA-43
4-Channel Headphone Amp



The HA43 is for expanding or generating headphone channels. It has a 1/4˝ stereo input, 4 volume controls, 4: 1/4˝ stereo output jacks and is powered by an AC adapter (included). The HA43 has a 20Hz - 20K Hz response, small size, and is very easy to use.



PA-71
7-Watt Amp

The PA71 Plus mixes an XLR microphone signal with stereo (mixed to mono) RCA signals. The combined signal is sent to spring loaded output terminals via a seven watt power amplifier. The unit also features a 1/4˝ Line/Powered output. The XLR input features switchable 12v DC phantom power. This PA-71 PLUS is deal for small room applications where a simple mix of a mic with a cassette or CD player is needed.

RA-53 5-Channel Headphone Amplifier



The RA53 is a five-channel headphone DA (distribution amplifier) with a variety of input options. There are four different ways to send a master mix to the RA53; Discreet 1/4˝ Right and Left inputs, Right and Left RCA jacks, a 1/4˝stereo TRS plug connected to the Right input, or a 1/4˝ mono TS plug connected to the Left input. In addition to the four master mix input configurations, three separate mixes can be monitored using the main inputs and the two Insert jacks on Channels 1 and 2. Inserting a 1/4˝ TRS jack into Inserts 1 and/or 2 interrupts the master signal, sending different mixes to Channels 1 and 2, leaving the master mix on Channels 3 through 5.

- A 1/4˝ Link Input/Output jack is provided for connecting two RA53s together, allowing up to ten stereo channels of headphone monitoring.
- Each channel has a Level control and a stereo 1/4˝output jack.
- The RA53 is loud, has plenty of gain for operation in noisy environments, and works with most any headphone impedance.

RA62HA 6-Channel Headphone Amplifier



The RA62HA is a six-channel headphone amplifier for high quality audio monitoring. Since the Mono/Stereo switch is after the pan control, it will mix 2 Mono signals and their balance can be adjusted with the Pan control. Up to six individual monitor mixes may be listened to via the rear panel insert jacks. There are two auxiliary inputs (right and left) on the rear panel.

- 6 Headphone outputs with level controls
- 6 Insert jacks for separate use of channels
- Up to one-half watt output per jack
- Pan control
- Stereo/Mono switch allows mixing 2 mono channels
- Front and rear input jacks
- DC to 30khz response for accurate sound

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HEADPHONE AMPLIFIERS

PM52 Headphone Tap



The PM52 is for connecting headphones to a speaker line. The 1/4˝ mono/stereo loop jacks are for connecting many boxes in series. There are 2 headphone jacks and volume control. The PM52 uses the speaker line signal for power, and uses no batteries.

The PM50s/PM50sOB Personal Monitor Amplifiers

The PM50s and PM50sOB mix a microphone signal with a line-level stereo monitor signal to two headphone outputs. The microphone signal is passed through (hard-wired for no signal degradation) to a mixer or pre-amplifier. They are ideal for performers who need to monitor their own signal, such as their vocal, with the main mix.



- Both units feature a monitor level control, mic level control, mic In and pass through XLR jacks, two headphone output jacks, and a stereo TRS 1/4˝ monitor input jack.
- The ideal monitor tools for drummers who need to hear a click track, choir singers who need to hear themselves, etc. Also useful as a line tester for troubleshooting an audio system.
- The PM50s works on AC only and includes an AC adapter. The PM50sOB can be powered via 9 volt battery or AC with the optional PS27 AC adapter.

HM Series

The HM Series headphone mixing system offers a great-sounding, cost effective and flexible way to provide custom monitor mixes in recording and live sound environments. The combination of unique functions and signal processing add functionality and convenience to any single or multiple-user monitoring application. The system provides each musician with six volume controls: four monaural (for mixer channels or busses), plus one for an overall stereo mix and one for a stereo effects return.





- The HM60 master unit can stand alone or serve as a front end for HM59 and/or HM58 remotes.
- The remotes come with a 25' DB25 MF cable and can be “daisy chained” around the stage or studio as desired. The DB25 cable is a standard 25 pin computer serial cable and is available in several lengths.
- The HM60, HM59 and HM58 units are ideal as low distortion headphone drivers for the most critical listening situations.
- The HM57 Headphone Monitor interface provides an inexpensive solution for getting the audio out to the HM59 and HM58 Remotes.
- They have mixing capabilities, as well as important signal processing functions to provide a clear and comfortable listening environment.
- The HM60 and HM59 each have a nine-band graphic equalizer, and the HM58 has Bass, Mid, and Treble controls.
- Each unit employs a Rolls 2:1 AGC compressor to provide even signal dynamics.
- In addition to providing high current differential buffered signals, the HM60 and HM57 also provide power and ground to the HM58 and HM59 remote mixers.
- The units can be clamped to any mic stand using the optional MSC106 Microphone Stand Adapter.
- They are housed in rugged painted steel chassis, use only the finest quality components, and come complete with a limited One-Year Warranty.

HA-6AB

Headphone/Monitor Amplifier



The HA-6AB is an economical 20 watt per channel stereo amplifier. However, unlike a conventional power amp, it has six headphone outputs (stereo phone jacks) on the front panel, each with volume control. When used for overdubbing or rehearsal, this provides more than adequate loudness in all headphones (even older low-impedance models), and allows each musician to set his or her own comfortable volume level. When the overdubbing is complete and the musicians remove their headphones, the HA-6AB can power the tape playback through either (or both) of two sets of studio monitor speakers.

- Combination headphone amp and a conventional power amp—powers headphones and speakers for added space/cost-savings.
- Utilizes an efficient toroidal power transformer for silent operation and low magnetic leakage,
- Two balanced XLR and two unbalanced 1/4" phone inputs
- 6 headphone outs, each with volume control
- With 20 watts-per-channel the HA-6AB can drive headphones of any impedance to their full rated listening level, and do so with minimal noise and distortion.
- Just right for the small "cube" reference speakers used in most recording studios as a comparison with the main control room monitors—the HA-6AB powers them without overpowering them.
- Convenient front panel input level control.
- Switches for two sets of speakers
- Automatically switches to mono mode when right channel is not used
- Overload and signal-present LEDs for each channel; ground lift, power-on indicator
- Provision for remotely controlling volume
- Fully protected against thermal overload, it can also withstand a short circuit on any or all outputs for an indefinite time without damage.

Optional HR-2
Headphone Remote Station

Designed for the HA-6A, the HR-2 is a compact, unobtrusive box that clamps to a mike stand or belt, providing two headphone jacks, each with its own volume control. Any number of HR-2 stations may be connected to a HA-6A using standard mic cables. Comes with snap-on, snap-off mike stand clamp.

HDS-6 / HR-6

Headphone Audio Distribution System and Remote Mixer



The HDS-6 with the HR-6 offers a great-sounding, cost effective and flexible way to provide custom mixes for up to sixteen people in recording and live sound environments. The combination of unique functions add unprecedented functionality and convenience to any multiple-user audio application.



The system consists of two components: the rackmount HDS-6 Audio Distribution System, and one or more HR-6 remote mixers. Linking them is as easy as plugging in cables — multiple HR-6's are simply daisy-chained together. A pair of 25-foot linking cables is included with each HR-6. For applications where daisy-chaining is not practical, a twelve port or larger ethernet patch panel can be wired to allow multiple HR-6 units to connect from a central location. Up to eight HR-6 remote mixers can be linked to an HDS-6, and two pairs of headphones may be plugged into each HR-6.

HDS-6 / HR-6, *continued*

HDS-6

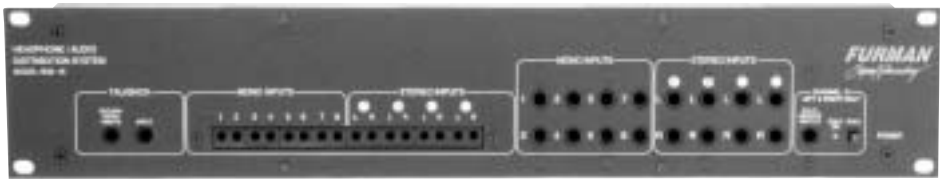
- Every performer in a studio or live sound setting can now have a personalized mix without any adjustments at the board. Perfect for use as a headphone system in the studio, plus drives powered monitors in live sound applications.
- HDS-6 provides gain trims and overload LEDs for each input, ground lift and on/off switch, and system status LEDs
- The HDS-6 connects to the mixer/console to provide interface and power supplies to drive a group of HR-6 remote mixers. Trimpots are provided on the front panel to adjust input levels.
- Ideal as a low distortion headphone driver for critical listening situations. In addition to providing high current differential buffered signals, it also provides power and ground to the HR-6 remote mixer.

HR-6

- The HR-6 is a compact six-channel, five-pot remote mixer that clamps to any mic stand. When used with the HDS-6, the HR-6 allows musicians to customize their own headphone or monitor mix, and the engineer doesn't have to touch the board.
- A "Submixes Included/Excluded" button mutes the four mono pots, allowing only the stereo source to be heard, without having to alter the HR-6's four monaural settings.
- The HR-6 mixer provides each musician with five volume controls: four monaural (for mixer channels or busses), plus one for an overall stereo mix or a stereo effects return
- 25-ft. linking cables are included with the HR-6, as is a universal mic stand clamp

HDS-16 / HRM-16

Headphone Audio Distribution System and Remote Mixer



An enlarged and enhanced version of the HDS-6/ HR-6, the HDS-16 Headphone Distribution System and HRM-16 Personal Headphone Mixer are designed to function equally well for both live (in-ear or stage monitors) and studio applications, allowing greater flexibility and ease for both the user and the engineer.



The HDS-16 with the HRM-16 cleanly routes 8 mono and 4 stereo signals to remote mixing locations. Up to eight HRM-16's can be connected to a single HDS-16, allowing as many as sixteen users (two per each HRM-16) to customize eight different mixes. A unique talkback system allows each user to communicate with other users and the main mixing location. Additionally, a SOLO system allows the engineer to mute all but one stereo send from the mix location- very handy for reviewing recent takes or rough mixes. Additional HRM-16 features include local effects send/return, line out, overall bass/treble controls, talkback level, and a master level control.

HDS-16

- Provides full patching functions on its front panel via 1/4" and TT input jacks.
- Provides power to drive the HRM-16. In addition, it includes two 1/4″ jacks for the two-way talkback functions, and a SOLO REMOTE switch with an indicator that allows muting of all but one stereo feed at all HRM-16's—ideal for isolating tracks for critical listening without having each HRM-16 user turn down all but one channel.
- The rear panel includes six Centronics 50-pin connectors to link up to six HRM-16'.
- Talkback system is configured two ways:

—Allows the talkback mic on each HRM-16 to be fed to all of the other HRM-16 units as well as the HDS-16.

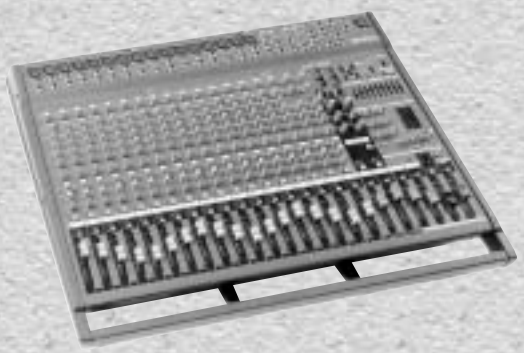
—Allows communication to the HDS-16 only. This is useful for stage use where the user's talkback mic shouldn't be fed into all of the monitors (including the users).

HRM-16

- Attaches to a mic stand with included hardware. Includes flexible 25-ft. cable that can be extended or routed through walls and daisy chained to other HRM-16s.
- The eight mono and four stereo signals have level controls.
- Each mono channel also has a pan control and an effects send control which feeds its own rear panel 1/4″ send/return jacks; the affected signal is returned into the mix and controlled via a stereo effects return knob.









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The Professional Audio SourceBook

Section 9

Sound Reinforcement/PA

Anchor **662-663**

EV **664-670**

FBT **671**

Fender **672-673**

JBL **674-681**

Mackie **682-687**

Samson **688-693**

Yamaha **694-701**

XTREME & LIBERTY SOUND SYSTEMS

AC- and DC-Powered Portable Sound Systems



With a built-in CD player and up to three built-in Anchor Wireless UHF Receivers, the Xtreme and Liberty expands their capabilities beyond any other system in the industry.

An AC-powered sound system, the 130-watt Xtreme delivers a powerful 123dB of crystal clear speech projection—enough to reach crowds of up to 2,500 people. The 50-watt Liberty is a DC-powered system that delivers 117dB of clear sound to crowds of 1000 or more.

With the convenience of a built-in CD player and a wireless microphone, the Xtreme is great for large indoor and outdoor events such as press conferences, sports events, graduation ceremonies, musical productions or any other settings needing powerful and clear voice or music amplification. Rugged, reliable and crystal clear, Liberty is used at the White House for Presidential press conferences, at thousands of schools around the world, at corporate training centers and on many U.S. aircraft carriers.

Xtreme

- Clear and powerful sound, thanks to an 8″ woofer and constant-directivity horn inside a ported enclosure.
- Delivers up to 123 dB of clear sound
- Handles crowds of 2,500 or more.
- Biamplified; separate high and low frequency amplifiers for enhanced performance.
- Active limiter—keeps sound clean even when driven hard.
- The XLR input allows the use of line level or microphone sources, with or without condenser mic power
- Line-level, balanced output supplies signal to additional sound system
- Neutrik Speakon connector for use with XTR-6001
- Bass and treble controls
- Speech projection mode provides crystal-clear voice projection
- Comes with a hassle-free 6-year warranty

Liberty

- Comes with a hassle-free 6-year warranty.
- Tough, injection-molded case is remarkably durable
- 8″ woofer and titanium dome compression driver in tuned enclosure
- Features lead-acid batteries and a built-in charger
- Handles crowds of 1,000 people or more
- External speaker output powers optional companion speaker
- Integrated circuit amplifier with short circuit and thermal overload protection
- Speech projection mode (on/off selectable) provides crystal-clear voice projection



XDP-6000
Xtreme Deluxe
2-Speaker Package

XBP-6000 Xtreme Basic Package:

Includes Xtreme XTR-6000CU1 sound system, built-in CD player and built-in Anchor Wireless UHF with16-user selectable channels, speaker stand (SS-550) and choice of wireless mic: handheld mic (WH-3000) or body-pack transmitter (WB-3000) and headband mic (HBM-3000) or lapel mic (LM-3000) or collar mic (CM-3000)

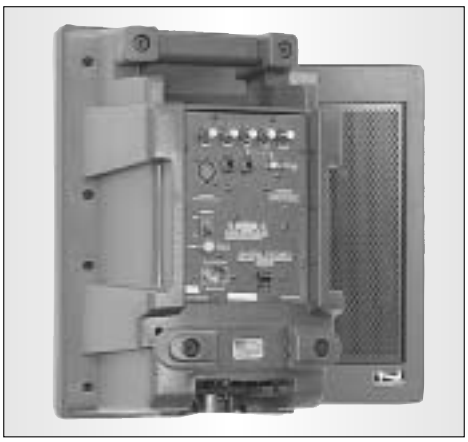
XDP-6000 Xtreme Deluxe 2-Speaker Package:

Includes Xtreme XTR-6000CU1 sound system, built-in CD player and built-in Anchor Wireless UHF with16-user selectable channels, unpowered companion speaker (XTR-6001) two speaker stands (SS-550), 50' cable connection (SC-50NL) and choice of wireless mic: hand-held mic (WH-3000) or body-pack transmitter (WB-3000) and head-band mic (HBM-3000) or lapel mic (LM-3000) or collar mic (CM-3000)

EXPLORER SOUND SYSTEMS

Strong, Full-Range Sound for Crowds Up To 250 People

Explorer is a favorite of audio rental companies and hotels for a quick setup in a conference room. Weighing 20 lbs. this sound system has also traveled the world with missionaries and aboard cruise ships. Delivers a great balance of speech clarity and music fidelity. (A battery-powered model is also available.)



- Heavy-duty amp and full-range speaker system deliver fidelity sound
- 6.5″ woofer coupled with a high-output horn tweeter
- Simple to set up and operate
- Durable injection-molded polypropylene case
- Short circuit and thermal overload protection
- Weighs less than 20 pounds
- Speaker output to power companion speakers
- Comes with a 6-year warranty

EBP-2500 Explorer Basic Package:
Includes Explorer PA-2500W sound system, (SS-550) speaker stand and choice of wireless mic: (WH-58) Shure handheld mic or (WB-II) body-pack transmitter and (HBM-60) headband mic or lapel mic (LM-60) or collar mic (CM-1000).

EDP-2500 Explorer Deluxe Package:
Includes Explorer PA-2500W sound system, Explorer P-2501 companion speaker, two (SS-550) speaker stands and choice of wireless mic: (WH-58) Shure handheld mic or (WB-II) body-pack transmitter and (HBM-60) headband or lapel mic (LM-60) or collar mic (CM-1000).

MegaVox PA System

Police, firemen, coaches, FBI agents, Marine drill sergeants, auctioneers, all choose the MegaVox. This PA system delivers an amazing 126dB of clear speech projection that can operate for up to 10 hours on a charge. Incorporating a rechargeable battery, this unit weighs just 15 lbs. and is available with wireless microphone for convenient field operation.



- Injection-molded case
- High-efficiency re-entrant horn driver
- Speaker output powers optional companion speaker
- Built-in wireless microphone receiver
- Powerful amplifier and horn driver combine for 126 db output
- Runs up to 10 hours on a charge
- Speaker stand receptacle molded into bottom
- 6-year warranty

MEGA-BP MegaVox Basic Package: Includes MegaVox with built-in 2-channel selectable wireless receiver, (SS-550) speaker stand and choice of wireless mic: (WH-1000) handheld mic or (WL-1000) body-pack transmitter and (HBM-40) headband mic

MEGA-DP MegaVox Deluxe Package: Includes MegaVox with built-in 2-channel selectable wireless receiver, MegaVox P-351 unpowered companion speaker, (SC-50) 50" cable connector and two (SS-550) speaker stands and choice of wireless mic: (WH-1000) handheld mic or (WL-1000) body-pack transmitter and (HBM-40) headband mic.

AN-100 Sound Systems

AN-100 packages are the perfect choice for classrooms small conference rooms and corporate road warriors who rely on their sound systems to command the audience. Plus, it's the perfect companion for a video projector. The AN-100 produces such accurate sound that it is used by movie studios to hear their mix on the master sound tracks. Weighing in at just 8.5 lbs. this powerful sound system is designed for audiences up to 100 people.



- 25 watt amplifier provides lots of sound.
- 2-way speaker system combines 4.5″ woofer and dome tweeter
- Durable injection-molded case
- 6 year warranty
- Magnetically shielded speaker.
- Has mic, instrument & 2 line inputs
- Built-in two selectable channels RF wireless receiver

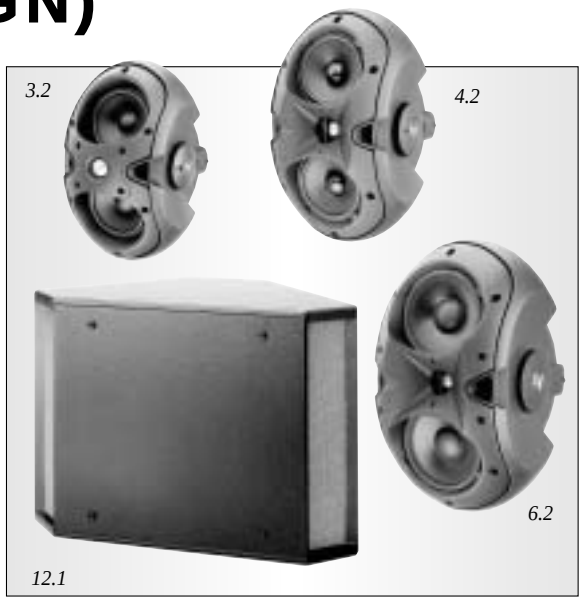
AN-100:
AN-100 all by itself with nothing else. Just plug in a microphone or the video projector cables, and you're ready to go.

AN-100BP AN-100 Basic Package:
Includes AN-100 Wireless VHF Receiver and choice of wireless mic: (WH-1000) handheld mic or (WL-1000) body-pack transmitter and (HBM-40) headband mic

EVID (EV INNOVATIVE DESIGN)

Premium Surface Mount Speaker System

The EVID concept means innovation. From it's unique, superior design to its innovative acoustical integrity, EVID delivers value, performance and ease-of-use to the contractor and end listener alike. In designing EVID, Electro-Voice evaluated all the current products available and asked contractors and typical end users, "What do you want?" End users wanted something attractive and superior sounding amidst a world of plain black boxes. Contractors wanted a comprehensive line that would provide flexible choices for easier system design and installations. EVID achieves these goals. The result is a comprehensive background/foreground speaker line with the key characteristics installers need and users want. Why settle for anything else?



FEATURES

- Cast aluminum Strong-Arm-Mount (SAM) system provides excellent strength and reliability
- Dual woofers for extended bass, greater power handling and output
- Coherent coverage Waveguide Horn design on models 4.2 and 6.2
- Dual low frequency ports provide extended bass output
- Treated woofer cones for added weather resistance and increased reliability
- Over 100° of swing and 45° of rotation for incredible installation flexibility
- Titanium dome tweeters with Neodymium magnet structures provide clean highs and added weather resistance
- One-piece ABS cabinet provides outstanding strength and acoustic rigidity
- Durable zinc-plated steel grille
- Simple keyed hex head adjustments make installations quick and easy
- Smooth lines and innovative cosmetic design complement any décor or environment
- Magnetically shielded transducers allow placement close to sensitive equipment

3.2 Very Compact

Full-Range Speaker System

Loaded with dual 3.5″ LF drivers and a 3/4″ Ti direct radiator, the EVID 3.2 produces sound that defies its small size. Its unobtrusive profile is ideal for background music and paging support in a wide variety of environments (e.g., offices, retail stores, restaurants, etc.) Coupled with the 12.1 Subwoofer, two to four EVID 3.2 speakers deliver amazing SPL levels for foreground music.

4.2 Compact

Full-Range Multi-Use Speaker System

Perfect for a variety of environments where full-range audio is needed but space is limited: restaurants, bars, bistros, department stores, and outdoor venues. Offering the bass response of a much larger speaker, dual 4-inch woofers make the 4.2 a phenomenal performer. The 1″ waveguide coupled titanium tweeter delivers controlled clean mids and highs. Combine it with the 12.1 Subwoofer for an amazing listening experience. Full system protection and EV's unique Strong-Arm-Mount (SAM) make the EVID 4.2 a solid choice in any environment.

6.2 Extended-Range

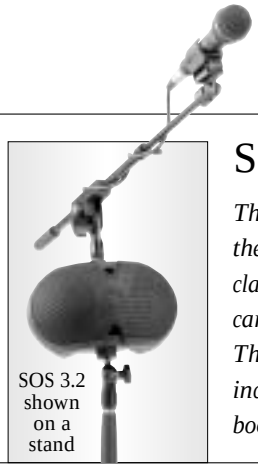
High Output Speaker System

Comprised of dual 6″ LF drivers and a 1″ Ti/waveguide, the EVID 6.2 is perfect for applications that require exemplary audio and the capability to fill large areas with sound. Additionally, the flexible Strong-Arm- Mount (SAM) system provides the versatility to accommodate any installation challenge. EV's unique 6″ dual woofer design delivers crisp and clean program for the utmost in listening experience regardless of the location. Great for larger areas that demand higher SPL levels, the 6.2 delivers the outstanding, full bass output required for shopping malls, stadiums, airports, etc.

12.1 Dual Input Subwoofer

To augment and extend the low frequency response of any full-range models, an attractive and powerful subwoofer completes the EVID line. The 12.1 provides an easy installation by incorporating dual voice coils to handle both channels of a stereo pair. In addition, the unit can power up to four satellites with its four crossed-over audio outputs on the back panel. For extended bass performance in demanding installations.

EVID (EV INNOVATIVE DESIGN)



SOS 3.2 shown on a stand

SOS 3.2 Monitor Speaker *Mic Stand Personal Monitor*

The EVID SOS 3.2 is a new approach to personal monitors! Its unique design enables the use of only one stand for the microphone and monitor. It frees up stage space and puts the monitor closer to the performer for additional clarity. The SOS 3.2 features a 1/4˝ input connector for easy connection to the amplifier. In addition, the performer can precisely adjust the speaker volume level using the L-Pad adjustment on the back of the unit.

The compact design of the SOS 3.2 makes it unobtrusive on stage and easy to pack and carry. The SOS 3.2 package includes the SOS 3.2 speaker, all stand mounting adapters and a shock mount clip ready for use on your favorite boom-mount mic stand.

EVID ACCESSORIES

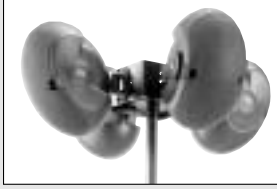
AB-34 and AB-64 360° Mounting Bracket Kits



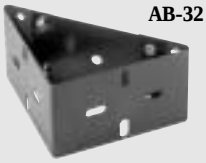
AB-34

AB-34 and AB-64 series kits feature a 4-sided powder coated steel mounting assembly that allows the mounting of 4 EVID speakers into a 360 degree pattern array configuration. The mount offers additional flexibility in your EVID system designs. The AB-34 works with the EVID 3.2 series and can be used with a threaded rod for suspension from the ceiling. The AB-64 is used with the EVID 4.2 or 6.2 series speakers and can be used with a speaker stand tripod or hung from the ceiling using a threaded rod. All necessary accessories and parts needed for use are included.

AB-34 shown with optional speakers



AB-32 and AB-62 180° Mounting Bracket Kits



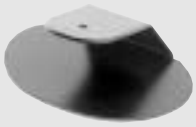
AB-32

AB-32 and AB-62 series kits feature a 2-sided powder coated steel mounting assembly that allows the mounting of 2 EVID speakers into a 180 degree pattern array configuration. The AB-32 works with the EVID 3.2 series and can be used with a threaded rod for suspension from the ceiling. The speakers can then be positioned on the bracket to cover a variety of patterns to best serve the installation requirements. The AB-62 allows the EVID 4.2 or 6.2 series speakers to be used with a speaker stand tripod or hung from the ceiling utilizing a threaded rod. All necessary accessories and parts needed for use are included.

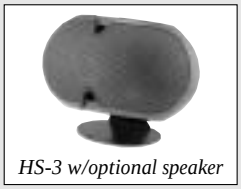


AB-32 shown with optional speakers

HS-3 Horizontal Tabletop Stand

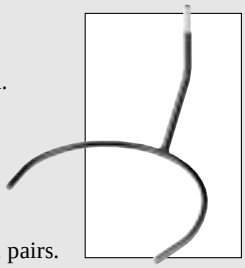


The HS-3 horizontal tabletop stand allows the EVID 3.2 or 4.2 to sit on a table, meter bridge, desk, bookshelf or other flat surface. Ideal for portable applications, the stands are made of steel for strength and durability and include rubber feet to protect surfaces. Sold in pairs.



HS-3 w/optional speaker

VS Series Vertical Tabletop Stands



VS series vertical tabletop stands are available for any EVID full range model. The stands allow the hassle-free vertical orientation of EVID speakers on a table, desk, bookshelf or other flat surface. Ideal for portable applications, the stands are made of plastic coated steel for strength and durability. Sold in pairs.

TC Series Terminal Covers

TC-4 and TC-6 terminal covers protect the input connections on EVID speakers from the long-term effects of moisture. The covers are available for the EVID 4.2 and 6.2 speakers. The covers easily attach to the rear input panel of the speaker and include a weatherproof cable connector.



MA-3 Mic Stand Adapter

The MA-3 microphone stand adapter allows use of the EVID 3.2 with a standard boom mount mic stand. This unique kit includes adapter pieces for both the mic stand and boom sections. These adapter pieces allow a solid and secure junction between the microphone boom on one side and microphone stand side on the other.

Sx-SERIES

Speaker Systems

Designed by the company who pioneered the concept and use of injection-molded structural foam enclosures to make compact speaker systems lighter and stronger, the Sx-Series provides outstanding performance and flexibility in cabinets designed for portable and permanent installation applications. Featuring a wide range of models—from 8-inch two-way to 15-inch two-way systems—to meet your specific requirements, each is equipped with PRO driver protection circuitry and RMD for level-independent fidelity and incredible vocal reproduction. Sx-Series speakers weigh as little as 18 lbs. and have power capacities up to 400 watts. Molded-in attachment points and optional mounting and arraying brackets make them ideal for a wide variety of small-to mid-size portable or fixed applications.

Using exclusive EV manufacturing processes and quality control measures, the Sx-Series far surpass normal industry standards. Designed for musicians, DJs, performers and presenters in clubs, smaller auditoriums, churches and theaters, Sx-Series systems are used at the House of Blues in Las Vegas and have been used at prestigious events such as ShoWest and the Sundance Film Festival.

Ring-Mode Decoupling (RMD), the same technology that gives the EV X-Array concert touring series its incredibly clean and powerful performance has been applied to all SX-Series and components. This gives the SX-series precise speaker intelligibility and true dynamic integrity independent of volume levels by virtually eliminating distortion caused by spurious component vibrations.

Each SX-Series speaker system is created with a precise driver, horn and woofer combination. Pure titanium-diaphragm compression drivers with Time Path phase plug and Resonant Drive technology are coupled to EV-exclusive Constant Directivity HF horns with VariPath throat geometry for silky smooth, extended response, plus detailed and accurate highs.

Sx-Series woofers were developed to complement the performance of the high-frequency section. Sx-Series woofers use a cone material with extremely high internal damping for suppression of unwanted resonances and feature beryllium-copper lead wires, low-mass, edge-wound voice coils and cast-frame aluminum frames for the highest sound quality and rock-solid reliability. And the performance of the Sx-Series speakers is protected with EV's proprietary PRO circuit that prevents the compression drive from being damaged if a dangerous condition occurs.

Sx80



The Sx80 is a compact, high efficiency, constant directivity speaker system for sound reinforcement and monitoring. The small size, high sensitivity and high power-handling capacity of the Sx80 make it an excellent choice for use in studios, clubs, bars theaters or other applications.

- Two-way 8″ system with 175 watts continuous, 700 watts peak power handling
- EV DH2005 pure titanium compression driver for great sound and reliable performance
- 90° x 65° constant-directivity high-frequency horn
- Ring-Mode Decoupling for superior vocal reproduction and smooth, accurate vocals and mids
- Suspension points accommodate Sx80MBB and Sx80MBW wall/ceiling brackets and OmniMount Series 75 mounting hardware
- Optional Sx80SM stand-mount adapter allows use on 100BK and other speaker stands with 1⅜″ diameter

Sx80B (Black)

Sx80B *(#11483160)*: Push-terminal connectors, flying and stand mount (with Sx80SM adapter), paintable black polystyrene

Sx80BE *(#11483254)*: Sx80B with Neutrik Speakon connector

Sx80BP *(#11483253)*: Sx80B with 1/4″ connector

Sx80TB *(#11483162)*: Sx80B with internal 70.7/100-volt, 60-watt transformer (60-, 30-, 15-, 7.5-, 3.8- and 1.9-watt taps)

Sx80W (White)

Sx80W *(#11483161)*: Push-terminal connectors, flying and stand mount (with Sx80SM adapter), paintable white polystyrene

Sx80WE *(#11483256)*: Sx80W with Neutrik Speakon connector

Sx80WP *(#11483255)*: Sx80W with 1/4″ connector

Sx80TW *(#11483163)*: Sx80W with internal 70.7/100-volt, 60-watt transformer (60-, 30-, 15-, 7.5-, 3.8-and 1.9-watt taps)

Sx80P (Weather Resistant)

Sx80PI *(#11483164)*: Weather resistant version of Sx80B

Sx80PIX *(#11483169)*: Same as Sx80PI with internal 70.7/100-volt 60-watt transformer (60-, 30-, 15-, 7.5-, 3.8- and 1.9-watt taps

Sx-SERIES

The term versatility is testament to the inherent design considerations of the Sx-Series. A high performance Sx cabinet using wall mount accessories can cover an audience. The same cabinet using adapter hardware becomes an unbeatable low-profile stage monitor for a rock band. They meet the needs of both musicians and audio engineers by offering adjustable stand mounts and stacking footprints built-in to service various portable setups. Comfortable carrying handles assist in moving Sx systems from job to job. Industry-standard rigging and hanging accessories accommodate any application from concert sound to a local church. To satisfy the demands of professional installations, the Sx full-range systems are available in weather-resistant packages with or without transformers.

Sx100+



The Sx100+ is a compact 12″ two-way system with 200 watts continuous, 800 watts peak power handling. Durable polypropylene enclosure is light and very strong.

- 65° x 65° constant-directivity horn has excellent projection capability and allows the system to be used vertically or horizontally with identical coverage
- Ring-Mode Decoupling delivers smooth, accurate vocals, instruments and mids. Sound quality remains even over a wide dynamic range.
- Rubber feet attached to the bottom of the cabinet and mating sockets molded into the top provide a means of stacking systems
- Built-in stand mount and inserts accept a wide range of accessories to allow safe suspension, arraying and use as a floor monitor

Sx100+ *(#11053282)***:** 1/4″ connectors, flying and stand mount, black

Sx100+E *(#11053314)***:** Sx100+ with Neutrik Speakon connectors

Sx100+W *(11053285)***:** Same as above in white polypropylene

Sx250



Only 40 lbs. the Sx250 is designed for applications that require the warmer low end that its 15″ woofer provides while maintaining small size and light weight. Multi-angle enclosure allows use in different sidefill and floor monitor orientations

- 15″ two-way speaker system with 350w continuous, 1,400w peak power handling.
- 80° x 55° constant-directivity horn has better projection capabilities than 90° horns
- High-performance EV DL15BFH cast-frame woofer and DH2010a pure titanium compression driver
- Ring-Mode Decoupling delivers rich, smooth vocals and accurate, detailed mids
- Dual Neutrik Speakon connectors for loss-free signal transfer
- Built-in metal socket allows mounting on a pole or speaker stand
- Built of 18 mm plywood and finished in black Futura, a sprayed-on polyurethane finish that is very rugged
- Optional SK-1 hardware kit makes flying the Sx250 vertically safe, easy and cost-effective

Sx250 *(#301253001)***:** Flying and stand mount, black Futura

Sx300+



The Sx300 is a 300-watt, 12″ two-way, high-efficiency, constant-directivity compact speaker system with Ring-Mode Decoupling (RMD). The high-frequency section utilizes a molded-in, 65° x 65° constant-directivity horn. This unique pattern contributes to high intelligibility. Vocals sound natural yet "cut through" in reverberant, noisy rooms.

- The world's best lightweight, high-power speaker system with 300w continuous, 1,200w peak power handling. Delivers over 125 dB continuous, 131dB peaks at full power.
- Constructed of high-impact polypropylene structural foam, the cabinet provides a stiff and extremely durable enclosure.
- Integral carrying handle and a stand socket for mounting on 1⅜″ stands such as the EV 100BK. Rubber feet and mating sockets provide a means of stacking systems.
- 65° x 65° constant-directivity horn allows vertical or horizontal use with identical coverage. Delivers better pattern control and projection than typical 90° horns
- Trapezoidal-shaped cabinet, in combination with uniform, 65° x 65° directivity, make the Sx300 ideal for use alone or in arrays.
- Proven, reliable EV components including DL12BFH cast-frame 12″ woofer and DH2010a pure titanium compression driver
- EV PRO Circuit protects the compression driver from damage if a dangerous condition occurs.
- Built-in stand mount and inserts in enclosure allow the system to be safely suspended and arrayed using a wide range of optional hardware. Can also be used as a floor monitor with F200 attachment

Sx300E *(#11053031)***:** 300-watt 12″ two-way, 65° x 65°, Neutrik Speakon connectors, flying and stand mount, black polypropylene

Sx300P *(#11053030)***:** Same as above with 1/4″ connectors

Sx300PI *(#11052991)***:** Weather resistant (including full grille)version of the Sx300E

Sx300PIX *(#11052992)***:** Weather resistant (including full grille) model with internal 70.7/100-volt, 100-watt transformer, black polypropylene

Sx300WE *(#11053029)***:** Sx300E in white polypropylene

Sx300WP *(#11053032)***:** Sx300P in white polypropylene

Sx-SERIES

Sx500+

The Sx500+ is a 400-watt, 15˝ two-way, biampable, high-efficiency constant-directivity speaker system featuring a vented-horn woofer section.



- Two-way 15˝ full-range system with 400w continuous, 1,600w peak power handling
- DL15SX 15-inch woofer and large enclosure volume extend and increase bass output
- High-frequency horn is driven by the DH2T one-inch throat, wide-bandwidth, titanium-diaphragm driver.
- The driver uses a unique, convex-drive TimePath phasing plug structure for smooth and extended high-frequency performance.
- Dual Neutrik Speakon high-current connectors
- PRO circuit for high frequency driver protection
- Ring-Mode Decoupling (RMD) technology for increased intelligibility
- Asymmetric 75° x 60° constant-directivity HF horn aims down 10° – helps direct sound at the audience when vertically stand mounted
- Horn loading of the woofer controls the coverage pattern down to 500 Hz– unheard of in a product of this size– for higher vocal intelligibility and musical clarity. Also provides more bass punch at greater distances than conventional front-loaded woofers
- Attachment points for secure suspension with optional brackets
- Mb600 and horizontal array kits available for side-by-side arraying of two systems (requires one Mb500 wall/ceiling bracket per speaker)

Sx600PI / Sx600PIX



Weather-resistant indoor/outdoor speaker, the Sx600PI is ideal for applications requiring high intelligibility sound at high SPL's such as arenas and stadiums. The Sx600PIX adds an internal 600-watt line transformer for operation on 70, 100, 140 and 200v distributed line systems.

- Dual 12˝ two-element vertical array based on a powerful Neodymium woofer.
- HF section features a DH2T compression driver coupled to a molded-in 65° x 65° constant directivity horn using EV's Varipath throat geometry.
- Exclusive SuperSAM mounting system allows 60° of vertical adjustment and 180° of horizontal pan with simple adjustment.
- Molded polymer enclosure combined with aluminum & neodymium components provide a lightweight, easy to handle system.
- Multilayer stainless steel grille protects against rain, sprayed water, dust, humidity and salt spray to military and environmental grade specifications.

SB121 *300-Watt 12˝ Subwoofer*



The Sb121 is a compact, lightweight subwoofer designed to add bass enhancement to Sx Series speakers. The Sb121 combines a long-throw 12˝ woofer with a lightweight, yet strong polypropylene enclosure. The Sb12 woofer has a specially designed deep frame geometry with a magnetic structure optimized to yield the best possible combination of performance and output capable in this enclosure size. Its Kevlar composite cone and special surround give it nearly 2 inches of total excursion capability. The Sb121 can handle 300w continuous, 1200w peak and can be safely used with amplifiers in that power range, as long as they are not driven to clipping. Using 1/2 or 1/4-space loading as well as mutually coupling multiple Sb121's give this small box very impressive output capability. The Sb121's parallel Neutrik Speakon connectors allow loss-free signal connection. The enclosure can be stacked, stand-mounted and safely suspended using a variety of flexible,inexpensive optional hanging kits.

- 12˝ low-frequency system with 300 watts continuous, 1200 watts peak power handling
- Very durable polymer enclosure can be safely suspended
- Integral handle makes lifting easy
- Dual Neutrik Speakon high current input connectors
- Cast-frame 12˝ woofer with Kevlar reinforced cone
- Adds bass impact and range to systems using Sx80, Sx100+ or Sx300 speaker systems
- Black polypropylene enclosure is light, strong and non-resonant
- Accommodates Mb100 / Mb200 / Mb300 hardware accessories for safe, cost-effective hanging and arraying
- Built in pole mount
- Compact and lightweight, weighs only 32 lbs.

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Sx/Sb System Accessories

100BK: Speaker stand for speakers with integral stand mount or any system up to 110 lbs (1-3/8″ shift diameter), black

F200: Pair of monitor feet for Sx100+/300

Mb100: Forged eyebolt attachment kit for Sb121, Sx100/300, black

Mb200: Wall/ceiling U-bracket kit for Sb121, Sx100+ and Sx300, black

Mb200W: Same as above in white

Mb300: Kit for side-by-side arraying of two Sb121, Sx100+, Sx300 (Mb300 one Mb200 U-bracket kit per speaker system), black

Mb300W: Same as above in white

Mb500: Wall/ceiling U-bracket for Sx500+, black

Mb600: Kit for side-by-side arraying of two Sx500+'s (Mb600 requires one Mb500 U-bracket kit per speaker system), black

Mb700: Forged eyebolt attachment kit (set of three) for a single Sx500, black

PD500: Padded case for Sx500+

PDSx: Padded case for Sb121, Sx100+ and Sx300

SK1: Suspension kit for Sx250

Sx80MBB: Wall/ceiling U-bracket kit for Sx80, black

Sx80MBW: Same as above in white

Sx80SM: Stand-mount adapter for Sx80

VPC500: Vinyl cover for Sx500+

VPCSx: Vinyl cover for Sb121, Sx100+ and Sx300

SB180 *600-Watt 18″ Subwoofer*



The Sb180 is a 600-watt subwoofer which combines the attributes of both horn-loaded and vented-box designs. It utilizes the unique SubScoop cabinet construction with a EVX-180B woofer, as used in the X-Array concert systems. The design behaves like a horn over much of its range, but without the time delay and weight problems usually associated with conventional designs. Vented-box principles take over for the lowest octave and assist in producing exceptionally high levels of clean, punchy bass from a sub-compact box. The reverse woofer orientation has been used to optimize the rear chamber volume and decrease the horn flare rate for extended low-end performance. The Sb180 will augment the bass performance of any EV stage system. A 1⅜″ stand-mount receptacle on the top of the Sb180 allows satellite speaker systems such as the Sx500 and Sx300 to be used. A 34″ mounting pole is also included.

- Concert-proven EVX180B woofer with 4″ voice coil has unsurpassed long-term fatigue resistance for excellent reliability and incredible response to 31 Hz
- Rugged, road-ready cabinet, metal grille. Enclosure is built with 18mm plywood and is finished in Futura, a sprayed-on polyurethane finish that is very rugged.
- SubScoop enclosure combines the extended response of a vented box and the projection and punch of a horn-loaded system
- Natural high-frequency roll-off
- 600-watt continuous, 2,400 watts peak power handling
- Dual Neutrik Speakon inputs
- 1⅜″ stand mount on top allows use of satellite system.
- Includes 34″ long mounting pole for elevating satellite speakers.
- Only 23.75″ tall, includes metal handles for ease in transportation

	Sx80	Sx100+	Sx300	Sx250	Sx500+	Sx600PI
Frequency Range (-10 dB)	51 Hz - 20 kHz	55 Hz - 20 kHz	55 Hz - 20 kHz	50 Hz - 20 kHz	43 Hz - 16 kHz	100 Hz - 18 kHz
Recommended High-Pass Frequency	51 Hz	(12 dB/Oct.) 55 Hz	(12 dB/Oct.) 55 Hz	(12 dB/Oct.) 45 Hz	(12 dB/Oct.) 43 Hz	(12 dB/Oct.) 80 Hz
Axial Sensitivity SPL 1W/1m (Biamp mode)	92 dB	98 dB	100 dB	99 dB	100 dB (100/107 dB)	105 dB
Max. SPL / 1m (calc.); full space	121 dB, (100V: 110 dB)	127 dB	131 dB, (100V: 123 dB)	130 dB	132 dB	138 dB
Continuous Power Handling (low Z)	175 W, (60 W/100V)	200 W	300 W, (200 W/100V)	350 W	400 W	600 W
Coverage (nominal -6 dB) H° x V°	90° x 65° (CD Horn)	65° x 65° (CD Horn)	65° x 65° (CD Horn)	80° x 55° (CD Horn)	75° x 60° (Asym. CD Horn)	65° x 65°
LF Driver	8″	12″	12″ (DL12Sx)	15″ (DLH15BFH)	15″ (DLX155X)	2 x 12″
HF Driver	1″ (DH2005)	1″ (DH2010A)	1″ (DH2010A)	1.25″ (DH2010A)	2″ (DH2T)	1″
Crossover Frequencies (slope in Biamp mode)	2,200 Hz	1,500 Hz	1,500 Hz	1,600 Hz	1,600 Hz (24 dB/Oct.)	1,800 Hz
Nominal Impedance (non-transformer)	8 Ω	8 Ω	8 Ω	8 Ω	8 Ω	4 Ω
Minimum Impedance (non-transformer)	7.2 Ω	5.6 Ω	6.0 Ω	7.5 Ω	5.2 Ω	3.5 Ω
Dimensions (H x W at front x D)	15.75 x 11.5 x 8.75″	23.1 x 17 x 12.3″	23.1 x 17 x 12.3″	24.6 x 17 x 12.3″	33 x 26.5 x 17.63″	45.8 x 17 x 12.3″
Net Weight	18 lbs.	32 lbs.	39 lbs.	40 lbs.	69 lbs.	80 lbs.

SxA-SERIES

Self-Powered Speaker Systems

If you need solid, clean powerful audio in the lightest package possible, direct your ears to the SxA-Series. These self-powered speakers work well in a wide range of sound reinforcement applications, from a simple vocal reinforcement PA to a large, multi-box, high-SPL sound system.

The SxA100+ and SxA250 full-range systems set new performance standards for powered speakers. They provide multiple inputs with full mix capability, a flexible EQ section and plenty of power in compact, easy-to-move enclosures. They make setting up a high-performance sound system quick and easy.

The compact SbA750 and full-size SxA180 powered subwoofers add hard-hitting bass to the SxA or other full-range systems.



SxA100+/SxA250 Powered Speakers

Full range loudspeaker ideal for most applications requiring high-quality sound. Their design provides multiple inputs with mix capability, an easy to use EQ section, and plenty of power in very compact, attractive packages.

- Designed to provide high SPL without excessive heat, the high quality mic preamp provides great sounding vocals or speech. A master level control sets the desired SPL, and the line input can be mixed and balanced with a mic with its level control.
- Linear response is easily optimized for speech or music with the on board multiband equalization.
- Built-in pole mount and optional hardware makes it easy to position them without expensive accessories.
- High frequency section utilizes a molded-in 65° x 65° and 80° x 55° constant-directivity horn coupled to a DH2010A. The result is vocals that sound natural yet "cut through" in reverberant, noisy rooms.

SbA750 15″ Powered Subwoofer

The SbA750 is a high-power compact sub designed to match the SxA100+ and SxA250. The state-of-the-art class H power amplifier module delivers 750 Watts RMS power to an EV DL15Y 15″ woofer. The patented LPN (Low-Pass-Notch) filter compensates for transient distortion of dynamic signals and allows, in combination with the PowerMax 12 crossover filter, the SbA750 to achieve a higher acoustic output than expected from a compact cabinet that size.

SxA180 18″ Powered Subwoofer

- Built-in crossover with Linkwitz-Riley filters continuously variable from 90 -125 Hz
- Built-in storage compartment with ample room for cables and included 34″ pole
- Detachable transport dolly with 4 heavy-duty casters
- Active balanced XLR-type input and outputs for both filtered and unfiltered signals

	SxA100+	SxA250	SxA180	SbA750
Frequency Response (-10 dB)	50 Hz - 20 kHz	55 Hz - 20 kHz	38 - 120 Hz	45 - 150 Hz
Rated Output Power long term	LF150 W/HF 50 W	LF150 W/HF 50 W LF	650 W	750 W
Rated Output Power 10 ms. burst	LF350 W/HF 80 W	LF350 W/HF 80 W	LF 300 W	
Max. Sound Pressure Level	124 dB	126 dB	126 dB	128 dB
HF Coverage (nom.)	65° x 65°	80° x 55°	—	—
LF Components	12″ woofer	15″ woofer	18″ woofer	15″ woofer
HF Components	DH2010A	DH2010A	—	—
Inputs	XLR Mic XLR/1/4″ Line Level XLR out (slave)	XLR Mic XLR/1/4″ Line Level XLR out (slave)	XLR Line Level XLR out	2 x XLR/jack combo 2 x XLR slave through 2 x XLR Mid/Hi out
Level Controls	Mic Level (-35 dBu to 0 dBu) Master Level (- ∞ to 0 dB)	Mic Level (-35 dBu to 0 dBu) Master Level (- ∞ to 0dB)	Master Level (- ∞ to +10 dB)	
2-Band EQ	LF: ± 6 dB; HF: ± 4 dB	LF: ± 6 dB; HF: ± 4 dB	—	—
Dimensions (HWD)	23.1 x 16.9 x 12.3″	24.6 x 17.9 x 13.1″	31.5 x 21 x 23.5″	23.7 x 16.9 x 26.2″
Net Weight	43 lbs.	49 lbs.	88 lbs.	94.7 lbs.

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ONE HOUR FREE PARKING

AT 349 W. 34th STREET (with purchase of $100 or more)

FBTMAXX

Powered Speaker Systems

The new generation of FBTMaxX speakers is made of polypropylene with gas injection molding, producing an ultra-light FBT product that is extremely sturdy. The special design, coupled curves and rounded edges, permits multi-functional application in all audio-professional application systems: such as stage monitors, for wall or ceiling installation and with special adapters to stack the enclosures. Special attention has been paid to the accessibility of controls in all conditions of use. FBTMaxX is the perfect marriage of power, mechanics and leading-edge technology.



FBTMaxX 2A

- Extremely light and sturdy enclosure made of polypropylene with gas injection molding
- 200-watt RMS 10″ low frequency woofer with a Neodymium magnet and die-cast aluminum frame, and a 50-watt RMS 1″ high frequency compression driver with Titanium diaphragm coupled to a custom-designed constant directivity 90° x 60° horn.
- The control panel designed for accessibility under all operating conditions, features both Neutrik XLR and 1/4″ inputs, mic/line switch, and ground lift switch. Volume control, variable gain control, 3-band EQ, and LED clipping indicators are also provided.
- Incorporates ADAP (Advanced Dynamic Active Protection) Processor for total protection of the LF & HF drivers and a 24dB/oct active crossover filter.
- Frequency response is 50Hz to 20kHz and it will handle 250 watts @ 8 ohms.
- The built-in ergonomically designed carry handle allows for a “balanced” feel when transporting.
- Built-in stacking adapters enable cabinet stacking and two M-10 flying points for wall or ceiling mounting, either vertically or horizontally, as well as a standard 1⅜″ speaker socket for speaker stand mounting.

FBTMaxX 4A

Same as FBTMaxX 2A except:

- 300w RMS 12″ LF woofer with a Neodymium magnet, die-cast aluminum frame, and a100w RMS 1″ HF compression driver with Titanium diaphragm coupled to a custom constant directivity 90° x 60° horn.
- Frequency response is 50Hz to 20kHz and it will handle 400 watts @ 8 ohms.
- Four M-10 flying points

FBTMaxX 6A

Same as FBTMaxX 4A except:

- 700w 15″ LF woofer with a Neodymium magnet and die-cast aluminum frame, and a 200w RMS Class G high frequency compression driver with Titanium diaphragm coupled to a custom-designed constant directivity 90° x 60° horn.

FBTMaxX 9SA

- 900 watt RMS 15″ dual coil low-frequency woofer with a Neodymium magnet and die-cast aluminum frame.
- Frequency response is 40Hz to 125Hz and it will handle 400 watts @ 8 ohms.
- Two M-10 flying points for wall or ceiling mounting, either vertically or horizontally, as well as a standard 1-3/8″ speaker socket for speaker stand mounting.
- Control panel features left and right channel inputs with a built-in crossover, both input channels with XLR/Neutrik input, XLR Neutrik link connectors, XLR high pass filtered output and volume control.

ACTIVE VERSION				
	FBTMaxX 2A	**FBTMaxX 4A**	**FBTMaxX 6A**	**FBTMaxX 9Sa**
LF Driver	10″	12″	15″	15″
HF Driver	1″	1"	1.4″	—
Built-in LF amplifier	200W	300W	700W	900W
Built-in HF amplifier	50W	100W	200W	—
Frequency Response	50 Hz - 20 kHz	50 Hz - 20 kHz	40 Hz - 20 kHz	40 Hz - 125 kHz
Sensitivity@1W/1m dB	97	98	98	97
Input Impedance	22kΩ	22kΩ	22kΩ	22kΩ
Max. SPL	121dB	123dB	128dB	126.5dB
Dimensions	13.9 x 21.7 x 12.1″	16 x 25 x 13″	19 x 29.9 x 15.7″	19.5 x 25.5 x 19.1″
Net Weight	28.6 lb.	35.3 lb.	72.8 lb.	64 lb.
PASSIVE VERSION				
	FBTMaxX 2	**FBTMaxX 4**	**FBTMaxX 6**	**FBTMaxX 9S**
Power Handling	250W	300W	700W	900W
Impedance	4 or 8Ω	4 or 8Ω	4 or 8Ω	4Ω
Net Weight	22 lb.	28.6 lb.	64 lb.	57.3 lb.

PASSPORT SYSTEMS

Complete Self-Contained Portable Sound Systems



The innovative Passport Sound Systems are truly portable PA system. Designed with simplicity, flexibility, versatility, reliability, ease-of-use, and efficiency in mind, each Passport system is ideal for working musicians, acoustic groups, club and lounge performers, school and day-care teachers, coaches, auctioneers, religious organizations, aerobics instructors, corporate educational presenters — anyone or any application that requires a portable audio sound system that sets up easily and quickly (in under 5 minutes!). The truly remarkable aspect of a Passport Audio System design is that for all their high-tech technology, you don't have to be a sound engineer to get professional quality sound. Fender has incorporated years of pro audio experience into the Passport Systems' unique design and powerful features. These include proprietary elements like Vocal Input Priority circuitry that automatically lowers all other signals in the system when you speak into the microphone –(particularly useful when you want to make an announcement over music being broadcast), and One Touch EQ, an extremely powerful tone control circuit that takes the guesswork out of getting great sound quickly. Plus, each custom cabinet contains specially designed, full-range speakers positioned at critical angles to maximize projection and clarity without feedback.

Each system offers flexible input and output connections, tone control on each channel, system tone control, effects send and returns, digital reverb, recording outputs and inputs (for CD, tape, video, DVD, or computer connections), speaker and microphone cables, and Fender's P-51 microphones. All systems are compatible with the new Passport Wireless systems for the ultimate in flexibility and portability. The Passport's durable power supply delivers consistent power - in stereo - whether you are using the system with an AC or DC source, and the entire system is packaged into a unique, weather-resistant, portable case utilizing a patented latching system.

They All Feature:

- Mic/line inputs are balanced XLR, plus either balanced or unbalanced 1/4″ TRS. (On the P-150 and PD-150 Deluxe, Input 3 also has stereo RCA connectors buffered to mono). Stereo AUX Return is 1/4″ TRS.
- Digital reverb provided with EFX / AUX Send on each channel. EFX Return to main control provided with remote bypass
- Single switch configures for stereo (R/L) or (Main/Monitor) use. Master level controls provided for each power amplifier. (P-250 and PD-250 Deluxe systems only)
- Pan controls for Mic/Line channels; balanced controls for stereo input channels
- Overall system EQ (for room conditions) and individual channel EQ
- Tape out and reverb footswitch jacks
- Exclusive VIP (Vocal Input Priority) control provides automatic priority of channel 1. Perfect for voice over announcements, and a systems operator isn't required
- Docking connector for passport wireless receiver
- Innovative deluxe speaker array design for superior sound in any application
- Superb feedback rejection allows the speakers to be placed behind the user, eliminating the need for separate monitor speakers
- 1⅜″ stand adapters included
- Switching power supply delivers rated performance even when encountering low power line voltages
- Switchable line voltage for international use
- Systems can run on 12v DC power with optional DC adapter
- Systems include one (P-150/PD-150 Dlx) or two (P-250/PD-250 Dlx) professional cardioid dynamic microphones, 18′ microphone cable(s), two 20′ speaker cables and a detachable power cable. For super convenience, they have a mic/cable storage compartment with latching lid

PASSPORT SYSTEMS

PD-150



- 28 lbs.
- Four input mixer: Three mic/line, One mono/stereo
- Operates on AC or DC power
- 150 watts (75 per channel) stereo, with digital reverb
- One P-51 microphone (with mic clip, cable and carrying pouch)
- Proprietary DSA speaker design featuring three custom speaker drivers per cabinet

PD-250



- 55 lbs.
- Six input mixer: Four mic/line, Two mono/stereo
- Operates on AC or DC power
- 250 watts (125 per channel) stereo, with digital reverb
- Proprietary DSA speaker design featuring four custom speaker drivers per cabinet
- Two P-51 microphones (with mic clips, cables and individual carrying pouches)

ACCESSORIES

Wireless Executive System
The ultimate in convenience! Plug the docking receiver into a 150 or 250, pop a 9v battery into the transmitter, and away you go! Comes with lavalier mics and instrument cable.....**229.95**

Executive Kit
Same as above without the receiver **159.95**

Wireless Hand-Held Mic System
Includes a handheld electret-condenser cardioid mic and a custom docking receiver.....**179.95**

Wireless Mic
Same as above without the receiver **149.95**



P-51 Microphone Kit
A high quality, cardioid pattern dynamic mic. Comes with a cable, stand clip and a custom pouch **49.95**

12v DC-DC Power Converter
Allows Passport systems to run off a DC power source. The converter connects between any 12-volt automotive or marine battery and the Passport system.....**77.95**

Passport Battery Pack
For use with the 12v DC-DC Power Converter. This 12v deep-cycle 17 amp battery powers a Passport system for 6 to 8 hours. Included AC adapter recharges the battery **109.50**

Travel Pack 250
A steel adjustable luggage cart and padded water-resistant canvas cover **89.95**

250 Series Protective Cover
Padded, water-resistant canvas cover.....**42.50**

150 Series Protective Cover
Same as above for the 150 series **26.95**

ST-275 Stand Kit
Two heavy-duty tripod stands constructed of anodized aluminum alloy tubing with a nylon carrying bag.....**119.95**

ST-280 Stand Kit
The same as the ST275, with two additional mic tripod stands **139.95**

Speaker Wall Mount Kit
Custom fitted, universal position wall mounting bracket for a Passport speaker **49.95**

Dual Speaker Mount
Allows you to mount two speakers (P-150 or P-250) on a single speaker stand.....**34.95**

DC Power Accessory Bag
Holds the Passport Battery Pack, DC-DC converter, all cables and microphones.....**59.95**

Passport P-10 and P-10W

A highly portable, personal sound system that is ideal for paging applications, tour guides, classroom and day-care activities, crowd communications and all situations where you need to raise the volume of your voice to communicate effectively with a crowd or small audience. The systems are battery powered and weigh only 6.5 lbs.



P-10W - Comes with a built-in wireless receiver and hand-held wireless cardioid, electret condenser mic, battery, recharger and shoulder strap **299.95**

P-10 - Comes with a dynamic microphone and a 3´ coiled cable, battery recharger and shoulder strap **209.95**

SOUNDFACTOR



The SoundFactor series offers musicians, DJs and other users of live sound reinforcement speakers the performance that goes with JBL speakers at prices they can afford. To achieve this, SoundFactor includes only those features that contribute to sound quality, reliability, and durability—without sacrificing JBL's legendary standards. For example, SoundFactor speakers must pass the same 100-hour, full rated-power torture test that JBL high-end touring systems must survive.

SoundFactor's crossover uses advanced network topology designs that are far more sophisticated than the simple crossovers typically seen in lower-prices systems. Heavy gauge wire is used for inductors to assure that distortion due to coil saturation is minimized and SonicGuard high-frequency driver protection guards against damage caused by inadvertent overpowering of the system. All SoundFactor models are constructed of carpet-covered, 3/4˝ MDF. All-metal handles and an 18 gauge, full metal grille are also used. SoundFactor also uses JBL's Progressive Transition Waveguide design for well-controlled coverage, low-distortion, and good frequency response Tough construction, great performance, gentle on your budget – JBL SoundFactor.

FEATURES

Hardware

All the SoundFactor models use tough, non-resonant all-steel handles with big, beefy bars. And the entire baffle (not just the woofer) is protected by 18 gauge steel grills.

Enclosure Construction

For SoundFactor, JBL selected a rugged and acoustically superior 3/4˝ MDF enclosure material and put it all together with advanced adhesives and closely spaced mechanical fasteners. Not only are the cabinets rugged, but the denser enclosure walls are more rigid, delivering better low end as a result.

Crossover Network Design

While the drivers are the muscle of a speaker system, the crossover network is the heart and soul. SoundFactor uses the same kind of sophisticated advanced network that JBL applies to top-of-the-line systems. SoundFactor networks are built using heavy coils with massive cores and heavy gauge wire so they won't saturate when you drive them hard, and high-voltage capacitors that can handle what you send them. SoundFactor networks use an advanced design that shapes frequency response for coherent summation in the critical crossover region for better mid-range clarity and more consistent coverage at all frequencies.

SonicGuard

Feedback, dropped microphones, pushing the amps into clipping – all these things can quickly destroy even the best high-frequency drivers. That's why JBL SoundFactor speakers include SonicGuard protection. SonicGuard circuitry absorbs excess energy without interrupting the performance. No circuit breakers to reset, no fuses to replace, and no "poly-switches" to degrade sound quality.

Progressive Transition Waveguide

Another example of JBL's innovation is the PT Waveguide design. For decades, horn designers were faced with trade-offs between well-controlled coverage, low-distortion, and good frequency response. JBL developed proprietary algorithms which allow the design of waveguides that give the PT waveguides the best performance in all three areas.

The JBL 100 Hour Torture Test

SoundFactor designs have to survive the same 100 hour torture test as JBL's most expensive systems. When JBL specifies a speaker as having 250 watts continuous power capacity, that means samples of the speaker system have survived a grueling, 100 hour test with IEC noise at 250 watts continuous and 1000 watts peak power. No one in the industry uses a more demanding standard.

SoundFactor speakers have quite a family name to live up to. Just consider its closest relatives - SR-X, MPro, EON G2… legendary JBL performers in the world of professional loudspeakers; products that outperform anything else their respective classes have to offer. Now, that same commitment to performance, quality and technical excellence is also available with a new ingredient: affordability. By keeping everything that makes a speaker perform and sound its best and eliminating things that don't, SoundFactor empowers your performance through speakers designed and built by the worlds leading supplier of professional sound reinforcement products.

B&H PHOTO - VIDEO - PRO AUDIO EQUIPMENT LEASING AVAILABLE

SF12M

12˝ Two-way Stage Monitor Speaker System



The SF12M isn't just a PA speaker in a wedge-shaped box - it was designed from the ground up as a monitor. It is extremely compact, so it won't block audience sightlines or take up too much of the limited real-estate on stage. And it's voiced to let the performers hear the pitch, harmony, and rhythm cues they need for a tight sound. With its high sensitivity (99 dB SPL), it will crank out plenty of sound even with a moderately powered amplifier. For compatibility it's equipped with Neutrik Speakon and 1/4˝ input connectors. Continuous power capacity is 250 watts and Peak power capacity is 1000 watts. A 35 mm pole mount socket lets it do double duty as a main PA speaker.

SF22SP

Dual 12˝ Powered Subwoofer

The SF22SP is a dual 12˝, powered subwoofer using a fourth-order symmetrical band pass enclosure design. It has built-in 500 watt amplifier with 300-watts continuous power or 500-watts peak. For versatility and ease of use, you can plug either a line-level or speaker-level signal into the SF22SP. Its balanced, line-level stereo inputs combine both channels for use in systems with a single sub and two satellites. The speaker-level input is especially useful, as this allows connection of the sub via the loop-through connectors, which then pass on the amplified signal to the passive satellite speakers, such as the SF15.

Accepts stereo or mono line level inputs via balanced XLRs as well as single channel speaker level input via a 1⁄4˝phone jack. Integral signal processing includes a stereo crossover and limiting for the internal amplifier. A pair of XLR outputs may be used to pass the input signals to another system or to send a highpassed, line-level signal to a pair of powered, full-range speakers. Satellite speakers can sit atop the 24˝ pole that is included with the subwoofer.



The SF22SP measures 31.6 x 18.3 x 34.8˝ and weighs 126 pounds. Transport is aided by four removable swivel 3" casters for easy load-in and load-out.

Subwoofers are usually thought of as having large 15˝ or 18˝ woofers, but a pair of 12˝ actually outperform a single 18˝. That's because how low and loud a speaker can play is determined by how much air it can displace, which is determined by the area of the speaker's cone times the maximum distance the cone can travel. A pair of 12˝ actually displaces more air than a single, comparable 18˝ speaker.

Designed for dance music and live reinforcement of drums and bass, which can demand more than a basic two-way speaker can deliver, the SF22SP can work with either powered or non-powered speakers, external amps or a powered mixer. Add the SF22SP, and you've added strong low frequencies to your sound system.

SF15

15˝ Two-way Sound Reinforcement Speaker System



This SF15 is a trapezoidal 15˝ front-of-house speaker that is the choice for live sound, dance music and speech reinforcement. It includes SonicGuard high-frequency protection, a 35 mm pole mount socket, and an impressive peak power capacity of 1000 watts (250 watts continuous power capacity) For compatibility it's equipped with Neutrik Speakon and 1/4˝ input connectors.

SF25

Dual 15˝ Two-way Sound Reinforcement Speaker System



The SoundFactor SF25 is a dual 15˝, two-way speaker system designed for live music performance and playback. However, in reality it is more of a three-way system in that the lower woofer is actually rolled off at a lower frequency than the upper woofer. That makes it right at home in applications where additional low-frequency power and extension is needed but adding a subwoofer to the system is not practical. Continuous power capacity is 500 watts and Peak power capacity is 2000 watts. For compatibility it's equipped with Neutrik Speakon and 1/4˝ input connectors.

Quasi Three-Way Crossover Design

Dual 15˝ systems can crank lots of low-end with their double woofers, but they represent a design challenge. The high-frequency driver and both mid-frequency drivers are all trying to produce mid-range. But since the mid-range is coming from three different places there's lots of uncontrolled "phase cancellation." The result is what engineers call "comb-filtering" and what listeners call "muddy mids". The SF25 crossover design solves this problem by rolling off the mids from the lower woofer so it's working more like a built-in subwoofer. With only the upper woofer and the high-frequency driver covering the mid-range, the SF25 delivers "coherent summation" for clear mids.

MPRO SERIES

The MPro Series speakers are designed for music pros looking for professional performance and appearance in live sound reinforcement speakers. The MPro line has two segments, the 200 Series with carpet-covered enclosures and the 400 Series with higher power handling, DuraFlex™ coated enclosures and a Crown-powered Subwoofer. The MPro 200 Series features premium-grade carpet covered 18mm plywood enclosures. The edges of the carpet are captured by the joint detail to prevent visible seams or peeling. The large-mouth horn design of the MP212, MP215 and MP225 delivers smooth, natural reproduction—especially in the critical mid-range. A 16-gauge, steel grille protects the woofers.



MPro 200 family

The MP400 Series features JBL's DuraFlex coated enclosures and a a Crown-powered Subwoofer. The MP412 and MP415 feature the ARD (Annular Ring Diaphragm) compression driver to deliver extraordinary high-frequency extension.

FEATURES

They All Feature

- 70° x 70° horn designed for even coverage and smooth response on and off axis
- Ferro-fluid cooled compression driver with titanium diaphragm
- SonicGuard™ protects high-frequency driver from excess power without interrupting the performance
- Low frequency drivers with 2-1/2″ (64 mm) edgewound voice coil
- Rectangular (MP212/225)and trapezoidal (MP215) enclosures constructed of 18 mm plywood using tongue-and-groove joint detail for durability and years of service.
- Molded baffle is constructed of SMC (sheet molding compound), a durable, cross-linked thermo-plastic also used in automotive body panels
- Laminar Flow Baffle™ reduces distortion and diffraction effects by providing smooth, rounded baffle contours
- Precision engineered crossover for optimal power response and performance in the critical mid-range region.
- Covered in dark gray, premium grade, 12 oz. carpet - the most durable and attractive carpet covering available
- An attractive 16 gauge, hexagon perforated, steel grill protects speaker components from damage.
- Input panel includes a Neutrik Speakon NL-4 input and a 1/4″phone jack for flexible connectivity.
- 35 mm pole mount receptacle (all except the MP225)

The MPro line offers a unique, professional appearance, superb acoustical performance and cutting-edge configurations. MPro includes the first mid-priced subwoofers to offer VGC (Vented Gap Cooling) technology and the first speaker systems offering JBL's Laminar Flow Baffle technology. The JBL Laminar Flow Baffle incorporates smooth, contoured surfaces that greatly reduce distortion caused by turbulence and diffraction. Constructed of Sheet Molding Compound (a super tough, cross-linked, polymer also used for automotive body panels), this one-piece baffle integrates horn, ports and woofer-mounting into a single part. This permits features to be tightly spaced, allowing a large horn-mouth (for improved mid-range and directivity performance) without making the cabinet too large.

MP212
12″ Two-Way Speaker

Compact, portable, 12″ two-way speaker in a multi-angle enclosure that's equally at home as a FOH (front-of-house) speaker or as a stage monitor. Designed for portable applications in live performance, music playback and speech reinforcement, the MP212 delivers excellent performance by itself or as a satellite over the MP255S subwoofer.

MP215
15″ Two-Way Speaker

The MP215 is a portable, 15″, two-way speaker designed for live performance, music playback, and speech reinforcement.

MP225
Dual 15″ Two-Way Speaker

The MP225 is a portable, dual 15″, two-way speaker system designed for live performance, music playback, and speech reinforcement. The crossover network employs separate low-pass filters for each woofer. The lower woofer is rolled off at a lower frequency than is the upper woofer. This reduces mid-range phase anomalies while retaining the strong low-frequency output that is the primary appeal of a dual woofer system.

B&H PHOTO - VIDEO - PRO AUDIO

MPRO 200 / 400 SERIES

MP255S
Dual 15˝Subwoofer



The MPro MP255S is a dual 15˝ subwoofer designed to supplement and extend the low-frequency output of full-range systems in a variety of portable and fixed applications.

Bandpass design (uses two tuned chambers to shift its energy down into the low-frequency region) combined with an internal Low Pass Filter (to get rid of the remaining mid-range output) assures solid performance in bi-amplified or single amplified systems. While configured for vertical orientation, the speaker may also be used in a horizontal position when a larger footprint and lower vertical height are preferred.

- Dual 15˝ band-pass design puts the energy in the bass region where it's needed for "punch" and power.
- Low-pass filter blocks mid-range output from the subwoofer. Combined with the band-pass design, this assures that the MP255S will combine with the mid / high speaker for excellent low-frequency performance - even without a separate subwoofer amplifier. Even greater performance and control is possible in bi-amplified configurations.
- Dual Neutrik Speakon NL-4 input connectors provided for secure connection and loop-through capability.
- Input connections may be reconfigured for pin 1± or pin 2± for compatibility with any cabling scheme.
- Heavy duty casters mounted on the rear of the enclosure for easy transport.
- 35 mm pole mount receptacle. 24˝ steel pole included

MP410 Series

The MPro 400 Series steps up with enclosures coated with JBL's exclusive DuraFlex - a rugged, advanced, poly-urea finish. They also feature dual Neutrik Speakon NL-4 input connectors. Designed for portable applications in live performance, music playback, and reinforcement of speech, the MP410, MP412 and MP415 deliver excellent performance by themselves or as satellites over the MP418S and MP418SP subwoofers.



MPro 400 family

MP410 10˝ Two-Way Speaker

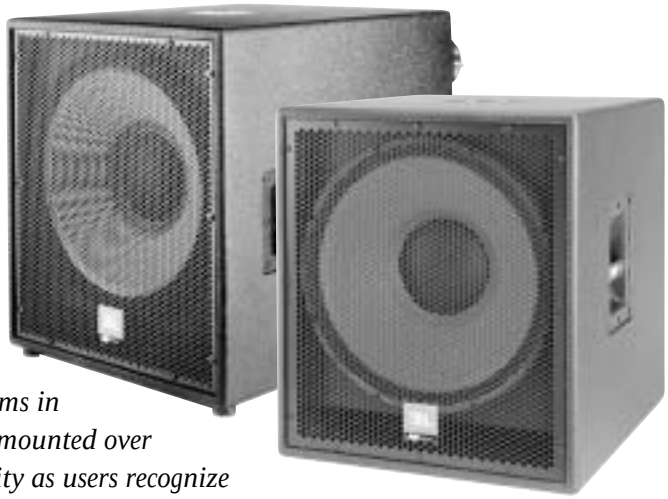
The MP410 is a compact, portable, 10˝, two-way speaker system using JBL components in a professional quality enclosure that looks and sounds like the high-end, specialty speaker systems that top audio pros rely on. It incorporates a 10˝, woofer with 2˝, edge-wound voice-coil. Surround design is optimized for maximum excursion and low distortion using FEA (finite element analysis) modeling. The compression driver with pure titanium diaphragm and ferro-fluid cooling provides improved smoothness and thermal power handling.

MP412/MP415 12˝ and 15˝Two-Way Speakers

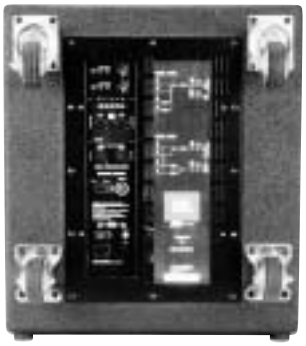
The MP412 and MP415 are portable 12˝ and 15˝ two-way speaker systems in a multi-angle enclosure that supports main PA or stage monitor applications. Their high-frequency performance makes them outstanding choices for high-level music playback, especially when combined with subwoofers.

- JBL 2023 12" and 2033 15˝ woofer with 3˝, edge-wound voice-coil. SHG (Selective Harmonic Geometry) technology reduces harmonic distortion and improves power response
- With a design that combines very high motor strength with low moving mass, their unique, ARD (Annular Ring Diaphragm) compression driver delivers smooth, extended high-frequency output to 20 kHz
- High power handling - 350 watts (average) and 1,400 watts (peak) power capacity

MPRO 400 SERIES



The MPro MP418S is a compact, single 18˝ subwoofer in a bass reflex enclosure. It is designed to supplement and extend the low-frequency output of full-range systems in a variety of portable and fixed applications. Achieving the power and performance characteristics of the MP418S requires the use of a woofer with very high motor strength - the JBL 2241. This woofer uses VGC (Vented Gap Cooling) an exclusive JBL technology that greatly reduces the loss of output resulting from power compression.

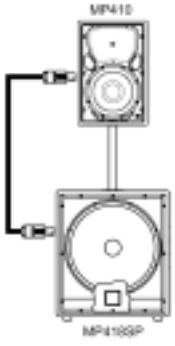


"Subwoofer / satellite" systems (systems in which a smaller, two-way speaker is mounted over a subwoofer) have grown in popularity as users recognize the benefits of elevating the mid and high frequencies while positioning the subwoofer on the floor or stage. To support such applications, the MP418S includes a specially designed, top-mounted, 35 mm pole mount receptacle. The input panel incorporates a pair of Neutrik Speakon NL-4 connectors wired in parallel. The subwoofer is supplied with pins 1± connected to the woofer. This may be easily reconfigured to work with cabling systems intended to drive subwoofers on pins 2±. The MP418S is also designed to be used as a "slave" to the powered MP418SP. When used in this manner, the result is a dual 18˝, powered subwoofer system with 1,320 watts total power.

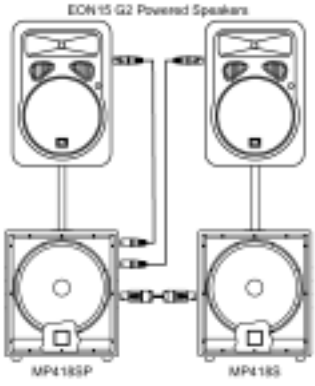
The MP418SP is a compact, powered, single 18˝, bass-reflex subwoofer system. Designed for portable applications, the MP418SP includes heavy-duty, 3˝ casters and ergonomically positioned steel carry handles to facilitate transport.

The MP418SP uses the same driver and enclosure tuning as the non-powered MP418S. It is designed for portable applications and includes a heavy-duty, 3˝ casters and ergonomically positioned steel carrying handles for transport.

MP418SP: OPERATION OF DUAL-CHANNEL CROWN AMPLIFIER



"SUB + SAT MODE



"SUB + SUB MODE

A dual-channel Crown amplifier with signal processing is incorporated into the system. While one amplifier channel (the "Internal" channel) is dedicated to powering the internal 18˝ VGC woofer, the other channel (the "External" channel) may be used in one of two ways:

In the **"Sub + Sat"** mode, an MP418SP may be used with a full-range, passive speaker for a single channel, two piece system. The external amplifier channel powers the satellite.

In the **"Sub + Sub"** mode, the external amplifier channel is used to drive the MP418S (a passive version of the MP418SP). Stereo, high-passed line-outputs are provided to drive powered satellite speakers or an external amplifier / speaker system.

The MP418SP is a compact, powered, single 18˝, bass-reflex subwoofer system with the perfect formula for great sound—just power the best speakers with the best amplifiers. The dual channel internal amplifier is rated at 660 watts @ 4 ohms; 400 watts @ 8 ohms with less than 0.5% THD. One channel is dedicated to the internal 18˝ woofer. The other can drive your choice of an MP418S passive subwoofer or MPro two-way satellites.

EON 15P-1 / EON 1500

MPRO SERIES SPECIFICATIONS					
	MP410	**MP412**	**MP415**	**MP418S**	**MP418SP**
System Time	10″ two-way bass-reflex	10″ two-way bass-reflex	15″ two-way bass-reflex	18″ two-way bass-reflex	Powered 10″ two-way bass-reflex
Frequency Range	50Hz - 20kHz (-10dB)	50Hz - 20kHz (-10dB)	44Hz - 20 kHz (-10dB)	36Hz - 300kHz (-10dB)	36Hz - 150kHz (-10dB)
Frequency Response	62Hz - 14kHz (±3dB)	67Hz - 20kHz (±3dB)	57Hz - 20kHz (±3dB)	40Hz - 300kHz (±3dB)	40Hz - 120kHz (±3dB)
Sensitivity (1w, 1m)	94 dB SPL	99 dB SPL	99 dB SPL	101 dB SPL	
Nominal Impedance	8Ω	8Ω	8Ω	4Ω	4Ω
Maximum SPL @1m	125 dB	130 dB	130 dB	135 dB peak	132 dB SPL (peak@1m)
Power Capacity	300w	350w	350w	600w	
Peak Power Capacity	1200w	1400w	1400w	2400w	
Nominal Dispersion	70° x 70°	70° x 70°	70° x 70°		
Transducers: LF	1x JBL 127H-4	1x JBL 2023H	1x JBL 2033H	1x JBL 2241G	1x JBL 2241G
HF	1x JBL 2412H	1x JBL 2406H	1x JBL 2406H		
Enclosure	18mm plywood	18mm plywood	18mm plywood	18mm plywood	18mm plywood
Input Connectors	2x Neutrik Speakon NL-4, parallel	2x Neutrik Speakon NL-4, parallel	2x Neutrik Speakon NL-4, parallel	2x Neutrik Speakon NL-4	2x Neutrik Speakon NL-4, parallel
Net Weight	34 lb.	47 lb.	59 lb.	64 lbs.	119 lbs.
Dimensions (HxWxD)	20 x 13.3 x 11.8″	23.9 x 15.6 x 13.6″	28.3 x 18.2 x 15.1″	24.3 x 21.2 x 23.6″	24.3 x 21.2 x 30.6″

THE ORIGINAL EON SERIES

The EON system is a made-to-match system. All components are designed to give you hassle-free professional-sound quality and performance. EON was the first professional sound system made to be truly portable. The secret behind EON's feather-weight is a rare earth material called neodymium with 10 times the magnetic strength of ceramic magnets, enabling a few ounces of neodymium to replace nearly 20 lbs. of conventional magnetic materials.
For flexibility, the EON speakers are built to work as both upright speakers and wedge monitors. With ergonomic handles and light weight, they are remarkably easy to carry. Simple to set up. And a pleasure to use.



EON 15P-1

The EON 15P-1 is a two-way powered speaker system which incorporates a discrete 130 watt power amplifier for low frequencies and a 50 watt power amplifier for the high-frequency driver in a light weight rugged enclosure. Designed for multi-purpose usage, the EON-15P-1 may be driven from a mixer or a single microphone.

EON 1500

The EON 1500 features the same light weight, portability and great sound of other EON speakers, but is designed for use with a powered mixer or external amplifier. EON 1500 features proven JBL components like a liquid-cooled compression driver and SonicGuard protection. It's unique design allows you to use them as a main speaker, tripod mounted or a dual-angle floor monitor. The EON 1500 accepts a 1/4″ phone jack or Speak-On input connection wired in parallel for ease of hookups.

EON G2



Hassle-free with professional sound quality and performance JBL's EON G2 series is designed for applications ranging from live sound reinforcement, speech and vocals to music playback in entertainment, A/V and institutional environments. Consisting of the compact EON10 G2 (10˝ woofer) and full-range EON15 G2 (15˝ woofer) powered speakers with three input mini-mixer, and the EONSUB G2 powered subwoofer, they are not only ideal for musicians, but for hotels, meeting planners, AV companies and just about any application where portable compact sound reinforcement is required.

They deliver 125 and 300 watts of power (respectively) to a Neodymium Differential Drive low frequency driver. The driver boasts a dual neodymium magnet and dual voice coil motor that delivers high output and low distortion from a lightweight woofer. Meanwhile, the high frequency amplifier delivers 50- and 100 watts of power to a titanium diaphragm compression driver with ferro-fluid cooling. JBL's revolutionary "ThermoMaster Total Thermal Management System" eliminates the need for any internal or external cooling fans. Instead, a single piece, cast-aluminum baffle integrates the woofer frame, horn and amplifier heat-sink while air movement through the finned ports dissipates heat. With their integrated three-input mixer, the speakers can directly accommodate electronic keyboards, drum machines, electrified acoustic guitars, and microphones. An additional XLR output allows the speakers to be daisy-chained together or send the mixed output to an external mixing console.

With 125 watts for low frequencies and 50 watts for the highs, the EON10 G2 can be used in AV applications, smaller musical and spoken word performances. Versatile cabinet design allows use in several ways: As a front of house speaker, sitting on a flat horizontal surface or raised in the air using standard tripod speaker stands. Angled back allows use as a vocal or instrument monitor on stage. The EON10 G2 can also be snugly placed atop the EONSUB G2 powered subwoofer for added volume and extended bandwidth. Combined, they provide a full range sound system for small and medium sized events.

With 400 watts of power, excellent sound quality, easy to transport and set up, the EON15 G2 is at home as a main PA or on stage as a monitor or instrument amplification system. Use by itself for casual solo or duo work, or combine multiple EONs into a complete, powerful PA system. Electric guitarists using amp modeling modules will find that the clean, full-range sound of EON15 G2 reproduces all of the subtle tones the modules can produce.

EON10 G2

- Multi-angle enclosure for FOH or stage monitor applications
- 60° x 90° constant directivity horn
- 10˝ Differential Drive low frequency driver with neodymium magnet for light weight and reduced distortion
- 1˝ JBL compression driver with titanium diaphragm and ferro-fluid cooling
- 125 watt low-frequency and 50 watt high-frequency (bi-amplified) amplifiers – actual power delivered to the drivers
- Selectable balanced mic/line input and one dual-channel, 1/4˝ TRS line-level input
- Balanced loop/mix output. "Daisy-chain" additional EON speakers or send the mixed output to a mixing console
- Rugged, lightweight, black co-polymer enclosure
- Integral 35 mm pole mount receptacle with securing thumb screw
- Attachment points for fixed-angle and adjustable mounting brackets

EON15 G2

Step-up Features

- 15˝ Differential Drive low-frequency driver with dual neodymium magnet for light weight and reduced distortion
- 1˝ (throat diameter) JBL compression driver with 1.75˝ titanium diaphragm and ferro-fluid cooling
- 300 watts low-frequency and 100 watts high-frequency (bi-amplified) - actual power delivered to the drivers
- One balanced mic/line input, two 1/4˝ phone line-level inputs
- Two-band equalization
- Extended low-frequency performance — delivers powerful bass down to 39 Hz
- Tailored dynamic processing protects amplifiers and speakers from being over-driven without producing extreme changes in frequency response as level varies

EONSUB G2

The EONSUB G2 offers powerful low frequencies in a compact, durable enclosure, making it a perfect companion for either the EON10 G2 or EON15 G2 powered speaker, or any sound system that needs additional low end.

When you combine the EONSUB G2 with either the EON10 G2 or EON15 G2, you have a full range sound system whose clarity, volume and low end are truly amazing, considering their small size. Because the power amp and crossover are internal to all EON speakers, set up time and outboard equipment are minimized. Just plug in and play.

- 250 watts of low-end power and a frequency range of 40 to 200 Hz
- Patented Neodymium Differential Drive woofer generates maximum efficiency with minimum weight
- Thermomaster cooling system allows the amplifier and 15˝ woofer to stay cool, even under the most challenging circumstances, without the need for noisy fans or protruding external heatsinks
- Specifically designed to securely hold the EON10 G2 using receptacles built into the top of the EONSUB G2
- Scratch-forgiving black covering and embossed grill cover for added rigidity

EON G2 Accessories:

ESK15: Suspension kit for EON15 .....**29.95**

ESK10: Suspension kit for EON10 .....**22.95**

EONBRK1: Mounting bracket (fixed angle) for EON 15˝ models.....**39.95**

EONBRK2: Mounting bracket (fixed angle) for EON 10˝ models.....**33.95**

SS2-BK: Black anodized aluminum tripod speaker stand **134.50**

EON10 Bag: Zippered, plush-lined Cordura bag for EON 10 G2 .....**39.95**

EON15 Bag: Zippered, plush-lined Cordura bag for EON 15 G2 .....**49.50**

EON15WB: Zippered, plush-lined Cordura bag with wheels for EON 15 G2.....**99.95**

EON G2 Systems

JBL offers the EON10 and EON15 G2 in complete full-function sound systems that include two speakers, a mixer, two JBL dynamic M50S microphones and four 25´ XLR cables for connecting the mixer to the microphones and speakers.



Ideal for smaller musical performances, speeches, coffeehouses, schools, churches or any application that requires quality sound in a smaller venue, the 14 input, 350 watt EON10 G2 sound system includes the MusicMix 14 mixer and a pair of EON10 G2 speakers. When combined with the EONSUB G2 subwoofer, the power of the system is more than doubled. Featuring twice the power and greatly enhanced sound, the 16 input, 800 watt EON15G2 System sound system includes the MusicMix 16 mixer and two EON15 G2 speakers.

EON10 G2 System	*EON15 G2 System*
2x EON10 G2	2x EON15 G2
EON MusicMix 14	EON MusicMix 16
2x M50S Microphones	2x M50S Microphones
4x 25´ XLR Cables	4x 25´ XLR Cables



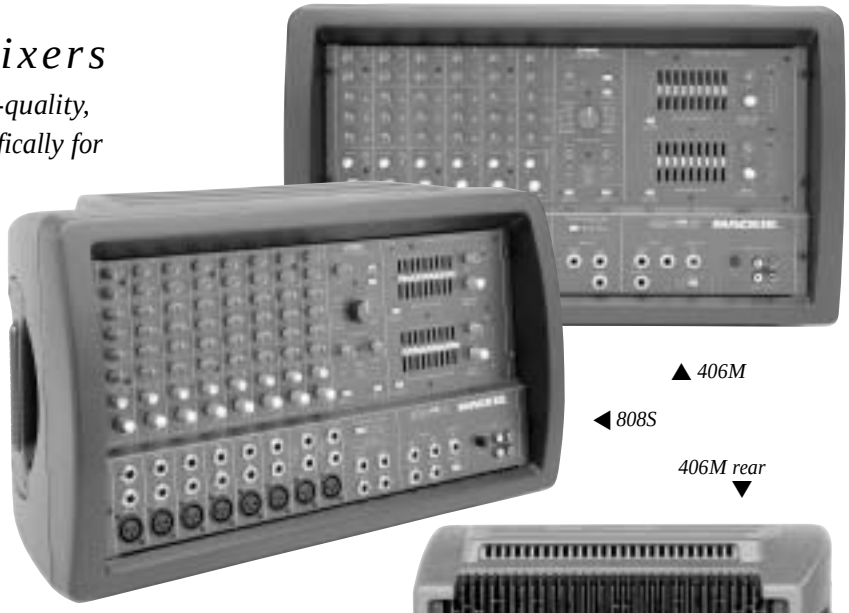
The EON MusicMix 14 and 16 are feature-packed mixers with six or eight mono inputs, two stereo inputs, two auxiliary inputs, and three band equalizer. Their third auxiliary channel can be selected as either pre- or post-fader, for greater control and flexibility with monitors and effects devices.

- Six or eight mic/line input channels
- Two stereo input channels
- Two auxiliary input channels
- 3-Band channel EQ with variable mid-frequency for precise tailoring of the sound
- Individual channel on/off switches let you mute an input without disturbing your mix
- Low-cut filter to reduce unwanted noise
- 48v Phantom Power
- Three Auxiliary Mix Busses: 1 Monitor, 1 Effects and 1 Selectable
- Complete 'listen' (PFL/AFL) system for setup and trouble shooting
- Smooth 100mm faders faders

PPM SERIES

Professional Powered Mixers

Mackie's PPM Series consists of three professional-quality, high-performance powered mixers designed specifically for applications such as local and regional club bands, rehearsal rooms and general gigs, small-to-medium club PA systems, small-to-medium houses of worship, and corporate A/V. From the 6-input, 500-watt mono 406M to the the-top-of-line 8-input, 1200-watt stereo 808S, you can count on ultra-low noise, high headroom, graphic EQ, sixteen 32-bit digital effects, rugged composite -molded cases, and a real RF Series high-current amplifier inside.



▲ *406M*

◀ *808S*

406M rear ▼



FEATURES

- In addition to 16 ultra-realistic EMAC 32-bit processing effects, there are two parameter controls that expand the range of each effect for almost unlimited sound possibilities. There is also a special EFX WIDE circuit that can psychoacoustically enhance the width and depth of reverb, chorus, and flange— perfect for set-ups where your speakers have to be placed close together.
- Studio-grade 9-band mains and monitor graphic EQs utilize multiple summing amps and precision infinite feedback band-pass filters to yield exceptionally low-noise low-distortion sound.
- 75Hz low-cut filters in both EQ sections remove microphone thumps, P-pops and stage rumble. Sharp 18dB/octave circuitry works without cutting audible bass.



808M

- Mic/line mixer section based on the MicroSeries 1202-VLZ and enhanced with new mic preamps and many ingenious extra features.
- Level-setting control and LED on every mixer channel makes setting input levels a snap. This is the key to optimizing headroom and reducing mixer noise.
- Break switch mutes Channels 1-6 during set breaks so you can play music (via the tape input and stereo channels 7 or 8) without having to mess up your levels, deal with feedback loops, or worrying about somebody grabbing the microphone while you are on a break.
- Built-in soft-knee compressor works wonders on vocals and percussion, letting you get more out of your speakers.
- EFX to Monitor lets you add effects to on-stage monitor mixes independently of the main mix.
- Dual line inputs on channels 7 and 8 (mono models), and stereo line inputs (stereo models) lets you hook up keyboards or CD players without burning up two channels. (Not on the 406M).
- Dozens of conveniences including separate input level control, external effects loop, effects foot switch jack, and a gently tilted front panel that makes it easy to make adjustments.

Power Amp Section

- Power amp routing switch lets you assign power amps to mains or mains + monitors.
- Massive 10,000 microfarad storage capacitors to satisfy current-sucking peak power demands.
- Massive toroid transformer for high current output
- Massive, full-width 1-piece die-cast heat sink is custom-cast for maximum heat dissipation—keeps output devices running cool even when you're playing hot. Black paint coating improves efficiency by additional 15%.

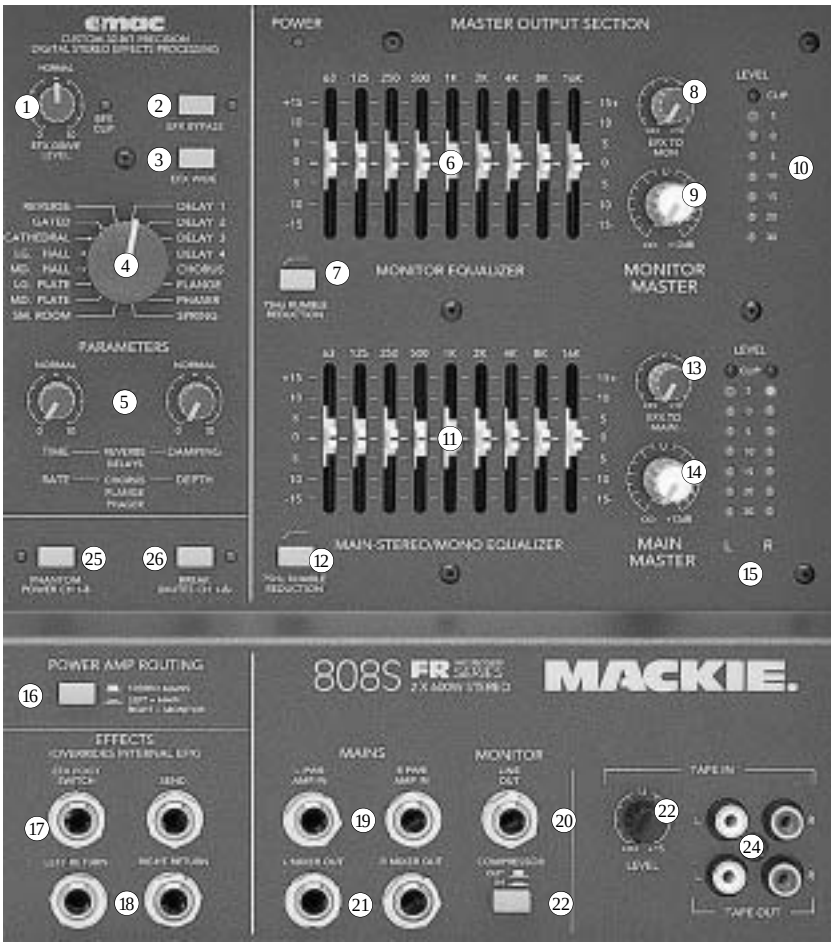
Rugged Construction

- Rugged injection-molded chassis resists impacts that would destroy most other powered mixers. Built-in handle allows easy transport.
- For durability and reliability, they use mil-spec fiberglass circuit boards (instead of cheap phenolic) to resist physical impact. Rugged, sealed rotary controls resist dust, moisture and other contamination.

PPM SERIES

MASTER SECTION

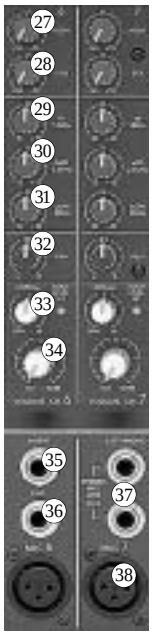
1	Effects Drive level control & clip OL LED
2	Effects Bypass with LED
3	Effects Wide spatial expander
4	EMAC effects preset select-o-matic
5	Time/Rate & Damping/Depth parameter controls
6	Monitor graphic EQ with centers ±15dB at 63Hz, 125Hz, 250Hz, 500Hz, 1kHz, 2kHz, 4kHz, 8kHz, & 16kHz
7	Monitor 75Hz, 18dB/oct low cut filter
8	Effects to Monitor level control
9	Master Monitor output level control
10	Monitor output level LED ladder
11	Main graphic EQ (same centers as #6 above. EQ is stereo on 808S and 408S)
12	Main 75Hz 18dB/oct low cut filter
13	Effects to Main level control
14	Main output level control
15	Main output LED ladder
16	Power amp routing switch
17	External effects foot switch jack
18	External effects loop - mono send (stereo return on stereo models)
19	Power amp line in jacks
20	Monitor line output
21	Line level mixer output(s) (bal/unbal)
22	Soft-knee compressor switch
23	Tape input level control
24	RCA Tape inputs and outputs (unbal)
25	Phantom Power switch with LED
26	Break Switch with LED (chs. 1-6 only)



PPM SERIES SPECIFICATIONS

	808S	808M	406M
Total Channels	8	8	6
Total Inputs	18	18	12
Configuration	stereo	mono	mono
Mic Preamps	8	8	6
Mono line inputs	6	6	6
Dual Line Inputs	—	2	—
Stereo Line Inputs*	2	—	—
2-ohm Power	600+600w	600+600w	250+250w
4-ohm Power	450+450w	450+450w	200+200w
6-ohm Power	300+300w	300+300w	125+125w
Channel EQ	3-band	3-band	3-band
Main Graphic EQ (main)	true stereo 9-band +75Hz low cut	mono 9-band +75Hz low cut	mono 9-band +75Hz low cut
Monitor Graphic EQ (monitor)	mono 9-band +75Hz low cut	mono 9-band +75Hz low cut	mono 9-band +75Hz low cut
Channel Inserts	6	6	6

** same jack set as above, used fr Hi-Z mic inputs*



CHANNEL STRIP

27	Monitor send for stage monitor feeds
28	EFX send to internal EMAC effects
29	12kHz HI shelving EQ. ±15dB
30	2.5kHz MID peaking EQ. ±12dB
31	80Hz LO shelving EQ. ±15dB
32	Pan control (stereo models only)
33	Input Level Control & Level Set LED
34	Master channel level control
35	Channel insert (mono channels only)
36	Line input also handles Hi-Z mics (bal/unbal)
37	Dual (mono unit) or stereo (stereo unit) line inputs on chs. 7 & 8 (not available on 406M)
38	The best mic preamps on any powered mixer

MACKIE

C300/300i



Passive 12″ 2-Way Speakers

The C300 (Speakon connectors) and the C300i (1/4″ connectors) uses the same basic transducers as the active SRM450 — with two differences. The C300's 1.75″ high frequency compression driver uses a composite mylar diaphragm; the 12″ low frequency transducer incorporates a 3″ inside/outside temperature- resistant voice coil. Both of these compensate for the fact that the speaker system must be able to handle a wide range of external power inputs (including Mackie's mega-powered PPM808S or PPM808M).

The C300/C300i uses the same studio-quality horn assembly as the SRM450, with multi-cell aperture horn throat and large format exponential/conical high-frequency horn mouth for superb dispersion. This means everyone in your audience hears clearly... not just the people in the middle between the speakers.

They have an asymmetrical trapezoidal cabinet for minimum internal resonance and the ability to recline for floor monitor use. Their enclosure is weight-balanced with dual side handles and top handle, built-in pole mount and ten built-in hanging points. Passive speakers (no internal amplification) the C300/C300i have a Low-Impedance Compensated Crossover (LICC) with full built-in transducer protection that just beg to be energized to shock volumes with a powered mixer or amp.

- Wide-dispersion, studio-quality horn assembly with multi-cell aperture horn throat & exponential / conical HF mouth
- 1.75″ high frequency compression driver
- Composite mylar diaphragm increases clarity, eliminates resonances
- 12″ long-throw low frequency transducer
- Reclines for use as a stage monitor
- Weight-balanced asymmetrical trapezoidal cabinet - strongest & stiffest in it's class
- 10 built-in fly points & built-in pole mount
- Two comfortable side handles for easy carry & set-up + top handle for easy positioning
- 300 watts RMS power handling
- 8-Ohm nominal impedance
- Choice of Speakon or 1/4″ jack connectors

S500 *Passive 15″ 2-Way Speakers*



Although not the cheapest 15″ 2-way system on the market, the S500 is one of the most accurate and easy on the ears at high output levels. Make beautiful music with a Mackie powered mixer—especially when used with a high-wattage model, it really starts shaking your audience up. Tight bass, smooth midrange, spine-tingling treble... at shock volumes up to 127dB.

Low-distortion, high-output horn coupled to a 1.75″ titanium diaphragm compression driver provides the correct power response and phase alignment to deliver a wide 75° x 65° dispersion pattern that lets everybody in your audience hear the same great sound. The S500 employs a trapezoidal enclosure that's both strong and resonance damping. It is built of rugged 16-ply Baltic Birch with tough end caps made of pressure-injected structural resin. The perforated steel grille is coated with a special scuff and weather-resistant polymer coating. Cast aluminum side handles are placed at the speaker's center of gravity for easy carrying and set-up. Rear panel inputs include both 1/4″ and Neutric Speakon connectors wired in parallel.

- 1.75″ high frequency compression driver
- Composite mylar diaphragm increases clarity, eliminates resonances

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SRM450

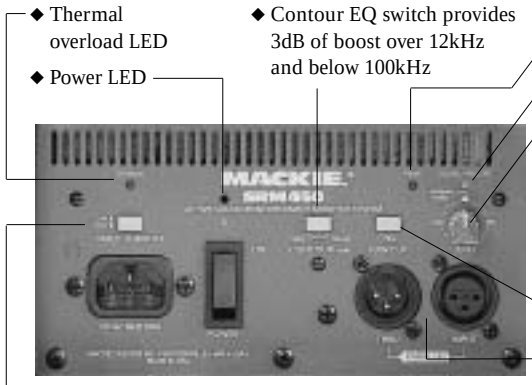
Active Sound Reinforcement Monitor



Forget about what other small PA speakers sound like. Especially forget what so-called "powered" speakers sound like. The active SRM450 is that different. The most sophisticated, versatile portable sound reinforcement speaker ever developed, the SRM450 translates the fidelity and performance found in Mackie's award-winning HR824 Active Near Field Studio Monitor into a tough, portable and easy to use composite enclosure. The SRM450 delivers tight, steel-fisted bass to below 55Hz and detailed, articulate treble out to 20kHz. Midrange is warm, natural, and free from ear-fatiguing harshness. Active technology with wide dispersion, studio quality horns lets the whole audience hear clearly, whether they're in the front, back, or way out on the sides of the room. And above all, the SRM450 cranks! Without a hint of distortion, it effortlessly achieves sound pressure levels that would melt conventional speakers into a pile of molten slag.

FEATURES

- Combination mic/line level control incorporating Mackie studio-quality low noise mic pre-amp. Built directly into the SRM450, this allows it to also function as a complete stand-alone PA system.
- 2-way bi-amplified active system —built-in monolithic 300 watt bass & 100 watt mid/high amplifier—no external power amplifier needed
- Servo-controlled RCF Precision 12″ (300mm) long-throw frequency transducer with inside/outside voice coil
- Studio-quality maximum dispersion horn design
- High (127dB) SPL output/low distortion
- 1.75″ high-output precision compression driver with damped titanium diaphragm assembly
- Level control, contour & low cut switches
- Built-in electronic equalization, phase alignment, time compensation, precision 24dB/oct. Linkwitz-Riley electronic crossover and protection circuitry.
- Power On, Thermal, Peak and Signal Present LEDs
- Molded composite, monocoque asymmetrical enclosure and metal grille
- Three balanced ergonomic handles make it easy to carry
- Timed Auto-on/Auto-off feature
- Pole-mountable and multiple fly points make it easy to set up—also a great floor monitor



Big studio monitors use an exponential horn design that keeps the air impedance constant and the sound smooth as silk. However, off-axis pattern control isn't great. Lots of speakers claim 90° horizontal dispersion. But when you examine their polar patterns, you discover that they don't achieve 90° dispersion at all frequencies. When a speaker has uneven dispersion (poor pattern control) at higher frequencies, acoustic energy projects straight out in a beam—great for the listeners right in front of the speaker, but muddy, dull, uneven sound off-axis.
The SRM450's horn design is a unique three-piece assembly derived from high-end studio monitor technology where wide dispersion and precise reproduction of voice and high frequencies is critical. Mackie applied this technology in a high powered, sound reinforcement application coupled with high output compression drivers. The result is an extremely smooth off axis response, which allows everyone in front of the SRM450 to experience the same high-resolution audio no matter where they are seated.

MACKIE

SA1232/SA1521/SR1530

Active 2- and 3-Way Sound Reinforcement Speakers





Mackie's active sound reinforcement speakers are designed for extreme accuracy, high output, and smooth dispersion across the entire audio frequency range. They integrate FR series amplifier power, active control electronics, and RCF Precision components to provide unparalled high resolution and clarity in the critical low-mid and mid-range frequencies Offering exceptional price/performance ratio, they are used in garage bands, one-man/duos/trios, local club bands, practice room rehearsal halls and school music rooms. They are equally suited for small, fixed or traveling church revivals, small music and dance clubs, assembly halls, conferences and outdoor meetings . The active 15˝ two-way SA1521 and 3-way SR1530 are ideal for mobile DJ and small house mix, clubs, bars and restaurants, while the active dual 12˝ 3-way SA1232 is designed to provide high definition and extreme output level sound reinforcement to club or regional touring bands, club PA systems, or medium to large size churches.

- They have a signal input panel containing XLR connectors for line-level input and loop-through connections, a level control, and four status LEDs for Power On, Signal Present, Limit, and Thermal functions. (The SR1530 doesn't have level control or Signal and Thermal LEDs)
- Their internal processors provide complete system management of all electronic and acoustic functions. These include an electronic active crossover, electronic phase alignment, electronic time correction, electronic equalization, and complete amplifier and component protection.
- To ensure long term reliability, their amps are mounted on a massive extruded heatsink, which is convection cooled and dramatically extends the life expectancy of the heat-producing components.
- Their trapezoidal cabinet is designed to withstand Neanderthal-levels of road and rental company abuse. The top and bottom of the enclosure are molded using a high-impact resistant composite resin. The front, back and sides are constructed of heavy duty, 13-ply Baltic Birch used in expensive Hi-Fi speaker systems.
- Weight-balanced for stable stand-mounting, they also feature two large-grip, cast-aluminum side handles while top and bottom handles allow optimal portability, set-up and repositioning. Full-size, oval punched, weather-resistant steel grille protects the front of the cabinet.

SA1521 *(Active 15˝ 2-Way)*

A 'high output big brother' to the SRM450, the 500 watt SA1521 provides the same accuracy but with even more bottom end and a higher SPL. A two-way, bi-amplified, horn loaded, active speaker system, the SA1521 includes a high efficiency, light mass, 15˝ low frequency transducer with 2.5-inch inside/outside high temperature coil technology. This innovative woofer is combined with a state of the art, 1.75˝ titanium dome, 1˝ exit compression driver. This unique combination allows the system to be crossed over at 1300 Hz enabling the woofer to more efficiently keep up with the horn/driver combination. The result is that the SA1521 offers much better mid-band performance than other popular 15 +1 designs.
The fully integrated amplifier module contains two separate amps - one for each driver, which together generate 500 watts continuous RMS of total power and a maximum SPL of 130 dB. The low frequency amplifier uses Mackie's switch mode and Class G hybrid topology to deliver 400 watts of continuous output, and over 650 watts peak at 90% efficiency. The result is very little power being dissipated thermally through the heat sink.

SA1530 *(Active 15˝ 3-Way)*

The SR1530 is a three-way, tri-amplified, horn loaded, active system designed for high definition and extreme output levels. Precision components include a 15˝ woofer with inside/outside high temperature coil technology, a high-output, horn loaded 6˝ midrange speaker, and a one-inch exit compression driver mounted on a wide dispersion, extreme output high/mid horn design. The SR1530 is powered by an integrated three-channel amp module that generates 500 watts of total power.

SA1232 *(Active Dual 12˝ 3-Way)*

The SA1232 is also a three-way, tri-amplified, horn loaded system that includes two 12˝ woofers with 3" inside/outside high temperature coil technology, a high-output horn loaded 6˝ midrange speaker and a 1˝ exit compression driver mounted on a wide dispersion, 90° x 40° mid/high frequency Optimized Wavefront horn design. This unique horn design combines the mid and high frequencies to eliminate unwanted phase shifting.The SA1232 is powered by an integrated amp module that contains three separate amplifiers—one for each driver, which together generate a staggering 1300 watts RMS of total power and a maximum peak SPL of 133 dB.

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ONE HOUR FREE PARKING
AT 349 W. 34th STREET (with purchase of $100 or more)

SWA-1501/SWA-1801

15- and 18˝ High Output, Active Subwoofer Systems



Ideally suited for small to medium sized touring groups and mobile DJs as well as fixed installations in churches, corporate facilities, and nightclubs, the SWA1501 and SWA1801 feature high-quality 15- and 18˝ RCF Precision transducers with high-temperature inside/outside coil combined with a 500-watt continuous/700 watt peak (SWA-1501) or 900-watt continuous/ 1400-watt peak (SWA-1801) application-specific amplifier. Their cabinets are constructed with 18mm thick, 13-ply Baltic-Birch plywood finished in back splatter paint and equipped with cast aluminum side handles for efficient loading and transport as well as a female pole mount fixture. Together, these robust components allow them to deliver an impressive 128dB (SWA1501) or 135dB (SWA-1801) of peak output. Input output options allow them to be easily configured for most any application. Stereo and mono inputs are provided through XLR style connectors while a high-pass output allows for connection to active full-range speakers or amplifiers powering passive full range speakers. Four function LEDs provide power on, limit, signal present, and thermal condition indication. A rotary level control allows adjustment from -15dB to +5dB.

MACKIE SPEAKER SPECIFICATIONS

	C300	S500	SRM450	SA1521	SR1530	SA1232
Frequency Response (-10dB)	—	—	45Hz - 20kHz	49Hz - 20kHz	40Hz - 20kHz	38Hz - 20kHz
Frequency Response (±3dB)	70Hz - 16kHz	50Hz - 20kHz	55Hz - 18kHz	57Hz - 18kHz	45Hz - 18kHz	45Hz - 18kHz
Peak SPL (@1w/1m)	124dB	127dB	127dB	133dB	126dB	136dB
HF Transducer Diameter	1.75˝	1.75˝	1.75˝	1.75˝	1˝	1.75˝
HF Transducer Freq. Response	1kHz-20kHz	1kHz-20kHz	1kHz-20kHz	1kHz-20kHz	2.5kHz-19kHz	1kHz-20kHz
MF Transducer Diameter	—	—	—	—	6˝	6˝
MF Transducer Freq. Response	—	—	—	—	700Hz-3kHz	700Hz-3kHz
LF Transducer Diameter	12˝	15˝	12˝	15˝	15˝	2x 12˝ (18˝ equiv.)
Height	26˝	31˝	26˝	31˝	43.9˝	47.24˝
Front Width	15.4˝	19.1˝	15.4˝	19.1˝	19.1˝	19.1˝
Rear Width	—	13.9˝	—	13.9˝	13.9˝	13.9˝
Depth	14.8˝	18.1˝	14.8˝	18.1˝	18.1˝	18.1˝
Weight	46 lbs.	75 lbs.	51 lbs.	83.6lbs.	111.25 lbs.	132 lbs.
Enclosure	Composite molded	Baltic birch ply	Composite molded	Baltic birch ply	Baltic birch ply	Baltic birch ply
Enclosure Geometry	trapezoidal	trapezoidal	asymmetrical	trapezoidal	trapezoidal	trapezoidal
Mounting Methods	Pole & Integral M10 mounting points	Floor applications only	Pole & Integral M10 mounting points	Pole mount	Floor applications only	Floor applications only
Male XLR Signal pass-through	—	—	yes	yes	yes	yes
Level Control	—	—	yes	yes	no	yes
Timed Auto Turn-Off	—	—	yes	no	no	no
Power On, Signal Present, Limit and Thermal LEDs	—	—	yes, yes, yes, yes!	yes, yes, yes, yes!	power on/ OL	yes, yes, yes, yes!
Low Cut Filter	—	—	100Hz/2nd order	—	—	—
Mic Preamp input	—	—	yes	—	—	—

S62/S63

6-Channel Rack-Mountable/Standalone Powered Mixers



S62

The S62 and S63 are ideal all-in-one device for recording, live concert and contracting applications. Built tough for reliable performance and long life, they feature 6 electronically balanced mic/line inputs (XLR and 1/4˝ connectors), a CD/tape input (phono connectors), switchable 48v phantom power, internal reverb, an Aux send and return, along with amp in and preamp and monitor out patch points. Rugged and versatile, they easily integrate into a variety of installations such as small clubs, theaters, schools, restaurants and houses of worship. Like all Samson's powered mixers, they combine comprehensive mixing and powering capability with extensive EQ and a wide range of useful features.

The S63 steps up with more power, 3-band EQ on each channel and a 10-band master graphic EQ that lets you "tune" the unit to particular environments, eliminating feedback problems.

- 6 channels with balanced XLR mic/line inputs for each channel allows you to blend a variety of source signals, including mics, keyboards, CD players, etc.
- Balanced XLR connectors (for mic inputs) and electronically balanced 1/4" jacks (for line-level inputs) for each channel; there is also a dedicated CD/Tape input that provides dual phono (RCA) jacks.
- 48v phantom power for mic inputs
- Clean, high-quality amplifier, delivers 150 watts into 4 ohms. Any standard speaker (2, 4, 8, or 16- ohms) can be connected to the rear-panel 1/4" speaker output jacks.
- A built-in three-spring reverb unit allows you to add reverberation effects to your vocals or instruments without having to use an expensive external signal processor. Front-panel reverb master volume control lets you precisely define the amount of reverb to be added to the signal.
- 2-band EQ on each channel enables you to shape the sound of your input signal sources
- Two Effects sends per channel (one pre-fade and the other post-fade) allow you to route multiple signals to the internal reverb unit or to external signal processors with a single front-panel control for the two. The pre-fade send (Effects send 1) can be used as a Monitor control, allowing you to set up an onstage monitor mix that is independent of the main house mix. A separate Monitor level control is provided on the front panel.
- In addition, a five-band graphic master EQ allows you to "tune" the output of the S62 to the particular room environment you are in. This can be particularly useful for eliminating ringing or feedback problems.
- Master Protection and Peak LEDs shows you at a glance when the internal amplifier is overloading.
- Protection relay circuitry prevents "thumps" when powering on or off. Allows you to use the S62 with a single power strip into which other audio devices are connected, without danger of damage to connected speakers.



S63

- 2 speaker outputs
- Portable design with tough reinforced corners and a carry handle. Can also be rack-mounted (3RU high) allowing integration into any setup. Supplied with rack ears.

S63 Step-up Features:

- Clean, high-quality amplifier delivers 200- watts into four ohms. Any standard speaker cabinets (two, four, eight, or sixteen ohms) can be connected to the rear-panel 1/4" speaker output jacks.
- 3-band EQ on each channel lets you shape the sound of your input signal sources.
- 10-band graphic master EQ allows you to "tune" the output of the S63 to the particular room environment you are in.
- Independent input Trim controls for each channel allows you to precisely set the correct input gain.
- A rear-panel amplifier input allows you to bring external signals from other mixers or audio devices into the SS63 power amplifier
- Rear-panel preamplifier output lets you connect the S63 to external power amps when more power is required and/or when additional amp feeds are necessary.
- 2 Aux sends per channel
- 2 stereo returns with dedicated level controls for outboard effects
- Master monitor and reverb level controls
- Front-panel LEDs show the current status of the amp's protection relay circuitry and whether phantom power is being applied.
- Portable design with tough reinforced corners and a carry handle. Can also be rack-mounted (4RU high) allowing integration into any setup. Supplied with rack ears.

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S83



8-Channel Rack-Mountable/ Standalone Powered Mixer

Fully loaded in terms of features and performance, the S83 steps up with 8 mic/line inputs , 300-watts of power into 4 ohms, 4-band EQ on each channel, front-panel output level meter, and headphone amp. The S83 also includes enough Sends and Returns so you always have adequate mixing capability.

- 8 channels with balanced XLR mic/line inputs for each channel
- Clean, high-quality amplifier, delivers 300 watts into four ohms. Any standard speaker cabinets (two, four, eight, or sixteen ohms) can be connected to the rear-panel 1/4" speaker output jacks
- 4-band EQ on each channel lets you shape the sound of your input signal sources
- Peak LEDs for each channel show you at a glance when an input signal is on the verge of overloading. Other front-panel LEDs show the current status of the amplifier's protection relay circuitry and whether or not phantom power is being applied.
- Four Effects returns give you the ability to blend in the return signal from external signal processors or other line-level devices without having to utilize input channels. Two front-panel Effects return level controls (one for Effects returns 1-2 and the other for Effects returns 3-4) allow you to bring in stereo signals, which are then automatically mixed together in mono. In the S63, two Effects returns are provided, with a single front-panel control for the two.
- Built-in headphone amplifier, with a front-panel 1/4″stereo connector and dedicated level control allows you to monitor your main mix
- Convenient front-panel meter let you see at a glance the continuous output signal level

PA324

Six-Channel Powered Mixer with 24-bit DSP Effects



The PA324 is a compact six-channel, 300-watt powered mixer with built-in 24-bit DSP effects. Optimized for live sound reinforcement and commercial installations, it handles up to six mic/line inputs with power to spare. The power amp section can run in three modes: two channel PA (150 watts per side), PA and monitor (150 watts each) or bridged for 300 watts of power. Each channel features a low noise, high quality mic preamp, 3-band EQ, monitor and effects sends, a level control and a 30dB pad on each of the six channels.

The onboard 24-bit DSP effects range from Hall, Room and Vocal reverbs to Chorus and Delay and can controlled via front panel footswitch jack. Other practical features include dual 7-band graphic EQ, tape in and out RCA connectors, Aux in, phantom power and dual 7-segment LEDs. Packaged in a rugged and reliable enclosure, the PA324 comes mounted in a durable carpet covered wood cabinet complete with corners and handle.

- Six mic/line inputs with 1/4″ phone and XLR connectors
- Built-in, 24-bit DSP with 8 selectable presets including Reverb, Delay and Chorus
- Advanced circuit design, utilizing low noise operational amplifiers
- Three power operating modes: 2x 150-watts, or 150-watt main/150-watt monitor, or 300-watts bridged
- Low-noise mic pre-amps with 48v phantom power
- 3-Band EQ on each channel
- Effects and monitor sends on each channel
- Dual 7-band graphic EQ for Main/Monitor
- Tape/CD Input.
- Mounted in a durable carpeted wood cabinet complete with corners, convenient carrying handle and large rubber feet

SAMSON

TM300/500

6-Channel Stereo Powered Mixer

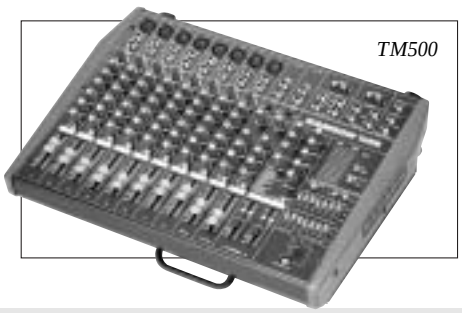


The TM300 is a practical, full-featured 6-channel stereo powered mixer with built-in signal processing in a compact table top design. The mixer can also be rack-mounted with the supplied rack ears. Ideal use in rehearsal studios, small club venues, lounge acts and other sound reinforcement applications, the TM300 can be used wherever there are a number of microphone or line-level sources that need to be mixed down to a stereo output. Simply hook it up to a set of PA speakers and you're ready to go.

The TM300 features a comprehensive mixer section with extensive routing and metering capability. Stereo power amp produces 150 watts a side into 4 ohms and uses a fan-cooled, bipolar design for greater stability and long life. A built in DSP section provides 3 different preset effects. 3-band EQ is complemented by a dual 7-band graphic EQ for more sonic control.

FEATURES

- Six independent channels, each with XLR mic and electronically balanced 1/4″ line inputs. A dedicated stereo Tape input provides dual phono (RCA) jacks.
- Flexible switching allows the TM300 to be used either in standard stereo configuration or in a unique "split mono" mode, where it can drive both main PA speakers and onstage monitors simultaneously.
- Built-in stereo power amplifier delivers 150 watts per side into 4 ohms. Any speaker cabinet (4, 8, 16-ohms) can be connected to the four rear-panel 1/4″ output jacks.
- Amp utilizes a variable-speed fan and bipolar design for maximum reliability and long life
- Onboard digital signal processor (DSP) with three high-quality effects
- 48v phantom power for mic inputs
- Two Aux sends per channel (one pre-fader and the other post-fader) allow you to route multiple signals to the internal DSP or to external signal processors
- Dual stereo aux returns—with dedicated front-panel level controls—give you the ability to blend in the return signal from the internal DSP and/or external signal processors or other line-level devices without having to utilize input channels
- 3-band EQ for each channel, with 15 dB of cut or boost for low (100 Hz) and high (10 kHz) frequencies, and 12 dB of cut/boost for the mid (1.8 kHz) frequency
- In addition, a dual 7-band graphic master EQ allows you to "tune" the TM300's output to the particular room environment you are in —ideal for eliminating ringing or feedback problems. The graphic EQ can be used in stereo or can be "split" so that one side affects the main PA and the other side affects stage monitors
- Constant level Pan controls for each channel that allow you to precisely place each input signal in the left-right plane
- Center detents for all Pan and EQ controls, makes it easy to use the TM300 even in low-light live performance situations
- Continuously adjustable wide-range input Trim controls for each channel allow you to precisely set the correct input and output gain structure



TM500

- Independent 65mm faders for each channel and for the stereo bus
- Pre Fader Listen (PFL) soloing for each channel, allowing headphone monitoring of individual channels, pre-fader (but post-EQ) without affecting the signal being output either by the speaker or various line-level outputs.
- Front panel ten-segment level meter allows you to view at a glance the continuous levels of the main stereo output and/or Aux sends
- Peak LEDs for each channel show when an input signal is on the verge of overloading
- Dedicated status LEDs show main power and phantom power on/off
- Separate Record out jacks (with dedicated level control) allow you to directly connect the TM300 to any tape recorder. In addition, a handy Mono output jack provides a line-level signal for connection to external monitor power amplifiers.
- An independent front-panel headphone output with dedicated volume control for private monitoring of the main stereo output, Aux sends, or soloed channels.
- Main inserts and four channel inserts allow you to use outboard signal processors such as equalizers, compressor/ limiters, or noise gates in a standard "effect loop."

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ONE HOUR FREE PARKING

AT 349 W. 34th STREET (with purchase of $100 or more)

EXPEDITION EXPRESS

All-In-One Portable PA System

Expedition Express is the total solution to professional sound, made portable. It gives you everything you need to make a professional presentation anywhere you can roll it. Expedition Express is a great solution for performers on the move. Bands, mobile DJ's, schools, houses of worship, corporate A/V and institutional applications-anywhere portable, high quality sounds required.

Unlike most portable PA systems, the Expedition Express is designed to sound great. Not just for public speaking either—but for live and recorded music, too. The system is comprised of a pair of rugged, acoustically balanced two-way loudspeakers, a 4-channel mixer with DSP, a powerful 250-watt amplifier and all the accessories you need. What's more, the system interlocks, and packs up into a single rolling unit so it's easy to transport.





The Expedition Express combines a powered speaker with built-in amplification and controls, and a light-weight second speaker or "slave". When set up for performance, the two cabinets are connected by a single cable. When they are packed of for the road, the lightweight passive speaker cabinet locks into the top of the amplified speaker cabinet with a unique auto-locate mechanism. Each loudspeaker is housed in a lightweight, yet heavy-duty, molded resin cabinet with extensive internal bracing to give it rigidity, low resonance and the maximum possible volume.



Pro Quality Electronics

The Expedition Express's integrated 250-watt amplifier is perfectly matched to the drivers for surprisingly high efficiency. The sound is clear and hard-hitting. Built-in digital signal processing (DSP) adds dramatic pro-quality effects like reverb and echo at the flip of a switch. Selecting from six preset effects makes getting great sound easy and fast.

- It features custom designed high and low frequency drivers. The High Frequency section employs an elliptical wave-guide horn that offers extra wide dispersion for maximum coverage.
- With molded-in handle and wheels, you can roll the cabinet anywhere you want. This makes Expedition Express ideal for boardrooms, classrooms and fitness studios, either as a portable speaker or mobile all-in-one sound system.
- The Expedition Express takes the guesswork out of public presentations. Everything you need is in one package: Speakers, built-in mixer with DSP, Samson microphone and all the cables. Best of all, you don't have to be an engineer to hook it all up. Sets up (and packs up) in less than five minutes.

ACCESSORIES

Rechargeable Battery
RB2030 rechargeable battery kit is for high quality sound anywhere electricity is not available - from the street to the beach, from parks to pool-side.....**89.95**

Optional Wireless Module
The Expedition Express is prewired to accept Samson UM-1 or Micro 32 Series wireless mics.

Built-in Cassette Player/Recorder **34.95**

EXPEDITION PRO SERIES



Three Enclosures, Unlimited Possibilities

The Expedition series are versatile two-way loudspeaker enclosures with Celestion woofers, telescoping handle and locking wheels, available in three different versions: passive (XP100), powered (XP200) and powered with a built-in 4-channel mixer and digital signal processing (XP300).

Delivering huge sound with lots of headroom, a smooth top end and articulate bass response—without taking up much space, these loudspeakers represent smart solutions for clubs and performance halls, mobile DJ use, houses of worship, corporate A/V, fixed installations and any application where portable audio is required.

Incredibly rugged, Expedition enclosures are injection-molded with Polypropylene construction with substantial internal bracing supporting the nearly 1/2″ thick side wall construction. Convenient fly-points, a steel grille and a scuff-resistant textured finish ensure extreme versatility, ruggedness and dependable performance.

To finish your system, a range of Expedition options are available including a rechargeable battery kit, a cassette recorder/player with pitch control and plug-in Samson VHF and UHF micro wireless systems.

XP100

The passive XP100 enclosure is housed in a sleek molded cabinet with substantial internal bracing to support its thick side wall construction. Designed to be lightweight and compact, the XP100 is still rigid enough to allow maximum energy to be transferred from the cabinet into the room. Two 1/4″ Speakon connectors, convenient fly-points, a steel grill and scuff-resistant textured finish ensure exceptional versatility and long life.

The XP100 is a passive enclosure equipped with a custom designed, Kevlar-impregnated 12″ Low Frequency driver with a 2.5″ voice coil and a 49 oz. barium ferrite magnet to produce an accurate, super-tight bass response. The XP100's High Frequency section employs a 1″titanium compression driver along with an elliptical wave guide horn that controls dispersion. This in combination with a phasing plug produces a linear overall response with a clear, musical top end.

XP200

Using the same cabinet and speaker design as the XP100 the XP200 adds 200-watts of bi-amped power. It also includes an active crossover and built-in limiter with two XLR inputs.

The XP200's power module gives you three stages of speaker protection, including relay switching for power up and power down. The onboard limiter circuit ensures a clean output even when you push it to maximum sound pressure levels. Because of its perfectly matched amplifier and speakers, the XP200 is particularly effective as a powered monitor.

AC daisy-chain output for adding additional Expeditions enclosures in stereo, recording and presentation or performance applications.

exceptional versatility



Because of their telescoping (as well as retractable) handle and wheels, you can pull out the handle and roll the cabinet anywhere you want. This makes the Expedition series ideal for boardrooms, classrooms, and fitness studios, either as a portable speaker (XP100 or XP200) or mobile all-in-one sound system (XP300).

Optional hardware allows you to mount Expedition cabinets on wall and ceilings. The MP1020 mounting plate makes it easy to install the cabinets anywhere you want and is compatible with industry standards such as Omnimount.

In an angled position, they make perfect wedge monitors



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EXPEDITION PRO SERIES



XP300

The most comprehensive version of Expedition, the XP300 is a powered loudspeaker with a built-in 4-channel stereo mixer and digital effects processing. Designed as a completely portable, all-in-one PA system, the XP300 is ideal for people who want superior audio anywhere they go. Whether you just want rugged and versatile loudspeaker enclosure that consistently deliver big sound, or a complete professional audio system in a highly mobile design perfect for today's fast-paced world, the XP300 is the answer.

The XP300's mixer features two bands of equalization, two Mic/Line channels and a Stereo Channel with a mic preamp. In addition to a standard TRS stereo line input, channel three also has RCA inputs that allow you to easily interface a CD, Mini Disk or Tape player. The digital processing section consists of six high-quality presets that range from a touch of room ambience to lush, reverberant halls. Versatile and easy to use, the XP300's mixer has balanced XLR Left and Right outputs for daisy-chaining additional enclosures (XP200) in stereo, recording and presentation or performance applications. The mixer's output can be monitored via its multi-segment VU meter, and there is also an LED meter indicating power level available from the optional RB20/30 rechargeable battery cartridge.

Like the XP100/200, the XP300's enclosure is equipped with a custom designed, Kevlar-impregnated 12″ Low Frequency driver with a 2.5″ voice coil and a 49 oz barium ferrite magnet to produce an accurate, super-tight bass response. The XP300's High Frequency section employs a 1″ compression driver along with an elliptical wave guide horn that controls dispersion. This in combination with a phasing plug produces a linear overall response with a clear, musical top end.

EX500 Active Subwoofer

The EX500 active subwoofer can produce 500 watts of power in tandem with a custom, heavy-duty 15″ low frequency driver. It functions as an ideal complement to Expedition series cabinets, or any loudspeaker system, when the application requires a focused yet deep bass response. The built in stereo crossover lets you use the EX500 in mono or as a common subwoofer in a stereo system. It also features a steel grill and rigid corners for durability, as well as an integral pole mount for conveniently mounting Expedition cabinets.



db500a Active Monitor System



Inside the rigid custom designed cabinet is an exclusively designed 15″ driver, matched with a 1.75″ Neodymium high frequency driver, aimed through a 1″ diameter horn. The dB500a features 500 watts of Class H power (400 watts low frequency/100 watts high frequency) and has a crossover designed for high SPL. The exclusive Optimax signal processing circuitry provides total speaker protection but won't cut the low end at high volumes, as similar speakers with sliding filters do.

The back panel has a built-in mixer-like preamp section with mic and line inputs, XLR connectors and level controls. Controls for the selectable low pass filter and Optimax processing are also included.

The dB500a speaker cabinet features an array of mounting options. Designed with floor monitoring in mind, they are also easily stacked or permanently installed. A 1⅜″ pole receptacle is built in, and 10 reinforced fly points are available.

YAMAHA

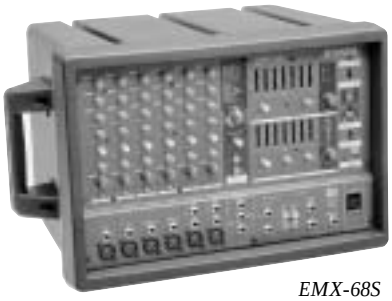
EMX-62M/66M/68S/88S

Powered Mixers



EMX-66M

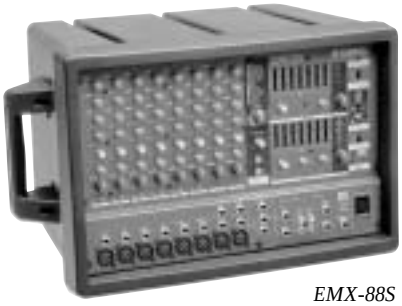
Ideal for small-to-medium sound reinforcement applications such as solo performers, bands, houses of worship and corporate use, Yamaha's EMX series of powered mixers offer portability, simplicity, economy and reliability. They feature built-in amplification and onboard digital effects, including Yamaha's acclaimed reverb technology. Internal amplifiers deliver 200-400 watts of efficient power and contain proprietary YSP circuitry that tailors the response curve to match Yamaha Club Series speakers. They are housed in rugged high-impact molded enclosures, and since many of today's permanent and portable applications have less space —they can be rack-mounted for integration into any system or installation (optional rack-mounting kit required). This is also made possible by a highly efficient fan cooling system that ensures reliable, stable operation in just about any environment.



EMX-68S



EMX-62M



EMX-88S

EMX62M

The six-input EMX62M features a single 200-watt (4 ohm) amplifier and built-in limiter to prevent damage from feedback or overload. Channels 1-4 have balanced XLR connectors with switchable pads and phantom power, plus balanced 1/4″ jacks. Inputs 5 and 6 each include two unbalanced inputs. All channels contain 2-band EQ, plus independent monitor and effects sends. RCA connectors with separate level control for a tape recorder or a CD player are also conveniently located on the front panel. The unit also contains a professional quality 7-band graphic equalizer and eight digital effects. Offers solo performers and public speakers who need a really clean, simple system, all the creative control and power they need for small shows and events.

EMX66M

The 6-channel EMX66M has all the features of the EMX62M plus adds two mono 300-watt (4 ohm) amplifiers (600 watts bridged) that can drive two house speakers or; one can drive the house speaker while the other drives a monitor system, 3-band channel EQ, and two 7-band graphic EQs (mains and monitors) for feedback control and tone shaping.

EMX68S/EXM88S

The EMX68S features a 6-channel stereo design with two switchable 400-watt (4 ohm) amplifiers, an All Channel Mute switch and 16 built-in digital effects - based on the popular Yamaha SPX effects processors. The EMX88S is the same as the EMX68S, with the addition of an 8-channel stereo design.

Why a YAMAHA Powered Mixer?

So you've decided that you need a powered mixer, why should choose a Yamaha? Start with a solid reputation for professional quality based on more than 25 years experience in the field. And don't forget that Yamaha also makes some of the world's finest acoustic pianos, stringed instruments, wind instruments, drums, and electronic instruments as well as industry-standard sound reinforcement gear. In short, the Yamaha name is synonymous with leading-edge technology that also embodies a unique sensitivity to the artistic needs of the musicians and performers who use it. But since we're talking about powered mixers, here is the main reason—superior sonic quality. Yamaha's engineers, sound specialists and instrument designers listen carefully to each product at every possible design stage, making changes and refinements that may not appear in the specifications, but that make an audible difference to the final sound.

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EQUIPMENT LEASING AVAILABLE

EMX-62M/66M/68S/88S

Inputs

- Four mono XLR mic and 1/4″ line inputs. A global +15V phantom power switch is provided for the mic inputs, and an independent pad switch on each channel allows input level matching.
- Channels 5 and 6 provide the option of 1/4″ TRS line level inputs channel or 1/4″ Super Hi-Z instrument inputs optimized for direct input of acoustic and electric guitars and basses.
- Each input channel features level control and 2-bands of EQ (Hi/Low) as well as separate Monitor and Effect sends
- The Effects sends can be used to control the level to the internal effects (or an outboard effect processor)

Digital Effects

- Choice of three high-quality digital reverb effects (vocal reverb, large hall, small hall) to add ambience and depth to your sound.
- Separate Effect Returns are provided for the main and monitor sections.

Main Section

- Main section Master control adjusts the overall level of the main house sound
- A 5-segment LED level meter is provided as well as a LED Limiter indicator that tells you when the internal limiter circuitry is active due to power amplifier overload
- 7-band graphic EQ provides flexible sound-tailoring and feedback control for the main output.
- 1/4″ Aux In jack allows the signal from an auxiliary source or sub-mixer to be mixed into the main program.
- 2TR (RCA) input, with a dedicated level control, can be used to mix the output from a tape deck or other stereo source into the main outputs.
- 1/4″ main output can be used to drive an external power amp or sound system from the main program.
- The dual RCA Rec Out jacks delivers the main mix signal to an external recorder

Monitor Section

- A Master control knob adjusts the volume of the monitor output
- 1/4″ monitor output delivers a line-level signal to an external sound system.

Output Section

- 1/4″ speaker 1 and 2 outputs allow connection of two 8-ohm speakers or a single 4-ohm speaker
- Yamaha Speaker processing switch for smoother high frequencies and enhanced low-end output from Yamaha speakers
- A 1/4″ headphone output is also provided

Conveniences

- 1/4″ Effect output delivers the signal from the channel Effect send controls to an external effect processor
- A 1/4″ jack accepts an optional FC5 foot switch to turn the effects on or off

EMX66M Adds

- A 3-band EQ (Low, Mid and Hi) is provided for each input channel
- XLR and 1/4″ TRS inputs on channels five and six
- 8 effects programs including Vocal Echo and Plate
- Two 300W amplifiers (4 ohms) can be operated in three modes: Main + Main; Main + Monitor; Main Bridged

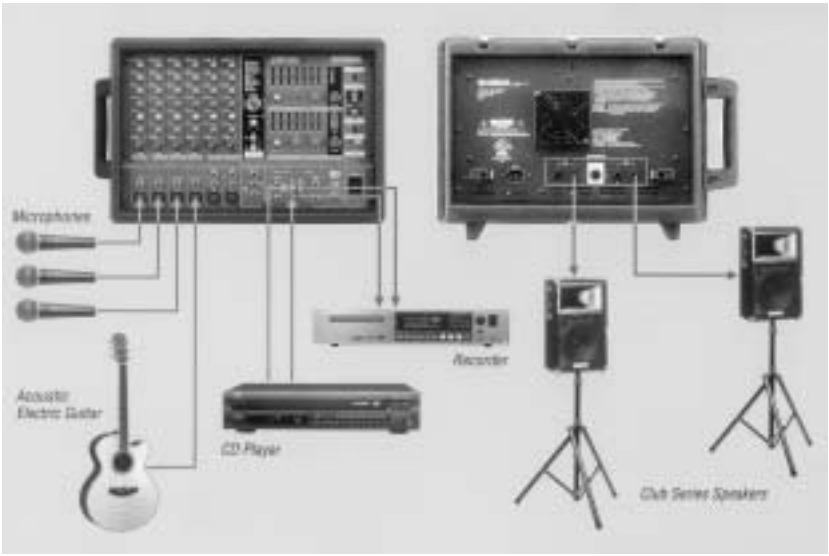
EMX68S Adds

- Two 400W amplifiers (4 ohms) can be configured for left and right Stereo Main outputs; Main (L+R) + Monitor outputs or Main (L+R) Bridged
- Pan controls are provided for each input channel
- Channels five and six can each be used as stereo line or mono mic inputs
- The 1/4″ Auxiliary inputs, RCA Tape inputs and Record outputs are all configured for stereo operation
- SPX 32-bit digital effects processor and 24-bit A-D-A converters with 16 programs including Reverb, Gated Reverb, Echo, Delay, Vocal Doubler, Flanger, Distortion and more
- A second 1/4″ FC5 compatible footswitch input is available for used as a Tap Termpo control for the effects processor
- A Standby switch allows you to instantly mute all inputs except for the Tape Inputs

EMX88S Adds

- Two additional XLR mic 1/4″ line inputs are provided for a total of six mono inputs and two stereo/mono inputs

YAMAHA MIXERS APPLICATIONS

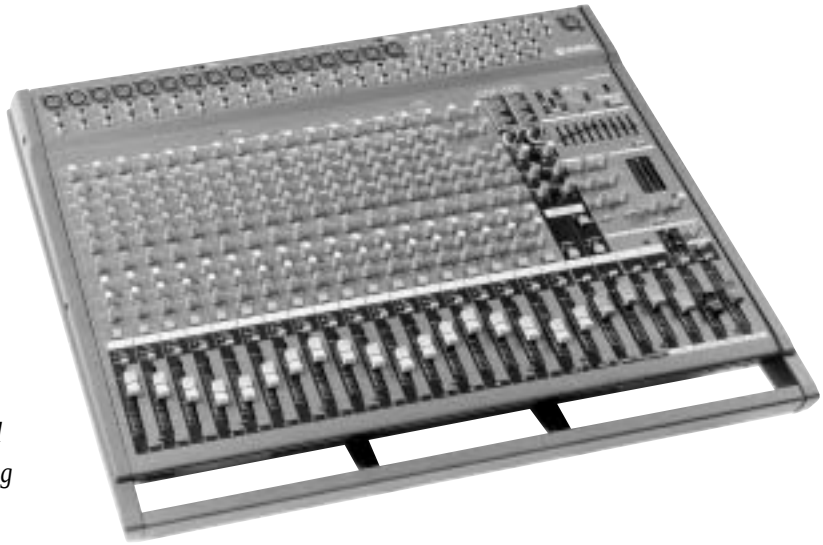


YAMAHA

EMX5000

Powered Mixers

If your show demands serious input capacity and power, the 12- channel EMX5000-12 or the 20-channel EMX5000-20 will probably do the job with performance to spare. Both feature a 500-watt per channel multi-mode stereo power section that will fill even fairly large venues with outstanding sound. And although they're integrated powered mixers designed with convenience and portability in mind, both models deliver features and performance that will satisfy even the most discerning performers and sound engineers.



FEATURES

Input Channels

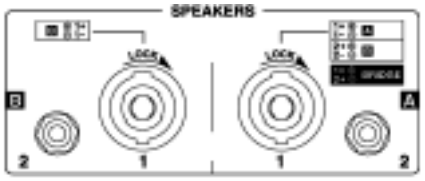
- Available in 12 channel (8 mono and 2 stereo) and 20 channel (16 mono and 2 stereo) configurations
- The mono input channels offer a choice of balanced XLR mic and 1/4″ TRS line inputs as well as 1/4″ TRS insert points for insertion of external compressors, equalizers, or other processing gear
- Balanced 1/4″ TRS and unbalanced RCA inputs are provided for the stereo input channels
- Each input channel features an input gain control, a high-quality linear fader for controlling levels; a Pan control; a PFL solo switch for monitoring individual channels; signal and peak LEDs; two Effect sends; Aux 1 and Aux 2 send controls with pre/post fader switching; and a Channel ON switch for muting individual channels
- Mono input channels also feature a 26 dB pad, 3-band EQ with sweepable mid band as well as a switchable 80Hz high pass filters for elimination low-frequency noise and rumble – stereo channels have a 3-band EQ with a fixed mid band
- The two Effect sends controls can be used to feed a channel's signal to the dual high-performance internal effect systems, or to outboard effect processing gear
- 48V phantom power is switchable in groups of 8 channels for using condenser mics
- Dual 1/4″ TRS Stereo Aux inputs with level controls and PFL switches allow convenient connection to sub-mixers or other stereo sources
- Unbalanced RCA 2 track inputs with level control can be used to mix the output from a tape deck or other stereo source into the main and monitor outputs
- Unbalanced RCA record output jacks allow you to route the main stereo program signal to an external recorder

Dual Effects Processor

- Each processor offers a selection of 16 professional-quality SPX digital effects
- A parameter control for each processor can be adjusted to tailor the effect. Settings are memorized for each effect program
- A Tap-delay button makes it easy to produce tempo-synchronized delays
- 1/4″ TRS Effect Send jacks allow you to route the signal from an channel's Effect send control to an external effect processor
- Aux 1 and 2 send controls adjust the amount of effect signal sent to the auxiliary buses

Outputs

- 1/4″ outputs for Aux sends 1 and 2 deliver the corresponding auxiliary signal for external processing or monitoring
- 1/4″ stereo outputs can be used to drive an external power amplifier or sound system from the main program
- 1/4″ power amp input connectors facilitate the insertion of external equalizers or other processors into the main program
- Stereo 1/4″ Sub outputs controlled by a level control, above the mains fader, allows sub-mix cascading to a second mixer
- A 1/4″ mono subwoofer output is provided featuring a switchable low-pass filter with a screwdriver adjustable cutoff frequency variable from 80 to 120Hz



- High-quality SPEAKON speaker connectors for fast, reliable connection. 1/4″ jacks are also provided
- Headphone output jack with independent level control

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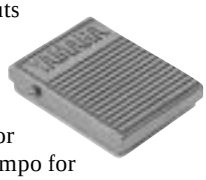
EMX5000

Master Section

- A Stereo master fader with both PFL and AFL switches
- Linear Aux 1, Aux 2, and Mono Out master faders with AFL (after-fader listen) monitor switches
- Precise 12-segment stereo level meters
- A 9-band graphic equalizer with ON switch provides flexible sound-tailoring and feedback control
- Linear Effect return faders with PFL and ON switches
- A Power amplifier output selector allows selection of 500, 300, or 100 watt output per channel
- A Power Amp mode selector allows the two-channel power amplifier to be quickly configured for Main + Main, Mono + Aux 1, Aux 1 or Aux 2, or Bridged operation
- Internal limiter circuitry with LED indictors help control power amplifier overload
- A Yamaha Speaker processing switch delivers smoother high frequencies and enhanced low-end output from Yamaha speakers

Conveniences

- A Stand-by switch allows you to instantly mutes all mono inputs
- Foot Switch jacks accept optional FC5 foot switch to turn the effect 2 on or off, and tap in the tempo for the tempo delay effect "on the fly"



- Carrying handle doubles as an armrest for extra control comfort
- Top Panel BNC Lamp socket

EMX SERIES SPECIFICATIONS

	EMX62M	EMX66M	EMX68S	EMX88S	EMX5000-12	EMX5000-20
Maximum Output Power @0.5% THD at 1kHz	200w/4Ω 135w/8Ω	300w+300w/4Ω 205w+205w/8Ω 600w/8Ω bridge	400w+400w/4Ω 270w+270w/8Ω 800w/8Ω bridge	400w+400w/4Ω 270w+270w/8Ω 800w/8Ω bridge	500w+500w/4Ω 325w+325w/8Ω 1000w/8Ω bridge	500w+500w/4Ω 325w+325w/8Ω 1000w/8Ω bridge
Input Connectors	CH 1-4: XLR & TRS phone CH5, 6: 2x TRS phone	CH 1-4: XLR & TRS Phone CH5, 6: XLR and 2x TRS phones	CH 1-4: XLR & TRS phone CH5, 6: XLR and 2x TRS phone	CH 1-6: XLR & TRS Phone CH7, 8: XLR and 2x TRS phones	CH 1-8: XLR & TRS phone CH9/10, 11/12: 2x phones & 2x RCA pins	CH 1-16: XLR & TRS Phone CH17/18, 19/20: 2x phones & 2x RCA pins
Channel Inserts	—	—	—	—	8 (mono channels)	16 (mono channels)
EQ (Mono CH) High	10kHz shelving	10kHz shelving	10kHz shelving	10kHz shelving	10kHz shelving	10kHz shelving
EQ (Mono CH) Mid	—	2.5kHz peaking	2.5kHz peaking	2.5kHz peaking	250Hz-5kHz peaking	250Hz-5kHz peaking
EQ (Mono CH) Low	100Hz shelving	100Hz shelving	100Hz shelving	100Hz shelving	100Hz shelving	100Hz shelving
EQ (Stereo CH) High	—	—	10kHz shelving	10kHz shelving	10kHz shelving	10kHz shelving
EQ (Stereo CH) Mid	—	—	2.5kHz peaking	2.5kHz peaking	2.5kHz peaking	2.5kHz peaking
EQ (Stereo CH) Low	—	—	100Hz shelving	100Hz shelving	100Hz shelving	100Hz shelving
Aux Sends	2 (1 monitor,1 effect)	2 (1 monitor,1 effect)	2 (1 monitor,1 effect)	2 (1 monitor,1 effect)	4 (2 AUX [pre-Post], 2 effect)	4 (2 AUX [pre-Post], 2 effect)
Phantom Power	+15V	+15V	+15V	+15V	+48V	+48V
Graphic Equalizer	7 band	7 band	7 band	7 band	9 band	9 band
Digital Effects	3 programs	8 programs	16 programs	16 programs	Effect 1 (16 programs), Effect 2 (16 programs)	
Power Amp. Mode	Main	Main + Main, Main + Monitor, Main Bridge,	ST L + ST R, Main (L+R) + Monitor, Main (L+R) Bridge	ST L + ST R, Main (L+R) + Monitor, Main (L+R) Bridge	ST L + ST R, Aux1 + Mono, Aux 1 + Aux 2, MONO Bridge	ST L + ST R, Aux1 + Mono, Aux 1 + Aux 2, MONO Bridge
Power Select Switch	—	—	—	—	500/300/100w	500/300/100w
Meters	5 points (Main)	5 points (Main), 5 points (Monitor)	2x 5 points (Main), 5 points (Monitor)	2x 5 points (Main), 5 points (Monitor)	2x 13 points	2x 13 points
Yamaha Speaker Processing	Yes	Yes	Yes	Yes	Yes	Yes
Low Pass Filter	—	—	—	—	Yes (MONO OUT)	Yes (MONO OUT)
Stand-by switch	—	—	Yes	Yes	Yes	Yes
LAMP Connector	—	—	—	—	Yes (XLR 3 pins)	Yes (XLR 3 pins)
Foot Switch	Effect On/Off	Effect On/Off	Effect On/Off, TAP	Effect On/Off, TAP	Effect 2 On/Off, TAP	Effect 2 On/Off, TAP
Dimensions (WHD)	18 15/16 x 12 x 12 7/8″	18 15/16 x 12 x 12 7/8″	18 15/16 x 12 x 12 7/8″	18 15/16 x 12 x 12 7/8″	18 13/16 x 6 1/4 x 21 3/16″	26 7/8 x 6 1/4 x 21 3/16″
Weight	26.5 lbs.	33.1 lbs.	34.2 lbs.	34.2 lbs.	33.1 lbs.	41.9 lbs.

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YAMAHA

CLUB SERIES IV

Ideal for many applications including bands, clubs, church installations and small theaters, these rugged, durable performers have a proven track record. For over five years Yamaha Club Series IV speakers have been making performances everywhere easily heard by the audience and performers alike. Club Series IV speakers feature rugged steel handles, metal corner protectors, and heavy-duty steel grille for easy and secure handling. Two input jacks mounted on a metal jack plate allow for loop-through connections. Rugged carpet-covered, trapezoid-style cabinets allow seamless array arcs. Black carpet covering also helps them stand up to abuse without showing ugly "road rash."





S112IV *and* S115IV

12- and 15˝ 2-way Loudspeaker Systems

The S112IV and S115IV are trapezoid shaped cabinets, with oversize steel handles, steel corner protectors, steel jackplate with two heavy duty 1/4˝ jacks, and a built-in pole socket (1⅜˝) for easy stand mounting. The S112IV's 12˝ woofer is a high power, custom design, the S115IV's 15˝ woofer is a high power, cast aluminum frame, with a cooling port in the pole piece for extended high power handling. The high frequency driver on both the S112IV and S115IV is a custom 2˝ titanium diaphragm (1˝ exit) design with an exclusive phase plug that reduces harmonic distortion inside the driver itself. This driver is mated to a 90° x 40° constant directivity horn to give smooth, even coverage over the pattern of the horn. The internal passive crossover has protection circuitry to help guard against damage that may occur from over powering or sending clipping signals to the speakers.

SM10IV/SM12IV/SM15IV

10-, 12- and 15˝ 2-way Floor Monitor Systems

The SM10IV, SM12IV and SM15IV are are low-profile, (horn beside woofer style) floor monitor cabinets, with steel corner protectors, steel grille to protect the woofers, and a heavy steel jackplate with 2 heavy-duty 1/4˝ jacks. A 1⅜˝ pole socket is installed on the end of the cabinet for easy pole mounting, FOH (front of house) duty. They are covered in tough, abuse resistant black carpet.



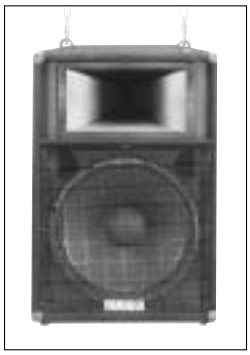


Their low frequency driver is a high power, cast aluminum frame, with a cooling port in the pole piece for extended high power handling. The high frequency drivers are a custom, 2˝ diaphragm (1˝ exit) design with an exclusive phase plug that reduces harmonic distortion inside the driver itself. This driver is mated to a 40∞ x 90∞ constant directivity horn to give smooth, even coverage over the pattern of the horn. The internal passive crossover has protection circuitry to help guard against damage that may occur from over powering or sending clipping signals (underpowering) to the speakers.

CLUB SERIES IV

S115IVA

Flyable 15″ 2-way Speaker



The S115IVA is the same as the S115IV 15″ 2-way speaker, except it has flying hardware tested and pre-installed in the cabinet. Steel strapping spanning these points inside the cabinet further ensures safety. Two 3/8″-16 forged eyebolts are included.

S215IV

Dual 15″ 2-way Speaker

The S215IV is a trapezoid shaped cabinet, with oversize steel handles, steel corner protectors, and a steel jack-plate with 2 heavy-duty 1/4″ jacks. Unique cabinet bracing for extra rigidity. Its covering is tough black carpet, and it has a heavy gauge perforated steel grille for driver protection. The woofers, or low frequency drivers are high power, Cast aluminum frame, with a cooling port in the pole piece for extended high power handling. The High frequency driver is a custom, 2″ diaphragm (1″ exit) design with an exclusive phase plug that reduces harmonic distortion inside the driver itself. This driver is mated to a 90∞ x 40∞ constant directivity horn to give smooth, even coverage over the pattern of the horn.





SW118IV

18″ Subwoofer

The SW118IV is a low profile subwoofer system. Using a cast frame 18″ driver, it is capable of delivering powerful low frequency energy to augment smaller systems. It is one of the most cost effective ways to add extra bass to your system. Large, heavy steel handles make transporting the SW118IV an easy one person job.

SW215IV

Dual 15″ Subwoofer



The SW215IV is a dual 15″ subwoofer in a particle board enclosure (needed for rigidity). The drivers in the SW215IV are butyl surround, long travel, 3″ voice coil, vented pole drivers. They assure very high SPL (sound pressure level) operation without damage or fatigue. The enclosure is rectangular in shape, and the baffle is designed to fold back in the center of the cabinet to form the port for this bass reflex design. This unique construction keeps the large top and bottom sheets that form the top and bottom of the cabinet from resonating. This makes the cabinet one of the most rigid boxes available today. Installed in the top of the cabinet is a metal socket that accepts a 1⅜″ diameter tube to allow satellite placement of another loudspeaker.

CLUB SERIES IV 'S' SPEAKERS



The S version of the popular Club Series IV provides the same high level of sonic performance while adding an attractive cosmetic alternative to the line. The Club S Series features a high-quality, extremely durable black polyurethane elastomeric bonded exterior that creates a high-tech look and offers greater shock absorption. The durable, all-weather elastometric coating also has a low gloss, understated appearance that provides a professional appearance and allows the cabinet to be easily painted for integration into any environment. Like the Club IV and Club IV-OAK, the Club S cabinets include dual 1/4″ connectors, corner protectors and heavy-duty carrying handles. Heavy gauge, perforated steel grills provide additional component protection and guarantees years of great performance.



YAMAHA

OAK SERIES IV



Speakers that look as good as they sound

Yamaha took high quality components and advanced engineering from the Club Series and placed them in trapezoidal oak finish cabinets. Now, you can enjoy a fine furniture appearance or paint the cabinets to match your specific surroundings.

CP2000 Power Amplifier



The CP2000 delivers a maximum 2000 watts of clean power into 4-ohm load when bridged. In a stereo configuration it delivers 1000 watts of power into 2-ohms, 650 watts into 4-ohm loads or 450 watts into 8-ohm loads. Yamaha's exclusive EEEngine technology, combined with a MOSFET amplifier allows the CP2000 to deliver high performance along with exceptional efficiency.

Yamaha Speaker Processing Circuitry enhances audio performance when the CP2000 is used with Club Series speakers. Limiter circuits (same as in Yamaha's high end H Series amps) provide optimum protection from hard clipping.

CP2000 features three output modes for maximum flexibility. The normal Stereo mode (separate input signals routed to independent channels with L/R signal control). Parallel Mode feeds a single input to two independent channels with L/R signal control, and Bridged Mono for extra high power output. Equipped with balanced XLR jacks and balanced 1/4″ phone jacks for input connections, and 1/4″ phone jacks and 5-way binding posts for speaker connections.



SE SERIES

The Se Series is a cost-effective, high-quality alternative for touring or small-to-medium size permanent systems. Combining Yamaha's advanced acoustics research with computer-controlled woodworking technology, they offer state-of-the-art components and rugged enclosures that will satisfy the requirements –and budget– of any consultant, contractor or engineer who needs a reliable PA system. Designed for optimal individual performance or as part of a system, each speakers includes a proprietary constant-directivity horn coupled to a 1″. 90º x 40º compression driver, which produces a flatter response, greater intelligibility and better overall performance than piezo-type tweeters. Each has a crossover frequency of 3 kHz, and are built to handle the power delivered by contemporary amplifiers.

In addition to dual 1/4″ connectors, all Se Series speakers offer a host of roadworthy features including durable "carpet" finish, corner protectors and heavy-duty carrying handles. Heavy-gauge, perforated steel grills provide additional component protection and guarantee years of great performance and durability.

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CLUB SERIES SPECIFICATIONS

Model	Woofer	Tweeter (diaphragm)	Frequency Response (-10dB)	Power Handling (noise*/program/peak)	Sensitivity 1W, 1m	Dimensions W x H x D	Weight (lbs.)
CLUB Series IV Series							
S215IV	15″ Cast Basket (x2)	2″ Titanium	42Hz-16kHz	500/1000/2000	99	20 x 48 x 24″	108
S115IV/S115IVA	15″ Cast Basket	2″ Titanium	55Hz-16kHz	250/500/1000	99	19 x 28 x 14″	61
SM15IV	15″ Cast Basket	2″ Titanium	55Hz-16kHz	250/500/1000	99	28 x 19 x 14″	57
S112IV	12″ Steel Basket	2″ Titanium	60Hz-16kHz	150/300/600	97	25 x 16 x 14″	43
S112IV	12″ Steel Basket	2″ Titanium	60Hz-16kHz	150/300/600	97	26 x 25 x 13″	43
SM12IV	12″ Steel Basket	2″ Titanium	60Hz-16kHz	150/300/600	97	25 x 16 x 14″	43
SM10IV	10″ Steel Basket	1″ Phenolic	70Hz-20kHz	100/200/400	95	22 x 13 x 11″	27
CLUB Series IV S Series							
S115IVS/S115IVAS	15″ Cast Basket	2″ Titanium	55Hz-16kHz	250/500/1000	99	19 x 28 x 14″	61
S112IVS	12″ Steel Basket	2″ Titanium	60Hz-16kHz	150/300/600	97	16 x 25x 13″	43
SM12IVS	12″ Steel Basket	2″ Titanium	60Hz-16kHz	150/300/600	97	25 x 16 x 14″	43
Subwoofers							
SW118IV/118IVS	18″ Cast Basket	n/a	30Hz-2kHz	250/500/1000	96	21 x 26 x 31″	71
SW215IV	15″ Steel Basket (x2)	n/a	33Hz-2kHz	500/1000/2000	98	44 x 21 x 24″	141
OAK Series							
S115IVA-OAK	15″ Cast Basket	2″ Titanium	55Hz-16kHz	250/500/1000	99	19 x 27 x 14″	61
SM15IV-OAK	15″ Cast Basket	2″ Titanium	55Hz-16kHz	250/500/1000	99	27 x 19 x 20″	57
S112IV-OAK	12″ Steel Basket	2″ Titanium	60Hz-16kHz	150/300/600	97	16 x 24 x 12″	40
SM10IV-OAK	10″ Steel Basket	1″ Phenolic	70Hz-20kHz	100/200/300	95	21 x 12 x 15″	23
SE Series							
S15e	15″ Steel Basket	1″ Phenolic	60Hz-14kHz	125/250/500	98	19 x 26 x 15″	44
S12e	12″ Steel Basket	1″ Phenolic	65Hz-14kHz	125/250/500	97	16 x 22 x 13″	32
S12Me	12″ Steel Basket	1″ Phenolic	65Hz-14kHz	125/250/500	97	22 x 15 x 18″	30
S10e	10″ Steel Basket	1″ Phenolic	70Hz-14kHz	100/200/400	96	16 x 21 x 13″	30

MS300 *300-Watt Powered Speaker System*



The MS300 is a 2-way speaker system that delivers high-quality, high-powered sound in a light-weight, easy to carry and setup unit. Each unit is equipped with an internal bi-amplified power system that delivers a total of 300 watts—225 watts to the 12″ low frequency driver and 75 watts to the 2″ high frequency driver.

It is also equipped with an easy to use but effective built-in mixer that provides independent MIC and LINE inputs, complete with a separate gain control on each input, a two-band EQ (Low, High) and a Master Volume control. The LINE input signal can be sent to other speakers via the Parallel output jack which allows the daisy chaining of multiple speakers. Equipped with a variable high cut frequency control to optimize integration of a subwoofer into a variety of sound systems.

For true versatility, the MS300's design allows a wide range of setups since it is flyable, stackable and pole mountable right out of the box. It can even be used as a floor monitor when laid on its side. Optional mounting brackets allow the MS300 to be mounted on the wall or from the ceiling.









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The Professional Audio SourceBook

Section 10

Wireless Microphones

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WIRELESS MICS

INTRODUCTION TO WIRELESS MICROPHONES

The decision to go wireless represents both a commitment to your craft as well as the faith that technology will deliver the results you have in mind. To glean the most of both of those ideals, a clear understanding of the possibilities and limitations of wireless operation should be established.

Wireless systems afford freedom of movement that isn't possible with a traditional hard-wired signal interface. However, in order for this mobility to occur, a specialized radio link is required. Thus, not only do the operational requirements of standard microphone capsule apply, but also the laws of radio, both legal and electrical, apply as well.

Wireless microphones must meet legal requirements—they must have a FCC license number—as well as the electrical laws of radio. The #1 rule of radio is that two signals cannot successfully operate on the same frequency simultaneously within the transmission range of the system. Should this occur, drastic performance problems will arise. A wireless receiver "listens to" any signal that is being transmitted on the frequency it is tuned to, and if more than one signal is being "heard" by the receiver, neither one will be usable.

Given this most essential rule, it is impossible to operate more than one wireless transmitter on any given frequency, and to operate on a frequency that is being used by any other signal source such as television or nearby wireless communications. When specifying a wireless mic system with more than one transmitter, the number of receivers must match the number of transmitters and a plan for multiple frequency occupation must be established where each system operates on their own discrete frequency. The #2 rule of radio is that there is no limitation on the number of simultaneously operated receivers. Just as in radio or TV where any local receiver can pick up a broadcast, the receiver-to-transmitter ratio is unlimited. This aspect is especially convenient in wireless mic systems when there may be more than one destination of the transmitted signal.

Since contemporary wireless microphone systems are designed to operate in either the VHF or the UHF radio spectrum, an understanding of those terms is important. TV broadcast requirements have greatly influenced the development of radio frequency spectrum usages. The portion of the frequency spectrum determined as suitable for TV signals is relatively wide and is broken down into two general areas. VHF (very high frequency) is defined as frequencies between 54 MHz and 216 MHz, with a large gap between 88 MHz and 174 MHz that is used for signals other than television. UHF (ultra high frequency) is currently defined as frequencies between 470 MHz and 806 MHz. These two major bandwidth sections (VHF and UHF) are divided up into smaller segments commonly known as TV channels. In order to accommodate the complex signal information required for color television broadcast, a TV channel is actually 6 MHz wide. The VHF bandwidth is dedicated to TV channels 2 – 13; UHF to TV channels 14 – 69. Since the UHF spectrum is far wider than VHF, it is more desirable for wireless microphone usage due to reduced congestion. Each TV channel has an assigned bandwidth, and knowing the frequencies that are affected by these signals is very important when choosing wireless microphone system tuning bandwidths and operating frequencies.

Because wireless microphone systems share the RF (radio frequency) spectrum with television, and due to the fact that attempting to operate a wireless microphone system within the bandwidth of a local TV broadcast signal will render poor results, it is advisable that you research your local RF environment to determine what bandwidths are being used by TV broadcasters, which will in turn identify which bandwidths are clear for wireless system operation. The web makes this information readily available. The FCC regulates TV broadcasters and they publish information at http://www.fcc.gov/mb that may help you find sources of local interference. Manufacturers also offer similar information. Check out the manufacturer's web site for information on local TV broadcasts and tuning recommendations.

SYSTEM CONSIDERATIONS

The typical wireless system consists of a transmitter (with accompanying input device), a receiver, and associated antennas. Variables lie in the type of transmitter and receiver required to meet the application needs.

Handheld Transmitters:

The most visible transmitter is the handheld. This device resembles a standard handheld microphone with the distinction that it also contains the transmitter circuitry and antenna. Handheld transmitters employ an integrated microphone capsule that is electrically connected to the input stage of the transmitter. These capsules are usually available in a variety of polar patterns and transducer types. Typical polar pattern choices are cardioid or super-cardioid. Cardioids are most popular and are chosen for vocal performance and interview applications. They allow for good off axis pickup for working the microphone in-close and then away from the mouth, for situations where more than one voice is addressing the capsule, or where close proximity to the capsule is not possible. Super-cardioid polar patterns provide superb off-axis rejection, and are generally chosen in high gain-before-feedback situations such as high volume stage performances.

Another consideration in capsules is whether a dynamic or condenser type is appropriate. Dynamic capsules employ a moving coil within a magnetic field to produce voltage. A large percentage of microphone capsules are dynamic and are generally chosen for their high sound pressure level handling capabilities. The other choice is the condenser capsule, which requires an external voltage applied to elements within the capsule to produce voltage. Condenser capsules produce a sharper, more articulate image. Because of their accuracy in measuring and conveying changes in transient impulses, condenser capsules are often regarded as sensitive, snappy and bright. Care should be used in choosing a condenser element as they are far less forgiving of input overload compared to dynamic capsules. Handheld transmitters require some type of DC battery as a power supply, and AA or 9 volt are the general sources.

Wireless microphone introduction courtesy of Sennheiser USA.

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EQUIPMENT LEASING AVAILABLE

INTRODUCTION TO WIRELESS MICROPHONES

Bodypack Transmitters:

Bodypack transmitters are far less visible and are often chosen for that reason. Popular in video/film production, theater, and general presentation situations, bodypacks require an input source and an antenna. Bodypack transmitter systems are available in four basic varieties that are defined by the desired input source:

1, 2) Lavalier mics are the most common as they apply to general hands-free presentation, and are used for close body-miking in theatrical performance, and in video/film production. Lavaliers are available in either an omnidirectional or cardioid version.

3) Bodypack systems can also contain a headworn microphone, which is ideal for hands-free operation in vocal stage performance, athletic and aerobic instruction, and often in on-camera situations. The headworn microphone element is attached to a flexible boom that extends down the cheek and in front of the mouth, making them ideal for close miking of the mouth in active situations.

4) Instrument systems use the same bodypack transmitter as the lavalier and headworn systems, but employ a standard instrument cable with a 1/4″ plug on one end, and a transmitter input plug on the other. Very popular with guitar and bass players, this system allows the performer the mobility to roam the stage and/or venue without the fear of coming unplugged.

Bodypack transmitters require some type of DC battery as a power supply, generally AA and 9-volt batteries.

Plug-on Transmitters:



Plug-on transmitter modules convert a wired mic into a wireless one. Popular in broadcast interview and ENG applications, the plug-on transmitter module attaches directly into the XLR output jack of a low-impedance mic (usually adding 3-4″ to the overall length). They are also used in video & film production to convert the overhead boom or shotgun mic to wireless, allowing the boom operator freedom of movement. Plug-on transmitter modules are available in models with or without a 48v phantom power feed required for condenser mic powering. Models without 48v phantom power can only be used with dynamic mic elements. Note, that these plug-on modules use pin 1 of the XLR (ground) as the antenna, and the grounding network of what it is attached to becomes the antenna. Plug-on transmitters require some type of DC battery as a power supply, generally AA or 9v batteries.

Wireless mic systems use FM (frequency modulation) as the method of signal transfer. This means that the audio source signal (output of your input source) modulates a carrier signal that is transmitted through the air. It is important to note that the strength of the source signal determines how strong of a transmitted audio signal is present at the receiver. Transmitters usually have some sort of input gain setting that must be set properly for optimum system performance. Typically, this is known as input sensitivity and setting it properly involves viewing the receiver's audio meter while performing a sound check. While addressing the microphone (or other input source), adjust the transmitter input sensitivity until the receiver's audio meter displays an optimum level. This optimum level is found in the owner's manual.

Lavalier Microphones:

Choosing Between an Omnidirectional or Cardioid

Wireless lavalier systems are unquestionably the most popular for the presentation of the spoken word. These inconspicuous little microphones consistently accomplish the task of concealment quite well, however there are variables that influence how successfully they perform with a sound reinforcement system. Lavaliers are available in a variety of polar patterns, and understanding how these pickup characteristics incorporate into the environmental elements of a room will help guide you in choosing the right lavalier for your application.

OMNIDIRECTIONAL: An omnidirectional pattern microphone has a fairly unlimited field of vision. Their width of panorama is spherical allowing them equal sensitivity to sounds arriving from any direction, and their depth of field is extended due to their liberal sensitivity to sound sources. Delivering a natural openness due to the perceived low ratio of close to distant sounds, omnidirectional lavaliers have a reputation of sounding true-to-life even when attached to clothing very near the mouth.

CARDIOID: A cardioid pattern microphone has a limited field of vision. Their width of panorama is generous at the front of the diaphragm, limited at the sides, non-existent at the rear, and their depth of field is somewhat shallow and is dependent on the angular orientation of the sound source to the diaphragm. Delivering an up-close sound due to the perceived high ratio of close to distant sounds, cardioid lavaliers have a reputation of sounding dry due to the lack of ambience. Due to the nature of the pattern, there's a low frequency boost when the microphone is positioned very close to the sound source. This proximity effect is an attractive characteristic of a handheld vocal microphone but it can be problematic with lavaliers.

The next element to consider when evaluating a wireless lavalier is the nature of the performance. Wireless lavaliers afford the speaker the freedom to roam about a venue and to perform with all of the animation required. Consider the amount of gestures, head turns and overall body movements a performer makes during a performance, and how the different polar pattern characteristics will react to these changes in sound pressure. Lavaliers are generally clipped to the tie or lapel at a distance of 5 – 7 inches below the chin and aimed upward towards the mouth. The omnidirectional lavalier will respond favorably to head turns as its field of vision is quite broad, and changes in vocal amplitude are relatively unnoticeable. Conversely, head turns will be quite noticeable with cardioid lavaliers. As the position of the mouth enters and leaves the microphone's zone of highest sensitivity, the changes in vocal amplitude and tone will be pronounced. This can be distracting to the listener, especially if the presentation is being recorded either on audio or video tape. The general rule of thumb is that if a performance is for broadcast, video or audio recording, omnidirectional lavaliers will render the best results.

WIRELESS MICS

INTRODUCTION TO WIRELESS MICROPHONES

Room geometry, cubic area and loudspeaker positions are unique in any venue, and their relationship to any microphone plays a great role in overall system performance. Sound reinforcement systems are designed to lend amplification to sound sources and provide even distribution to the listeners. The amount of sound reinforcement required for even distribution is dependent on two factors; the cubic area of the room, and the positioning of the loudspeakers. Lower cubic areas require less gain, and larger cubic areas require higher gain. The objective of any sound system is to amplify microphone signals to adequate levels without feedback.

Feedback is caused when the input to a microphone is heavily weighted with the output from a loudspeaker that is amplifying that microphone's signal. The number-one cause of feedback is a speaker placement that is directed towards the microphone diaphragm. The second most common cause is poor microphone choice or placement that requires abnormally high gain for the desired sound level to be achieved. Suffice it to say that any microphone/sound system combination will feedback if the gain of the microphone is raised too high. Positioning the microphone closer to the sound source and away from the radiating pattern of a loudspeaker generally improves gain before feedback problems.

Choosing the right lavalier involves understanding the aforementioned sound reinforcement criteria, as well as the gain before feedback characteristics of the different polar patterns. Because of their limited field of vision, cardioid lavaliers have a high gain feedback characteristic and are generally chosen when high levels of sound reinforcement in a large space are required or when the speaker placement is problematic in medium and smaller spaces with high sound level requirements. Conversely, due to their unlimited field of vision, omnidirectional lavaliers have a lower gain before feedback characteristic and are generally chosen when lower levels of sound reinforcement are required, or when the performance is being recorded for broadcast or reproduction.



Receivers

The proper receiver for your system is determined by your needs. Receivers are separated into two general categories: Rack-mountable (for fixed applications where AC powering is present) or portable (battery powered for field work where AC power isn't available). These two categories can be further defined by either non-diversity, or true diversity.



The receiver's job is quite simple. It completes the wireless link by tuning into the transmitter's radio signal, processing it back into an audio signal, and outputting it for connection to a chosen destination. Determined by the application, the destination is usually a sound system mixing board, powered speaker system, field audio mixer, or the audio inputs on a video camera.

Receiver performance is critical to overall system performance. A receiver's ability to seek out and capture the transmitted signal is essential and can be adversely affected by several variables. The distance between transmitter and receiver should be logical, and mistakes are often made in system design when relying on the wireless nature of the signal to travel too far. Wireless systems have a rated effective transmission range (ETR) indicating the maximum distance a signal travels. Bear in mind that the farther a signal travels, the weaker it becomes, and while it is airborne it is volatile. The best plan is to mount your receiver antenna(s) so that the distance does not push the system's ETR limits. The ETR is also rated in direct line of sight. A radio signal weakens drastically with each wall it goes through. The type of material a wall is made of also plays a role; the harder the surface, the more dampening may occur. An ETR of 500 feet is easily reduced to 100 feet or less after it passes through 3 or 4 walls. Because of these aspects, it is advisable that you plan to locate your receiver antenna(s) within the same room as your transmitter and mount them so they are out in the open. Mounting receiver antennas so that they are enclosed inside of a rack can prove problematic and should be prevented if possible. Many manufacturers make racking accessories that provide the hardware to accomplish this important task.

A system's ETR may also be affected by the setting of the receiver's squelch control. Squelch functions much like an audio noise gate, where the audio signal path will close (mute) when certain conditions no longer exist. The squelch setting establishes the threshold of RF voltage required to open the audio signal path. When the RF signal strength weakens to the point determined by the threshold, the audio output of the receiver mutes. This prevents unwanted noise from going through your sound system. If the squelch is set too tight (a high level of voltage required to keep the audio path open), the receiver may mute the audio prematurely causing dropouts in the transmission and decreasing the ETR.

True Diversity

Diversity receivers compensate for dropouts. Just as sound reflects within an acoustical environment, radio signals behave in the same manner, bouncing from surface to surface. Direct signals from a transmitter arrive at the receiver antenna ahead (in time) of the reflected signals, and it is physically and electrically possible for the combination of these signals to occasionally be 180° out of phase in single-antenna, non-diversity receivers. When this happens a dropout occurs. It cannot be prevented and it happens all of the time.

INTRODUCTION TO WIRELESS MICROPHONES

High-quality receivers employ a "true diversity" design to compensate for the inevitable dropouts. "True diversity" receivers employ two antennas that feed two independent receiver circuits (within the unit) whose voltages are measured at all times. A comparator circuit identifies the stronger of the two incoming signals and allows it to pass through to the output stage. The shifting from antenna A signal to antenna B signal (and vice versa) happens in microseconds and there is never audible evidence that it is taking place. Care should be taken in choosing a two-antenna system because not all are "true diversity". Some models on the market display two antennas that merely sum together and feed a single receiver circuit. To insure that you are choosing the most reliable receiver, look for the term "true diversity".

Multi-system operation is common, and when using "true diversity" receivers for the best system performance, the number of antennas is twice that of the number of receivers. As discussed earlier, antennas enclosed within a rack should be avoided, as are excesses in the physical distance between transmitters and receiver antennas. Many manufacturers offer antenna splitters that accomplish several necessary tasks. Splitters are designed to deliver minimum quantities of antenna signals to maximum quantities of receivers. Common designs are 2 x 4 (two antennas feeding four receivers) and 2 x 8 (two antennas feeding eight receivers). Combined with front-mounting antenna hardware, splitters allow one pair of antennas to be exposed (best idea) and feed multiple receivers. System performance will be drastically improved by getting receiver antennas closer to the transmitter, as in an auditorium or situations where the receivers are located in a sound booth or another room. Splitters combined with antennas that are specifically designed for remote use will maximize the potential of a wireless microphone system. When using remote antennas, high-grade coaxial cable specified by the wireless manufacturer will allow greater distances to be achieved.

Many manufacturers package transmitters and receivers in kits that provide the ideal pairing of components to meet customer needs. For example, a bodypack transmitter, omnidirectional lavalier and rack-mountable receiver may be available in a kit for presentation purposes, whereas the same transmitter and lavalier paired with a portable receiver is an available kit for ENG field production. The variable is the receiver type.

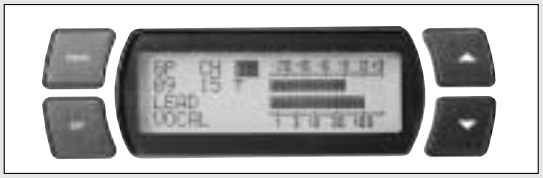
Multiple System Compatibility

The typical contemporary wireless system consists of several systems operating simultaneously. This is due in part to advancements in technology that have provided higher quality frequency agile systems, in wider tuning bandwidths, with highly selective receivers at affordable prices. Manufacturers of high-quality wireless systems should publish a maximum number of simultaneously operated systems per bandwidth spec in their specifications. This number will identify how many systems can operate together, under ideal RF conditions, by following the manufacturer's frequency-planning scheme. The compromising element here is that you must subtract frequencies from that total that fall inside of local active TV signal bandwidths. Having access to local TV activity information will aid you in choosing bandwidths with the fewest number of TV broadcast signals.

Operating multiple systems requires that precise system tunings take place. Frequency compatibility is basically a set of numbers (frequency tunings), that when used together will not produce interference amongst members of the system. When precisely calculated frequencies are not observed, and random numbers are chosen for multiple system operation, drastically unacceptable results are likely. Manufacturers of quality wireless systems publish recommended frequency selections. Many products have some or all of these numbers programmed into memory registers at the factory. It is not recommended that you get creative with multi system tunings. Refer to your manufacturer's web site or published documentation for their recommendations. Frequency agile systems with flexible tuning capabilities will render effective multiple system compatibility.

Understanding the capabilities of wireless systems is the key to met expectations, and identifying the type of system that best suits the needs of your craft will guide you in making the right decision.

Frequency Agility



Frequency agility is a term indicating that a system can be tuned to a variety of frequencies. This feature is far superior to fixed-frequency systems that could not be tuned. In today's "RF economy", where more and more TV signals take to the air, it is advisable that the user be equipped with wireless systems that can compensate for the changing RF environment. TV stations vary from city-to-city, and if your plans are to tour or use your system in a variety of cities, it is imperative to have the means to find open frequencies, and frequency agile systems are the only tools for the job.

How frequency agile is it? There are degrees of frequency agility that the specs of a wireless system should identify. The first is tuning bandwidth, or the lower and upper limits of its tuning capabilities. Tuning bandwidths are identified in overall MHz width, and should cite from and to frequencies. Obviously, the wider the bandwidth, the better, but to further quantify the frequency agility, the manufacturer should also identify how many frequencies there are at your disposal. Once again, the greater the number, the better the spec. For example, an ideal spec would be: Tuning bandwidth = 32 MHz, from 518 – 530 MHz, 1280 user selectable frequencies. Manufacturers of high-quality wireless systems will offer their products in a variety of tuning bandwidths, giving the user the opportunity to choose the most ideal bandwidth for their area. Knowing your local RF environment will help in making this an informed decision.

AKG

WMS40

UHF Wireless Mic System



The WMS40 is an entry-level wireless system for applications requiring quality audio on a limited budget. It is ideal for small stages, clubs, houses of worship and hotels. The system operates in the UHF range and uses advanced circuitry and SAW (Surface Acoustic Wave) filtering for trouble-free operation under difficult operating conditions. Up to two channels may be used simultaneously. The utter simplicity of the system and its preset operating frequencies allow the user to "plug and play" the system immediately. Beltpack and handheld transmitters feature extended bandwidth for optimum transmission of vocal and instrument signals, On/Off/Mute Switch and battery status LED.

SR40 Receiver

The SR40 receiver may be rack-mounted with the optional RMU60 kit. Indicators on the SR40 include RF status, power and mute status. A squelch control ensures noise-free reception. A front panel output level control varies the output signal from microphone to line level; both professional XLR and unbalanced 1/4″ jacks are provided.

- Adjustable squelch for noise-free reception
- Clear status display including RF, power, and mute indicators
- Output adjustable from mic to line level
- Special strain relief prevents power cable from being unplugged unintentionally
- 1/4″ and XLR-balanced output
- Easily mounts into any 19″ rack using the optional RMU60 rackmount kit

HT40 Handheld Transmitter

The HT40 handheld transmitter uses the AKG D880 microphone element to provides wide range and linear response for all applications. Delivers brilliant, uncolored vocal reproduction. Operates in a choice of frequencies: 710.400 mHz (TV channel 54), and 734.600 mHz (TV channel 58). The transmitter provides On/Mute/Off switching and battery status indication.

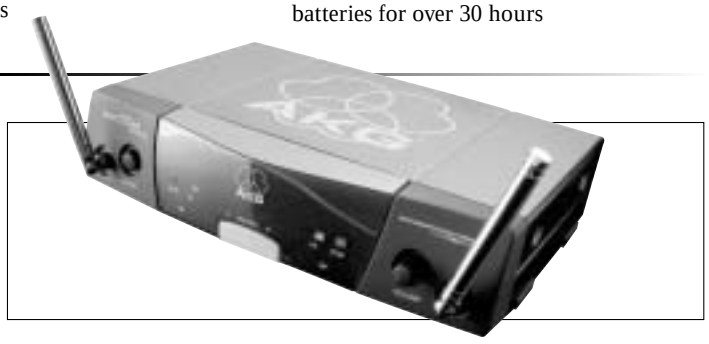
- Integrated antenna for optimized RF output and perfect transmission quality
- Controls include on/off and mute switches, and a battery status indicator
- Replaceable color code clip for channel identification
- Runs 30 hours on 1.5 V AA alkaline or rechargeable batteries

PT40 Bodypack

The PT40 is made of rugged plastic and can be used with a wide variety of mics, along with direct instrument pickup via the MK/GL instrument cable. Sophisticated audio and RF circuitry ensures that audio quality is identical to that of the equivalent hardwire mic. Same frequency assignments, operating controls and features as the HT40.

- Professional mini XLR connector allows use with all AKG "L" type microphones
- Extended audio bandwidth makes it well suited for use with guitar and bass as well
- Controls include on/off and mute switches, and a battery status indicator. Also has an input level control and mic/line switch for exceptional flexibility
- Runs on 1.5v AA alkaline or rechargeable batteries for over 30 hours

SR40/DIV Diversity Receiver



Slightly more expensive than the standard version, the SR 40 Diversity is a sophisticated receiver designed to be very rugged and more reliable in environments hostile to RF transmission to meet professional requirements.

- The SR40/DIV comes with an all-metal case, and can be installed in any 19″ rack with the optional rack kit.
- Front panel is finished in soft-touch enamel. Readouts are protected by an impact, scratch resistant clear Lexan window. Indicators include three RF and two audio level LEDs that include one peak and two diversity LEDs indicating which antenna is active.
- Rear panel provides two outputs; one balanced XLR and one unbalanced jack output. Each output can be adjusted from mic to line level with the output level control.
- A cable strain relief next to the supply cord input prevents unintentional interruptions of the supply voltage.
- Fully compatible with all current WMS 40 series transmitters.
- Available in all standard frequencies, a large front-panel color code label indicates the permanently programmed carrier frequency. The color code makes it easy to identify the various channels in a multichannel system.
- Preset squelch mutes the receiver if the received signal is too weak so the self-noise of the receiver, will not become audible when the transmitter is switched OFF.

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ONE HOUR FREE PARKING
AT 349 W. 34th STREET (with purchase of $100 or more)

WMS40 / MICROTOOLS

The ultimate modular wireless system, the WMS 40 Series and microtools make everything possible. Combining small size, superior quality, and great looks, the WMS 40 microtools make the totally wireless show a reality. Whatever the application, from vocals to instruments, they provide more flexibility and versatility than any other system.Microtools are compatible with the ultra-compact PR40 receiver as well as either SR 40 receiver and have integrated charging contacts.

- They operate on a single AAA alkaline or rechargeable battery
- Integrated charging contacts and optional charger for using rechargeable batteries
- Integrated antenna for optimum transmission
- On/Mute/Off switch and battery status indicator
- Replaceable color code element
- Input gain control for optimum modulation
- DC/DC converter for constant, transmission quality throughout battery life

MP **40** micro**pen**



Small but packed with innovative technology, the MP-40 Micropen provides the most inconspicuous and professional way to transmit a speech signal to the sound system. Just put the miniature transmitter in a convenient pocket or use the supplied cord to hang it round your neck. The mic is nested in the transmitter. You can also detach the mic from the transmitter to get a stronger signal.

- Integrated, detachable C 407 lavalier microphone with cable and attachment clip



SO **40** snap**on**

Use professional wireless technology with your existing equipment. The SO-40 Snap-on transmitter turns every dynamic microphone or mixer into a wireless one. Professional features include an input gain control for optimum matching to microphone output level and a DC/DC converter for constant, transmission quality throughout battery life. To convert your mic to wireless, simply plug the SO-40 Snapon transmitter into the XLR connector on the mic.

- Glass fiber reinforced miniature case
- XLR connector matches all dynamic mics

GB **40** guitar**bug**

The GB-40 gives you the freedom of unlimited guitar sounds without the cable mess—all you need to do is plug it into your instrument. Features include an input gain control, battery status indicator, replaceable frequency color code element, and a DC/DC converter. But the best thing about the guitarbug is its "FlexJack" swiveling swivel jack that connects the GB-40 to any guitar jack, oblique, on the top, or on the side. Mates with all guitars from the Stratocaster or Telecaster to a Les Paul.







The "FlexJack" connects the GB 40 to any guitar jack, oblique, on the top, or on the side.



PR **40** UHF Pocket Receiver



Ideal for many applications, guitarists can use the PR 40 in their pedal board in conjunction with the GB 40 guitarbug, allowing complete freedom on stage. Or, use with the MP 40 transmitter in camera mount applications for broadcast use, or with headsets and headphones in tour guide and small conference systems. A belt clip attaches the PR 40 to the user's clothes.

- Ultra durable glass-fiber plastic case plus heavy-duty swivel joints on the antennas add to the reliability of the receiver.
- Powered by 2 AAA alkaline or rechargeable batteries. Integrated charging contacts let you to charge the batteries by simply placing the receiver in the CU 40 charger. The battery compartment also accepts a special adapter for powering the receiver from a camera or DC power supply.
- Fixed-line level and adjustable headphone level outputs
- Metering includes RF, audio level and battery status LEDs
- A preset squelch mutes the receiver if the received signal is too weak so the self-noise of the receiver won't be audible when the transmitter is switched OFF.
- Supplied Velcro fastener allows mounting on a mixer, pedal board, or a video camera.

CU **40** charging **station**

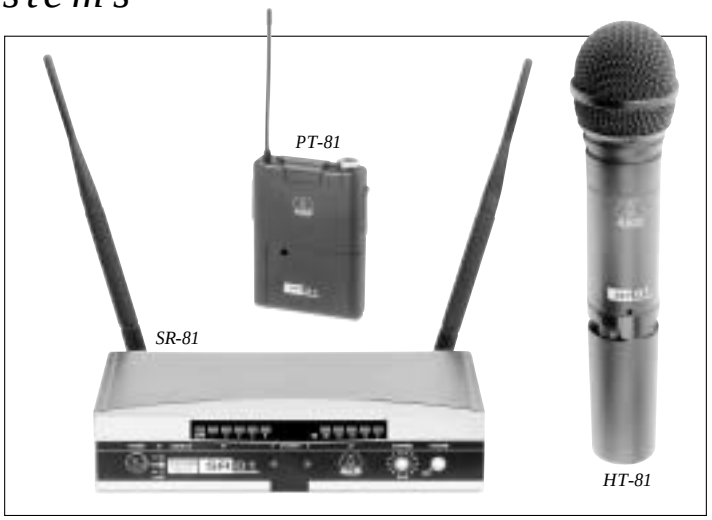


The perfect accessory for the Microtools. Simply set the PR 40, MP-40, GB-40 or SO-40 into the CU-40 Charging Station, and the internal battery is automatically and gently recharged.

AKG

WMS61/WMS81

VHF and UHF Wireless Mic Systems



With their modular system architecture, the WMS 61 (VHF) and WMS 81 (UHF) provide a custom solution for every application. They are multi-channel systems (up to 20 channels simultaneously), and they both feature SAW (Surface Acoustic Wave) filters and 15 selectable frequencies to ensure interference-free operation. An intelligent battery management system monitors the transmitter battery voltage and computes residual capacity. A pilot tone system transmits battery data to the receiver and to control the Tone Code Squelch and Auto muting functions. WMS61 and WMS81 bodypack and handheld transmitters have the same transmission characteristics (10 mW output) and via signal companding (compression and expansion), a dynamic range greater than 100 dB is attained at the receiver's output. Audio bandwidth extends from 50 Hz to 20 kHz. Through diversity reception, their performance is uniform over a wide range—up to 330´. Handheld microphone and body-pack transmitters are factory matched to a given receiver. Ideal for vocals, instruments and speech in theaters, conference centers, small sound systems, places of worship, and A/V productions.

WMS80

UHF Wireless Mic System



The WMS80 is an ideal choice for any application when performance and multi-channel operation (simultaneous use of up to 19 channels) is required at an exceptionally affordable price. All the features of the WMS81 (SAW filters and 15 selectable frequencies) except there is no intelligent battery management system to monitor voltage, no inputs for external boosters or antennas, and no pilot tone system to control the Tone Code Squelch and Automuting functions. Ideal for small and medium size stages, clubs, theaters, places of worship as well as conference centers, open-air performances and sports facilities.

SR 80 Receiver

- 15 selectable carrier frequencies make it easy to set up multichannel systems and avoid interference frequencies
- Microprocessor controlled, digital "Four Level Diversity" antenna management system and SAW (Surface Acoustic Wave) filters provide reliable and noise-free reception.
- Adjustable squelch
- Professional outputs include balanced XLR (mic/line switchable) and unbalanced 1/4″ jack
- Output level control
- Strain relief prevents power cable from being unintentionally unplugged
- Supplied with 19″ rackmount kit for easy rack mounting. (Half-rack size)
- Clear status display including RF and audio level, diversity, and mute indicators

SR 61/SR 81 Receivers

(STEP-UP FEATURES)

- Tone Code Squelch function (TCSQ) for automatic noise suppression
- Transmitter battery status indicator
- Inputs for external boosters or directional antennas for ranges longer than 100m (330´)

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WMS61/WMS81

PT 61/PT 81
Bodypack Transmitters



- Miniature bodypack transmitter with rugged plastic case and mini XLR input
- Extended audio bandwidth makes them perfect for use with guitar and bass
- They work with the C417L, C419L, C420L, C 444L, CK77L, and MKG/L instrument cable

All Transmitters Feature

- 15 selectable carrier frequencies make it easy to set up multi-channel systems and avoid interference frequencies.
- Pilot tone for battery status data transmission
- Controls include frequency selector, on/off, and mute switches, input level control, audio overload and battery status indicators
- Security covers for all controls prevent unintentional readjustment.
- They use one 1.5 V AA or rechargeable battery to keep operating costs low
- Intelligent battery management system computes transmitter battery capacity
- DC/DC converter maintains constant transmission quality throughout battery life

HT 61/HT 81
Handheld Transmitters



- Three mic elements for different applications are available
- Integrated dipole antenna for optimized RF output and perfect transmission
- Available in various colors for channel identification in multi-channel setups

PR 81
Bodypack Receiver



The PR 81 UHF camera mount receiver offers unbeatable performance for the price. This portable miniature receiver with its rugged plastic case and mini-XLR output has been primarily designed for use in video and TV camera mount applications. However, the portable receiver can also be used as a distribution system for multiple language translations, in houses of worship, conferences and guided tours.

- Miniature bodypack receiver with rugged plastic case and mini XLR output
- 15 selectable carrier frequencies make it easy to set up multi-channel systems and avoid interference frequencies
- Microprocessor controlled "Four Level Diversity" antenna management system for reliable reception with no dropouts
- Clear status display including RF and audio levels, diversity, and mute indicators
- Fixed line level and adjustable head-phones outputs on mini XLR connector
- Battery status indicator
- Volume control for headphones output
- Security covers protect the channel selector, squelch control, and on/off switch from unintentional readjustment
- Complete with versatile camera mount kit and belt clip
- An optional camera power adapter allows the receiver to be powered from the camera
- Uses AA alkaline or rechargeable batteries
- Designed for use with WMS 80/WMS 81 handheld and body-pack transmitters

PSU Central Power Supply
Switched power supply for powering four SR61/SR81 receivers or three PS61/81 antenna splitters with up to ten SR61/81 receivers

UAM1
Universal Antenna Mount
Blank panel with BNC connectors for mounting receiving antennas on the front of a 19″ rack



PSU Central Power Supply



Color Coding Kit



Universal Antenna Mount

Color Coding Kit
Identifies the individual channels of a multi-channel system by different colors. Includes 8 rotatable security covers for the HT61/81 and 8 color coded clips for the PT 61/81 bodypack transmitters and SR61/81 receivers.

WMS61/WMS81

PS 61/81 Antenna Splitter



The PS61 or PS81 is an active wide-band antenna splitter for up to four SR61/SR81 receivers or three receivers and one additional antenna splitter.

- Stackable four-way antenna splitter
- Compatible with all system components for easy configuration
- Powers remote booster antennas directly through the antenna cables to eliminate cable noise
- Direct receiver powering through antenna cables (using a PSU) saves another AC adapter
- Integrated line terminals for ease of handling
- Cable attenuation
- Status display

RA61B/RA81B External Antenna



Remote receiving antennas with integrated booster, the RA61B or RA81B can be powered form a PS61 or PS81 Antenna Splitter (respectively) through antenna cable. Includes a stand adapter.

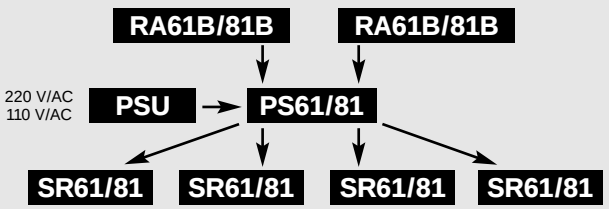
- Omnidirectional wideband antenna, compatible to all system components
- Integrated booster
- Remote powering through antenna cable
- Status indicator
- Capable of cable runs up to 330´

SRA1 External Directional Antenna

The SRA1 is a remote wideband directional antenna compatible with all UHF wireless receivers. Includes a stand adapter.

TYPICAL CONFIGURATION

Four-channel system with
antenna splitter and central power supply



SR receivers are powered through the antenna cables

WMS80

UHF Wireless Systems

WMS80/*** Handheld system with D880WL/1, D3800WL/1, or C535WL/1 head

WMS80/** Bodypack system with C417/L omnidirectional lavalier or CK55L cardioid lavalier mics

WMS80/** Bodypack system with C419/L wind/brass instrument mic, C420/L headset, C444L headset or MK/GL guitar cable

WMS61

VHF Wireless Systems

WMS61/*** Handheld system with D880WL/1, D3800WL/1, or C535WL/1 head

WMS61/** Bodypack system with C417/L omnidirectional lavalier, CK55L cardioid lavalier, CK97C/L cardioid lavalier, or CK77WR-L premium omnidirectional lavalier mics

WMS61/** Bodypack system with C419/L wind/brass instrument mic, C420/L headset, or MK/GL guitar cable

WMS81

UHF Wireless Systems

WMS81/*** Handheld system with D880WL/1, D3800WL/1, or C535WL/1 head

WMS81/** Bodypack system with C417/L omnidirectional lavalier, CK55L cardioid lavalier, CK97C/L cardioid lavalier, or CK77WR-L premium omnidirectional lavalier mics

WMS81/** Bodypack system with C419/L wind/brass instrument mic, C420/L headset, or MK/GL guitar cable

WMS61/WMS81

Combination VHF or UHF Wireless Systems

Combination systems include one handheld transmitter + mic capsule, one bodypack transmitter + lavalier mic, and one receiver + rack mount kit.

WMS/880/17** With D880WL/1 head and C417/L lavalier

WMS/880/55** With D880WL/1 head and CK55L lavalier

WMS/880/97** With D880WL/1 head and CK97C/L lavalier

WMS/880/77** With D880WL/1 head and CK77WR-L lavalier

WMS/3800/17** With D3800WL/1 head and C417/L lavalier

WMS/3800/55** With D3800WL/1 head and CK55L lavalier

WMS/3800/97** With D3800WL/1 head and CK97/L lavalier

WMS/3800/77** With D3800WL/1 head and CK77WR-L lavalier

WMS/535/17** With C535WL/1 head and C417/L lavalier

WMS/535/55** With C535WL/1 head and CK55L lavalier

WMS/535/97** With C535WL/1 head and CK97C/L lavalier

WMS/535/77** With C535WL/1 head and CK77/L lavalier

WMS80/61/81

D880



Dynamic performance mic with a frequency response specifically designed for lead and backing vocals. AKG Varimotion diaphragm for brilliant sound and frequency independent supercardioid polar response for high gain before feedback. (Not for the WMS300).

D3700



Rugged construction, attractive Tri-Power styling and superior sound for vocalists. The D3700 features an extended frequency response with a slight boost between 3kHz and 15kHz that places vocals up front. A hypercardioid pattern allows close-in usage with maximum gain before feedback right next to monitor speakers. (Not for the WM61/80/81).

D3800



Unlike other dynamic mics, the D3800 provides a unique combination of high output and an exceptionally tight hypercardioid pickup pattern for very high gain before feedback. Easily handles limit-pushing SPLs. Moving Magnet Suspension system dramatically reduces mechanical and handling noise.

C5900



Combines the detailed transparency of an AKG studio condenser mic with the road-tough construction and high gain before feedback required of a true stage performance mic. InterSpider 3-point internal shock mount effectively suppresses handling noise. (Not for the WM61/80/81).

C535



Provides the clean, open sound of AKG studio condenser mics, with a gentle 2dB rise between 7 and 12 kHz that makes voices cut through the mix.

MICS FOR ALL OCCASIONS

CK77 Lavalier ***From Opera to Movies***



The CK77 is the world's smallest twin-diaphragm lavalier. It is ultra-light (1 oz.) and only 0.56″ long. Acoustically optimized for vocal performances, the mic can be discretely attached to clothing or covered with make-up.



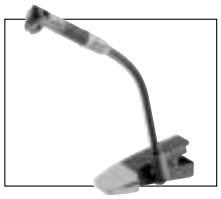
CK97 Lavalier ***From Newscasting to Talkshows***

Designed mainly for TV and radio work, CK97 provides high feedback rejection and a cardioid pattern. Very unobtrusive and easy to handle.

C417 Lavalier ***From Public Address to Conference***



For inconspicuous miking of lecturers, emcees, vocalists, etc. Fix the C417 to the lapel and hide the cable beneath the jacket.



C419 Instrument Mic

Brass and wind instruments can now go wireless. The C419 sounds natural without any loss of quality in the instrument's acoustics. Attaching the mic is quick and easy. Just clamp the C419 on the bell, aim at the rim, and find the optimum position.

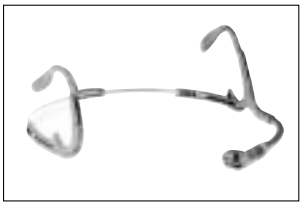


C420 in use



C419 in use

C420 Headset Microphone



The C420 is a condenser design with a full bandwidth 20 - 20 kHz frequency response. It captures the voice with a detailed, brilliant sound quality, providing the performer with a pure, powerful sound that carries the punch and presence to cut through to the audience.

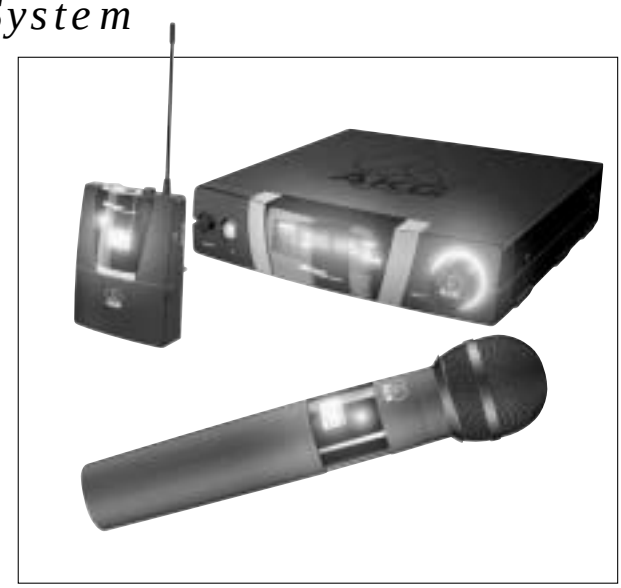
MK/GL Guitar Cable

Eliminate endless instrument cable connections. This 32″ cable with an WMS61/80/81 system offers total mobility. Just plug it and play, and get great sound.

AKG

WMS-4000

UHF True Diversity Wireless Mic System



The WMS 4000 is one of the most advanced and innovative wireless systems for professional use available today. It offers an inspiring combination of sophisticated engineering, exceptional flexibility, surprising versatility, and maximum ease of use. The application range of the WMS 4000 is almost unlimited, from fixed installations or mobile systems in architecturally difficult environments to motion picture and broadcast work. In large-scale projects including worship center sound systems, theatrical productions, or stadium concerts, the WMS 4000 shows off its professional qualities.

The advanced backlit display on the transmitters as well as on the receiver indicates the selected Preset, remaining transmitter battery life, gain setting, or intermodulation-free frequencies. Transmitters provide a silent mode and a "hidden" pilot tone system for status monitoring. An Environment Scan function on the SR 4000 receiver finds interference frequencies within the RF environment. The Rehearsal function lets you track important system parameters during the soundcheck to avoid potential problems before the show. The CU/BP 4000 battery powering system is controlled by the SBMS Smart Battery Management System. A unique innovation, it permanently monitors battery status and prevents overcharging, while Charge Balance Management provides maximum safety and reliability. The antennas available for the WMS 4000 use future oriented technologies, too. Choose from directional and omnidirectional antennas for every application.

SR 4000 True-Diversity Frequency-Agile UHF Receiver



The SR 4000 is a state-of-the-art, true diversity receiver that provides maximum ease of use and the most useful functions in its class. Many features of the SR 4000 were never before available or limited to extremely expensive systems. The accurate battery life readout is an invaluable advantage in live sound applications. The transmitter uses a pilot tone to transmit battery status data to the receiver that displays the remaining battery capacity in hours. The pilot tone decoder also detects and displays other important information including the current position of the MUTE switch on the transmitter. The integrated software not only permits automatic setup and frequency scanning but remote control and monitoring from a PC via a dedicated interface as well. Optional MCS 4000 Mission Control Software makes setting up and monitoring highly complex systems incredibly easy. A logic output allows remote control of specific functions on an external device, e.g., an automatic microphone mixer. Naturally, the SR 4000 operates in an extremely wide UHF band (30 MHz) and each frequency preset provides up to 24 intermodulation-free subchannels. Housed in a half-rack, all-metal case, the SR 4000 is the most compact, reliable, and powerful UHF receiver in its price bracket.

- Operates on a 30 MHz-wide UHF wide bandwidth containing preset inter-modulation-free frequencies and access to up to 1,200 switchable frequencies. Three separate 30 MHz channels may be selected for use in the U.S., offering a maximum of approximately 50 usable channels that are limited only to other external RF interference.
- Unique signal compression and expansion circuits provide a dynamic range greater than 112 dB.
- An output level switch allows the selection of three output levels at -30, 0, +6 dB for optimum system gain structure.
- Internal software features AutoSetup for inter-modulation free channel selection, Environment Scan for scanning of the unit's RF range, and Rehearsal Mode selection for saving system data during rehearsal set-up. Settings like squelch threshold, carrier frequency selection and user's name can be edited and stored in an easy menu structure.
- Programmable status indicators on the SR 4000 receiver include mute, audio level bar graph, RF level bar graph, and a 6 x 14 segment alphanumeric display and menu control. The unique, programmable Status Control Ring can display the most important system data at a glance.
- The large, backlit color display and a jog wheel make the SR 4000 easy to use and a programmable warning light indicates selectable critical conditions within the system.

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EQUIPMENT LEASING AVAILABLE

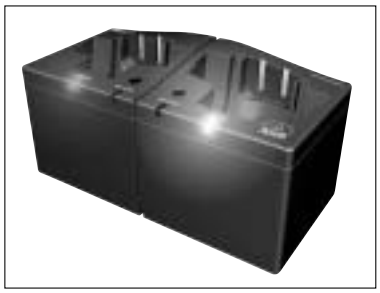
WMS-4000

PT 4000 Bodypack and HT 4000 Handheld UHF Transmitters

The PT 4000 bodypack and the HT 4000 handheld UHF transmitters incorporate programmed, pre-matched, optimized frequency groups that allow for quick, easy frequency selection. Other features include backlit LCD displays for system status monitoring and a jog wheel for quick system setup and navigation. These transmitters are encased in magnesium metal chassis for rugged on-stage use, and have a usable operation distance of up to 330 ft., with a system audio bandwidth from 35 Hz to 20 kHz.



- The HT 4000 handheld transmitter allows for interchangeable microphone modules with three dynamic models D880, D 3700, D 3800 and three condenser models C 900, C 5900, and the C 535.
- The PT 4000 bodypack accepts both microphone and line-level input signals and has an adjustable input sensitivity. A lockable TA-3F Mini-XLR connector is used for easy connection of a broad range of AKG headsets, lavaliers, instrument microphones and instrument cables. The PT 4000 bodypack transmitter also has a 2.5-mm jack connector for the optional RMS 4000 remote switch device.
- Both HT 4000 handheld and PT 4000 bodypack transmitters can be powered either by 2 AA batteries or a BP 4000 rechargeable battery pack and incorporate integrated charging contacts for easy use with the optional CU 4000 charging unit.



CU 4000 Charger

The CU 4000 charging unit features two charging bays that are designed to hold two separate HT 4000 handheld, PT 4000 transmitters or BP 4000 battery packs. The CU 4000 can charge two battery packs simultaneously in one hour. A three-step LED display on the CU 4000 displays the charging status. The unit is based on a trickle charge system with inflection point switch-off automation to preventing the battery pack from overcharging. The CU 4000 identifies possible battery memory problems and suggests starting an optional recovery cycle via a recovery button on the CU 4000's front panel. This function refreshes the BP 4000 battery pack by charging and discharging cycles to determine the actual capacity of the battery.

- The CU 4000 has a lockable DC connector on its rear panel that allows it to be powered by either an external DC adapter or by the optional PSU 4000 central power supply.

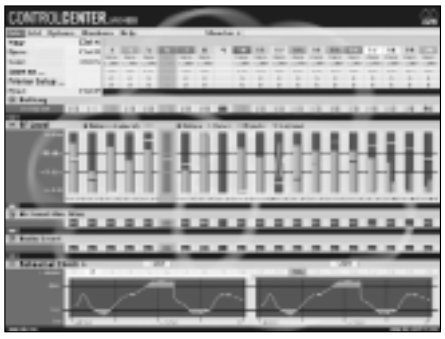
PS 4000 Multi-Coupler



The PS 4000 multi-coupler provides an antenna signal and DC power for up to our diversity wireless receivers. It has a wide-band antenna signal amplifier/divider with a 220 MHz-wide UHF bandwidth custom-tuned to all available WMS 4000 channels. Two additional BNC outputs are also available on the rear panel for cascading the diversity antenna signal to another multi-coupler. Up to three multi-couplers can be cascaded to supply antenna signal to 12 receivers from a single pair of antennas.

- The PS 4000 has several operation and failure status LED's that allow monitoring of correct cable and antenna signal operation. A 10-position switch on the front panel of the PS 4000 will adjust for signal loss caused by cable length, thus providing reception optimization.
- Either a single DC adapter or an ASU 4000 central power supply unit can power the PS 4000. The PS 4000 is encased in all-metal chassis and is one standard rack unit high and one-half rack unit wide. When mounted it can be placed side-by-side with an SR 4000 receiver, another PS 4000 multi-coupler, an ASU 4000 and other WMS 4000 rack-mountable accessories.
- The PS 4000 uses three different external antenna options, SRA1 passive directional, SRA2 active directional and the RA 4000B active omnidirectional. When cable length between external antennas and PS 4000 exceed the maximum compensation setting an AB 4000 in line amplifier can be added to gain an additional 17 dB of amplification.

Additional Accessories:



An expansive line of additional accessories include the ASU 4000, an in-line power supply unit for use in large antenna networks, HPA 4000 Headphone Amplifier with eight line inputs, HUB 4000 Network Interface (designed for multi-complex systems) that features an ID switch, status display and 8 x SR in, an Ethernet out and the MCS 4000 PC Software that provides additional programming and monitoring of connected receivers, detailed RF and AF functions, a graphical scanner and enhanced auto setup mode.

FREEWAY WIRELESS SYSTEMS



200 Series (VHF)

With the Freeway 200 Series, Audio Technica again sets the standard for a comprehensive, cost-effective and dependable VHF wireless miking solution. Designed for musicians, live performance, houses of worship, PA and aerobic instruction, the 200 Series is a single-channel, fixed-frequency systems available in three VHF frequencies (169.505, 170.245, and 171.905 MHz) and in multiple configurations.

600 Series (UHF)

An exciting range of frequency-agile UHF wireless, the Freeway 600 Series provides interference-free performance and exceptional audio quality in a wide variety of applications such as live performance, houses of worship, PA and aerobic instruction. Available in multiple configurations, the Series 600 features 10 selectable UHF channels, a removable docking power supply, and an advanced dipole antenna design. Two independent frequency bands allow operation for up to eight simultaneous channels: Band A: 470-480 MHz (TV channels 14-15) and Band B: 482-492 MHz (TV channels 16-17).



ATW-R200 (VHF) and ATW-R600 (UHF) Receivers

The ATW-R200 (non-diversity, single-channel receiver) and ATW-R600 (10-channel frequency-agile receiver) both feature an advanced dipole antenna design (improves operation by providing adjustable ground element in addition to the usual "signal" element) and a unique removable docking power supply that slides into the receiver. For further flexibility, the power adapter can easily be removed from the receiver case and be used as a conventional in-line AC adapter. They also offer squelch and volume controls along with AF Peak, RF and power indicator lights.

- Can also be operated from external 12V DC source for portable applications.
- Front panel channel selector switch (ATW-R600 only) and output level control
- High impact molded case with storage for antennas and docking power supply
- 1/4″ output jack

ATW-201/ATW-601 Bodypack Transmitters

Bodypack's professional 4-pin locking HRS-type connector provides solid secure connections and works with a wide variety of mics and cables. Dual-impedance input for instruments and mics (high and low impedance audio connections) plus a bias voltage to allow use with dynamic and condenser mics as well as Hi-Z instrument pickups.

Bodypack and Handheld Transmitters Both Feature

Both transmitters operate on a single 9-volt battery with a battery-condition indicator light, easily-accessible, recessed three position Off/Standby/On switch, internal channel selector (600 Series only) and a variable trim control to provide for maximum RF modulation with minimum distortion.

ATW-202/ATW-602 Handheld Mic/Transmitter

Handheld transmitter has rugged unidirectional dynamic element with internal shock mount to minimize handling noise.

ATW-201/G Guitar System .....**Call**
ATW-201/H Headworn Mic System.....**Call**
ATW-201/L Lavalier Mic System.....**Call**
ATW-201/H Handheld Mic System.....**Call**

ATW-601/G Guitar System .....**Call**
ATW-601/H Headworn Mic System.....**Call**
ATW-601/L Lavalier Mic System.....**Call**
ATW-601/H Handheld Mic System.....**Call**

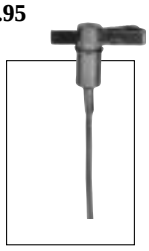
UNIPAK BODY-PACK TRANSMITTERS

Wireless Essentials for A-T UniPak Body-Pack Transmitters

Wireless Essentials are a collection of microphones and cable assemblies pre-terminated with a HRS-type type connector for use with Audio-Technica wireless bodypacks. Included are omnidirectional and cardioid lavalier mics, headworn mics, clip-on instrument mics, and instrument cables along with a boundary mic and a podium gooseneck mic for wireless installations. By providing this wide variety of choices, Audio Technica allows you to "personalize" your wireless system for a specific application. Plus, you can interchange the Essential without having to purchase another wireless system or transmitter. Essentials are properly terminated so that they connect to the appropriate input on the transmitter. This enables the Essential to take advantage of the individual transmitter gain control so that plugging in an instrument cable will not affect the mic trim level and vice versa. The result is maximum flexibility.

AT829cW
Miniature cardioid condenser mic 29.5″ cable terminated with locking 4-pin connector.....**49.95**

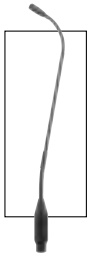
AT831cW
Miniature cardioid condenser mic; 29.5″ cable terminated with locking 4-pin connector. Includes clothing clip and windscreen.....**79.95**

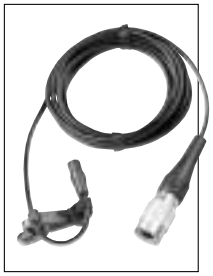




AT851cW
Unidirectional condenser boundary mic; 29.5″ cable terminated with locking 4-pin connector **119.95**

AT857AMLcW
Cardioid condenser 19″ gooseneck mic; mounts to 5/8″-27 thread; 29.5″ cable terminated with locking 4-pin connector Includes windscreen **174.95**





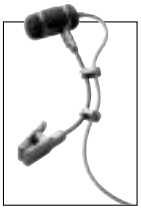
AT889cW
Headworn noise-canceling condenser mic; 55″ cable terminated with locking 4-pin connector. Rugged, moisture-resistant construction designed for aerobics instruction and other demanding applications. Includes windscreen and cable clip.....**159.95**

AT899cW
Subminiature omnidirectional condenser mic; 55″ cable terminated with locking 4-pin connector. Includes AT899AK accessory kit **159.95**

AT899cW-TH
"Theater" model, same as AT899cW except beige color mic and cable for concealment. Includes AT899AK-TH (beige) accessory kit....**159.95**



AT-GCW
Hi-Z instrument/guitar cable with 1/4″ phone plug, terminated for use with UniPak body-pack transmitters **23.95**



ATM35cW
High-intensity cardioid condenser mic; 55″ cable terminated with locking 4-pin connector. Includes AT8418 instrument mount for brass, reeds, drums, etc **114.95**

ATM73cW
Headworn cardioid condenser mic; 55″ cable terminated with locking 4-pin connector. Includes windscreen **119.95**





ATM75cW
Headworn cardioid condenser mic; 55″ cable terminated with locking 4-pin connector. Includes windscreen and cable clip **79.95**



MT830cW
Subminiature omnidirectional condenser mic; 55″ cable terminates with locking 4-pin connector. Includes clothing clip and windscreen.....**74.95**

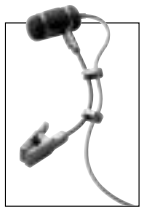
MT830cW-TH
"Theater" model, same as MT830cW except beige color mic and cable for concealment.....**74.95**



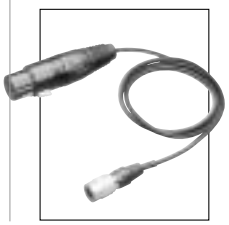


PRO 8HEcW
Headworn hypercardioid dynamic mic; 55″ cable terminates with locking 4-pin connector. Includes windscreen and cable clip.....**79.95**

PRO 35xcW
Cardioid condenser mic; 55″ cable terminates with locking 4-pin connector. Includes AT8418 instrument mount for brass, reeds, drums, etc **99.95**



XLRW



Input cable for UniPak body-pack transmitters with XLRF to 4-pin connectors. For Lo-Z mics with XLRM-type output terminations...**23.95**

AUDIO TECHNICA

1100/1200 SERIES

True Diversity VHF Systems

Ideal for schools, garage bands and small clubs, the 1100 and 1200 series are affordable crystal-controlled, true-diversity wireless mic systems. Their receivers are equipped with two antennas that feed two completely independent RF sections on the same frequency, while automatic logic circuitry continuously compares and selects the superior signal, providing better sound quality and reducing interference and dropouts. The versatile UniPak body-pack transmitter has both low- and high-impedance inputs plus a bias connection, for use with dynamic and condenser microphones, as well as Hi-Z instrument pickups. Both the handheld and UniPak transmitters use 9-volt batteries and have Off/Standby/On switches, battery condition indicators, and battery-save switches. The 1100 Series receiver is made to be mounted in a standard 19˝ rack (1U), while two 1200 Series receivers can be mounted side-by-side in a single 19˝ rack space.

System Features

- VHF Band operation (169.505 - 213.800 MHz TV channel 7-13) crystal controlled for interference resistant operation
- Robust VHF operation minimizes dropouts caused by spurious reflections and is less dependent on “line-of-sight” operation
- True Diversity reception with two independent RF sections in the receiver automatically selects the better signal for highest reliability
- Full FM deviation improves signal/noise ratio for greater headroom and dynamic range
- Choice of 20 VHF frequencies available for minimum interference, including US and Canadian traveling frequencies

Transmitters

- Battery condition indicators alert you when battery needs replacing
- Locking HRS-type type connector on the bodypack provides a secure connection and works with a wide variety of mics and cables
- Bodypack features both high and low impedance audio connections plus a bias voltage to allow use with dynamic and electret condenser microphones as well as Hi-Z instrument pickups
- Handheld Dynamic Transmitter incorporates Hi-Energy element and internal shock mounting to minimize handling noise
- Dual power output switches extend battery life and minimize receiver input stage overload when transmitters are used next to the receiver
- They operate on a single 9V Battery with 15-20 hours of battery life

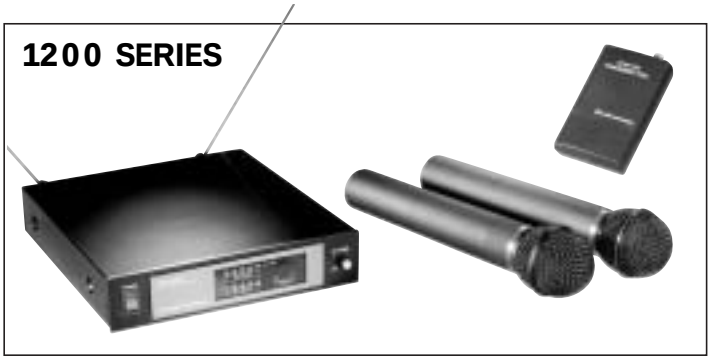
Receiver

- Heavy-duty metal construction reduces interference and shields receiver from other components
- Mounts in a single 19˝ rack space (with included rack mount ears) for fixed installations
- Front panel LEDs show power, AF peak and diversity tuner operation
- Balanced (with ground lift switch) and unbalanced audio outputs simplify use with a variety of mixers, amplifiers, and sound systems
- Removable rear mounted antennas enable the receivers to be used in a rack with remote antennas and antenna distribution accessories



ATW-1127 Bodypack System
with ATW-T27 UniPak transmitter.....**274.95**

ATW-1128 Handheld System
with ATW-T28 handheld unidirectional dynamic mic/transmitter.....**329.95**



ATW-1235 Body-pack System
with ATW-T35 UniPak transmitter.....**329.95**

ATW-1236 Handheld Microphone System
with ATW-T36HE handheld hypercardioid dynamic mic/transmitter...**299.95**

ATW-1237 Handheld System
with ATW-T37C handheld hypercardioid condenser microphone/transmitter (accepts interchangeable elements for a variety of polar patterns).....**499.95**

1400 SERIES

True Diversity UHF Systems

The 1400 Series is very similar to the 1200 Series except it works in the UHF bandwidth (728.125 - 740.500 TV channel 57 -59) crystal controlled for interference resistant operation.



Same features as the 1200 Series PLUS—

- UHF Band operation (728.125 - 740.500 TV channel 57 - 59) crystal controlled for interference resistant operation
- Factory pre-selected frequencies enable up to 10 systems to operate simultaneously
- Available in single transmitter (bodypack or handheld) or dual transmitter "combo" configurations depending on your application
- Removable front mounted antennas enable the receiver to be used in a rack for portable applications
- Adjustable squelch control allows system to be optimized for specific use location to minimize interference
- Locking HRS-type type connector on Bodypack provides solid secure connections and works with a wide variety of mics and cables

ATW-1451 UniPak Body-pack System
ATW-R14 receiver and ATW-T51 UniPak transmitter (with Super Hi-Z input for optimum load for guitar pickups)

ATW-1452 Handheld Microphone System
ATW-R14 receiver and ATW-T52 handheld unidirectional dynamic mic/transmitter (with Hi-ENERGY dynamic element)

ATW-1453 Dual Transmitter System
ATW-R14 diversity receiver, ATW-T52 handheld and ATW-T51 UniPak transmitters, and AT831cW lavalier microphone



7000 SERIES

Frequency Agile True Diversity UHF Systems

The 7000 Series features UHF operation with two separate bands (656.125 - 668.500 MHz (TV channel 45-47) and 728.125 - 740.500 (TV channel 57 - 59) for maximum flexibility and to increase the number of simultaneous operating systems. In addition, 100 phase locked loop (PLL) synthesized channels per band with selectable switches on receiver and transmitter provide ease of setup.

Same features as the 1400 Series PLUS—

- Available in two frequency ranges:
  656 - 668 MHz (TV channels 45 - 47) and
  728 - 740 MHz (TV channels 57 - 59)
- 100 PLL-synthesized channels selectable via switches on receiver and transmitter
- Removable rear mounted antennas enable the receiver to be used in a rack with remote antennas and antenna distribution accessories
- Condenser handheld transmitter with the legendary AT4033 studio condenser element for discriminating applications
- AC power uses standard line cord—no wall warts and line lumps

ATW-7375 UniPak Body-Pack System
ATW-R73 receiver and ATW-T75 UniPak transmitter (includes Super Hi-Z input that provides optimum load for guitar pickups)

ATW-7373 Handheld Microphone System
ATW-R73 receiver and ATW-T73 handheld unidirectional condenser mic/transmitter (featuring the same condenser element used in the AT4033 studio mic)

ATW-7376 Handheld Microphone System
ATW-R73 receiver and ATW-T76 handheld unidirectional dynamic mic/transmitter (featuring a Hi-ENERGY dynamic element with superior internal shock mounting

AUDIO TECHNICA

3000 SERIES

200-Channel Frequency-Agile UHF True Diversity System

Designed for a wide range of applications including live performance, regional touring companies, fixed installation, public address and places of worship, the 3000 Series operates in the UHF 600 MHz frequency band (655-680 MHz, TV channels 44-49) with 200 selectable frequencies and true diversity wireless technology. All components have soft-touch controls for quick, easy access to the most important functions during set-up and operation. An easy-to-read LCD information display in each unit provides convenient visual indications of settings and operation status.



FEATURES

ATW-R310
True Diversity Receiver

The ATW-R310 offers 200 PLL-synthesized UHF frequencies, true diversity operation, advanced Digital Tone Lock Squelch, soft-touch controls with LCD information display and step-through menus to minimize set-up time

- Adjustable-slope high-pass filter (Off, 6dB, 12dB, or 18dB)
- Meter hold setting function permits the AF bar graph in the LCD window to capture and display the highest level audio modulation received from the transmitter - very useful during setup or sound check.
- Transmitter battery-life fuel gauge
- XLR-balanced and 1/4˝unbalanced outputs
- All-metal housing for superior shielding, and a ground-lift switch
- Includes two flexible UHF antennas and rack-mount adapters
- Two receivers can be mounted side-by-side in a single 19˝ rack space

ATW-T310
UniPak Transmitter

- The versatile ATW-T310 offers a multi-function LCD display and soft-touch controls for easy set-up and operation.
- Low- and high-impedance inputs (with a bias connection) permit use with dynamic and condenser mics as well as Hi-Z instrument pickups.
- In addition to its programmable functions, the ATW-T310 has a three-position sliding cover to limit access to controls as appropriate for the application and user.

ATW-T341
Handheld Mic/Transmitter

- Using the same cardioid dynamic capsule (specifically tailored for a more aggressive sound quality that keeps the vocals up-front in the mix) as ATs acclaimed Artist Elite AE4100 microphone, the ATW-T341 provides a focused vocal pickup while offering superior gain-before-feedback.
- Features rugged metal construction, soft-touch controls and a multi-function LCD display.

Both transmitters have dual-power setting: select high for maximum range or low for extended battery life and are powered by two 1.5V AA batteries for 6-10 hours of operation.

3000 Series Systems

ATW-3110 UniPak Body-Pack System: Includes the ATW-R310D receiver and ATW-T310D UniPak transmitter (655 - 680 MHz range). Requires a Wireless Essentials microphone or cable to complete the system.

ATW-3141D Handheld Microphone System: Includes the ATW-R310D receiver and ATW-T341D handheld cardioid dynamic microphone/transmitter (655 - 680 MHz range)

3000 SERIES SPECIFICATIONS

Operating Frequency	UHF band “D” system: 655.500 MHz to 680.375 MHz	Normal Deviation	±10 kHz
Number of Channels	200 total	Operating Range	300´ typical
Frequency Stability	±0.005%, Phase Lock Loop frequency control	Operating Temperature Range	41° F (5° C) to 113° F (45° C)
Modulation Mode	FM	Frequency Response	70 Hz to 15 kHz

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ONE HOUR FREE PARKING
AT 349 W. 34th STREET (with purchase of $100 or more)

4000/5000 SERIES

Frequency-Agile True Diversity UHF System

Building on the 3000 Series, the 4000 and 5000 Series wireless systems can suit the most demanding performance applications. Multiple receivers of both series can be linked, and both systems feature IntelliScan frequency selection and dual companion. All components can store up to five preset configurations, with customized "names" if desired. The heart of the 4000 and 5000 Series is their true diversity frequency-agile receivers. The half-rack AEW-R4100 and full-rack, dual-receiver AEW-R5200 offer two hundred selectable UHF channels in the range of 655.500 to 680.375 MHz (TV channels 44-49), with an operational range in excess of 300 feet.



- Receivers may be linked to form a complete system in which the built-in IntelliScan capability can automatically determine and set the best available frequencies on all linked receivers, minimizing interference and intermodulation. An AC pass-through system allows multiple receivers to be powered using only one outlet.
- Dual-band compander circuitry compresses and expands low- and high-frequency audio signals separately, minimizing artifacts such as pumping and breathing and maximizing audio quality.
- Digital Tone Lock squelch system ignores RF that does not carry the wanted signal's unique signature. Concurrently, the digitally encoded tone also communicates transmitter data for receiver display
- External mute switch jack to mute the system quickly and easily.
- The AEW-R5200 also includes Ethernet ports, enabling monitoring of system parameters via the supplied Mac or Windows software.
- To complement both the 4000 and 5000 Series receivers, five wireless transmitter options are available. All feature rugged, ergonomic metal bodies, programmable on/off/mute switching, and 10 mW/35 mW switchable RF power, and are compatible with both systems. Five user-programmable settings, plus a default setting, make these transmitters extremely versatile.
- The AEW-T1000 UniPak transmitter offers advanced features such as a recessed dual-impedance 4-pin locking connector, durable metal body, soft-touch controls with sliding cover, and an input level that is adjustable in 2 dB increments.
- Developed from AT's Artist Elite line of wired mics, the handheld transmitters are specially optimized for wireless operation. Options include two condenser and two dynamic types:
- The AEW-T5400 and AEW-T3300 cardioid condenser mics boast superior clarity and realism, very low handling noise, and sturdy, die-cast construction. The cardioid AEW-T4100 and hypercardioid AEW-T6100 dynamic mics offer assertive sound, outstanding pop protection, and rugged construction.

4000 Series System Configurations

UniPak Transmitter System (Including AEW-R4100 Receiver)

AEW-4110: With AEW-T1000 UniPak transmitter ..... **Call**

4000 Series Handheld Systems (Including AEW-R4100 Receiver)

AEW-4230: With AEW-T3300 cardioid condenser mic/transmitter..... **Call**
AEW-4240: With AEW-T4100 cardioid dynamic mic/transmitter ..... **Call**
AEW-4250: With AEW-T5400 cardioid condenser mic/transmitter ..... **Call**
AEW-4260: With AEW-T6100 hypercardioid dynamic mic/transmitter..... **Call**

4000 Series Dual Transmitter Systems (Including AEW-R4100 Receiver)

AEW-4313: AEW-T1000 and AEW-T3300 ..... **Call**
AEW-4314: AEW-T1000 and AEW-T4100 ..... **Call**
AEW-4315: AEW-T1000 and AEW-T5400 ..... **Call**
AEW-4316: AEW-T1000 and AEW-T6100 ..... **Call**

5000 Series System Configurations

Dual UniPak Transmitter System (Including AEW-R5200 Dual Receiver)

AEW-5110: With two AEW-T1000 UniPak transmitters ..... **Call**

Dual Handheld Systems (Including AEW-R5200 Dual Receiver)

AEW-5230: 2 AEW-T3300 cardioid condenser mic/transmitter ..... **Call**
AEW-5240: 2 AEW-T3300 cardioid condenser mic/transmitter ..... **Call**
AEW-5250: 2 AEW-T3300 cardioid condenser mic/transmitter ..... **Call**
AEW-5260: 2 AEW-T3300 cardioid condenser mic/transmitter ..... **Call**

5000 Series Dual Transmitter Systems (Including AEW-R5200 Receiver)

AEW-5313: AEW-T1000 and AEW-T3300 ..... **Call**
AEW-5314: AEW-T1000 and AEW-T4100 ..... **Call**
AEW-5315: AEW-T1000 and AEW-T5400 ..... **Call**
AEW-5316: AEW-T1000 and AEW-T6100 ..... **Call**

U100 SERIES

UHF Multi-Channel Wireless Mic System

UHF camera mountable with a rock-solid RF design and rugged metal construction, the U100 system is available with either a body-pack or plug-on transmitter. Each system features the ATW-R100 true diversity receiver and operates in the channel 57 to 59 TV range avoiding upcoming DTV stations in the channel 67 to 69 range. The ATW-U101 system includes the ATW-T101 body-pack transmitter with standard Switchcraft TA5F input connector that is compatible with several lavalier and headworn mics. The ATW-U102 system includes the ATW-T102 XLR plug-on transmitter compatible with all dynamic and most self powered microphones.



System Features:

- 100 Phase Lock Looped (PLL) -synthesized channels selectable via switches on receiver and transmitter
- 300′ operating range typical
- Frequency-agility allows operation in both analog and DTV operating environments
- 100 Hz to 15 kHz frequency response
- Operates in the less crowded 728.125 - 740.500 MHz UHF frequency band or TV channels 57-59
- Hinged battery doors on receiver and transmitters

ATW-R100 Receiver

- True diversity operation automatically selects the better signal from two independent receivers, reducing dropouts
- LEDs indicate A or B antenna status, AF peak, and power status
- XLR balanced output with level control
- Removable antennas (BNC) connectors
- Operates on two internal 9 v batteries (for approx 6 hours) or on external 12 v DC
- Headphone jack with volume control
- Helical filter design improves rejection of unwanted signals
- Measures 3.35 x 4.53 x 1.42″ (WHD) and weighs 12.0 oz.

ATW-T101 Beltpack Transmitter

- Features industry-standard Switchcraft 5-pin mic input connector
- Runs on a single 9V battery with approximately 8-10 hours battery life
- Measures 2.52 x 3.78 x 0.91″ (WHD) and weighs 5 oz.

ATW-T102 Plug-on Transmitter

- 3-pin XLRF-type input connector compatible with all dynamic and most self powered mics
- Runs on a single 9 volt battery with approximately 7-9 hours battery life
- Measure 1.57 x 4.32 x 1.57″ (WHD) and weighs 6.2 oz.

U100 Wireless Systems

ATWU101
Includes ATW101 beltpack transmitter and ATW-R100 receiver (no mic)

ATWU101830
Includes ATW101 transmitter, AT831cT5 lavalier mic and ATW-R100 receiver

ATWU101831
Includes ATW101 transmitter, MT830cT5 lavalier mic and ATW-R100 receiver

ATWU102
Includes ATW102 plug-on transmitter and ATW-R100 receiver

U100 Components

ATWT101 Beltpack transmitter

ATWT102 Plug-on transmitter

ATW-R100 True Diversity receiver

AT831cT5
Miniature cardioid lavalier mic

MT830cT5
Subminiature omni-directional condenser lavalier mic

MT830cT5TH
“Theater” model (same as above except in beige)

UHF COMPARISON CHART

	Receiver	AE 5000 Series AEW-R5200	AE 4000 Series AEW-R4100	3000 Series ATW-R310	1400 Series ATW-R14	Freeway 600 ATW-R600	U100 Series ATW-R100
receiver	Fixed Frequency / Agile	Agile	Agile	Agile	Fixed	Agile	Agile
	Band Frequencies	TV 44 - 49	TV 44 - 49	TV 44 - 49	TV 57 - 59	TV 16 - 17, TV 14 - 15	TV 57 - 59, TV 45 - 47
	Number Channels / System	200	200	200	1	10	100
	Number of Simultaneous Channels	20 per band	20 per band	20 per band	10	8 (using both bands)	18
	Intelliscan (Automatic System Channel Plan)	■	■				
	True Diversity with Indicators (Tuner LEDs)	■	■	■	■		■
	RF Level	Bargraph	Bargraph	Bargraph	Bargraph	Peak LED	Peak LED
	TX Mute Indicator	■	■	■			
	AF Level	Bargraph	Bargraph	Bargraph	Bargraph	Peak LED	Peak LED
	Digital Tone Lock Squelch	■	■	■			
	Adjustable Squelch	■	■	■	■	■	
	TX Battery Status & System Alert Indicators	■	■	■			
	Channel Selector Switch					■	■
	Channel Selector (Soft Touch Control)	■	■	■			
	Output	XLR, 1/4″ TRS	XLR, 1/4″ TS	XLR, 1/4″ TS	XLR, 1/4″ TS	1/4″ TS	XLR
	Ground Lift Switch	■	■	■	■		
	External Mute Contact Closure	■					
	Power Supply Type	IEC	IEC	12v DC In-Line	12v DC WW	Docking IEC	2 x 9v
	Headphone Monitor Output	■	■	■			■
	Rack Mount	■	■	■	■	Optional	
	Half-rack Case		■	■	■		
	Metal Case Construction	■	■	■	■		■
	Antenna Location	Rear/Front	Rear/Front *	Rear	Front	Front	Top
	BNC Antenna Connections	■	■	■	■		■
	Antenna Power (12V DC)	■	■				
	Optional Antenna Splitter (DA)	■	■	■	■		■
	Optional Ground Plane & Directional Antennas	■	■	■	■		■
	Optional Powered Antennas	■	■				
transmitter	Battery Status LED	Bargraph	Bargraph	Bargraph	LED	LED	LED
	Batteries	2 x 1.5v	2 x 1.5v	2 x 1.5v	1 x 9v	1 x 9v	1 x 9v
	Frequency Display/Mute Indication	■	■	■			
	Configurable/Defeatable Power/Mute Switch	■	■	■			
	Channel Selector Switch					■	■
	Soft Touch Configuration	■	■	■			
	Control Cover (Bodypack Transmitter)	■	■	■			
	HRS-type 4-pin Connector (Bodypack)	■	■	■	■	■	TB5M
	Interchangeable Mic Elements (Handheld)	■	■				
	Metal Case Construction (Handheld/Bodypack)	■	■	Handheld Only			
	Digital Tone Lock Squelch	■	■	■			
	Plug-on Transmitter						■
	** Needs to be done at factory*						

AZDEN

PRO SERIES

VHF Wireless Mic Systems

The Azden Pro Series brings you high performance VHF wireless at a price you can afford. Built around a two-channel design that allows selection of the frequency that provides the cleanest signal — for reliable RF performance up to 250 feet away. They all feature two switchable frequencies — 169.445 and 170.245 MHz for a range of 250 feet, high performance VHF transmitters and receivers incorporating SMD (surface mount device) technology. The compact receivers connect to the camera's external mic jack with the attached mini cable and attach to the camera with a supplied shoe mount or Velcro, and have LED indicators for battery condition, transmit and receive operation.

COMPLETE SYSTEMS

WHX-PRO

Handheld Mic System



- Completely self-contained VHF wireless handheld microphone/transmitter
- Has two switchable frequencies for selecting the cleanest sound
- Delivers crystal-clear sound at a range of more than 200 feet
- On/off switch with standby position for quiet switching
- Includes shoe mount and Velcro for mounting the receiver to the camera, and an earphone for monitoring
- Both microphone and receiver have LEDs for operating verification

WHX-Pro.....**149.95**

WLX-PRO

Lavalier Mic System



- Compact, lightweight, transmitter clips to subject's belt
- Sensitive electret condenser lavalier mic with attached tie clip
- Two switchable frequencies for selecting the cleanest sound
- Delivers crystal-clear sound at a range of more than 200 feet
- Receiver includes shoe mount and Velcro for mounting to the camera, and an earphone for monitoring

WLX-Pro.....**134.95**

CAM-3

Miniature Mic Mixer



This miniature 3-channel microphone-mixer allows you to add and mix sound from three microphone sources simultaneously. Only 3¼″ x ¼″ x 2⅛″, it weighs only 3 ounces, uses no batteries, and attaches to the camcorder or the hand strap.

CAM-3 Mixer **44.95**

WMS-PRO

Two Mic System



- Complete wireless mic system with two microphones (one lavalier, one handheld)
- Delivers crystal-clear sound at a range of more than 200 feet
- Two switchable frequencies for selecting the cleanest sound
- Miniature transmitter attaches to a belt or fits into a pocket
- Receiver includes belt-clip, Velcro, shoe mount for added versatility.

WMS-Pro.....**143.95**

COMPONENTS (RECEIVERS)

WR22-PRO

Discrete 2-Channel VHF Receiver

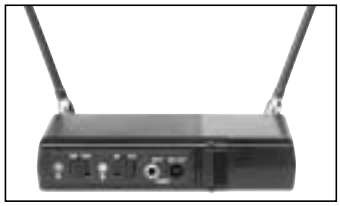


- Unique VHF receiver allows use of two wireless microphones simultaneously.
- Captures the voices of two people at the same time, with each on a separate channel.
- Each transmitter can be from 10 to more than 250 feet from the WR22-PRO receiver.
- Smaller than a pack of cigarettes, the WR22-PRO mounts on a camera with supplied shoe mount or Velcro.
- Operates on 9 volt battery and can be powered (with optional adapters) by 12 volt DC or 120 volt AC.

WR22-Pro **154.95**

WDR-PRO

On-Camera True-Diversity Receiver



A VHF, true diversity receiver that incorporates two complete 2-channel receivers inside the case and two separate antennas to eliminate dropouts and help you create the highest quality audio. Both receivers are set to the same frequency, and the WDR-PRO's circuitry selects the receiver receiving the stronger signal from the microphone and accepts this signal. The LEDs on the WDR-PRO turn from red to green as one receiver, and then the other receives a stronger signal. Unlike the WR22-PRO, you can only use one microphone at a time.

- Diversity circuit avoids sudden dropouts and loss of signal
- Compatible with all Azden "PRO" series microphones
- Two selectable VHF frequencies
- Range of over 300 feet
- Attaches to your camera with supplied shoe mount or Velcro
- Balanced output
- 9-volt battery or 12-volt DC operation

WDR-PRO **169.95**

The incredible success of the "PRO" Series among videographers has prompted Azden to add two more frequencies, 171.105 and 171.845, to the WR-22-PRO and WDR-PRO receivers and the WL/T-PRO, WM/T-PRO and WX/T-PRO transmitters. These new frequencies will not interfere with Azden's long-established 169.445 and 170.245 MHz frequencies. (When ordering the new frequencies, specify "S" models.

COMPONENTS (TRANSMITTERS)

WM/T-PRO

Handheld Mic/Transmitter

Same self-contained, handheld microphone as used in the WHX-PRO, it can also be used with WLX-PRO, WDR-PRO or WR22-PRO receivers **104.95**

WL/T-PRO

Beltpack Transmitter

Same lapel microphone and beltpack transmitter as used in the WLX-PRO, it can also be used with WHX-PRO, WR22-PRO, or WDR-PRO receivers **77.95**



WX/T-PRO

XLR Plug-in Wireless Transmitter



The WX/T-PRO converts any dynamic microphone with an XLR connector to wireless operation when combined with the WR-22 PRO or WDR-PRO receiver. Adjustable audio level accommodates a variety of microphones. Superb frequency response allow the true sound of your mic to come through. Power on/off, audio mute and status LED complete the full array of onboard controls needed for operational ease. It has a sleek ergonomic design in a durable ABS housing, and can operate for up to 10 hours on a single AA battery

WX/T-PRO **119.95**

AZDEN

VHF PERFORMANCE SERIES

VHF Wireless Systems

Designed for professional use, the Performance Series offers six receivers including a portable receiver, four transmitters, and is available in a choice of 25 frequencies in the VHF band (174-216 MHz), for the ultimate in flexibility and versatility. With the ability to use up to 10 mics simultaneously, these systems are ideal for schools, bands and house of worship. Working range is 300´ under average conditions.

COMPONENTS (TRANSMITTERS)

31 HT *Handheld Transmitter*



- High quality uni-directional dynamic element
- Unique rubber-coated durable ABS housing for comfortable feel and low handling noise
- Has 3-position power switch with standby for muting the audio

31 XT *Plug-In Transmitter*

- Unique plug-in transmitter turns any dynamic "wired" microphone with XLR output into a wireless handheld microphone
- Input level control for maximum flexibility
- Controls include power on/off, audio mute and status LED





31 LT
Lavalier Bodypack Transmitter

- Choice of high quality plug-in omni or uni-directional electret condenser microphone
- Input level control for maximum flexibility
- Constructed of durable ABS housing
- 3-position power switch with standby for muting audio

31 IT *Instrument Transmitter*

- Fixed 1/4″ cable for electric guitars and basses
- Durable ABS housing
- Input level control for maximum flexibility
- 3-position power switch with standby for muting audio

32 BT *Bodypack Transmitter*

For use with all Performance Series and Producer Series receivers. It has the same features as the 31LT *plus*-

- On-off-standby switch
- Power LED
- Mini locking 3.5mm jack
- Metal belt clip
- The lavalier microphone is designated EX-503L (locking) **84.95**



VHF FREQUENCY CHART

Channel	MHz	TV
A1	169.505	None
A3	171.045	None
A4	171.905	None
B1	175.400	7
B2	178.225	7
B3	179.200	7
B4	176.425	7
B5	177.650	7
C1	181.400	8
C2	184.025	8
C3	185.125	8
C4	181.850	8
C5	182.550	8
D1	187.800	9
D2	188.200	9
D3	190.600	9
D4	191.300	9
D5	189.325	9
D6	190.075	9
E1	194.400	10
E2	195.425	10
E3	196.800	10
E4	197.325	10
E5	193.825	10
E6	195.975	10

VHF PERFORMANCE SERIES

COMPONENTS (RECEIVERS)

200R
Single Channel Receiver

The result of Azden's design and experience, the 200R Receiver is their newest "value" leader. Housed in a durable ABS case, with power on/off, volume control, and available in 25 frequencies, it gives you excellent RF performance at an attractive price. With Power (red) to Receiving (green) LEDs, and 1/4″ output jack with volume adjustment **74.95**

211R
Single Channel Receiver

Same features as the 200R except the 211R is housed in a metal case for better RF shielding. This results in excellent RF performance even under adverse conditions, and audio quality formerly associated with much more expensive equipment.....**114.95**

221R
Discrete 2-Channel Receiver

Designed to allow the use of two mics simultaneously, the 221R receiver has 2 complete receiver sets built into one case. With Power (red) to receiving (green) A & B LEDs, and two 1/4″ output jacks with independent volume adjustments. Housed in a metal case for better RF shielding and excellent RF performance **185.00**



311DR
True Diversity Receiver

For the professional who needs "rock-steady" RF signal strength, less chance of drop-outs, and exceptional audio performance. With Power and A/B true diversity reception LEDs, 1/4″ output jack, and volume adjustment **149.95**

311DRH
Half-Rack True Diversity Receiver

The half-rack version of the popular 311DR, the 311DRH adds RF/Audio displays, On/Off switch, external squelch and rear mounted detachable antennas. One of two units can be rack mounted with either the HR-1 or HR-2 adapter kits, respectively.....**219.95**

111R
Portable Receiver

Designed for professional videographers who need excellent audio reproduction, the 111R delivers excellent RF performance and wide frequency response - even under adverse conditions. The 111R features output volume adjustment, balanced or unbalanced output capability and adjustable mute/squelch. It also has headphone output with volume adjustment, removable rubber ducky antenna and 9-volt battery or DC 12-volt power input. Includes a mini-mini output cable, with mini-XLR and mini-phone plug, available as options.....**194.50**

FMX-2 *2-Input Field Mixer*

A portable, all-metal 2-channel low-noise field mixer, the ultra-compact (5 x 3.75 x 1.625″ WDH) and ultra lightweight (8 oz.) FMX-2 runs on a single 9-volt battery for up to 12 hours making it almost a requirement for any videographer's camera bag.



- Two balanced XLR mic/line switchable outputs plus a stereo mini-jack output for high-quality DV cameras.
- One mic can be mixed to both outputs as well as either single output while two mics can be mixed to either single output or mixed together to both outputs.
- Two balanced XLR inputs with individual level controls
- Twin "PEAK" level input indicators, a "MONITOR" output (stereo or mono) with level control and a POWER On/Off/Low Battery LED indicator.
- Can be attached directly to a camera with provided velcro or belt-worn using the included 3-position belt clip.....**269.95**

AZDEN

UHF SERIES



51BT
Bodypack Transmitter

Housed in a sleek modern case, the 51BT has 63 user-selectable channels, input level control, standby switch, Hirose 4-pin connector and metal belt clip. Available with a variety of lavalier and headset mics, or as instrument transmitter **114.95**

41HT
Handheld Mic Transmitter

Handheld with super-cardioid uni-directional mic element and 63 user-selectable channels. Uses 2 AA alkaline batteries **179.95**

51XT *Plug-in Transmitter with Phantom Power*

63 user-selectable UHF channels. Special "lockdown XLR connector for secure grip allows you to use your favorite microphone with an XLR output. Features phantom power, adjustable input level control, power on/off and audio mute switches. LED AF Peak and Power indicators **199.95**

UHF Broadcast Quality Wireless System

In today's crowded RF world you need the ability to find clear channels and at the same time, to deliver rock-solid performance without interference. Available with 63 "onboard" user-selectable channels, Azden's UHF Series transmitters and receivers let you quickly and easily find the clearest channels for superb performance at a very reasonable price. Consisting of three transmitters and three receivers, the UHF Series works in the 794-806 MHz band and incorporates crystal-controlled PLL-synthesized circuitry for drift-free reception. The receivers are all housed in heavy-duty die-cast chassis' for better RF shielding and durability.



411DRH *UHF Half-Rack Receiver*

A half-space rackmount crystal-controlled, PLL synthesized UHF receiver with 63 user-selectable channels in the 794-806 MHz band. Up to 9 systems can be used simultaneously. It has detachable antennas mounted on BNC connectors, and both 1/4″ and XLR outputs with volume adjustment. Has LED indicators for RF and AF levels **329.95**

411UDR *UHF Receiver*

A crystal-controlled, PLL synthesized UHF receiver with 63 user-selectable channels in the 794-806MHz band. Up to 9 systems may be used simultaneously. It has both 1/4″ and XLR output jacks and volume adjustment.

500UDR *Portable True Diversity UHF Receiver*

Redefining the parameters of cost-effective, high-band portable wireless technology, the 500UDR is a PLL synthesized UHF receiver with 63 user-selectable frequencies in the 794-806 MHz range. True diversity design houses two separate receivers with twin removable antennas in a ultra lightweight, all-metal chassis that is only 4.1″ square and 1.37″ thick.

- Tone-code squelch
- Dual antennas with BNC connectors
- Ideal for camera mounting, can run over 8 hours on 6 AA batteries, or from your camcorder's battery via a 4-pin XLR 12v DC input
- XLR and headphone output with volume control
- LED indicators for AF peak level, A or B antenna reception, and battery condition
- Compatible with the 41BT bodypack, 41HT handheld mic and 41XT plug-in mic transmitter, the 500UDR easily attaches to your camcorder with the supplied velcro. Monitor the action via the ear phone jack with separate volume control.....**356.95**



100LT SYSTEM

Event Series UHF Wireless Mic System

Introducing a new standard of performance for entry level on-camera UHF. Consisting of the 100UPR Receiver and the 10BT Transmitter, the Event Series 100LT system is small, lightweight and has 63 user-selectable (794-806MHz) crystal-controlled and PLL synthesized channels.



The 100UPR is a true antenna-diversity receiver with 1/8″ mini-jack output, headphone output with volume control, and Power On/Receiving Signal (Red to Green) LED. Powered by 9-volt battery (not supplied), it includes a shoe mount, metal belt clip and velcro. Also includes mini to mini output cable, but can be used with balanced XLR inputs using optional MX-1 cable.

The 10BT Transmitter includes a metal belt clip and the EX-503 omni lapel mic with optional lapel and headset mics available. Features power switch with On LED and separate Standby switch for muting audio. Uses 9V battery (not supplied)

UHF 1000 BROADCAST SERIES

Digitally-Selectable UHF Wireless Mic System

Designed for broadcast professionals, the 1000 Series UHF offers 121 digitally selectable UHF frequencies (723 to 735MHz) in a very compact and lightweight package. Offers performance levels previously attained only with systems costing twice as much.

1000URX • 1000URX/AB • 1000URX-Si

The 9.9 oz. 1000URX receiver features true diversity electronics in conjunction with removable twin high-gain antennas and Azden's proprietary DLC (Diversity Logic Control) circuity for superior, noise-free reception. In addition, the receivers use state-of-the-art dielectric filters for improved image rejection and 5th order filters for an improved S/N ratio. It also features a multi-function LCD readout, XLR mic/line output and a 3.5mm headphone output with level control. The rugged, black metal case measures a mere 3.3 x 3.9 x 1.2″ (WHD).

Exactly the same model, the 1000URX/AB is a 1000URX integrated with the Anton Bauer "Gold Mount" system which allows the receiver to be powered by the camera battery.

The 1000URX-Si is the drop-in version of the 1000 series receiver, designed for Ikegami and Panasonic "Slot-in" video cameras.

1000BT

1000BT Bodypack with LCD readout, digital frequency selection and various microphone options. To ensure a long working life (up to 8 hours) the 1000BT draws less than 60mA from a single 9V alkaline battery. Available with the Azden EX-503H, Sony ECM-44H or ECM-55H lavalier mics.

1000XT

The 1000XT XLR Plug-in converts both dynamic and phantom powered mics to wireless. To ensure a long working life (up to 8 hours), the 1000URX draws less than 60mA from a single 9V alkaline battery.



RE-1

Professional Wireless Mic System



The loudspeaker arrays have gone up. The stage design is finished. The pyrotechnics have been coordinated. The FOH engineer has everything tweaked to his liking. Yet, he takes the time to test and retest the wireless several times. Why? Because like most engineers worth their weight, this guy knows that poor wireless can ruin a show. It is absolutely essential that everything work 100% — with no drop-outs or interference. With over 25 years of experience in wireless behind them, EV's RE-1 system takes frequency agility to a whole new level... touring market power with unmatched simplicity. The clear channels have always been there, but it took EV to bring them to your fingertips. Now you can feel secure that even the most crowded RF environments — places like New York or the San Francisco — are fully manageable.

CSR-1000 Receiver

- Optimized channel groups allow up to 16 systems to operate simultaneously in one frequency band.
- Programmable in 25 kHz steps across 24 MHz operating bandwidth, there are over 950 possible channels so you'll always find a clear channel.
- Advanced ClearScan, automatic group and channel selections, allows quick and simple setup.
- Backlit LCD display shows Group/Channel, transmitter battery status, diversity operation, RF & audio level meters, and space for a name (2 lines, 10 characters each).
- Specially designed "Sound Check" mode lets a person walk test the mic in the performance space with tangible results.
- DSP Posi-Phase Diversity System for maximum range and audio quality.
- Balanced XLR mic output and 1/4-inch unbalanced adjustable line-level output
- Includes rackmount hardware and Three-Year Limited Warranty.

CDR-1000 Dual Receiver

All the features as the CSR-1000 (Advanced Clear Scan, Auto Channel Select, Sound Check and 950-channel frequency agility), the CDR-1000 includes internal power supply, headphone monitoring jack, balanced line-level output, antenna pass through, DC power on antenna jacks for optional RF amplifier, and RE-OneLink PC software for remote monitoring and control.

CSB-100 Bodypack Transmitter

- Cast magnesium housing weighs only ounces but is durable enough to take the pounding on tour.
- Detachable 1/4-wave antenna allows use of different antenna options.
- Cell phone style beltclip included for quick and easy attachment. Optional pouches also available for wearing under costumes or for more vigorous applications.
- Wide selection of lapel and headworn mics and mic accessories for the bodypack
- Unique features for guitar applications:

—Dual band compander circuit provides the audio bandwidth required to cover the lows of a bass guitar to the highs of a solid body.

—Mic/Instrument switch adds a 20 dB pad in the transmit path so the standard bodypack can be used for microphone and guitars.

— Guitar patch cord featuring George L's cable with solderless, interchangeable right angle and straight plugs included to fit your guitar.

Bodypack and Handheld Transmitters Both Feature



- Unique "smart" battery circuit means there is no way to put the battery in wrong.
- On/off button can be disabled to prevent accidental turnoffs during a performance. Also acts as a mute, great for pauses in presentations and worship services.
- Low battery LED will light when the battery needs a replacement.
- LCD display and the same four control buttons as the receiver so programming a channel or frequency is quick and easy.
- Normal and High power transmit means you use just enough power for the application, which maximizes the number of simultaneous systems and limits RF spill over into adjacent buildings/theaters.

CSH-100 Handheld Transmitter

- Interchangeable head allows a choice of elements to fit vocal style and environment.

– N/DYM 767a premium dynamic vocal microphone with VOB, excellent gain-before-feedback for high SPL stages.

– RE-510 premium condenser vocal mic for vocalists, spoken word and quieter stages.

- Internal 1/2-wave antenna for excellent range; stays out of harms way.
- An over-molded Warm-Grip handle reduces handling noise and encourages proper microphone technique for better performances.

100 SERIES

Camera Mountable Wideband UHF Synthesized Systems



Ideal for corporate and event videography, the Lectrosonics 100 System is designed for field production with compact DV camcorders or portable mixers. The system offers 256 selectable UHF frequencies to alleviate interference problems. The tiny UCR100 receiver provides an adjustable output from -50dBv to 0dBv to match mic or line level requirements. Each transmitter offers a full 100mW of output power to deliver a fail-safe system with an exceptional operating range, free from dropouts. The audio processing includes the Lectrosonics exclusive dual-band compander, which lowers distortion and improves the signal to noise ratio over conventional companders. Machined aluminum housings, door and panel provides the ruggedness required to withstand the rigors of daily use.



TRAM INC.

TR50 Lavalier Microphone

The market leader for many years, the miniature TR50 has established itself as the most universal microphone of it's kind with a combination of features not offered by any other mic. The TR50 is a miniature professional electret-condenser microphone designed to provide high sound quality with low, or even no-visibility. The tiny omni capsule features minimum sound coloration and is therefore easily combined with boom-mics during post production. A variety of accessories makes mounting the mic on or under clothing easy and reliable. The TR50 is available with the TR79 power supply for direct connection to a mixer, or with any of a large range of connectors for direct connection to virtually any wireless transmitter including Beyerdynamics, Sennheiser, Lectrosonic and many more.

- Available in Black, Grey, White and Flesh
- External or internal battery powered
- Minimum sound coloration for universal use
- Easily combined with boom microphones
- Available with TR79 for direct XLR connection to audio mixer. The TR79 provides powering and balanced XLR output to the TR50. Connection between the TR50 and TR79 is through a connector of choice, matching the connector used by an accompanying wireless microphone set if required.

TR50 Configurations (Black or Gray):

Each includes Cable Holder (CH), Clip On Holder (COH), Tie Bar (TB), Windscreen (WS), Carrying Case (CC), Tape Down (TD), Tie Tack (TT).

TR50: TR50 with XLR connector.....**294.95**

TR50S: TR50 with Sony 4-pin Hirose **254.95**

TR50SQ: TR50 for use with Sennheiser's EW Series Transmitters.....**216.95**

TR50TA5: TR50 with TA5F connector.**216.95**

add $25 for White or Flesh (Tan) colored versions

Optional Accessories

MC (MIC CAGE:) Makes it easy to mount the TR50 under clothing. The mic is protected from contact noise by two "rails" which keep fabric away from the capsule.

MC-1 Plain Black.....**19.95**

MC-2 Clip-on Black.....**24.95**

TIEBAR: Holds the TR50 while securing itself through the springloaded clamps.

BDTB Tie Bar (Black only).....**18.00**

SAMSON

VM-1/UM-1 SYSTEMS



VHF and UHF On-Camera Micro Diversity Wireless Microphone Systems

With their compact, lightweight, and streamlined design, Samson's VM-1 (VHF) and UM-1 (UHF) wireless systems are ideal for professional videography. They offer great performance along with a host of smart, practical features. Up to six systems can be used simultaneously in either VHF or UHF bandwidths. Designed for camera mounting, they offer a wide variety of lavalier, handheld and headset microphones to suit any application from video deposition and wedding videography to corporate and broadcast video. Virtually identical (except bandwidth) the VM-1 has unbalanced mini-plug output, the UM-1 features balanced-XLR output.

FEATURES

VH-3 and UH-1
Handheld Transmitters

- Transmitters offer a 14 hour typical battery life and three multi-color LEDs indicate relative battery power (low, medium and high)
- External audio On/Off switch allows the audio signal to be muted without inducing a thud or popping sound when disabling the transmitter
- Battery compartment conceals On/Off switch and an audio input trim pot, controlled with a supplied plastic screwdriver lets you optimize audio level performance when using different microphone elements

VT3L and UT1L
Beltpack Transmitters

- A number of lavalier and headset microphones can be connected to the beltpack transmitters using a 3-pin Switchcraft mini-XLR jack
- Permanently attached pig tail antenna on the bodypacks maximize RF transmission
- Beltpack transmitters have a belt clip that can be rotated to any desired position or removed



VM-1 System

VM-1 Micro Series Systems (Includes VM-1 Receiver)

VM-1 Lavalier Systems

with VT-3L Beltpack Transmitter, Mini Cable, Mic Clip and Carrying Case

VML: Without microphone **199.95**
VML-350: With Audio Technica MT350 Lavalier Mic **199.95**
VML-831: With Audio Technica AT831 Lavalier Mic **249.95**

VM-1 Handheld Systems

VMH-HQ: Includes VH3 handheld transmitter with Samson QMic, mini cable and two 9-volt batteries **209.95**

VM-1 Headset Systems

with VT-3L Beltpack Transmitter, Samson Headset Mic, Mini Cable and Carrying Case

VMH-SQE: With QE headset mic **229.95**
VMH-SQV: With QV headset mic **229.95**

VM-1 System Components Only

VM-1 Receiver ..... **169.95** **VT-3** Instrument transmitter **112.50**
VT-3L Beltpack transmitter with P3 connector **124.95**
VH-3Q Handheld transmitter with Samson QMic mic capsule **119.95**

B&H PHOTO - VIDEO - PRO AUDIO **EQUIPMENT LEASING AVAILABLE**

VM-1/UM-1 SYSTEMS

VM-1/UM-1 Micro Diversity Receivers

- Two discrete receiver circuits are housed in each VM-1 and UM-1 receiver while a processor decides which of the two receiver circuits has the better signal
- They run 12 hours on a single 9v battery
- Both offer 1/8″ stereo headphone output with level control
- A/B LEDs indicate whether the left or the right receiver channel is currently in use
- 3 multi-color LEDs (low, medium and high) are switchable to indicate either battery power or RF signal strength and can be disabled to conserve battery power
- Peak LED indicator lights red when approaching clipping which may cause audible distortion
- Battery compartment houses the on/off switch and a Variable Squelch control (with plastic screwdriver) that sets the maximum system range before drop out
- Both antenna mountings allow full rotation for flexibility in placement and can be folded inward for transporting
- Attaches to camera using supplied Velcro



- DC input can accept voltage between 6 an 13 volts from a video camera
- VM-1 has an unbalanced 3.5mm mini stereo phone jack, the UM-1 has a balanced-XLR output (as well as an unbalanced mini). Both include output cable

UM-1 Micro Series Systems and Components

UM-1 Lavalier and Headset Systems

They all include:

UT-1L Beltpack Transmitter, UM-1 Receiver, Balanced mini XLR-to-XLR output cable, Mic Clip and Carrying Case

UM-LP3 Without microphone.....**359.95**
UM-L350 With Audio Technica MT350.....**319.95**
UM-L831 With Audio Technica AT831 lavalier mic.....**364.95**
UM-L44 With Sony ECM-44 lavalier mic.....**409.95**
UM-L55 With Sony ECM-55 lavalier mic.....**514.95**
UM-L77 With Sony ECM-77 lavalier mic.....**574.95**
UM-LMKE With Sennheiser MKE-2 lavalier mic.....**629.95**
UM-HSQE With Samson QE headset microphone.....**399.95**
UM-HSQV With Samson QV headset microphone.....**399.95**
UM-HS75 With Audio Technica ATM75 headset mic.....**429.95**
UM-HS311 With Crown CM311E headset mic.....**529.95**
UM-HSI With Crown Isomax headset mic.....**561.95**

UM-1 Handheld Systems

They all include:

UH-1 Handheld Transmitter with UM-1 Receiver, Balanced mini XLR-to- standard XLR Cable and Carrying Case

UM-HHQ With Samson QMic mic capsule.....**324.95**
UM-HH757 With Electro Voice 757A mic capsule.....**Call**
UM-HH857 With Electro Voice 857A mic capsule.....**Call**

UM-1 System Components

UM-1 Receiver.....**299.95**
UT-1L Beltpack transmitter.....**178.95**
UH-1Q UH-1 handheld transmitter with QMic mic capsule.....**189.95**
UH-1757 UH-1 handheld transmitter with EV 757A mic capsule.....**Call**
UH-1857 UH-1 handheld transmitter with EV 857A mic capsule.....**Call**

UHF MICRO 32

32-Channel UHF On-Camera Micro Diversity Wireless Mic Systems

UHF Micro 32 is a versatile frequency-selectable diversity system offering the highest level of performance for broadcast ENG/field work, corporate video and professional videographers. This remarkable system features 32 channels in both the receiver and transmitter. Sophisticated Diversity circuitry ensures optimum RF performance. The system features sophisticated SAW (Surface Acoustic Wave) filters in the receiver that focuses exclusively on the incoming signal to maximize reception and transmission quality. PLL-synthesized VCO circuitry in the transmitters reduces RF interference from spurious emissions in multi-user applications. The "system of choice" for demanding video professionals.



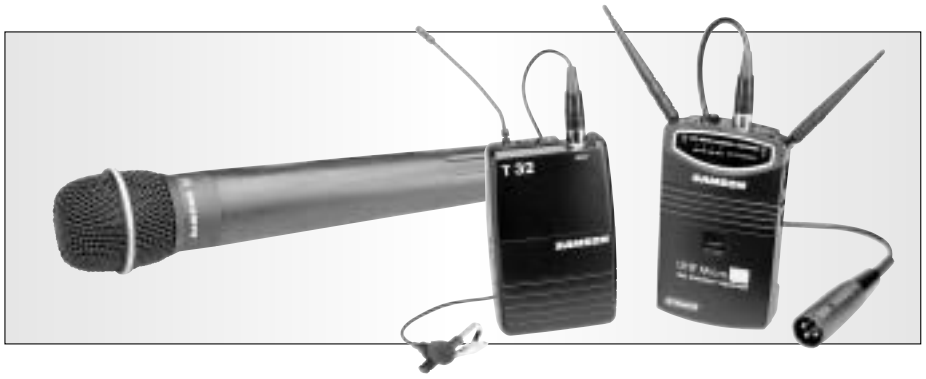
FEATURES

System

- Three bands of operation (low, medium and high), with 32 different available channels per band, all operating UHF bandwidth. Up to six Micro 32 systems (each tuned to a different frequency within the same band) can be used in the same location without interference
- Uses SAW (Surface Acoustic Wave) filters like the one used in high-end television receivers for extremely precise and stable tuning. Ensures that the receiver focuses only on the incoming signal, not extraneous noise or overlapping signals.
- True Diversity technology maximizes active range (up to 300 feet) and reduces potential interference problems through automatic switching between two independent receivers
- Channel and group selectors
- Optimized for use in videography applications, the Micro 32 system is highly customizable, combining a UHF "walkaround" receiver with your choice of beltpack or handheld microphone transmitters, as well as headset systems
- Built-in companding noise reduction in all components for crystal-clear sound with minimal background noise and hiss
- 3-year limited warranty

Transmitters

- Transmitters utilize PLL synthesized VCO technology to absolutely minimize spurious emissions such as RF interference in multi-system applications
- Turning off the audio signal while leaving the carrier signal on, allows audio to be muted without "pop" or "thumps"
- Convenient three-segment multi-color battery strength LED meter for monitoring remaining power in the installed battery
- They run on 9-volt batteries for 8 hours
- 3-pin mini-XLR jack offers connection to guitar cables or a variety of popular headsets and lavalier microphones
- Controls include Power On/Off switch, Audio Frequency Level adjustment, and Group/Channel selectors for 32 channels
- Compact "beeper-sized" belt-pack transmitter is extremely lightweight (less than 4 ounces with a 9-volt battery installed)



M32 Receiver

- Using the supplied velcro strip, the compact, lightweight M32 receiver easily attaches to any video camera
- 3-pin mini-XLR receiver output with supplied XLR to mini-XLR cable
- Includes a pair of tuned antennas and provides balanced and unbalanced outputs (with a three-position output level switch) and a headphone monitor output with adjustable level. Also includes an audio peak LED, dual receiver indicators, and an RF strength meter.

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SYNTH 32

32-Channel Frequency-Selectable UHF True Diversity Wireless Systems



Designed for use in both live sound and sound contracting applications, the UHF Synth 32 system utilizes state-of-the-art technology to provide a high performance, cost effective wireless solution. Frequency-agile, it is capable of operating over any of a number of selectable frequencies, making it ideal for touring setups or where multiple wireless systems need to be used simultaneously. The system features three bands of operation with 32 different available channels per band. Up to 6 Synth 32 systems can be used in the same location without interference. Designed for world-wide use, Synth 32 features a compact half-rack receiver with a large, amber-colored LCD display on the front panel. The system includes PLL synthesized VCO circuitry in the transmitters to optimize use in multi-user situations and SAW filters in the receiver for optimum RF performance. Built-in Noise Squelch circuitry is available to reduce wide band noise.

R32 Receiver

The R32 receiver is a half-rack unit that can be used freestanding or mounted in any standard 19″ rack, making it easy to integrate into any traveling or fixed installation audio system. It includes a pair of tuned, molded antennas for better reception. It provides both balanced and unbalanced outputs, line/mic output level switch, and continuously adjustable Volume and Mute (noise squelch) controls.



- Large, easy-to-read, backlit, front-panel LCD display indicates the status of key functions such as band/group/channel information, dual receiver indicators, audio and RF level strength meters, and Mute status.
- Intelligent Noise Squelch circuitry automatically mutes wide band noise in the receiver whenever it rises above the threshold level of the carrier signal.
- Two R32 receivers can be mounted in a 1U rack space with the optional DR2 Rack Adapter.

Micro 32 and Synth 32 UHF Systems

With T32 Beltpack Transmitter and M32 or R32 True Diversity Receiver

Transmitter only with detachable P3 connector (no mic)

With Audio Technica MT350 Omnidirectional Mic with detachable P3 connector

With Samson QL1 Uni-directional Mic with detachable P3 connector

With Audio Technica AT831 Cardioid Mic with detachable P3 connector

With Sony ECM-44 Omnidirectional Mic with detachable P3 connector

With Sony ECM-55 Omnidirectional Mic with detachable P3 connector

With Sony ECM-77 Omnidirectional Mic with detachable P3 connector

With Sennheiser MKE-2 miniature Omnidirectional Mic with detachable P3 connector

With Samson QE aerobics mic with detachable P3 connector

With Samson QV vocal mic with detachable P3 connector

With Audio Technica ATM-75 headset condenser mic with detachable P3 connector

With Countryman Isomax mic with detachable P3 connector

With Audio Technica Pro 35X uni-directional mic with detachable P3 connector

With Samson HM40 Brass/Woodwind Mic

With GC32 Guitar Cable

With H32 Handheld Transmitter and M32 or R32 True Diversity Receiver

With Samson QMic Hypercardioid element

With Audix OM-5 Cardioid element

SAMSON

AirLine Series

The Revolution in Wireless

The whole point of wireless microphone and instrument systems is to eliminate cables for complete freedom of movement. AirLine Systems takes this idea to the next level with miniaturized transmitters that eliminate conventional body pack transmitters and connecting cables so vocalists, musicians, fitness trainers, public speakers, on air talent and more can cut ALL the cables for the first time! A revolution in wireless technology, Samson's transmitters are so small, lightweight and aerodynamic, they are nearly invisible, providing a completely "hassle-free" user experience. To create the world's smallest wireless transmitters, Samson developed new proprietary technology. Featuring miniaturized circuitry and the ability to operate on a single tiny AAA battery (14 hours typical battery life), these transmitters also feature significantly improved wireless reception and sound quality.





AH1

Fitness and Performance Headset Transmitter

The AH1 Headset Transmitter mounts right on to Samson's comfortable and lightweight headset. Incredibly small, the AH1 still features an easy-to-reach Power On/Off switch; an Input Sensitivity control to adjust the mic level; a Mute (Audio on/off) switch and a multi-function LED that indicates when the transmitter is on and when your battery's running low. And the AH1 comes equipped with with either the Samson QE fitness microphone which is ideal for vocal reproduction or the QV vocal mic with a balanced response for performance. Samson's breakthrough micro circuitry allows the AH1 to operate on a tiny triple-A battery for 14-hours while giving you great sound and reliable wireless transmission over long ranges, wherever you use it.



AL1

Presentation Transmitter

A revolutionary idea in lavalier wireless, the AL1 is an all-in-one miniaturized presentation transmitter with a built-in electret condenser mic! Tiny, the AL1 still has all Power and Mute switches, an input level control and a two-function red LED that indicates when the transmitter is on and transmitting. Highly flexible, it lets you use the transmitter's built-in unidirectional electret condenser mic or plug your favorite lavalier mic into the mini input jack. Phantom power is provided for condenser microphones. An alligator clip allows it to be mounted on a lapel, pocket or belt. Or you can use the convenient lanyard to wear it around your neck. Operates on a tiny triple-A battery with 14-hours of life while giving you great sound and reliable wireless transmission over long ranges.



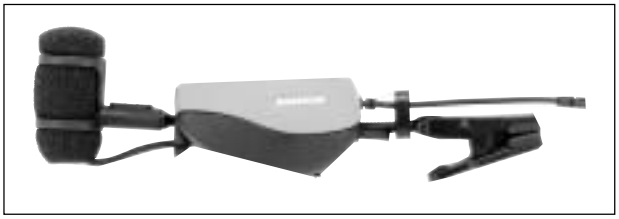
AX1

Handheld Microphone Transmitter

The super-compact AX1 Handheld Transmitter lets you convert any dynamic microphone into a wireless mic. Just plug in the AX1 and you're good to go! And, because it's a little smaller than an XLR connector, you won't even know it's there. Like all AirLine transmitters, the AL1 operates on a triple-A battery with 14-hour battery life and delivers reliable reception and transparent sound over long ranges.

AH1/35x and AH1/Model One

Wind Instrument Transmitters



The first wireless system for wind instruments without a bodypack or microphone cables to limit your movement on stage. The AH1 Headset Transmitter clips right onto your horn. Super compact, the AH1 still has every control you need: Power On/Off switch, input sensitivity control, a Mute (audio on/off) switch and a multi-function LED that indicates when the transmitter is on and the battery is running low. The AH-1 operates on a tiny triple-A battery with 14-hours of life while giving you great sound and reliable wireless transmission over long ranges.



The AH1 is available either with Audio-Technica's popular Pro 35X or AMT's (Applied Microphone Technology) Model One microphones to capture the full range of your instrument.

About the AMT Model One

The ability to securely clamp a microphone to an instrument has always been problematic. The mic either interferes with performance or cannot be properly placed for accurate sonic quality. Due to the irregular shapes of instruments a conventional pinch type clip simply will not hold. Handling noise is another troublesome issue associated with attaching a microphone to an instrument.

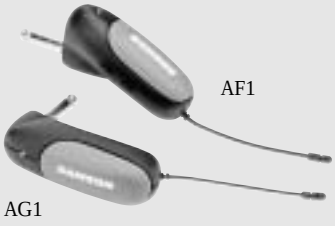
AMT solves these problems by implementing a unique vibration suppression mount into Model One. The four point suspension reduces handling noise as much as 15dB, allowing for much less annoying transients, such as key noise made by touching the instrument. The result is high-quality audio reproduction and an ease of use that is superior to any other microphone in its category.

About the AT 35x

The Audio Technica Pro 35x high-intensity cardioid condenser mic clips on to brass, reeds, piano, acoustic bass, snare and toms, and handles high-intensity sound without distortion. The UniMount clip permits accurate positioning while protecting the element.

AF1 and AG1 Guitar Transmitters

The revolutionary AirLine "plug-in" guitar transmitters come in two versions: the AF1 for Strat(R)-style inputs and the AG1 for traditional Gibson(R)-style end-mount inputs and all others. Less than half the size of conventional body packs, these miniaturized transmitters run on a tiny triple-A battery with 14-hour battery life. Both feature an array of on-board controls: A Power On/Off and Mute switch, red/green LED for Power On/Off, a Peak LED and an input level control for different pickups.





AP1 Guitar Pedal Receiver

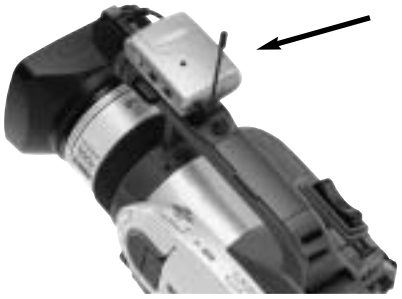
Designed for use with your compact effect pedals, the AP1 diversity receiver gives you reception you can always depend on. It features a 1/4˝ output jack, Power switch, Peak LED and a Volume control. You can use a 9v battery or an AC adapter to power the receiver. When you unplug your the unit, the battery is disabled to conserve power.

SAMSON

AirLine Series

RECEIVERS

AM1 Miniature On-Camera UHF Receiver



The AM1 is camera-mounted micro receiver developed especially for the AirLine system. Even smaller than the typical wireless transmitter, the AM1 uses the same miniaturized circuitry and 800–805 MHz UHF frequency band as the AirLine Series transmitters .

The AM1 is perfect for all video applications requiring outstanding audio and RF performance. Mounting on the hot shoe of your camera or utilizing the belt clip, the AM1 features an audio out, a headphone out with level control for monitoring, a Mic/Line switch for optimized output, a Power Switch and a multi function LED displaying RF, Low Battery and Power. The AM1 receiver can operate 5-8 hours on a single AAA battery. Measures just 2″ x 1-5/8″—making it Samson's smallest receiver ever.

AR1 UHF Micro Receiver



AR1 Micro Receiver is small and lightweight—smaller than even the typical wireless transmitter, yet operates in the 800–805 MHz "clear sky" UHF bandwidth, and comes equipped with a full complement of controls. These include: Output (volume) level control; power on/off switch; LEDs that tell you if the AR1 is receiving a signal (RF LED), and that audio level is okay (Peak LED); Squelch control to eliminate noise; and balanced XLR and unbalanced 1/4″ outputs for connecting to different sound systems. Swivel mounting allows full rotation for optimum placement of the antenna. A special holder is also included, so the receiver can be easily wall mounted to stay out of the way.

UM1 UHF On-Camera Receiver



Compact and lightweight, the UM-1 is a camera mountable receiver ideal for any application from video deposition and wedding videography to corporate and broadcast video. It features A/B LEDs for channel usage as well as LEDs for battery or RF strength and peak LEDs. Battery compartment houses a Variable Squelch control. Both antenna mountings allow full rotation for flexibility in placement and can be folded inward for transporting. DC input accepts voltage between 6 an 13v from a camera. Runs 12 hours on a 9v battery and has a balanced-XLR output as well as an unbalanced mini.



UR1 Rackmount Receiver

Fully professional, the UR1 operates in the 800–805 MHz UHF bandwidth, and features a True Diversity design with two independent receivers, S.A.W. (Surface Acoustic Wave) technology and built-in tuned, molded antennas for the most stable RF reception possible. Housed in a sleek one space, half rack package, it has a front-panel with RF level meter, squelch control and peak audio LED. It also has a balanced XLR (mic/line switchable) and unbalanced 1/4″ output. Two UR1s can be mounted side by side with the optional DR2 Rack Adapter.

- True Diversity system switches to the strongest available signal so you get less dropouts and interference
- 3-color RF level meter indicates signal strength being received
- Built-in noise reduction system accurately tracks the signal for clear sound with minimal noise and hiss
- Squelch control sets minimum threshold level for carrier signal to eliminate unwanted noises and RF interference
- Receiver A/B LEDs indicates which receiver is currently operating
- AF level control adjusts the signal level for both balanced and unbalanced outputs on rear panel
- Audio Peak LED shows when incoming signal is clipping
- Power On/Off switch with status LED
- Tuned molded antennas for best reception
- Rear-panel balanced XLR and unbalanced 1/4″ outputs with a mic/line output level switch

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ONE HOUR FREE PARKING
AT 349 W. 34th STREET (with purchase of $100 or more)

AirLine Series Systems

ELECTRIC GUITAR/BASS SYSTEMS:

SWQSGF: AF1 Transmitter (Fender style) and AP1 Pedal receiver

SWQSGG: AG1 Transmitter (Gibson style) and AP1 Pedal receiver

SWQSBG: AG1 Transmitter (Flat End Mount style) and AP1B Pedal Bass receiver with shape circuitry

SWPSGF: AF1 Transmitter (Fender style) with UR1 receiver

SWPSGG: AG1 Transmitter (Gibson style) with UR1 receiver

HANDHELD SYSTEMS:

SWQSHX: AX1 Plug-in Handheld Transmitter with AR1 receiver

SWPSHX: AX1 Plug-in Handheld Transmitter with UR1 receiver

SWPSHX: AX1 Plug-in Handheld Transmitter with AM1 receiver

PRESENTATION SYSTEMS:

SWQSHX: AX1 Plug-in Handheld Transmitter with AR1 receiver

SWQPHX: AX1 Plug-in Handheld Transmitter with UR1 receiver

SWQPHX: AX1 Plug-in Handheld Transmitter with UR1 receiver

WIND INSTRUMENT SYSTEMS:

SWQSWX: AH1/35X Transmitter with Audio Technica Pro 35X Mic and AR1 Receiver

SWQSWA: AH1/AMT1 Transmitter with AMT Model One Mic and AR1 Receiver

SWPSWX: AH1/35X Transmitter with Audio Technica Pro 35X Mic and UR1 Receiver

SWPSWA: AH1/AMT1 Transmitter with AMT Model One Mic and UR1 Receiver

AEROBICS AND VOCAL HEADSET SYSTEMS:

SWQSCE: AH1/QE Transmitter with Samson QE aerobics Mic and AR1 receiver

SWQSCV: AH1/QV Transmitter with Samson QV vocal Mic and AR1 receiver

SWPSCE: AH1/QE Transmitter with Samson QE aerobics Mic and UR1 receiver

SWPSCV: AH1/QV Transmitter with Samson QV vocal Mic and UR1 receiver

COMBINATION SYSTEM

SWQSWX: AX1 Plug-in Handheld Transmitter, AL1 Transmitter with built-in electret condenser mic, AM1 Receiver, R11 and MT350 microphones

Stage 5 and Stage 55 VHF Wireless Systems

STAGE 5

The Stage 5 system is based on the VR5 receiver. The Super Heterodyne VHF receiver has great RF reception for interference-free performance. Front-mounted, tuned antenna, Tone Squelch control, half-rack VHF receiver design with XLR and 1/4″ outputs and easy to read battery, audio and signal level meters. Available in a guitar, hand-held and headset mic configurations.



Instrument System: ST5 beltpack, SR5 receiver, GC5 instrument cable

Lavalier System: ST5 beltpack, Samson LM5 lavalier mic, SR5 receiver

Handheld Microphone System: HT5 handheld transmitter with Q7 cardioid element, SR5 receiver

Headset System: ST5 beltpack, Samson HS5 headset mic, SR5 receiver

All Stage 5 systems come with an AC power supply and a carry case.



STAGE 55

Dual antenna True Diversity system at a price never seen before. Super Heterodyne VHF receiver has great RF reception for interference-free performance. Front-mounted, tuned antennas, Tone Squelch control, half-rack VHF receiver design with XLR and 1/4″ outputs and easy to read battery, audio and signal level meters. Available in a guitar, hand-held and headset mic configurations.

Instrument System: ST5 beltpack, SR55 receiver, GC5 instrument cable

Lavalier System: ST5 beltpack, Samson LM5 lavalier mic, SR55 receiver

Handheld Microphone System: HT5 handheld transmitter with Q7 cardioid element, SR55 receiver

Headset System: ST5L beltpack, Samson HS5 headset mic, SR55 receiver

All Stage 55 systems come with an AC power supply and a carry case.

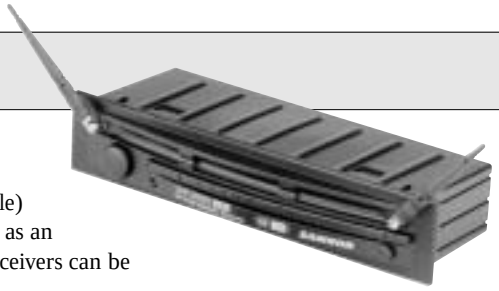
UHF Series One

UHF True Diversity Wireless Mic Systems



Designed to bring high performance UHF wireless within everyone's reach, the truly affordable, UHF Series One operates in the 800-806 MHz band and employs patented PLL (Phase-Lock Loop) transmitter technology to ensure reliable transmission. Built-in companding noise reduction in all components delivers crystal-clear sound with minimized background noise and hiss. Series One also features S.A.W. (Surface Acoustic Wave) receiver circuitry and True Diversity technology for extremely precise tuning and maximum stability of signal reception for extended periods of time. Both the extremely compact "beeper-sized" UT1 beltpack transmitter and sleek, lightweight UH1 handheld transmitter feature a 3-segment LED "battery-life" level meter. Exceptionally compact, the single-space half-rack receiver is equipped with built-in tuned, molded antennas. An optional 19" rackmount adapter allows two receivers to be mounted in one rack space. Ideal for the performer, public speaker or fitness instructor who wants to take their wireless to the next level, or sound contractors looking for "performance critical" wireless that fits into their budget.

UR1 True Diversity Receiver



The UR1 receiver is a half-rack unit that can be used freestanding or can be mounted in any standard 19″ rack, making it easy to integrate into any traveling or fixed installation audio system. It includes a pair of tuned antennas, both balanced XLR (mic/line switchable) and unbalanced 1/4″ output, continuously adjustable AF level and mute controls, as well as an audio peak LED, dual receiver indicators, and a five-segment RF level meter. Two UR1 receivers can be mounted in a 1U rack space with the optional DR2 Rack Adapter.

- True Diversity system switches to the strongest available signal so you get less dropouts and interference
- Multi-segment, 3-color RF level meter indicates strength of signal being received
- Built-in noise reduction system accurately tracks the signal for clear sound with minimal noise and hiss
- Receiver A/B LEDs indicates which receiver is currently operating
- Audio Peak LED shows when incoming signal is clipping
- Power On/Off switch with status LED
- Squelch control sets minimum threshold level for carrier signal to eliminate unwanted noises and RF interference
- Tuned molded antennas for better reception
- Rear-panel balanced XLR and unbalanced 1/4″ outputs with a mic/line output level switch
- AF level control adjusts the signal level for both balanced and unbalanced outputs on rear panel



Two UR1 receivers mounted in the optional DR2 19″ Dual Rackmount kit

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EQUIPMENT LEASING AVAILABLE

UHF Series One



UT1 Beltpack Transmitter

Incredibly compact, the "beeper-sized" UT1L and UT1G beltpack transmitters weigh less than 4 ounces with a 9-volt battery installed, and incorporate PLL technology to lock in the signal for more stable transmission. The UT1G comes prewired with a permanently attached 1/4″ plug for electronic instruments such as electric guitar or bass, while the UT1L provides a mini-XLR connector for connection to a variety of popular headsets and lavalier microphones including Samson's Qe (fitness) and Qv (vocal performance) headset mics. Rotating belt clip lets you position the transmitter any way you want. Permanently attached "pigtail" antenna enables optimum reception.

- Patented PLL (Phase-Lock Loop) and dielectric resonation transmitter technology lock in the signal to ensure stable transmission
- External audio On/Off switch instantly mutes the audio signal without "pops" or thumps while leaving the carrier signal on)
- Convenient three-segment multi-color "Battery-Life" LED meter tells you when your battery is low
- Audio Input Control optimizes the UT1 and UH1 for use with different lavalier and headset mic models, or different mic elements
- They use 9-volt batteries with a battery life of at least 12 hours

UH1 Handheld Transmitter



Comfortable and lightweight, the UH1 handheld transmitter also features PLL technology to ensure stable signal transmission. The UH1 is available with the Samson Qmic and a variety of other mic elements. The Qmic is a high performance wireless dynamic mic design with a modern neodymium element and hypercardioid polar pattern for maximum gain before feedback.

UHF Series One Mic Systems

LAVALIER SYSTEM

SWLSL3: UT1L Beltpack Transmitter, Audio Technica MT350 omnidirectional mic with detachable P3 connector, and UR1 True Diversity Receiver

SWLSL1: UT1L Beltpack Transmitter, Samson QL1 uni-directional Mic with detachable P3 connector, and UR1 True Diversity Receiver

SWLSL8: UT1L Beltpack Transmitter, Audio Technica AT-831 cardioid mic with detachable P3 connector, and UR1 True Diversity Receiver

SWLSP3: UT1L Beltpack Transmitter, detachable P3 connector (no mic included) and UR1 True Diversity Receiver

SWLSL4: UT1L Beltpack Transmitter, Sony ECM-44 Omnidirectional Lavalier mic with detachable P3 connector and UR1 True Diversity Receiver

SWLSL5: UT1L Beltpack Transmitter, Sony ECM-55 Omnidirectional Lavalier Mic with detachable P3 connector and UR1 True Diversity Receiver

SWLSL7: UT1L Beltpack Transmitter, Sony ECM-77 Omnidirectional Lavalier Mic with detachable P3 connector and UR1 True Diversity Receiver

SWLSLM: UT1L Beltpack Transmitter, Sennheiser MKE-2 Omnidirectional Lavalier Mic with detachable P3 connector and UR1 True Diversity Receiver

HANDHELD MICROPHONE SYSTEM

SWLSHQ: UH1 Handheld transmitter with Samson Qmic Hypercardioid element and UR1 True Diversity Receiver

SWLSH5: UH1 Handheld transmitter with Shure SM58 cardioid element and UR1 True Diversity Receiver

SWLSHO: UH1 Handheld transmitter with Audix OM-5 cardioid element and UR1 True Diversity Receiver

HEADSET SYSTEM

SWLSCE: UT1L Beltpack Transmitter, Samson QE aerobics mic with detachable P3 connector, and UR1 True Diversity Receiver

SWLSCV: UT1L Beltpack Transmitter, Samson QV vocal mic with detachable P3 connector, and UR1 True Diversity Receiver

SWLSC2: UT1L Beltpack Transmitter, Audio Technica ATM-75 condenser mic with detachable P3 connector, and UR1 True Diversity Receiver

SWLSH7: UH1 Handheld Transmitter with Shure SM87 condenser mic element and UR1 True Diversity Receiver

SWLSC3: UT1L Beltback Transmitter, Crown CM-311E Cardioid Mic with detachable P3 connector and UR1 True Diversity Receiver

SWLSCI: UT1L Beltback Transmitter, Countryman Isomax Mic with detachable P3 connector and UR1 True Diversity Receiver

INSTRUMENT SYSTEM

SWLSGT: UT1G Beltpack and UR1 True Diversity Receiver

SWLSWX: UT1L Beltback Transmitter, Audio Technica Pro 35X uni-directional mic with detachable P3 connector and UR1 True Diversity Receiver

SENNHEISER

UHF WIRELESS

The Evolution Series Wireless brings Sennheiser's philosophy of affordable performance to the realm of compact wireless systems. Designed from the ground up to give performers, producers and engineers the highest standard of wireless microphone technology, they are ruggedly constructed to survive the toughest performances—night after night. Available in four interchangeable series—the 100, 300, 500 and 550 Series—each system can be configured to fit the requirements of virtually every artist, band, sound system or any other application. Each system is based on a compact, high performance, steel-construction UHF receiver, and comes with your choice of Evolution wireless handheld and bodypack transmitters. The 100 Series is an excellent choice for adding UHF wireless capabilities without sacrificing sound quality or ease of use, and is extraordinarily affordable. The 300 Series offers enhanced performance and is available as an in-ear monitoring system for the ultimate in listening quality and freedom. The 500 and 550 (dual-channel) Series provide utterly professional performance, and are as comfortable on the stage of a national touring act as they are the conference table.

- System operates using switchable UHF frequency technology—previously available only on systems costing several thousand dollars. 1,280 digitally synthesized frequencies are instantly selectable in 25 kHz increments, so it's easy to pick a wireless channel that's free of interference.
- Guarantees interference-free operation with the highest transmission reliability. Advanced diversity techniques receive the clearest signal at all times, as every receiver contains dual receiver circuits picking up signals independently as the performers move around the stage. Constantly comparing the two signals, the receivers select the stronger one at any instant for an audio output that's solid and free of interference.
- Utilizing Sennheiser's unique HDX compander technology, they deliver crystal-clear sound. A noise suppression system specifically developed for stage applications, HDX compander optimizes the mic's dynamic range while minimizing unwanted modulation and transmission noise. The result is powerful, dynamic sound quality, equal to a good wired microphone.
- All transmitters feature a flashing low battery warning LED. 300 Series also provide a real-time battery level display on the transmitter body, while on 5000 Series this information is transmitted to and displayed on the off-stage receiver, so the sound engineer can keep an eye on power levels without interrupting the performance.
- Designed by Sennheiser's engineers to take hard knocks, each transmitter and receiver is protected by a rugged metal housing, built to stand up to the grind of regular stage work.
- High pressure on-stage situations require that a wireless mic system is simple to set-up and use—no go is no show. Evolution Wireless incorporates easy-to-read LED and LCD displays and user-friendly functions, with a logical approach to operations. Frequency, channels, sensitivity and other parameters can be viewed and adjusted in an instant. All components offer this same operating system, so as soon as you're familiar with one product, you know how the rest of the range works.

Evolution 100 Series

Simply the most affordable way to get professional-caliber UHF wireless. The amazing 100 Series is perfect for any musical style, and offers four switchable presets (reprogrammable to any of the 1280 available frequencies), a true diversity receiver for stable, high-quality transmission, and high-quality metal construction that can stand up to the rigors of life on the road Makes any stage performance possible, from extreme rock to sensitive ballads, and performers and presenters of any kind.



- 1,280 user-selectable UHF frequencies
- A choice of four preset frequencies saves you much time when operating several channels simultaneously on stage.
- HDX compander for crystal-clear sound
- High output power (30mW transmitters) for long transmission range
- True diversity receivers for highest transmission reliability
- Transmitters and receivers with rugged metal housings
- User-friendly display on the mount receiver allows you to quickly read off all the important transmission parameters.
- User-friendly menu control via LCD display
- Low battery consumption allows longer performances (low-battery warning as well)
- Preconfigured, ready-to-use systems for every application and every requirement

UHF WIRELESS

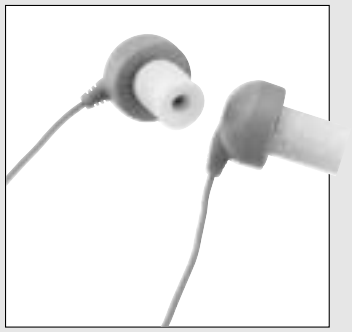
Evolution 300 Series

For increased flexibility and power, the 300 Series steps up with eight switchable presets (reprogrammable to any of the 1280 available frequencies), peak-hold level and battery meter status indicators, and variable-level balanced XLR and TRS 1/4″ connectors. In addition, an alphanumeric name can be assigned to the receiver.



- Eight switchable preset memory locations (using any of the 1280 available frequencies)
- Peak-hold level indicator
- Name, store and recall individual channel presets (e.g. 'lead vocal') in a multi-microphone situation
- Battery status indicator

Evolution 300 Series Wireless In-Ear Monitor System



Why bother with the trouble and expense of traditional floor wedge monitoring when you can go wireless? You get much greater quality, without on-stage feedback problems. The 300 Series In-Ear Monitor System features eight switchable presets, which can be reprogrammed to any one of the 1280 available frequencies, so you can operate several systems simultaneously. It also features the ability to create personalized in-ear monitor mixes for each performer and a unique Focus function, which switches monitoring to "dual mono" mode, enabling performers to individually separate their voice or instrument from the total mix. A switchable limiter protects performers from excessive sound pressure levels.

Evolution 500 Series

Offering the ultimate in performance for concert sound, theater and more, the state-of-the-art 500 Series steps up with sixteen switchable presets (reprogrammable to any of the 1280 available frequencies), assignable alphanumeric names for each of the sixteen presets, headphone output on the receiver (for monitoring), variable level-balanced XLR and TRS 1/4″ connectors, and extensive metering including battery telephony. It also features an innovative Soundcheck mode, which allows an artist or engineer to perform their own on-stage level and RF-check





EM-550 Dual Channel RF Receiver

The EM550 dual-channel true-diversity RF receiver is designed for professional touring and installation applications where unwavering RF performance and rugged construction is mandatory. Incorporates an active antenna splitter, all metal frame with reinforcement, transformer-balanced outputs and headphone monitoring on the front panel.

SENNHEISER

EVOLUTION WIRELESS MICS

135/335/535

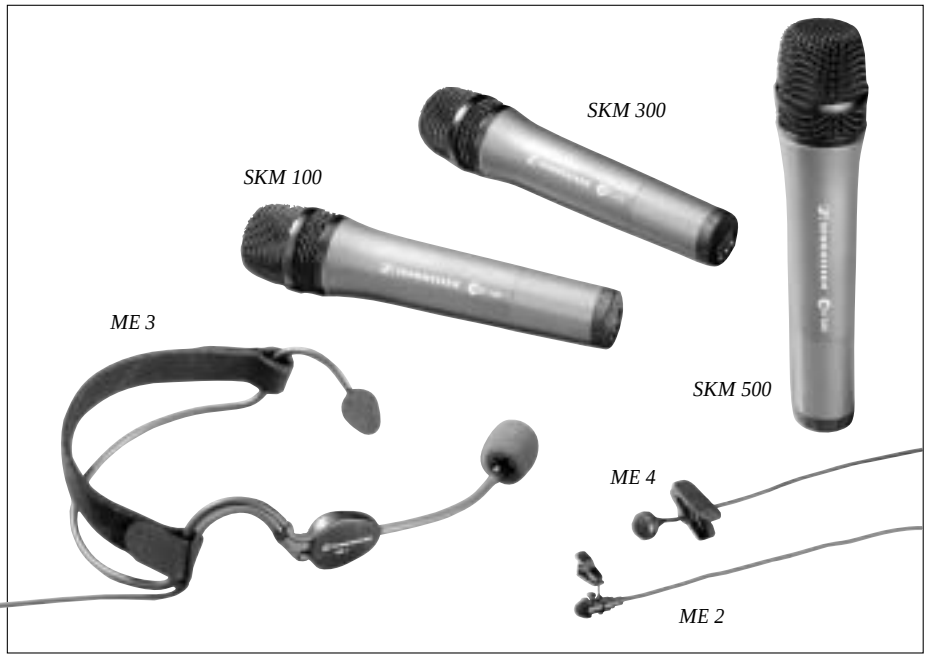
The wireless version of the Evolution e835 classic all-around performer. This mic is designed to complement any vocal style, delivering a punch output with a gentle presence boost. Cardioid pattern offers good signal isolation and feedback rejection. Robust metal construction and shock-mounted capsule minimize handling and impact noise.

145/345/545

The wireless version of the Evolution e845 high-quality vocal dynamic microphone. Ruggedly constructed with a super-cardioid pick-up pattern for a higher signal output to cut through high on-stage volume. A full, warm tonal response is matched with an extended high frequency presence lift for clarity and projection. Consistent on/off axis and proximity response maintains the tonal balance when moving around. Full metal construction and extensive shock-mounting minimize handling and impact noise. A tough high performance stage microphone.

165/365/565

The wireless version of the Evolution e865 condenser mic, the sound is marked by an exceptionally smooth, linear frequency response, capable of exceptionally detailed reproduction with the lowest possible levels of noise and distortion. A super-cardioid pick-up pattern handles high on-stage sound levels without feedback. Also well-suited for broadcast and presentation work. Full metal construction and sophisticated shock-mount suspension shield and protect the capsule for use in the most rigorous conditions.



ME 2

Ideal for presentation and interview work, this compact lavalier design clips discretely onto any item of clothing. With an omni pick-up pattern it projects signals with a flat and extended frequency response for increased vocal intelligibility.

ME 3

A headset microphone of exceptional quality, the ME3 is designed for music and speech applications that require movement and close proximity signal reproduction. The headband is comfortable and immovable, enabling use during exertive dance or sports activities. Super-cardioid condenser design offers excellent feedback rejection.

ME 4

A high quality dynamic lavalier cardioid design—perfect for clear, effective signal projection even with high on-stage sound pressure. Provides levels of ambient signal and feedback rejection not associated with the use of a lavalier microphone. Ideal for presenters, vocalists, and acoustic musicians.



ew100ENG Kit:

Camera Mountable UHF Wireless Lavalier and Plug-On Bundle

Consists of EK100 Receiver, SKP100 Plug-On Transmitter, SK100 BodyPack Transmitter and ME2 Microphone , Battery, (CL100) Unbalanced Mini-to-XLR Cable (Receiver Out-to-Camera In), (CL-1) Unbalanced Mini-to-Mini cable (Receiver Out-to-Camera In), Antennas, Microphone Clip, Clothing Clip and 2 Year Warranty

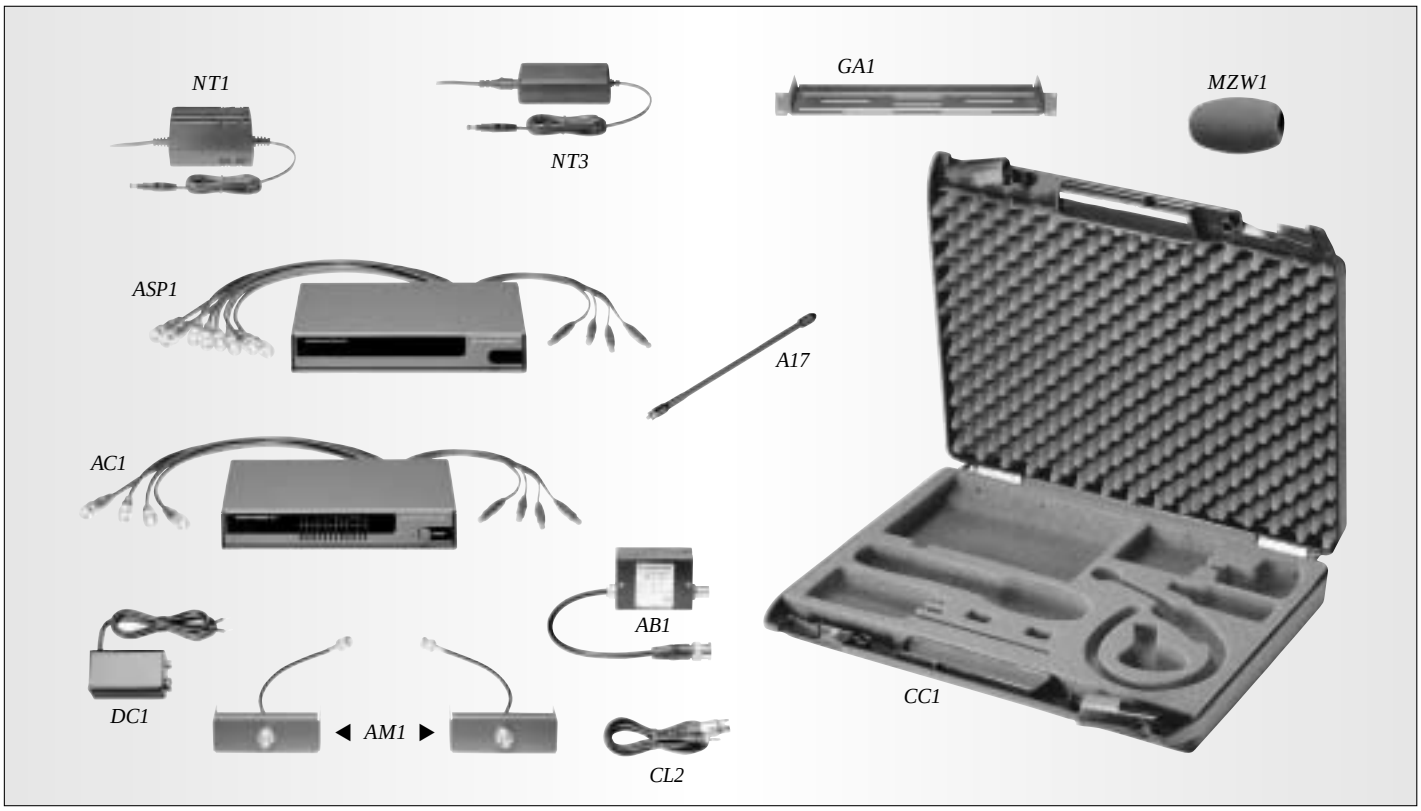
ew500ENG Kit:

Camera Mountable UHF Wireless Lavalier and Plug-On Bundle

Consists of EK500 Receiver, SKP500 Plug-On Transmitter, SK500 BeltPack Transmitter and ME 2 Lavalier Microphone, Battery, (CL500) Balanced Mini-to-XLR Cable (Receiver Out-to-Camera In), Antennas, Microphone Clip, Clothing Clip and 2 Year Warranty

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EVOLUTION WIRELESS MIC ACCESSORIES



ASP-1 Antenna Splitter
Passive antenna splitter allows a single antenna unit to feed up to four fixed receivers simultaneously, reducing the number of antennas required for simplified set-up. Power is delivered direct to the receivers via the ASP-1 without the need for additional cables **429.95**

AC-1 Antenna Combiner
Enables the signal from four separate In-Ear Monitor transmitters to be transmitted with a single antenna **519.95**

A1031-U Antenna
Passive, omni-directional antenna for use as a receiver or transmitter antenna **229.95**

AB-1 Antenna Booster
Particularly useful for compensating for the slight attenuation that occurs when antenna splitters are being used, an antenna booster increases signal levels to receiver. The AB-1 is frequency selectable to avoid interference from unwanted RF sources **279.95**

GA-1 Rack Adapter
Offers convenient side-by-side mounting of two fixed receivers or transmitters in a 19-inch rack.....**39.95**

AM-1 Antenna Mount
This adapter is used to mount antennas at the front, when used with receivers or transmitters in a 19-inch mounting arrangement, moving them from their usual rear position. This enables easier access and installation when rack mounting **39.95**

NT-120 AC Adapter
AC adapter for use with the ASP-1 Antenna Splitter **38.50**

NT-3 AC Adapter
Mains power adapter for use with the AC-1 Antenna Combiner.....**64.95**

get more out of evolution

MZW-1 Windscreen
Foam “pop shield” for handheld microphone transmitters, the MZW-1 effectively suppresses wind and popping. Simple fitting to all handheld mics.....**11.95**

A17 Receiver Antenna
Optional flexible, short receiver antenna, with conductor wrapped around a flexible rod. For use with cameras or camcorders **29.95**

DC-1 Camera Power Adapter
Power supply adapter for use with EK receivers and SK transmitters. Draws power for bodypack directly from camera’s own power supply.....**49.95**

CL-2 Line Cable
Optional line cable for portable bodypack transmitter **14.95**

CC-1 Carrying Case
Hard-shell foam-lined case provides effective protection for your wireless set **58.50**

VP SERIES

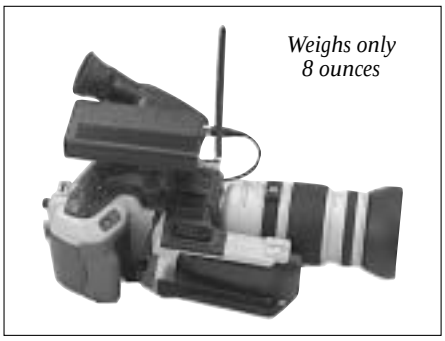
VHF On-Camera Wireless Microphone Systems

Portable, camera mountable wireless system operating in the 169.445 to 216 MHz VHF, the VP series offers beltpack or handheld transmitters using either the WL93 omni-directional lavalier or the world standard SM58 uni-directional handheld microphone. A total of 10 channels are available, and up to six systems can be used simultaneously in the same environment. The VP3 receiver runs on 9-volt batteries and connects to either balanced or unbalanced camcorders and mixers. Ideal for documentary, broadcast and event videography, the system operates up to 300´ and features a frequency response of 80Hz to 15kHz ± 3 dB.



VP3 Receiver

- Mounting options include velcro, removable belt clip or camera shoe adapter.
- Runs for 8 hours on one 9-volt battery
- Available in 10 frequencies
- Stereo/mono compatible 1/8˝ (3.5mm) headphone output with level control.
- Balanced mic level output



- 3- pin “Tini Q- G” audio output connector provides balanced output to video camera or audio mixer when used with the optional WA450 XLR male adapter.
- Red peak light indicates audio overload or flickers when receiving loud signals
- Power/RF indicator illuminates green when the receiver is turned on and orange when an RF signal is present.
- Recessed squelch control (adjustable with supplied screwdriver) mutes the receiver when no transmitter signal is received.
- Pivoting antenna

T2/58 Handheld Microphone Transmitter

- Includes SM58 cardioid mic capsule
- Audio Hi/ Lo gain switch in the battery compartment controls the audio level of the microphone

T2/58 AND T1 TRANSMITTERS BOTH FEATURE

- On/ Off switch mutes the mic without having to turn the transmitter off. This eliminates “pop” or “thump” sounds that occur when the mic is turned on and off.
- Indicators show when the power is on and the battery is good (approx. 18 hours on a 9v battery) or when the battery has less than one hour’s worth of power

T1 Body-Pack Transmitter

- Tini Q-G microphone jack connects to the supplied WL93 subminiature omni-directional lavalier mic or the WA310 XLR microphone adapter cable
- Flexible antenna wire is permanently attached to the transmitter
- Audio level control is achieved with the supplied small screwdriver
- Includes belt clip allowing it to be worn on a belt or waistband

VP WIRELESS SYSTEMS, COMPONENTS AND ACCESSORIES

Complete Systems *(including)*: VP3 receiver, WA460 TA3F to 3.5mm stereo mini cable, (2) 9v batteries, velcro mount and camera “Hot Shoe” mount

VPL93 Lavalier System: With T1 Belt-Pack Transmitter and WL93 lavalier mic **284.95**

VPH58 Handheld System: With T2/58 Handheld Mic/Transmitter.....**324.95**

Components:

VP3 Receiver.....**159.95**

T1 Belt pack transmitter (no mic) **119.95**

T2/58 Handheld transmitter.....**194.95**

WL93 Subminiature lavalier mic.....**69.95**

Accessories:

WA450 Balanced Tini-Q-G to XLR male cable for VP3 receiver.....**22.50**

WA360 In-line audio switch for T1 transmitter.....**57.95**

WA395 System carrying case **19.95**

WA310 Microphone adapter cable for T1 transmitter.....**19.95**

PS20 AC Adapter for VP3 receiver **8.95**

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ONE HOUR FREE PARKING
AT 349 W. 34th STREET (with purchase of $100 or more)

UP-4

UHF Frequency Agile Portable Wireless Receiver

Designed for professional broadcasters and videographers, the UP4 is a portable UHF wireless receiver that provides the perfect combination of crystal-clear audio and rugged dependability — no matter the environment—at a price that will easily fit your budget. The UP4 is available alone or as part of a complete UP Portable UHF Wireless System with your choice of Shure UC or UHF series transmitters. (See UC Wireless Systems [page 756] and UHF Wireless Systems [page 758] for transmitter features and specifications.)





Engineered to avoid interference and deliver a clear, steady signal:

- Frequency Agility - provides over 100 selectable UHF channels
- IF (Intermediate Frequency) Filtering - selects the desired signal
- Phase Locked Loop (PLL) Circuitry - ensures the signal is steady
- Tone Key Circuitry - blocks out other RF signals that could interfere with reception when the transmitter is off
- Noise Squelch Circuitry - analyzes signal quality to evade noise bursts when the transmitter is off
- Patented Shure Predictive Diversity technology anticipates dropouts before they occur, switching antennas when necessary for consistent audio and RF reception

Designed and constructed for dependable and reliable operation even under the most challenging environments:

- Runs up to 10 hours on a single 9V battery
- Extruded aluminum chassis - built tough to withstand the real rigors of the road
- Attaches securely to NP-style and Anton-Bauer Gold Mount battery outfits
- Unique mounting design allows the battery to be changed without removing the receiver from the camera
- Innovative weather-resistant cover snaps easily around the top of the receiver for protection against the elements
- Includes a rugged system carrying case
- Legendary Shure quality - subjected to one of the most stringent environmental and drop testing programs in the industry.

Additional Features:

- Bi-colored LEDs to monitor multi-feature operation status
- Polarity protected battery case
- SMA antenna connectors with removable antennas
- Thread-locking DC input jack for external power option
- Push-lock audio gain control
- Mic level XLR output

UP Systems with UC Series Transmitters

UPC14/93: UC1 Bodypack Transmitter and WL93 Omnidirectional Micro Lavalier

UPC24/58: UC2/58 Handheld Transmitter with SM58 Capsule

UPC24/87: UC2/87 Handheld Transmitter with SM87A Capsule

UPC124/93: UC1 Bodypack Transmitter with WL93 Omnidirectional Micro Lavalier mic and UC2/58 Handheld Transmitter with SM58 Capsule

All UP Systems include UP4 Portable Receiver with 3´ XLR output cable , weather resistant cover, system carrying case, two 9v or AA batteries, 9v battery, mounting box and DC power cable

UP Systems with U Series Transmitters

UP14/93: U1 Bodypack Transmitter and WL93 Omnidirectional Micro Lavalier

UP24/58: U2/58 Handheld Transmitter with SM58 Capsule

UP24/87: U2/87 Handheld Transmitter with SM87A Capsule

UP124/93: U1 Bodypack Transmitter with WL93 Omnidirectional Micro Lavalier mic and U2/58 Handheld Transmitter with SM58 Capsule

T SERIES

VHF Wireless Mic Systems

Whether it's for vocal, speech, or instrument performance, the T Series has a wireless system for you. Working in the VHF frequency band, the T Series systems give you the reliability and flexibility you need, at a price you can afford. Pre-packaged specifically for vocalists, guitarists, presenters and instrumentalists, all T Series systems feature:



- *Convenient mute switch for noiseless control of On/Off function*
- *Noise squelch circuitry to virtually eliminate noise bursts*
- *A durable carrying case for easy storage and travel*
- *Unique "battery saver" technology allows up to 12 hours of operation from a single 9-volt battery. This makes the T Series ideal for live sound reinforcement, conference rooms and other installation applications that require long uninterrupted system performance.*

T1/T1G Beltpack and T2 Handheld Transmitters

- 300´ transmission distance
- Audio Gain Control accommodates input sources with different signal strengths (e.g., speaking or playing an instrument).
- On/Mute switch mutes audio without turning the transmitter off.
- Powered by 9v batteries, a low battery indicator glows red when less than an hour of operating time remains.
- Power On/Off switch is recessed to prevent it from being accidentally turned off.

T1/T1G Beltpack Transmitters Only

- Belt clip secures the transmitter to a belt, waistband or guitar strap. On the T1, the Tini Q-G input connects to lavalier and headset mic cables, and Shure WA302 instrument adapter cable. The T1G accepts a 1/4″ or guitar cable plug.

T3 Non-Diversity Receiver



- Volume control of receiver output
- Telescoping antenna
- Unbalanced 1/4″ output
- RF (radio frequency) signal indicator glows yellow when RF signals are received
- Audio peak indicator flashes red when the input signal approaches clipping levels
- Noise squelch circuit analyzes signal quality as opposed to signal strength, discriminating between noise and desired signal. This reduces noise burst due to environmental RF noise.



T4A Diversity Receiver

Same features as T3 PLUS—

- Exclusive MARCAD (MAximum Ratio Combining Audio Diversity) circuitry continuously processes the RF signal from each antenna and combines them to produce one signal of optimum quality.
- Unbalanced 1/4″ phone plug and balanced XLR connectors for simultaneous output to different devices. XLR output is selectable between mic and line level.
- Can be rack-mounted with the optional URT rack mount tray

T88 Dual Channel Non-Diversity Receiver

The T88 receiver offers dual channel performance without having to compromise your budget or standards. Ideal for applications from DJs and Karaoke, to schools and houses of worship, the T88 delivers full, rich sound and allows two transmitters to be used simultaneously.

- Power, RF signal and audio peak indicators
- Audio output level (volume) controllers
- Telescoping antennas
- Independent 1/4″ audio output connectors



T SERIES



The Vocal Artist—
Offers wireless freedom combined with Shure's true-to-life sound and reliability in a single, economical package. Includes T3 Receiver and T2 Handheld Transmitter with SM58 or PG58 mic head. Also includes a swivel adapter, carrying case and 9v battery.

TVS/PG58: Includes PG58 Cardioid Mic and T3 Receiver **176.95**
TV58S: Includes SM58 Cardioid Mic and T3 Receiver **229.95**
TVD/PG58: Includes PG58 Cardioid Mic and T4A Receiver **231.95**
TV58D: Includes SM58 Cardioid Mic and T4A Receiver.....**279.95**

The Presenter—

From stage to podium to pulpit, this system delivers crisp, clean sound and outstanding reliability for speech and theater applications. Rugged design and maintenance-free operation provides the big-time performance you need and the freedom to go where you want.



TPS: WL93 micro-miniature lavalier mic, T1 body-pack transmitter with 4-pin mini connector, and T3 Receiver **208.95**

TPD: WL93 micro-miniature lavalier mic, T1 body-pack transmitter with 4-pin mini connector, and T4A Receiver **299.95**

TPD/84: WL184 lavalier mic, T1 body-pack transmitter with 4-pin mini connector, and T4A Receiver.....**299.95**

TPD/85: WL185 lavalier mic, T1 body-pack transmitter with 4-pin mini connector, and T4A Receiver.....**294.95**



The Headset— As comfortable as it is durable, The Headset features a lightweight, fully adjustable wire frame that's tough enough to stand up to continuous highly active use. Unique elastic band keeps it in place from start to finish. Ideal for aerobic use, DJing, or for singing musicians. Aerobics instructors and DJs will appreciate the freedom, comfort and quality of this system. The WH20 dynamic microphone features a fully adjustable, lightweight frame. The gooseneck microphone can be quickly repositioned for optimal sound clarity or moved aside during breaks.

THS: WH20 Cardioid Dynamic headset mic, T1G bodypack transmitter with 4-pin mini connector and T3 Receiver.....**208.95**

THD: WH20 Cardioid Dynamic headset mic, T1G bodypack transmitter with 4-pin mini connector and T4A Receiver.....**278.50**



*The Guitarist—*Designed exclusively for guitars - The Guitarist delivers the crystal clear highs and deep, resonant lows that you'd expect from a wired set up. Shure's rugged Surface Mount Technology and input gain adjustment allow for the best signal-to-noise ratio with the widest variety of guitars and playing styles.

TGS: Includes T1G transmitter, two WA303 Guitar Cables and T3 Receiver **169.95**

TGD: Includes T1G transmitter, two WA303 Guitar Cables and T4A Receiver **229.95**

The Instrumentalist —
Versatile solution for high volume wind, brass or percussion. Features the high SPL WB98H/C mini-cardioid condenser microphone with integrated gooseneck and clip-on clamp for secure fit and positioning.



UT14/98H: Includes WB98H/C Gooseneck Clip-On Instrument Mic, T1 body-pack transmitter with 4-pin mini connector, and T4A Receiver .....**Call**

Dual Systems with T88 Dual Channel Receiver
Offering dual channel performance without having to compromise budget or standards.

TwinPro Vocal Artist—

T288: Two T2 Handheld Transmitters with PG58 Cardioid Mics, and T88 TwinPro Dual Channel (non diversity) Receiver **393.50**

TwinPro Presenter—

T188/93: Two WL93 micro-miniature lavalier mics, two T1 Bodypack Transmitters with 4-pin mini connector and T88 TwinPro Dual Channel (non diversity) Receiver **393.50**

T188/85: Two WL185 professional cardioid lavalier mics, two T1 Bodypack Transmitters with 4-pin mini connector and T88 TwinPro Dual Channel (non diversity) Receiver **393.50**

TwinPro Vocal Artist/Presenter—

T1288/93: T2 Handheld Transmitter with PG58 Cardioid Mic, WL93 micro-miniature lavalier mic, T1 Bodypack Transmitter with 4-pin mini connector and T88 TwinPro Dual Channel (non diversity) Receiver.....**393.50**

T1288/85: T2 Handheld Transmitter with PG58 Cardioid Mic, WL185 professional cardioid lavalier mic, T1 Bodypack Transmitter with 4-pin mini connector and T88 TwinPro Dual Channel (non diversity) Receiver.....**393.50**

PERFORMANCE GEAR WIRELESS

UHF Diversity Wireless Mic Systems

Whether for vocal, speech, or instrument performance, Shure's Performance Gear Wireless has the right system for you. Offered in UHF bandwidths, they give you the reliability and flexibility you need, at a price you can afford. Pre-packaged specifically for vocalists, guitarists, presenters and instrumentalists, all Performance Gear Wireless systems feature:



- Noise squelch circuitry to virtually eliminate noise bursts
- Incorporating Shure's exclusive Marcad Diversity technology, the UT4A receiver seamlessly combines audio signals from two independent receiver sections—improving operating range and clarity while virtually eliminating dropout and switching noise.
- Convenient mute switch for noiseless control of On/Off function
- A durable carrying case for easy storage and travel
- Unique "battery saver" technology allows up to 18 hours of operation from a single 9v battery. This makes the Performance Gear Series ideal for live sound reinforcement, conference rooms and other installation applications that require long uninterrupted system performance.

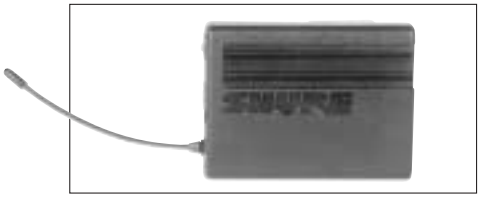
UT4A Receiver



- Exclusive MARCAD (MAximum Ratio Combining Audio Diversity) circuitry continuously processes the RF signal from each antenna and combines them to produce one signal of optimum quality.
- Unbalanced 1/4″ phone and balanced XLR connectors for simultaneous output to different devices. XLR output is selectable between mic and line level.
- Multiple receivers can be stacked or rack-mounted, with two receivers fitting into the optional URT rack mount tray
- 3″ rubberized antennas offer greater durability than telescoping antennas
- Yellow diversity A/B lights glow when RF (radio frequency) signals are received from the UT1 or UT2 transmitter on one or both of the antennas
- Audio peak indicator flashes red when the input signal approaches clipping levels
- Volume control of the receiver output
- Noise squelch circuit analyzes signal quality as opposed to signal strength, allowing the circuit to discriminate between noise and desired signal. This reduces noise burst due to environmental RF noise
- Retainer secures the AC power cable to the receiver

UT1 Body-Pack Transmitter

- Input attenuation switch selects between 0 dB and –20 dB attenuation, for greater range of audio gain control



- Audio gain control accommodates various input signal strengths. A small screwdriver is supplied to make adjustments.
- Tini Q-G connector allows connection to a variety of lavalier and headset microphones and cables
- Flexible wire antenna is permanently attached to the bottom of the UT1 body-pack transmitter.

UT1 and UT2 Transmitters Both Feature

- Transmitters work at a distance of up to 300′ from the receiver.
- Red light warns you that there is less than one hour of battery life left
- Power On/Off switch is recessed to prevent it from being accidentally turned off.
- 9-volt battery operation, a green light indicates that they are on. Runs up to 18 hours on a single battery.
- On/Mute switch prevents unwanted sounds from being picked up without turning the transmitter off.

UT2 Handheld Transmitter

- Audio gain control hidden in battery compartment, allows sound levels to be matched with the transmitter for better performance.



PERFORMANCE GEAR WIRELESS



The Vocal Artist - Offers wireless freedom combined with Shure's true-to-life sound and reliability in a single, economical package. Choice of either a T4A or UT4A diversity receiver with power supply, and a T2 Handheld Transmitter with an SM58, PG58 or BETA 58 mic head. Each also includes a swivel adapter, carrying case and 9v battery.

UT24/58: Includes UT2 Handheld Transmitter with SM58 Cardioid Microphone and a UT4A Diversity Receiver

UT24/PG58: Includes UT2 Handheld Transmitter with PG58 Cardioid Microphone and a UT4A Diversity Receiver

UT24/BETA58: Includes UT2 Handheld Transmitter with a Beta 58 Microphone and a UT4A Diversity Receiver



The Headset - As comfortable as it is durable, The Headset features a lightweight, fully adjustable wire frame that's tough enough to stand up to continuous highly active use. Unique elastic band keeps it in place from start to finish. Ideal for aerobic use, DJing, or for singing musicians. Aerobics instructors and DJs will appreciate the freedom, comfort and quality of the Headset system. The WH20 dynamic microphone features a fully adjustable, lightweight frame. The gooseneck microphone can be quickly repositioned for optimal sound clarity or moved aside during breaks.

UT14/20: Includes WH20TQG Cardioid Dynamic headset microphone, UT1 Bodypack Transmitter, and a UT4A Diversity Receiver

The Vocal Artist Headset - Combines Shure's true-to-life sound and reliability with the comfortable WH30 headset that provides high-quality voice pickup for active vocalists. Headset includes adjustable wireframe, elastic headband, microphone cartridge and windscreen, microphone boom, isolation shock mount, boom hinge and cable.



UT14/30: Includes WH30 Cardioid Condenser headset microphone, UT1 Bodypack Transmitter, and a UT4A Diversity Receiver



The Presenter - From stage to podium to pulpit, the Presenter delivers crisp, clean sound and outstanding reliability for speech and theater applications. Choose from omni, cardioid & supercardioid lavalier mics-each providing the big-time performance you need and the freedom to go where you want. Features rugged design and maintenance-free operation.

UT14/93: Includes WL93 micro-miniature lavalier microphone, UT1 Bodypack Transmitter, and a UT4A Diversity Receiver

UT14/84: Includes WL184 supercardioid lavalier microphone, UT1 Bodypack Transmitter, and a UT4A Diversity Receiver

UT14/85: Includes: WL185 cardioid lavalier microphone, UT1 Bodypack Transmitter, and a UT4A Diversity Receiver

The Instrumentalist — Versatile solution for high volume wind, brass or percussion. Features the high SPL WB98H/C mini-cardioid condenser microphone with integrated gooseneck and clip-on clamp for secure fit and positioning.



UT14/98H: Includes WB98H/C Gooseneck Clip-On Instrument Mic, UT1 Bodypack Transmitter, and a UT4A Diversity Receiver



The Guitarist – Designed exclusively for guitars, The Guitarist delivers the crystal clear highs and deep, resonant lows that you'd expect from a wired set up. Shure's rugged Surface Mount Technology and input gain adjustment allow for the best S/N ratio with the widest variety of guitars and playing styles.

UT14: Includes WA303 Guitar Cable for connection to UT1, WA302 Instrument Cable for connecting UT4A to Guitar Amplifier , UT1 Bodypack Transmitter, and a UT4A Diversity Receiver

Vocal Artist/Presenter - Includes the legendary SM58 hand-held transmitter plus the WL185 lavalier and body-pack transmitter. This package is for people who want the flexibility and versatility of two microphone options.



UT124/85: Includes: SM58 Handheld Mic, WL185 cardioid lavalier microphone, UT1 Bodypack Transmitter, and a UT4A Diversity Receiver

LX SERIES

VHF Wireless Mic Systems

Widely used in installed sound and live music situations, Shure's LX wireless systems deliver top-notch performance and VHF reliability at a moderate price. With a full line of professional handheld, lavalier, instrument and headworn microphone options, as well as instrument cable, and single-antenna or diversity receivers, a wireless system can be configured for almost any application. Plus, guitar players who demand the highest degree of sound quality and reliability will really appreciate the low noise and uncolored sound of an LX Wireless System.







LX1 Beltpack Transmitter

- Low-profile, highly durable LX1 body-pack transmitter features a wide dynamic range and unsurpassed reliability, plus convenience features like vertical and horizontal actuated toggle mute switches, which are easily identified by touch.
- Belt clip has a wider contact surface for a better grip, and firmly holds thinner, more slippery materials. It can be removed or inverted for special positioning.

Both Transmitters Feature

- 18-20 hour battery life
- Power and mute switch
- Adjustable audio gain control
- Both transmitters include a 3-segment LED that indicates both "power on" and the amount of battery life remaining.

LX2 Handheld Transmitter

- Beneath the sleek, elegant exterior of the LX2 handheld transmitter lie the electronics and engineering that deliver the sound that's spelled success for so many users.
- Battery mounting provides easy access and outstanding transmitter balance.
- The unbreakable WA555 grip/switch cover accessory, supplied with all LX2 hand-held mic transmitters, prevents accidental movement of the power and mute switches and provides a "grip" feel.

LX3 and LX4 Receivers

- Half-rack wide, the all-metal LX3 and LX4 receivers are supplied with both single and dual rack-mount hardware. An optional WA503 accessory kit lets you front-mount antennas.
- Noise Squelch control located on the front panel for easy access, analyzes signal quality instead of signal strength. This virtually eliminates the possibility of annoying noise bursts coming through your receiver.
- A five-segment audio meter helps to optimize transmitter gain setting and lets you monitor audio level during operation.
- UHF-type connectors provide connection to the supplied 1/4 -wave antennas or to coax cable used with remote antennas. They also provide connection to the optional WA421 remote antenna cable kit used with optional WA380 and WA490 1/2 -wave antennas.
- Supplied AC adapter incorporates small, in-line transformers that save space on AC power strips. It also has mounting tabs so it can be secured to any surface, as well as locking DC power connectors to prevent accidental disconnection from the receiver.

LX4 Only

- Instead of a single RF meter, the LX4 has two 5-segment antenna meters. The dual meters indicate received signal strength at each antenna, and make it easier to identify and troubleshoot RF "dead spots."
- MARCAD (MAximum Ratio Combining Audio Diversity) circuitry constantly monitors signals from both receiver sections and combines them into a single output signal. Provides improved reception and freedom from dropouts.

LX88-II

The LX88-II is a VHF wireless mic system that provides two channels in one single-rack sized receiver– perfect for DJ or karaoke applications.



- Supplied with anti-roll device for handheld transmitters
- Metal chassis contains two single antenna (non-diversity) receiver sections
- LED indicators: two RF (one per channel), two audio peak (one per channel), power-on
- Two balanced XLR connectors and three unbalanced 1/4″ connectors
- Detachable racks ears with holes for optional front-mount antenna connector kit
- Two rear-mounted, detachable antennas with UHF connectors

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LX SERIES

Headworn Systems

LX14/16:
LX1 bodypack transmitter with WCM16 headset mic and LX4 receiver

LX14/20:
LX1 bodypack transmitter with WH20TQG headset mic and LX4 receiver

Instrument System

The most popular miniature instrument mic in the world, the cardioid WM98 wireless system is ideal for horns and other acoustic instruments. Included the A98KCS horn clamp which provides mounting capability for most brass and woodwind instruments.

LX14/98:
LX1 bodypack with WM98H/C clip-on instrument mic and LX4 receiver

Guitar Systems

The LX Wireless Guitar System features crisp, clean sound and transmitter batteries that can survive several gigs. And Shure redesigned the belt clip to attach more securely to the guitar strap so guitarists can "lose their leashes" but keep all of their tone for hours longer than before.

LX13:
LX1 bodypack transmitter with WA302 Guitar/Bass Cable and LX3 receiver

LX14:
LX1 bodypack transmitter with WA302 Guitar/Bass cable and LX4 receiver

Handheld Systems

Available in four versions: with the world standard SM58, the SM87 condenser mic, or the high-performance, premium-quality Beta 58 dynamic and Beta 87 condenser, which are the mics of choice for concert sound. And all heads are instantly interchangeable.

LX23/58:
With SM58 and LX3 receiver

LX24/58:
With SM58 and LX4 receiver

LX24/87:
With SM87 and LX4 receiver

LX24/BETA58:
With Beta 58 and LX4 receiver

LX24/BETA87A:
With Beta87A and LX4 receiver

LX Lavalier Wireless Mic Systems (with LX1 Bodypack Transmitter

LX13/93:
WL93 lavalier mic and LX3 receiver

LX14/93:
WL93 lavalier mic and LX4 receiver

LX13/83:
WL183 lavalier mic and LX3 receiver

LX14/83:
WL183 lavalier mic and LX4 receiver

LX13/84:
WL184 lavalier mic and LX3 receiver

LX14/84:
WL184 lavalier mic and LX4 receiver

LX13/85:
WL185 lavalier mic and LX3 receiver

LX14/85:
WL185 lavalier mic and LX4 receiver

LX3/LX4 OPTIONAL ACCESSORIES

WA380
1/2 Wave Antenna Kit

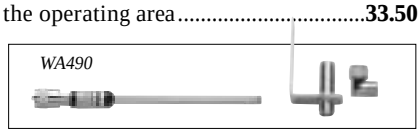
The WA380 is a multi-section, telescoping 1/2 wave antenna that provides up to 3 dB more gain than the supplied quarter-wave antennas. Kit includes wall mounting bracket and bulkhead adapter. LX4 requires two WA380 antennas.....**53.95**

WA421: 20´ antenna cable with connectors for the WA380 or WA490.....**19.95**

WA503: Front mount antenna kit.....**29.95**

WA490
1/2 Wave Cable Antenna Kit

The WA490 provides up to 3 dB more gain than the supplied quarter-wave antennas. A low-cost alternative to telescoping antennas, the WA490 offers unobtrusive wall- or ceiling-mounting in or around the the operating area.....**33.50**



WA405 *Antenna/Power Distribution System for LX3/LX4*

The WA405 combines an amplified antenna distribution system and DC power distribution system in one compact unit. The WA405 uses two antennas, splitting the wireless transmitter signals to feed as many as four receivers. Signal amplification compensates for insertion loss. In addition, four 12v DC outputs provide a combined current of up to 2 amps to power four receivers, eliminating the need for external power supplies. The built-in, high-efficiency switching power supply accepts 120v AC power.....**434.95**

WA470 *Passive Antenna Splitter/Combiner*



When used as an antenna splitter, the WA470 allows two LX4 or four LX3 receivers to receive signals from only two antennas. This minimizes the number of antennas required and helps prevent interference problems in installations with multiple receivers. When used as an antenna combiner, the WA470 routes signals from four antennas to the inputs of a single VHF diversity receiver, while retaining true diversity performance. This is useful in applications where one wireless system must provide coverage in two areas **79.50**

LX88 Dual Systems

LX288/58: Two LX2/58 Handheld Transmitters with LX88-II Dual Channel Receiver and PS20 Power Supply, two swivel mic adapters, rack mounting kit, Anti-Roll Devices

LX1288/58: LX2/58 Handheld Transmitter, LX1 Bodypack Transmitter (no mic) with LX88-II Dual Channel Receiver and PS20 Power Supply, one swivel mic adapter, rack mounting kit, Anti-Roll Device

ULX STANDARD/ULX PROFESSIONAL

Professional UHF Wireless Systems

Affordable systems designed for both live performance and systems integration applications, the ULX Standard and ULX Professional UHF Wireless Systems feature Shure's world-renowned audio and proprietary Automatic Frequency Selection circuitry, an innovation which simplifies system set-up by automatically scanning a user's environment for open frequencies. Over 30 system configurations available with a variety of handheld, headworn, instrument, and lavalier microphone options.

The ULX Standard is designed for regional and touring performers who have multiple wireless system users, want quick setups between gigs, and want the security of less signal interference and dropouts.

The ULX Professional with advanced controls and displays is designed for professional sound installers who require programmable and lockable front panel settings, rely on advanced LCDs, and need a durable, metal-chassis, rack-mounted system.



Both frequency-agile receivers operate between 554-590 MHz (J1) and 662-6987 MHz (M1) and offer up to 1440 selectable frequencies organized into pre-programmed compatible groups. This enables you to quickly set-up and operate up to 40 compatible systems simultaneously (area dependent) when both frequency bands (JI and M1) are utilized. ULX receivers also include Predictive Diversity circuitry which virtually eliminates RF drop-out, as well as an easy-to-read, multi-function LCD display for monitoring group/channel selection, battery level, and TV channel.

Equipped with more advanced features and controls, the ULX Professional steps up with group scan function (the ability to scan all frequency groups to locate the one with the maximum number of compatible frequencies), frequency and volume lockouts, advanced front panel controls, indicators for RF signal strength, furnished rack hardware, remote mountable 1⁄2 wave antennas, and an extruded metal chassis.

ULX1 Bodypack and ULX2 Handheld Transmitters

- Backlit LCD display indicates group, channel setting and battery power level.
- They run for up to 8 hrs. on a single 9v battery
- 300´ operating range (normal conditions)



- -20dB pad switch lets you select 0dB for Voice/Low Output; 20dB for High Output
- Adjustable Audio Gain Control
- Control lockout

ULX/S4 Standard and ULX/P4 Professional Diversity Receivers



- Automatic Frequency Selection scans for open channels—provides a straight shot to a clear channel—allowing for fast setup
- Ultra-wide UHF band agility operates across multiple TV channels for clear operation anywhere
- Shure microprocessor-controlled Predictive Diversity anticipates and avoids sound dropouts for consistent reception
- Multi-function LCD display indicates group and channel setting, current UHF TV channel and remaining transmitter battery power
- Up to 20 compatible systems can be used simultaneously (depending on location)
- LED indicators confirm RF reception and level, as as transmitter audio level
- Detachable 1/4-wave antennas allows front mount capability
- Durably constructed 1/2-rack plastic chassis

ULX/P4 Step-up Features:

- Lockout settings (frequency and volume) protect from accidental changes
- Easy to use LCD interface and advanced front panel controls
- Remotable 1/2-wave antennas allow optimum antenna placement for enhanced signal reception
- Ruggedly constructed 1/2-rack metal chassis with supplied rackmount hardware

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ONE HOUR FREE PARKING
AT 349 W. 34th STREET (with purchase of $100 or more)

ULX STANDARD/ULX PROFESSIONAL

Headworn Mic Systems with ULX1 Bodypack and Receiver

ULXS14/30: WH30TQG Cardioid Condenser Headworn Mic and ULX/S

ULXP14/30: With WH30TQG Cardioid Condenser Headworn Mic

Guitar/Bass and Instrument Systems with ULX1 Bodypack and Receiver

ULXS14: WA302 1/4˝-mini 4-pin instrument cable and ULX/S Receiver

ULXS14/98H: WB98H/C Cardioid condenser clip-on mic and ULX/S Receiver

ULXP14: WA302 1/4˝-mini 4-pin instrument cable and ULX/P Receiver

ULXP14/98H: WB98H/C Cardioid condenser clip-on mic and ULX/P Receiver

Handheld Systems with ULX/S Receiver

ULXS24/58: SM58 Cardioid Dynamic Mic/ Transmitter

ULXS24/BETA58: Beta 58 Supercardioid Dynamic Mic/Transmitter

ULXS24/87: SM87 Supercardioid Condenser Mic Transmitter

ULXS24/BETA87A: Beta 87A Supercardioid Condenser Mic/Transmitter

Handheld Systems with ULX/P Receiver

ULXP24/58: SM58 Cardioid Dynamic Mic/ Transmitter

ULXP24/BETA58: Beta 58 Supercardioid Dynamic Mic/Transmitter

ULXP24/87: SM87 Supercardioid Condenser Mic Transmitter

ULXP24/BETA87A: Beta 87A Supercardioid Condenser Mic/Transmitter

ULXP24/BETA87C: Beta 87C Supercardioid Condenser Mic/Transmitter

Please Note:
Combo systems (two transmitters and one receiver) and dual system (two transmitters and two receivers) are also available.....Call

OPTIONAL WIDE BAND ACCESSORIES

Optional wide band accessories include the UA844 Antenna/Power Distribution Amplifier, which enables the use of up to four ULX receivers with just one pair of antennas; the UA830WB Line Amplifier for boosting the RF signal from remote mounted antennas; and the UA870WB Powered Directional Antenna.

UA506 Rackmount Kit For one ULX receiver **29.95**

UA507 Rackmount Kit For two ULX receivers.....**29.95**

UA600 Front Mount Antenna Cable Kit **39.95**

WA610 Universal Hard Carrying Case.....**78.95**

UA505 Remote Antenna Mounting Bracket Kit: Lets you mount a 1/2 wave receiver antenna away from the receiver or the equipment rack.....**27.50**

UA830 Active Remote Antenna Kit
A low-noise, in-line antenna amplifier, the UA830 is designed for remote-mounting antennas to a ULX receiver in a large installation, effectively shortening the distance from the transmitters. When an antenna is remote-mounted, the UA830 amplifies the received radio signals to compensate for insertion loss in the coax cable. A gain selector switch adjusts for longer lengths of coaxial cable (3dB for under 25´, 10dB for over 25´).



- ◆ Mountable on supplied desk-/wall-mount or on a microphone stand
- ◆ Isolates antennas from cable interactions

UA844 Wideband UHF Active Antenna and Power Distribution System
Splits a pair of antennas to multiple receivers allowing the expansion of wireless mic systems. It also amplifies RF signals to compensate for insertion loss that results from splitting signal power. A single UA844 system can support up to 4 wireless receivers. Multiple UA844 systems can be used in a tiered configuration until the maximum number of compatible frequencies is reached.....**341.95**



UA870WB UHF Wideband Active Directional Antenna
The UA870WB is an active UHF wireless directional antenna featuring a log periodic dipole array. When compared to an omnidirectional 1/2-wave antenna, the UA870WB's cardioid pattern offers enhanced reception of the desired coverage area and greater rejection of RF signals outside the coverage area.
On-board amplifier offers 3 or 10 dB gain to compensate for cable signal loss. 12v DC power for the antenna amp is provided by a Receiver or UA844 System. The UA870WB can be mounted on a mic stand, suspended from the ceiling, or mounted to a wall using the integrated swivel adapter bracket. It is also weather resistant for use outdoors. For best diversity performance, use two UA870WB antennas **239.95**

Lavalier Systems with ULX1 Bodypack Transmitter

With ULX/S Receiver

ULXS14/93:WL93 Omni

ULXS14/83: WL183 Omni

ULXS14/84: WL184 Supercardioid

ULXS14/85: With L185 Cardioid

With ULX/P Receiver

ULXS14/50: WL50 Subminiature Omni

ULXS14/51: WL51 Subminiature Cardioid

ULXS14/83: WL183 Omni

ULXS14/84: WL184 Supercardioid

ULXS14/85: L185 Cardioid

UC SERIES

100-Channel UHF Diversity Wireless Mic System



The Shure UC series gives you more of what you want most in a UHF wireless microphone system. Its flexibility and affordable price makes it the ideal choice for a wide range of applications including houses of worship, trade shows, schools, businesses and club installations. And while each UC system features more than 100 fully selectable frequencies and the ability to operate up to 16 systems simultaneously, its ease of operation eliminates the extended learning curve you'd expect from such a high-performance system.

100 user-selectable frequencies combined with Shure's tried and true MARCAD Diversity technology creates an exceptionally reliable wireless system, free of interference and drop outs—even in the most congested environments. The UC Series also incorporates microprocessor-controlled, Phase Locked Loop (PLL) circuitry for a clear, steady signal. It offers two UHF frequency ranges (692-716 and 782-806 MHz), 500´ minimum working range, and 45Hz to 15kHz frequency response.

UC4 Receiver

- 1/2 rack space design, includes hardware for single and dual rack mounting
- Tone Key Squelch Circuitry prevents unwanted noise from entering including the "pop" noise that occurs when the transmitter is turned on or off
- Noise squelch circuitry analyzes signal quality rather than signal strength, eliminating possible noise bursts
- Dual RF meters and 5-segment antenna LEDs indicate received signal strength at each antenna, making it easy to identify "dead spots" in the performing area. Also has a LED low battery transmitter warning
- Incorporates a two band EQ with 6dB cut/boost to match the sound of wired and wireless products in an installation. Helps compensate for off–axis lavalier mics or to adjust the sibilance of vocal microphones
- Adjustable high pass filter reduces low-frequency signals
- Group/Channel Display indicates the current group and channel settings. Facilitates easy set-up of multiple channel systems
- Audio meter lets you monitor levels to optimize the transmitter gain setting
- BNC connectors provide connection to the supplied antennas or to the coaxial cable used with a distribution amplifier
- Logic In/Out provides logic interface with external devices such as the Shure SCM810 automatic mixer and AMX or Crestron compatible control systems
- Unbalanced 1/4˝ and balanced XLR output connectors with mic/line slide switch

UC1 Body-Pack Transmitter



The UC-1 body-pack has a slim, low-profile design, and functions with the full line of Shure lavalier and instrument microphones as well as guitar cables. Supplied belt clip allows the transmitter to be worn on a belt, waistband, or guitar strap. Belt clip can be rotated 180°.

- Rugged plastic case design
- TA4F (4-pin mini connector) provides connection with a variety of lavalier, headworn and instrument mics or cables. Transmitter with LEMO 4-pin input connector (UC1L) is also available.
- Remote audio mute connector allows external audio and/or RF muting capability (requires the UA-101 Remote Mute Switch)
- Two position attenuator switch (0 for normal applications including voice and low output instruments, and –20dB for high output instruments such as electric guitars.)

UC1 and UC2 Transmitters Feature

- Combined power On/Off and mute switch saves time and hassle
- Eight hour battery life with single 9v battery
- Three-segment battery fuel gauge
- Audio gain control to accommodate various sound sources (singing or speaking)

UC2 Handheld Transmitter

- Grille protects the microphone cartridge and helps reduce breath sounds and wind noise
- Available with choice of mic capsules from some of Shure's most popular microphones.

Handheld Mic Systems with UC2 Handheld Transmitter and UC4 Receiver with two 1/4-Wave Antennas, Single and Dual Rackmount Kits, Swivel Adapter and WA555 Grip/Switch Cover for UC2



UC24/58: UC2/58 Handheld Transmitter with SM58 mic

UC24/87: UC2/87 Handheld Transmitter with SM87A mic

UC24/BETA58: UC2/BETA58 Handheld Transmitter with BETA58A mic

UC24/87A: UC2/BETA87 Handheld Transmitter with BETA87A mic

UC24/87C: UC2/BETA87C Handheld Transmitter with SM58

Headworn and Lavalier Mic Systems with UC1 Bodypack Transmitter and UC4 Receiver with two 1/4-Wave Antennas, and Single and Dual Rackmount Kits

UC14/16: WCM16 Hypercardioid Headworn Mic

UC14/20: WH20TQG Cardioid Headworn Mic

UC14/30: WH30TQG Cardioid Condenser Headworn Mic

UC14/50: WL50B Subminiature Omni Lavalier Mic

UC14/51: WL51B Subminiature Cardioid Condenser Lavalier Mic

UC14/83: WL183 Omnidirectional Lavalier Mic

UC14/84: WL184 Supercardioid Lavalier Mic

UC14/85: WL185 Cardioid Lavalier

UC14/93: WL93 Omnidirectional Micro Lavalier Mic

UC14/98H WB98H/C Clip-on condenser instrument mic w/shock mount

OPTIONAL UC SERIES ACCESSORIES

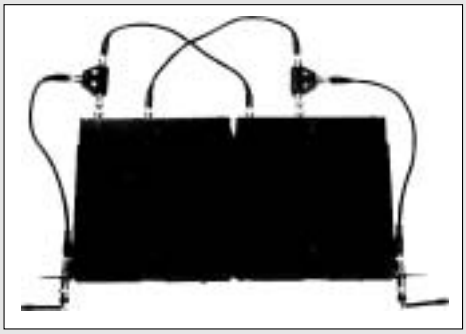
Although UC Wireless systems come with everything you need to operate them, you can enhance the system to new levels with a host of optional accessories:

UA-101 Remote Mute Switch:
The Shure UA101 Remote Mute Switch lets you remotely mute the audio and/or RF from the UC1 bodypack transmitter during presentations or performances. Ideal for vocalists, lecturers, clergy, referees — anyone who needs easy, instant ability to mute the bodypack transmitter **47.50**

UA-500 Remote Mount Antenna Kit: For mounting one 1/2 wave antenna.....**48.50**

UA-600 Front Mount Antenna Kit: Mounts antennas for the receiver **39.95**

UA-220 *Passive Antenna Splitter/Combiner*



The UA220 includes two adapters that can be used as one–input, two–output or two-input, one-output passive antenna splitter/combiners for UC Wireless systems. The UA220 Kit lets you split an incoming signal into two outgoing signals or combine two incoming signals into one signal. Each kit includes two UA220 Splitter/Combiners, four coaxial cables, and attaching hardware.

The UA220 is designed for use with Shure's UA400 1/4 wave UHF antennas, UA820A 1/2 wave antennas, and UA802/UA825/850 50 ohm low loss antenna cable.....**119.95**

UA-845 *Antenna Power/Distributor*



658.95

The UA845 is an amplified, UHF Antenna Distribution System designed to expand a wireless microphone system by splitting one pair of antennas to multiple UC4 wireless receivers. It also amplifies RF signals to compensate for insertion loss due to splitting signal power to multiple output connectors. Each UA845 allows up to four receivers to use the same antennas. CASCADE connectors allows connections to a fifth receiver or a second UA845. There are also power connectors for powering UC Wireless systems. The UA845 comes with hardware to front-mount the antennas, if desired.

- The UA845 maintains clean signals with low noise and minimal distortion
- Whenever a signal is split to multiple output ports, there is a loss in signal strength. The UA845 amplifies signals to compensate, ensuring a strong signal to the receivers.

UHF SERIES

Premium UHF Wireless Systems



Engineered for the most demanding touring environments and installations, Shure's UHF Wireless Series provides the ultimate in wireless technology, with exclusive characteristics, unique flexibility, reliability and an affordable price. Offers up to 1200 fully selectable, user programmable frequencies per frequency band. Individual frequency bands deliver up to 24 compatible systems per band. Using multiple frequency bands, over 75 compatible systems can be used simultaneously. The ideal solution for a wide range of high-end applications—from touring and installed sound to theater to broadcast and video production. But system versatility goes beyond multiple systems, interchangeable components, and a choice of single or dual channel diversity receivers. The UHF Series also offers computer control and monitoring of either single or multiple receivers. Using the optional UA111 Computer Interface Adapter gives you quick setup and monitoring of an individual receiver. Or monitor and control multiple receivers simultaneously from a remote PC with the UA888 networking interface system.

FEATURES

U1 Body-Pack Transmitter

The U1 body-pack is a compact transmitter that offers all the benefits of Shure handheld transmitters. The U1 functions with the full range of Shure lavalier, headworn, and instrument microphones, as well as guitar cables.

- Power-on LED
- 1/4- wave, efficient whip antenna
- TA4F (4-pin mini connector) or optional LEMO connector (U1L Body-Pack)

U1 and U2 Transmitters Both Feature

- Digitally-controlled programmable LCD display
- Combined power On/Off and mute switch
- Five-segment battery fuel gauge
- Power and frequency "lock" feature
- Adjustable audio gain control
- Durable all-metal housing
- 12 hours battery life

U2 Handheld Transmitter

U2 handheld transmitters are the top choice of live sound and install professionals worldwide with state-of-the-art UHF technology, five microphone options, and accessories included.

- Grille protects the mic cartridge and helps reduce breath sounds and wind noise
- 1/4-wave, efficient helical antenna



UHF transmitters

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UHF SERIES

Unparalleled Performance

- Choose from an assortment of legendary handheld mics, including the Beta 87A, Beta 87C and Beta 58A, known worldwide as the mics of choice for live concert sound. Lavalier and headworn designs are also available for a variety of applications
- Exclusive MARCAD diversity circuitry improves reception and eliminates dropouts
- Tone Key Squelch keeps unwanted signals and noise from entering the system
- Patented Power Defeat protection lets you "lock" transmitter power on. System controls prevent accidental transmitter and receiver program changes
- Reliable and easy to use; batteries last a full 12 hours
- Receiver's rugged front plate resists abuse encountered in transport

Unprecedented Compatibility

- Four operating bands are available, letting you tune across 108 MHz of bandwidth
- 192 selectable frequencies in 125 kHz increments in UA and UB frequency bands
- 1200 selectable frequencies in 25 kHz increments (M4 and J4 frequencies) allow you to "squeeze out" additional channels
- Pre-programmed compatible frequency groups assure an optimal number of simultaneous channels in any location. Up to 20 (UA and UB) or 24 channels per frequency band (J4 and M4) can be achieved. With all four operating bands in use, up to 78 compatible channels can be achieved.

Unbelievably Informative

- No other UHF system gives you more ways to monitor vital system functions:

– Digitally-controlled, programmable transmitter and receiver LCDs indicate group/channel setting

– Receiver display shows frequency, user name, squelch setting and menu "lock out" function, LED indicators provide RF and audio level monitoring

– Five-segment battery power display on the transmitter and three-segment remote battery indicator on the receiver show how much transmitter battery life remains

U4 Receivers

For added flexibility and performance, the system offers two receiver choices; the U4S with a single receiver design, and the U4D with, a dual-channel receiver. Both feature Shure's exclusive MARCAD Diversity circuitry, 5-segment audio meter(s), two half-wave flexible rubber antennas and 5-segment antenna meters, 3-segment transmitter battery fuel gauge, programmable multi-function LCD, tone key squelch, mic/line switch, full rack design, internal power supply, headphone jack, rugged metal chassis, network connection, and switchable 120/220 power supply. They also include front mount antenna cables and accessories.



U4S



U4D

Programmable, multi-function LCD display to set and view:

- Actual frequency in MHz
- Group/channel setting
- Performer name
- Squelch setting
- Menu lock-out status
- UHF TV channel
- Transmitter battery fuel gauge

Optional Accessories

UA820A: 1/2 Wave Dipole Receiver Antenna (774-862MHz)
UA820B: 1/2 Wave Dipole Receiver Antenna (692-746 MHz)
UA400: 1/4 Wave Dipole Receiver Antenna (774-862 MHz)
UA400B: 1/4 Wave Dipole Receiver Antenna (620-722 MHz)
UA820C: 1/2Wave Dipole Receiver Antenna (662-698 MHz)
UA820D: 1/2Wave Dipole Receiver Antenna (554-590MHz)
UA870A: Active Directional Antenna (774-863MHz)
UA870UB: Active Directional Antenna (692-716 MHz)



UA870

UA870WB: Wideband UHF Active Directional Antenna (500-900 MHz)
UA500: Remote Mount Antenna Kit
UA600: Front Mount Antenna Kit
UA830: In-Line Antenna Amplifier
UA845: Antenna/Power Distribution System



UA845

SHURE

UA888



Networking Interface System

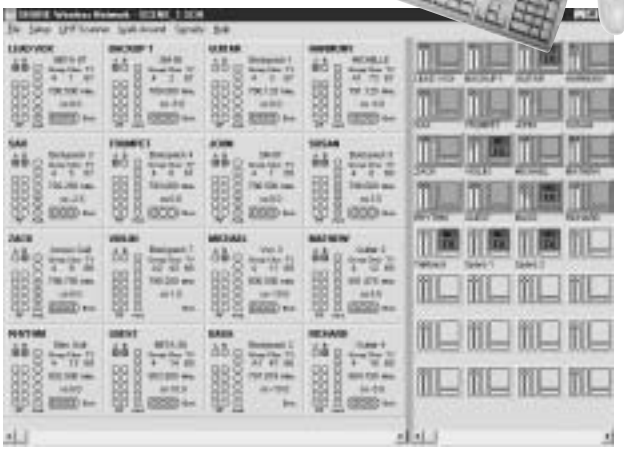
Now you can have complete knowledge of the up-to-the-second status of your multi-channel Shure UHF wireless microphone system—and the power that goes with it—with the UA888 Networking Interface System. The UA888 is an affordable computer interface accessory that allows you to monitor and control the functions of up to 32 dual channel receivers from a remote site using a PC. Using the supplied ShureLink cable, monitoring of up to 64 channels can be achieved by connecting multiple UA888 units.

Ideal for installed sound applications requiring multiple systems such as theater, houses of worship, and for touring sound and audio/video rental.



The UA888 continually monitors an array of functions, including RF level, audio level, A/B antenna diversity and transmitter battery level. What's more, the system provides monitoring and control capability of group/channel selection, frequency in MHz, squelch, user name and lock/unlock status.





- Includes rack-mountable interface module, Windows-based software, and connecting cables for using with the U4S or U4D receivers.
- Full monitoring and control of all vital system parameters with complete real-time information. Displays all functions on a single computer screen.
- Sophisticated troubleshooting capabilities include a "walk-around" RF plotter that measures RF signal strength and identifies "dead" spots in the performing area, and a highly accurate frequency scanner that helps minimize RF interference by scanning the airwaves and identifying present frequencies
- Scene setup and implementation, saves and instantly recalls multiple receiver settings. A password option prevents unauthorized changes to settings.

UA111 *Computer Interface Adapter*



Using the same software as the UA888 , the UA111 is a computer interface that allows you to monitor and control one U4D or U4S receiver (monitors and controls up to two UHF wireless microphones when used with the U4D dual receiver).Use in touring sound, installed sound, and audio/video rental applications. Has all the of the software features of the UA888 Networking System, including:

- UHF frequency scanner
- Walk around RF plotter
- Password protection
- Scene setup
- Squelch control
- Indicators for transmitter battery level, frequency in MHz, group/channel, audio level, A/B antenna diversity, RF levels

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ONE HOUR FREE PARKING
AT 349 W. 34th STREET (with purchase of $100 or more)

SPECIFICATION CHART

RECEIVER SPECIFICATIONS

	T3	T88	T4A	UT4A	LX3	LX4	LX88	ULXS4	ULXP4	UC4	U4S/U4D
# of compatible systems	6	6	6	8	12	12	3 (dual)	20	20	16	20-24
RF/audio metering	■	■	■	■	■	■	■	■	■	■	■
Tone key squelch								■	■	■	■
Frequency synthesized								■	■	■	■
Frequency agile								■	■	■	■
Programmable LCD								■	■		■
Headphone monitor											■
Balanced XLR output			■	■		■		■	■	■	■
Unbalanced 1/4" output	■	■	■	■	■	■	■	■	■	■	■
Mic/line switchable output				■		■		■	■	■	■
Rack mountable receiver			■	■	■	■	■	■	■	■	■
1/2-rack size receiver					■	■		■	■	■	
Front/remote antenna					■	■	■	■	■	■	■
Dual channel available		■			■	■	■	■	■	■	■
Transmitter battery fuel gauge					■	■	■	■	■	■	■
In-Line power supply	■	■	■	■	■	■	■	■	■	■	
Internal switching power											■
Networking option											■
User adjustable EQ										■	
TTL logic control capacity										■	
LCD for group/channel status								■	■	■	■
Noise-sensing squelch	■	■	■	■	■	■	■	■	■	■	■
CE mark approval	■	■	■	■	■	■	■	■	■	■	■

TRANSMITTER SPECIFICATIONS

	BODYPACK TRANSMITTERS						HANDHELD TRANSMITTERS					
	T1	UT1	LX1	ULX1	UC1	U1	T2	UT2	LX2	ULX2	UC2	U2
Attenuator switch		■		■	■	■	NA	NA	NA	NA	NA	NA
Lavalier mic option	■	■	■	■	■	■	NA	NA	NA	NA	NA	NA
Headworn mic option	■	■	■	■	■	■	NA	NA	NA	NA	NA	NA
Instrument mic option	■	■	■	■	■	■	NA	NA	NA	NA	NA	NA
"LEMO" connector option					■	■	NA	NA	NA	NA	NA	NA
Battery light	■	■					■	■				
Battery fuel gauge			3 segment	3 segment	3 segment	5 segment			3 segment	3 segment	3 segment	5 segment
Batteries (battery/hours)	9V/18	9V/18	9V/18	9V/8	9V/8	2 AA/12	9V/18	9V/18	9V/18	9V/8	9V/8	2 AA/12
Separate mute switch	■	■	■				■	■	■			
On/off/mic mute switch				■	■	■				■	■	■
LCD for group/channel status				■		■				■		■
Remote mic mute					■						■	
Power/frequency control lock				■		■				■		■
Beta 58A	NA	NA	NA	NA	NA	NA		■	■	■	■	■
Beta 87A & Beta 87C	NA	NA	NA	NA	NA	NA			■	■	■	■

WIRELESS MICROPHONES

UHF Synthesized Wireless Microphone System



Sony's UHF Synthesized Wireless Microphone System, consisting of handheld and bodypack transmitters and 7 different receivers, is widely recognized as the outstanding wireless mic system for professional applications. Operating in the 800 MHz UHF band range and equipped with a PLL (Phase Locked Loop) system, they provide up to 94 channels, space diversity reception and LCD indicators. Reliable and sophisticated circuit technology assure low noise, wide dynamic range, and extremely stable signal transmission and reception. An affordable system with superb audio quality and easy handling, the Sony UHF Synthesized system is the ultimate in convenience for applications at broadcasting stations, film production facilities, and in ENG/EFP work.

FEATURES

PLL Synthesized System

A PLL (Phase Locked Loop) synthesized system controls the transmission and reception frequencies of the system. Employed both in transmitters and receivers, these PLL controlled systems provide highly stable, selectable frequencies to be generated in increments of 125 kHz. Users may choose from 94 frequencies on all A-series models and from 188 frequencies on B-series models.

Space Diversity Reception System

A space diversity reception system is used to eliminate noise and signal dropouts, thus ensuring stable reception. Dual antenna inputs and reception circuits incorporated in the diversity system receive signals over two different paths and select the stronger signal as the output. This switching operation is undetectable on the audio output of the receivers. An additional benefit of this system is stable reception and cleaner audio when working at extended operating distances.

Pre-programmed Frequency Groups

Wireless channels are conveniently preset in the CPU of each tuner for use in a multi-channel operation at one location. These are optimum combinations of calculated and practically tested intermodulation-free frequencies. These frequencies are arranged in groups, with each group pre-programmed to allow inter-modulation free operation. The use of preset channels makes it easy to choose the correct frequencies for multi-channel operation, and unlike random selection of channels, prevents interference such as beat noise in multi-channel operation and guarantees better unit isolation.

Advanced Filtering

The Sony wireless units all employ Helical, Ceramic and/or SAW (Surface Acoustic Wave) filters as appropriate. This offers stable reception and excellent audio quality, and also enables the unique miniaturization seen in most of the Sony products.

Wide Dynamic Range and Low Noise

A compander (compressor/expander) system is employed for optimum transmission over a wide dynamic range with low noise and interference. The time constants for attack and release times are carefully chosen in the various transmitters to match the application, providing minimized noise levels and smooth and excellent audio quality.

Informative, Easy-to-Read LCD

Most receivers and transmitters feature a built-in, easy-to-read LCD, which provides extensive status information. All receivers will display the diversity RF and AF signal strength, frequency and channel groups chosen, and battery status of the transmitter. Most transmitters do in essence have the same information available in the display; such as AF input level, and RF output, battery status, current channel number, selected frequency, input attenuation setting, and accumulated battery operating time.

WIRELESS MICROPHONES

Tone Squelch Circuitry

To ensure accurate audio reproduction, the UHF Synthesized System is equipped with tone squelch circuitry control. Here's how it works: The wireless microphone transmitters transmit a 32 kHz tone signal along with the program audio signal. In the receivers, a squelch circuit is present, and the program signal is only output when the tone signal has been correctly received. This squelch feature is designed to eliminate undesired signals and noise from other equipment and eliminates RF noise and popping when the transmitter is powered off.

Consistent RF Power Output

DC-DC convert circuit built into the power supply section of microphones/transmitters provides consistent output power over the life of the batteries.

Remote Battery Alarm

For added assurance of continued operation, you can monitor the battery reserves of handheld, body pack and plug-on transmitters. Remaining battery life is indicated on both the transmitter and receiver, with an additional indication approximately one hour before the battery goes dead. On the receiver, the battery indicator on the LCD and LED displays also flash, helping to avoid the chance of battery failure at a critical moment.

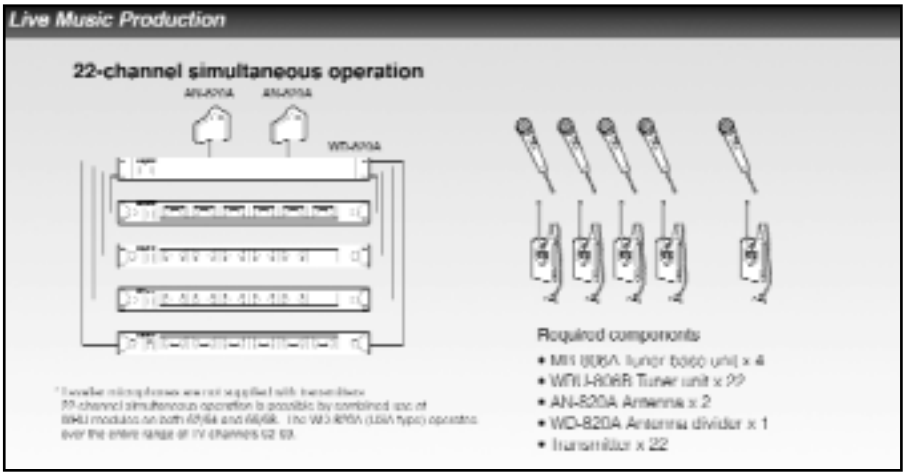
Memory Back-Up

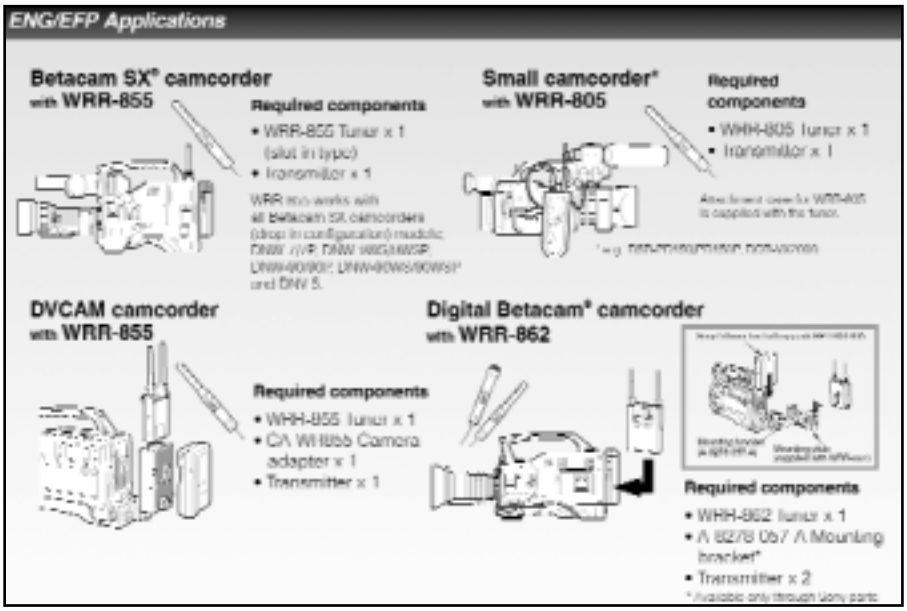
Each unit has a convenient memory back-up capability. When the Power switch is turned on, the previous channel setting is automatically recalled and displayed.

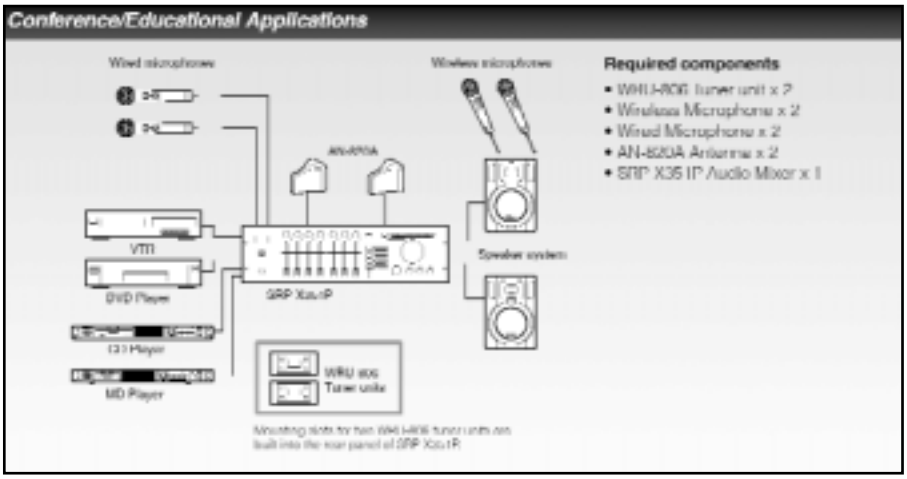
System Versatility

In recent years, Sony has enhanced its line-up to fulfill the ever-increasing needs for wireless operations. Offering a broad range of choices, this family presents a solution to virtually any wireless mic application from the very simple to the most sophisticated.

Such applications include ENG and EFP location, conference and entertainment, TV studio production, theater, and live performance to name just a few. A variety of options are also available so the systems can be made to suit different operational needs.







SONY

UHF SYNTHESIZED WIRELESS BODY-PACK TRANSMITTERS

WRT-805B



Although a low-cost transmitter, the lightweight WRT-805B offers 188 PLL-synthesized channels and a host of high-end features.

- Compact, lightweight, tapered bodypack
- 10mW RF output power
- Easy-to-read LCD shows channel number, attenuator, RF and AF condition, battery status and accumulated operating time
- 5/32″ 3-pole locking mini microphone jack
- Switchable -40/-60 dB input level and phase switch for guitar and lav mic
- Up to 6 hours of continuous operation with a single AA battery
- Low-battery alarm transmitted to compatible receivers

WRT-822B



The WRT-822B provides stable RF performance and high audio quality in a sturdy yet lightweight ergonomic package. It is suitable for use with any Sony UHF receiver and all Sony lavalier microphones, making it ideal for ENG, broadcast or theatre work.

- Compact, lightweight, magnesium alloy body - only 2½″ x 4⅛″ x 11/16″.
- 20mW RF output power
- Easy-to-read LCD shows channel number, attenuator, RF and AF condition, battery status and accumulated operating time
- Audio attenuation setting from 0 dB to - 21 dB in 3 dB steps
- Eight hours continuous operation with 2 AA batteries
- Low-battery alarm transmitted to compatible receivers
- Includes low profile 1/4 wave length wire antenna and leather case

WRT-8B

Compact and lightweight, the WRT-8B Body-pack transmitter has all the essential qualities for use in fast-moving TV and theater musical productions. Inside its tough, die-cast magnesium body is a highly stable, low noise transmitter providing superb audio quality. It operates for up to 13 hours on two AA-batteries, and features switchable AF input (Mic/Line) and RF output levels (10mW/50mW), variable input attenuator, and easy-to-read indicators. As well as theater and studio use, the WRT-8B has the functionality and rugged construction that makes it ideal for the demanding environments of news gathering, documentary production and sports events.



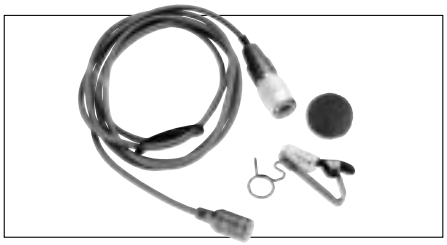
- Rugged constructed of die-cast magnesium
- Accepts professional lavalier microphones fitted with an SMC9-4P connector, such as the Sony ECM-77BC, ECM-44BC, ECM-166BC lavalier microphones
- Switchable Mic/Line input level and variable attenuator
- Easy-to-read LCD indicates operating channel/frequency, AF input and RF output levels, transmitter battery status and accumulated operating time.
- A red LED indicator also flashes when the AF level exceeds a designated level.
- Selectable output levels: 10mW for multi-channel operations such as in theater and studio productions; 50mW for the long-distance transmissions typically needed for sports and news coverage.
- Runs for up to 13 hour on two AA-batteries at 10mW, up to 6 hours at 50mW output
- Transmits a 32 kHz tone signal along with the program audio signal. This tone is recognized by the squelch circuit in Sony receivers so only the program signal is output when this tone signal is received. This eliminates the output of undesired signals and noise from other equipment, as well as RF noise and popping when the transmitter is powered on or off.
- Easily removable antenna with SMA connector
- As well as displaying battery status via the LCD indicator, the WRT-8B transmits a “low battery warning” which can be monitored by most Sony receivers. In these receivers, a red LED starts flashing one hour before the transmitter’s battery will be exhausted.

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EQUIPMENT LEASING AVAILABLE

ELECTRET CONDENSER LAVALIER MICS

FOR SONY BELTPACK TRANSMITTERS



ECM-44

Omni-Directional Microphone

- Superior sound quality
- Frequency response: 40 Hz to 15 kHz
- Sensitivity: -40 dB SPL (10 mV)*
- Microphone head: 11/32″ x 19/32″; 0.07 oz.
- Supplied with holder clip (single/horizontal type), Urethane wind screen, case

ECM-44BC For WRT-822B & WRT-8B..**104.95**

ECM-44BMP For use with WRT-805B.....**99.95**

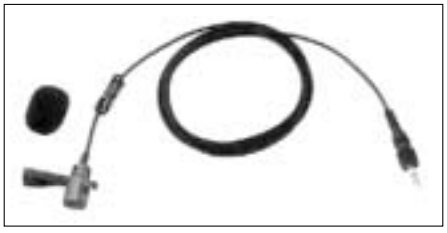


ECM-66

Uni-Directional Microphone

- Designed for instrumental applications
- Frequency response: 70 Hz to 14 kHz
- Sensitivity: -36.5 dB (15 mV)*
- Mic. head: 7/16 x 31/32″; approximately 0.3 oz.
- Includes holder clip, urethane wind screen

ECM-66BC For WRT-822B & WRT-8B...**237.95**



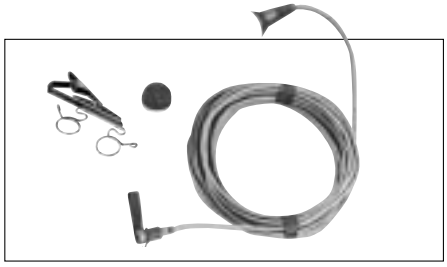
ECM-166

Uni-Directional Microphone

- Resists howling and rejects indirect sound
- Ideal for installed sound and contractor applications such as lectures & conferences
- Frequency response: 100 Hz to 10 kHz
- Sensitivity: -45 dB (5.6 mV)*
- Microphone head: 1/2″ x 15/16″; 0.12 oz.
- Includes urethane wind screen, holder clip

ECM-166BC For WRT-822B & WRT-8B..**104.95**

ECM-166BMP For the WRT-805B.....**99.95**



ECM-77

High-Performance, Ultra-Miniature, Omni-Directional Microphone

- Frequency response: 40 Hz to 20 kHz
- Sensitivity: -39.0 dB (11.2 mV)*
- Mic. head: 1/4″ x 1/2″; approximately 0.05 oz.
- Supplied with 2 holder clips (single/horizontal type and single/vertical type), Metal-mesh wind screen, microphone case

ECM-77BC For WRT-822B & WRT-8B..**253.95**

ECM-77BMP For use with WRT-805B...**241.95**



ECM-55

Uni-Directional Microphone

- Frequency response tailored for enhanced presence and improved voice quality
- Frequency response: 30 Hz to 18 kHz
- Sensitivity: -38.5 dB (11.9 mV)*
- Mic. head: 7/16″ x 27/32″; approximately 0.2 oz.
- Includes holder clip, metal-mesh wind screen

ECM-55BC For WRT-822B & WRT-8B...**224.95**

ECM-55BPT 9.8′ cable without battery or connector (pig tail connection).....**189.95**



ECM-310

Lightweight, Wide-Cardioid Headset-style Microphone

- Provides crisp and clear sound – isolates desired sound from surrounding ambience
- Adjustable hinge and goose-neck
- Frequency response: 70 Hz to 12 kHz
- Sensitivity: -44 dB (6.3 mV) ±3 dB*
- Includes urethane wind screen

ECM-310BC For WRT-822B & WRT-8B.....**97.95**

ECM-310BMP For the WRT-805B.....**86.95**

BC models come with a 1.2m cable and a SMC9-4P connector. BMP models have a 3-pole mini-plug with a lock mechanism for secure cable clamping.
**(0 dB = 1V/Pa, at 1 kHz)*

MICROPHONE KITS							
	ECM-166BC	ECM-44BC	ECM-55BC	ECM-66BC	ECM-77BC	EC-15CF	K1161
WRT-805B Mic Kits	Call	Call	Call	Call	Call	Call	Call
WRT-822B Mic Kits	Call	Call	Call	Call	Call	Call	Call
WRT-8B Mic Kits	Call	Call	Call	Call	Call	Call	Call

CABLES

EC15CF XLR adapter cable for BC Series transmitters.....**59.95**

GC0.7MP 1/4″ guitar cable for WRT-805B transmitter.....**44.95**

K1161 1/4″ guitar cable for BC Series transmitters.....**89.95**

WRT-847B

UHF Synthesized Transmitter Unit

The WRT-847 UHF Synthesized Transmitter Unit accepts five optional microphone capsules, providing a choice of characteristics to suit a range of different applications. Mic capsules are available individually. (One head is required for it for function.). The CU-F780, CU-G780 and CU-E700 capsules are designed for vocal and speech applications in broadcasting, live music performances and concert halls, while the CU-E672 and CU-F117 options are suitable for interviews in news gathering and field productions. Other features include PLL frequency synthesized circuitry for highly stable operation and an easy-to-read LCD screen.



- Audio compander time constant is switchable to suit different capsules
- Selectable RF output level: 10 mW for simultaneous multi-channel operation and 50 mW for long working distance
- Audio gain and attenuation setting from +9 dB to -12 dB in 3 dB steps
- Lockable power switch to avoid accidental power off
- Battery status information is transmitted to the receiver and shown on an LCD for additional assurance of continuous operation
- Two AA-size alkaline batteries provide up to eight hours of continuous operation
- Easy-to-read LCD with back light indicates extensive information on operating conditions such as channel number, wireless channel frequency in MHz, audio input level, compander time constant, battery status and accumulated battery operating time.

CU-F780
Dynamic Supercardioid Mic Capsule



- Ideal for vocal applications including live music performance
- Dynamic element for extreme durability and high SPL handling
- Super cardioid polar pattern minimizes off-axis pickup and feedback
- Uses the same high quality edgewise winding Copper Clad Aluminum Wire voice coil used in Sony's acclaimed F-780 wired mic.
- Wide 50 Hz to 18 kHz frequency response
- Built-in windscreen
- Dimensions: ø 51 x 90 mm (6.3 oz)

CU-G780
Dynamic Cardioid Mic Capsule

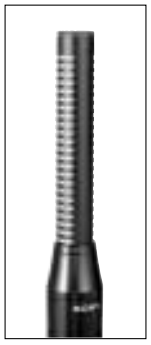


- Ideal for live vocal applications, especially those requiring increased immunity to feedback
- Dynamic element for extreme durability and high SPL handling
- Super cardioid polar pattern minimizes off-axis pickup and feedback
- Designed to cope with high sound pressure level vocals and to provide outstanding feedback rejection
- Wide 50 Hz to 20 kHz frequency response
- Built-in windscreen
- Dimensions: ø 51 x 90 mm (6.3 oz)

CU-E700
Supercardioid Condenser Mic Capsule



- Designed for critical vocal and speech applications
- Super cardioid polar pattern minimizes off-axis pickup and sensitivity to acoustic feedback
- Built-in windscreen
- Electret element for smooth, clear sound, excellent transient response
- Wide frequency response (50 Hz to 18 kHz) for natural sound reproduction
- Max. SPL: 150 dB
- Dimensions: ø 51 x 98 mm (6.0 oz)



CU-E672
Hypercardioid Condenser Mic Capsule

- A shotgun mic capsule (similar to the Sony ECM-672). Ideal for a wide variety of applications in news-gathering, sports and interviews
- Frequency response: 50 Hz to 16 kHz
- Dimensions: ø 37 x 172 mm (5.3 oz)
- Includes urethane wind screen
- Extreme directional response minimizes off-axis pickup
- Electret element for clear sound, excellent transient response and wide frequency response



CU-F117
Omnidirectional Dynamic Mic Capsule

- The perfect mic for various interview applications, the CU-F117 is an omni-directional dynamic microphone capsule designed to help reject wind noise and popping
- Dynamic element for extreme durability and high SPL handling
- Ideal for use in interviews and ENG applications
- Built-in windscreen reduces wind noise & popping
- Frequency response: 50 Hz to 15 kHz
- Dimensions: ø 44 x 105 mm (6.0 oz)

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HANDHELD TRANSMITTERS

WRT-800A



- Wide-cardioid electret condenser capsule for outstanding sensitivity and response
- 12 hours of continuous operation with a single AA battery
- Selectable RF output of either 2.5 mW or 10 mW depending on transmission distance and operating environment.
- Ergonomic grip and miniature low profile helical antenna
- Low-Battery alarm works with compatible receivers
- Smoothly tapered body with compact lightweight design
- Includes easily detachable mic holder

WRT-807A



- High sound quality for vocals - powerful, crisp and clean sound as well as presence in the low end and mid frequency range
- Dynamic microphone capsule, same as used in the Sony F-780 professional vocal microphone
- LCD for display of channels, attenuator and accumulated hours
- Up to 5 hours of continuous use with a single AA battery
- Battery alarm transmitted to compatible receivers
- Newly-developed lockable power switch to prevent accidental operation

WRT-808A



- RF power output: 10mW
- Plug-In transmitter works with any dynamic microphone with an XLR connector
- 94 channel frequency agile
- Runs on two AA batteries
- Battery level indicator
- Level control
- AF/Peak indicator shows condition of audio input level
- Low-Battery alarm works with compatible receivers

WRT-867A/X



- Uni-directional dynamic mic capsule incorporates a high quality edgewise-winding voice coil with lightweight CCAW (copper clad aluminum wire) and AlNiCo magnet for powerful, crisp and clear sound as well as presence in the middle and high frequency range.
- Four hours of continuous operation at 10 mW RF output level with one AA battery.
- Ideal for critical vocal applications in broadcast and film production.

	WRT-800A	WRT-807	WRT-808A	WRT-867A/X
Carrier Frequency	794 MHz to 806 MHz (68 only) 94 channels selectable	770 MHz to 782 MHz (64) 782 MHz to 794 MHz (66) 794 MHz to 806 MHz (68)	770 MHz to 782 MHz (64) 782 MHz to 794 MHz (66) 794 MHz to 806 MHz (68)	794 MHz to 806 MHz (68)
RF Power Output	5 mW (50 Ω load)	10 mW (50 Ω load)	10 mW (50 Ω load)	10 mW (50 Ω load)
Antenna	1/4 wave length wire antenna	1/4 wave length wire antenna	Internal antenna	Helical antenna
Frequency response	100 Hz to 15 kHz	50 Hz to 15 kHz	70 Hz to 15 kHz	40 Hz to 20 kHz
Signal-to-Noise Ratio	> 57 dB (A-weighted)	> 60 dB (A-weighted)	> 68 dB (A-weighted)	> 60 dB (A-weighted)
Attenuator Range (pad)	—	21 dB, variable in 3 dB steps	0 dB to 50 dB (continuous)	0, 6 dB, 12 dB (switchable)
Max. Input SPL	128 dB SPL (1 kHz)	151 dB SPL (1 kHz)	N/A	130 dB SPL
Directivity	Uni-directional	Uni-directional	N/A	Uni-directional
Battery Life	Approx. 12 hours; (1) AA	Approx. 5 hours; (1) AA	Approx. 4 hours; (2) AA	Approx. 4 hours; (1) AA
Dimensions	2¼ x 8⅝″	2⅛ x 9⅜″	1⅝ x 1⅝ x 4¾″	2⅛ x 7¼″
Weight with Battery	7.1 oz.	15.5 oz.	6.18 oz.	11.5 oz.
Supplied Accessories	Mic holder, Metal screw adapter	Mic holder, Metal screw adapter	Soft Case	Mic holder, stand adapter

CAMERA MOUNTABLE UHF SYNTHESIZED RECEIVERS

WRR-805A



Loaded with features and ready to perform, the compact WRR-805A features PLL frequency synthesized circuitry for highly stable operation and whip antenna that transmits battery status to compatible receivers. An easy-to-read LCD screen and LED's indicates all the information a user might need.

- Compact and lightweight, the WRR-805A works with any camcorder that has a stereo mini jack for Mic input.
- 94 selectable frequency synthesized UHF channels
- RF, battery alarm indication by both LCD and LED to double check operating condition
- Switchable RF muting (squelch); ON (15 dBμ) or OFF.
- LCD for display of operating channel, AF, RF, accumulated operating hours, the tuner's battery status
- 3.5 mm diameter, balanced line output connector with a lock mechanism for secure camera cable clamping.
- Six hours of operation (with two AA alkaline batteries)
- Convenient headphone monitoring with volume control
- Flexible mounting position and direction ensures secure operation when mounting on DV, DVCAM, Betacam SX, and Digital Betacam camcorders. Optional mounting bracket required for rear-mounting of up to 2 units.)



- Dual diversity reception for elimination of dropouts and outstanding reception.
- Dual-powered, they can receive power from a Sony camcorder with a DC output (via supplied DC cable) or 4 AA batteries for up to 8 hours of operation.

WRR-861B/WRR-862B

The WRR-861B (single-channel) and WRR-862B (dual-channel) are cost-effective portable space-diversity receivers designed to deliver high-quality/high-reliability reception in broadcast news and EFP applications. Dual-powered with a lightweight but rugged body made from die-cast magnesium, they are designed for mounting onto Sony professional camcorders with the supplied attachment kit/case, as well as for a variety of other portable applications such as field sound mixing in combination with a portable mixer. High-end features include easy-to-read LCD/LED indicators, selectable RF squelch function, transmitter battery alarm and headphone monitoring function.

- LED indicators display RF input status and transmitter battery alarm. Easy-to read LCD displays operating channel/frequency, AF output/RF input levels, receiver battery status and accumulated battery operating time.
- Frequencies are arranged pre-programmed groups to allow intermodulation-free operation. Also makes it easy to choose the correct frequencies for multi-channel operation.
- Headphone jack for monitoring the output (with volume control)
- To eliminate ambient noise, a selectable RF squelch function allows audio to be muted when the RF signal decreases to a certain level (can be set to 5 dBu, 10 dBu, 15dBu).
- Battery alarm function monitors their battery level as well a transmitter's. LED flashes one hour before transmitter's battery goes dead.
- Rugged, compact and lightweight, the WRR-861B (10 oz.) and WRR-862B (14 oz.) are easy to mount with the supplied attachment kit and case.

WRR-855B



A broadcast quality true diversity receiver designed for use with Sony's SX cameras, the unobtrusive WRR-855B takes its power and supplies audio directly to the Betacam's SX's plug-in slot, or can be used with the BTA-801 adapter to provide external connections on 4-pin DC power cable and XLR audio. For reassurance on location, the WRR-855B also includes an RF scan function, allowing you to identify if any RF noise is present. The Pilot Tone Squelch, Low RF Muting and Excessive White Noise Muting functions work together to ensure optimum noise free and seamless performance. Preprogrammed groups of intermodulation free frequencies make it easy to select an used channel on busy shoots

UHF SYNTHESIZED DIVERSITY RECEIVERS

- Water resistant structure
- Muting RF Level: 10dBu or Off
- Space diversity tuner for camcorder
- Squelch circuitry operated by ultrasonic tone for positive transmitter reception
- The WRR-855A has a 15-pin connector for audio output to a Betacam SX camcorder
- Can be mounted on Sony camcorders with the optional BTA-801 mounting adapter. The camcorder can power the WRR-855A when you use the DC cable supplied with the BTA-801.
- The audio is recorded to either (or both) track 1 or 2 on an SX camcorder, either with manual or (AGC) automatic gain control. In addition, audio is always recorded onto track 3 with AGC regardless of any other settings on the camera.
- Compact and lightweight— only 11 oz

BTA-801 mounting adapter **399.95**

WRR-802A

Single Channel Rackmount Receiver



Half size 19˝ rack width and 1U height, single-channel space diversity tuner with LCD display of channel and battery alarm. Output volume control. Balanced output connectors with XLR & TRS phone jack. Front mount antennas.

- Two front mounted antennas allow the receiver to constantly compare signals and to choose the strongest one
- Mic/line selectable XLR and 1/4˝ TRS balanced output
- Convenient front panel located volume control knob for quick level adjustments
- AF, RF and battery alarm indication by both LED and LCD to double check operating condition (channel indication by LCD only).
- Also monitors the battery power of any transmitter that transmits battery power (includes WRT-800A & 805A transmitters
- Half 19˝ rack width and 1U high

MB-806A and WRU-806B

Tuner Base Unit and UHF Synthesized Tuner Unit



The MB-806A is a tuner base unit featuring slots for six WRU-806B diversity receiver modules, with an internal antenna divider for six channels of diversity operation in a single rack space. Using the optional WD-820A Antenna Divider, you can go even further, with up to 12 wireless mics operating simultaneously on a single pair of AN-820A active antennas! The MB-806A also features a convenient autochannel assignment function. Once an operating group is set on one of the installed WRU-806B receiver, channels within the same group are automatically assigned, and unusable channels for the additional receivers will be automatically detected and skipped. This enables instant and automatic multi-channel setting of the entire rack of receivers.

- Modular 19˝ rack mount tuner frame (1RU)
- Six XLR-balanced output connectors plus an XLR balanced mix output
- Selectable mic/line (-58dBm/-20dBm) output level switch
- Accommodates up to 6 WRU-806B diversity receiver modules for up to six simultaneous channels of operation. With the optional WD-820A Antenna Divider, up to 3 systems (18 mics) can be operated simultaneously
- Auto channel assignment for extra tuner modules for instant programming of interference-free multi-channel operation
- AF, RF and battery alarm indication by both LED and LCD to double check operating condition (channel indication by LCD only)
- RF input attenuator (10db / 0dB) switch
- Supplied passive antennas for rear mounting (with provision for front mounting
- Built-in antenna divider with 9v DC power

WRU-806B

UHF Tuner Module



MB-806A.....**Call**
WRU-806B.....**Call**

SONY

MB-8N and WRU-8N

Wireless Mic Tuner Base Unit and UHF Synthesized Tuner Unit(s)



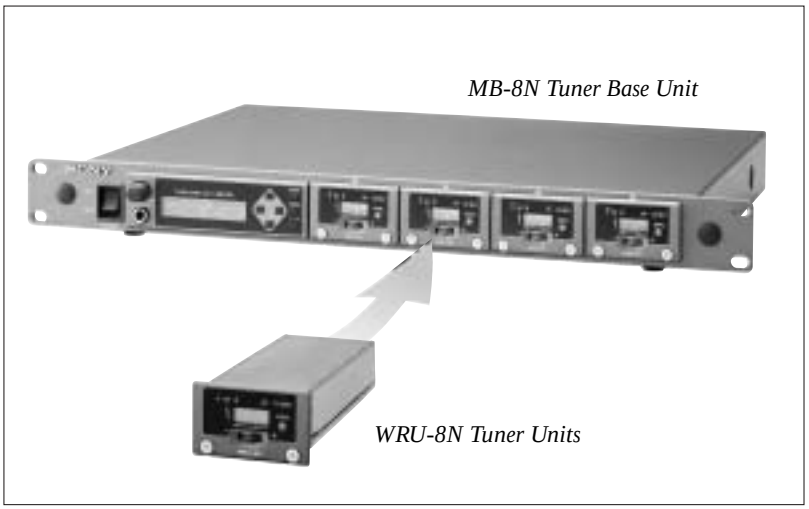
Incorporating cutting-edge wireless microphone technology, the breakthrough MB-8N Tuner Base Unit together with the WRU-8N UHF Synthesized Tuner Unit set new standards for performance, flexibility and functionality in a wireless mic system. Although only 1U high, the MB-8N tuner base unit accommodates up to four WRU-8N tuner units, and it includes a built-in antenna divider making it easy to daisy-chain up to four of these base units to form a 16-channel system. The WRU-8N offers exceptional standards of RF performance. It features space diversity reception, and a wide dynamic range with low levels of noise and distortion. But what sets this system apart from any other, is the level of remote operation and monitoring the MB-8N/WRU-8N combination provides. Over standard Ethernet connections, a complete system can be set up and controlled from a PC using the supplied GUI software. This includes the store and recall of different channel configurations, a feature that is particularly useful for location use and equipment rental organizations. A separate audio monitoring feature is included to allow the individual outputs of the tuner units to be checked.

MB-8N Tuner Base Unit

- Easy to install in both fixed and mobile applications, the MB-8N Base houses up to four WRU-8N tuner units in a compact 1U high, 19˝ rack mountable design. This allows flexibility in expanding the number of wireless channels in a system according to budgetary and operational needs.
- Using the controls and LCD on the MB-8N's front panel, users can quickly generate and store channel names for each WRU-8N Tuner Unit installed. Since the stored names can be recalled to the LCD panel, operators can immediately verify the sound source assigned to a given unit. Additional controls include output level (mic/line selectable), variable RF squelch level for each tuner unit, antenna attenuator value, and power to the AN-820A UHF Antennas connected to the MB-8N Tuner Base.
- The MB-8N has a built-in antenna divider, allowing four units to be daisy-chained without the need of an external divider. Since each tuner base unit accommodates up to four tuners, a 16- channel system is easy to configure (requires 16 tuner modules).
- With the use of the optional Sony WD-880A Antenna Divider, simultaneous operation on up to 42 channels is possible. This combined with the pre-programmed channel plans, makes it easy to build a high-quality, sophisticated, multi-channel wireless system.
- Headphone monitor jack and level control are located on the MB-8N front pane. Output can be selected to monitor the sound from individual WRU-8N tuner units or the mixed sound from all installed units.

WRU-8N Tuner Unit(s)

- Wide dynamic range ensures that the WRU-8N can provide optimum reception of signals from current and future Sony wireless mics and transmitters.
- The three technologies used in the WRU-8N —PLL frequency synthesized system, pre-programmed frequency groups and Space Diversity Reception System—result in extremely stable RF reception, a vital factor for any wireless microphone system.
- Each tuner has a jog dial for quickly choosing channel/group number settings as well as as LCD panel and LED indicators for information on operating conditions.
- LCD indicates the operating channel group and frequency, and has four-segment displays of AF output and RF input levels, while the LEDs indicate RF input and AF output status and the battery reserve of the transmitter. A red LED starts flashing one hour before the transmitter's battery will run out.



B&H PHOTO - VIDEO - PRO AUDIO
ONE HOUR FREE PARKING
AT 349 W. 34th STREET (with purchase of $100 or more)

MB-8N and WRU-8N

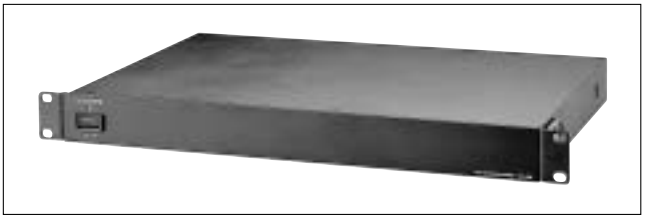
Computer-Based Control



For added operational convenience, the MB-8N/WRU-8N system allows computer-based remote control. To provide straightforward and intuitive operation, the MB-8N and installed WRU-8N units can be set up, controlled and monitored from a PC interfaced via a simple Ethernet connection and running the supplied software. Full monitoring of operating status in one window. The GUI simultaneously displays the status of up to 16- channels in one window, which can be switched to a 25- channel or 49-channel display. Both a quick reference to each tuner's reception status and that of the entire configuration are displayed on left side, while details of the channel status are displayed on the right side. Through a simple drag-and-drop operation, the operator can designate the channels to appear on the right side for access to an extensive range of operating controls and information which includes the channel name, selected channel group/frequency, tone squelch status, noise squelch status, RF squelch level, AF/RF input levels and transmitters battery alarm. The AF level is indicated in steps of approximately 2.5 dB, while the RF level is indicated in steps of approximately 5 dB. The GUI also allows the control of all other settings available on the MB-8N Base Unit.

The pre-programmed channel frequencies can also be selected from this GUI. This is especially useful for complex multi-channel operations, since operators can adjust frequencies while simultaneously viewing all channels. Once the appropriate frequencies are selected, the setup can be stored as a file. This is extremely useful, since the setup for a particular multi-channel system can be easily re-established by loading the appropriate file.

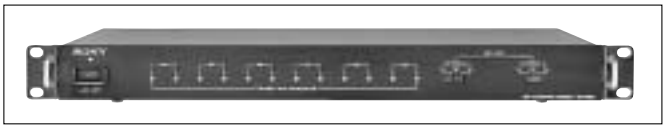
WD-820A UHF Antenna Divider



- ◆ Four diversity distribution output connectors for distributing the RF input signal from up to three MB-806A tuners
- ◆ Two pair of antenna input connectors for connection of up to four AN-820A antennas– allowing expansion of the operating area
- ◆ Cascade three WD-820A's to control up to 19 channels

WD-820A Antenna Divider.....**1113.95**

WD-880A UHF Antenna Divider/Channel Multiplier



- ◆ Expands simultaneous operation from 19 channels to up to 42 channels
- ◆ Incorporates RF bandpass filtering technology for stability and reliability
- ◆ Supplies 9 or 12v power to the AN-820A

WD-880A Antenna Divider/Channel Multiplier.....**3579.95**

AN-820A UHF Antenna



The AN-820A incorporates an amplifier to compensate for signal deterioration caused by cable transmission and distribution.

- ◆ Unique and inconspicuous design allows installation on a wall or in a mic stand with the supplied adapter
- ◆ LED indication for installation check
- ◆ Powered by the MB-806A or the WD-820A via coax cable

AN-820A UHF Antenna..... **189.95**

UWP SERIES

Low-Cost UHF Synthesized Wireless Microphone System



Although low-cost wireless systems has proliferated dramatically in recent years, they usually lack in transmission stability and have noise problems that are overlooked. The ideal solution for budget-conscious users seeking rock-solid transmission, the UWP Series incorporates sophisticated wireless technologies from Sony's top-of-the-line UHF Series including PLL-synthesized system, space-diversity reception and a tone squelch function. Whether for low-cost ENG, EFP or PA systems, the UWP Series delivers the convenience of non-compromised wireless microphone operation at a very affordable price.

The UWP Series consists of five core elements — a lavalier/bodypack transmitter, a wireless handheld mic, a portable tuner, a half-rack-size tuner, and a tuner module. These are available in six turnkey packages, each comprising a microphone, transmitter and tuner, for a ready-to-go system straight out of the box. Each package has been carefully compiled to address almost any application—from video production and A/V presentations, to live performance and electronic news gathering.

FEATURES

- PLL Synthesized System used in the transmitters and tuners, achieves solid transmission and reception by using a stable carrier signal to avoid interference with other frequency channels and to allow the selection of a preferred channel from multiple frequencies (188 selectable frequencies).
- Space Diversity Reception System—used in all UWP tuners – reduces signal dropout to a minimum. Achieves stable reception by using dual-antenna inputs/reception circuits that receive signals over two different paths and automatically select the stronger RF signal for output.
- The antennas of the portable and half-rack-size tuners each allow for angle adjustments, which helps to further eliminate signal dropout.
- Tone Squelch Circuitry prevents the output of unwanted signals or noise from other signal transmissions in the air, as well as the RF noise and popping noise that occur when the transmitter is powered on or off.
- Allows simultaneous operation of up to 16 wireless mics. Optimum combinations of practically tested, intermodulation-free frequencies (stored in the UWP tuners) simplifies the task of system setup.

Lavalier Microphone and Bodypack Transmitter



- Compact and lightweight design
- 3.5mm, 3-pole mini-jack input connector with lock mechanism
- Includes belt clip
- Supplied lavalier is a uni-directional, electret-condenser mic with windscreen and mic clip

Bodypack and Handheld Transmitter Both Feature

- LCD screen displays the operating channel number and its frequency in MHz, attenuator level, RF-output level (High/Low), audio-input and RF-output status, transmitter-battery status and accumulated operating time
- Six hours operation on two AA batteries
- Selectable RF-output level: 5 mW for simultaneous multi-channel operation: 30 mW for long-distance transmission
- Attenuator function allows adjustment of the mic-input level to suit each user's voice

Handheld Microphone



- Uni-directional, dynamic mic capsule
- Internal antenna design
- Supplied with a microphone holder and a screw adapter

UWP SERIES



Half 19″ Rack-Size Tuner

- Equipped with both XLR (balanced) and 1/4″ phone (unbalanced) type output connectors. The output level on the XLR-type connector can be switched between mic and line levels.
- LCD screen displays the operating channel number and its frequency in MHz, plus the audio output status and RF-input level
- Stereo headphone jack with monitor volume-control
- Supplied with an AC/DC adapter
- Angle-adjustable antennas to help eliminate signal dropout

Portable Tuner



- LCD screen provides extensive information, including the operating channel number and its frequency in MHz, audio-output status, RF-input level, tuner-battery status and accumulated operating time.
- Six hours of operation with two AA batteries
- Stereo mini jack with volume control
- Angle-adjustable antennas help eliminate signal dropout and allow mounting position flexibility when mounted on a camcorder.
- Supplied shoe-mount adapter enables easy mounting on Sony camcorders. Also includes mic stand adapter, screw adapter, mic cable and belt clip.

Plug-in Diversity Tuner Module



- Up to modules can be installed into a Sony mixer/amplifier (SRP-X700P or SRP-X351P), while up to six modules can be installed in the Sony MB-806A tuner base unit
- An LCD screen displays the operating channel number and its frequency in MHz, plus the audio-output status and RF-input level

All Tuners (receivers) Feature

- Space diversity reception system for stable RF reception
- RF squelch function virtually eliminates ambient noise and unwanted signals from other wireless microphone systems
- A green LED indicator illuminates when RF-input signals are appropriately received

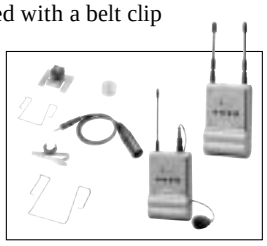
Lavalier Systems

Each system includes a lavalier mic supplied with a windscreen and mic holder clip, and a bodypack transmitter supplied with a belt clip

UWP-C1

Includes lavalier microphone, bodypack transmitter and portable tuner

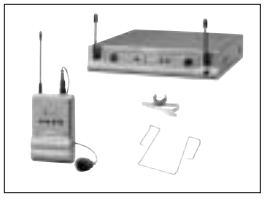
- Suitable for a wide range of applications, from news gathering and interviews to talk shows and conferences
- The portable tuner includes mic stand adapter, screw adapter, shoe-mount adapter for mounting on a camcorder and mic cable (3-pole mini plug/XLR-type)



UWP-S1

Includes lavalier mic, bodypack transmitter and half rack-size tuner

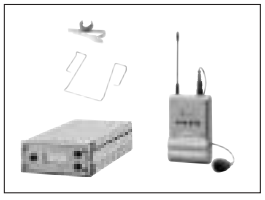
- Suitable for use in PA systems
- The half-rack-size tuner is supplied with an AC/DC adapter



UWP-X1

Consists of a lavalier mic, bodypack transmitter and tuner module

- Suitable for use in PA systems



Handheld Systems

Each system includes a handheld microphone supplied with mic holder and screw adapter

UWP-C2

Consists of a handheld microphone and portable tuner

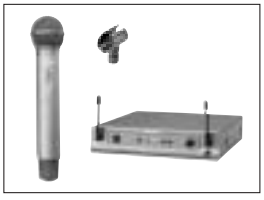
- Suitable for news gathering and for use in PA systems
- The portable tuner includes mic stand adapter, screw adapter, shoe-mount adapter for mounting on a camcorder and mic cable (3-pole mini plug/XLR-type)



UWP-S2

Consists of a handheld microphone and half rack-size tuner

- Suitable for use in PA systems
- The half-rack-size tuner is supplied with an AC/DC adapter



UWP-X2

Consists of a handheld microphone and tuner module

- Suitable for use in PA systems



SRP-X351P

Powered Mixer



An audio/video mixer and power amplifier, the SRP-X351P is designed for a wide range of applications, including corporate boardrooms, conference rooms, houses of worship and sports/karaoke bars. Cost-effective and flexible, it has all the key elements of an outstanding presentation system within a rugged, compact design. A built-in video switcher facilitates easy selection of a monitor or projector and the corresponding audio sources. Wide range of connections includes four balanced-XLR mic channels with switchable 48V phantom power; three stereo line inputs; stereo and master record outputs, as well as Echo send out and mic group out that allow for flexible system configuration and easy expansion. Finally, the SRP-X351P offers a unique wireless tuner option. It has two plug-in slots in the back of the unit that accommodate two Sony WRU-806A UHF tuner modules with a built-in antenna divider. Mounts in a 19˝ rack (3U high).

FEATURES

Inputs

- Four balanced mic inputs are provided via rear-panel XLR connectors. The fourth input is also accessible via a 1/4-inch jack on the front panel. A sharp cut-off LF filter and a Signal Present LED are provided on all four inputs. 48v phantom power is available on the XLR connectors.
- Three stereo channels with line level inputs. Channel 3 has four input connections, with input selection made by the 4-way A/V selector.



Outputs

- Mic Group bus (usually used to route selected mics)
- An insertion point is provided for an optional EQ, dynamics control unit or feedback reducer.
- Stereo Record Bus feeds both the power amps and the record output via an EQ and an insertion point for external signal processing. This means that the stereo record output can feed external power amplifiers with the same signal as the internal power amplifiers.

Video Switcher

- Has four sets of composite and S-Video inputs. Any one set of inputs is selected by the four-way A/V selector and routed to two pairs of outputs, one associated with the stereo Master output, and one with the stereo Record output.

Remote Control

- RS-232C and Control-S ports allow external remote control of A/V input selection and Master Volume, as well as muting of the power amp and Record outputs. Also includes an infrared wireless remote .

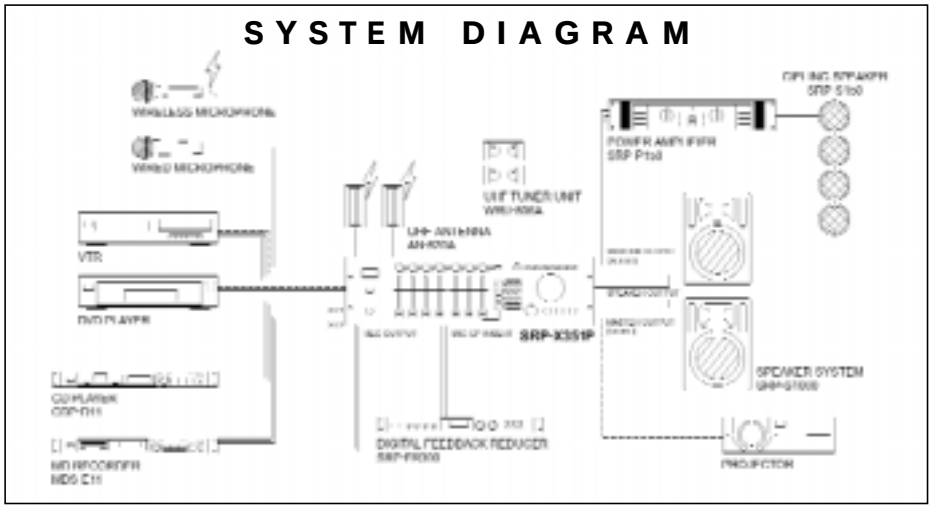
Wireless Tuners

- Two Sony WRU-806 wireless tuner units can be directly installed into the rear of the SRP-X351, to integrate high quality Sony wireless mics into the presentation system. 9v DC is provided on the two antenna connectors to power optional Sony AN-820A omni-directional active antennas.

Power Amp

- Low-distortion stereo power amplifier outputs 170W per channel into 4 ohms. Loudspeaker protection circuit and clip indication are also provided.

SYSTEM DIAGRAM



SRP-X700

Digital Powered Mixer

Designed for the modern presentation system, the SRP-X700P has the ability to process material from a wide range of sources including microphones, video, audio, DVD and PC graphics. In fact, it provides the functionality of up to seven tradition presentation devices in a compact, 3U high rack-mounting chassis. And by utilizing digital processing and control technology, the SRP-X700P combines an impressive set of features with high quality processing and simple operation.



WRU-806 tuner slots

FEATURES

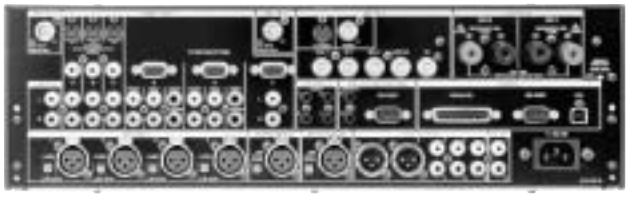
6x1 Multimedia Switcher

The SRP-X700P can handle a wide variety of signal sources such as wire/wireless microphones, PC graphics, DVD playback and surround sound audio, and has a 'six into one' switcher that allows all the different material to be routed to the presentation system.

- Six picture inputs: 3-RGB/component, 3-composite/S Video
- Six audio inputs: 4 stereo and 2 surround sound (5.1 format)
- High resolution component signal processing, including 480p or 1080i
- High-performance switching, with a 150MHz frequency response for RGB signals (1280 x 1024 pixels, SXGA)

Integrated Machine Control and Automation Control Interfacing

The SRP-X700P has extensive machine control capabilities with four CRTL-S control ports for wired machine control, as well as parallel and RS-232C ports for extensive control from professional presentation controllers.



- RS-232C port allows for remote control of Sony projector and plasma monitor
- USB, RS-232C and parallel input ports for remote control from an external PC (or other control systems).
- Control-S ports allow for remote operation of Sony VCRs, CD and DVD players, Minidisc recorders and projectors.
- A parallel output port for remote control of such devices as a video screen (up/down), window curtains (open/close), projector (up/down), lighting (on/off), etc.
- Up to two Sony WRU-806 wireless tuner units can be directly installed into the SRP-X700P, to integrate high quality Sony wireless mics into the presentation system.

Bundled Software for Improved System Efficiency

The SRP-X700P is supplied with two PC applications (SRP-X700P Manager and User Control Panel). The X700P Manager is a 'set and forget' application, supplied to allow the unit to be fully configured during installation and provides a complete set of control GUIs. The User Control Panel provides a simple control GUI for real time presenter control. Additionally, the unit's 20 internal Scene memories, four of which can be recalled from front panel buttons or the user control panel, allow an instant reset all of the internal processing, so down-time is minimized between operating modes.

- 6-mono and 2-stereo channels audio mixing with processing to maximize sound quality

– Can mix and process 6-mono (4 mic, 2 mic/line) and 2-stereo (a stereo line and the 5.1/stereo line input of the switcher) inputs signals to 10 outputs (8 general purpose outputs and a stereo record output).

– Comprehensive audio processing including EQ, dynamics, routing and delay, allows the audio signals to be maximized and provides tools for correcting for room acoustics.

- Two additional tools allow 'professional quality audio' processing to be setup at the touch of a button:

– 'Feedback Reducer' analyzes and 'nulls out' potential howl round frequencies.

– 'Automix' function allows the SRP-X700P to analyze, in real time, the incoming signals and make adjustments to maintain the quality of the output.

- 24-bit/48 kHz A/D -D/A conversion
- Multiple operating modes with 10 outputs and the on board stereo power amplifier
- A routing function allows the SRP-X700P to offer many different operating modes to support surround sound matrix switching or 'Zoned' speaker configurations. In addition, the on board 150W stereo power amplifier can be switched between high and low impedance modes.
- Power rating 200 W + 200 W (4•), 150 W + 150 W (8•), Max 150 W (70V Line)

TELEX

ENG-100

100-Channel UHF Wireless Mic System

The versatility of the ENG-100 receiver makes this the UHF system of choice. Designed to perform reliably even under the harsh conditions of news gathering, the compact but rugged receiver offers 100 channel frequency agility and delivers superior audio performance on any of 100 selectable channels in the 668 to 746 MHz frequency range. Two integral antennas and a built-in Posi-Phase diversity system assure reception in the most difficult situations. The ENG-100 also features an internal headphone amplifier and an adjustable audio output control to accommodate any situation from news gathering operations to live performances. Equally suited for corporate, industrial and event video applications, the ENG-100 is available in two frequency groups; Group A which corresponds to TV channels 47-48 and Group B which corresponds to channels 58-59.



FEATURES

ENG-100

UHF Camera Mountable Receiver

The frequency-agile 100-channel ENG-100 receiver mounts on and interfaces with virtually all professional camcorders. It uses dual integral antennas and Posi-Phase diversity to ensure clear, superior quality, drop-out free audio. The receiver provides side-mounted controls, XLR mic output, headphone output and internal (two AA batteries) or external power for hassle-free field operation.

- Operates in the 668 - 722 MHz range
- Posi-Phase II smart diversity reception and Posi-Squelch III audio suppression ensure reception in the most difficult conditions.
- Tone-code squelch
- XLR mic output and headphone monitor output jack
- Audio level control with mic/line selector switch
- Each transmitter matches two channels on the ENG-100 receiver for easy changes in the case of interference on location. Additionally, with 4 different channel groups to choose from, you get up to 8 simultaneous users in one location.

$399.95

LT-100/SH-100

UHF Beltpack and Handheld Transmitters

- 100 selectable frequencies
- Power on/off and audio mute switches
- Four segment battery life indicator
- Hi/low transmit power output
- They operate on 2 AA batteries
- Handheld transmitter with Audix OM3 cardioid dynamic mic capsule (SH-100)
- TA-4M mic jack (LT-100)
- Has external charging pins for use with BC-100 drop-in charger (LT-100)

LT100 Beltpack Transmitter.....**377.50**

SH100 Handheld Transmitter **469.95**

UT-102

UHF Crystal-Controlled Plug-on Mic Transmitter

The UT-102 is a top-of the-line, crystal-controlled, frequency-agile plug-on transmitter that converts standard wired dynamic mics into wireless-capable transducers. This gives sound and live broadcast engineers the ability match the individual location and application with different microphones.

- Housed in an all metal aircraft alloy case with water-resistant seals.
- Although compatible with other dynamic microphones, the UT-102 is perfect for use with the EV RE50, the world standard in broadcast interview microphones.
- Also provides +6V phantom power for electret microphones.

U102 Plug-in Transmitter.....**749.95**

ENG-100L System

ENG-100 receiver, LT-100 beltpack transmitter, ELM-22S omni-directional lapel microphone, SEB-1 single earbud with cord, XLR-to-pigtail audio cable **679.95**

ENG-100UT System

ENG-100 receiver, UT-102 plug-on two-channel transmitter, SEB-1 single earbud with cord, XLR-to-pigtail audio cable.....**1049.95**

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ENG-100

Optional Accessories for the ENG-100

DC-GM Anton Bauer "Sandwich" Mount

Allows the ENG-100 to be sandwiched between the camera and an Anton Bauer battery. Fully compatible with Anton Bauer Gold Mount **229.95**



DC-SM Anton Bauer Side Mount Kit

Attaches ENG-100 to Anton Bauer brand side mount accessory.....**42.95**





DC-UM Universal Strap Mount Kit

The DC-UM is a universal mounting device that fits virtually any video camera. The adjustable, self-locking Velcro strap ensures a secure mount.....**42.95**



DC-NP1 Standard NP-1 Mount Kit

The DC-NP1 allows the ENG-100 to be mounted on standard NP-1 battery holder (easily installed and removed with thumb screws **42.95**

PROSTAR VR-12 *VHF Wireless Microphone System*

Going wireless has never been more affordable and natural. Built around the VR12 receiver, the ProStar system offers an audio signal that is processed by the fewest possible components. The allows the signal to follow a virtual "straight line," producing pure and natural sound. Available in 8 frequencies, up to three systems can be used simultaneously.

The VR12 receiver is rugged reliable and includes the same diversity Posi-Phase technology found in professional Telex wireless systems. It features squelch control, 1/4˝ output with volume control and a built-in antenna system that eliminates interference and substantially increases the range of the system.



Made in the USA with the highest reliability in their class, the ProStar system delivers clean drop-out free performance every time you power up. Choose from handheld, lavalier or headworn mic systems. For guitarists, the VGR12S system was conceived, designed, and constructed for one instrument only – the guitar. The VGR12S captures everything from the window-rattling lows and the searing highs of a solid body, to the subtlety and nuances of a jazz guitar or acoustic/electric.

VR12C Lavalier System

Includes the VR12 receiver, VB-12 beltpack transmitter with TA4M connector and Audio Technica AT-831b unidirectional condenser lavalier mic with TA4F connector.....**259.95**

VR12L Lavalier System

Includes the VR12 receiver and VB-12 beltpack transmitter with attached WLM-10 electret condenser lavalier mic.....**189.95**

VR12N1 Handheld System

Includes the VR12 receiver and a VH12N1 handheld transmitter with EV 167 neodymium dynamic mic element.....**238.95**

VR12H Headworn System

Includes the VR12 receiver and VB-12H beltpack transmitter with EV HM2 headworn mic **289.95**

VR12S Guitar System

Includes the VR12 receiver and VB-12 beltpack transmitter with 1/4˝ plug and cable .....**Call**

VR12A Aerobic System

Includes the VR12 receiver, VB-12 beltpack transmitter (with sports pouch) with TA4M connector and Special Projects WPHS-746 aerobics mic with TA4F connector.....**359.95**

TELEX

FMR-10

10-Channel UHF Frequency-Agile Wireless Mic System



Designed for sound contractors, A/V rental facilities and vocalists, the FMR-10 system offers high performance audio, frequency agility and simplicity. Ideal for schools, houses of worship and DJs it includes optimized channel frequencies to allow up to 10 simultaneous operating systems by combining channels from low and high bands (up to seven simultaneous systems within one band). Additionally, a 10-channel UHF frequency-agile system with ClearScan auto channel selection allows quick setup, transmitter matching and new channel selection if RF conditions change. At the push of a button, C1earScan scans for and selects a clear channel. Operation is further enhanced with front panel 4-segment RF and audio level LEDs, balanced-XLR mic level output and by the Posi-Phase diversity system for clear and drop-out free audio. For clear vocals the HT-10D handheld microphone transmitter features an Electro-Voice N/DYM 767A dynamic mic element with VOB (Vocally Optimized Bass) technology.

FEATURES

HT-10D/WT-10
Handheld and Bodypack Transmitters

- 10 selectable channels
- Power on/off and audio mute switches
- Power and low battery indicator LEDs
- EV N/DYM 767A mic element with VOB (HT-10D)
- TA-4M mic connector (WT-10)
- Run 10 hours on a single 9v battery

FMR-10
UHF Synthesized Receiver

- Posi-Phase Diversity receiver technology provides clear, drop-out free audio
- Adjustable amplitude squelch
- Optimized channel frequencies allow up to 10 systems to operate simultaneously by combining channels from low and high bands, or up to 7 channels within one band.
- 10-channel UHF frequency-agile system with ClearScan auto channel selection allows quick setup, transmitter matching, and new channel selection if RF conditions change.
- Front panel RF and audio level indicators
- Balanced XLR output
- Two 1/4-wave antennas
- Rackmount hardware for one or two receivers
- Wall pack power supply

FMR-1000

950-Channel UHF Frequency-Agile Wireless Mic System



The FMR-1000 is a breakthrough frequency-agile system that mixes the power of fully programmable channels with the simplicity of Advanced ClearScan. With a push of two buttons, Advanced Clear Scan finds the clearest group of channels and you can set up 16 system installations in minutes with confidence you have the clearest channels in the clearest group. A unique sound check screen allows one person to walk test a transmitter with tangible results. For clear vocals, the HT-1000 handheld microphone offers a choice between two Electro-Voice mic elements. The ideal solution for the needs of sound contractors, A/V rental facilities and performers, the FMR-1000 carries a 3-year warranty and very affordable price.

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EQUIPMENT LEASING AVAILABLE

FMR-1000

FMR-1000 System Features

- Programmability in 25 kHz steps across 24 MHz operating bandwidth for over 950 possible channels.
- Channels are set-up in optimized groups to allow the simultaneous operation of up to 16 units within one frequency band.
- At the simple push of two buttons, Advanced ClearScan finds the clearest group of channels, allowing you to configure up to 16 system installations in minutes with the confidence that the clearest channels in the clearest group have been programmed.
- Power users can also program their own groups and channels for custom installations and tours.
- UHF operation is further enhanced by the patented DSP Posi-Phase diversity system for clear and drop-out free audio.
- Front panel parametric equalizer with Level, Q, and frequency controls for sound shaping without a mixing board.

FMR-1000 Receiver

- 1⁄2 rack width metal receiver housing design, RF and audio level indicators, tone code plus adjustable squelch, 950 selectable channels, balanced XLR output, adjustable 1⁄4″ line output, two 1⁄4-wave antennas, rack-mount hardware for one or two receivers and an in-line power supply cord. Also features a plug-able terminal strip for use in permanent installations
- Backlit LCD display shows group/channel, transmitter battery status, diversity operation, RF and audio level meters and allows space for a custom label.
- Unique sound check mode allows a person to walk-test a microphone throughout the performance area with quantifiable results shown on the LCD display.
- Front panel parametric equalizer with Level, Q, and frequency controls for sound shaping without a mixing board.
- Optional AD-450 antenna distributor provides power and antenna connections for four FMR-1000(s) from just two antennas.

WT-1000 *and* HT-1000 Bodypack *and* Handheld Transmitters

The bodypack transmitter is made of cast magnesium and the over-molded Warm-Grip handheld transmitter features detachable microphone elements for rugged performance in the most demanding environments. Both transmitters feature a unique “smart” battery circuit, which makes it impossible to put the battery in wrong. Powered by 9v battery for up to 8 hours.

For clear vocals, the HT-1000 wireless handheld transmitter offers a silent power on/off and LCD channel display, low-battery indicator LED, and the choice between the EV N/DYM 767a dynamic mic element with VOB and the RE 510 professional condenser element.

THE BEC GROUP

Wireless Receiver Holders



BEC’s Wireless Receiver Holders are manufactured using plate aluminum, formed and welded to the exact size of your receiver. Their skilled metal mechanics use the latest in computer assisted machinery to meet the exacting standards of your cameras and receivers. They apply a durable Dry Powder coating to each holder in a smudge resistant black wrinkle finish. And your receiver is protected by a felt lining that keeps it from the bangs and bumps of everyday use.

Holders mount on all cameras with the Sony screw configuration that is most commonly found on the NP/1 battery holder, the IDX NP/1battery system and the Anton Bauer Gold Mount system. Other options like the Sony V-lock system, other Pro cameras and the smaller DV cameras, may need BEC camera brackets, slide plates and other accessories to complete the installation. Everything you need for a no hassle installation is supplied. Includes all the screws, instructions with pictures and a reversible screwdriver.

BEC-185: Fits Lectrosonics CR-185, CR-187, UCR190 ..... **74.95**
BEC-195: Fits Lectrosonics CR-195, UCR-200/205/211/411.. **74.95**
BEC-201: Fits Lectrosonics UCR-201 ..... **74.95**
BEC-210: Fits Lectrosonics UCR-210 ..... **74.95**
BEC-100: Fits Lectrosonics UCR-100 ..... **69.95**
BEC-500: Fits Sennheiser Evolution 500 ..... **69.95**
BEC-3041 Fits Sennheiser 3041 ..... **74.95**
BEC-AT100: Fits Audio Technica UCR100 ..... **69.95**
BEC-AZ400: Fits Azden UDR-400/500/1000 ..... **74.95**

PORTACOM

2-CHANNEL WIRED INTERCOM SYSTEMS

A flexible, high-performance series of two-channel wired intercom, built smart, simple and affordable without sacrificing power or sound quality. Portacom's low-noise, broadcast-quality intercom systems are easily expandable and can be tailored to any environment. They are available in pre-configured systems or as individual components. Connection between components are made with standard mic cables and quickly adapt and grow with your needs without having to start all over with a new system. The two channel capability offered by these systems allow a director separate lines of communication to crew members and talent. The small, lightweight aluminum belt packs have their own channel switches and volume controls and Headsets are equipped with mic on/off switches. The noise-canceling microphones reduce background noise and you can power the system with an optional lightweight battery pack. Ideal for use in television/theater/lighting and church production, educational institutions, industrial production, sound contracting and football coaching.





- Full duplex
- High performance/low cost
- Durable lightweight metal construction
- Standard cables/ 3-pin XLR connectors
- Comfortable noise cancelling headsets
- Battery powered option
- Call light function
- Offers 20-headset capability with simple branch-box operation

Pre-Configured 2-Channel Intercom Systems


COM-60FC

2-channel wired intercom systems consist of a central power supply, belt packs with A/B channel select, volume control, call light and mic on/off switch. With dual muff headsets, carrying case, and with or without standard mic cable. (H-200S single muff headsets can be substituted– mixed or matched, but you must specify.)

COM-40 4-Headset System:
Includes PC-100 power console, 4x H-200 dual earpiece headsets, 4x BP-200 belt-packs, 4x EX-50M 50´ extension cables, CC-15 carrying case

COM-40FC 4-Headset System: Same as COM-40 without cables

COM-60 6-Headset System:
PC-100 power console, 6x H-200 dual muff headsets, 6x BP-200 belt-packs, B3-100 branch box, 7x EX-50M 50´ extension cables, CC-15 carrying case

COM-60FC 6-Headset System: Same as COM-60 without cables

COM-80 8-Headset System:
PC-100 power console, 8x H-200 dual muff headsets, 8x BP-200 belt-packs, 2x B3-100 branch boxes, 10x EX-50M 50´ ext. cables, 2x CC-15 carrying cases

COM-80FC Same as COM-80 without cables

Components



PC-100: 2-Channel Power Console- Two channel power console with 4 outputs; powers up to 20 BP-200 belt-packs. Includes wall mount AC adapter (operable on 30-40v DC)

BP-200: 2-Channel Belt-Pack- Lightweight belt-pack with volume control and A/B channel switch, call light and mic on/off switch. Has standard XLR-3 line connector and XLR-4 headset connector. Rugged construction. For use with H-200 headsets

H-200S Single Earpiece and H-200 Dual Earpiece Headsets- Lo-Z dynamic mic and headphones with 6' cable and XLRF-4 connector. Feature mic on/off switch, and noise cancelling mic

B3-100 Branch Box- For expanding PortaCom systems, 1 input and 3 outputs

CC-15F Deluxe Carrying Case- Custom-cut foam inserts for PC-100, AC adapter, 6 headsets, 6 belt-packs, Branch Box

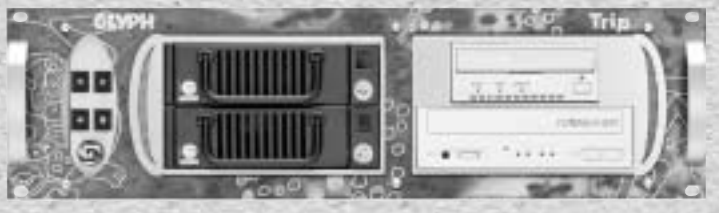
CC-15 PortaCom Carrying Case- Without foam inserts

PBS-100 Portable Battery Supply- Uses 6 D-cell batteries to power 8-10 headsets for 6 to 8 hours

RM-100 PortaCom Rackmount-
Mounts 1 or 2 PC-100 consoles in a single 19˝ rack space


Duo




GLYPH
Trip



B&H

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The Professional Audio SourceBook

Section 11a

Computer Audio Hardware

Computer Audio Peripherals

Aardvark **792-799**
Apogee Digital **800-807**
Edirol **808-819**
ESI **820-825**
Event Electronics **826-827**
Lynx **828-831**
M-Audio **832-841**
Mackie **842-845**
Mixed Logic **846-847**
Presonus **850**
Radikal Technologies **848-849**
Roland **851-854**
Tascam **854-857**
Terratec **858-861**
Yamaha **862--863**

Computers/Monitors & Storage

Apple Computers **864-771**
Carillon Audio **872-875**
Sony **876-877**
Glyph Technologies **878-882**
ADS/Medea **883**
LaCie **884-887**
Pinnacle Systems **888**
Roxio **889**

DIRECT PRO Q10

10-in/10-out Audio Interface with 8 XLR Mic Preamps



Looks great. Sounds even better. You've never heard mic preamps sound this good. Aardvark's unique Q10 brings a new approach to computer recording. Its 8 studio quality XLR mic preamps have an ultra quiet .002% THD+N so you can professionally record an entire band direct to the computer without a mixer. Any of the 8 inputs can also be used as high quality 1/4˝ line inputs and take advantage of Aardvark's legendary sound quality and low-jitter performance. The Q10 has tons of studio-friendly features like 4¼˝ analog effects inserts for easy integration with outboard processors, phantom power, 2 EFR guitar Hi-Z inputs, front mounted headphone & line monitor out level controls, S/PDIF, MIDI, a shielded PCI card and comes with a full version of Cakewalk's Pro Audio 9.0 recording software.

The Q10 is compatible with almost all audio recording software, and is often used with Steinberg Cubase & Nuendo, Cakewalk Sonar, Emagic Logic Audio Platinum, Syntrillium Cool Edit Pro 2000 and gets ultra low-latency performance with software synths & samplers like Native Instruments, GigaStudio and Propellerheads Reason.

FEATURES

The Q10 embodies all of Aardvark's great hardware technologies to bring to you one of the highest quality systems available. It offers great mic preamps and converters, plus all the appropriate I/O's of an analog mixer. This means that with one unit, you can record direct to you computer without having to go through a mixer, noisy mic preamps and all the extra noisy cables just to record audio on your computer.



- 8 flexible and high quality inputs. The XLR inputs go directly to the discrete mic preamp circuitry , the 1/4˝ inputs go directly to the line level inputs.
- Software controllable analog input gain, applied before the A/D converter, maximizes headroom and gives you clean gain just like an analog mixer. There's even Aardvark's proprietary EFR™ Hi-Z guitar input circuitry to warm up any guitar you connect directly into the Q10.
- Unlike other soundcards that use standard "mic-pre-on-a-chip" solutions, which are not discrete and can lack the clarity and definition needed for professional use, the Q10's mic-pre's are a hand crafted, discrete transistor design that yield outstanding noise specs and pristine sound quality.
- The mic preamps are shielded in the breakout box and combine with top of the line converters to deliver professional audio quality from your mics to your computer.

The true brilliance of the Q10 is the combination of great sounding but flexible hardware & brilliant DSP-powered software. Discrete transistor mic preamps, 24-bit A/D-D/A, shielded PCI card and Aardvark's legendary digital clocking technology combine to deliver audio quality that is second to none. The flexibility of the hardware completely replaces an outboard mixer, so you can record directly to computer without sacrificing audio quality or studio functionality. The software-based DSP Control Panel uses the powerful DSP processor on the PCI card, so your computer keeps running lightning fast and in real-time. It's laid out much like an analog mixer, and leaves native power for you to run more plug-ins and other processing.

4 Inserts for Outboard Effects

The Q10 has four 1/4˝ TRS inserts that work just like those of an analog mixer. They connect directly to outboard signal processing gear so you can 'insert' the processed analog signal between the mic preamps and the A/D converter. This means you can integrate your classic analog signal processing gear such as EQs, compressors, and limiters with the Q10s crystal clear mic preamps, when you record direct to the computer.

2 EFR Hi-Z Guitar Inputs

The selectable front panel Hi-Z inputs (7+8) feature EFR (Enhanced Frequency Response) input circuitry specifically tailored for recording electric and bass guitars with a unique warmth and clarity. If you disliked the sound of a cheap DI box, then you'll love the warm, rich sound the EFR provides.

Headphone & Monitor Output

There is a headphone output on the front and stereo Monitor outputs on the back to connect directly to your speakers. Both outputs have individual level controls to give you mixer-like monitoring capability. Combine that with the 20 channel real-time DSP Mixer, and you have an incredibly flexible and professional system.

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DIRECT PRO Q10

Shielded PCI Card

With custom shielding on the PCI card, both the breakout box and the card are noise free, keeping harmful computer noise and EMI out of your precious audio.

Expandable

You can run up to 4 Q10's in the same computer for a maximum of 32 analog inputs and 8 channels of digital I/O. (In fact, all Aardvark audio interfaces can be expanded up to 4 units…mix and match!) Just connect the word clock in & outs on the Q10's to each of the devices and your running sample accurate with an unprecedented 40 channels of great sounding I/O. This is perfect for portable rigs that need tons of I/O and great sounding mic preamps.

Aardvark Low Jitter Clock

Every Aardvark computer audio card uses clocking technology similar to the legendary AardSync II master clock generator. The AardSync II is the industry standard master clock, used to sync Pro Tools, digital mixers, A/D converters and other devices, improving their sound via its low-jitter clock. This proprietary clocking technology gives our computer audio interfaces improved stereo imaging and better resolution on extended high and low frequency audio tracks.

Output Routing (Patch Bay)

Dealing with the software routing of audio interfaces can sometimes be a big hassle. So Aardvark took a photo of the back panel, and gave easy to use drop down menus to select what you want routed. This includes analog through to the outputs (Analog 1-8), any of the playback tracks (Playback 1-10), generating a tone and using the stereo output of the Control Panel's Monitor Bus (MONITOR).

Logo Brightness Control

One of the coolest features of the Q10 is that the power LED's brightness, on the front in the shape of the Aardvark logo, can be controlled from the DSP Control Panel to match your studio's light level. Even the phantom power button on the front activates both an LED on the front of the unit and one in the control panel.

Q10 Software Features



The Q10 also sets itself apart from other interfaces by having a fully DSP powered software control panel. This control panel is packed with user friendly features, and is laid out just like an analog mixer with stereo monitor outputs. The hardware & software combination of the Q10 truly replaces every aspect of traditional analog mixers, except you get better sound and tons more flexibility with digital presets and routing.

Q10 Control Panel: — The brain of the Q10, the DSP powered Control Panel replaces an external hardware mixer, while still providing the real-time mixing and easy monitoring that you get with an analog mixer. And since it's powered by the powerful DSP processor on the PCI Card, mixing, monitoring and metering won't slow down your computer. Works in combination with the Q10 hardware to provide a total digital mixing solution that includes flexible routing, zero-latency monitoring, snapshot preset recall, input level trim, digital peak metering and is +4/-10 selectable.

The Control Panel can be seen as a 20 channel mixer with a 2 channel monitor output that is routed to 1/4″ stereo line outs on the back panel, and a 1/4″ front panel headphone out. Ten of the channels are from the physical 10 inputs of the box (8 analog and stereo digital), and the other 10 channels are from the software. Any combination of previously recorded audio tracks can be played back, mixed and monitored individually or in subgroups, while simultaneously recording additional audio tracks. The control panel even comes in 4 cool colors that you can change as often as you'd like. This lets you match it to your studio, or make life easy on the eyes during an all night recording session.

3-Stage Input Gain: The input trim (found at the top of each mixer strip on the Control Panel) provides up to 75dB gain on mic inputs. This gain is done in the analog stage (pre A/D conversion), just like an analog mixer. In contrast, most other audio cards provide the gain after the A/D, which dramatically increases the noise. The Q10 software avoids this by sending commands to the analog circuitry in the breakout box to get the best sound possible. Actually has 3 separate gain stages; m2 (Mic 2), m1 (mic 1), and L (Line level), which are each optimized for that particular gain range for the best possible sound.

ASIO Turbo Mode: Enhanced ASIO 2 driver provides some of the best latency performance in the industry. Down to 4ms or less with some applications such as software synths and VST instruments. Great for real-time performance.

ASIO Buffer Slider: Not only is the ASIO driver really efficient, but it has a unique 31-step ASIO Buffer Slider which lets the user dial in the exact buffer size that is optimal for the computer. No other card goes this far to get real-time performance.

Snapshot Preset Recall: Hitting the Presets button on the Control Panel activates the Presets window, which let you save an unlimited number of screen shots to recall at any time This is particularly useful when you need to change the patch bay settings or the input levels for different studio sessions. It's also nice to save different or special mixes so you can get up and running in no time.

DIRECT PRO LX6

4-In/6-Out Audio Interface with Realtime DSP FX



Today's project studios need more than just a sound card. They need the same great sounding 24-bit/96kHz, multi-channel audio and DSP effects that the big guys have. Much more than just a sound card, the Direct Pro LX6 is a high-precision audio and MIDI interface that gives you everything that you need to record and playback professional audio on your computer. The intuitive user interface, professional monitoring capabilities and amazing sound quality make the LX6 a true recording revolution. Features include 4-in/6-out 24-bit/96kHz sampling capability, real-time DSP mixing & effects and a shielded PCI card to keep out computer noise. The built in S/PDIF makes flawless 24-bit digital transfers to outboard gear, and the integrated MIDI I/O makes it an ideal solution for audio & MIDI workstations. The powerful on-board DSP processor provides real-time, zero latency DSP effects and keeps your computer running lighting fast. There are also separate headphone and line outs for easy monitoring.

The LX6 and Direct Pro 2496 are compatible with almost all audio recording software including Steinberg Cubase VST/32 & Nuendo, Cakewalk Guitar Tracks Pro, Home Studio & Sonar, Emagic Logic Audio and Syntrillium Cool Edit Pro 2000. Their 6 outputs and ultra low-latency performance make the Direct Pro LX6 and Direct Pro 24/96 a perfect match for software synths & samplers like Native Instruments, GigaSampler and Propellerheads Reason. Both also come bundled with a full version of Cakewalk's Pro Audio 9.0 recording software.

DIRECT PRO 24/96

4-In/6-Out Audio Interface with Realtime DSP FX and 4 XLR Mic Preamps



A truly professional audio interface for the project studio, the Direct Pro 2496 has all the features of the LX6 but adds XLR mic inputs and four mic preamps to combine all the pieces of a professional studio into one complete and easy-to-use 4-in /6-out system. Its four studio quality XLR mic preamps are quieter than most mixers and outboard mic preamps, with a THD+N of .002%, and can even be used as 1/4˝ line inputs for those looking to maximize studio quality and flexibility. The Direct Pro also offers 48v phantom power and 3-stage input level gain.

Headphone Output

They have a headphone out right on the front of the unit to give you all the control over your mixes. The headphone gain is easily adjustable via the DSP software, which also does the real-time mixing of 10 channels to help you create the perfect mix.

Shielded PCI Card

Only Aardvark provides custom shielding on our PCI cards. Both the breakout box and the PCI card are designed to be completely noise free. This extra shielding keeps harmful computer noise out of your precious audio to give a pristine, high quality sound.

S/PDIF Digital I/O

The 2 RCA connections on this PCI card provide the 24 bit, 96kHz digital in and out to your computer for lossless digital audio transfers from outboard digital audio gear. A must have for any pro or project studio.

LX6 & DIRECT PRO 24/96

6 Analog Outputs

The rear-panel of the breakout box features four analog main outputs with 24 bit, 96 kHz +4dBu on 1/4″ TRS jacks and 2 additional outputs on RCA jacks. All 6 outputs can be used simultaneously to take to a mixer or outboard effects units. These give excellent mix down flexibility and provide plenty of connectivity to the rest of your studio. The six outputs can also be used for surround sound mixing.

MIDI In/Out

Built-in MIDI interface lets you integrate MIDI instruments (synthesizers, drum machines, etc) into your digital audio system.

The Direct Pro 24/96 breakout box features both a MIDI In and MIDI Out jack so you can send MIDI data to and from your favorite synth. And since the audio and MIDI are integrated in one box, your guaranteed perfect audio/MIDI synchronization.

Expandable

f you feel like expanding your Direct Pro 24/96 system, you can expand up to 4 units into one computer. In fact, all Aardvark audio interfaces can be expanded up to 4 units…mix and match! No other line of computer interfaces allows as much flexibility and expansion as the Aardvark Direct Pro Series.

Inputs

The LX6 has four 1/4″ line level inputs, while the Direct Pro 24/96 has for combo inputs that accept either XLR mic or 1/4″ line level inputs, or any combination of the two. This gives you the ultimate in flexibility, allowing you do all the work right in your computer.

4 Discrete Mic Preamps

Unlike other soundcards that use “mic-pre-on-a-chip”" solutions, which are not discrete and can lack the clarity and definition needed for professional use, the mic preamps in the 2496 are a hand crafted, discrete transistor design that yields outstanding noise specs and pristine sound quality. They also have phantom power for condenser microphones. The mic preamps are shielded in the breakout box and combine with top of the line converters to deliver professional audio quality from the mic to your computer.

LX6 and Direct Pro 24/96 Bundled Software

The LX6 and 2496 set themselves apart by have a fully-DSP powered, user-friendly software control panel laid out just like an analog mixer with stereo monitor outputs. The hardware & software combination of the LX6 and Direct Pro 2496 replaces every aspect of having an analog mixer, audio interface, and 3 cool effects units, so you can monitor in real-time while recording direct to the computer with ZERO-latency! This frees up power for the native processor so you can record more tracks and use more plug-in effects processors.

The Control Panel replaces an external mixer, while still providing real-time mixing and easy monitoring. And since it’s powered by the DSP processor on the PCI Card, mixing, monitoring and metering don't slow down the computer. Works in combination with the hardware to provide a total digital mixing solution that includes flexible routing, zero-latency monitoring, snapshot preset recall, input level trim, +4/-10 selectable I/O and true digital peak metering.

Control Panel

The Control Panel has complete mixing capabilities and is very easy to use. It can be seen as a 10 channel mixer with a 2 channel monitor output that can be routed to any of the 6 line outputs or the headphone output for monitoring. 6 of the channels are from the physical 6 inputs of the box (4 analog and 2 digital), the others are from the software. Any combination of previously recorded audio tracks can be played back, mixed and monitored individually or in subgroups, while simultaneously recording additional audio tracks from the LX6 or 2496. The panel comes in 4 colors, so you can match it to your studio, or makes life easy on the eyes during an all night recording session.



Compressor, EQ & Reverb

The 100% DSP generated Reverb/Echo, Compressor/Limiter, and Multi-band EQ let you add effects to incoming audio “on-input” while recording in real time with no-latency–just where you need it most! Choose between simply monitoring the effects while recording “dry” to multitrack software, or you may record DSP effects. You can also audition the effects individually with the effects bypass feature.

Snapshot Preset Recall

Save an unlimited number of screen shots to recall at any time later. This is particularly useful when you need to change the patch bay settings or the input levels for different studio sessions. Saving different preset mixes you can get up and running in no time.

ASIO Turbo Mode

The enhanced ASIO 2 driver provides some of the best latency performance in the industry. Down to 4ms or less with some applications such as software synths and VST instruments. Great for real-time performance.The ASIO also has a unique 31-step ASIO Buffer Slider which lets you dial in the exact buffer size that is optimal for your computer.

Output Routing (Patch Bay)

The patch bay allows routing of any input signal (as well as the DSP monitor mix) to any output (including the headphone output) in real time with no latency. You simply drag the source to where you want it to go. Tone and Silence signals can also be routed to any output for calibration purposes and system checks.

Cakewalk Pro Audio 9.0



The LX6 and 2496 are bundled with Cakewalk Pro Audio 9.0 multi-track recording and MIDI sequencing software. 28 tracks, 24-bit/96kHz support, real-time effects, MP3 enabled, import and sync digital video to audio, edit & print notation w/guitar tablature, real-time Guitar Fretboard display, time expansion/compression, audio-to-Midi Conversion, fade/envelope & crossfade, Direct X plug-in compatible, Parametric EQ, Reverb, Delay/Echo, Flange, Chorus, Pitch Shifting, Vintage Amp Simulation.

AARK 24

10-in/10-out Audio Interface with S/PDIF and ADAT I/O

The Aark24 is a professional quality 10 x 10 PC interface that is in use everywhere, from home project studios to the most modern advanced studios, making amazing digital recordings worldwide. The Aark 24 easily integrates into your studio with eight 24-bit A/D and D/A converters, stereo S/PDIF and ADAT sync/optical digital I/O, all shielded in a rugged rack-mountable external breakout box. Aark24's superior 24 bit converters assure every nuance of your audio is captured faithfully. The digital link to any ADAT or digital mixer enables you to keep your audio tracks in the digital realm and achieve the kind of fidelity your audio projects deserve.



FEATURES

10 Inputs/10 Outputs

The Aark 24 has eight front-panel mounted, balanced TRS 1/4″ analog inputs and outputs (individually switchable from +4 to -10 in the Control Panel software) connected to eight high-quality 24-bit A/D and D/A converters. The front panel of the shielded breakout box also features digital S/PDIF I/O, giving a high resolution 24-bit connection to digital devices such as external CD-burners .

ADAT Sync & Optical I/O

The rear-panel of the breakout box features ADAT optical and ADAT sync interfaces for professional recording flexibility. The optical I/O and ADAT Sync allow you to easily interface any ADAT or digital mixer to your PC.

Toslink

The Aark 24's fiberoptic jacks are software switchable to accept the popular two-channel TOSLINK format as well. This lets you fly in tracks optically from DAT and CD players and even allows you to edit your Sony MiniDisc so you can record in the field and then master & edit back home on your PC.

MIDI In-Out

Integrate MIDI instruments (synthesizers, drum machines, etc.) with your PC's digital audio system via the Aark 24's built-in MIDI interface. (Although Aardvark's digital interfaces don't synthesize MIDI data, the inactive interfaces are developed with the same high standard applied to their active interfaces.

Control Panel Software



The Aark 24's DSP powered Control Panel lets you meter each input and output signal, route any input or playback signal to any output, and create your own custom monitor mix with full level and pan control. Its all generated by the on-board DSP, so it won't slow down your PC's precious CPU resources! The monitor mix or any input signal can be routed to any physical output of the Aark 24 breakout box via the digital patchbay. The patch bay allows the routing of any input signal (as well as the monitor mix) to any output in real time with no latency - true direct monitoring. Tone and Silence signals can also be routed to any output for calibration purposes.

ASIO Turbo Mode

ASIO drivers drastically reduce the latency inherent in PC-based audio systems. Low-latency, real-time integration makes the Aark 24 one of the most powerful workstations available. The ASIO control panel allows limiting of input and output channels, Direct Sound buffer size, ASIO buffer size and ASIO sample bit size. All assisting you in configuring your computer for optimum efficiency and the lowest latency possible.

Assignable Stereo Monitor Bus

Monitor any input or playback track with the click of a mouse. You can even mix, pan and level control each track independently as you're tracking or recording to disk!

Switchable I/O Levels

Switch from professional +4dBu to consumer -10dBv, one at a time or change all 8 inputs and outputs at once. This means that the Aark24 interfaces to any mixer, MIDI instrument, or outboard gear. The signal levels for each analog input and output can be switched between balanced and unbalanced levels in the Aark Control Panel.

AARK 24

Word Clock Sync

Word Clock synchronization is standard on all professional digital audio gear and is used to keep “sample accurate” synchronization between all concerted digital audio devices in your studio. Aardvark’s legendary reputation for high quality digital audio synchronization is prominently implemented in the Word Clock I/O, allowing seamless integration and rock-solid sync with other digital audio gear.

DSP Engine

The heart and soul of the Aark 24 is a powerful Motorola 56301 DSP engine. The DSP handles all the software extras that are included in the control panel like digital peak metering, mixing, panning, level control and the flexible output router. It also manages up to four Aark 24 cards in the same PC.

USB3

Portable USB Interface with Mic, Guitar and Line In



The perfect system for musicians and guitarists worldwide, the USB3 eliminates the need for an external mixer, sound card, guitar stomp box or even other recording software. The USB3 comes with everything needed to record, mix and play audio from any computer. Just plug it in and go. Extremely portable, you won’t find an easier and better sounding audio experience on your laptop than the USB3—clean 24-bit converters, solid steel construction, premium mic preamp, brilliantly warm EFR Hi-Z guitar input and easy headphone monitoring.

Inputs/Outputs

Classic design includes easy access for headphones, mic & guitar input, stereo line I/O and knobs and sliders to control audio levels.

- Convenient 1/4″ mic preamp captures the audio with absolute clarity and definition.
- Selectable EFR circuit (Enhanced Frequency Response) Hi-Z input adds extra warmth and clarity to any guitar or bass. The EFR lets guitarists connect directly into their computer and get a sound that adds clarity and "punch" to any high-impedance instrument, and warmly enhances the sound of electric guitar or bass.
- Stereo 1/4″ line inputs and outputs give you full stereo I/O capability so you can record and play easier than ever before.
- The USB3 is powered from the USB bus of the computer. No extra power cord or bulky wall-wart adapter necessary.
- A clever mixing control surface allows for precise adjustments for all input and output audio levels. This gives you total control over your headphones, speakers and microphones, and gets you the best recording levels as well.

Highest Quality Sound

The USB3 sounds great because it uses high-resolution, 24 bit A/D converters so you're getting the most audio detail possible.

- Record at either 44.1 kHz for CD recordings, or 48 kHz when doing audio-for-video applications.
- Solid steel construction ensures proper shielding from magnetic and electrical interference—far surpassing built-in laptop audio, cheap sound cards and other plastic USB audio interfaces.
- Real-time input metering lets you maximize your input levels without clipping.
- Simultaneous Record/Playback feature lets you record and playback audio tracks at the same time, perfect for overdubbing, and it's enhanced by the high quality drivers which provide ultra low latency.
- Record Select switch allows separate monitoring of recording busses similar to working with an outboard mixer and sound card. This allows for up to 5 channels of low-latency monitoring for recording, mixing and playing back multiple tracks from any audio software.

Software Drivers

Record with one application, then transfer the audio files to another application and edit, mix and master to your heart’s content.

- ASIO drivers get down to 4 ms latency and guarantee seamless integration with ASIO programs like Cubase, Emagic Logic, Bias, Native Instruments and Propellerheads. This means you get real-time performance with soft synths & samplers and can monitor with VST plug-in effects in real-time.
- Control your buffer settings to get the best performance from your computer. The TURBO, FAST and NORMAL mode each have specific buffer settings so you're guaranteed uninterrupted audio with the best latency possible.
- Bundled with Cakewalk Guitar Tracks 2 for the PC and Cakewalk Metro for the Mac. Everything you need to achieve professional results with the USB3, they are creative music production tools simple enough for novice users to start recording, editing and mixing their own digital audio and MIDI tracks right away, but powerful enough for the experienced engineer.

AARDSYNC II

Low-Jitter Master Clock for Digital Audio

The AardSync II is the master word clock sync generator of choice for professional recording studios throughout the world. Not only does it provide a common word clock source for the digital studio, but it has become legendary in the industry for its sonic benefits. Innovative, proprietary digital technology ensures exceptional jitter reduction in the critically audible region. The resulting pure and precise audio clock signal enables today's 24-bit A/D & D/A converters to operate more efficiently, improving their accuracy and resolution. The benefits of clocking to an AardSync II are immediate and spectacular.





Back panel has 4 BNC outputs, 2 AES/EBU outputs and a video input BNC. These outputs can be configured in several configurations via internal jumpers

FEATURES

All digital audio equipment needs some form of master clock to derive its timing reference or to operate at a desired sample rate. Because only one clock source can be the master clock at any given time, problems can occur when multiple digital devices are synced or "daisy-chained" together. These problems usually manifest themselves in audible "clicks and pops" and a general loss in audio fidelity and stereo imaging.
As the sole master clock for the entire digital studio, the AardSync II solves these issues. It also gives the A/D and D/A converters a more accurate timing reference, causing them to operate more efficiently, which results in improved sonic clarity and stereo imaging.
To sync to an AardSync II, each piece of digital equipment should be connected to the card's BNC Word Clock, SuperClock (256), or AES/EBU outputs. For larger studios, additional outputs to the AardSync II can be provided by Aardvarks Word Clock or AES/EBU distribution amps.

- Low Jitter improves A/D D/A resolution!
- Eliminates clicks and pops
- Multiple Clock Outputs
- AES/EBU sync, WC, 2X WC & Superclock outputs
- NTSC 4% & PAL .1% Sample Rates
- Problem Free Synchronization
- 2x Word Clock for 96 kHz /88.2 kHz

What Does it Do

The AardSync II guarantees a sample-accurate sync lock between all digital audio gear in the studio, and will lock to video black burst if necessary. This means the audio workstation and everything connected to it will remain in perfect synchronization from beginning to end, recording to playback. The AardSync II can be set to generate 44.1, 48, 88.2 and 96kHz rates as well the standard Pull-Ups and Pull-Downs to easily handle film and video transfers. Keeps perfect sync between digital mixers, A/D/D/A converters, CD-Rs and DAWs. Compatible with digital audio gear from Digidesign, Avid, Mackie, Sonic Solutions, Tascam, Yamaha, and more.

What is Word Clock?
Word clock is a square-wave signal that is transmitted via a 75 Ohm BNC connector. A square wave alternates between high and low states, which makes it ideal as a timing reference. Because many devices can use word clock signals as their master timing reference, it has become a standard type of digital audio system clock.

Makes A/D Converters and Pro Tools Sound Better

When an A/D converter or Pro Tools 888/24 references the AardSync II's ultra low-jitter clock, it will sample the analog audio wave more accurately than when referencing to its own internal clock. This improvement in accuracy of A/D conversion results in:

- Increased clarity and definition for all musical instruments and voices.
- Bass frequencies become tighter, more focused and easier to place in the mix
- High frequency instruments sparkle and shimmer with clarity.
- Ambience, reverb and delay effects become easier to manage in the mix.
- Stereo imaging and separation between instruments is vastly improved
- Digital "harshness" becomes a thing of the past

Optional Racks:

19″ Mono Rack: The 2U Mono Rack is a 2 space, 19″ rack plate/rack tray enclosure that gives your Aardvark PC audio gear a truly professional look. The 2U Mono Rack is compatible with AardSync II, , Sync DA, or Aard DDA flush in the center of the 2U rack.

19″ Dual Rack: The 2U Dual Rack will mount any two AardSync IIs, Sync DAs, or Aard DDAs side by side in the same 2U rack.

B&H PHOTO - VIDEO - PRO AUDIO
ONE HOUR FREE PARKING
AT 349 W. 34th STREET (with purchase of $100 or more)

AARDDDA

1x6 AES/EBU Audio Distribution





The AardDDA is a 1x6 AES/EBU audio distribution amp that helps you avoid impedance mismatches, clock jitter & signal dropouts, and in general will make your digital life worry free. Combining the standard 1x6 AES/EBU I/O with the optional S/PDIF adapters, you can configure the input and the 6 outputs to perfectly match your setup. For example: S/PDIF input, 4 AES/EBU & 2 S/PDIF outputs. Or an AES/EBU input, 1 AES/EBU & 5 S/PDIF outputs. AardDDA lets you easily interface any number of consumer and professional pieces of equipment easily and efficiently. The AardDDA can also distribute AES/EBU or S/PDIF sync ensuring worry-free digital audio and digital sync distribution. And since the AardDDA's transformer isolated outputs are also individually buffered you can not only distribute a clean digital audio signal to different devices, but over longer distances as well.

- ◆ 1 x 6 digital audio distribution
- ◆ 6 individually buffered digital XLR (AES/EBU) outputs
- ◆ Filters out ground loop noise
- ◆ Channel Status: Source channel status is echoed
- ◆ No re-clocking.
- ◆ Optional S/PDIF I/O adapter available

SYNCDA *Word Clock Distribution*





Because only one clock source can be the master clock at any given time in the digital studio, problems can occur when multiple digital devices are synced or "daisy-chained" together. The Sync DA solves this by providing 6 additional Word Clock outputs from a single clock source. It allows any digital device, such as the AardSync II master clock or a digital mixer to be the master clock in any given situation.

The Sync DA feeds Word Clock on 6 BNC outputs, allowing each piece of connected digital equipment to receive a separate feed and be in sync with the master clock device. Two of the outputs can be configured to SuperClock 256, if the Sync DA is receiving an AES/EBU input. The SyncDA is a great way to expand the AardSync II's outputs, allowing larger digital audio studios to take advantage of its superior clocking technology.

Compatibility

Compatible with any digital audio device that has word clock or AES/EBU I/O, including:

Aardvark's AardSync II Master Clock Generator, Direct Pro Q10 and Aark 24

Digital Mixing consoles

Hard Disk multitrack recorders

Outboard A/D/D/A converters, CD-Rs

Features

Extremely low jitter distribution!

Makes any DAT, DAW or mixer the Master Clock

Distributes audio clock throughout the facility

Verify exact sample rate

Specifications

Clock Source Inputs:
Word Clock (BNC), AES/EBU (XLR)

Clock Outputs:

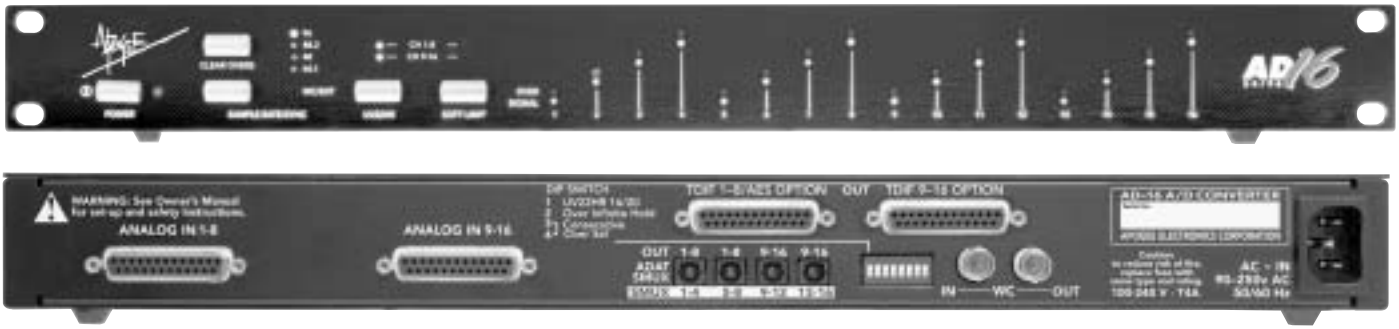
5 Word Clocks (BNC)

1 256 Superclock (BNC)

1 AES/EBU Thru (XLR)

AD-16

16-Channel 24-bit/96kHz A-D Converter

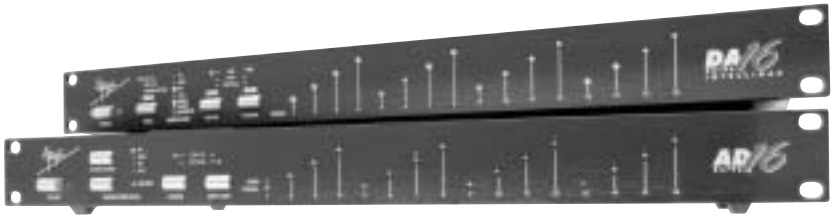


An affordable 16-channel 24-bit/96kHz A-D converter, the AD-16's powerful multi-channel configuration makes it the ideal front end for many digital audio workstations as well as for surround applications.
The system offers ADAT optical outputs and optional AES/EBU and TDIF, for compatibility with many digital audio systems, plus syncing to word clock or its own internal high stability clock. Apogee Word Clock output is also provided. Optical outputs on the AD-16 support the Sonorus S/MUX specification for sample-splitting of high sample-rate audio data into multiple ADAT-style optical interfaces. Additional light-pipe outputs are provided to deliver up to 96 kHz digital audio from all 16 channels in this mode. An expansion port allows the addition of optional TDIF or AES/EBU daughter cards.

FEATURES

- An elegant and effective user interface characterizes the AD-16, including a power switch, sample rate/sync selector, clear "overs" button, and two buttons for Soft Limit and UV22HR respectively. Designed to closely match its companion, the DA-16 (16- channel D/A) in appearance.
- Balanced analog inputs are organized into two groups of eight channels on 25-pin D connectors.
- LEDs indicate signal status on each channel, with the intensity modulated by the signal level to give an "analog-like" display. A second LED per channel indicates "overs", which may be user-configured. T
- Apogee's unique SoftLimit system maximizes digital output level without overs. These features are easily activated with a pair of buttons, and may be optionally applied to channels 1–8, 9–16 or all.
- Includes Apogee's industry-standard word-length reduction system, UV22HR. Reduce word length from the converter's 24-bits to 16-bit for CD mastering, Internet audio, etc, or 20-bit for DVD. High resolution detail is retained without creating artifacts.
- Rear panel DIP switches select regular ADAT versus S/MUX mode, over settings, UV22HR output resolution and infinite hold for the over indicators.

AD/DA-16 HD Pack—The Ultimate Pro Tools HD Upgrade

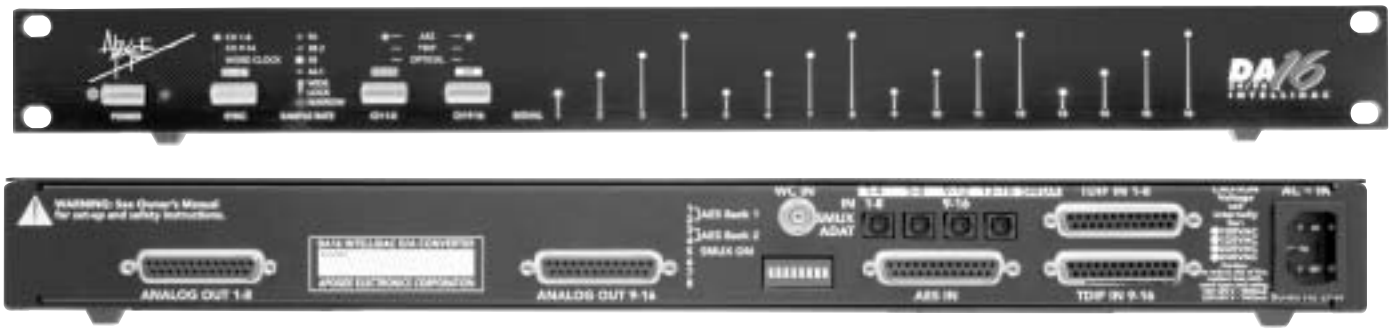


Put your money where your sound is… For a small percentage extra, you can audibly and significantly improve the quality of your Pro Tools HD system. The AD/DA-16 HD Pack gives you everything you need to add to a 192 Digital I/O. It includes the AD-16 (with AES Expansion card), the DA-16 and the special digital interconnect.

- Digidesign's 192 Digital I/O is a 24-bit, 192 kHz multichannel interface designed to allow the interconnection of digital audio sources with the Pro Tools HD environment. 16 channels of AES/EBU, TDIF and ADAT I/O, along with S/PDIF I/O.
- Connect the AD-16 and DA-16 to the 192 Digital I/O using AES/EBU at up to 96 kHz. A special cable, part number AES16-DIGI-IFC, makes the direct connection between the AD/DA-16 and the 192 Digital I/O.

DA-16

16-Channel 24-bit/96kHz D-A Converter



Designed to interface seamlessly with the latest hard-disk recording systems, the DA-16 'IntelliDAC' offers 16 channels of Apogee quality and 24-bit/96kHz D/A conversion in a compact, convenient 1RU package, at an exceptionally affordable price. The converters can source their digital input from either AES/EBU, ADAT optical or TDIF sources, and the unit can sync to word clock or a specified input. ADAT and TDIF inputs are organized into two groups of eight, and channels 1-8 and 9-16 can have different sources. In addition, the light-pipe inputs accept the Sonorus S/MUX protocol for sample-splitting high-resolution signals into optical interfaces, allowing access to all 16 channels -the first time this protocol has been included on an Apogee converter.

FEATURES

- Input sample rate is automatically detected, and a two-level "Lock" indicator shows "wide" (up to ±150°) and "narrow" (5°) lock.
- LEDs indicate signal status on each channel, with LED intensity modulated by the signal level to give an "analog-like" display.
- Balanced analog outputs come in groups of eight channels on 25-pin connectors, in keeping with other Apogee D/A conversion systems using these connectors (the pinout is the same as on Tascam systems).
- "IntelliDAC" relates to the DA-16's unique "intelligent" two-stage re-clocking system. Apogee excels at removing jitter from the incoming clock signal, but for extremely jittery input sources, even more control is needed.
- The DA-16 does this by using two clocks. A fast-responding 'read' clock, with a wide locking range, fills a dedicated FIFO buffer, while an ultra-low-jitter 'write' clock writes the data out of the buffer, and is used to clock the converters. The advantage is that both incoming clock and data are now de-jittered.
- The system is also less sensitive to phase errors between synchronous digital sources. Errors up to ±150° can be corrected, substantially reducing the chances of glitching, and enabling the D/A converter to offer superior performance even with extremely unstable input signals.
- A calibration mode is also included, allowing quick and easy adjustment of the reference level for each channel.

NativeTools Studio Edition



The power of Nuendo; the sound of Apogee. The ultimate pro-audio combination! Combine Apogee's superb-sounding AD-16 & DA-16 16-channel 24/96 converters with the power of Steinberg's Nuendo workstation - on either Macintosh or PC - and you've stepped into a new world. That's Apogee's NativeTools: the ultimate combination of sonic quality, native processing power and value for money - all in one package. If you're looking for the ultimate pro-audio combination, NativeTools is the answer. Everything you need (except the computer) in one box for the complete DAW system that is taking the audio world by storm.

APOGEE DIGITAL

MINI•ME

2-Channel Mic/Instrument Preamp



The Mini•Me is a portable 2-channel 24-bit/96kHz A/D converter with built-in mic/instrument preamps, featuring a special low-power, wide range supply-voltage design for maximum flexibility. But this compact unit is not a toy. It is a professional audio product and delivers true Apogee professional-quality audio performance, from input to AES/EBU and S/PDIF digital outputs. And with a low-latency USB interface, you can use almost any software you choose. Other features include headphone out with level and Direct/USB Return Mix knobs, unique multicurve comp/limiter, and flawless 16- and 20-bit output using UV22 technology.

FEATURES

- Front panel feature left and right input level controls with a click stop at the far left that activates a preset line/cal level, set with multi-turn trimmers. Accurate LED indicators between the input level controls give a clear display of available headroom.
- Two-channel universal XLR/TRS mic and instrument inputs with preamps and phantom power, and line-level input. (Instrument input is automatically selected when you plug in a TRS jack.)
- A single control selects the sample rate and word length. The Mini•Me's converters output a full 24-bit signal at any of the standard rates: 44.1, 48, 88.2 and 96kHz sampling. In addition, select 16- or 20-bit outputs at 44.1 or 48 kHz using Apogee's industry standard UV22 HR system for word-length reduction.
- Even if the sample rates get mismatched between the Mini•Me and your computer, it will convert rates automatically. And if you select 24-bit on the Mini•Me and 16-bit on your recording software, the unit will apply UV22HR automatically to the USB output.
- Apogee has long been famous for their Soft Limit process, designed to maximize digital level without overs – but now they've added Push-IT – a unique three-curve stereo compressor/limiter circuit. It's ideal when you need extra punch, or require a safety net when making live recordings. This powerful circuitry takes Apogee's dynamics control to a whole new level – from the people who invented soft limiting.
- A comprehensive mono/stereo monitor section allows you to balance the direct sound you're recording with audio returned via the USB port.
- Low-latency USB interface carries two channels 24-bit/48 kHz audio to and from the computer for monitoring. ASIO drivers are also supplied for minimum latency.
- Includes AES/EBU and S/PDIF (RCA/coax) outputs which are available simultaneously, clocked by a high-stability reference crystal oscillator. Makes it the ideal master clock.
- Low power consumption makes it ideal for battery operation. The supplied power unit supplies 12v, but the Mini•Me will run on AC or DC from 6 to 16 volts.

MINI-DAC

2-Channel 192kHz D/A Converter



Following the huge success of Mini-Me Apogee's latest addition to their Mini Series, the Mini-DAC is a professional quality,192kHz D/A converter—and the ultimate portable and compact solution for studio playback, reference monitoring, USB connectivity to your DAW, and premium home audio systems. Mini-DAC provides AES, optical (ADAT, S/MUX & S/PDIF) S/PDIF Coax, USB inputs, and analog output level control. Also included with Mini-DAC is "digital-thru-mode", a unique USB technology allowing the connection of non-USB digital devices to a computer/DAW.

- If you are using OS X version 10.2 or better, enjoy true plug and play with Apple's Core Audio. Simply plug Mini-DAC into your computer via USB and it will appear as a hardware option in Logic, Bias, Peak, and other Core Audio compatible DAW's.

"Digital-Thru-Mode"

These days, if you are doing digital, you need connectivity options. Through Mini-DAC's USB port, you can interface directly with your computer and virtually any DAW via. However, Apogee has expanded the advantage USB capability to offer what they call "digital-thru-mode". This gives the option of interfacing non-USB digital devices with your computer. To use, you simply set the Mini-DAC's input selector to "USB"… connect any two-channel digital device like Apogee's Mini•Me, Trak2, or Rosetta, and you have just USB enabled your favorite A/D converter and interfaced it with your computer/DAW. No other hardware required.

Precision Reference Monitoring

Are you using a massive mixer just to ride your monitor levels? Or do you have a great A/D but inferior D/A? Then make a little room for Mini-DAC. With comprehensive level control and a high quality headphone output you can hear exactly what you are committing to while recording. And when it is time to mix down, Mini-DAC provides the optimum stereo image and the most faithful analog reproduction of your carefully recorded digital audio that money can buy. Mini-DAC is also a great way to optimize your home audio system. Connect your premium CD/DVD player to Mini-DAC and then out to your speakers for a superb sonic experience that your discerning ears will appreciate.

Additional Features

- Inputs include AES (single & double wide), Optical (ADAT, S/MUX & S/PDIF), S/PDIF Coax, & USB
- Support for OSX Core Audio and USB drivers available for Windows XP
- Analog output level control for direct connection of powered monitors
- Low-current, low-voltage - ideal for location/ENG
- Stylish, compact and portable (5.5 x 10″)
- 1/3 rack width with a 1U rack-mount kit available

ROSETTA

48k and 96k 2-Channel A/D Converters



Ideal for project studio recording and mastering, the Rosetta A/D is a 24-bit analog to digital converter with the most common digital interfaces as standard. Like the Rosetta Stone, discovered by Napoleon's soldiers in Egypt in the 18th century which provided the key to translating the mysterious hieroglyphs of the ancient Egyptians, Apogee's Rosetta A/D is the key to flawless translation of analog signals into multiple digital formats. And, the feature of the Rosetta A/D has been carefully tailored to project studio requirements—you don't pay for features you'll never use.

- Analog inputs operate at pro or consumer levels, balanced or unbalanced. The highest quality Apogee 24-bit conversion, at 44.1/48 kHz sample rates (standard version) or 44.1/48/88.2/96kHz, feeds two AES/EBU outputs so you can feed two machines at once (44.1/48 kHz version is upgradeable to 96kHz).
- Built-in ADAT, TDIF, and S/PDIF coax/optical interfaces (two of each)
- User-defined over indication
- Easy to use front panel. Every function has its own control, clearly labeled. Metering features easy-to-read bar-graph LEDs with oversize over indicators and you can determine how many consecutive digital full-scale samples constitute an over.
- Soft Limit technology lets you record at the maximum level without overs, while the high-resolution UV22HR process lets you generate flawless 16- and 20-bit masters retaining as much of the original 24-bit resolution as possible.

ROSETTA 800

8-channel, 24/96k A-D/D-A Converter (upgradeable to 192k)



The Rosetta 800 takes two of Apogee's most celebrated products and the latest in high-definition digital and combines them into one impressive package. The Rosetta 800 gives you eight channels of superior AD/DA conversion at sample rates of up to 192k, Apogee's "SoftLimit", "UV22HR", and their advanced "Intelliclock". The result is a premium digital recording solution that is built for the professional recording facility yet economical enough for the project studio.

All the features of the Rosetta A/D PLUS—

- 96k standard and optional 192k sample rates
- Optional Firewire card lets you connect natively to any Firewire input device, without additional hardware.
- Intelliclock is really two clocks in one. A fast-responding 'read' clock, with a wide locking range, fills a dedicated FIFO buffer, while an ultra-low-jitter 'write' clock writes the data out of the buffer, and is used to clock the converters.

TRAK2

2-Channel Mic Preamp, DI and 24/96 Conversion System



Trak2 is the ultimate analog/digital combination. With the Trak2, you have it all, in one compact single-unit rack space: superb mic pre; incredible A/D; any interface you need; digital routing; and behind it, Apogee's legendary name and reputation for the very highest audio quality available in the industry today.

Trak2 features the highest-quality 2-channel mic preamp on the market. ±90 dB gain, front panel inputs, analog inserts/outputs and phantom power. Behind the mic preamp is a full 24-bit/96 kHz, 117 dB dynamic range Apogee A/D converter. Built-in AES and S/PDIF interfaces, plus the ability to plug in cards for offer almost any interface you can imagine including Pro Tools, ADAT, TDIF and more. Plus there is optional video sync, optional 2- or 8- channel D/A cards with virtually the same incredible specs, eight-channel routing and format conversion, a serial/MIDI connection, a powerful software-controlled interface giving access to a host of advanced features, and firmware you can update over the Internet.

FEATURES

Superb Mic Preamp...

The Trak2's discrete mic preamp is accessible via rear panel XLRs or the front-panel universal XLR/TRS connectors. The sound of this preamp will have you using it on every session, analog or digital. The preamp also accepts hi-Z instrument inputs at the front panel, for maximum versatility. An insert point is included to provide an analog output or to allow the insertion of analog processors into the Trak2 signal path.

...incredible A/D

Apogee's award-winning 2-channel, 24-bit A/D converter operates at 44.1, 48, 88.2 and 96 kHz with a dynamic range over 117 dB. The A/D also includes Apogee's Soft Limit with digital adjustment control for maximum digital level without overs, and Soft Saturate with digital adjustment to simulate analog tape compression. Both can be switched in and out on one or both channels.

Add a D/A

A special slot for an optional 2- or 8-channel D/A card with specifications matching those of the A/D section, turns the Trak2 into a complete digital audio conversion system with two A/D channels and up to 8 D/A.

Interface Versatility

Two Apogee Multimedia Bus (AMBus) slots are provided in addition to the built-in AES/EBU and S/PDIF output to interface with Pro Tools, ADAT, TDIF, SDIF-II, SSL HiWay, etc. Many cards support Apogee Bit-Splitting, which enables you to use standard 44.1/48, 16-bit recorders to store 20-, 24-bit, and even 96kHz signals.

UV22 HR

Apogee's industry-standard UV22 HR (High Resolution) system takes a high-resolution digital signal and reduces the word-length to 16 or 20-bits for CD or DVD mastering while maintaining 24-bit detail. Trak2 also includes Apogee's Ultra-Low Jitter Clock, used to lock the unit to incoming digital or video signals and virtually remove jitter.

Front Panel

- Clear LCD display shows configuration and signal flow.
- Cursor controls/delta-wheel for fast settings changes.
- Programmable Quick Keys for instant access to frequently-used functions.
- Easy-to-read 2-channel LED bar metering
- User-definable over detection

Additional Features

- 10 to 8 Digital Routing
- Low/High settings for multi-pass and mastering operation.
- Adjustable Soft Limit with digital adjustment control for maximum digital level without overs.
- Soft Saturate with digital adjustment control simulates analog tape compression
- MIDI In/Out/Thru and serial port
- Programmable 'Quick Keys' for instant access to frequently-used functions.
- 96kHz Headphone D/A with high output 40W power amplifier.
- User saveable/recallable presets
- Video sync capability
- Clear LCD display shows configuration and signal flow
- Cursor controls/delta-wheel for fast settings changes
- Easy-to-read 2-channel LED bar metering
- User-definable over detection (1-4 digital full scale)
- Ideal companion for Steinberg's Nuendo system

TRAK2

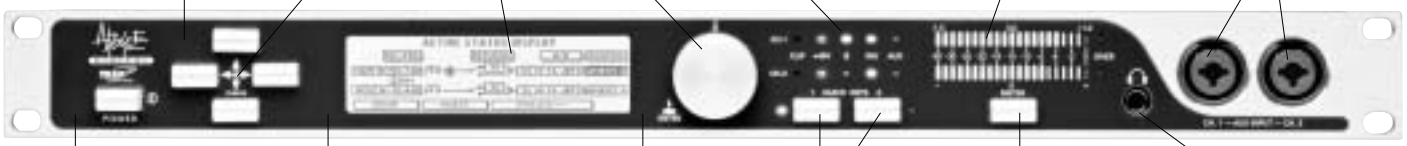
The Trak2 features Apogee's stunning 24-bit digital conversion. The quality of your A/D converter defines the quality of the entire digital signal from then on – so you need the best. Is it a superlative 2-channel mic preamp followed by the highest quality A/D, or a world-class A/D preceded by the best-sounding mic pre you've heard? Either way, you'll notice the Apogee difference.

The Trak2 is controlled from the large graphical backlit LCD display, which indicates signal flow and the status of the different parameters. Navigate with the cursor keys; change values with the delta wheel. Push the wheel to set the value.

LEDs indicate phantom power, polarity, insert and aux input status

Multi-segment, multi-mode peak/average LED meters give a clear indication of level, with user-defined over indication. Meter either before or after your recorder. Super-high resolution calibration mode, and a phase meter to keep you on track.

Front panel left and right connectors are universal XLR/TRS jack types and accept mic, line, balanced or unbalanced, –10 dBV or Hi-Z direct instrument inputs. TRS connectors are selected automatically when an instrument is plugged in.



Apogee's proprietary technologies are selectable on a per-channel basis. Soft Limit allows maximum level without overs; Soft Saturate adds the warmth and punch of analog tape compression.

Apogee's UV22HR system preserves maximum detail when reducing the Trak2's 24-bit conversion to 16- or 20- bit for CD or DVD. Low (multi-pass) or Normal (mastering) settings for maximum flexibility.

With an optional video card the Trak2 can be set to lock to PAL, NTSC and B&W video, with sample rate pull-up and pull-down.

Programmable keys pro-vide instant front-panel access to any function in the Trak2. Each Quick Key can store two functions.

Selects the meter mode – peak, average, hold, etc – and clears the over indicators when required.

Built-in stereo monitor D/A and headphone amplifier allow instant checking of selected signal pairs passing through the unit.

Balanced or unbalanced, the mic preamp offers up to ±90 dB of gain, adjustable from the front panel.

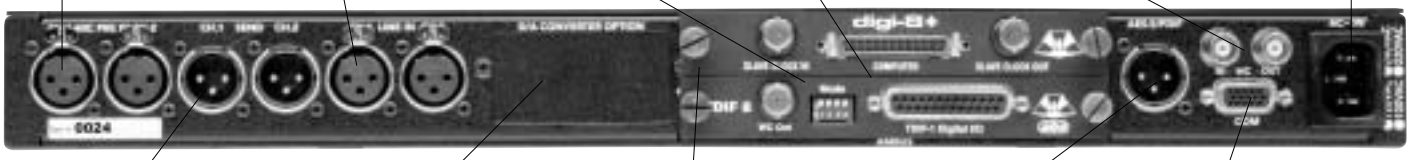
+4 dBu or -10 dBV, balanced or unbal-anced, the line level inputs provide direct access to the A/D section.

Two 8-channel AMBus slots are provided, sup-porting cards like those in the AD-8000. Interfaces include a direct Pro Tools connection, ADAT, TDIF, SDIF-II (DASH), SSL HiWay and others.

All installed interfaces can be used simultaneously; any signal input to the Trak2 is output from all available interfaces. AMBus technology allows full 8-channel digital format con-version, for maximum ver-satility in the studio.

BNC connectors provide word clock to other devices, and synchroniza-tion to word clock in. The input is also used for optional video sync.

The super-smooth linear power supply is built around a custom toroidal transformer, minimizing interference and avoiding potential spikes and noise. A generously-specified exposed heat sink is included.



Inserts allow you to patch analog processing gear between the mic pre and the A/D converter, or to derive an analog output from the mic pre-amp alone for processing elsewhere.

Plug in a 2- or 8-channel D/A converter card to build a complete conversion system. Cards offer the same 24-bit capability and superb performance as the A/D.

With the Digi-8+ AMBus card, the Trak2 becomes a unique Pro Tools interface, communicating direct with the computer for super-quality performance.

AES/EBU &S/PDIF output irrespective of the presence of AMBus cards, enabling the Trak2 to be used straight out of the box.

High-density connector provides a serial connection for transferring firmware updates from the Internet, planned MIDI remote con-trol capability, and more. Cable included.

PSX-100

2-Channel 24-bit/96kHz A-D/D-A Conversion System



Apogee's first 96kHz system, the PSX-100 offers many of the features of the Trak2 but without mic pre-amps and strictly 2-channels. The PSX-100 is equipped with built-in ADAT, TDIF, S/PDIF coax and optical, plus multiple AES/EBU interfaces. It features Apogee's proprietary UV22HR word-length reduction system (which can be applied to either the A/D or the D/A) and SoftLimit for maximum level without overs. An auxiliary output allows a full-resolution output to be derived even when UV22 is in use on the main interfaces. The wide range of standard interfaces provided and the multiple AES I/O ports allow use for signal distribution and format conversion as well as for A/D and D/A applications.

FEATURES

Three Operating Modes:

- The PSX-100 has separate A/D and D/A sections which may be clocked separately and configured to work together in different ways. The digital inputs and the A/D output are fed into a routing section which sends signals to the D/A and Aux output or to the main digital outputs, depending on which modes is selected.

—In Confidence Monitor mode the A/D and D/A are independently clocked (and can even run at different sample rates). The output of the A/D is available at all the digital outputs, and the D/A can receive its signal from any of the available digital inputs. The bargraph metering can be switched between A/D and D/A. Two separate signal paths through the unit in this mode, other modes utilize a single path.

—Digital Copy mode takes the selected digital input and provides it to all the PSX-100's outputs, digital and analog. Both clocks in this mode are synchronized to the digital input. This mode is ideal for format conversion as well as offering the maximum D/A flexibility.

—In Analog Monitor mode, the A/D is fed to all the system outputs and to the D/A. The main clock is used as the reference (locked to crystal or an external digital source), while the auxiliary clock is slaved to it.

- Two channels of Apogee's true 24-bit A/D and D/A conversion with 117 dB dynamic range. Level trim of A/D and D/A levels via easy screwdriver adjustment for instant calibration and gain structure alignment.
- Apogee quality at 44.1, 48, 88.2 and 96 kHz
- Built in interfaces include AES/EBU, ADAT, TDIF and S/PDIF (coax and optical)
- Balanced or unbalanced analog I/O
- Exclusive active self-balancing analog I/O
- Gold-plated XLR jacks for analog and AES/EBU I/O, RCA jacks for S/PDIF I/O
- AES/EBU distribution amplifier capability
- Monitor the A/D output or select S/PDIF coax or optical; ADAT; TDIF or either of two AES/EBU inputs (one AES input at 88.2/96kHz sampling). Synchronize the D/A to digital in or to the A/D sync source.
- Apogee Soft Limit helps you record at a higher level without overs
- Easy-to-read light-bar metering, includes "over" indication with 2-second/infinite hold. User-defined "over" setting (1-4 consecutive digital full-scale readings). Meter the A/D or D/A in "confidence mode". Calibrate with ±0.1 dB setup accuracy.
- Individual L-R channel muting at the touch of a button, instantly controlling the signal from the main digital outputs.
- Two proprietary Apogee low-jitter master clocks
- Optional video sync card allows the PCX-100 to lock to NTSC/PAL video
- Apogee Bit-Splitting (ABS) modes allow the two-channel 24-bit signal to be recorded using pairs of tracks of a 16-bit 44.1/48kHz sampling recorder, via ADAT, TDIF or AES/EBU. In ABS-96 mode, you can record stereo at 96kHz sampling on a 16-bit 8-track.
- UV22HR encoding process translates 24-bit signals to 20 or 16-bits with no quality loss (Not on 88.2 & 96kHz sample rates).
- Word Clock I/O on BNC connectors. The WC input socket can act as a video input when the video option card is installed.
- A 10-way DIP switch allows 'set-once' parameters for analog input/output level; pin 2/3 hot; 2-second or infinite 'over' indication; 'over' definition; ABS enable on AES input and special sync parameters.
- MDM input selection lets you select the pair of tracks you want to listen to (or the set of tracks for replaying a bit-split source).
- Special Edition (PSX-100SE) available with upgraded op-amps, along with additional level of power supply filtering. The D/A section includes a new third-order Bessel filter, the A/D includes a specially-designed anti-alias filter that smooths top end response.

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BIG BEN

Studio Master Clock and Word Sync Distributor



For years audio professionals have been asking when Apogee was going to make their legendary clocking technology available as a stand-alone master clock. Introducing Big Ben, a radical new studio timepiece that will set the pace for all master clocks to come.

FEATURES

C777 Clock Technology up to 192kHz

At the heart of any word clock regeneration is a phase lock loop (PLL). Currently, and most commonly in audio applications, the PLL is a combination of analog and digital elements without the ability to dynamically adapt to the nature of the external clocks. That's where Apogee's C777 clock comes in. The C777 utilizes an entirely digital process, that Apogee has developed using the most advanced Direct Digital Synthesis (DDS) technology available along with DSP based digital filtering. The result? The most aggressive jitter reduction ever. With a stable, crystal based digital PLL ticking away as it's heart, Big Ben is able to intelligently manipulate incoming signals and adapt to them accordingly.

Intelligent Clocking with Adaptive Loop Filtering (ALF)

In digital audio, as in comedy, timing is everything. But if your audio path has got the jitters that's no laughing matter! Enter Apogee's exclusive Adaptive Loop Filtering technology. ALF maximizes Big Ben's clocking ability with an intelligent low pass filter. Because the C777 has made all incoming signals into data (1's and 0's) Big Ben is able to dynamically compensate for excessive jitter and even improve already acceptable clock signals to levels unattainable until now. No matter what your input looks like, ALF can adapt and overcome all obstacles to create a stable, reliable, low jitter clock source.

Realtime Format Conversion

- Not that you would dare expect it from a master clock, Big Ben gives you realtime format conversion between all digital formats. Use Big Ben's intuitive digital display to go from S/PDIF to AES/EBU, AES/EBU to ADAT, ADAT to S/PDIF, etc.
- AES, S/PDIF, Optical I/O - Word Clock/Video In - 6 Word Clock Outs
- Optional Firewire connection to facilitate clocking and format conversion with other Firewire devices

"SureLock"

What happens when Big Ben encounters the ultimate disturbance in clocking, a dropped signal? With "SureLock" Big Ben will compensate by remaining locked on the last relevant frequency sent by the delinquent device. When the signal resumes, Big Ben will gradually, and smoothly re-synchronize without interruption.

Termination

Another feature to help you stop watching the clock is Big Ben's termination sensing ability. Very often in an elaborate audio chain word clock signals are improperly terminated. Big Ben will chime in with a visual indication of over or under termination on it's digital display for each word clock output, allowing you to troubleshoot more quickly and get back to business.

Inputs:

- 2 x AES/EBU on XLR 44.1-192kHz single-wide and 88.2 k–192k double-wide.
- S/PDIF optical on TOSLINK 44.1-48k
- S/PDIF coaxial on RCA 44.1-192k
- ADAT 44.1-48k
- ADAT/SMUX II for 88.2-96k
- ADAT/SMUX IV for 176.4-192k
- Word Clock BNC 44.1 –192k
- Video
- Firewire card option provides clocking and format conversion functionality to Firewire equipped devices.

Outputs:

- 2 x AES/EBU on XLR single or double wide 44.1-192k
- S/PDIF coaxial 44.1-192k
- S/PDIF optical on TOS-LINK 44.1-48k
- 6 x BNC Word Clock 44.1-192k (of which 2 can output 256fs at low sample rates)
- ADAT
- SMUX II/SMUX IV
- Optional Firewire

Specifications:

- Sample rates: 44.1, 48, 88.2, 96, 176, 192kHz
- Video: PAL/NTSC/60Hz
- "SuperClock" (256fs)
- Pull up/down 0.1% and 4% from any sample rate
- Three stage termination indicator (under, correct, over) on each Word Clock output
- 4 digit numeric true sample rate indication

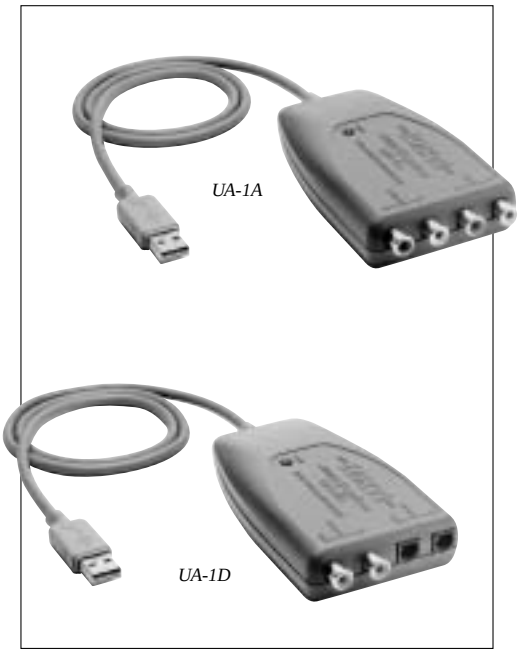
EDIROL

UA-1A/UA-1D

USB Audio Capture Interfaces

The UA-1A (analog) and UA-1D (digital) are straightforward USB audio I/O devices that turn your USB-equipped computer or laptop into a digital audio recorder— quickly and easily. Convenient USB connection provides clean digital data transmission as well as power – they are completely powered by the USB bus. And they are MME (WIN) and Sound Manager (Macintosh) compatible, so they will work with a host of digital audio recording software packages for both Mac and Windows.

The UA-1A is a mouse-sized device for analog input/output. Internal D/A and A/D converters operate at 16-bit/44.1kHz. A pair of stereo RCA input/output connections allows you to capture audio from CD players, keyboards or any analog device with line level output connections. The UA-1D Digital Audio Capture comes in a similar package, but offers both optical and coaxial S/PDIF digital connection, allowing you to connect any S/PDIF device, including component CD recorders, MiniDisc players/recorders, digital home receivers, DAT, etc. The UA-1D is also packaged with WinDVD so you can send Dolby Digital or DTS encoded audio to your home theater system.





UA-20

24-bit USB-powered Audio & MIDI "Dual" Interface

A complete USB audio recording device, the UA-20 brings portable 24-bit quality to your Mac or PC applications. Your laptop audio is now 24-bit resolution, powered by your computer through the USB bus, and flexible! Easily connect your guitar, mic and line level input with the portable and affordable UA-20.

- Two Line/Guitar/Mic inputs
- Two line (RCA), S/PDIF (optical) outputs
- Fast and stable 1x1 MIDI input/output with FPT
- Direct Monitoring function allows for low latency performance while recording.
- ASIO 2.0 (Win/Mac), MME, WDM drivers and Sound Manager (Mac) compatibility lets you create music with any music software program for Mac or PC.
- USB connection supplies power as well as data
- Headphone (1/8″) jack and volume control

FPT (Fast Processing Technology)

FPT is a technology that allows for high speed, high resolution MIDI data transmission. Using a combination of driver and hardware improvements to take advantage of the high-speed extensive transmission capability of USB, FPT can deliver high speed, low latency, low jitter MIDI transmission across multiple ports, while maintaining high data resolution.

Driver tuned for FPT

The driver for FPT-based hardware devices uses an optimal transmission method depending on the amount of MIDI data being transmitted. The driver then effectively utilizes the bandwidth of USB (depending on the amount of MIDI data) allowing communication to be stable and very fast.

Hardware tuned for the FPT

FPT also allows for high speed, high resolution processing through hardware using HDMR.

The high speed transfer capability of FPT allows for high resolution data transfer without compromising the performer's subtle expressions even in live performance.

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ONE HOUR FREE PARKING
AT 349 W. 34th STREET (with purchase of $100 or more)

UA-3D

Portable USB Audio Interface with 5.1 Surround Sound Output



USB connectivity brought a revolution in audio interfaces for your computer. Now, Dolby Digital and DTS brings surround as well! The UA-3D allows transfer of 5.1 digital data from your computer via USB to an outboard decoder for a full theater-style listening experience. The UA-3D is a portable, palm-sized USB audio interface, ready to use on both desktop and laptop computers. It features inputs for analog line signals, S/PDIF digital (optical), Hi-Z guitar preamp and 1/8˝ minijack selectable dynamic/condenser mic.
Includes Win DVD playback software and Cool Edit Pro LE for audio recording and editing. USB cable provides power and computer interface with no need for batteries or power adapter.

- Two RCA analog and S/PDIF digital (optical) input/outputs
- Guitar and microphone inputs
- High quality sound with USB audio capture and UA-3D's A-D and D-A converters
- Bundled WinDVD and Cool Edit Pro LE (Windows) lets you enjoy surround sound and experience theater style movies. (DVD-ROM and a DTS or Dolby Digital decoder is required for 5.1 surround sound.)
- Ideal for laptops, the UA-3D receives power supply from the computer via USB cable. No need for AC adapter. Connect your audio equipment to your Mac or PC with the UA-3D and just one USB cable.



UA-5

Compact 24-bit/96kHz Processing USB Audio Interface

Compact and stylish, the UA-5 is a complete USB audio recording system. Ready for use on both desktop and laptop computers, it provides the widest range of inputs, including phantom- powered mic inputs, optical and coaxial S/PDIF, and a Hi-Z guitar preamp. A comprehensive array of switches and buttons on the front panel allows for extremely fast and intuitive operation, ranging from gain control, sample rate switch and a unique Advanced mode (ADV) which enhances the hardware performance of the UA-5 when used ASIO or WDM enabled software.

- Two front-panel 1/4˝ TRS input connectors (XLR, Guitar, balanced/unbalanced)
- Two 1/4˝ TRS outputs (balanced or unbalanced) and stereo line (RCA) output
- Coaxial and optical S/PDIF I/O
- ASIO (Windows/Mac), MME,WDM drivers and Sound Manager (Mac) compatible
- 48v phantom power
- Unique switch for both native (ASIO) and generic drivers to enhance accuracy in recording environment with low-latency
- Direct monitoring function allows for zero-latency performance while recording/overdubbing
- Portable design, offers easy connection to computer through a single USB cable
- Self-powered via USB when not using phantom power or Advanced Mode
- Compatible with most sequencer/audio recording software programs

EDIROL

UA-700



24-bit/ 96kHz USB Digital Audio Interface with COSM Technology

Simply call it "The Everything Box", because that's just about what is has. The UA-700 combines a high resolution USB audio interface with a built-in effects processor and MIDI Interface into one home studio recording powerhouse. Plus, it has the power of COSM- Composite Object Sound Modeling, so you can tweak that mic to be just the way you want it. It even works as a standalone effects processor. Be careful- if you get your hands on it, you'll be hooked!

FEATURES

Versatile I/O Capability

- Two combo inputs, XLR/phones, +48v phantom power for condenser mic
- Input for guitar
- RCA Line/Phono input and RCA output
- Optical/coaxial S/PDIF input & output
- Low latency with Direct Monitoring
- ASIO 2.0 (Win/Mac), WDM (WinXP/2K), and Apple's Sound Manager compatibility

System Effects Section

- The UA-700 has a wide variety of effects processors including Noise Suppressor and Equalizer, that are indispensable for computer recording. Other effects include Reverb and Chorus to make it sound like you recorded your music in a concert hall.
- AUX effects: Center cancel
- System effects: 3-band EQ, Noise suppressor, Reverb (5 types), Chorus

Microphone Section

UA-700 combines Mic modeling engine of COSM technology. Control five kinds of high-quality microphone sounds with various characteristics. Features its original preset compressor for mic recording and also de-esser, a must for vocal recording. (Guitar/Mic effect cannot be used simultaneously).

– Reference Microphone: DR-20, Small Dynamic, Head-worn Dynamic, Miniature Condenser, AKG C3000B, Flat

– Modeling Microphone: Small Dynamic, Large Dynamic, Small Condenser, Large Condenser, Vintage Condenser, Flat

– Preset type compressor: Normal, Vocal, Kick, Snare, Kit, Acoustic guitar

Guitar Section

Powerful COSM engine provides a wide range of guitar sounds from heavy distortion with tube amp to soft distorted crunch-sound or clean sound. Sounds include delicate tone control of volume and picking. Tweak 11 guitar-amp sounds from every guitar sound in history with effect knobs.

– Amp modeling: JC Clean, Crunch, Lead, Black Panel, Tweed, American and Brit Combo, Vintage, Modern and Metal Stack

– Speaker Cabinet type: Original, 4x12″, 4x10″, 2x12″, 1x12″, OFF

– Guitar effects: Flanger, Tremolo, Phaser, Delay, Compressor

UA-1000 *High-Speed USB Audio Recording Interface*



Ideal as the centerpiece of a project studio and equally at home in your mobile studio, the UA-1000 is the world's first multi-port Hi-Speed USB (USB 2.0) audio recording interface. It delivers exceptional audio capabilities on 10 separate channels via a wide variety of connections, offering high quality microphone preamps using premium analog components. These include XLR/TRS combo jacks, Hi-Z guitar, S/PDIF, 8 channel ADAT, MIDI, and even inserts for effect send/return.

- Hi-Speed (480 Mbps) USB 2.0 support allows the UA-1000 to offer 10-in/10-out 24-bit/96Khz full duplex performance
- 4 front-mounted XLR/TRS combo jacks, with phantom power
- High quality microphone preamps
- S/PDIF (optical/coaxial), ADAT, MIDI, Word Clock
- 4 inserts I/O for effects sends and returns
- Zero latency direct monitoring
- Striking blue 1U high metal rackmount

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DA-2496

24-bit/96kHz 8-in/8-out Digital Audio Interface



The DA 2496 is a rack-mount system with a whole host of connections to help your computer become the center of your recording studio. Whether you are running to your computer from an outboard mixer or directly plugging your instruments into it, the DA-2496 is the best way to get from the instrument to the computer. Record multiple instruments, vocals and performers—each on their own track—all at the same time. Balanced and unbalanced inputs with individual gain control and 48v phantom power as well as digital audio connectors are conveniently located on the front and rear panel. Furthermore, the inputs on the front of the unit can supercede the inputs on the back. So if you have your DA 2496 setup in your rack but want to plug a bass directly into one of the Hi-Z inputs on the card, (no need for a direct box) you don't need to unplug the back of the unit.

- Up to 24-bit/96 kHz AD/DA conversion
- Up to 8-in/8-out simultaneous multi-track recording on PC and Mac via PCI interface.
- Multiple choice of balanced TRS connectors, including mic inputs with 48v phantom power and +4/-10 dB operation
- Direct input for guitar (Hi-Z)
- Optical/coaxial S/PDIF input/output on dedicated channels
- MIDI in/out
- Quick access to digital input sources on front panel—great for CD/MD/DAT inputs
- Full compatibility with the most popular sequencer/audio recording software
- Word clock synchronization via BNC input/output
- ASIO2.0 (Windows/Macintosh), MME WDM drivers and Sound Manager (Macintosh) compatibility
- 1U-rack mount type break-out box & PCI interface card

	UA-1A	UA-1D	UA-20	UA-3D	UA-5	UA-700	DA-2496
Main Function	Digital Audio Interface	Digital Audio Interface	Digital Audio Interface 1-in/1-out MIDI Interface	Digital Audio Interface	Digital Audio Interface	Digital Audio Interface with effects	Multiple 8-in/8-out Digital Audio Interface
Unique Feature	Cable style	Cable style	Cable style	5.1 channel Surround Dolby Digital output	24-bit/96 kHz	COSM effects controlled all by knobs	24-bit/96 kHz Word clock sync IN/OUT
Connection to Computer	USB	USB	USB	USB	USB	USB	PCI bus
System Compatibility	WindowsXP/Me/98/2000, Mac OS 9.0.4 or later						
Channels	1 Stereo Record, 1 Stereo Playback, Full duplex						8 inputs, 8 outputs Full duplex
Signal Processing	AD/DA 16-bit		AD/DA 24-bit	AD/DA 16-bit	AD/DA 24-bit	AD/DA 24-bit	AD/DA 24-bit (int. 36 bit)
Sample rate	AD/DA 44.1 kHz	Digital In 32/44.1/48 kHz Digital Out 48 kHz	Digital Out 44.1 kHz	Digital In 32/44.1/48 kHz Digital Out 44.1 kHz	Up to 96 kHz	Up to 96 kHz	Up to 96 kHz
Audio Driver Compatibility	MMe, Sound Manager	MME, Sound Manager	ASIO2.0 (Win/Mac), WDM, MME, Sound Manager	MME, MME, Sound Manager	ASIO2.0 (Win/Mac), WDM, MME, Sound Manager	ASIO2.0 (Win/Mac), WDM, MME, Sound Manager	ASIO2.0 (Win/Mac), WDM, MME, Sound Manager
Inputs	Line (RCA pin) x2	S/P DIF Coaxial/Optical	Line/Guitar/Mic x2	Dynamic or Plug-In powered mic x1 Guitar (Hi-Z) x1 Line (RCA pin) x2 S/P DIF Optical	TRS Balanced/Unbalanced XLR/Phone combo with Phantom power x 2 Guitar (Hi-Z) Line (L/R) Standard Jack S/P DIF Coaxial/Optical	TRS Balanced/Unbalanced XLR/Phone combo with Phantom power x 2 Guitar (Hi-Z) x 1 Phono or Line (1/4″ phone) x2 Aux Line (RCA pin) x2 S/P DIF Coaxial/Optical	TRS Balanced/Unbalanced XLR/Phone combo with Phantom power x 2 Dynamic mic x2 TRS Balanced/Unbalanced 1/4″ phone x 6 Guitar 1/4″ (Hi-Z) x 2 S/P DIF Coaxial/Optical
Outputs	Line (RCA pin) x2	S/P DIF Coaxial/Optical	Line (RCA pin) x2	Line (RCA pin) x2 S/P DIF Optical Headphones (mini-stereo) x1	Line (1/4″ pin) x2 Line (RCA pin) x2 S/P DIF Coaxial/Optical Headphones (1/4″ phone) x1	Line (1/4″ pin) x2 Line (RCA pin) x2 S/P DIF Coaxial/Optical Headphones (1/4″ phone) x1	TRS Balanced/Unbalanced 1/4″ phone x 8 S/P DIF Coaxial/Optical Headphones (1/4″ phone) x1
MIDI Interface			IN x1/OUT x1			IN x1/OUT x1	IN x1/OUT x1
Bundled Software		WinDVD	Drivers (Win/Mac)	WinDVD, Cooledit ProLE	Drivers (Win/Mac)	Drivers (Win/Mac)	Drivers (Win/Mac)

'STUDIO CANVAS' SD-20

USB Bus-Powered MIDI Sound Module



A compact, portable, USB bus powered MIDI solution—perfect for your laptop—the SD-20 gives you the right number of sounds, and works with your computer no matter what its age: you can connect through USB or serial ports. And its compact size makes it easy to connect and create anywhere.

- 32-part, 64-voice polyphony
- Complete studio module including 32-part MIDI synthesizer and 64-voice polyphony for sequencing, karaoke, huge orchestrations and more
- GM2/GS/XGlite 660 tones, 23 drum sets
- Optical Digital Output (S/P DIF)
- The standard GM2/GS/XGlite compatible sound selections provide a variety of music genres such as classical, contemporary, pop and beyond. Play the realistic instruments tones from Grand Pianos, Rippin' Guitars, Jazzy Sax and Ethnic Drums!!
- Optical digital connection allows for exceptional MIDI sound quality output not found on any sound module in its price range.
- USB bus powered, adapter free
- USB connection supplies power as well as data. The convenience of having all the mobile elements of music production.
- Easy connection to computer (Serial / USB)
- Legacy compatible with serial computers or simply connect through USB for Mac

'STUDIO CANVAS' SD-80

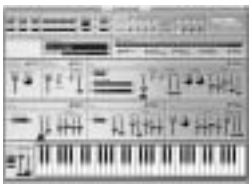
USB MIDI Sound Module and Synthesizer



A high performance MIDI sound generator with a powerful synthesizer engine and high quality waveforms in one slim body, the SD-80 is equally at home on your desktop or in a rack. Edirol started with a footprint that accommodates either application, added tons of connection options like USB, digital S/PDIF (coaxial and optical) interface and 1/4″ phone jacks, then made sure the sounds were just right–all 1050 of them. For flexibility, you can even run specific MIDI channels through assignable audio outputs for outboard effects processing to make the sound all your own.

- World leading technology of synthesizer sound engines and high quality waveforms (including the same wave ROMs as SD-90)
- Fast and stable MIDI data transmission realized by FPT (Fast Processing Technology of MIDI transmission)
- GM2/GS/XGlite compatible
- 32-part, 128-voice polyphony
- 1,050 tones, 30 drum sets, 3 MFX (90 types)
- Rack-mounted for studio use, or vertically mounted for desktop use
- Optical/Coaxial Digital output (S/P DIF)
- SD-80 Editor software included (Win/Mac)
- 2-stereo/4-mono multi-outputs
- 2-port external MIDI IN/OUT
- Letter sized design measures 11 1/16 x 10 3/16 x 1 3/16″ (WxDxH)
- Internal Sound Sets Sound Maps: 6 (Classical, Contemporary, Solo, Enhanced, Special 1, Special 2)
- Preset Sound Instruments: 128, Drum Sound Sets: 8
- System Effects: Reverb (6 types), Chorus (6 types)
- Insertion Effects: Multi-effects x 3 (90 types)
- Display: 20 characters, 2 lines (backlight LCD)

Bundled SD-80 Editor Software



The SD-80 ships with a powerful software editor that controls the SD-80's sound parameters by a graphical interface. The editor provides control over tone editing, patch changes, and effect parameters that can be preserved. (SD-80 has a built-in memory for preserving "user patches". Editor is Mac and Windows compatible.

'STUDIO CANVAS' SD-90

USB Audio Interface and MIDI Tone Module

The SD-90 effectively integrates in one unit a top-quality MIDI synthesizer, a complete USB digital audio interface, and powerful Digital Signal Processor — ideal for musicians using original tracks as their backing band, as well as for composers and remixers who can take advantage of its user-friendly environment. Through a simple USB connection to your computer, the SD-90 allows you to play MIDI instruments, record your own original audio from a microphone or a guitar, and process all your music content with a professional effects generator, just like you would in a studio. Works with most multi-track audio and MIDI recording software for Mac and Windows.



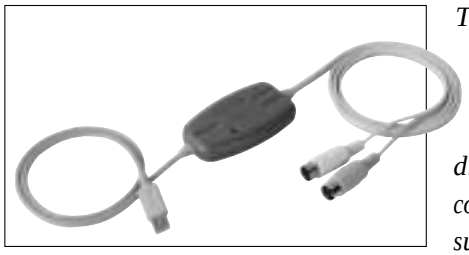
- 24-bit AD/DA converters
- 2 input - 2 output MIDI connections
- Optical/coaxial S/P DIF input and output jacks plus 1/4″ discrete microphone/guitar inputs
- Fast and stable MIDI data transmission realized by FPT (Fast Processing Technology of MIDI transmission)
- MIDI with 1,050 instruments, 36 drum sets (128 polyphony, 32 multi-timbrel)
- GM2/GS- compatible rich sound source selections
- Guitar, vocal, ambience and mastering effects for audio
- ASIO2.0 (WIN/MAC), MME (WIN), WDM (Win2000), Sound Manager (MAC)

	SD-20	SD-80	SD-90
Main Function	Sound Generator	Sound Generator	Sound Generator and USB Audio
Unique Feature	GM2 Format USB bus powered	Letter size design; Rack-mount type Character LCD(20x2)	GM2 Format
Connection to Computer	USB, MIDI, and Serial	USB and MIDI	USB and MIDI
System Compatibility	WindowsXP/Me/98/2000; MacOS/OS X	WindowsXP/Me/98/2000; MacOS/OS X	WindowsXP/Me/98/2000; MacOS
Instruments	660	1050	1050
Drum Sets	23	30	30
Voice x instrument	4-voice	4-voice	4-voice
Parts	32	32	32
Max polyphony	64	128	128
DSP	Reverb (6) Chorus (6)	Reverb (6); Chorus (6) 2 band EQ; Insert FX (90 types) x 3	Reverb (6); Chorus (6) 2 band EQ; Insert FX (90 types) x 3
Audio driver compatibility			ASIO2.0, WDM
MIDI Interface	IN x 1	IN x 2/OUT x 2	IN x 2/OUT x 2
Audio Inputs	Line (RCA pin L/R) x 1		24-bit audio I/F; Unbalanced 1/4″ (Mic/Guitar) x 1; Line (RCA pin L/R) x 1 S/P DIF Coaxial&Optical
Audio Outputs	Line (RCA) S/P DIF Optical Headphones (mini-stereo) x 1	1/4″ Line 1 (Master) 1/4″ Line 2 (Individual) S/P DIF Coaxial/Optical Headphones (1/4″ phone) x 1	Line 1 (Master) Line 2 (Surround) S/P DIF Coaxial/Optical Headphones (mini-stereo) x 1
Audio Effects			Guitar-Multi, Vocal-Multi, Mastering, and Spece-Multi; Effects (9 types)
Format	GM2/GS/XGlite	GM2/GS/XGlite	GM2/GS/XGlite
Bundled Software	Drivers (Win/Mac) SD-20 Editor (Win/Mac) Demo MIDI Songs	Drivers (Win/Mac) SD-80 Editor (Win/Mac) Demo MIDI Songs	Drivers (Win/Mac) SD-90 Editor (Win/Mac) Demo Audio&MIDI Songs Demo MIDI Songs
Power Supply	USB Bus powered or AC Adapter	AC Cord	AC Cord

EDIROL

UM-1/UM-1S/UM-2

1x1 and 2x2 USB MIDI Interfaces



The UM-1 is a 1x1 MIDI interface device that comes fitted with the USB cable on one end, and two (MIDI In & Out) leads on the other end. It is powered directly through the USB connection to the computer, so it doesn't require any power supply to function.

The UM-1S delivers the same specification MIDI interface as the UM-1, but allows the use of MIDI cables of your own choice and length. Conveniently offers an attached USB cable to the MIDI In and Out port connectors so you can choose the cables for connection to MIDI keyboard or sound module.



- Connect any MIDI instrument directly to your PC or Mac. This includes sound modules, MIDI keyboards, digital pianos, home keyboards and synthesizers
- Control up to 16 channels of MIDI
- True Plug & Play
- Includes CD-ROM with drivers for Windows 98/2000 and Mac OS
- Powered directly through USB connection
- Controls up to 32 channels of MIDI equipment
- Low latency with FPT (Fast Processing Technology of MIDI transmission)



This compact device lets you send and receive 32 channels of MIDI data. Just hook it up between your USB-equipped computer or laptop and 2 different MIDI devices—quickly and easily. Convenient USB connections provides data transmission as well as power -no AC power adapter needed. WIN/MAC compatible.

- Fast and stable MIDI data transmission realized by FPT (Fast Processing Technology of MIDI transmission)
- Controls up to 32 channels of MIDI equipment
- Adapter-free bus-type power supply through PC
- Signal indicators for 2-ins and 2-outs

UM-550 *5-in/5-out USB MIDI Interface*



If your creativity happens in a studio environment with multiple MIDI devices, the UM-550 provides a 5 x 5 near-zero-latency MIDI interface/ MIDI patching. Its 3/4˝ rack design combined with all the same utility functions as its "big brother" the UM-880, make it the "just right" solution for USB connectivity of medium complexity MIDI implementations.

- 5-in, 5-out USB MIDI Interface to computer (control up to 80 channels)
- Stand-alone hardware MIDI Patcher without connecting to computer
- Quick patching, easy operation, and low latency with HDMR (Hardware Direct MIDI Routing) technology
- Fast and stable MIDI data transmission realized by FPT (Fast Processing Technology of MIDI transmission)
- Comfortable patching operation with LED switches on the front panel
- Convenient MIDI utility function (Merge, MIDI Message filter, Preview, Cable check)
- Letter size compact body for desktop use, and variable rack-mount type with detachable adapters
- Multiple units (up to 4) can be connected to the same computer for control of up to 320 channels
- Windows and Mac compatible

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EQUIPMENT LEASING AVAILABLE

UM-880

8-in/8-out USB MIDI Interface with Hardware MIDI Patcher



The UM-880 is the most powerful, easy to use MIDI interface/patcher available. The vast majority of functions are available through either one or two button commands on the front panel. With eight sets of MIDI inputs and outputs, the UM-880 lets you simultaneously control up to 128 channels. It's easy to connect to your computer—simply connect a USB cable and you're done. And it is hot-swappable, so you can plug and unplug the UM-880 even while your computer is on. You can also use up to four UM-880 units simultaneously, expanding your system to 512 channels.



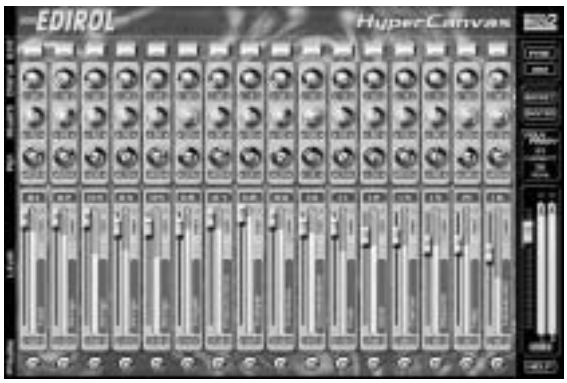
The UM-880 features hardware MIDI patcher functionality. MIDI can be routed directly, simply by pressing the panel buttons; and there are none of the complicated settings that are all too common on software patchers. The hardware MIDI patcher makes use of HDMR (Hardware Direct MIDI Routing) technology to ensure low-latency. If you have a lot of MIDI devices, you will find the UM-880 to be an integral part of your studio or stage rack, or both.

- The UM-880 can also act as a Stand-alone MIDI patcher with eight memory locations where you can store patch settings.
- Just a few major benefits of owning the UM-880 include:
- Fast and stable MIDI data transmission realized by FPT (Fast Processing Technology of MIDI transmission)
- Eight sets of input/out connections can be stored into memory.
- Easy USB connection to your computer.
- Convenient "Merge" function.
- Cable Check Utility.

	UM-880	UM-550	UM-2	UM-1	UM-1S
Main Function	MIDI Patcher/ MIDI Interface	MIDI Patcher/ MIDI Interface	MIDI Interface	MIDI Interface	MIDI Interface
Unique Feature	Hardware MIDI Patcher with HDMR FPT	Hardware MIDI Patcher with HDMR FPT	Powered via USB	USB & MIDI Cable included	USB Cable included
System Compatibility	Windows XP/Me/98/2000 MacOS, MacOS X	Windows XP/Me/98/2000 MacOS, MacOS X	Windows XP/Me/98/2000 MacOS, MacOS X	Windows XP/Me/98/2000 MacOS, MacOS X	Windows XP/Me/98/2000 MacOS, MacOS X
Inputs	8	5	2	1 (Male)	1 (Female)
Outputs	8	5	2	1 (Male)	1 (Female)
MTC (MIDI Time Code)	Yes	Yes	Yes	Yes	Yes
Bundled Software	Drivers, OMS, FreeMIDI	Drivers, OMS, FreeMIDI	Drivers, OMS, FreeMIDI	Drivers, OMS, FreeMIDI	Drivers, OMS, FreeMIDI
Body	1U rack	Latter size, rack mountable	Compact	Cable ends	Cable end
Power Supply	AC cord	AC Adapter	USB Bus powered	USB Bus powered	USB Bus powered

HQ HyperCanvas GM2

High Quality General MIDI Software Synthesizer

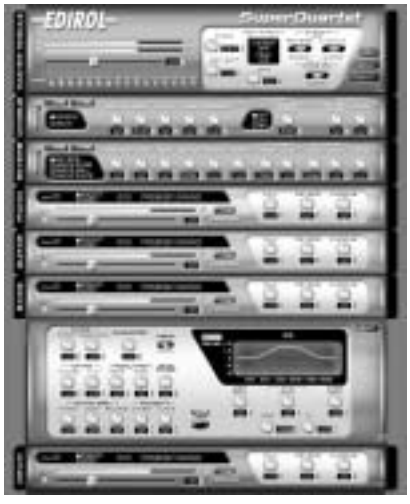


HyperCanvas GM2 is GM2 (General MIDI System Level 2) compatible, with 256 sounds and 9 drum sets. Play Jazz, Rock, Classics and more with your unique collection of studio quality instrument sounds. HyperCanvas GM2 contains multiple rich sound sources to help you create professional music of any kind. Features include a friendly user interface and a dedicated control panel to easily customize the sounds. Each instrument has levels of variation tones that are useful for a wide variety of music.
This high-quality software synthesizer is equipped with a sampling waveform and software synthesizer engine that generates amazing quality sound, ideal for new musical creation possibilities. Roland's synthesizer technology is used to make expressive waveform of various kinds possible. Compatible with two plug-in formats: DirectX Instruments and VST Instruments. This sound source can now be integrated for excellent results using the best sequencing software in the world!

- GM2 (General MIDI System Level 2) compatible, with 256 sounds, 9 drum sets, 16 part, and 128-voice polyphony
- Hyper Studio GM2 provides over 500 tone variations that can be saved for use in future projects. 2 types of plug-in software synthesizers: DXi and VST Instruments
- 24-bit/96 kHz performance, 32-bit floating point signal processing
- Dedicated control panel to easily customize the sounds
- 256 preset instrument sounds and 9 preset drum sets (User can add 512 variation sounds and 128 variation drum sets)
- Excellent for SMF data playback
- High Quality reverb chorus/delay & part EQs
- 128-note polyphony with 16 part multi-instrument playback
- Includes Cakewalk's Music Creator and exceptionally well-produced original compositions

SuperQuartet

Plug-in Software Synthesizer for Piano, Bass, Guitar, Drums



Create your own exceptional sound. This plug-in software synthesizer offers outstanding variations of Piano, Bass, Guitar and Drums. SuperQuartet is equipped with a sampling waveform and software synthesizer engine that generates amazing quality sound, ideal for new musical creation possibilities. Roland's synthesizer technology makes it possible to use expressive waveforms of various kinds.
Include 3 drums sets - expandable to 128, 67 instruments - expandable to 384, 16 part multi-instrument playback, 128 voice polyphony, chorus and reverb effects, 24-bit/96 kHz processing performance and a full array of customizable editing parameters.

- 24-bit/96 kHz performance, 32-bit floating point signal processing
- Dedicated control panels to easily customize the sounds. Each instrument has levels of variation tones that are useful for a wide variety of music:
  – Piano: Concert Grand, Upright, Electronic, Rock, etc.
  – Guitar: Nylon classic, Steel, Electronic Acoustic, Jazz, Clean, etc.
  – Bass: Wood, Finger, Picked, Fretless, etc.
  – Drums: Standard, Brush.
- DXi and VST compatible
- 60 preset instrument sounds and 3 preset drum sets (you can add 384 variation sounds and 128 variation drum sets)
- Unlimited sound expression tools to edit and save as user variation tone
- High Quality reverb chorus/delay and part EQs
- 128-note polyphony with 16 part multi-instrument playback
- Includes Cakewalk's Music Creator and exceptionally well-produced original compositions

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ORCHESTRAL

Lush Soundtracks and Beautiful Classical Arrangements



Orchestral instruments come alive with rich ambient piano and stereo sampled string sounds in this software synthesizer. The premium sounds in this synth are perfectly combined for lush soundtracks and beautiful classical arrangements. And its multiple output compatible with DXi 2.0 and VST 2.0 means you can enjoy your arrangements through several audio devices...simultaneously!

Orchestral provides premium acoustic sounds, focusing on strings, woodwinds, and brasswinds, as well as percussion and keyboards. These high quality samples include realistic articulations such as vibrato, spiccato, pizzicato, and tremolo. Using ample wave memory, Strings/Brass winds tones are selectable from solo, part section, and full section, making this ideal for the composition of contemporary music, sound tracks, and marching bands.

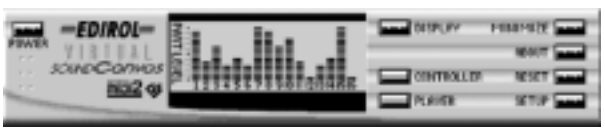
- DXi 2.0 and VST 2.0 support
- Acoustic sounds, focusing on Strings, Woodwinds, Brass, percussion, and keyboards
- Articulations such as vibrato, tremolo, pizzicato, and spiccato
- Up to 16-part and 128-voice polyphony
- Up to 24-bit/96kHz sampling resolution
- Multiple output
- Full assignable MIDI control
- Includes Cakewalk "MusicCreator2002"

VSC-MP1

Virtual Sound Canvas Multi Pack

This application allows you to take MIDI files and customize them to suite your taste. It includes standard effects such as reverb, chorus and delay. Use the playback list function to store and retrieve your compositions. Export your finished files into popular multi-media applications as .WAV or .AIFF file formats. This is MIDI made easy!





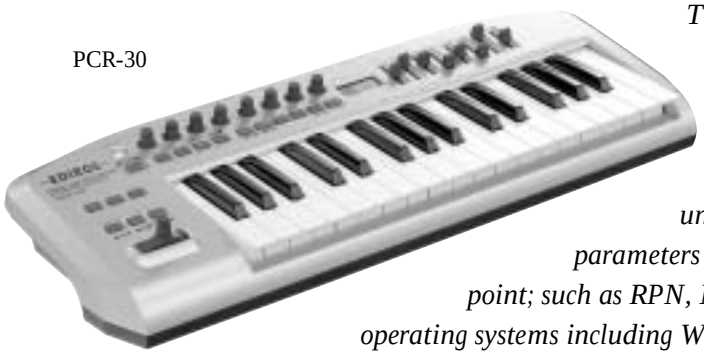


- Two plug-in versions (DXi and VST Instruments.)
- 16 parts, up to 128-voice polyphony
- 902 tones plus 26 drum sets
- Reverb, chorus and delay effects
- MIDI to WAV/AIFF file conversion
- Includes over 100 royalty-free MIDI file songs
- Playback list function permits easy storage and retrieval of your selections.
- VSC-MP1 is WIN/MAC compatible and ships with 100 royalty-free MIDI files.

PCR-30/PCR-50

32- and 49-Note MIDI Keyboard Controllers

PCR-30



The PCR-30 (32-note) and PCR-50 (49-note) are the advanced MIDI keyboard controllers available, period. Never before have there been better compact keyboards for use with popular sequencers and software synthesizers. The PCR-30 / PCR-50 take MIDI control to a new level with unprecedented versatility. They can control a number of parameters not accessible from other controller keyboards at this price point; such as RPN, NRPN, SysEx. They are fully compatible with the latest operating systems including WinXP and Mac OS X. They are the ideal keyboards for use with Arturia Storm, SONAR, Cubase, Reason, FruityLoops, and most other popular sequencers and software synthesizers. As such it comes equipped with templates for these and other programs.

- 32 or 49 full-sized, velocity sensitive keys
- 8 assignable rotary knobs (cut off, resonance, pan, tempo, etc.)
- 8 assignable faders (volume, EFX send, etc.)
- 9 assignable buttons (program change, track mute, start, stop, etc.)
- 1 sustain, 1 expression pedal connections
- MIDI in & MIDI out
- Pitch & Modulation controls
- Preset Memory & 15 User Memory locations
- USB Bus powered or AC adapter
- Template sheets for controls

UR-80 *USB Recording System and MIDI Controller*



The UR-80 is an all-in-one studio box for computer musicians, combining a fully assignable control surface, USB audio interface with 24-bit/96 kHz capability, MIDI and the GM2 Hyper Canvas software synthesizer. The integration of audio capture and highly customizable software control makes the UR-80 a "must-have" in the project studio. The control surface resembles a digital mixer, with transport controls, a time wheel and assignable switches for quick access to music software functions. It comes with a Control Map Editor and eight pre-loaded control maps supporting sequencers such as SONAR, Cubase and Logic, plus a variety of soft synths. (More control maps can be downloaded from Edirol). The UR-80 has a vast amount of MIDI implementation that allows you complete control of your favorite sequencing programs of software synthesizers as well as phantom power support for use with high-end studio microphones. Also offers full PC and Mac driver support, V-LINK support and a variety of input and output options.

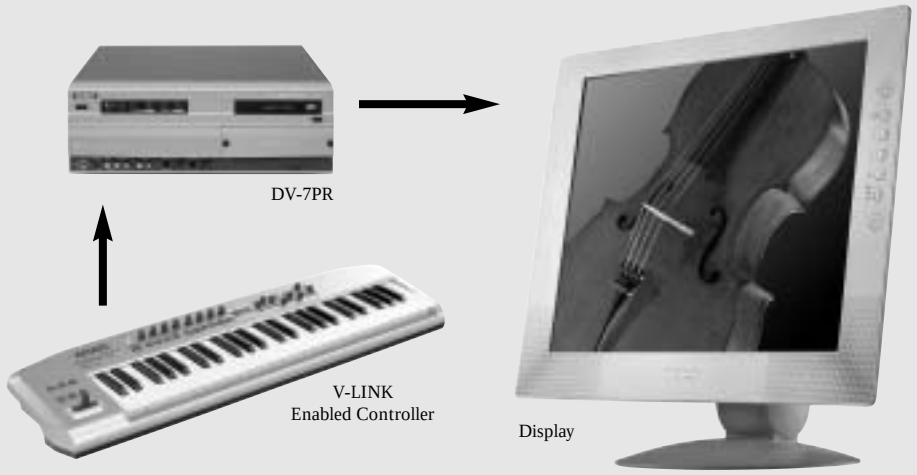
- 102 assignable parameters through 43 knobs, buttons and sliders on the control surface
- Up to 24-bit/96kHz recording via USB Audio Interface
- Support for any MIDI parameter, including CC,RPN, NRPN and SysEx
- Comprehensive surface layout for assignable switches for plug-ins and soft synths
- 2x XLR/TRS combo jack with mic preamp & phantom power
- Bundled with HQ-GM2, Hyper Canvas software synthesizer
- ASIO 2.0, WDM, Mac OS X compatible drivers, low-latency support.
- V-LINK for control of Edirol DV-7PR Real-time Video Presenter

DV-7PR

V-LINK

V-LINK is an exciting new technology from Edirol enabling the user to link musical performance to video. Using a V-LINK equipped device allows you to trigger clips, perform video effects and transitions all directly controlled by the expression being used in the musical performance. V-LINK uses a standard MIDI connection to control the Edirol DV-7PR. V-LINK is available on a number of Edirol and Roland devices.

Many creative artists have discovered the benefits of using audio and video together. V-LINK now makes it easier.





DV-7PR

The DV-7PR is a unique real-time visual performance system. It allows the instantaneous recall of hundreds of high-quality DV material. It is ideal for use it in concerts, churches, convention centers and seminars. The DV-7PR integrates a MIDI based control device, such as a MIDI Keyboard, to allow the user to combine DV clips, digital audio, and still images into a synchronized presentation. The DV-7PR can play and loop indefinitely drawing from up to 13 hours of unique content.

The DV-7PR is suitable for:

- Visual presentations in conjunction with concerts, live stage shows, and dance clubs (with MIDI Keyboard, trigger unit, etc.)
- Video kiosk systems
- Video performance presentation at show events such as trade shows, business presentations, weddings, etc.
- Includes DV-7 software, a powerful video editing application

On-Demand Playback of Digital Video

- Native DV audio/video signal output
- Dual image cross-dissolve



- Stores and plays back up to 13 hours of unique content even in a continuous loop

Multi-Interface Control

- Direct and easy operation with a mouse or ASCII keyboard
- Real-time effect/mixing with DV7C controller
- Use any MIDI controller (e.g. MIDI keyboard, touch-pad, or V-LINK enabled device)
- Touch screen via RS-232C control
- Compatibility with many third-party devices

Real-Time Operation

- MIDI control of digital video and audio content
- Allows fully automated loop playback
- Dynamic real-time dissolves
- Easy re-display of any sequenced clip with a single key stroke

Integration of Video Formats

- Allows editing of video and audio clips
- Can use removable hard disks (DV-HD60/ DV-HD120) as a media source for dynamic playback and archiving
- Will import DV video clips created on a PC through the CD-ROM drive and FireWire interface

GIGAPORT AG/DG

USB to 8-Channel Analog or ADAT Interfaces

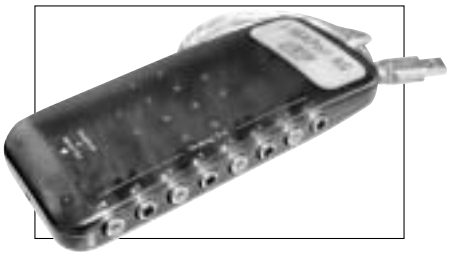
Not just another USB peripheral, GigaPort is the first of its kind, offering eight channels of audio output to either a single ADAT Lightpipe interface or eight individual analog outputs. The versatility of USB means that now you can have a viable high quality audio connection going out of your laptop or any USB compatible computer to the destination of your choice. And since GigaPort supports ASIO 2.0, GigaPort is compatible with both Mac and PC. GigaPort can also transform your laptop into a full-blown home theater system. It offers full compatibility with Dolby Digital 5.1 and DTS 5.1 surround formats and is ready to support DTS ES 6.1, and 7.1 formats as well.

Available in two versions:

The GigaPort DG with its ADAT digital connector is the world's first USB to ADAT optical interface. GigaPort DG is perfect for transferring files or mixing direct to ADAT or any piece of equipment that supports the ADAT optical interface. On the other hand, GigaPort AG is equipped with eight RCA analog outputs which make the possibilities endless. With eight high quality outputs, you could be mixing directly to DVD, transferring files, or just kicking back listening to your favorite DVDs straight out of your laptop in full 5.1 surround.

GIGAPort AG Specifications

- 8 x RCA analog outputs (8 x 24 bit D/A Converter),
- S/PDIF optical output
- ASIO 2.0 driver Mac & PC
- Headphone output
- WDM driver for Windows XP/2000/ME/98SE



GIGAPort DG Specifications

- 1 x ADAT digital output (8-channel out),
- ASIO 2.0 driver Mac & PC
- WDM driver for Windows XP/2000/ME/98SE





WAVETERMINAL U24

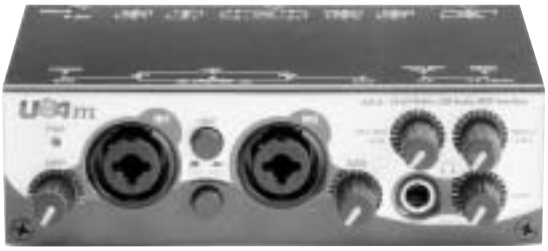
24-bit USB Audio Interface with Sample-Rate Converter

Perfect for laptop or desktop, the U24 is the most 'bang for your buck' USB peripheral going. Use it to record or perform wherever your gig takes you. Offering total USB flexibility and pure digital quality, coupled with true 24-bit capabilities, the U24 will easily qualify as the most useful device in your studio, wherever that may be. The U24 offers three stereo input options (coaxial, optical, analog) that you can select one at a time while the output signal is available on all output formats simultaneously. This allows the U24 to act as a signal converter between coaxial and optical audio. The analog I/Os of the U24 are supported with ultra-wide range 24-bit A-D/D-A converters, and unbalanced 1/4˝ phone jacks.

- 24-bit playback & record resolution
- S/PDIF coaxial & optical digital I/O
- 24-bit A-D and D-A converters : 100dB(A/D), 110dB(D/A) dynamic range
- Signal to Noise ratio (D/A) : 110dB
- -10dBv unBalanced 1/4˝ phone jack
- Optical <-> Coaxial Signal converter
- Simultaneous audio output through both analog and S/PDIF for easy-mix monitoring
- Internal digital mixer for input monitoring.
- 32, 44.1 and 48kHz sampling rates
- Real-time hardware sample rate converter
- Plug and play installation : no "IRQs" or "DMAs" to set, no addresses to configure
- Headphone amp
- Compatible with Windows 98SE/ME/2000, and XP along with Apple Sound Manager and MAC ASIO 2.0. Includes 6´ USB cable.

AQUA U24M

4-in/4-out USB Audio/MIDI Interface



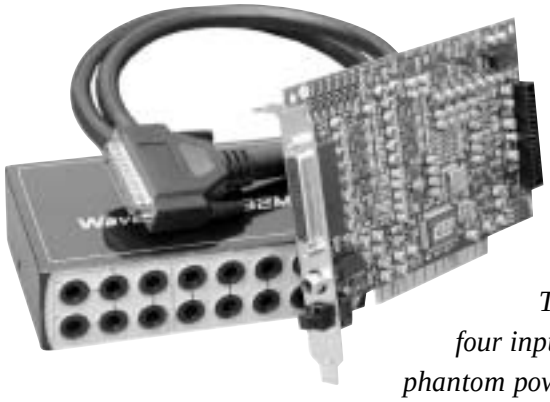
Built on the foundation of the U24, the U24M is a 24-bit/96kHz 4-in/4-out unit equipped with a 1x1 MIDI I/O port, S/PDIF digital input and output and two mic preamps. Housed in a 1/3 rack space size box, the U24M front panel has two combo connectors that can be switched between mic and line inputs. The mic preamps are equipped with switchable +48v phantom power. Also includes individual input gain controls and a headphone output with its own monitor level volume. The back panel layout consists of two 1/4˝ line inputs for channels 1 and 2, and two phono jacks for line inputs 3 and 4. Outputs are comprised of four phono jacks for outputs 1 through 4, and a separate left and right mix out that are serviced by two 1/4˝ connectors.



- Supports up to 24-bit/96 kHz
- Analog 4-in/4-out
- S/PDIF digital coaxial I/O
- 2 mic preamps with +48v phantom power
- 2 Inserts to use the external effect processor
- Headphone amp with volume control
- Internal monitoring mixer with In/Out level control
- No-latency monitoring with internal mix bus output
- 1x1 MIDI interface
- Windows OS/MAC OS compatible

WAVETERMINAL 192M

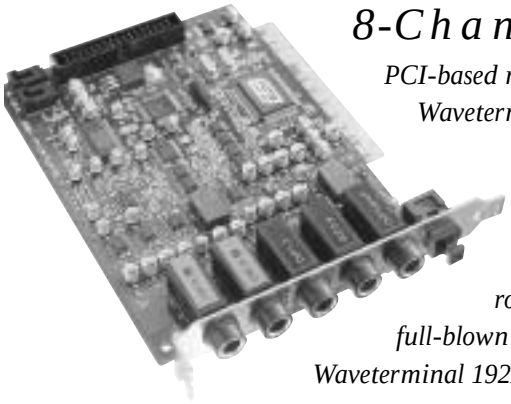
4-in/8-out PCI Audio Interface



The stylish Waveterminal 192M features a 4-in/8-out breakout box that connects to a half-length PCI card. The input A/D features high quality 24-bit/96kHz converters, while the D/A output has superior 24-bit/192kHz converters. Other features include support of multiple sample rates up to 192kHz, headphone amp output, 24-bit/192kHz S/PDIF stereo digital coaxial and 24-bit/96kHz optical and outputs, and full duplex (simultaneous record/playback) capability. The Waveterminal 192M's versatile breakout box makes it easy to access its four inputs and eight outputs. Two separate balanced TRS mic inputs with +12v DC phantom power can be found along side the 4 unbalanced +4dB line inputs.

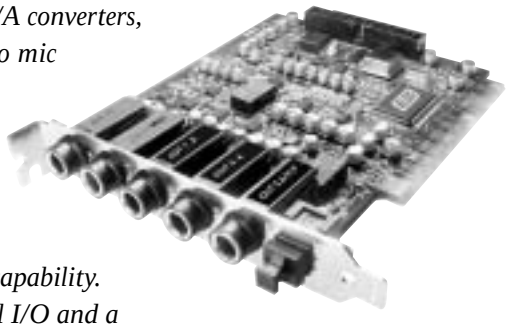
- High-quality 24-bit/96kHz A/D (100dB dynamic range) and 24-bit/192kHz D/A (104dB dynamic range
- Four 1/4˝ unbalanced +4dBu line inputs
- Eight 1/4˝ unbalanced +4dBu line outputs
- 24-bit/192kHz S/PDIF coaxial output & 24-bit/96kHz optical output
- 2 mic preamps with +12v phantom power
- Two 1/4˝ stereo headphone outputs with level control (output 1/2 or selectable mixed output)
- Supports multiple sampling rates : 16, 22, 24, 32, 44.1, 48, 88.2, 96, 176.4, 192kHz
- Full duplex capability—simultaneous record/playback
- 32-bit PCI slot: PCI Bus-Mastering support
- ESI's powerful E-WDM driver provides perfect compatibility with Windows XP, 2000, ME, 98SE and Mac OS 10.x, offering ultra low-latency performance (ASIO 2.0) with applications such as Nuendo, Cubase, Cakewalk, Gigastudio, Sonar and Logic
- Accepts the optional MI/ODI/O add-on card (see next page).

WAVETERMINAL 192X/192L



8-Channel PCI Audio Interfaces

PCI-based multichannel audio interfaces, the 2-input/8-output Waveterminal 192L and Waveterminal 192X are equipped with high-quality A-D and D-A converters, advanced audio capabilities and support ESI's E-WDM driver for compatibility with Windows XP, 2000, ME, and 98SE offering low-latency performance with applications such as Nuendo, Cubase, Gigastudio, Cakewalk, Sonar and Logic. They also offer complete compatibility with Dolby Digital 5.1 and DTS 5.1 surround formats. Used with a software DVD player, they transform your PC into a full-blown 5.1 home theater system. And with its true 24-bit/192kHz resolution the Waveterminal 192X also supports the DVD-Audio standard.



The Waveterminal 192L features 24-bit/96kHz A/D converters and 24-bit/192kHz D/A converters, all with 100 dB dynamic range. Other features include stereo analog line inputs, mono mic preamp with 12v DC phantom power, headphone amp output, 24-bit/192kHz S/PDIF stereo optical digital output and full duplex (simultaneous record/playback) capability.

For even higher quality, the Waveterminal 192X features 24-bit/192kHz A-D and D-A converters with up to 123dB dynamic range. Also standard is a 24-bit/192kHz S/PDIF stereo optical digital output and full duplex (simultaneous record/playback) capability. Optional MI/ODI/O card adds S/PDIF optical digital inputs, 192kHz S/PDIF coaxial I/O and a 1-in/1-out MIDI interface to the Waveterminal 192L and 192X cards.

FEATURES

- Two-channel (four-channels with the optional MI/ODI/O) recording and 8-channel playback at the same time
- 24-bit 96kHz S/PDIF optical digital out
- They support multiple sampling rates : 16, 22, 24, 32, 44.1, 48, 88.2, 96, 176.4, 192kHz
- Full duplex capability—simultaneous record/playback
- 32-bit PCI slot: PCI Bus-Mastering support
- Headphone amp
- Support the EWDM driver: Multiple MME, Multiple Direct sound, ASIO 2.0 and GigaStudio support
- Compatible with Windows XP/2000/ME/98SE & MAC OS 10.x

Waveterminal 192L Only

- 24-bit/96kHz A/D 100dB Dynamic range
- 24-bit/192kHz D/A; 104dB Dynamic range
- Two+4dBu unbalanced 1/4″ TRS inputs
- Three +4dBu unbalanced 1/4″ TRS stereo outputs
- 24-bit/96kHz optical stereo digital output
- MIC preamp with +12v phantom power

Waveterminal 192X Only

- 24-bit/192kHz A/D; 123dB Dynamic range
- 24-bit/192kHz D/A; 106dB Dynamic range
- Two +4dBu balanced 1/4″ TRS inputs
- Three +4dBu unbalanced 1/4″ TRS stereo outputs
- Supports multiple cards in one computer system (192X only)

MI/ODI/O



The optional MI/ODI/O card lets you get more from your Waveterminal 192 series (192M, 192L, 192X) by adding 24-bit/96kHz S/PDIF optical input and coaxial I/O, and a 1x1 MIDI I/O. Easy to install, MI/ODI/O requires no additional drivers , power supply or new IRQ set up.

- 1X coaxial digital I/O (192kHz)
- 1X optical digital input
- 16 Channels MIDI I/O
- Turns any Waveterminal 192 card into a 4-in 8-out/16 MIDI channel integrated audio interface

WaMi Rack 192X/192L

10-Channel Integrated Audio Interfaces



The flagship WaMi Rack 192L and WaMi Rack 192X are housed in a single-rack space box designed to form the central core of a complete 192kHz/24-bit recording system. They feature four studio-quality discrete XLR mic preamps (line/mic switchable) inputs with +48v phantom power, headphone output with independent level control, built-in MIDI interface and coaxial S/PDIF connectors. Otherwise identical, the 192L features a 24-bit/96kHz A/D converter, the 192X features a 24-bit/192kHz A/D converter.

- 4 studio-quality, ultra-clean discrete XLR mic preamps with +48v phantom power
- The WaMi Rack 192L has a high-quality 24-bit/96kHz A/D converter (up to 100dB dynamic range), the 192X's 24-bit/192kHz A/D converter handles up to 123dB.
- Both have state-of-the-art 24-bit/192kHz D/A converters with 104dB dynamic range
- 1x1 MIDI Interface
- Four analog inputs and 8 outputs on +4dBu unbalanced 1/4″ TRS phone jacks
- They support multiple sampling rates: 16, 22, 24, 32, 44.1, 48, 88.2, 96, 176.4, 192kHz
- 24-bit/192kHz supported S/PDIF coaxial digital I/O
- They offer full compatibility with Dolby Digital and DTS 5.1 as well as DTS ES 6.1, and 7.1 surround formats. WaMi Rack 192X fully supports DVD Audio as well.
- Full duplex capability—simultaneous record/playback
- Headphone amp
- ESI's powerful E-WDM driver provides perfect compatibility with Windows XP, 2000, ME, 98SE and Mac OS 10.x, offering ultra low-latency performance (ASIO 2.0) with applications such as Nuendo, Cubase, Cakewalk, Gigastudio, Sonar and Logic.

MAXIO XD

The Ultimate Solution for High Quality Audio Production



The MAXIO XD is a complete 24-bit/192kHz recording system designed to handle 32 simultaneous channels of analog inputs and outputs. The basic system consists of the EX-8000, a 2-rack space unit with 8-in/8-out balanced XLR connectors and a choice of a PCI interface for desktop systems, or a cardbus interface for laptop recording. Each interface is equipped with four firewire-like EDI (ESI Digital Interface) connections, each capable of supporting one EX-8000, giving you 32-channel capability on one card. In addition, the PCI version can support up to 4 cards on the same computer making a total of 128 channels of recording possible. The front panel of the EX-8000 features 8 balanced mic preamps., while combo connectors offer the choice of using XLR or TRS inputs. The EX-8000 also includes +48v phantom power and 10-step LED peak metering lights for each channel, as well as AES/EBU and S/PDIF digital I/O, and a headphone monitor.

- MAXIO PCI card and the MAXIO Cardbus interface include four EDI protocol ports and a buslink breakout that hosts a 192kHz S/PDIF coaxial I/O, Word Clock I/O BNC connector, and a 1x1 MIDI interface.
- Hardware input monitoring
- Comes standard with DirectWire, ESI's powerful and unique virtual patchbay for digitally connecting between popular audio software programs.
- 8-channel digital I/O: AES/EBU & coaxial support up to 24-bit/192kHz
- Eight XLR/1/4″ TRS analog combo inputs and eight 1/4″ TRS/XLR analog outputs (+4dBu/-10dBV); plus eight 1/4″ inserts
- Headphone amplifier
- Internal 64-channel digital mixer & assign channel to port

QUATA-FIRE

24-bit 192kHz FireWire 4x4 Audio/2x2 MIDI Interface.



The ultimate mobile solution for audio professionals, Quata-Fire is a versatile 4-in/4-out 1/3 rack space size unit with high-quality 24-bit/192kHz audio performance. Features include two front panel mounted combo connectors that can be switched between mic and line inputs. The mic preamps are equipped with switchable +48v phantom power, individual input gain controls and a headphone output with its own monitor level volume. Also features 2x2 MIDI interface, and a coaxial S/PDIF I/O.

OCTA-FIRE

24-bit 192kHz FireWire 8x8 Audio/4x4 MIDI Interface



Based on ESI's popular WaMi Rack Series, OctaFire is an 8-in/8-out device features four combo connector mic preamps with +48V phantom power along with a headphone monitor on the front panel. Other great features include 4x4 MIDI interface offering up to 64 channels of MIDI, coaxial S/PDIF I/O, and Wordclock I/O.

The QuataFire, OctaFire and HexaFire are all Windows XP and Mac OS X compatible and all feature:

- ◆ Powered by IEEE1394 Bus or external power supply
- ◆ No-latency monitoring with internal mix bus output
- ◆ Cascade (link) up to 4 units via standard IEEE1394 hub

HEXA-FIRE

24-bit 192kHz 16x16 High Definition FireWire Audio/MIDI/SMPTE Interface



The flagship of ESI's Fire Series is the HexaFire, a 16-In/16-Out box that features 8 balanced mic preamps via combo connectors offering the choice of using XLR or TRS inputs. There is +48v phantom power on each channel as well as 10-step LED peak metering lights for channel inputs and outputs on the front panel. Additional features included S/PDIF digital I/O, and a headphone monitor, along with S/PDIF coaxial I/O, Word Clock I/O, and a 1x1 MIDI interface. Sixteen channels of AES/EBU in and out are accessible via four D-25 connectors. Two IEEE-1394 ports can connect the HexaFire to a computer or daisy chain the unit with up to three other HexaFires.

MIDI INTERFACES

ESI introduces the power MIDI trio. Three different USB MIDI interfaces to choose from that best fit your budget and production needs. The RoMI/O is a 1-in 2-out, 32 channel interface that fits in the palm of your hand. It offers the flexibility to take your music anywhere and everywhere you need to go. The M4U is a 4-in/4-out device housed in a half-rack sized box that can handle up to 64 MIDI channels. The M8U is built into a full single rack space unit. It has 8 inputs and 8 outputs and is designed to accommodate 128 MIDI channels. The M8U also acts as a stand-alone MIDI patchbay when not connected to a computer. All three units are designed to for use with Windows XP and Mac OS X. Just plug and play.



◀ *RoMI/O*

MIDITerminal M4U ▶





MIDITerminal M8U

THEY ALL FEATURE:

- High-speed connection to USB-equipped PC or Mac
- MIDI activity indicator for each port

RoMI/O

1-In 2-Out, 32 channel USB to MIDI Interface for Mac and PC

- Complete cable type body from PC end to MIDI ends
- USB powered - requires no external power supply
- Extremely compact 83 x 52 x 28mm (WDH)

MIDITERMINAL M4U

4-In, 4-Out, 64 channel USB to MIDI Interface for Mac and PC

- USB powered - requires no external power supply
- Perfect MIDI timing
- Compact 1/3 rack size

MIDITERMINAL M8U

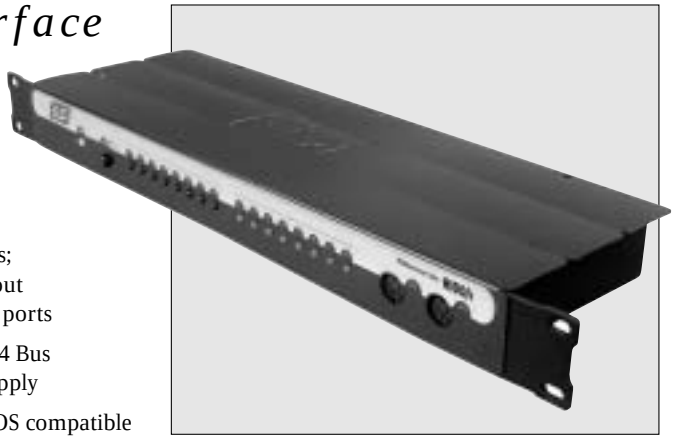
8-In, 8-Out, 128 channel USB to MIDI Interface for Mac and PC

- Perfect MIDI timing
- Various MIDI THRU function
- Unit can be used as a stand-alone MIDI patch bay

8180FS - MIDITERMINAL 1394

8x8 IEEE1394 MIDI/SMPTE Interface

The 8180FS is an 8x8 FireWire to MIDI interface offering up to 128 channels of MIDI along with full SMPTE and Wordclock synchronization. The front panel is equipped with MIDI activity lights. The unit can be used together with up to three other 8180's.



- 8in/8out (128 channels) MIDI interface
- H/W based MIDI processing with large size of buffer memory
- SMPTE (LTC) I/O
- Video Sync In, Word Sync Out (BNC connector)
- Stand-alone MIDI Thru/Patch bay BOX feature
- Cascade up to 4 units; provides 32-inn/32-out (512channels) MIDI ports
- Powered by IEEE1394 Bus or external power supply
- Windows OS/MAC OS compatible

EZBUS

The Ultimate Three-in-One Audio Tool

A true triple-threat, Event took a computer audio recording interface filled with an extremely versatile collection of analog and digital inputs and outputs, a deeply programmable and remarkably flexible software control surface and a 24-bit/96kHz stand-alone digital mixer, and built them all into one incredibly convenient, compact, powerful package.





Analog Input/Output

When it comes to getting music into and out of your computer, there's no more flexible, powerful, or versatile recording interface than the EZbus. Dual low-noise 24-bit/96kHz mic preamps—complete with +48V phantom power—provide a super clean path for vocals, acoustic guitars, and other miked signals. High impedance, high gain instrument inputs (also 24-bit/96kHz) make tracking superb sounding electric basses and guitars an everyday affair. And Event's unique triple-summed 24-bit/96kHz line inputs can each accept up to three independent signals, so it's EZ to plug in multiple synths, or a rack of outboard effects—or both. You won't find the EZbus skimping on analog outputs either. Six, discrete, balanced 24-bit/96kHz outputs plus a TRS stereo headphone output (complete with its own amplifier), make it a snap to interface with your outboard gear, set up control room and cue mixes, monitor a surround mix—or accommodate just about any other audio application you can imagine.

Processing

Each of the EZbus Primary Audio Channels features three bands of super responsive, musical EQ—high- and low- shelving plus a fully parametric mid band. You also get real-time dynamics processing on every channel. Tame your vintage gear with the Noise Gate. Or tap into the fully programmable look-ahead Compressor and never again worry about clipping that perfect vocal take.

Hardware Mixer

If you're making music with computers, the EZbus can make the process simpler and more fun. Take mixing, for example. Without an EZbus, you adjust each of your magnificently crafted tracks one at a time with a mouse. With an EZbus, you reach for an honest-to-goodness hardware mixer, and move eight faders at a time. (Now that's mixing!)

LCD Display

This custom display not only shows you all the normal stuff—levels, mute and solo status, and the like—but cool stuff, like the amount of gain reduction your compressor settings generate, and handy alert messages, such as input overload and clock signal dropout warnings.

Transport Controls

The EZbus sports programmable transport controls that let you easily start, stop, record, rewind, and fast forward through your projects. Use them to set multiple locate points on the fly, then recall them at the push of a button. Navigating through your projects has never been this easy.

Full Control

Virtually every parameter of a DAW, plug-in, virtual instrument, or MIDI-capable device can be controlled from the front panel. Use the factory control surface profiles or create and customize your own. With up to 32 control setups in on-board memory, the EZbus immediately takes center stage as command central of your studio or live performance rig.

Power Routing

Event's exclusive any-input-to-any-output audio routing matrix provides unmatched flexibility for virtually any audio application. Tap into the signal at the input, pre- or post-trim control, pre- or post-EQ and dynamics, pre- or post-fader—and route it to the destination of your choice. Even route digital signals directly from input to output for bit-accurate digital copying. Now add four Sends, each independently configurable pre- or post-fader, two mono Returns and one stereo Return, and both Main and Alt Mix buses, and there's almost no end to your routing options.

Easily generate independent control room, stage monitor, and front-of-house console mixes. Create a multi-bus setup for recording—and monitor it with zero latency. Save your custom I/O configurations for instant recall from the front panel, via footswitch, or using MIDI program change commands. Up to 32 unique configurations can be on-board at any time, and you can archive and organize thousands more using the included EZbus Mix Librarian software.

Control Surface

A single button-push is all it takes to instantly transform the EZbus from an Audio Recording Interface into a Control Surface. Then operate your favorite MIDI and audio software using the EZbus's hardware controls. Customize the controls with your own command set, or use the convenient factory profiles. Presets for major software programs, including VST, Nuendo, Cakewalk, Sonar, and many more are included.

MIDI

With three independent MIDI I/O ports, the EZbus puts you in complete command of your MIDI system. Connect your controller keyboard to either of the MIDI inputs on the EZbus, and trigger your virtual instruments via USB. Now route the audio from the virtual synths back into a couple of EZbus channels via USB, and mix the signals with your regular hardware synths (which are plugged into other EZbus channels). Your entire keyboard rig—real and virtual—is now a fully integrated system!

Digital I/O

What could be easier than hot-plugging the EZbus into your computer's or laptop's USB port and getting your musical ideas recorded fast? Connect to lightpipe-equipped gear via the ADAT optical ports. Interface with digital effects units, DAT recorders, or any S/PDIF device via the dual coaxial S/PDIF outputs and the coaxial and optical S/PDIF inputs. You can even mix and match sample-rates via the coaxial S/PDIF input, so that killer track you recorded at 44.1kHz can easily be imported into your 48kHz session. And the low jitter word clock output will keep all your digital devices locked with

EZ8 24-bit/96kHz Optical Audio Interface

Expand your EZbus into a full-blown multitrack recording system with the EZ8 Optical Audio Interface. This low-cost/high-performance PCI card provides eight discrete channels of full-duplex 24-bit/48kHz digital I/O via ADAT lightpipe.

JL COOPER
CS-32 MINIDESK

Amazing 32-Channel Miniature Control Console that fits in the Palm of Your Hand!



The CS-32 MiniDesk is a miniature control surface for fast access to the most commonly used functions in most audio programs. It works with ProTools, X-Track, Nuendo, Cubase SX, VST, Digital Performer, Soundscape REd, Cakewalk Pro, Sonic Foundry, Ableton Live, Merging Technologies Pyramix, Logic Audio and most audio software programs! The CS-32's radically small size does not reduce it's intense capabilities as an extremely powerful controller with 32 Dedicated Faders/Channel Strips for real-time control of volume levels, Mute, Solo, Track Arming and more—record, edit and mix your projects to perfection.

The "Mini" has Tactile Transport Buttons, Cursor Keys and Weighted Jog/Scrub Wheel, just like the big guys. Six Rotary Knobs give you hands-on, automation control of panning and plug-ins. A full bank of quick function keys are ready for frequently used editing features.

MiniDesk takes up less space than a sheet of paper. Use it almost anywhere, even on the most crowded desk. Perfect for portable recording applications with a laptop. With a controller this small, you might want to get an extra one to keep in your car.

- Adds tactile controls to most digital audio editing systems, MIDI sequencers and animation software
- 32 dedicated channels with 20mm faders and channel buttons eliminate fader bank switching confusion
- Puts over 150 dedicated controls at your fingertips
- Perfect for project and home studios
- Weighted Jog/Scrub Wheel (optically encoded)
- 2-digit alphanumeric LED display
- Dual LED activity indicators
- Sends various messages —CS-10 Emulation, General Controllers
- Low profile and compact design with heavy duty construction
- MIDI/MMC, USB and RS-232 models available
- USB version includes software (Mac and Win) and Keysets

LYNXONE

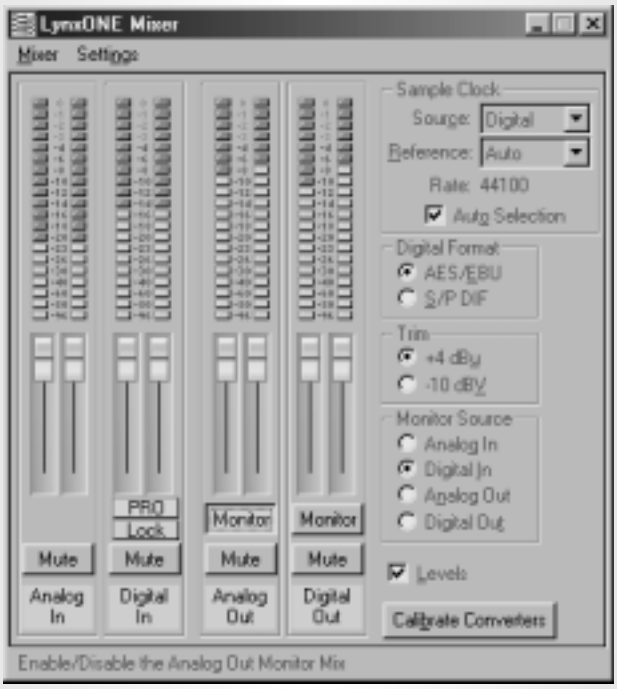


24-bit Digital Audio/MIDI PCI Card

Representing a great value in integration and professional capabilities, LynxONE is the perfect "front-end" for any studio quality audio or MIDI workstation. LynxONE combines studio-quality 2-channel analog I/O (sampling rate up to 50kHz), bit-perfect AES/EBU or S/PDIF digital I/O supporting up to 96kHz, and 32 channels of low latency MIDI to create a new standard in interface hardware for audio and MIDI workstations. And LynxONE's compatibility with all popular Windows-based audio editing and MIDI sequencing software allows users to choose their own working environments. Applications include critical audio recording and editing, CD mastering, restoration, audio for video, music composition, and MIDI sequencing.

FEATURES

- Two-channel 24-bit A-D/D-A converters and a simple analog signal chain ensure exceptional sound quality. (Software selectable +4dBu or -10dBV levels).
- If you are working in an all digital environment, the LynxONE delivers bit-perfect digital audio at up to 96kHz, via an AES/EBU or S/PDIF I/O (software-selectable in the LynxONE mixer application).
- Low-jitter PPL sample clock allows the LynxONE to be used as a slave or master timing source, enabling up to four cards to be linked together for multi-channel operation.
- The flexible mixer application also provides keyboard and mouse control over the sample clock generator, monitor source, analog trim and levels of all four channels, with peak level metering.
- PCB layout and circuit design techniques reject computer related noises. Because of this, LynxONE offers 103dB of "real-world" dynamic range when installed in a typical computer.
- The ability to easily monitor the recording inputs is a must during overdubbing sessions. LynxONE lets you select either the analog or digital inputs as the monitor source. This signal can be mixed with playback audio on either the analog or digital outputs or both simultaneously. This capability also allows "stand-alone" A/D or D/A conversion completely on board LynxONE. Because mixing is performed in hardware, recording input-to-output delay is extremely low.
- Two independent MIDI ports provide 32 channels of low-latency MIDI I/O, with real-time transfer of Note On/Note Off and MTC messages over the PCI bus for accurate timing and synchronization in music composition and MIDI sequencing applications.
- Compatible with Windows-based audio editing and MIDI sequencing software, LynxONE delivers exceptionally high levels of performance in critical audio recording and editing, CD mastering, broadcast, audio restoration, audio for video and MIDI sequencing applications.
- Comes complete with balanced analog, digital and MIDI/clock cables for easy integration into your existing setup.



Complete software control of LynxONE's features is provided with the LynxONE Mixer. As shown to the right, this application permits keyboard and mouse click control of volume, sample clock generator, digital I/O format, analog trim, and input monitoring. Peak level meters are also provided.

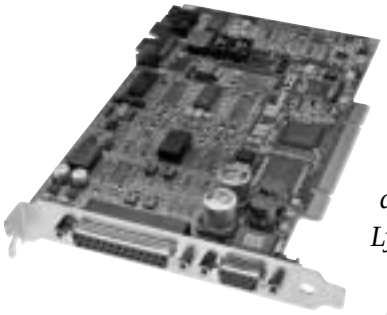
LYNXTWO

192kHz Multichannel Audio Reference Interface



Incorporating state-of-the-art converter technology, the LynxTWO is capable of precision never before achieved in a PCI audio card. With performance that exceeds stand-alone converters, LynxTWO sets new standards for signal purity, versatile synchronization, powerful hardware mixing, and extensibility to support a variety of multichannel formats including ADAT and TDIF. The LynxTWO incorporates the latest generation 24-bit/192kHz converter technology to create a PCI audio which rivals the performance of many high-end studio converters. A choice of three multi-channel analog configurations makes it ideal for DVD-authoring, multi-channel recording, mastering, broadcasting, synchronization and video applications, with non-audio digital I/O support provided for Dolby Digital.

- 24-bit analog inputs and outputs with three options: 4-in/4-out (LynxTWO-A), 2-in/6-out (LynxTWO-B) or 6-in/2-out (LynxTWO-C), each with line levels of either +4dBu or -10dBV, selectable in channel pairs through the mixer software.
- The 32-channel mixer application also controls the card's routing, monitoring and sample clock settings.
- Includes balanced audio and sync cable sets
- Has one digital I/O supporting both AES/EBU and S/PDIF formats at 16, 20 and 24-bit depths, and features a sample rate converter for up to 3.1 rate conversion from the digital input signals.
- The internal port allows data routing and synchronization between multiple LynxTWO cards, while the external port allows the connection of Lynx, ADAT, TDIF and other multi-channel interfaces for great system flexibility.
- Two LStream expansion ports support multi-channel interface modules, with each port offering 8-input/output channels of 24-bit/96kHz operation.
- Powerful synchronization and time code facilities include an LTC reader/generator which accommodates all standard frame rates, and a low-jitter sample clock generator, which is capable of locking to NTSC or PAL video signals, and to standard reference and word clock signals.



Lynx L22

192kHz PCI Digital Audio Interface

The Lynx L22 is a professional audio interface card designed for the most demanding audio recording, post-production, broadcast, and measurement applications. Based on LynxTWO technology, the L22 can capture or generate signals beyond the audio bandwidth up to 100 kHz with extreme accuracy. Using proprietary circuit topologies and custom components in the analog stages coupled with the latest converter technology, the L22 achieves a 117 dB dynamic range. Most importantly, advanced noise-rejection design techniques maintain this performance in real world use from computer to computer.

- Large on-board buffers and extremely efficient zero-wait state DMA engine ensure glitch-free recording and playback. This provides tolerance to system latencies and significantly reduces load on the host CPU.
- Extremely flexible onboard digital mixer allows the L22 to adapt to any studio or workstation configuration.
- Each of the mixer's 16 outputs is capable of mixing signals from 16 physical inputs or 16 playback tracks with 32-bit precision.
- A choice of dither algorithms is available on each input to mask low-level artifacts while reducing word widths to 8, 16, or 20 bits.
- The L22 provides I/O expansion via its two LStream ports. Combined, these ports allow 16 channels of additional I/O from external ADAT, AES/EBU, or TDIF devices. The ports can also be used to route data and synchronize to other L22's or LynxTWO's to create high channel-count systems.

LS-ADAT

Multi-channel ADAT Expansion Card



The LS-ADAT is a full-function ADAT interface that provides two ADAT lightpipe inputs and outputs and an ADAT sync input for the LynxTWO and Lynx L22 audio cards. Operating at a sample rate of 48 kHz, the LS-ADAT I/O channel capacity is 16. Higher sample rates are supported using S/MUX technology: 8 channels at 96 kHz and 4 channels at 192kHz.

- The ADAT sync input provides two methods of sample-accurate synchronization.
  - In systems using ASIO compatible audio applications, the LS-ADAT utilizes Steinberg's ASIO Positioning Protocol for synchronization to ADAT time code.
  - Using the LS-ADAT's cue point capability initiates recording or playback with sample period resolution.
- Sample clock source can be derived from a word clock or composite video received by the LynxTWO. In addition, either lightpipe input or the Sync In port can also be selected as the clock source.
- The LS-ADAT can also transmits time code via MTC for general compatibility with many applications.
- Installs in an empty bracket internally in your computer, no PCI slot required.

Now you can easily increase the I/O capacity of the LynxTWO and L22 digital audio cards. The LS-AES and LS-ADAT provide high-performance multi-channel AES/EBU and ADAT connectivity for any Windows or Macintosh based audio or video workstation at sample rates up to 192kHz.

Channels from the expansion cards are connected directly to the LynxTWO/L22 digital mixer. This allows almost unlimited routing capabilities with expansion card I/O's, as well as the analog and digital I/O's of the host LynxTWO or L22.

Using the LynxTWO/L22 Mixer application, "stand-alone" functions can be configured that place no demand on the host computer. In a LynxTWO system these functions include but are not limited to:

- Four-channel, 192 kHz analog-to-AES or analog-to-ADAT converter
- Four-channel, 192 kHz AES-to-analog or ADAT-to-analog converter
- Eight-channel AES digital patch-bay
- 16-channel ADAT digital patch bay
- AES to ADAT or ADAT to AES format converter
- Composite video or word clock to AES or ADAT synchronizer

LS-AES *LStream Expansion Card*



The LS-AES is an 8-channel AES/EBU or S/PDIF LStream interface module that is designed for use with the LynxTWO and Lynx L22. The LS-AES provides four inputs and outputs that support professional AES/EBU and consumer S/PDIF digital I/O formats. Connections are transformer coupled with pro-quality sample rate conversion (SRC) available on all inputs. The LS-AES is also compatible with Dolby Digital and DTS encoded formats.

- Running at a sample rates up to 96 kHz, the LS-AES provides eight I/O channels in single-wire mode and four channels in dual-wire mode. At 192 kHz, four channels of I/O are available in dual-wire mode and two channels in quad-wire mode.
- For additional channel capacity, two LS-AES cards can be used simultaneously with one LynxTWO or L22.
- The LS-AES can handle both asynchronous and synchronous studio configurations. SRC can be enabled independently on each digital input to accommodate asynchronous signals from multiple digital devices Alternatively, all inputs can be run synchronously and any input can be selected as the system clock source. The card can also be synchronized to a word clock or composite video received by the host LynxTWO.
- For S/PDIF coaxial connections, the optional Lynx XLR to RCA S/PDIF adapters CBL-XFDRM18 and CBL-XMDRM18 are required. The LS-AES installs in an empty bracket internally in your computer, no PCI slot required.
- Includes: LS-AES Internal Card LS-AES LStream Ribbon Cable 6' AES/EBU Breakout Cable

LYNX COMPARISON CHART

	LynxONE	Lynx L22	LynxTWO-A	LynxTWO-B	LynxTWO-C
Analog I/O					
Number of Input Channels	2	2	4	2	6
Number of Output Channels	2	2	4	6	2
Maximum Sample Rate	50kHz	200kHz	200kHz	200kHz	200kHz
Dynamic Range (Analog In or Analog Out)	>106 dB	117 dB	117 dB	117 dB	117 dB
Digital I/O					
Number of Channels	1 Stereo	1 Stereo	1 Stereo	1 Stereo	1 Stereo
Maximum Sample Rate	96kHz	96kHz	96kHz	96kHz	96kHz
AES/EBU and S/PDIF Support	Yes	Yes	Yes	Yes	Yes
Bit Perfect Transfers for Dolby & DTS	Yes	Yes	Yes	Yes	Yes
3:1 Sample Rate Converter		Yes	Yes	Yes	Yes
Subcode Support		Yes	Yes	Yes	Yes
LStream Ports					
Number of LStream I/O Ports		2	2	2	2
MIDI					
Number of MIDI Ports	2				
Multi-Client MIDI Record	Yes				
Time Code					
LTC Reader			Yes	Yes	Yes
LTC Generator			Yes	Yes	Yes
LTC to MIDI Time Code (MTC) Conversion			Yes	Yes	Yes
Synchronization					
Word Clock	Yes	Yes	Yes	Yes	Yes
256 Word Clock	Yes	Yes	Yes	Yes	Yes
13.5MHz Pixel Clock	Yes	Yes	Yes	Yes	Yes
27MHz 2x Pixel Clock	Yes	Yes	Yes	Yes	Yes
SMPTE Longitudinal Time Code (LTC)			Yes	Yes	Yes
NTSC/PAL Composite Video Genlock			Yes	Yes	Yes
Miscellaneous					
Simultaneous Record and Play (SRP)	Yes	Yes	Yes	Yes	Yes
Simultaneous Stereo Record Devices	2	8	8	8	8
Simultaneous Stereo Play Devices	2	8	8	8	8
4 Channel / 32-bit Digital Mixer	Yes				
32 Channel / 32-bit Digital Mixer		Yes	Yes	Yes	Yes
General Purpose Frequency Counters		8	8	8	8
16 Channel Bus Master Scatter-Gather Engine		Yes	Yes	Yes	Yes
Driver Support					
Windows 95/98/ME					
Multimedia Extensions (MME)	Yes	Yes	Yes	Yes	Yes
DirectSound	Yes	Yes	Yes	Yes	Yes
ASIO 2.0	Yes	Yes	Yes	Yes	Yes
Windows NT/2000/XP					
Multimedia Extensions (MME)	Yes	Yes	Yes	Yes	Yes
ASIO 2.0	Yes	Yes	Yes	Yes	Yes
Macintosh					
ASIO 2.0	Yes	Yes - Beta	Yes - Beta	Yes - Beta	Yes - Beta

M-AUDIO

DUO USB



2-in/2-out USB Audio Interface with Mic Preamps and Digital Output

Record from mics or any line level sources. Optimize levels with gain controls and pads, as well as signal and clip LEDs. Use the Direct Monitor button for zero latency monitoring (direct from the sources without going through the computer and software). Adjust the global output by toggling between +4db and -10db. Headphones can be connected for convenient monitoring in the field. When disconnected from the USB port, the Duo functions as a 24-bit standalone A/D converter. Select sample rates via DIP switches (44.1, 48, 88.2 and 96kHz), connect the S/PDIF output to your digital device, and you re ready to record digitally anywhere, anytime.

- 2x2 analog I/O
- 2 mic preamps
- 2 XLR-balanced mic inputs
- 2 balanced/unbalanced 1/4″ TRS line inputs
- Available gain of 66dB
- 20db pad on each mic input
- 48V phantom power
- S/PDIF digital output (16- and 24-bit)
- Zero-latency direct monitoring greatly simplifies recording and overdubbing
- Multitrack recording and mixing without PCI card installation

USB/Firewire Mobile Studio Interfaces—Take your music anywhere you want

Mobility is about more than just field recording, it's about ease and flexibility. Today the studio is no longer a place, its a process. M-Audio's mobile audio solutions let you use your laptop or any USB- or Firewire-compatible computer to record and play audio whenever and whatever inspiration strikes. One simple connection gives you all the functionality you need for professional desktop music production or making music on the go. M-Audio's mobile audio solutions offer a variety of of configurations to suit every need—including preamps, MIDI and digital I/O—all with M-Audio's legendary driver support spanning all popular operating systems and industry communication standards.

- Mobility and simple connectivity for audio and MIDI
- 24-bit/96kHz audio
- Highest quality components, specifications and fidelity
- Low driver latency, zero-latency hardware monitoring
- Software-controlled internal digital mixing and routing
- Unequalled driver support and software compatibility

QUATTRO USB



4-in/4-out USB Audio/MIDI Interface

The ultimate USB audio interface for professional recording needs, Quattro USB lets you use your laptop or any USB-equipped computer to access the world of digital audio. The Quattro USB couples today's latest technology with pristine A-D/D-A converters to bring you a recording tool that is easy to use and delivers the sound quality of systems that cost many times more.

- 4-in/4-out analog I/O via balanced/unbalanced 1/4″ TRS connectors
- 1x1 MIDI I/O
- Multitrack recording and mixing without PCI card installation
- AC-powered components for optimal fidelity
- Selectable +4dBu/-10dBv input and output levels (in stereo pairs) via hardware supports guitar/keyboard/line inputs as well as professional and consumer equipment.
- Near zero-latency direct monitoring in stereo pairs
- More professional connectivity via the optional Omni I/O (takes on the functionality of OmniStudio USB)

MOBILEPRE USB

USB Bus-Powered Preamp and Audio Interface



A bus-powered USB interface with built-in two-channel mic/instrument preamps, the MobilePre USB eliminates the need for an external power supply, making it ideal for field recording and other laptop-based mobile recording situations. It features two XLR-balanced mic inputs with switchable +48V phantom power, 1/8˝ mini stereo mic jacks for desktop mics, and two 1/4˝ TRS balanced instrument/line inputs. Each input channel also features a gain control knob (+40dB maximum), phantom power indicator, and a signal presence/clipping indicator. Outputs include two unbalanced 1/4˝ connectors, 1/8˝ mini stereo line output, and 1/8˝ mini stereo headphone output with level control. USB MobilePre also offers zero-latency direct hardware monitoring with software control of levels.

- 2x2 16-bit/48kHz analog I/O
- Two XLR-balanced mic inputs with 48v phantom power
- Stereo microphone input (1/8˝)
- Two balanced/unbalanced 1/4˝ TRS high-impedance instrument/line inputs optimized for guitars and basses
- Two unbalanced 1/4˝ line outputs
- 1/8˝ stereo line output
- 1/8˝ stereo headphone output with levels
- Zero-latency hardware direct monitoring (mono/stereo) with software level control

MAXIMUM AUDIO TOOLS BUNDLE



All M-Audio mobile audio solution include M-Audio's Maximum Audio Tools bundle - a great collection of software software that lets you make music immediately. Bundle includes:

- Ableton Live Delta audio sequencer (a basic version of Ableton's revolutionary Live audio sequencer software
- M-Audio ProSession series WAV/REX samples (125MB)
- Arkaos VJ VMP visual performance software (special edition)
- IK Multimedia SampleTank SE VST sample player (special edition)
- DSound RT Express VST host with WAV player (special edition)

AUDIOPHILE USB

USB Audio/MIDI Interface with S/PDIF I/O



Audiophile USB is an affordable, no-hassle USB audio and MIDI interface that builds on the success of the Audiophile 2496—to deliver great frequency response, dynamic range and noise specs. AC-powered components that deliver better fidelity than bus-powered circuitry and the bundled Mac and PC drivers let you independently scale bit depth and sample rate (up to 96k) to deliver the lowest USB latency anywhere—whether you're on stage or in the studio.

- Two pairs of analog line-level audio inputs (1/4˝ unbalanced or RCA)
- Two analog line-level audio outputs (RCA) with level control knob
- S/PDIF in and out (coaxial)
- Supports AC3 and DTS surround sound setups
- 1x1 MIDI I/O
- 1/4˝ headphone output with level control
- AC-powered for high-performance A/D-D/A

OMNISTUDIO USB

Complete USB Mobile Recording Solution



More than just a simple USB interface, the OmniStudio USB gives you more connectivity, flexibility and core studio functionality than any other USB audio product available today. Based on the award-winning Omni Studio, the OmniStudio USB puts everything that you need for audio and MIDI recording, monitoring and mixing in one easy-to-transport package. The on-board mixer provides tremendous routing and monitoring control over a full complement of I/O including mic/line preamps, aux ins, direct outs, separate stereo monitor and record outs, effects send/return and more. OmniStudio USB even has two independent headphone outs with level controls so that you can work with a partner anywhere.

- Two front-panel analog inputs featuring:
  - Neutrik connectors (XLR and 1/4″ TRS)
  - Mic/instrument preamps with 66dB gain
  - 48v phantom power, individual gain controls, 20dB pad and signal/clip indicators
- Line inputs 3/4 (bal./unbalanced 1/4″ TRS)
- Direct monitoring and mono/stereo mode for each input pair
- 1 x 1 MIDI I/O
- 4 stereo aux inputs individually routable for monitoring or recording
- 4 balanced/unbalanced 1/4″ TRS outputs
- Stereo monitor outputs with level control (balanced/unbalanced 1/4″ TRS)
- Stereo record outputs independent of monitor level (balanced/unbalanced 1/4″ TRS)
- Two stereo headphone outputs with individual level controls (1/4″ TRS)
- Effects send with level controls for each of the four D/A or direct monitor signals (unbalanced 1/4″)
- Stereo effects return (bal/unbal. 1/4″ TRS)
- +4dBu/-10dBV hardware switching for line inputs 3/4, direct outs 1/2 and 3/4

About USB and Firewire

Offering excellent plug-and-play mobility compared to PCI, the USB and Firewire protocols each have their own inherent bandwidth that dictates how much audio it can transfer. Where PCI handles 132MB/sec., Firewire delivers 30MB/sec and USB 1.5MB/sec. Greater bandwidths allow you to record more tracks simultaneously than do lower bandwidths. USB meets the the needs of the majority of users who tend to record one or two tracks at a time. Supplied driver control panel allows you to manage bandwidth according to the needs of each project.

FIREWIRE 410

4-in/10-out Firewire Mobile Recording Interface



Delivering powerful performance for Firewire (IEEE1394) equipped computers, the 410's 4-in/10-out configuration is perfect for personal recording where you build up tracks one at a time yet need multiple outputs. FireWire 410's ten outputs allow your computer-based recording system to behave more like a traditional analog recorder or sampler, where signals are bussed to multiple channels of an analog mixer or to multi-channel monitoring systems. Software-controlled DSP handles all internal routing and mixing.

- Two analog input channels each featuring:
  - Balanced XLR and unbalanced 1/4″ mic/line inputs
  - Preamp with level control, signal/clip LEDs and amazing available gain of 66dB
  - 48v phantom power (globally switched)
  - 20 dB pad
- 8-line outputs with signal/clip LEDs (unbalanced 1/4″ @ -10dBV)
- Stereo monitor level control
- Two-channel S/PDIF digital I/O (coaxial and optical, switched)
- 1 x 1 MIDI I/O with bypass for stand-alone operation
- Two headphone outs with individual level controls (1/4″ TRS)
- Two Firewire high-speed ports for connection convenience
- Low-latency software monitoring;
- Zero-latency direct hardware monitoring
- AC3 and DTS surround support via digital output
- Analog outs can directly drive up to 7.1 surround w/o decoder using included software bass management
- Standalone operation via AC adapter

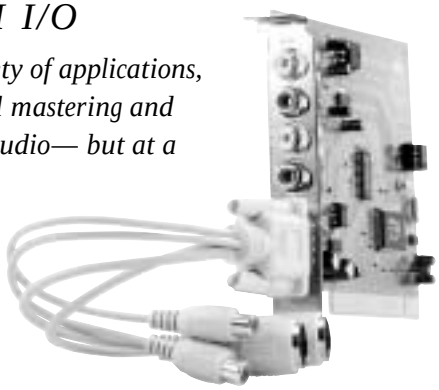
PCI INTERFACES

The best-selling audio cards in the world, the Delta series span all studio configurations, feature sets, operating systems, and price points. Full-duplex architecture allows recording and playback simultaneously, while low-latency drivers and zero-latency hardware monitoring insure that your overdubs will be in sync. Each card features outstanding driver compatibility and stability.

- Ultimate support for multitrack hard disk recording and software synths/samplers
- Highest quality components, specifications and fidelity
- Unequalled driver support and software compatibility
- 24-bit/96kHz full-duplex audio
- Low driver latency (as low as 1.5ms)
- Zero-latency hardware monitoring
- 36-bit internal DSP with powerful software-controlled digital mixing and routing
- Expansion flexibility in combining and synchronizing Delta cards
- S/PDIF provides bit-accurate digital transfers and SCMS copy protection control
- Surround-sound ready-outputs AC3 and DTS over S/PDIF to an external decoder

AUDIOPHILE 2496

4-in/4-out PCI Audio Card with S/PDIF and MIDI I/O



The Audiophile 2496 is an all-in-one high fidelity soundcard solution for a wide variety of applications, ranging from multi-track recording, LP/cassette-to-CD and digital transfers, to digital mastering and computer-based home theater. It delivers uncompromising professional grade digital audio— but at a price anyone can afford.

- 2x2 analog I/O (unbalanced gold-plated RCA jacks) utilizing professional 24-bit, 96kHz conversion.
- Digital S/PDIF (coaxial) I/O and MIDI I/O provide connectivity to both digital devices, and the world of MIDI.
- Use the S/PDIF I/O for DVD surround sound playback with an external decoder.
- Offers powerful digital mixing, routing and monitoring capabilities with included software, as well as control over SCMS (Serial Copy Management System).



DELTA 410

4-in/10-out Audio Card with Digital I/O

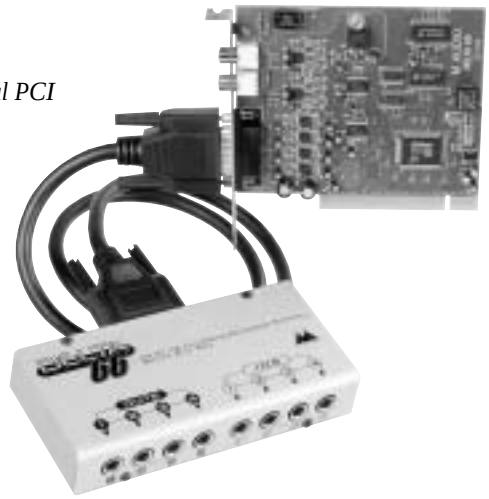
Recordings are often put together one track at a time (a guitar solo here, a vocal there) as opposed to being all tracked simultaneously. So why pay for inputs you don't need. The Delta 410's 4-in/10-out configuration is created with exactly this type of recording approach in mind. You get two analog and one stereo digital input to accommodate keyboards, preamps, CD/MiniDisc players, DATs and other gear. Ten outputs place the I/O emphasis where you need it—flexibility in connecting to a mixer and effects. This configuration allows your computer-based recording system to behave more like a traditional analog one recorder or sampler, where signals are bussed to and from a mixer in the analog domain.

- 4x10 24-bit/96kHz full-duplex recording interface
- Two analog inputs (-10dBV unbalanced RCA) for line-level gear
- 8 analog outputs (unbalanced RCA)
- S/PDIF digital I/O with SCMS copy protection control (coaxial)
- AC3 and DTS surround support via digital output. Directly drive up to 7.1 surround without decoder with included software.
- Comprehensive digital mixing, routing, and monitoring capabilities with the bundled Delta Control Panel software

M-AUDIO

DELTA 44/DELTA 66

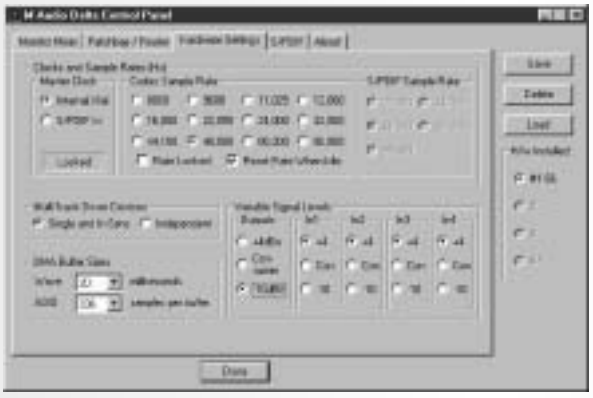
Professional 4-in/4-out Audio Cards



A top choice in PCI-based digital audio solutions, the Delta 44 and Delta 66 are professional PCI audio cards connected to a analog break-out box that provides four analog audio input and outputs (1/4″ TRS), compatible with balanced or unbalanced +4dB or -10dB signal levels. Also included is Delta Control Panel software, which allows you to control their comprehensive routing, monitoring, and mixing capabilities. Besides their superb sound quality, performance, and driver support, they provide 96 kHz recording, zero latency monitoring, digital mixing, and controllable SCMS. Otherwise identical, the Delta 66 adds a S/PDIF digital I/O with SCMS copy protection control (coaxial).
If you need more professional connectivity, simply add the optional Omni I/O (see Omni Studio below). With the Omni I/O as their front end, they gain 2 pristine mic preamps, an additional 8 inputs to its mixing section, 2 headphones sends, an FX send and return, dedicated monitor outs and record outs, auxiliary input record bussing, and more.

- 4x4 or 6x6 24-bit/96kHz full-duplex recording interface
- PCI Host card with external audio break-out box
- 4x4 analog break-out box (balanced/unbalanced 1/4″ TRS)
- Analog I/O configurable for +4dB and -10dB signal levels
- PCI card provides S/PDIF I/O on RCA jacks (Delta 66 only)

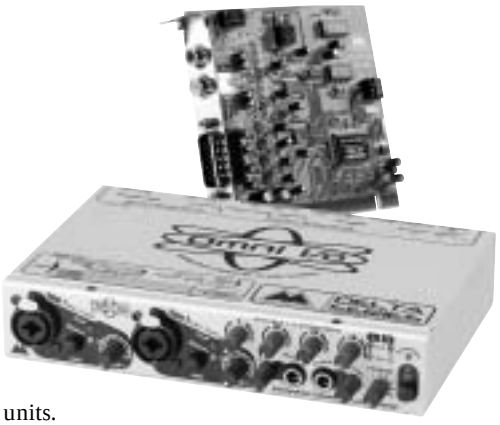
The Delta control panel offers full control over features such as sample rate, direct monitoring, master/slave sync, surround and bass management, latency and more—for up to four Delta cards



All Delta cards contain a 36-bit embedded DSP that offers the feature set of a full project studio mixing console. The Delta DSP seamlessly integrates analog and digital inputs with the ability to route them via a software-driven patchbay/router. You get flexibility in analog and digital configurations with extremely fast throughput for low-latency software monitoring. A single unified control panel provides settings for clock and sample rates, buffer sizes, individual signal levels for every input and output, adjustable +4dBu/-10dBV pad controls, and digital I/O control on up to four installed Delta cards. All controls are also easily accessible through most professional audio software applications for seamless integration and operation.

OMNI STUDIO

Delta 66 + Omni I/O -
The Complete PCI Desktop Studio Solution

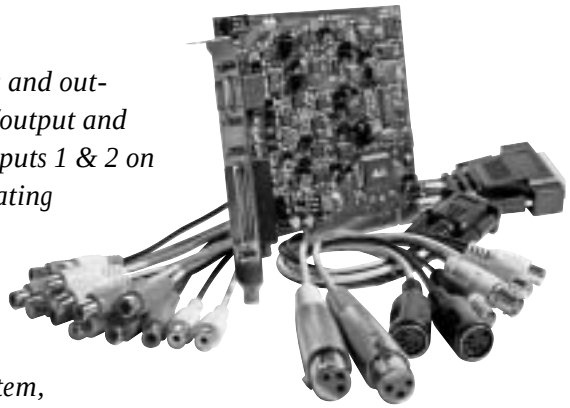


A unique and powerful songwriting/production tool — the Omni Studio is the ultimate complete recording package for Mac and PC, offering versatility and performance that will change the way you make your music. Consisting of the Delta 66 PCI card, and the Omni I/O interface, Omni Studio is designed to let you record, monitor and mix projects while streamlining your setup and giving you the tools you need to produce music more professionally and easily than ever before. Omni Studio allows you to:

- ◆ Mix to digital (CD, DAT, minidisk) and cassette without re-patching cables.
- ◆ Record vocals, guitars, keyboards, software synths, loops, samples, turntables, and external FX units.
- ◆ Monitor keyboards, MIDI performances, FX sends before recording the FX, using 2 headphones and/or speakers.

DELTA 1010LT

10-in/10-out Virtual Studio

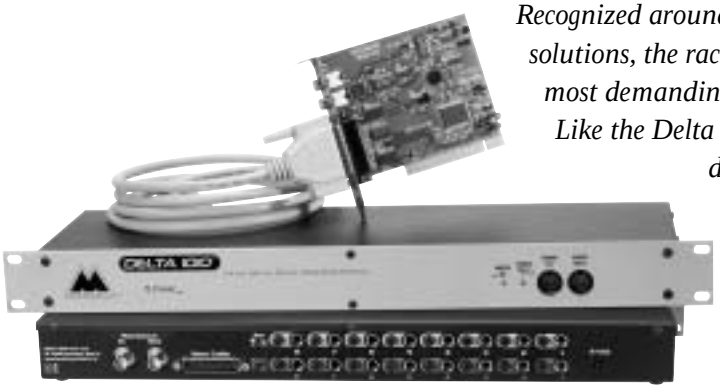


The Delta 1010-LT offers a host of interfaces including, eight analog inputs and outputs, two XLR microphone inputs with preamps, S/PDIF and MIDI input/output and wordclock I/O, allowing audio from a variety of sources to be connected. Inputs 1 & 2 on XLR jacks can be set to receive a balanced line or a mic level signal, eliminating the need for an outboard mic preamp. The S/PDIF input offers control over the SCMS copy protection protocol. The card offers zero latency hardware monitoring, and the bundled Delta Control Panel software can set and even fine-tune the input line level.
Delta 1010-LT outputs can be connected to an external mixer or sound system, while the S/PDIF outs feed your digital mixdown deck. The Control Panel Software also controls a hardware mixer embedded in the card, giving still more routing and mixing control. Use the MIDI I/O for your MIDI controllers and sound modules, and finally, connect to wordclock capable devices for rock-solid sample clock synchronization.

- 10-in/10-out 24-bit 96kHz audio card with mic preamps, S/PDIF and MIDI I/O
- 8x8 analog I/O (unbalanced gold-plated RCA connectors at -10dBv to +4dBu)
- Two balanced-XLRmic/line inputs with preamps (mic or line level is selected via hardware jumpers).
- +4dBu/-10dBV operation individually switched in software
- 1x1 MIDI I/O
- S/PDIF digital I/O with SCMS copy protection control (coaxial)
- External synchronization via word Clock I/O on BNC connectors
- AC3 and DTS surround support via digital output. Directly drive up to 7.1 surround without decoder with included software.
- Comprehensive digital mixing, routing, and monitoring capabilities with bundled Delta Control Panel software

DELTA 1010

10-in/10-out PCI-Based Digital Audio System



Recognized around the world as the top choice in host-based PCI digital audio solutions, the rackmount Delta 1010 is designed to handle the needs of the most demanding project and professional studios.
Like the Delta 1010LT, the Delta 1010 functions as a 10-input, 10-output digital recording interface. However, it consists of a PCI card and breakout box and features eight balanced/unbalanced 1/4˝ TRS analog inputs and outputs plus coaxial S/PDIF I/O—and doesn't have mic preamps. The sturdy rack-mount unit houses all of the A/D and D/A converters, keeping them away from the internal noise of your computer and assuring the best audio performance possible. (D/A converters rated at 117dB, A/D converters at 110dB). Also located on the rack unit is Word Clock and MIDI I/O.

- 8x8 analog I/O (balanced and unbalanced 1/4˝ TRS)
- S/PDIF digital I/O with SCMS copy protection control (coaxial)
- 1x1 MIDI I/O
- External synchronization via word Clock I/O on BNC connectors
- AC3 and DTS surround support via digital output. Directly drive up to 7.1 surround without decoder with included software.
- Combine several Delta 1010's for as much I/O as you need
- Comprehensive digital mixing, routing, and monitoring capabilities with bundled Delta Control Panel software

M-AUDIO

USB MIDISPORT SERIES

Midiman USB MIDI Interfaces

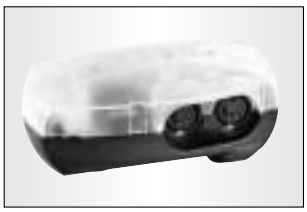
Midiman's USB MIDISPORT series is the world's best selling of family of USB MIDI interfaces. Available in various port configurations, they work with USB-equipped Mac and Windows computers to assure reliable, immediate MIDI transfers. Installation is a breeze—no computer disassembly, no jumpers to set, and no I/O addresses, IRQ's or DMA channels to configure. Expansion is just as easy—just plug in additional MIDISPORT units for more MIDI ports.

They All Feature:

- Available in 16, 32, 64 and 128 MIDI channel configurations
- High speed connection between the interface and computer via USB
- Easy expansion — just plug in additional MIDISPORT units as needed
- MIDI activity LEDs for each port
- True plug-and-play installation -no tools required
- Interfaces are hot swappable meaning they can be plugged or unplugged from the USB bus at any time
- The smaller interfaces get power from the host computer via USB so no external power supply required — ideal for laptop computers. Each includes a 6´ USB cable
- Each interface is covered by a lifetime warranty

USB MIDISPORT 1x1

Sub-Compact, Single Port USB MIDI Interface



- 1 MIDI In port and 1 MIDI Out port provides 16 MIDI channels
- Power to the interface is supplied by the computer via the high-speed USB connection
- LED indicates the USB port is functioning properly.
- Sub-compact lightweight design — ideal for use with notebook computers, as an entry level interface or as a simple interface where only limited MIDI I/O is required.

USB MidiSport 1x1.....**48.95**

USB MIDISPORT 2x2

Sub-Compact, Dual Port USB MIDI Interface



Same as 1x1 PLUS—

- 2 MIDI In and 2 MIDI Out ports provide 32 MIDI channels
- USB/MIDI Thru button lets you play your gear even when the computer is turned off — data from MIDI In-A is sent to MIDI Out-A and data from MIDI In-B is sent to MIDI Out-B.
- Sub-compact design with all-metal casing — ideal for use with notebooks, as an entry level interface or simple MIDI setups

USB MidiSport 2x2.....**69.95**

USB MIDISPORT 2x4

Multi-Port USB MIDI Interface

Same as 2x2 PLUS—



- 2x4 architecture lets you go beyond the MIDI specs limit of 16 channels. Two separate 16-channel input ports allow your system to accept up to 32 discrete MIDI input channels, and four individual 16-channel output ports allow you to address up to64 discrete MIDI output channels.

USB MidiSport 2x4.....**Call**

USB MIDISPORT 4x4

Multi-Port USB MIDI Interface

Same as 2x2 PLUS—



- 4 MIDI In and 4 MIDI Out ports provide 64 MIDI channels
- 1/2 rack design with all metal casing — ideal for desktop music systems with moderate MIDI requirements, for extra I/O in an expanded system, or as a truly flexible and roadworthy interface for notebook computers.

USB MidiSport 4x4.....**134.95**

USB MIDISPORT 8x8

Rackmounted Multiport USB MIDI Interface/ SMPTE Synchronizer



The flagship of the MIDISPORT line, the USB MIDISPORT 8x8 is a powerhouse with features and bandwidth to spare. It features rock-solid drivers on both Mac and Windows, USB and serial capability (in addition to being a gateway to the high speed and convenience of USB, the MIDISPORT 8x8 can be connected to the "legacy" serial ports of older computers) and full SMPTE implementation.

FEATURES

- 8 MIDI In and 8 MIDI Out ports provide 128 MIDI channels with the 8th I/O ports located on the front panel for easy access
- High-speed connection to USB-equipped PC or Mac computers as well as a 8-pin serial connection for older PC or Macs
- Select between computer-based modes—USB, PC Serial and Mac Serial as well as the four standalone modes — MIDI Patchbay, Thru, Cable Test and SMPTE Writer
- MIDI Reset (panic) button send an "all notes off" and "all controllers off" message to the MIDISPORT outputs in the event of a stuck note or MIDI overload.
- THRU Mode sends every message at the MIDI In ports on to all of the Out ports, helping diagnose connection problems.
- Includes an 8-Pin serial port for non-USB Mac and Windows 95/98 computers.
- Operates as a stand-alone MIDI patchbay and a SMPTE to MTC synchronizer
- Bundled remote control software (Mac and Windows compatible) enables you to control the interface's SMPTE functions and MIDI patchbay configurations from your computer. Configuration settings can be saved to disk and reloaded for later use.
- Built-in "MIDI cable tester"mode lets you test a MIDI cable by connecting it between In and Out ports on the front panel.
- Create and store up to 8 MIDI patchbay programs (routing any combination of MIDI Input to MIDI Output).
- MIDI In/Out Indicator LEDs (1-8)
- 1 RU all metal construction — ideal for larger MIDI or hard disk recording systems
- Powered by a DC 9v wall wart power supply

SMPTE Timecode Operation

- SMPTE Time Code Writer/Reader supports 24, 25, 29.97, 30 drop or 30 non-drop frame LTC SMPTE formats.
- Converts SMPTE LTC to MIDI Time Code (MTC) for syncing a video deck or multitrack recorder with MTC capable software.
- "Writing" and "Locked" LEDs indicate SMPTE writer and reader/regenerator status, which indicates that a valid SMPTE code is recognized at the SMPTE In jack.
- The Flywheel Control, found in the included Remote Control software, selects the number of frames, from 1 frame to 255 frames, or infinite flywheel (Jam sync), that will be tolerated if there is a dropout in the incoming SMPTE timecode signal.
- 1/4″ SMPTE I/O jacks on the rear panel

MIDIMAN PC & MAC COMPATIBLE MIDI INTERFACES

	BiPort 2x4	MIDISport 1x1	MIDISport 2x2	MIDISport 4x4	MIDISport 8x8
# of MIDI In Ports	2	1	2	4	8
# of MIDI Out Ports	4	1	2	4	8
# of MIDI In Channels	32	16	32	64	128
# of MIDI Out Channels	64	16	32	64	128
Computer Connection	9-Pin Serial	USB	USB	USB	USB/Serial
Mac OS Requirements		8.6 or higher	8.6 or higher	8.6 or higher	8.6 or higher
Required Mac MIDI driver	OMS (included)	OMS (included)	OMS (included)	OMS (included)	OMS (included)
Windows Compatibility	95/98/NT/2000	98/ME/2000	98/ME/2000	98/ME/2000	98/ME/2000
SMPTE Reader/Generator	Yes	No	No	No	Yes
Footprint	Compact	Sub-Compact	Compact	1/2 Rack	1RU
Power Requirements	9V DC wall wart	Self powered	Self powered	Self powered	9V DC wall wart

M-AUDIO

OXYGEN 8



25-Key Portable USB MIDI Keyboard Controller

A new breed of controller designed to meet the needs of today's electronic musicians, Oxygen 8 puts you in control of any 8 MIDI-assignable parameters within your favorite software programs. Oxygen 8 is also a USB MIDI interface that speaks directly with your computer without any extra devices. You don't need a MIDI interface for this keyboard interface.

Applications:

- The software synth is here to stay, and along with it comes dozens of parameters to be tweaked. Oxygen 8 puts your hands on any 8 parameters of your choice in real time.
- Oxygen 8 lets you get deeper into your software programs, to get more out of them, to experiment and push them to their limits.
- Gives you more freedom on the stage than you've ever had before, dynamically increasing the possibilities available to you.
- Instead of taking hours to program drums with a mouse, use Oxygen 8 to play grooves into your program in real time. Use the 8 knobs to manipulate the dynamics, feel, and FX.
- Synthesists can use Oxygen 8 to gain access to, and to automate parameters that exist deep within their hardware, making it easier to get more out of them without having to navigate through dozens of menus.

Features:

- 8 MIDI assignable knobs, one assignable slider
- Inputs and outputs 16 channels of MIDI to and from your computer using the power and convenience of your USB port.
- Fully functional, velocity sensitive 25 key MIDI controller keyboard, capable of sending all MIDI messages, such as modulation, pitch-bend, MIDI volume, and more.
- Powered via USB port or 6 "AA" batteries, or with the supplied power supply.

What is a MIDI Controller?

Unlike traditional keyboards, MIDI controllers have no sound built into them. They are optimized to control other MIDI sound sources such as synthesizers, samplers and today's popular software synths. For example, here's how you play sounds within a software program:
Because of M-Audio's driver technology, your computer will recognize your Keystation as a MIDI interface with which it can communicate. By playing the keyboard, you send MIDI information to your computer, which triggers sounds in a software program that are played via the audio output to your computer's sound card.You can also send MIDI via the Keystation's dedicated MIDI output port to external MIDI sound modules like synths and samplers.

OZONE

USB Audio/MIDI Mobile Workstation



Your laptop changed the way you live—and now new mobile tools can change the way you make music. The revolutionary Ozone is an all-in-one keyboard, MIDI control surface, mic preamp, USB audio interface and USB MIDI interface that's no bigger than a laptop. Combined with your computer and software like Reason and Live, Ozone provides a complete personal mobile studio from input to output.

MIDI Features:

- Built-in 16-channel USB MIDI interface
- 25-note keyboard with full-sized keys
- Eight assignable MIDI controller knobs
- Pitch and mod wheels
- Sustain pedal jack
- Two MIDI outs (one from computer, one from Ozone)

Audio Features:

- Built-in 2 x 2 24-bit/96kHz audio interface
- Stereo out (balanced 1/4" TRS)
- Mic input (XLR) with built-in preamp and phantom power
- Instrument and Stereo aux in (bal. 1/4˝ TRS)
- Stereo headphone out
- Monitor switch for zero-latency monitoring

RADIUM

49- and 61-Key USB MIDI Controllers with Realtime Knobs and Sliders



A new breed of controller, Radium is designed to meet the needs of today's electronic musicians. The Radium 49 (49-key/4 octave) and Radium 61 (61-key/5-octave) give even the most serious players a keyboard that they can love, combined with 16 controllers that can be mapped to any MIDI-assignable parameters in your hardware and software rig. The complement of 8 MIDI- assignable sliders are ideal for mixing with virtual faders, while the 8 MIDI-assignable knobs are great for panning and effects send— and both are great for real-time performance control of synth parameters.

They are also a USB MIDI interface that speaks directly with your computer without any extra devices. Radium is perfect for applications like real-time soft synth control, virtual mixing, drum programming, live performance and more. And they weigh in at only 7 lbs. so you can take them anywhere.

MAXIMUM AUDIO TOOLS BUNDLE



All M-Audio MIDI controllers include M-Audio's Maximum Audio Tools bundle - a great collection of software software that lets you make music immediately.
Bundle includes:

- Ableton Live Delta audio sequencer (a basic version of Ableton's revolutionary Live audio sequencer software
- M-Audio ProSession series WAV/REX samples (125MB)
- Arkaos VJ VMP visual performance software (special edition)
- IK Multimedia SampleTank SE VST sample player (special edition)
- DSound RT Express VST host with WAV player (special edition)

Radium Features

EASY CONTROL

Controls aren't buried mysteriously in obscure menus. The keys have a second layer of functions, allowing you to program MIDI control information directly from the black and whites. Here's how it works:

- Press the MIDI/Select button
- Press the note on the keyboard with label corresponds to the parameter you wish to change
- Use the Data Entry slider to select the value or enter numeric values by pressing the keys labeled with numbers

- 49- or 61-note keyboard
- Who wants to mix with a mouse? Use Radium's 8 sliders to control your virtual faders, while Radium's 8 knobs control pan, effects send amount and much more.
- USB port routes 16 channels of MIDI I/O directly to and from your computer
- Separate MIDI Out jack routes MIDI signals from your computer to control external gear—no separate interface needed
- MIDI Out jack routes MIDI signals from Radium to control external gear
- Pitch and modulation wheels
- Octave up and down buttons extend keyboard range
- Fully functional, MIDI controller keyboard sends all MIDI messages, such as modulation, pitch-bend, MIDI volume and more
- Keyboard doubles as a data input device
- Lightweight (7 lbs.)—easy to take wherever the music is
- Powered via USB port, 9v DC power supply or AC Adapter

MACKIE CONTROL

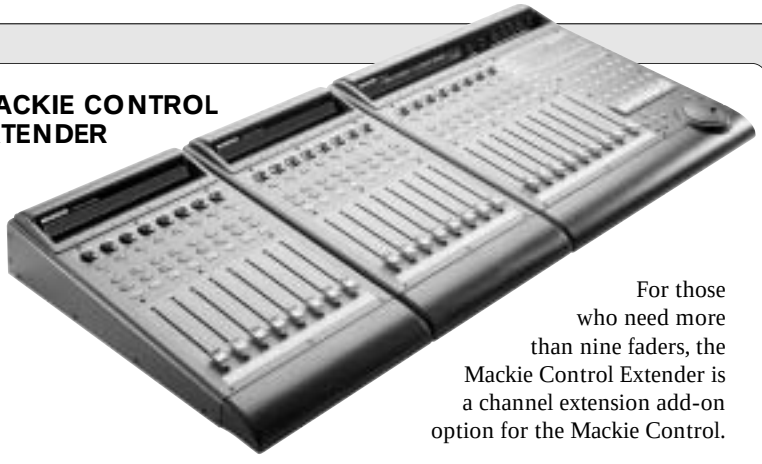
Automated Touch-Sensitive Control Surface



Mackie Control is a nine-fader (eight channels and master) MIDI controller that provides in-depth mixing, editing, automation, and navigational control for any supported digital audio workstation. The Mackie Control adheres to a compact, desktop-style design and offers a full range of professional-grade features and controls. Mackie Control has eight channel strips, each of which includes a Penny and Giles 100mm touch-sensitive motorized fader, mute, solo, and record arming buttons, Signal LED, a Channel Select button, and a V-Pot that can be used to control panning, send levels, EQ, dynamics, and other DSP plug-in modules. The eight channels can be bank-switched to control as many channels as the user-selected software supports. Various software parameters are displayed on a back-lit LCD screen that provides channel metering and track naming in addition to DSP and dynamics parameters. Mackie Control also supports a full automation assignment section, edit buttons, Undo, Redo, and Save buttons, and a Shift Modifier that offers dual functionality for specific commands. Complete navigational control is provided in a tape-style transport with Jog/Shuttle wheel, Loop In/Out points, and timeline quick jump buttons. All controls live within a small, tabletop, analog-style control surface with a backlit LCD display that is comfortable and enjoyable to use.

Mission-specific software "hooks" (or specialized commands) have been written by MOTU, Steinberg, and Cakewalk to support the Mackie Control. DAW-specific Lexan overlays that fit right over the master section can be used to "personalize" the Control for SONAR, Digital Performer, Nuendo or Cubase SX. Additional support and overlays will be available in the future for applications like Syntrillium's CoolEdit Pro.

MACKIE CONTROL EXTENDER



For those who need more than nine faders, the Mackie Control Extender is a channel extension add-on option for the Mackie Control.

Slightly narrower than a full Mackie Control, the Mackie Control Extender has all the channel strip features of the Mackie Control—only without the master section. Multiple Extenders can be used with a single Mackie Control to emulate a large-scale control surface with as many faders and pots as desired.

What is a "Lexan Overlay"?

A Lexan overlay is a thin sheet of polycarbonate that has DAW-specific nomenclature on the front side and an adhesive on the back. It is designed to be placed over the master section of the Mackie Control and associate the buttons and controls with specific software commands. Because the Mackie Control is supported by a number of different DAW applications, each DAW manufacturer has been given the opportunity to design and implement their own overlay. Using the overlay that corresponds with your DAW software will give you a personalized, mission-specific controller with all the commands for your particular platform at your fingertips

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ONE HOUR FREE PARKING
AT 349 W. 34th STREET (with purchase of $100 or more)

MACKIE CONTROL

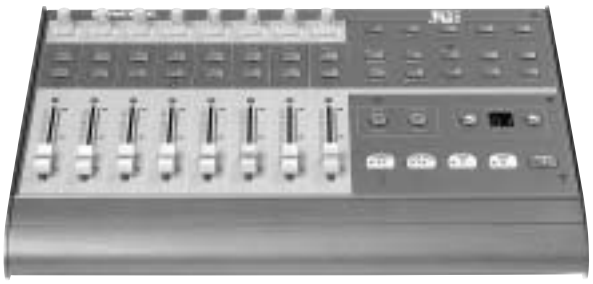
Mackie Control C4—Dedicated Plug-In/Virtual Instrument Controller



The Mackie Control C4 is a highly sophisticated plug-in and virtual instrument controller that provides four banks of eight V-Pots each, and four full-size displays for instant access to as many as 32 separate software parameters without the need for scrolling or bank switching. The perfect tool for desktop music makers, the C4 lets you intuitively control multiple plug-ins and virtual instruments simultaneously with shortcut buttons and four independent backlit LCD displays. Computer-based music-making has never been this fast.

- Eliminates the need to constantly "bank switch" a standard control surface display in order to access parameter-rich software modules.
- Has multiple displays and corresponding knobs for simultaneous access of up to 32 parameters of a given software module.
- Function buttons beneath the displays and knobs let you select the arrangement of those parameters and how they show up on screen.
- The C4 also acts as a "quick fix" tool for fader and pan settings across 32 channels, and can be used independently or with additional Mackie Control and/or Mackie Control Extenders.

BABY HUI *Compact Touch-Fader Master Controller*



Baby HUI is the direct descendant of HUI (Human User Interface). Originally released by Mackie in 1997, HUI quickly became one of the audio industry's first totally integrated control surfaces for digital audio workstations. With a similar feature set of single-button automation, editing, and navigational commands, complimented with eight full-featured channel strips that feature a touch-sensitive motorized fader, Baby HUI offers professional-grade features and capabilities in a compact, affordable package.

- Baby HUI is an eight-fader MIDI controller that provides in-depth mixing, automation, and navigational control for any digital audio workstation that supports the HUI MIDI mapping protocol.
- Supports all the major features found on the original HUI - only in a smaller footprint and at a more affordable price point.
- Perfect for DAW enthusiasts on a budget, or space-conscious engineers who have a very small area to use with a controller.
- Eight channel strips that include a 60 mm touch-sensitive motorized fader, mute button, solo button, and a Signal LED.
- Each channel strip also has a multi-functional rotary encoder that can be used for pan or send level control, or can be pushed in to provide Channel Select, Record Arming, or Automation Arming functions.
- One of the key features of Baby HUI is the extensive software support that already exists among the various DAW platforms - today! Baby HUI will operate and interact with any software that supports the HUI MIDI Mapping protocol. This is a HUGE advantage over generic MIDI controllers that require you to map every single control with every software parameter - a process that can take hours away from being creative and enjoying the product.
- Software Window Open/Close Buttons
- For any HUI-supported system, simply plugging Baby HUI into your computer's MIDI interface, accesses the control surface or console manager within the software, and activates HUI as a controller. As long as the MIDI connections and software interfacing is correct, Baby HUI will "wake up" and allow itself to be used.
- In/out buttons for location point insertion on the fly
- Easy to use MIDI In/Out connections
- Supports a full automation assignment section, display buttons, and Undo button and a Shift Modifier to double up certain controls.
- Basic navigational control can be achieved via the tape-style transport and RTZ/END buttons, and any channel can be accessed with the Bank/Channel Select buttons.

Powered By:

- Digidesign Digi001 and Pro Tools
- MOTU Digital Performer
- Steinberg Nuendo
- Steinberg Cubase SX
- Mackie Mixtreme and Soundscape 32

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HUI

Large Touch-Fader Master Controller for Digital Audio Workstations



Designed for video, film, multimedia and recording studio professionals, Mackie's HUI (Human User Interface) was the original hands-on control surface for Pro Tools, Digital Performer, Cubase and Nuendo digital audio workstations (DAWs). HUI did for digital audio workstation users what the mouse did for computer users: Familiar physical movement is instantly translated into precise creative control. For example, HUI includes what appears to the user to be familiar rotary knobs... but the DAW 'sees' a virtual interface that can be stored and recalled digitally. This complete hardware workstation console has the user-friendly features and ergonomics that Mackie is renowned for… including instantly touch-updatable 100mm motorized faders, transport controls, DAE plug-ins control, keyboard shortcuts, window buttons - even built-in mic preamps! What used to take multiple mouse moves and clicks can now be done at the push of a single button. Instead of individually adjusting "faders" on screen,users can now enjoy hands-on tracking and mixing with touch-updatable motorized faders. With full automation control and recall capabilities, HUI is anything but a conventional fader pack. For serious professionals who work day in and day out with Pro Tools, Digital Performer, Cubase or Nuendo, HUI can significantly boost productivity through direct hands-on control.

FEATURES

- Eight 100mm professional grade, touch update, readily re-assignable, motor faders. The quick reactive, conductive metalized fader knobs respond immediately to touch control without separate switch enabling or disabling.
- The motor fader design utilizes a high resolution, tightly tuned servo amplifier to reduce annoying fader chatter, and the conductive plastic fader servo track greatly extends fader lifetime.
- Per channel 4-character LED scribble strip, for updating the Channel ID on the fly and for displaying channel attributes: e.g. group membership, signal phase, source, routing, etc.
- Dedicated per channel Pan and Send encoders and controls for routing, assignment, status, edit and automation control.
- "V-Pots" is a cost effective and novel Mackie "soft" knob design, with illuminated LED segments in the collar that update to reflect values according to function.
- Highly tactile illuminated switches for mute, solo, automation write and record-ready enabling.
- Dual LED ladders per channel for level display status - e.g. mono/stereo levels, dynamics, etc.
- Dedicated plug-in section-to edit and automate your favorite DSP functions. Controllers include four rotary encoders with enable switches and a 2x40 LCD panel, with each parameter section containing callout display and precise values. Plus easy paging, assignment and bypass switching.
- Sectioned switch matrix containing automation controls, status views and F-keys for macro style operations.
- Fader view, recall, and reassignment/ dedicated bank and channel selectors.
- Analog control room section with talkback, and assignable routing for three stereo or surround audio pairs, with separate output trims and a Master control. Additional master mute, dim, mono monitoring and discrete one to one switching are included in this section.
- Transport controls, jog encoder, zoom and navigation cursors, locate/number pad.
- Rear panel includes two patchable mic preamp sections, an external talkback mic pre-amp and trigger, three pairs of analog I/O jacks (configurable for stereo or surround application), headphone out, four general purpose I/O trigger jacks, 2 sets of ADB mouse/keyboard through-ports, MIDI I/O, an expansion port for joystick or additional V-POTS control and an RS-232/422 port for direct computer connection.

HUI

HUI is organized into seven parts:

- In the upper-left corner is an assign section where internal mix bus and physical hardware I/O assignments are made.
- The main part of HUI is composed of eight assignable channel strips with big-console-grade, 100mm motorized faders and Mackie's innovative V-Pot rotary controls.
- In the upper-right corner of the interface is a vacuum fluorescent alphanumeric display for controlling the parameters of DAE-compatible plug-Ins.
- Below the DSP section is a switch matrix for assigning channel status and/or global session attributes.
- The lower right-hand section, with keypad and jog wheel, dedicated keys, and even extra cursor controls, is dedicated to transport/locate functions.
- The lower left contains a keyboard shortcut section bank and selector buttons to switch between various Pro Tools screens.
- Shoe-horned in the middle of all this is a complete analog control room section with 3 sets of speaker/phones controls and built-in talkback mic. (Trim controls for the two integrated mic preamplifiers and talkback mic are on the back.)



	Baby HUI	Mackie Control	HUI
Controls			
Faders, type	(8) 60mm motorized touch sens	(9) 100mm motorized touch sens	(9) 100mm motorized touch sens
Potentiometers	(8) Rotary Encoders w/Push Select	(12) V-Pots w/LED indicator ring	(12) V-Pots w/LED indicator ring
Channel Metering	Channel Sig LED	Sig LED & LCD Display Metering	Channel and Master LED Ladders
Dynamic Controls	No	EQ, Comp, Gate	EQ, Comp, Gate
DSP Control	Send Levels	Fat Channel LCD Display w/4 V-Pots	Fat Channel VFD w/(4) V-Pots
Monitor	Controller Only	Controller Only	6x6 Monitor Matrix
Expansion Options	Two or more on multi-port MIDI interface	Mackie Control Extender and Fat Channel	Two or more on multi-port MIDI interface
Automation Modes	Write, Read, Touch, Off	Write, Read, Touch, Trim, Off	Write, Read, Touch, Trim, Latch, Off
Automation Arming	Faders, Mutes, Pans, Send Levels	Faders, Mutes, Pans, Send Levels	Faders, Mutes, Pans, Send Levels
Window Open/Close	Edit, Mixer, Transport, Mem Loc	Edit, Mixer, Instruments, Returns, Busses	Edit, Mixer, Transport, Mem Loc
Editing Controls	Undo	Cut/Copy/Paste	
SOFTWARE			
Digidesign Pro Tools	Yes	No	Yes
MOTU Digital Performer	Yes	Yes	Yes
Steinberg Nuendo	Yes	Yes	Yes
Steinberg Cubase SX	Yes	Yes	Yes
Cakewalk Sonar	No	Yes	No
Syntrillium CoolEdit Pro	No	Yes	No
Soundscape 32	Yes	Yes	Yes
Mixtreme	Yes	No	Yes
Dimensions	10.1 x 14.6 x 3.4″	17.4 x 17.5 x3.8″ -Extender 10″ wide	22.6 x 20.8 x 6.9″
Weight	7.6 lbs.	14 .57 lbs.	37 lbs.

MIXED LOGIC

M24

Dedicated Remote Control Surface



The Mixed Logic M24 provides familiar hands-on control for the computer-based audio production environment. It offers a traditional recording console style interface designed to integrate with your favorite digital audio software and provide the best of the virtual and physical worlds. The M24 will complement your existing recording software, improve your efficiency and enhance the recording experience. Works with Cakewalk Sonar, Digidesign Pro Tools LE, TDM and HD, Steinberg Nuendo and Cubase, MOTU Digital Performer, Emagic Logic Audio, and C-Mexx C-Console for the Yamaha DSP Factory. Many more programs will currently work with the M24 under generic support.
The M24 has generic modes in addition to the dedicated host modes. One of the generic modes has bi-directional control of over 1000 different MIDI controllers at a time and keeps track of current values as you jump from channel to channel.

FEATURES

Channel Faders

- Has 24 motorized 100mm touch-sensitive Alps faders with 10-bit internal resolution (1024 steps) and scalable MIDI resolution depending on the host software requirements.
- The faders are assigned in banks to control the on-screen faders in the host software. If you move a channel fader the corresponding on-screen fader will follow the movement and moving any on-screen fader in the current bank will move the corresponding M24 fader to match the corresponding position.
- The M24 faders can also create or update automation. The fader range is in steps as fine as 0.1 DB.
- The M24 faders can control audio track faders, master faders, effect returns and aux inputs, group and volume levels of midi tracks in the normal mode. In flip mode or by reassignment, faders can control just about any other function.
- Moving any fader will cause the LCD display to display the corresponding fader's channel name and control name and fader value on the top line of the LCD display and update the corresponding channel scribble.

Pan Controls

- The M24's 24 rotary controls are used for panning and correspond to the on-screen pan controls in the current selected bank. The pan controls can also double for other functions depending on the host software.

Aux Send Controls, Dynamics

- The top row has 16 rotary controls and 16 switches and LEDs for auxiliary effect send control and dynamics control.
- The number of available auxiliary send controls per channel depends on the current host software.
- All 16 controls can also double for other functions including plug-ins depending on the host software.
- Each channel in the current bank has a select switch that is used to make the Aux Sends, dynamics and EQ and plug in controls active for the selected channel.
- The 16 switches in the top row are used for Pre / Post select or dynamics functions and double for Window select modes, Function switch modes and Aux send mutes.

Equalizer Controls

- Imagine having one type of plug in EQ on several channels and another type on several channels and with the push of one switch you could jump from channel to channel and be able to instantly use the same familiar controls to control similar plug in functions.
- The M24 is the first control surface to feature a complete dedicated equalizer section with 12 controls divided into 4 bands.
- Controls are provided for Gain, Frequency and Width on each band with a band bypass and EQ type switch for each band.
- A soft control is provided for access to additional bands assigned in pages (software dependent) or for use as a trim control.
- Depending on the host software the EQ section can be configured to control a large number of software plug in effects with the ability to use the same common controls for similar functions in effects from different manufacturers across all channels.

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M24

Transport and Navigation

- The M24 has traditional tape style transport controls with additional controls for navigation, selection, editing, menu control and parameter adjustment.
- The M24 features a large jog, shuttle control that can be used for audio scrubbing, shuttling and navigation. Audio Scrubbing is similar to rocking the reels back and forth on an analog tape recorder and is used during editing to help locate precise start or end points of a sound.

Zoom, Grouping and Mode Control

- The M24 has 10 switches that are used for mode select and include functions for Zoom, Grouping, Markers, Plug In mode select, Utility and Function modes and LCD and pan modes depending on host software. The M24 has input and output assignment capabilities as well.

Fader Flip Modes

- A SWAP switch below the Transport controls is used to swap the channel faders function to control the Aux Buss effect send levels for the currently selected send or the pan values if in Pan Flip mode. In send flip mode the pan controls can become send pan controls and send mutes are also available.

Bank and Channel Swap

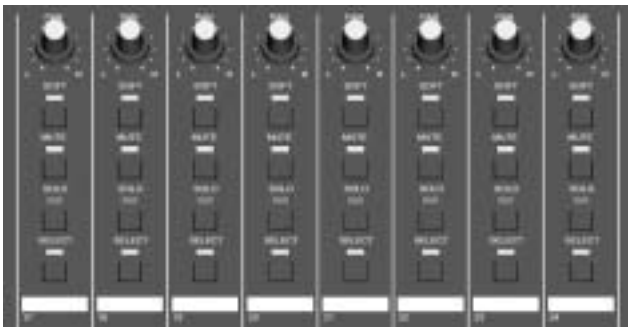
- You can swap the control assignments of the faders, pan controls, solos, mutes and soft controls in banks of 24 channels at a time using the BANK SWAP left, right arrow switches. If you are in the first bank the M24 faders are assigned to tracks 1-24. If you press the right BANK SWAP arrow switch once, the M24 faders would be reassigned to represent channels 25-48 in a song with over 24 tracks and the faders would move to match the new bank's on screen fader positions.
- Many of the supported software hosts can also shift the 24 assigned channels up or down one channel at a time. By pressing the BUSS 8 / CHANNEL switch and light the corresponding BUSS 8 / CHANNEL led, you will change the function of the BANK SWAP switches into channel shift switches.

LCD Display and Scribble Strip



The M24 features a 40 character by 4-line LCD display. The top row of the LCD display usually is used to display the most recent control and value including a long channel name, long control name and continuously updated control value. The next three lines of the display usually are used to display the abbreviated channel names for all 24 channels active in the current bank and act as channel scribble strips. Line 2 is for faders 1- 8, line 3 is 9- 16 and line 4 is 17- 24. The LCD has many additional modes and menus depending on the current host software. The P1 in the far right of the first row indicates the current page of controls that are active. The M24 has the ability to reassign just about every control on the unit instantly to other functions in several pages allowing the use of faders and other controls for very extensive control of plug ins and controllable instruments (software dependent) extending the total number of available controls.

Select, Solos, Mutes and Soft Switches



- The M24 has dedicated Channel Select, Solo, and Mute switches, 24 each that correspond to the same on screen channel functions in the host software depending on the current bank.
- The 24 Solo and Mute switches are always active, and can be easily reassigned to control additional functions.
- 24 Soft switches are assignable to different functions, such as record enable, depending on the host software. The Soft switches can also be assigned to automation arm, automation mode and punch control depending on the current mode of automation and the host software.
- The Channel Select switch is used to select the current active channel assigned to the remaining controls for effect sends (aux buses), dynamics, EQ and plug in controls.

Connections

- 3 MIDI in and 3 MIDI out connectors allow for multi program use or emulation of three eight channel controllers.
- A USB connector provides a high-speed connection to the host software.
- 25 pin accessory connector
- IEC320 Power connector for standard power cords

Weight and Construction

- Power supply: Two massive internal international switching power supplies (100-250V)
- Dimensions (WDH): 37.5 x 16.5 x 6″ (without armrest or wood trim accessories)
- Heavy Duty 16 gauge steel chassis
- Weight 38 lbs (without armrest or wood trim accessories)

SAC 2.2

Control Surface



The SAC-2.2 is a hardware control surface that allows you to control nearly all of the functions of your favorite software without using a mouse. It features 9 motorized faders, 12 rotary encoders with 31 segment LED rings, three large 2 x 40 LCDs, a jog wheel and a large locator display.
The SAC-2.2 is compatible with full-featured audio and MIDI recording software as well as virtual instruments and librarian/editor programs. In fact, the SAC-2.2 lets you control several different programs simultaneously, and switch between them in real time. The SAC-2.2 will automatically update itself with the correct operating mode and parameter settings as you switch between programs! And in addition, this feature is networkable, meaning it will allow you to control different programs on different computers on your network! Got a great Mac-based sequencer and an awesome PC-based sampler? No problem. One SAC-2.2 integrated into your network means you can control them both - at the same time, without the need to re-configure and re-boot!
The SAC-2.2 offers great fader movement, enhanced encoder LED visuals, USB support, and a very responsive and stable operating system. USB connectivity for both the Mac and PC greatly simplifies setup with Plug-n-Play. With its higher bandwidth, the SAC-2.2's fader performance when using USB is able to keep up with even the most complex automation moves. You'll love how responsive the button action is and how fast the LCD refreshes. And when connecting the SAC-2.2 to your computer using USB, the SAC-2.2's MIDI ports can be utilized as a built-in MIDI interface. Great for hooking up a desktop MIDI keyboard

The music tool which lets you concentrate on your mix and not your monitor. Works with every program that matters in pro audio software.

MOTU Digital Performer:
Superior audio and MIDI software for the professional musician. The SAC-2.2 gives complete access to every significant parameter within the software. An excellent plug-in driver done together with MOTU gives fast access to every essential nuance to create the best professional product. The SAC-2.2 and Digital Performer make a deadly combination.

Digidesign Pro Tools:
The definitive professional post production and audio software. The SAC-2.2 provides total access to the aspects of Pro Tools that not only save you hours of creative time, but allow you to perfect your product like nothing you have ever used.

Nuendo:
One of the future indicators of where professional music production software is going. Native and powerful, taking full advantage of today's power computers. The SAC-2.2 utilizes a control drive written by Steinberg giving complete access to the critical elements of recording and production.

Cubase:
This single program is the most used software in the world. Simple, professional and full of the kind of tools that you need to create the successful mix. The SAC-2.2 works hand in hand to automate and accelerate this complex task.

SAC 2.2

Samplitude 2496:
Professional software from Germany. Broadcast, post-production and mastering with some of the finest real-time plug-ins available today.

Logic Audio:
The software for the user who wants to create and work within their own complex environment. The SAC-2.2 is there to help you automate and control your dream setup.

Sound Diver:
The defining program for synthesizer editing without leaving your work area. A broad array of nearly every popular instrument of the past 20 years may be accessed directly from the SAC-2.2 through a driver written by Emagic. Put one touch access and parameters back at your fingertips the way it should be.

As well as...

- Sek'D Sequoia • Soundscape • Creamware Scope/Pulsar
- Propellers Reason • TC Works Spark • Ableton Live
- Merging Technologies Pyramix • EMU Ensoniq Paris system
- Cakewalk Sonar • Waveframe • Final Cut Pro 2
- Bias • Native Instruments

Virtual Instruments

- PPG Wave • Steinberg Model E • Pro Five/52
- TC Works Mercury One • Emagic EXS-24, ES-1
- Native Instruments B4
- Pulsar DES8i, EZ Synth, Inferno, Miniscope, BlueSynth, Vocoder, U-Know 007, Vectron Zarg Orion,

The goal of the SAC-2.2 is to make your work with both audio and MIDI tracks fast, effortless and far easier than you ever thought possible. Music is all about interaction and organization. The SAC-2.2 organizes every pertinent audio function needed to make a successful mix all within one touch. No searching and no mousing around. Simple, straight forward and designed to let you concentrate on your music as it was meant to be. All tracks are instantly accessible to edited together or singly with enough information in the displays that you will find yourself rarely having to look at the monitor.



FIRESTATION

8x8 FireWire Workstation For Mac and Windows



The FireStation is a 24-bit audio interface that provides eight simultaneous inputs and outputs to your favorite computer-based recording environment via a high speed FireWire connection. The FireStation utilizes Yamaha's open mLAN audio networking protocol that can support hundreds of channels of bi-directional digital audio, MIDI and word clock on a single FireWire bus. I/O features include two mic/instrument preamps, switchable between solid state or tube operation; eight 1/4˝ TRS line inputs two outputs; eight channels of ADAT optical I/O; S/PDIF I/O as well as control room and headphone outputs with level controls. Word clock I/O via BNC connectors are provided as well as a MIDI port for connecting synths and control surfaces etc... The FIREstation can also be used as a stand alone 8x2 mixer or A-to-D converter and is expandable to 48 Channels by daisy-chaining multiple units.

Two Dual-Servo Preamps with Switchable Tube Circuit

- Two Dual-Path Mic / Instrument preamps deliver maximum tonal flexibility with the choice of a solid state or 12AX7 tube enhanced audio path
- Preamp gain and the amount of signal routed to the vacuum tube are controlled independently with a dual concentric knob – greater levels of tube drive can deliver a desirable warmth or richer sound or distortion at extreme more settings
- XLR mic and 1/4˝ Hi-Z instrument are easily accessible from the front panel via Neutrik combo connectors
- 48 volt phantom power and 20dB pad switches are available for each channel
- Each preamp channel has it's own 1/4˝ TRS direct output send that allow an external processor, such as a compressor, to be placed in the signal chain and returned into the FIREstation via line inputs

Inputs and Outputs

- Eight balanced 1/4˝ TRS analog inputs and outputs
- The ADAT optical I/O supports eight channels of 24-bit audio, up to 48kHz to be transferred simultaneously in one direction
- Coaxial S/PDIF I/O

8x2 Line Mixer

- Each channel can be easily routed to the main outputs of the onboard line mixer. This provides zero-latency monitoring when used with a computer based DAW
- The mixer can be used whether connected to a computer or not
- The output level for the headphone and master outputs is controlled by a dual concentric volume knob

Dual mLAN/FireWire Interface

- The FIREstation can be networked to your computer and other digital devices via two standard IEEE-1394 FireWire interfaces that utilizes Yamaha's advanced mLAN network protocol
- mLAN (Music Local Area Network) an open platform that allows digital audio, MIDI and Word Clock information to be sent to and from multiple digital devices including multiple FIREstations, digital mixers and keyboards such as the Yamaha Motif or Korg Triton Studio

Compatibility

- The FIREstation works with any computer based audio system (Mac or PC) using ASIO drivers

MIDI I/O

- The onboard MIDI I/O port can be used for interfacing a keyboard or fader control surface for sequencing and mixing within your computer based DAW
- MMC (MIDI Machine Control) and MTC (MIDI Time Control) are also supported

Synchronization

- BNC Word Clock I/O allows the FIREStation to be configured as the slave or master device in a digital system
- The ADAT I/O can also provide or carry Word Clock as well.
- A 1/4˝ footswitch input, on the rear. allows you punch in/out recording

Specifications

Preamp Bandwidth	10Hz to 50kHz
Preamp Gain	70dB
A-to-D D-to-A Converters	24-bit
Supported Sample Rates	44.1kHz or 48kHz
A-to-D Dynamic Range	107db
D-to-A Dynamic Range	107db
Internal Word Clock Jitter	Ultra-Low
mLan Speed	200mbps
Internal Power Supply	

The IEEE-1394 FireWire standard offers the possibility of hundreds of channels of digital audio and midi on one network, easy connection of additional interfaces between products, master clock control bus without the need for word clock distribution



M-1000

10-Channel Digital Line Mixer



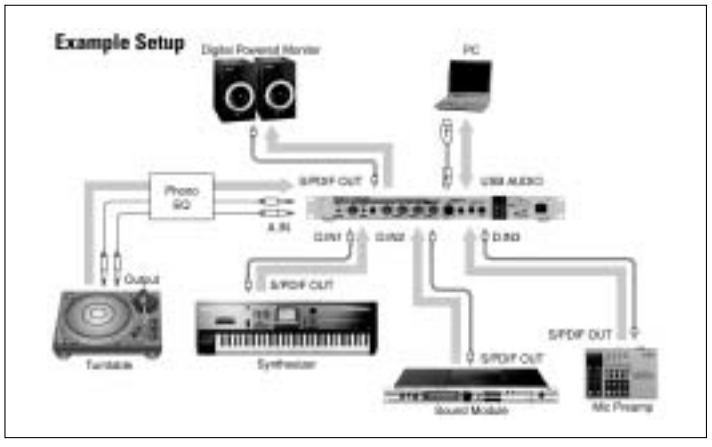
The M-1000 is a 1U rackmount mixer designed to mix digital signals of varying sample rates. The M-1000 comes with four stereo S/PDIF digital inputs and a stereo analog input—all with 24-bit/96kHz sound quality—making it perfect for mixing electronic musical instruments and recording gear equipped with digital outputs. The M-1000 can even mix audio from a PC via its USB port. Word Clock is also provided, and multiple units can be linked for greater mixing power.

FEATURES

Mix Multiple Digital Sources

Today's synthesizers, sound modules and recording gear often feature digital outputs, but musicians have never had a way to mix all these sources at once. That's where the M-1000 comes in. This 1U rackmount mixer can mix up to four stereo digital signals of varying sample rates—along with a stereo analog input and digital audio from a PC—with professional 24-bit/96kHz sound quality.

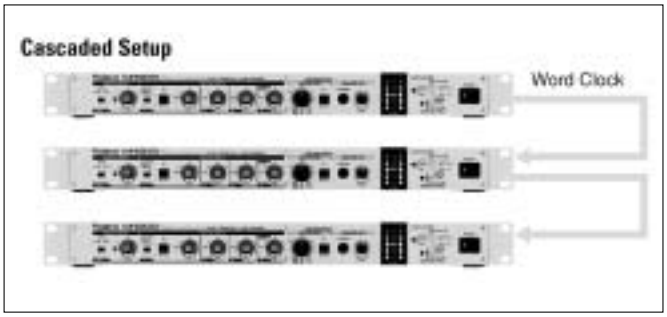
Take a look at the back panel, and you'll find four S/PDIF digital inputs and a stereo analog input with high-quality A/D converters. That's a total of 10 input channels that can be mixed—regardless of sample rate—using the front-panel knobs. The result can be output via coaxial and optical digital outputs or the M-1000's analog Master and Monitor outputs.



USB Port for PC-Based Recording

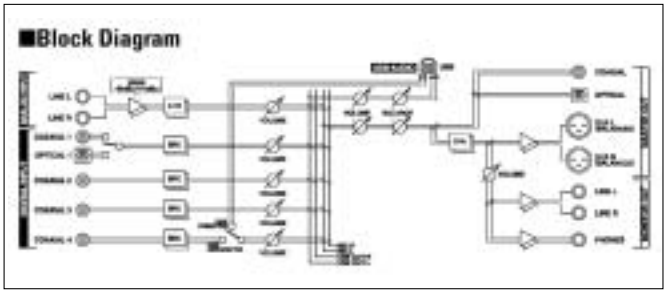
The M-1000 even lets you integrate your USB-equipped PC into the mix. Using a USB cable, you can route digital signals from a computer into the M-1000, where they can be mixed with other input channels. The results can then be sent back out to the computer for recording. The M-1000 supports sample rates up to 96kHz—even via USB*—so there's never been a better or easier solution for computer-based recording.

Flexible Sync Options and Expandable Design



With its Word Clock input and output, the M-1000 can be easily integrated into professional studios. The clock source itself is selectable among Internal, Word Clock, Digital In and USB* for maximum flexibility. And with its pro-quality specs—including pristine24-bit/96kHz AD/DA converters and 56-bit internal processing—the M-1000 can be used in a range of applications from home studio setups to world-class recording and editing suites. Best of all, multiple M-1000s can be cascaded for situations when additional inputs are required.

** At 96kHz, it is not possible to send audio in and out simultaneously via USB.*



SI-24 STUDIO PACKAGE

Studio Package Pro



The Studio Package Pro is a complete hardware and software bundle that gives musicians everything they need to professionally record music on a computer. The Studio Package Pro consists of three elements:

The SI-24 Studio Interface, RPC-1 PCI Interface Card, and Emagic's Logic RPC Pro software for Mac or PC. This affordable combination provides 24-bit/96kHz recording and professional MIDI sequencing with a motorized control surface—a powerful combination at an incredible value.

The SI-24 Studio Interface is a mixing surface equipped with eight XLR/TRS inputs, a Guitar Hi-Z input, plus digital I/O in S/PDIF and R-BUS formats—all with 24-bit/96kHz sound quality. The SI-24's 13 motorized faders, 12 rotary knobs and 5.1 surround joystick can all be used to control Logic's mixing functions and plug-ins. There's even a Screen Set function for changing Logic's screen sets remotely.

Audio is passed from the SI-24 to and from the computer via the included RPC-1 R-BUS Interface Card, a PCI audio card with drivers for Windows, Macintosh and ASIO 2. The SI-24 connects to the RPC-1 via a single R-BUS cable, which can handle eight channels of bi-directional 24-bit/96kHz digital audio, along with MIDI information received from the SI-24's MIDI In and Out ports—making the SI-24 a useful MIDI interface as well.

All recording is handled in the computer via the bundled Logic RPC Pro software. This special version of Emagic's award-winning music production software for Mac or PC provides eight channels of simultaneous 24-bit/96kHz recording, 24 tracks of playback, 5.1 surround mixing, level and pan automation, a host of killer plug-ins and more. And since the SI-24 is designed to control many of Logic's mixing functions—including track level and panning, EQ, plug-in parameters and soft synths—users gain hands-on control of their software for an unbeatable combination. The SI-24 also includes controller templates for popular software such as Pro Tools and Cubase, plus a "Learn" mode for setting up user presets more easily.

FEATURES

Professional Sound Quality

As with any digital recording system, sound quality is a primary concern. That's why Roland gave the SI-24 Studio Interface the best 24-bit/96kHz converters. You can use up to eight analog inputs simultaneously, choosing from either 1/4″ phone jacks or XLR connectors. Six analog outputs are also provided, plus S/PDIF digital I/O. Once audio is routed into the SI-24, it is transferred digitally in and out of the computer via R-BUS. And since Logic supports 24-bit/96kHz recording, the sound quality is never compromised. You can even dither back down to 16-bits when you're ready to burn a CD.

Fully Automated Mixing

With 13 motorized faders, mixing on the SI-24 couldn't be easier. Simply switch fader banks to access more channels and watch as the faders instantly snap to the proper positions! Naturally, the SI-24's faders can also be used to control Logic's "virtual"faders for more natural mixing. And when you really want to create complex mixes, tap into Logic's wealth of automation functions. Tools like HyperDraw let you draw in fader moves and panning, while the new track-based automation makes it even easier to automate any mixing or plug-in parameter without leaving the Arrange window.

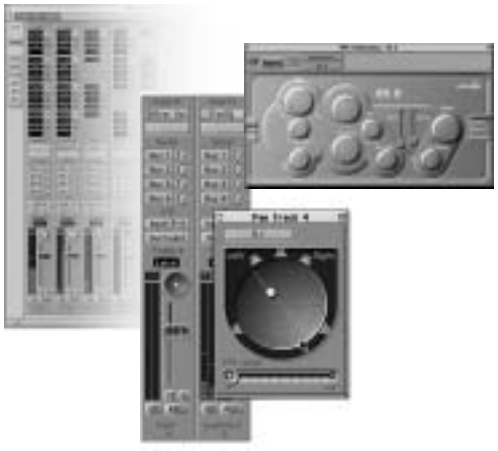
Transport/Function Buttons

Besides functioning as a mixer, the SI-24 works as a remote control for Logic. Dedicated transport buttons and a large Jog wheel make moving around your song easy, while ten numeric keys perform multiple functions: Push the Screen Set button and they automatically select Logic's screen sets, making it easy to switch between the Arrange window, Mixer window or other screens. Press the Short Cut button and the numeric keys automatically perform functions like Save, Undo, Copy, Paste and Cycle. They can even call up automation views for volume and pan. And that's just the beginning!

SI-24 STUDIO PACKAGE

Easy Editing with Real Faders and Knobs

Logic's mixer makes it easy to see levels, panning and effects—but who wants to mix with a mouse? Instead, try using the SI-24's faders, buttons and knobs for a more "hands on" approach. To adjust a track's level, for example, just grab one of the motorized faders and watch it change onscreen. And when you want to pan a track, simply select the desired channel and twist one of the Pan knobs. Similarly, you can use the SI-24's multi-function buttons and knobs to set EQ and send levels. There's even a dedicated Surround joystick so you can maximize the 5.1 surround capabilities of Logic RPC Pro.



RPC-1 R-BUS Interface Card

The RPC-1 is a cross-platform PCI audio card that serves as the gateway to your computer. Since all the A-D conversion is handled by the SI-24, the RPC-1 busses audio in and out of the computer using the R-BUS port. R-BUS is a multi-channel audio format that permits the simultaneous exchange of eight channels of 24-bit/96kHz digital audio, along with MIDI control information. The supplied R-BUS cable is all it takes to use the SI-24 with Logic RPC Pro.

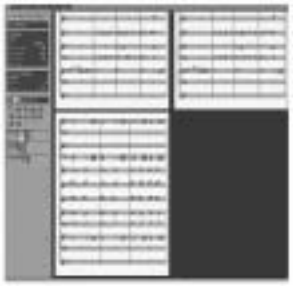
Logic RPC Pro

Based on Emagic Logic 5 Technology



Logic RPC Pro is a powerful sequencing and audio program based on Logic 5 technology—praised by professional musicians and producers the world over. You get high-resolution MIDI sequencing and 24 tracks of up to 24-bit/96kHz recording. Editing is ultra-intuitive using Logic's graphic editors and specialized tools. You also get access to award-winning Emagic plug-ins including effects and soft synths. This is serious software for serious producers.

Professional MIDI Scoring



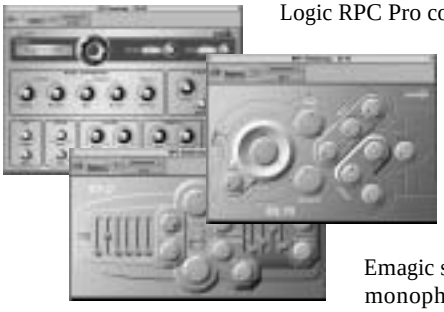
Using Logic's Score Editor, you can create professional score printouts quickly and easily. Simply record a MIDI performance and Logic RPC Pro will convert the data into clear and legible sheet music. Notation can then be edited onscreen with or without changing the original MIDI data. Logic also provides helpful tools to make your notation clearer, such as selectable note colors and instrument sets for separating individual parts from a score.

Intuitive Recording and Editing



Whether recording MIDI or audio, Logic RPC Pro makes it easy. The Arrange window is where all the action is, allowing you to build complex arrangements with simple and powerful tools. Detailed MIDI editing can be done in the Matrix (piano roll), List, and Score Editors, while audio can be non-destructively edited in the Sample Editor. In addition to all the standard audio editing functions, Logic RPC Pro also makes pitch and tempo editing available.

A World of Emagic and VST Plug-Ins



Logic RPC Pro comes with over 20 high-quality Emagic plug-ins. These include insert effects like dynamics and EQ, plus studio essentials including reverb, chorus and tape delay. Logic RPC Pro even comes with three Emagic soft synths: the ES M for monophonic synth sounds, the ES P for polyphonic sounds, and the ES E for ensemble sounds. Additional instruments like the EXS24 Sampler and EVP88 Piano can be purchased separately. And since Logic is VST-compatible, you can add optional VST plug-ins at any time for even more mixing power.

MMP-2



Mic Modeling Preamp

The Roland MMP-2 is a unique two-channel mic preamp with COSM Mic Modeling, four bands of fully parametric EQ, modeled tube compression, an enhancer and de-esser—plus ultra-intuitive control software for Mac or PC, superb-quality analog circuits, and 24-bit/96kHz A/D conversion. You can even download new effect plug-ins via USB, making the MMP-2 an indispensable tool for recording microphones and acoustic instruments. Uses the same carefully selected analog circuits and components as the mic preamps used in Roland's VS-2480.

COSM Mic Modeling and DSP Effects

- COSM technology models the sound of vintage and high-end studio microphones
- Four bands of fully parametric EQ
- Modeled tube compression, enhancer and de-esser effects are also provided

Built-In USB Port

- Connect to a Mac or PC and use the bundled control software to draw EQ curves or set compressors using a graphic interface
- Download effects such as noise suppressor, hum canceller, feedback eliminator and modeled tape saturation

Inputs and Outputs

- Analog I/O includes two balanced XLR / 1/4″ TRS combo mic inputs with phantom power and two XLR line outputs
- Coaxial digital input, plus coaxial S/PDIF and AES/EBU (XLR) digital outputs

4-band Fully Parametric EQ

- Low, Low-mid, Hi-mid and High EQ bands
- Filter types – Peaking, Low-Shelving, High-Shelving, Low-pass, Highpass, Low-pass2, Hi-pass2, Band-pass and Band-Eliminate
- Sweepable 20Hz to 20kHz frequency range with a ±15dB Boost/Cut and a variable Q of 0.36 to 16

Dynamics

- Three compression models to choose from: Solid, Tube1, Tube2. Each with variable Threshold (-24 to 0dB), Ratio (1:1 to ∞:1), Attack (0 to 800ms) and Release times (0 to 8000ms) as well as switchable Knee curves (soft/hard) and Level control (±24dB)
- AutoGain function automatically makes up for any level lost during compression
- Expander has a variable threshold (-80 to 0dBu), Ratio (1:1 to ∞:1 as well as Attack (0 to 800ms) and Release times (0 to 8000ms)
- Enhancer/De-esser has variable Sensitivity (0-100) and Frequency (1kHz to 10kHz)
- Adjustable enhancer mix level (-24 to +12dB)

TASCAM US-122

USB Audio/MIDI Interface

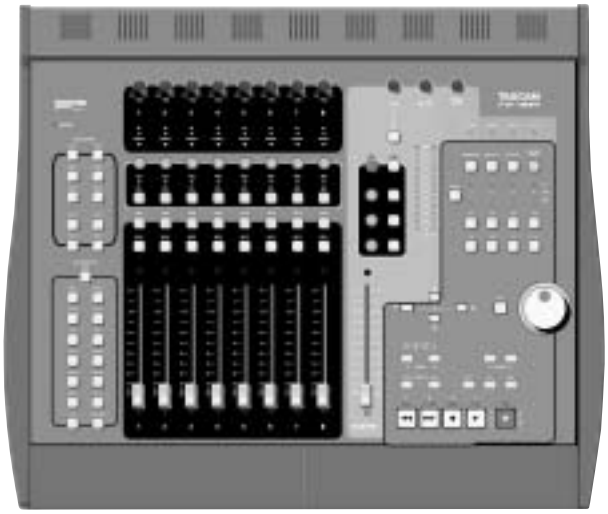


Combining affordability and ease-of-use with powerful capabilities and built-in compatibility with your computer, the US-122 is an excellent solution for several levels of computer recordists. With its compact, ergonomic design and rugged steel chassis, the US-122 meets the needs of everyone from hobbyist musicians through advanced professional recording engineers and producers. And it is bundled with Tascam's GigaStudio 24 sampling workstation software and a special version of Steinberg's Cubase recording/sequencing software, so you can begin creating and recording new music right out of the box.

- Two XLR-based, phantom-powered mic inputs with analog inserts that allow hardware devices (compressors, etc.) to be placed into the recording signal chain.
- Two line-level inputs on balanced 1/4″ TRS jacks are switchable to guitar-level for direct instrument recording.
- Adjustable zero-latency direct monitoring provides a sonically pure reference point before entering DAW processing.
- Two line-level outputs as well as headphone output with level control
- 16 channels of MIDI I/O for use with synthesizers and other MIDI equipment.
- Self-powered via USB making it perfect for location recording based around laptop computers as well as studio recording
- Mac and Windows compatible, ASIO, WDM, GSIF and Apple Core Audio (OS X) drivers turn any USB-equipped computer into a powerful audio recording system.

FW-1884

Professional Control Surface/ Firewire Audio-MIDI Interface



A professional DAW control surface and audio/MIDI interface equipped with high-bandwidth IEEE1394 (Firewire) capability, the FW-1884 provides audio and MIDI interfacing to computers as well as control of DAW parameters via eight 100mm motorized touch-sensitive channel faders, one master fader, and controls for pan, solo, mute and select functions on each channel. Tactile control for four bands of parametric EQ, a weighted jog wheel and a variety of shortcut keys for various popular software applications are also included.

The FW-1884s DAW control capabilities include comprehensive mixing, automation, editing and navigation tools. In addition to its surface control capabilities, the FW-1884 provides eight balanced XLR analog mic/line inputs with high quality mic preamps, phantom power and inserts on every channel. Using 24-bit/96kHz A/D and D/A converters, the FW-1884 offers full 96kHz operation on all analog I/O channels with compatible DAW software. In addition to its two IEEE1394 computer interfaces, the FW-1884 features eight channels of ADAT lightpipe as well as stereo S/PDIF inputs and outputs. Perfectly suited to multi-channel surround production as well as standard stereo mixing, the FW-1884 provides eight analog outputs for simultaneous connection to L/R and 5.1 surround matrices. A headphone output is also provided.

Four MIDI inputs and four MIDI outputs on the FW-1884 allow connection to a wide variety of MIDI sound generation and timing tools. Word Clock I/O allow computer systems to maintain accurate clock synchronization in digital recording environments. Also, its assignable footswitch jack can be used for tasks such as hands-free punching or MIDI control. In use, the FW-1884 allows up to 18 audio inputs to be accessed at once. As a DAW controller, the FW-1884 is compatible with a wide variety of popular applications including Digital Performer, Cubase, Nuendo, Sonar, and more.

For people who require simultaneous control of more audio channels, the optional FE-8 (eight-channel sidecar) expands the FW-1884 with eight additional channel control strips.

FEATURES

- Comprehensive mixing, automation, editing and navigation tools for DAWs
- Eight balanced XLR analog mic/line inputs with high quality mic preamps
- Phantom power and inserts on every channel
- 24-bit/96kHz A/D and D/A converters; full 96kHz operation on all analog I/O channels with compatible DAW software
- Eight channels of ADAT lightpipe, stereo S/PDIF inputs and outputs.
- Eight analog outputs, allowing connection of L/R and 5.1 surround matrices
- Eight 100mm motorized touch-sensitive channel faders, one master fader
- Dedicated controls for pan, solo, mute and select functions on each channel; tactile control for four bands of parametric EQ
- Shortcut keys for various popular audio software applications
- Dedicated headphone output
- Four MIDI inputs, four MIDI outputs
- Word Clock in and out jacks
- Assignable footswitch jack
- Up to 18 simultaneous inputs
- Compatible with a wide variety of popular DAW applications
- Optional FE-8 expands the FW-1884 with eight additional channel control strips (100mm motorized faders and dedicated control buttons). Multiple FE-8's may be added as needed for big console functionality with any DAW system.

TASCAM

US-428

Digital Audio Workstation Controller/USB Interface



The US-428 is a 24-bit digital audio workstation controller with the familiar interface of Tascam's Portastudio that professionals have come to recognize over the years. Essentially combining a hardware interface with computer functionality, the plug-and-play US-428 interfaces with Windows and Mac-based sequencing platforms via USB without any additional cards to be installed.

The US-428 offers full control over any number of software faders through eight hardware faders, groups of eight software faders selectable with buttons located beneath the unit's jog wheel. The control surface also supports transport, EQ and control functions for sequencers and DAWs. The US-428 supports inputs of 16- or 24-bit resolution at 44.1 or 48 kHz, and stereo or mono, with four analog inputs and a S/PDIF digital input. Two audio output channels also support 16- or 24-bit resolution in stereo or mono, and at 44.1 or 48 kHz. The US-428 also comes bundled with a custom version of Steinberg's Cubase VST recording/MIDI sequencing software.

FEATURES

Input Section

- There are four analog inputs and a stereo S/PDIF input, any four of which can be combined to record up to four simultaneous tracks via USB
- There are four Trim knobs with individual signal present and overload LEDs (one for each analog input) that ensure setting proper input levels before recording.

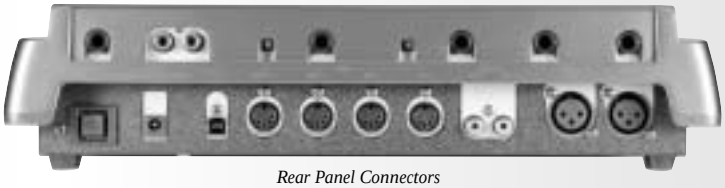
Channel Section

- Eight Channel Faders send continuous controller information to the current bank of eight faders delegated by the Bank select buttons. Each of the eight channels has its own Mute/Solo, record-ready and channel select switches with status LEDs.
- Master Fader controls the level going to the stereo bus output, and/or sends MIDI controller information to the host software.

The Master Section

- Dedicated Transport and Locate controls provide one-button access to the corresponding on-screen transport controls of your host application.
- Four Aux Send buttons with status LEDs access the corresponding Aux Send in your software and are controlled using the Datawheel
- The Datawheel also acts as a continuous controller for a variety of functions such as a shuttle wheel, moving the transport in the direction the wheel is turned.
- Dedicated EQ module controls the currently selected channel of your audio recording program and has continuous controllers for level, frequency and Q as well as selector buttons that access the four available EQ bands (High, Hi-Mid, Lo-Mid and Low)
- The Pan control allows left/right panning of the currently selected track
- The left and right Bank Select keys gives you unlimited access to successive banks of eight faders that control the corresponding channels in the software's internal mixer
- When changing to a different bank of faders, the Fader Null switch allows you to move the corresponding channel fader, using the Rec and Sel LED's as up/down guides until the US-428's fader matches the fader in the software's internal mixer.
- The Input Monitor switch allows you the Fader and Mute switches for channel's 1-4 to control the level of four analog inputs A,B,C and D to the US-428's stereo outputs. This is useful for monitoring the analog inputs with zero latency while recording.

- Two balanced XLR Mic/1/4″ TRS line inputs
- Two unbalanced 1/4″ inputs switchable from Mic/Line to HiZ for direct connection of a guitar or bass
- Coaxial S/PDIF digital I/O, US-428 to USB interface
- 1/4″ headphone output, stereo analog monitor outputs (RCA)
- Two independent 16 channel MIDI inputs and output



Rear Panel Connectors

US-428

US-428 Compatibility Reference

- Variable ASIO buffer size latency control
- ASIO 2.0 Direct Monitor ultra-low latency (1.5ms) monitoring with effects
- Sound Manager driver support (Mac) allows use as main sound controller for multimedia, video editing and games, or any editing program that rely on Sound Manager support including Pro Tools Free.
- Emulates Pro Tools CS-10 control protocol, allowing use as a control surface for any Pro Tools software (TDM and LE) and Pro Tools Free
- Cubase VST 5.0 or higher (Mac/Windows)
- Nuendo 1.5 - Mac/Windows
- eMagic Logic Audio 4.7 or higher
- Cakewalk Pro Audio 9.xx and Sonar1.xx
- Digital Performer v2.7 or higher - Mac
- Sonic Foundry Vegas, ACID and Sound Forge
- Syntrillium Cool Edit Pro
- Minnetonka MXTracks - Windows
- BIAS Deck and Peak
- Tascam GigaStudio
- Propellerheads Reason
- Most VSTi instruments
- Provides complete control of Native Instruments' B4 virtual organ software

US-224

Digital Audio Workstation Controller/USB Interface



Combining a hardware interface with computer functionality, the US-224 is a simplified version of the US-428 DAW controller and interfaces via USB. There are no additional interface cards to install, and it is compatible with applications that accept standard MIDI controller messages. Portable and incredibly affordable, the US-224 connects to your laptop for a complete studio you can take anywhere. The US-224's cool purple control surface gives you more than just mouseclicks for your DAW software. Like the US-428, the US-224 gives you real tactical controls like faders, transports, mutes and solos, panning and more. Plug in a guitar, mic or keyboard and get two channels of audio I/O in 16 or 24 bit quality, as well as 16 channels of MIDI I/O. The US-224 comes with a custom version of Steinberg's Cubase VST for Windows and Mac. With eight channels of audio and 16 MIDI tracks, the US-224 is a perfect all-in-one setup for anyone getting into the world of computer audio recording.

- The input section comprises a 24-bit audio (44.1 or 48 kHz) interface, capable of streaming two simultaneous tracks of audio into your computer.
- Inputs include two balanced XLR mic, two 1/4″ unbalanced line (switchable to high-impedance for direct input of a guitar, bass or other hi-Z source), and S/PDIF digital.
- Outputs include a pair of unbalanced line (RCA), S/PDIF and headphone
- Also has MIDI input and output ports, allowing you to send and receive data and MIDI Time Code (MTC) from your MIDI-based keyboards and other devices.
- The US-224 is powered via its USB connection to the computer with no external power supply required.
- The US-224 can also run in US-428 emulation mode offering users the ability to use software already supporting the US-428.
- The US-224's control surface includes a variety of controls which make working with audio software faster and easier. The US-224 allows control over any number of software faders via four hardware faders selectable in banks with buttons located beneath the US-224's jog wheel. Control surface also supports transport and control functions for sequencers and DAWs.

— The channel section of the control surface consists of four channel faders (plus one master fader), four MUTE/SOLO buttons, a switch to toggle Solo and Mute functions, and individual record-ready and select switches.

— The master section features a DATA wheel, PAN control (assigned by pressing the channel SELect key) and dedicated transport and locate controls, offering one-button access to these often-used functions.

TERRATEC

PHASE 26 USB

2-in/6-out USB Interface



The PHASE 26 USB is an external studio interface featuring TerraTec's tried-and-true audio technology in 24-bit, 96kHz resolution. This makes it the perfect mobile recording solution for all USB-equipped Windows notebooks or Apple PowerBooks. Featuring two analog inputs and six outputs, a MIDI interface, a microphone input, and an amplified headphones output, the PHASE 26 USB is a remarkably versatile audio system for musicians and producers. And with the popular and proven TerraTec Phono Preamp—designed to accept turntables with no further amplification required—it is also the first choice for DJs.

- Be it music productions, recording sessions, 5.1 surround productions, digital mastering, or DJing, the PHASE 26 USB offers peak performance and maximum mobility for all audio applications.
- An optical and a coaxial digital S/PDIF port, high-quality converters with resolutions ranging up to 24-bit/96kHz, and a dynamic range exceeding 106 dB ensure professional audio quality for mobile laptops and fixed desktop studios.
- Operates with or without power unit. It can also be switched on and off on the fly while the system is running. The computer detects the USB automatically, and its advanced software runs smoothly and flawlessly.
- Latency is extremely low, courtesy of generic drivers for Windows XP and Mac OS X. And that makes PHASE 26 USB an excellent choice for systems with real-time-enabled software synthesizers.
- Six (5.1) analog surround outputs
- Stereo input for line/studio devices
- Stereo input with preamp for turntables
- Mic input with Peak LED and Gain knob
- Headphones input with volume control
- S/PDIF optical and coaxial in/out
- 1x1 MIDI I/O
- Multi-platform (MAC + PC)
- Low-latency ASIO support



DMX 6FIRE 24/96

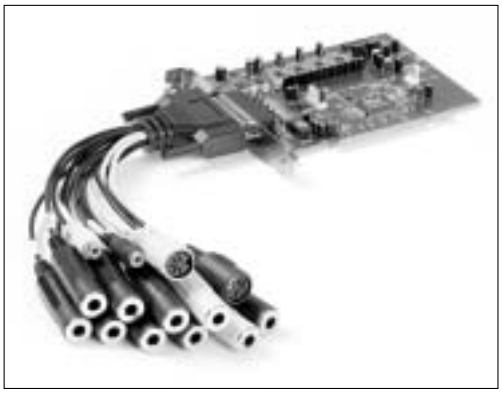
6-Channel Card for Music Recording, DVD, Games

Demanding users will experience an awesome sound experience, when using the DMX 6Fire24/96 Sound System: 6-channel audio reproduction in excellent 24-bit quality, advanced connection options on the computer's front side and full compatibility to audio standards of games, DVD entertainment and music. Uses only the best in sound technology: 24Bit/96kHz transducer components for all inputs and outputs guarantee a straight frequency response and a signal-to-noise ratio beyond-100db!

- Combines an EWX 24/96, Phono Preamp, MIDI, and an impressive microphone preamp in a single audio solution. Makes this audio interface the desire of any serious recording musician or hobbyist.
- For easy connections, the supplied 5-1⁄4″ front module I/O offers stereo inputs and outputs for analog and digital devices (optical and coaxial supporting 96kHz, AC3, and DTS) as well as MIDI instruments, headphone and microphone.
- Allows direct connection of tape deck and record player, while the supplied power pack of phono amplifier and professional software enable you to polish your oldest records with the finest digital sound!
- Gamers will be amazed when seeing the range of compatibility: Fast DirectSound, EAX, A3D, Sensaura 3D or MacroFX are just a few of the standards supported.
- Award-winning EWS technology with ASIO 2.0 and WDM drivers (2ms latency)
- Headphone output with volume level
- 16 channels of MIDI I/O
- Seamless connectivity with DVD players, AC3- and DTS reproduction via the digital interface, as well as full support of current Windows operating systems from 98SE to XP with WDM drivers
- Includes various applications from DVD player to audio editors and virtual sound studio— everything you can dream of.

PHASE 28

2-in/8-out 192-kHz Multi I/O Recording Interface



Offering a variety of ports, proven TerraTec technology, and transcendent audio performance on all Mac and PC systems, the PHASE 28 packs a powerful one/two punch - sophisticated studio features at a remarkably affordable price. PHASE 28 features two balanced 1/4″ analog inputs and eight balanced analog outputs, MIDI interface, and S/PDIF digital (coaxial) I/O designed for a variety of uses, on a specially shielded break-put cable. Inputs accept studio levels and signal resolution up to 24-bit/192 kHz. This, plus select converters and cleverly designed circuitry enable recordists to make crystal-clear stereo recordings and playback high-resolution 5.1 and 7.1 surround productions. Gain knobs serve to adjust input sensitivity individually to match analog signal sources. Two signal LEDs make it easy to dial in suitable levels for the premium quality preamp.

- 2 balanced analog 1/4″ inputs
- 8 balanced analog 1/4″ outputs
- 192kHz/24-bit resolution
- 1x1 MIDI interface for 16 channels
- Gain knobs and signal LEDs for all input channels
- Coaxial digital ports accept S/PDIF, AC3, and DTS data streams with all standard sampling rates up to 32-bits. What's more, the output is designed to handle sampling frequencies up to 192 kHz.
- Driver architecture ensures lowest latency ASIO support, top performance, and flawless compatibility with the audio platform of your choice. ASIO 2.0, GSIF, MME, WDM, Mac OS 9/X, or Windows XP, the PHASE 28's software integrates seamlessly into all prevalent systems and programs.

MIDI MASTER USB

Programmable USB-MIDI Keyboard



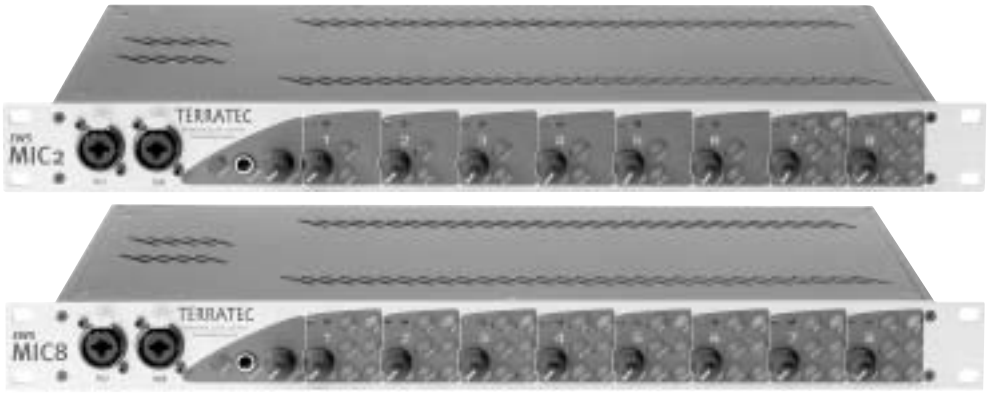
Making music on a PC or MAC is easier than ever before with MIDI MASTER USB—simply connect the keyboard with only one cable, turn it on and ...hit it!
MIDI MASTER USB features an extremely easy-to-play, touch-sensitive 49-key keyboard with pitch and modulation wheel, bright LED display, and ergonomically-placed control sliders and function keys. Additional features include MIDI controller commands freely assigned to controls, adjustable curves for touch sensitivity, MIDI reset and octave switches as well as numeric 10-key keypad.
Via USB, MIDI MASTER is directly available as input device, or as independent MIDI interface for connecting additional devices and accessed by every music software. And of course it can also be used without a computer as a pure MIDI keyboard. Last but not least, voltage supply can be selected from a power supply or USB, thereby saving an additional cable under your desk!

- 49 touch-sensitive keys with variable velocity
- Pitch-bend wheel and modulation wheel (controller assignable)
- Data entry slider (horizontal, MIDI controller assignable)
- USB MIDI-out interface
- 3 x 7 segment LED display
- Connector for optional sustain pedal
- 10 non-volatile memories
- 13 selectable velocity curves
- Integrated demo songs
- Powered via or optional 9v DC external power supply (9 V DC)
- Transpose function (transposing)
- Octave up/down button
- +/- keys
- 6 function keys
- 10 numeric keys
- USB MIDI driver for Windows 98, SE, ME, 2000 and XP and Mac OS 9 with OMS
- Includes 2 meter USB cable

TERRATEC

MIC2 & MIC8 SYSTEMS

8-Channel 24-bit /96kHz Digital Audio Interfaces



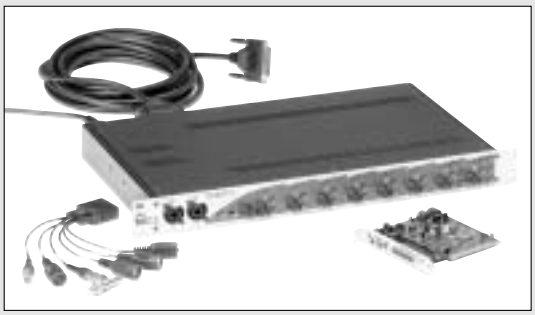
Incorporating state-of-the-art, versatile technology and built-in premium SPL quality microphone inputs, the MIC2 and MIC8 are advanced and powerful digital audio interface solutions for a host of professional studio applications. At the heart of the systems are the two (MIC2) and eight (MIC8) preamps developed in cooperation with SPL. All eight inputs on both devices can also accept line signals. The front panel offers two audio channels equipped with and XLR and 1/4˝ combo inputs. The many signal routing options enable recordists to configure their studio setups flexibly for the most diverse recording sessions without having to re-connect signal-carrying cords.

FEATURES

- 24-bit/96 kHz A/D and D/A converters
- Flexible control center for any studio. Besides analog I/O, the 19˝ (1RU high) housing also has ADAT and S/PDIF interface, connectors for MIDI and WordClock signals, and headphone output with levels.
- Two front-panel XLR balanced switchable mic and line/instrument inputs with 48v phantom power
- Eight analog inputs/outputs via switchable XLR and 1/4˝ jacks with +4/-10 dB settings
- Balanced I/O guarantee a low-noise signal management, allowing use as stand-alone preamps or audio mixers. All input channels can be adjusted at the rack module, and can be muted or directly forwarded to an output.
- 2 or 8 high quality mic preamps developed in cooperation with SPL of Germany
- 8 gain controls on front side for analog inputs (20 dB range)
- 8 input signal and clip LEDs on front side
- Direct out setting for all analog channels on front side (monitoring)
- Phase, low-cut and mute settings for each channel
- Integrated ADAT interface
- Toslink interface for ADAT or S/PDIF
- Up to 8 ADAT channels and audio channels can be mixed
- WordClock I/O
- High quality headphone DAC and amplifier for monitoring the mixed signal
- Internal power supply for 110/230v, 50Hz/60Hz operation
- The systems' open modular interfaces can connect to your computer via optional FireWire, mLAN or TerraTec's EWS88 32-bit PCI card (see box).
- Bundled, easy to use control software offers extensive routing and signal conversion possibilities. With a click of the mouse, analog, S/PDIF and even ADAT signals can be "wired" almost freely by the hardware and converted among each other.
- Sophisticated WDM driver for Windows 98SE, ME/2000 and XP, ASIO 2.0 and GSIF ensure smooth operation.

MIC2+ & MIC8+ SYSTEMS

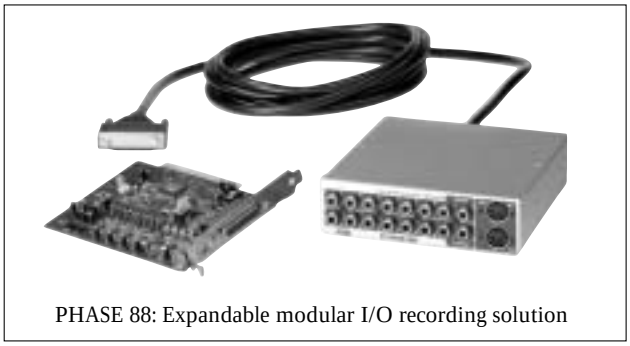
The Mic 2 + and Mic 8+ "Plus" versions add the EWS88 PCI card with a 4-meter cable to offer a professional, all-in-one computer interface/preamp/converter system. Otherwise, exactly identical to the Mic 2 and Mic 8, the "Plus" versions adds simultaneous recording and playback of all channels with up to 24-bit/96kHz, hardware mixer with internal 36-bit resolution, EWS-Connect, 2 MIDI ports, and a VU-meter in the control panel.



PHASE 88 / PHASE 88 RACK

Modular I/O and Multi I/O Recording Solutions

The PHASE 88 is a professional audio and MIDI interface that transforms any Mac or PC into a professional recording system. Eight high-quality analog input and output converters process audio signals (from start to finish and from hardware to software) with up to 24-bit/96kHz resolution. The PHASE 88's converters are housed in a 5¼˝ module that can be installed in or placed outside the computer. An additional S/PDIF (coaxial) digital input/output on the PCI bus card provides the perfect interface to the digital studio environment.



PHASE 88: Expandable modular I/O recording solution



PHASE 88 Rack: Multi I/O recording solution

PHASE 88 RACK offers all the ports required for a music production system. The modular system can be extended to 40 physical inputs and outputs. Based on the PHASE 88 recording interface, the rack version offers eight analog inputs and outputs with 24-bit/96-kHz resolution in an external 19˝ rack module. All inputs and outputs are balanced for noise-free signal routing; 1/4˝ jacks are mounted on the rear panel. In addition to the (coaxial) S/PDIF digital interface and two separate MIDI inputs/outputs, PHASE 88 RACK features two balanced microphone inputs on the front panel.

They Both Feature

- Based on proven TerraTec technology, the PHASE 88 interface features optimized conversion components and cleverly designed circuits, enabling recordists to cut crystal-clear recordings on up to eight tracks simultaneously and even play back high-resolution 5.1 or 7.1 surround sound.
- To make the most of the 24-bit converter's excellent dynamic range, they are equipped with an on-board 20-channel hardware mixer with 36-bit internal resolution and 20-in-4 routing. Perfect solution for studios that do without an external mixer.
- Every input port features a dedicated analog stage that can be switched from -10 dBv to +4dBv, as well as a variable amp with up to 18 dB gains. Channel thru monitoring generates zero latency, and latency introduced by the ASIO 2.0 driver comes to less than three ms.
- Drivers for Windows 98/ME, 2000, XP (ASIO 2.0, DirectSound, MME with Multi-Client mode, and ASIO GSIF Multi-Client), Mac OS 9.1, and OSX (ASIO 2.0).

Phase 88

Up to four PHASE 88 recording systems can be synchronized and cascaded within a single computer. This means that systems with up to 40 physical inputs and outputs and 64 MIDI channels can be configured without adding to the CPU load.

Phase 88 RACK

Up to four PHASE 88 RACK systems can be synchronized and cascaded within a single computer. This means that the system can be extended to 40 physical inputs and outputs and eight MIDI I/Os with 128 MIDI channels without adding to the CPU load.

PHONO PREAMP STUDIO USB



Barely the size of a deck of cards, phono PreAmp Studio USB lets you professionally edit records and cassettes on your computer at a level of quality and then immortalize them on CD via a USB port—without sound card!

- For all turntables with moving magnet (MM) cartridges
- External housing makes opening up the computer unnecessary. Simply connect the record player to the PC through the phono PreAmp. That's it! No extra drivers or even a power outlet required: power is taken directly from the USB port.
- Adjustable three-stage input capacity
- Precision RIAA curve equalization for true-to-life playback
- Bundled Sound Rescue software lets you dramatically reduce or even get rid of unwanted noise such as crackling and static in real-time while maintaining the original, authentic sound quality.

YAMAHA

01X

mLAN Music Production System



The Yamaha 01X delivers power and flexibility never before offered in any computer audio product and represents a dramatic breakthrough in total integration and flexibility. No more expensive interfaces and control surfaces, the 01X combines both functions in one single unit with mLAN compatibility delivering all MIDI and audio data via a single cable. Featuring Yamaha's next-generation mLAN technology, the 01X guarantees high-speed audio and MIDI networking between computers, synthesizers and other mLAN compatible devices. Connecting the 01X is simplicity itself thanks to the FireWire (IEEE1394) interface. Featuring low latency audio interfacing over mLAN for both Windows and Mac-based computers the 01X delivers remote control power never before offered in a multi-function device.

FEATURES

Total Control

- The 01X offers remote control power never before seen in a multi-function product of this type. With a level of integrated DAW support normally found only in dedicated remote controls costing as much as 01X itself, the 01X can also be the ultimate human user interface between you and your software.
- Functions which can be controlled directly from the 01X include track arming, transport control, plug in editing, mixing, window selection and many more using existing and popular control protocols. This means you have support straight out of the box for Cubase SX, Nuendo, Sonar, Logic, Digital Performer and many more software products besides.
- As well as being the perfect remote control partner for your sequencer or digital audio workstation software, 01X will also control a wide range of parameters with many of your software synthesizers and as the 01X also works as a multi-port MIDI interface, your existing hardware MIDI devices can be connected directly to the 01X, making it truly the heart of your working environment.

Total Mixing Power

- Built on world-renowned 96kHz DSP technologies, found in Yamaha's flagship DM2000, 02R96 and 01V96 digital mixers, the 01X not only works as a professional digital mixing interface for your computer but also as a total recall, stand-alone digital mixer, ideally suited for live performances.
- High quality mic pre-amps, 48V phantom power on balanced XLR and TRS, line inputs let you plug in guitars, microphones or indeed any musical instrument and know that the quality of your signal is maintained throughout.

DSP

- Full dynamics processing, including compressors, gates and limiters as well as 4 band parametric equalization on every one of the 28 digital mixing channels, and two world-class 32-bit effects processors means that the 01X puts no strain on your host computer for mixing or audio processing. Fully motorized faders mean that your mix is recalled at the touch of a button, and is accurately tracked during the course of your song.

Total Network

- Yamaha's 01X is the first product to feature Yamaha's next generation mLAN technology. Offering high speed audio and MIDI networking between computers, synthesizers and other mLAN compatible products, the 01X offers more than any previous computer music product has ever delivered.
- As Yamaha's mLAN technology is based on the industry standard IEEE1394 (FireWire) buss, connecting up your 01X is as simple as plugging in a single cable. Perfect for any environment, the 01X is as happy connected to a laptop as to a desktop computer.
- mLAN being a true network, and not just a point to point connection system, allows for up to 63 mLAN compatible devices to be linked together.
- And with mLAN using a single cable to send multichannel audio and MIDI, you need never have to worry about opening up your computer, fiddling with messy IRQ's or wrestling with awkward set-up configurations. 01X means you just plug in and play.

B&H PHOTO - VIDEO - PRO AUDIO EQUIPMENT LEASING AVAILABLE

01X

Total Expansion

- Based on mLAN, the 01X can be expanded to add more channels with optional mLAN channel expansion units, meaning that the system can grow with your needs.
- Up to 16 extra analog inputs can be added to over mLAN with the 01X internal digital processors and mixing channels handling the routing/effects/EQ and dynamics.



Inputs and Outputs

- Eight analog inputs including: two balanced XLR mic pres with switchable phantom power; and six 1/4″ TRS balanced line inputs. Input eight also features a dedicated Hi-Z 1/4″ instrument input.

TOTAL SOFTWARE SUPPORT

Unlike other solutions on the market, there is no need to switch to one specific application just to get the full benefit of 01X. With low latency 24-bit/96kHz audio driver support for Windows XP and Macintosh (OS9/OSX) computers as well as full MIDI capability, the 01X slip's into your existing working environment with absolute ease.

Bundled with the 01X is a professional suite of user interface control software and commercially available VST effects plug-ins to enhance even further your experience of working with the most powerful computer audio product ever conceived. While 01X itself has one of the most intuitive user interfaces yet designed, using 01X in conjunction with Yamaha's acclaimed Studio Manager, you can also instantly see and control every parameter directly from a computer screen. On top of this 01X comes with a VST compatible 01X channel module which mimics the hardware DSP and functions of the console, and lets you copy parameters from Studio Manager directly into your VST hosting application, offering a level of integration never before thought possible.

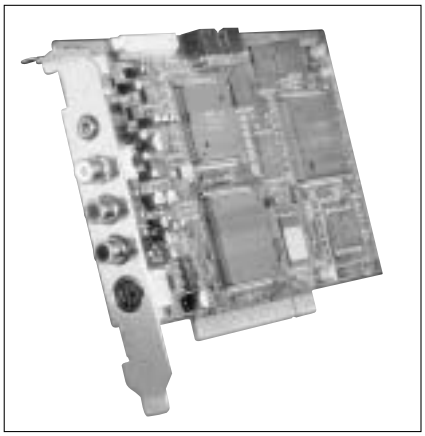
Also with 01X are three world-class VST plug-ins offering enhanced vocal processing, final mastering and studio quality, formant-accurate pitch correction.

SW1000XG

PCI XG/Hard Disk Recording/Effects Processing Card

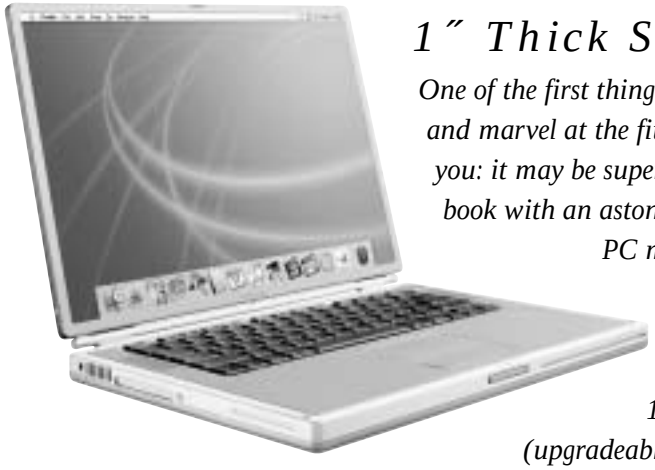
The SW1000XG is an audio production card with over 1,200 instruments including an incredible 46 drum kits. It features 12 mono (or six stereo) playback channels and multiple effects buses. And it can be set up so that its effects are available to both audio and MIDI tracks within your sequencing software. Additionally, effects can be applied to the output of a software synth package, or to a device that's connected to the card's line input. Any combination of these set-ups is also possible.

There are over 70 fully-programmable effects, which are configured as a mixed bag of insert and auxiliary processors. Up to six of them can be used in parallel (these include a reverb, chorus, variation, two inserts and a master parametric EQ) and effect types can be switched on-the-fly over MIDI—and all this without touching one drop of your computer's main CPU, leaving it free to handle even more audio tracks or additional software-based effects.



- 20 MB wave ROM with over1000 PCM samples
- 7 independent 24-bit effects sections including 12 Reverb, 14 Chorus, 70 Variation, 43 Insertion 1 and 43 Insertion 2 effects, 5-band multi EQs on the master stage, and channelized EQ with hi/low and gain on every single part. Optional PLG100VH adds another 4 unique "VH Insertion" effects for instant harmonies and "Gender Control".
- More than 1000 normal voices and more than 45 drum kits (kits... not sounds, including the way cool 'Coffin kit' and a host of new analog drum samples).
- Real SVA physical modeling synthesis in hardware so no latency or CPU overhead
- Two A/D inputs for independent analog effects processing, stereo analog outputs, S/PDIF, plus 48 channel MIDI support.
- S/PDIF digital connector.
- Tone Generation : AWM2 + optional PLG100VL (Virtual Acoustic Synthesis)
- Sound Module Mode : XG (including VL-XG if fitted), TG300B (GS), C/M (MT32) all fully programmable.
- Polyphony : 64 notes + 1 (VL) (if fitted)
- Multi-timbrel capacity : 32 parts internal, 16 part external (48 part total)

POWERBOOK G4



1″ Thick Studio that Travel With You

One of the first things you notice about the PowerBook G4— after you admire the looks, and marvel at the fit and finish — is the number of ports it has. And that's when it hits you: it may be superslim and ultralight, but it also happens to be a full-featured notebook with an astonishing lineup of built-in professional capabilities. And unlike some PC manufacturers who ship lightweight PC notebooks that shed pounds by shedding functionality, Apple makes no compromises. Available in three sizes, PowerBook G4 models feature optical drives — the legendary SuperDrive (DVD-R/CD-RW) on the 15- and 17″ models, and the DVD-ROM/CD-RW Combo drive (upgradeable to the SuperDrive) on the 12″ model.

- All feature FireWire and USB, with an additional FireWire 800 port on the 17″
- All feature S-Video output for connecting to a TV or projector
- Connect the PowerBook G4 to a PC network or to a Windows PC (and to most USB printers and scanners, as well as Firewire digital cameras, camcorders and external drives). Share .doc, pdf, jpg, gif, tiff, MPEG, Windows Media, QuickTime, MP3, MIDI and CD files with PC users.
- 56K modem is standard, while Bluetooth (wireless connectivity) is built into the 12- and 17″ models and is available as an option for the 15″ model.
- The 15- and 17″ come with Gigabit Ethernet (10/100BASE-T Fast Ethernet on the 12″ model).
- Supplied lithium-ion battery powers them for up to 5 hours depending on system configuration and usage.
- Housed in a lightweight and durable aluminum alloy enclosure, the 12- and 17″ are resistant to stains and scratches. They also have a drop-dead gorgeous design — perfectly smooth on all surfaces, with no doors, protruding latches or levers to break, no external buttons to accidentally press, and no sharp edges to catch on your clothes.
- The 15-inch is encased in titanium, the exceptionally strong and exceptionally light metal used in supersonic aircraft engines.

POWERBOOK CONFIGURATIONS

Order no.	M8760LL/A	M8858LL/A	M8859LL/A	M8793LL/A
Processor	867MHz PowerPC G4	867MHz PowerPC G4	1GHz PowerPC G4	1GHz PowerPC G4
Cache	256K SRAM Level 2 cache	256K SRAM Level 2 cache, 1MB DDR SRAM Level 3 cache		
Memory	256MB PC2100 DDR SDRAM	256MB PC133 SDRAM	512MB PC133 SDRAM	512MB PC2700 DDR SDRAM
Display (Diagonal)	12.1″, 1024x768, TFT XGA	15.2″, 1280x854, TFT	15.2″, 1280x854, TFT	17″, 1440x900, TFT widescreen
Graphics	NVIDIA GeForce4 420 Go with 32MB of DDR SDRAM	ATI Mobility Radeon 9000 with 32MB of DDR SDRAM	ATI Mobility Radeon 9000 with 64MB of DDR SDRAM	NVIDIA GeForce4 440 Go with 64MB of DDR SDRAM
Hard Disk Drive	40GB Ultra ATA/100	40GB Ultra ATA/66	60GB Ultra ATA/66	60GB Ultra ATA/100
Optical Drive	Slot-loading Combo (DVD-ROM/CD-RW)	Slot-loading Combo (DVD-ROM/CD-RW)	Slot-loading SuperDrive (DVD-R/CD-RW)	Slot-loading SuperDrive (DVD-R/CD-RW)
Ethernet	Built-in 10/100BASE-T	Built-in 10/100/1000BASE-T (Gigabit)		
Modem	Built-in 56K V.92 modem	Built-in 56K V.92 modem	Built-in 56K V.92 modem	Built-in 56K V.92 modem
Wireless Networking	Built-in Bluetooth 1.1; AirPort Extreme ready	Optional Bluetooth 1.1 adapter; AirPort ready	Optional Bluetooth 1.1 adapter; built-in AirPort Card	Built-in Bluetooth 1.1; built-in AirPort Extreme card
System Software	Mac OS X v10.2 Jaguar	Mac OS X v10.2 Jaguar	Mac OS X v10.2 Jaguar	Mac OS X v10.2 Jaguar
Software	QuickTime, iCal, iChat, iMovie, iPhoto, iTunes, iDVD (requires SuperDrive), DVD Player, Mac OS X Mail, Microsoft Internet Explorer, EarthLink (includes 30 days of free service), Acrobat Reader, Art Directors Toolkit, FAXstf, FileMaker Pro Trial, GraphicConverter, Microsoft Office v. X Test Drive (12-inch and 17-inch models), OmniGraffle, OmniOutliner, QuickBooks for Mac New User Edition, Developer Tools, Apple Hardware Test			
Video Accessories	Apple Video Adapter and Apple VGA Display Adapter	Apple DVI to VGA Adapter and S-Video to Composite Adapter		
Hardware Accessories	Modem cable, power adapter, AC wall plug, power cord.			



1GHz 17-inch iMac

The 17″ iMac dazzles with its brilliant 17″ widescreen flat-panel display that glides through the air, allowing you to adjust its height, depth or angle with just a touch of your finger. But what's more amazing is its compact base which packs a powerful 1GHz PowerPC G4 processor with Velocity Engine, 256MB of PC2100 (266MHz) Double Data Rate (DDR) main memory (expandable to 1GB), GeForce4 MX graphics with 64MB of DDR graphics memory and an Ultra-ATA/100 80GB hard disk spinning at 7200 rpm. You also get AirPort Extreme and Bluetooth wireless capability, and the best-of-breed digital lifestyle applications that have convinced so many PC users to switch to the Mac platform.

FEATURES

iLife—it all connects

Bundled with the iMac is a host of iLife software applications including iTunes, iPhoto, iMovie and iDVD. They let you be creative with your pictures, music and movies– and share them–in ways that PC users can only dream about. And whether you're creating your own music CDs, editing digital video, or authoring DVDs, you can do all without thumbing through a manual.

SuperDrive

The 17″ iMac is equipped with the SuperDrive (DVD-R/CD-RW) allowing you to burn audio and video on a DVD disc that plays in almost any standard DVD player. Even better, the SuperDrive burns DVDs at 4x speed, twice as fast as before.

54 Mbps AirPort Extreme

Say goodbye to having to set up your computer near a phone. Pop the AirPort Extreme Card into the 17-inch iMac, and you're ready for wireless networking from anywhere in your home or dorm — up to 150-feet from an AirPort Extreme Base Station.

Sync with Bluetooth

The 17-inch iMac can be custom-configured with an internal Bluetooth module to connect your digital devices wirelessly. Operating within a 30-foot radius, it enables wireless data transfers between a rapidly growing list of Bluetooth-enabled products. Using iSync, for instance, you can use Bluetooth to synchronize your personal information between your iMac, your mobile phone and your Palm OS-based handheld.

Extreme Graphics

The 17″ iMac has an NVIDIA GeForce4 MX graphics processor with 64MB of DDR RAM. The NVIDIA GeForce4 MX delivers 1.1 billion textured pixels per second. And since the graphics processors feature dedicated RAM — unlike some PC graphics cards that share main memory — they take over the transform and lighting calculation functions from the iMac's central processing unit (CPU), freeing the PowerPC G4 to perform essential system tasks faster than ever before.

Space-Saving Design

When developing the iMac, Apple used a rigorous design approach that optimized every available bit of space—and no wasted corners. Starting with a custom-designed circular logic board that fills the entire diameter of the base, Apple engineered enough space to fit in the full-height SuperDrive. On top of the optical drive are a high-capacity 3.5″ hard disk drive and even the power supply — a split design that tucks neatly into the top of the base.

Speakers and Audio

Includes Apple Pro Speakers that handle 18 watts of undistorted digital sound. And since the iMac also provides power to the speakers, you don't need an additional power brick. The result? Reduced cable clutter. Features an internal speaker, internal tripath digital amplifier, Apple Pro Speaker jack, headphone jack, and an audio line-in jack that lets you use it as a sound editing studio.

iBook

The Most Affordable Mac Notebook Ever

Perfect for everything from homework to playing games, and from watching DVD movies to burning your own music CDs. With its extra long battery life, compact size and ultralight weight, the 4.9-pound iBook can accompany you everywhere. And with a choice of 12.1″ or 14.1″ displays, 800MHz or 900MHz PowerPC G3 processors, 30GB or 40GB hard drives, 128MB or 256MB of RAM expandable to 640MB — plus optional AirPort (802.11b) wireless networking and FireWire, USB and Ethernet ports — the iBook gives you world-class performance in a slim, stunning design.



POWER MAC G5



Computers for Music Creation

The world's fastest personal computers and the first with a 64-bit processor — which means they break the 4 GB barrier and can use up to 8 GB of main memory, Apples new G5 line combines rock solid engineering reflective of the full-throttle Xserve architecture with new technologies for massively enhanced output and connectivity. Available at speeds up to dual 2GHz with a new ultra-high bandwidth system architecture featuring AGP 8X and PCI-X these turbocharged Power Mac rips through digital video and audio projects faster than Pentiums can say "uncle".

Jointly developed by Apple and IBM, the 64-bit PowerPC G5 processor supports up to 8GB of 400MHz, 128-bit DDR SDRAM — four times more than a typical PC. More main memory and fast 6.4 GBps throughput means you can write large projects to memory 40x faster than to a hard disk.

Available in three powerful configurations — 1.6GHz, 1.8GHz and a dual 2GHz model, they each feature the legendary 4X SuperDrive (DVD-R/CD-RW), three PCI-X slots, one FireWire 800 and two FireWire 400 ports, three USB 2.0 ports (plus two USB 1.1 ports on the keyboard), ADC connector, DVI connector, analog audio I/O, optical S/PDIF I/O, front headphone and speaker jack and built-in Gigabit Ethernet — as standard. In addition,all are AirPort Extreme-ready, and offer Bluetooth capability as a build-to-order option. Finally, each Power Mac G5 is supported by Mac OS X's robust foundation, which features multithreading and symmetric multiprocessing for an additional performance boost.

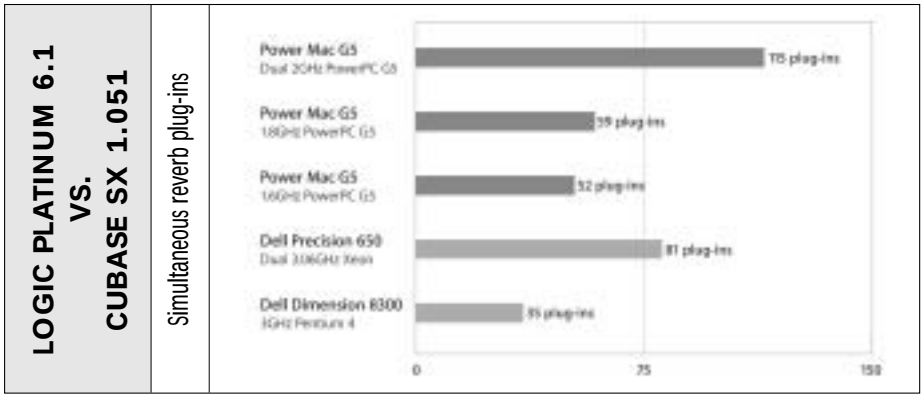
FEATURES

Exceptional Performance

- The latest in PCI technology, the PCI-X protocol is perfect for high-performance PCI devices, increasing speeds from 33MHz to 133MHz and throughput from 266MBps to 2GBps.
- Gigabit Ethernet, FireWire, USB 2.0 and optical digital and analog audio are all integrated through two bidirectional 16-bit, 800MHz HyperTransport interconnects for a maximum throughput of 3.2GB per second.
- Ultra-high bandwidth system architecture features a 1GHz frontside bus — one on each processor — for maximum throughput. And a point-to-point system controller lets data move directly between subsystems, without affecting processor function.
- Designed for whisper-quiet operation, the Power Mac G5's enclosure houses four discrete thermal zones to compartmentalize the primary heat-producing components. Fans in the zones spin at very low speeds resulting in a system three times quieter than the Power Mac G4.

Music and Audio

The Power Mac G5 supports more tracks and more plug-ins than any previous Power Mac—giving the audio pro a wealth of creative options all in a native environment. In addition, Mac OS X Core Audio (see box on next page) offers a scalable platform supporting 32-bit high resolution audio, a plug-in called Audio Units for DSP and Virtual Instruments, and plug-and-play connectivity for modern and legacy audio gear. And the built-in optical S/PDIF connects to other audio equipment for pristine sound quality without troublesome ground loops.



The dual 2GHz Power Mac G5 can play 115 tracks, compared with a maximum of 35 tracks on the Dell Dimension 8300 and 81 tracks on the Dell Precision 650. More impressively, the 1.6GHz single-processor Power Mac G5 played 50 percent more tracks than the 3GHz Pentium 4–based system.

64-bit Performance

The PowerPC architecture was designed from the beginning to run both 32-bit and 64-bit application code, offering seamless transition to 64-bit performance. Current 32-bit code — such as Mac OS X, the Mac OS 9 Classic environment and existing applications — runs natively at processor speed. With no interruptions to your workflow. And no additional investment in software required.

Graphics and Display Support

- Compared with the AGP 4X interface, the Power Mac G5's AGP 8X Pro graphics bus effectively doubles the maximum transfer rate and doubles the amount of data transferred in a single AGP bus cycle. The 66MHz AGP 8X Pro bus strobes eight times per clock cycle, achieving a 533MHz data rate and a maximum bandwidth of 2.1GB per second — ideal for the ultrafast ATI Radeon 9800 Pro or NVIDIA GeForce FX 5200 Ultra graphics cards they include.
- Supports 1920 x 1200 pixel digital resolutions and 1600 x 1200 analog resolutions
- ADC and DVI connectors; DVI to VGA adapter included
- Dual display support for extended desktop and video mirroring modes
- Support for two Apple flat panel displays

Storage and Expansion

- Two Serial ATA controllers supporting up to 150-MBps data throughput per hard drive
- Two 3.5-inch hard drive expansion bays with drive guides for a second drive

– One 80GB or 160GB 7200-rpm Serial ATA drive installed in standard configurations

– Support for up to two internal Serial ATA drives; 500GB maximum system capacity

–8MB memory buffers on all hard drives

- Optical drive bay with SuperDrive (DVD-R/CD-RW) installed; writes DVD-R discs at up to 4x, reads DVDs at up to 8x, writes CD-R discs at up to 16x and CD-RW discs at up to 10x, reads CDs at up to 32x speed
- One of the following PCI expansion configurations: Three open full-length 64-bit, 33MHz PCI slots, or one open full-length 64-bit, 133MHz PCI-X slot and two open full-length 64-bit 100MHz PCI-X slots.

Mac OS X Core Audio

Musicians, audio pros and music educators have long recognized the Mac as the superior platform for creativity. Historically, third party developers augmented the native capabilities of Macintosh with a wide variety of hardware, software and system add-ons, making it possible to use the Mac as a virtual recording studio. However, configuring such a studio became quite complex, and didn't allow for the highest level performance possible. With Mac OS X, professional-level audio is here. Mac OS X Core Audio integrates audio functions directly into the operating system in ways never before possible, enabling unprecedented performance and ease of use in your virtual studio. Core Audio lays a new foundation for the next generation of world-class audio and music applications from innovative developers.

Performance

Mac OS X delivers the best audio performance in desktop computing. The most fundamental measure of audio performance is throughput latency. That's the time it takes for audio to enter your Mac, travel through the system to your application and then pass back out to your monitoring system (speakers). Historically, Mac OS offered excellent latency of about 10ms. The Core Audio HAL (Hardware Abstraction Layer) provides ultra low latency of only 1m — and you get this performance in a full multi-channel environment.

Stereophonic Multi-Channel

Previously you had to install third-party system extensions to work with more than a 2-channel stereo mix. Now with Mac OS X offering native support for multiple channels, you can simultaneously record with more than two microphones, and perform large-scale professional tape transfer in a single pass, with greatly improved performance. The Core Audio HAL also allows multiple applications to share the same device. This means that you could assign channels 1-6 of a eight channel output device to a multi-track program, such as Logic, while leaving a virtual instrument like Reason with channels seven and eight.

High Fidelity

Core Audio manages all audio as 32-bit floating-point data. This allows your Mac to efficiently handle 24-bit/96 kHz audio as well as higher resolutions. Core Audio also delivers highly optimized sample rate converters to allow programs that do not yet use this high-resolution format to provide data to Mac OS X without truncation.

Audio MIDI Setup

Mac OS X has an easy to use setup application called Audio MIDI Setup. This provides applications with the ability to manage MIDI and define a system-wide MIDI configuration that is available to all of your applications. In addition, Mac OS X provides applications with Music Services, which are the fundamental functions of MIDI sequencers including cut, copy, paste, repeat and other common MIDI editing routines.

Reverb-erb-erb

In the absence of a system level plug-in architecture, third party developers created a vast array of DSP effects which are mutually incompatible. Core Audio enables developers to offer audio plug-ins in a more centralized manner, called Audio Units, making it simple to manage the audio capabilities you add to your setup. Apple includes several Audio Units in Mac OS X, including a Velocity Engine-optimized reverb and a sample rate converter.

Plug and Play

Mac OS X includes USB class drivers for connecting audio and MIDI devices to your Mac, and will automatically recognize those devices that comply with industry-standard specifications when you plug them in. You don't have to install extra software to use them.

POWER MAC G5

Peripherals and Audio

- The Power Mac G5 has a FireWire 800 port, the next generation high-speed serial interface, in addition to two FireWire 400 ports (one on the front panel and three USB 2.0 and two 1.1 ports.
- One of the fastest peripheral standards ever developed, FireWire 800 doubles the throughput of the original technology and increases the maximum connection distance –up to 100 meters. Also supports many new types of cabling, making it indispensable for transferring massive files
- Front headphone jack
- Built-in speaker
- Stereo audio line in and line out minijacks
- Optical digital audio I/O (toslink)

DVD-R/CD-RW SuperDrive

The SuperDrive, standard on all models,

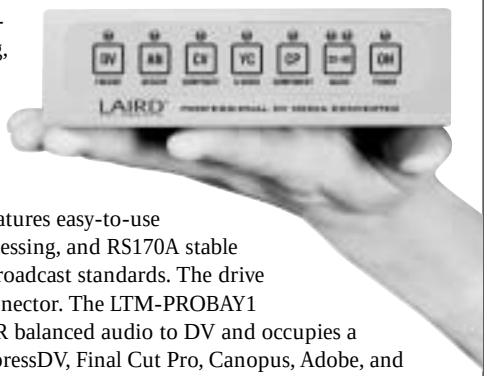


comes with the software needed to create your own DVDs — faster than ever before. With iDVD preinstalled and with optional DVD Studio Pro software, the G5 is a complete DVD production studio. And it's the fastest on the planet, too. And if you want to watch a DVD while your work is burning, or archive material at the same time, all Power Mac G5s offer a second optical bay. Choose a combo DVD/CD-RW to fill it up.

- SuperDrive-equipped G5s lets you read and burn both CDs and DVDs that can be played in most consumer DVD players.
- The SuperDrive (DVD-R/CD-RW) reads DVD titles at 8x and writes to 4.7GB DVD-R discs at 4x.
- Also reads CDs at 32x, writes to CD-R at 16x, and writes to CD-RW at 10x.
- It supports DVD-Video, DVD-ROM and DVD-R, as well as CD-ROM, CD-Audio, CD-R, CD-RW, CDI, CD Bridge, CD Extended, CD Mixed Mode and Photo CD.

Laird Telemedia LTM-PROBAY1

Slide-in Bi-Directional DV Media Converter for Power Mac G4 Systems



With little more than a screwdriver, the LTM-PROBAY slips into a single drive bay opening, turning an existing computer system into a complete DV editing workstation.
The LTM-PROBAY is a fully functioning analog-DV-analog media converter in a 5.25″ computer drive bay enclosure. Housed in the new DVMOD technology, PROBAY features easy-to-use front-panel controls, high-quality signal processing, and RS170A stable video output compliant with SMPTE/EBU broadcast standards. The drive simply plugs into a standard drive power connector. The LTM-PROBAY1 converts component, composite, YC, and XLR balanced audio to DV and occupies a single G5 drive bay. Compatible with Avid XpressDV, Final Cut Pro, Canopus, Adobe, and any other DV-based edit system software.

Communications

- 10/100/1000BASE-T Ethernet connector (RJ-45)
- Built-in 56K V.92 modem card
- Power Mac G5is Bluetooth-ready. Like AirPort Extreme, Bluetooth connects digital devices wirelessly. Operating within a much shorter range than AirPort Extreme, it enables wireless data transfers between a growing list of Bluetooth-savvy products.
- Built-in antennas and expansion slot for optional 54-Mbps AirPort Extreme Card. AirPort Extreme is a wireless networking technology based on the IEEE 802.11g standard. Just add an AirPort Extreme Card to the Power Mac G5 and it can wirelessly connect to an AirPort Extreme Base Station at speeds up to 54Mbps. That's nearly five times as fast as the transfer rates of the current 802.11b wireless standard.

Power Mac G5 Specifications:			
Model Number	**M9020LL/A**	**M9031LL/A**	**M9032LL/A**
G5 Processor	1.6 GHz	1.8 GHz	Dual 2GHz
L2 Cache (per processor)	512K	512K	512K
Frontside Bus	800MHz	900MHz	1GHz per processor
Main Memory (DDR SDRAM)	256MB PC2700 (333MHz) 4GB maximum	512MB PC3200 (400MHz) 8GB maximum	512MB PC3200 (400MHz) 8GB maximum
Ultra ATA/100 Hard Drive	80GB (7200 rpm)	160GB (7200 rpm)	160GB (7200 rpm)
Optical Drive	SuperDrive DVD-R/CD-RW	SuperDrive DVD-R/CD-RW	SuperDrive DVD-R/CD-RW
Graphics Support	NVIDIA GeForce FX 5200 w/64MB of DDR SDRAM	NVIDIA GeForce FX 5200 w/64MB of DDR SDRAM	ATI Radeon 9600 Pro w/64MB of DDR SDRAM
High Performance I/O	One FireWire 800 port, two FireWire 400 ports (one on front) three USB 2.0 ports (one on front), two USB 1.1 ports (on keyboard), AGP 8X Pro slot with graphics card installed, ADC and DVI connectors for dual display support.		
Audio	Optical digital audio I/O, analog audio I/O, front headphone minijack, speaker		
Networking	Built-in 10/100/1000BASE-T Ethernet and 56K modem; AirPort ready		

Each includes Apple keyboard and mouse, DVI to VGA adapter, AirPort antenna; Mac OS X v10.2 "Jaguar", iLife (iMovie, iPhoto, iTunes, iDVD), QuickBooks for Mac, FAXstf, Art Directors Toolkit, Microsoft Office v.X Test Drive, FileMaker Pro Trial, OmniGraffle, OmniOutliner, GraphicConverter, QuickTime, iChat, Safari, Sherlock, Address Book, iCal, iSync, DVD Player, Mail, Acrobat Reader, Classic environment & Apple Developer Tools.

FLAT-PANEL DISPLAYS

17˝, 20˝ and 23˝ Digital Flat-Panel Displays



An Apple display is an essential component of the Power Mac G4 user experience. Apple offers a lineup of pure-digital flat-panel displays that provide superior image quality and vivid color. The level of integration and the elegance of these displays present Power Mac G4 users with the best possible window on their work.

All Apple displays deliver the benefits of thin and light LCD technology, with twice the brightness, twice the sharpness, and twice the contrast of a standard CRT display. They can be calibrated for color-managed workflow environments and maintain consistent color and quality without requiring frequent recalibrations. Apple displays feature an adjustable base that allows users to adjust the viewing angle effortlessly. And since dual display capabilities are built into every Power Mac G4, it is easier and more cost-effective than ever to add a second display.

- With razor-sharp images, deeply saturated colors and brilliant screens, these best-of-class displays offer 16.7 million colors, a wide viewing angle (170° horizontal and vertical), incredible color fidelity, and an ultrafast pixel response you'll love when working with audio applications— to say nothing of elegant industrial design.
- Set up is a snap, thanks to the Apple Display Connector (ADC). A single cable carries the digital graphics, power, and USB signals to the display, dramatically reducing the number of cables you have to deal with. Built-in, two-port USB hub (discreetly positioned in the back of the displays) lets you connect peripherals.
- Twice the brightness, twice the sharpness, and twice the contrast ratio of the typical CRT display. And unlike other flat panel displays, they are designed with a pure digital interface to deliver distortion-free images that never need adjusting.
- The 17˝ Apple Studio Display with 1280-by-1024-pixel resolution, is an ideal alternative to a large CRT display
- The 20˝ Apple Cinema Display features 1680 x 1050 pixel resolution for display of application palettes. Widescreen format for simultaneous display of two full pages of text and graphics, or full-screen DVD and QT movies
- The 23˝ Apple Cinema HD Display provides unparalleled image quality and richly saturated color. Full high-definition resolution of 1920 x 1200 pixels for display of high definition (HD) still and video imagery. Even with such a wide viewing space, text remains sharp and colors are vivid and distortion free from edge to edge. The Cinema HD Display is perfect for enhancing large graphics—enabling high-performance image manipulation in gorgeous, richly saturated, flicker-free color.

FLAT PANEL DISPLAY SPECIFICATIONS

	Apple Studio Display	Apple Cinema Display	Apple Cinema HD Display
Model	M7649ZM/B	M8893ZM/A	M8537ZM/A
Screen Size and Type	17-inch (viewable) active-matrix LCD	20-inch (viewable) active-matrix LCD	23-inch (viewable) active-matrix LCD
Maximum Resolution	1280 by 1024 pixels	1680 by 1050 pixels	1920 by 1200 pixels
Viewing Angle	170° horizontal; 170° vertical	170° horizontal; 170° vertical	170° horizontal; 170° vertical
Color Support	16.7 million colors	16.7 million colors	16.7 million colors
Input/Output ports	Apple Display Connector; two-port, self-powered USB hub	Apple Display Connector; two-port, self-powered USB hub	Apple Display Connector; two-port, self-powered USB hub
Agency Approvals	ENERGY STAR, TCO 95	ENERGY STAR, TCO 95	ENERGY STAR, TCO 95

iPOD



Ultra-Thin Digital Music Players

The super-slim iPod redefines what a digital music player should be. It downloads music at blazing speed and can hold up to 7500 song—so you can take your entire music collection with you wherever you go. It weighs only 5.6 ounces—which is incredibly, 15% lighter than 2 CDs! It is also 65% smaller than a CD and just 0.62 inches thick, so it comfortably fits in the palm of your hand or pocket — and your life. Yet it offers a huge 10GB, 15GB or 30GB hard drive. Do the math: that's space enough to store three weeks of music — played continuously, 24/7 — or one new song a day for the next 20 years. Available for Mac and Windows the iPod incorporates hassle-free functionality and stunning, well-considered design that places it far ahead of the curve and the competition. Use the touch wheel to quickly and easily scroll through an entire music collection—using just one hand!

FEATURES

- The iPod is designed for listening to digital music away from your computer — Mac or Windows—and one simple connection does it all. A handy dock connector on the bottom enables the iPod to link to a computer and sync files via FireWire 400 (Mac or Windows) or USB 2.0 (Windows only). The result? Blazingly fast transfer speeds that let you load an album in as little as 10 seconds.
- The 15GB and 30GB iPod come with a dock that makes charging and syncing easier than ever. An elegant, convenient home base for iPod on a desk or connected to a home stereo system, the dock provides syncing via FireWire or USB 2.0, charging via FireWire and audio line out for connecting to powered speakers or a stereo system.
- Designed to give you the best portable digital music experience ever, the iPod delivers the highest sound quality from input to output. iPod supports the most popular audio formats including MP3 (up to 320 kbps), MP3 Variable Bit Rate (VBR) and WAV — giving you access to a wide range of audio file types. And iPod is the only portable digital music player that supports the AAC format (Mac-only), which features CD-quality audio in smaller file sizes than MP3, so you can fit even more songs on your iPod.
- Have a particular set of songs you want to hear? Queue up as many songs as you like in a playlist while away from your computer. Want to customize the iPod interface to fit your musical leanings? Just look under settings in the Main menu.

Bundled Software

- For Mac users, iPod ships with iTunes 4. iTunes 4 supports ripping music into the new AAC format which combines pristine CD-quality sound with smaller files sizes, compared to MP3s. And iTunes 4 lets you share and stream playlists from your personal music library to up to three other Macintosh computers for personal use.
- For Windows users, iPod includes the award-winning PC media player MUSICMATCH Jukebox Plus. As you add new songs or rearrange your playlists in MUSICMATCH Jukebox, the music will automatically update on iPod the next time it is connected to your PC.

Supplied Accessories

- Supplied earbud-style headphones offer a strong 20Hz to 20kHz frequency response range for high-quality sound with minimal distortion. And with Neodymium drivers, these headphones are five times more powerful than digital player headphones that use aluminum, cobalt or ceramic drivers.
- The 15GB and 30GB models come with a wired remote control that lets you fast-forward or rewind a track, play, pause and adjust volume — using just one hand.
- The 15GB and 30GB models also include an elegant carrying case with belt clip that protects your iPod. Clip it to your belt, purse or backpack for hands-free operation.

Optional Accessories

iPod Power Adapter:
Use an extra iPod Power Adapter for home, office or travel. The FireWire-based adapter allows you to charge your iPod when not connected to a computer.
iPod Dock

iPod Dock:
Add an additional dock for charging iPod at work or at home. Stereo line out lets the iPod Dock connects seamlessly to stereo speakers, making it a space-saving stereo.

Battery Pack:
Away from your computer for a few days? Optional backup battery pack gives you over 20 hours of battery life on four standard AA batteries.

Car Power Adapter:
Optional car power adapter includes a 3.5-mm lineout that plugs directly into a car stereo input jack so users can play iPod on the road.

World Travel Adapter Kit:
With the addition of this kit, iPod goes anywhere in the world. Includes six AC plugs with prongs that fit different electrical outlets around the world.

iSplitter:
Mini Stereo Y-Splitter connects two mini-stereo headphones or speakers to one jack so you can share your music with a friend.

iTUNES 4



Personal Digital Music Library

Bundled with the iPod (Mac version) as well as any Apple computer, iTunes 4 is easily the world's best best application for managing and enjoying your music. iTunes allows you to manage an entire music collection with one application. Everything is in one place—downloads are in the same library as other music you've collected. Create personal playlists or Smart Playlists that update automatically based on your listening patterns and preferences. Listen to your music on a Mac or your iPod, or burn a custom CD of favorite songs or playlists. iTunes 4 provides a soundtrack to your life—add music to slideshows, soundtracks to movies, and background music to DVDs.

FEATURES

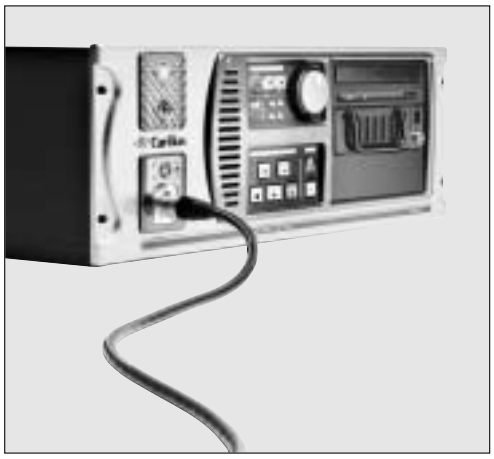
- iTunes 4 supports the AAC audio format (think high-quality sound with even smaller file sizes), and lets you share your music with other Mac computers on your local Ethernet or AirPort wireless network. And if you have a Mac with a SuperDrive, you can archive your entire music library to DVDs for safekeeping.
- A big part of MPEG-4 specifications, AAC (Advanced Audio Coding) is a cutting-edge encoding audio codec that's perfect for the Internet. AAC compresses much more efficiently than older formats like MP3 (which iTunes supports), while delivering quality rivaling that of uncompressed CD audio. In fact, AAC audio files compressed at 128 kbps (stereo) will be indistinguishable from the original uncompressed audio source.
- Music Sharing feature uses Rendezvous to give you remote streaming access to your personal music library from any room in your house. You don't have to manually configure anything, Rendezvous seeks out the other Macs on your local network and connects to them automatically.
- Generate dynamic Smart Playlists that reflect your preferences and listening habits. Instead of having to create playlists manually by clicking and dragging individual songs, you simply indicate what kind of music you like to hear. Set parameters with attributes such as Genre, Composer, Artist, Play Count, Last Played and so on — and then creates a personalized playlist that's in tune with your preferences.
- Automatically synchronizes with the iPod at high speeds. Simply connect iPod to your Mac via Firewire and download an entire music CD in just 10 seconds. Plus iTunes 4 lets you stay in sync with Play Count, Last Played and Song Ratings
- Access your iTunes digital music library and playlists from iPhoto, iMovie and iDVD. This allows you to use your music to build a slideshow in iPhoto, add a background soundtrack to your latest iMovie, or liven up your motion menus in iDVD.
- With iTunes it's easy to create CDs that play in your car, home stereo, Macs and on Windows-based PCs. With iTunes 4 though, CD burning gets even better with Soundcheck. Whenever you select a playlist in your library, the Browse button automatically turns into the Burn button. Click Burn CD, insert a blank disc and iTunes burns away. It's that simple.
- iTunes 4 can play spoken word content from Audible, the service that provides digital versions of books and publications.

The iTunes Music Store—Downloads Done Right



Instantly browse and buy music from your favorite artists. iTunes 4 comes with the fabulous iTunes Music Store stocked with hundreds of thousands of songs that you can preview and own with just one click.

AC-1



Audio Computer

When today's computers are on the drawing board, audio professionals are somewhere between the bottom and nowhere on the designer's list of target users. Working from the outside in, the average PC case does little more than keep the dog out. Covers and plastic front panels are wafer thin and employ clips rather than fasteners wherever possible, so as well as offering minimal sound insulation they also rattle like a snare drum when you monitor anything below 500Hz. They're also not designed to travel, so if you use a computer for audio, chances are you pretty much accept being tied to your own studio. On the other hand, the sophisticated manipulation of audio is one of the toughest jobs you could throw at a computer. Accurate timing and synchronization rely heavily on highly stable processing, and one minute of one track of full bandwidth audio soaks up the same amount of memory as the text of ten novels. But the unremitting pressure to lower the price of consumer PCs frequently means manufacturers resort to the cheapest available components. From simple things like switches, down to boards, drives and even memory. And with the emphasis on straightforward 'office' applications and web browsing, expandability and future proofing are also low on the list of priorities. Spare motherboard slots for example and bays for additional internal and removable storage, are frequently in short supply or absent altogether. . .

Welcome to the AC1 Audio Computer

19″ Rackmount Case Design

A PC fully loaded for audio can weigh as much as 50 lbs - heavier than most studio power amps. The design of a 19″ rack mount enclosure to safely support this type of load, and accommodate requirements for a high level of access and connectivity, presents several unique challenges.

Carillon uses a box (case) that is over- engineered from high grade aluminum and 2mm 14-gauge steel to provide an incredibly strong and totally rigid housing. A full 4U height contributes to a particularly well cooled, spacious interior with generous capacity for retro-fit components. The massive front panel is precision die cast in ADC 12/LM2 aluminum alloy by 10,000psi high pressure casting machines. This method is far more costly than regular gravity die casting but delivers a stronger, more consistent material, cast to much closer tolerances.

But the Carillon Audio Computer is equally at home on the desktop largely due to specially designed, removable Sorbothane feet. If you do prefer this arrangement, it's worth noting that the fourteen gauge cover supports a conventional 17″ monitor effortlessly. And nowhere is this heavy duty approach more appropriate than on the road—making these computers ideal for live and mobile recording use.

Internal layout is clean & spacious and the full-size ATX motherboards have six PCI slots and four USB ports for maximum expandability. Carillon uses Intel Pentium processors exclusively to consistently deliver the floating-point performance necessary for DSP intensive multitrack digital audio.

Front Panel



With software equivalents of most music technology hardware now available, 'software only' production is reality for a small but growing number of producers. But real controls are fast and fun so Carillon includes a bay to receive a variety of inexpensive optional control boards like 'tape' transport buttons, MIDI controller knobs and a jog-shuttle edit controller. The bay is user accessible by way of stainless steel hex bolts used to fix the main bezel.

High temperatures in some fully loaded enclosed racks, particularly live racks stacked with power amps, necessitate the intake of cooler air through the front panel. The intake features an easily replaceable filter.

There is a standard Neutrik bay housing the highest quality audio socket routed through to the computer's back panel. It can be permanently connected to the soundcard which often employs fiddly consumer grade connectors intended for infrequent 'install and leave' use. Alternatively, you can use it for a stereo headphone output.

In addition to three 5¼″ bays, the Pentium 4 compatible chassis can accommodate a further two drives mounted transversely inside the machine, both in noise cancelling Silent Drive enclosures.

AC-1

Quieter by Design



In an ideal world with little pressure on finances and floor space, noisy stuff like tape machines and computers are exiled to a 'machine room' which is acoustically isolated from the studio mixing area. But for many, even if the space is available, countless items come above a separate machine room on the studio shopping list, so the computer sits in the heart of the studio physically as well as well as functionally. The reality is that most studios operate from one room.
In this situation, comfortable, accurate monitoring is seriously compromised by a noisy computer (how can you hear your gate thresholds properly, or long fade-outs against a background of fan and drive noise?). Worse still, the need to record clean acoustic instruments and vocals can rule out the use of a standard PC altogether. Particularly when you can't run a microphone cable into the bathroom because you are recording yourself!

Solid State

The AC-1 enclosure design cuts hardware noise in two ways: First, the sound insulation properties of the thick steel and aluminum case substantially reduce transmitted noise from internal components. Second, the rigidity of these heavy gauge materials combined with the use of secure bolt fastenings, make for a highly vibration-damped structure. This effectively eliminates rattles and buzzes 'in sympathy' with fans, drives and the music you're monitoring.

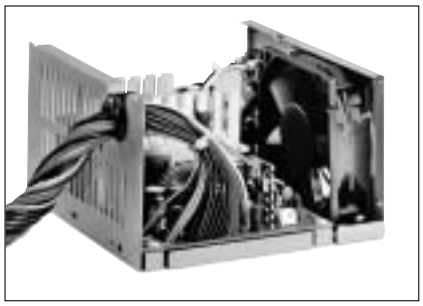
Splendid Isolation

Anyone who has ever rested an acoustic instrument on a wooden table while playing, knows that vibrations are transmitted and amplified when solid objects make contact. To combat this the AC-1 incorporates absorbent rubber gaskets to the rear of the 19" rack ears, to reduce metal-to-metal contact between the computer and the rack, and so reduce vibration transmitted from one to the other.

Big Foot



For desktop users Carillon designed large removable feet, utilizing visco-elastic Sorbothane. This unique material is used to isolate scientific instruments and audiophile equipment from vibration. Its high storage modulus means that an extremely high proportion of input energy is dissipated rather than transmitted. (Although Sorbothane is extremely expensive, the feet are standard.)

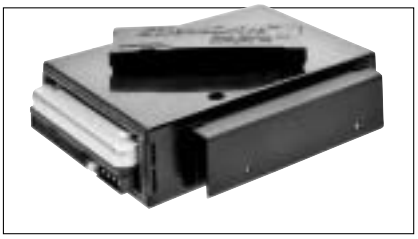


Today's hard drives run at speeds of up to 10,000 RPM. Power hungry applications need the fastest processors and processors will always be able to run faster if you cool them down.

We've selected the real 'church mouse' components. Super quiet hard drives, foam lined sound-proof enclosures, Silent Systems processor fans and our own Carillon low noise PSU.

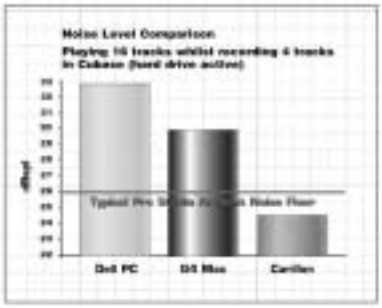
The power supply is typically a computers noisiest single component. The Carillon UltraMute PSU combines special low hum transformers with an all new fan, featuring substantially smoother fan-motor bearings and a reduced turbulence fan blade design.

A unique processor fan incorporates a radical 360% (radial fin heat sink). This patented design provides far greater surface area than conventional for increased heat dissipation with a reduced airflow requirement. In practice this means the fan can run far slower and more quietly.



However advanced the drive design, whirring drive platters and clicking heads can be further isolated by enclosing the drive completely in an acoustic sleeve. Carillon's Molex Silent Systems foam lined sleeve reduces drive noise by over 90% and includes two aluminum plates that conduct heat into the surrounding air and chassis mounts, and provide earth grounding.

Carillon fits Quantum Fireball plus hard drives as standard. They feature Quantum's own QDT (Quiet Drive Technology) and produce at least 4dB less noise than the average for comparably specified drives. We also offer the more expensive Fujitsu 'dynamic pressure bearing' drives as an option (marginally the quietest at the time of going to press) but find that the Silent Drive enclosure is so effective that, once fitted, the difference between the two brands is negligible.



Inevitably, a first impression of hardware noise is an early topic of conversation, and nearly everyone asks if we've switched it on yet! That really says more than a thousand dB measurements. Simply speaking, while Macs and PCs have always been clearly audible, the Carillon AC-1 is only barely audible - even in the quietest studio environments. But don't take our word for it. These graphs show the relative noise levels for a Dell PC, a Macintosh G4 and our Carillon Audio Computer, measured during busy hard disk activity and while idling.

AC-1

Carillonfix: Remote Control Technical Support



Although they'll be properly manned, Carillon does expect the telephone lines to be quiet for a computer music company. But if their on-screen manual doesn't show you how and they can't talk you through it, they'll be down but not out.

Carillonfix is a custom remote control application that lets their support engineers literally take the controls of your computer online. Book a session by phone or in the owners section on their website, launch the application at the arranged time and wait for them to dial in.

If you send a computer back to the supplier to fix a software problem you're none the wiser when you get it back. Working the Carillon way, you can watch them do repairs on screen (yes, the mouse actually moves) and communicate with their engineer via the onscreen chat window. Discuss the pattern of the problem for example, even ask to see a particular fix again.

In the event of an irreparable software problem Carillonfix gives offers one final option. When they build your system they burn a hard drive backup onto CD. It is supplied with the computer but in case you're not confident about re-installation, Carillon also keeps a copy filed with your personal details. With your permission they can reconfigure your entire system online for you to start afresh. Unfortunately though, even Carillon can't fix your hardware online, but they're confident enough about their build quality to cover hardware failure with an advance replacement guarantee for components. Alternatively they'll collect your machine, repair it and send it back to you within three working days.



Loopstation: On–Line Sampling

Carillons exclusive On-Line Loopstation Application, searches, previews and downloads from a huge range of studio quality samples free of charge. Each Carillon AC-1 audio computer has the Loopstation search engine pre-installed. Simply click on the Loop icon on your desktop and once the Carillon server recognizes your unique user ID, you're into the latest library. You'll soon discover that the sample search engine is by far the most intuitive and exhaustive anywhere. Sounds and loops are logically multi-layer indexed and you can browse a full list of relevant samples, however far you have refined your search. But despite being sophisticated, its remarkably quick, employing hardware style logic for the user interface with absolutely no screen redraws.

- Preview and download are simplicity itself (Carillon has their own dedicated servers and E1 connection guaranteeing the fastest possible transfer), but where Carillon really shines is the quality of the samples - these are definitely not your average coverdisc promos. Thousands of Neve-mixed, Apogee-converted, 24-bit samples as good, or better by category than anything else out there—free of charge.
- In addition to these free of charge sounds the Carillon Loopstation will also carry an increasing range of third party samples, available for Carillon owners to download at a substantial discount. (And remember, you don't need a software sampler to use samples in your music. The samples are encoded as .WAV files, compatible with all the leading sequencing software.)
- The right sample can make a track - but finding that sound from poorly classified data currently available can take longer than most of us can afford. The ultra flexible engine lets you direct the search. Say you're working on a disco house track and looking for a drum loop: search 'loops by style', select 'house', find 'disco' and refine by instrument (maybe choosing 'drums', then 'acoustic'), before listing samples.

B&H PHOTO - VIDEO - PRO AUDIO

AC-1

Complete Systems

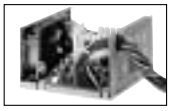
There are seven "Off the Shelf" Carillon Core Audio Systems. Each one incorporates their acclaimed heavy-duty, 19˝ all aluminum steel chassis—engineered for ultra-low noise and featuring Carillon's unique front panel patching and bays for their unique 'real controllers'. Every core system also includes:



standard with every system

Ultra-Quiet Components

- Carillon's own Ultramute Power Supply



- Molex Silent Drive enclosure
- Low-noise CPU cooling fan
- 1.44MB Floppy Disc
- 56Kbps Auto-Detect Modem
- Black Carillon keyboard and mouse

Bundled Software

Hundreds of dollars worth of killer applications, preinstalled and optimized.

- Steinberg Clean
- Steinberg Wavelab Lite
- Norton Ghost
- VST Plug-in Pack

Remote Support

- Carillon Fix literally enables their engineers to take control of your machine online
- Carillon's How & Help manuals and tutorials
- Carillon Image and Recovery Disk
- 3-Year Warranty

Loopstation Samples

Carillon's own ultra flexible search engine micro indexes by sound type A growing library of Neve mixed, Apogee converted 24-bit samples, including 5000 leading edge loops and instruments.



	CORE 1	CORE 2	CORE 3	CORE 4	CORE 5	GigaCore 1	GigaCore 2
Description	A solid reliable audio PC making an ideal starting point for many custom systems.	Step up to dual hard drives and greater processing power with confidence.	Take advantage of 533MHz front side bus processing speed for more serious applications.	Built for serious professionals, this unit features over 1GB of RAM.	The ultimate audio PC is ready to tackle the most serious applications with speed & grace.	Giga Studio users will benefit from expanded RAM & hard disc capacity & DVD-CDRW Combo Drive.	Serious Giga Studio users will appreciate the speed and capacity of this system.
Intel Processor	Pentium 4 1.7GHz	Pentium 4 2.0GHz	Pentium4 2.4GHz 533MHz FSB	Pentium4 2.53GHz 533MHz FSB	Pentium4 2.8GHz 533MHz FSB	Pentium 4 1.7GHz	Pentium4 2.8GHz 533MHz FSB
Intel Motherboard	D845EBG2L DDR 533 MHz	D845EBG2L DDR 533 MHz	D850EMV2L RDRAM 533MHz	D850EMV2L RDRAM 533MHz	D850EMV2L RDRAM 533MHz	D845EBG2L DDR 533 MHz	D850EMV2L RDRAM 533MHz
Video Card	ATI XPERT 2000 32MB	ATI XPERT 2000 32MB	Matrox G550 Dual Head 32MB DVI/CRT	Matrox G550 Dual Head 32MB DVI/CRT	Matrox G550 Dual Head 32MB DVI/CRT	ATI XPERT 2000 32MB	Matrox G550 Dual Head 32MB DVI/CRT
Memory	256MB 266Mhz PC2100 DDRRAM	512MB 266Mhz PC2100 DDRRAM	512 MB PC800 RD RAM	1024 MB PC800 RD RAM	1024 MB PC800 RD RAM	1024MB 266Mhz PC2100 DDRRAM	1024 MB PC800 RD RAM
Hard Drives	40GB Seagate Mark IV Ultra Quiet 7200 RMP	Two Seagate 40GB Mark IV Ultra Quiet 7200RMP	Dual 40GB/80GB Seagate Mark IV 7200RMP	Dual 40GB/80GB Seagate Mark IV 7200RMP	Two 80GB Seagate Mark IV 7200RMP	40GB plus 120GB WD SE HHD with 8MB buffer cache	40GB plus 120GB WD SE HHD with 8MB buffer cache
CD-RW Drives	Black Lite-On 40x12x40	Black Lite-On 40x12x40	Black Plextor 40x12x40	Black Plextor 40x12x40	Black Plextor 40x12x40	DVD-ROM	DVD-ROM
OS	Windows XP	Windows XP	Windows XP Pro	Windows XP Pro	Windows XP Pro	Windows XP	Windows XP Pro

Real Controllers

Carillon optional Real Controllers give you a traditional fingertip control of a variety of software functions and parameters. They connect internally, direct to the motherboard via their own drivers so you don't tie up valuable MIDI ports.

RK8 MIDI Controller Panel

A set of eight mappable MIDI controller knobs for use with virtual synthesizers, samplers, mixers, effects - pretty much anything with knobs or sliders. You can select General Midi, GS and XG or map parameters. We use high quality damped action pots and our own legible pointer knobs

JS1

Ideal for audio editing and also transforming the use of sequencers, the JS1 lets you set up to 20 locate points and jump directly to them with one or two button presses, in addition to having dedicated master in/out buttons. Jog/shuttle gives precise control over FF and REW as well as super-detailed scrubbing with unique jog calibration knob.

RTM 1 Transport Control

Feel comfortable with traditional transport controls? Then you'll love the RTM 1. Nikkai illuminating switches provide a reassuringly positive feel, and mimics your sequencer controls exactly. It also incorporates a switchable metronome. It's visual rather than audible with the metronome light flashing on the beat and the switch flashing on the bar.

SONY

FLAT PANEL LCDs

Sony Flat Panel LCDs are an artistic statement for your home or studio- even when they are turned off! Perfect for song writers, editors and producers. Superior focus and screen brightness, means sharp images and crisp text. You won't get flicker-induced headaches and eyestrain, even If you use the displays highest resolution. LCDs also use about half the power of comparable CRTs - a consideration if you run your computer for a good portion of the day.



BENEFITS:

Reduced Eyestrain -
Flicker free performance and minimal screen glare is easy on your eyes

High Brightness -
Dynamic, vivid images—especially under harsh lighting

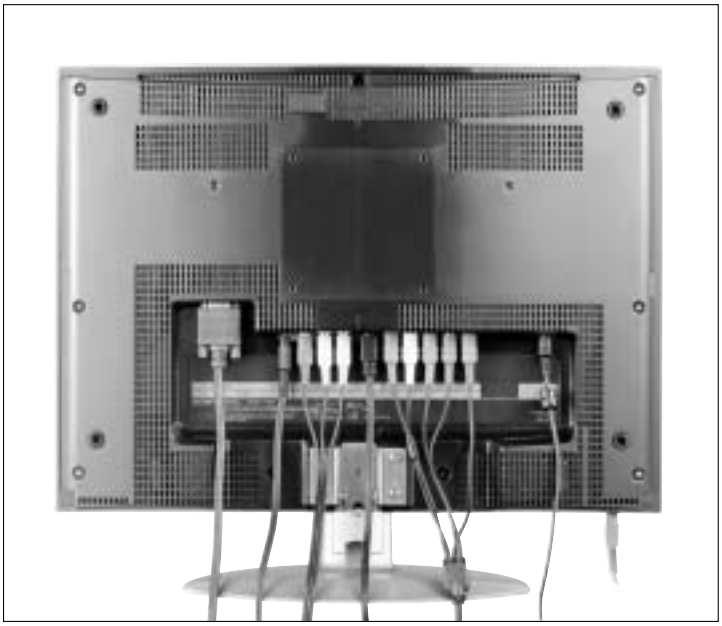
Space Saving -
Ultra-slim styling maximizes desktop space. Plus they're wall or arm mountable.

Energy Saving -
Low power consumption and energy bills

Regardless of which technology you choose, size does matter. A larger monitor offers larger viewing area and higher resolutions, making it easier to view text and graphics. More viewing area also means you have the ability to simultaneously open more windows side-by-side.

- ➢ 17- and 18″ are a good choice if you spend a lot of time in front of your screen
- ➢ 19″ are a the best choice if you use your display all day
- ➢ 21″ and above are optimal for design and finance professionals who work on detailed graphics and large spreadsheets

Sony's innovative advances in display technology have revolutionized the user experience. Remarkable picture quality, stylish designs and a variety of convenience features including compatibility with both Mac and Windows operating systems. And this year's line-up is no exception:



- ◆ Sony's Auto ImageSet and Digital FlexRes technologies maximize display performance by automatically adjusting images at multiple resolutions. Vibrant images are housed in an assortment of ultra-slim, energy saving designs.
- ◆ TFT active matrix LCD technology combined with anti-glare coating, provides clear, flicker free images that reduce eyestrain.
- ◆ Slim Bezel innovative design combined with integrated power supply and cable management system keeps space-constrained work spaces organized and clutter-free. Adds efficiency and style to the home or office.
- ◆ Digital & analog input connections maximizes video performance, dual inputs support simultaneous computer connections
- ◆ Automatic and instant adjustment of images at multiple resolutions optimize video performance
- ◆ Stereo speakers provide stereo audio for multimedia, music and internet content
- ◆ Lower power consumption provides increased savings throughout the life of the product
- ◆ Wall mountable, the stand base can be removed to support arm or wall mount applications

FLAT PANEL LCDs

You've made the decision to buy a computer system. Or you simply want to upgrade your monitor. Great! Now... which one should you choose? With so many sizes, features and technologies available, choosing the right display to fit your needs can be confusing. Sony understands that different people have different needs. So they've created a full line of Flat Panel LCDs -a comprehensive variety of models each offering the perfect blend of features and performance.

StylePro Series

StylePro Flat Panel LCDs combine slim styling and performance at a remarkable value. Enjoy flicker-free, bright and colorful screen images at multiple resolutions using Sony's Automatic Image Adjustments. An adjustable base maximizes desktop space and provides ideal viewing angle for maximum comfort. Upgrade to a Sony LCD today!

DeluxePro Series

DeluxePro Flat Panel LCDs feature bright color detail, superior video performance, and clear stereo sound - the perfect combination of performance and style. Packed with conveniences for the home or office including digital inputs, Automatic Image Adjustments, integrated power supply, smart cabling and special energy-saving ECO mode.

PremierPro Series

PremierPro Flat Panel LCDs deliver superior technology to the professional. Slim bezel combined with digital input, super bright and high contrast LCD panel make this display an ideal choice for financial applications. Wall mount design with built-in power supply and cable management system maximizes desktop space.

Entertainment Series

Driving entertainment beyond the PC, these 16:9 wide screen monitors connect directly to a variety of devices with stereo audio, composite, component, S-Video and PC inputs. They are also HDTV compatible (1080i and 720p) and feature SRS WOW Audio Technology to create a full, wide sound experience with deep, rich bass.

	STYLEPRO SERIES		DELUXEPRO SERIES			PREMIERPRO SERIES		ENTERTAINMENT SERIES	
Model Number	**SDM-S71 SDM-S71/B**	**SDM-S81 SDM-S81/B**	**SDM-X72 SDM-X72/B**	**SDM-X82 SDM-X82/B**	**SDM- X202B**	**SDM- P82/B**	**SDM- P232W/B**	**SDM- V72W**	**SDM- V72W/B**
Viewable Image Size	17.0″	18.0″	17.0″	18.1″	20.1″	18.1″	23.0″	17.1″	17.1″
Dot Pitch	0.264 mm	0.279 mm	0.264 mm	0.280 mm	0.255 mm	0.280mm	0.258 mm	0.291 mm	0.291 mm
Cabinet Color	Two-Tone Grey or Black	Two-Tone Grey or Black	Black & Silver	Black & Silver	Black & Silver	Black & Silver	Black & Silver	Dark Pearl Blue & Metallic Silver or Black	
Audio, Headphone Jack (HPJ)	N/A	N/A	Stereo Speakers, HPJ	Stereo Speakers, HPJ	Stereo Speakers, HPJ	N/A	N/A	Stereo Speakers with SRS WOW, HPJ	
Horizontal Scan 28-61kHz	28-92kHz	28-61kHz	28-92kHz	28-92kHz	28-92kHz	56-85kHz	28-92kHz	28-70kHz	28-70kHz
Vertical Refresh	48-85Hz	48-85Hz	56-85Hz	56-85Hz	48-85Hz	56-85Hz	48-85Hz	48-85Hz	48-85Hz
Response Time (Rise/Fall)	25ms	30ms	30ms	50ms	30ms	50ms	40ms	22ms	22ms
Native Resolution	1280 x 1024	1280 x 1024	1280 x 1024	1280 x 1024	1600 x 1200	1280 x 1024	1920 x 1200	1280 x 768	1280 x 768
Contrast Ratio	400:1	400:1	400:1	400:1	350:1	400:1	350:1	500:1	500:1
Luminance	250 cd/m²	200 cd/m²	280 cd/m²	300 cd/m²	250 cd/m²	300 cd/m²	200 cd/m²	300 cd/m²	300 cd/m²
Viewing Angle	160°/155°	160°/160°	160°/160°	170°/170°	170°/170°	170°/170°	170°/175°	150°/120°	150°/120°
Tilt & Height Adjust	Tilt	Tilt	Tilt & Height	Tilt & Height	Tilt	Tilt	Tilt	Tilt	Tilt
Input Connectors	Analog (HD15)	Analog (HD15)	Digital (DVI-D) and Dual Analog (HD15)	Digital (DVI-D) and Dual Analog (HD15)	Digital (DVI-D) and Dual Analog (HD15), USB Port	Digital (DVI-D) and Dual Analog (HD15)	Digital (DVI-D) and Dual Analog (HD15)	Component S-Video/ Composite, PC (HD15)	Component S-Video/ Composite, PC (HD15)
Color Temperature Control	9300K, 6500K, + user5000K							11500K, 9300K, 6500K	
Max Power/Eco Power	45 W	50 W	40/32 W	58/48 W	78/69 W	58/48 W	78/48 W	49 W	49 W
Internal Power Supply	N/A	N/A	N/A	Yes	Yes	Yes	Yes	N/A	N/A
Dimension (WHD)	16.7x15.7x9.2″	17.3x17.1x8.7″	17.2x16.1x8.9″	17.7x16.7x9.5″	20.5 x18.6x10.1″	17.7x16.7x9.5″	22.3x17.6x9.9″	15.9x18.6x7.2″	15.9x18.6x7.2″
Weight	14.4 lbs.	15.0 lbs.	15.7 lbs.	19.0 lbs.	26.5 lbs.	18.1 lbs.	28.7 lbs.	13 lbs.	13 lbs.
Limited Warranty	3 Years	3 Years	3 Years	3 Years	3 Years	3 Years	3 Years	3 Years	3 Years
TCO Compliant	S71 - TCO'99 S71/B - TCO'95	S81 - TCO'99 S81/B - TCO'95	X72 - TCO'99 X72/B - TCO'95	X82 - TCO'99 X82/B - TCO'95	TCO '95	TCO '95	TCO '95	N/A	N/A
Connects to	PC	PC	PC	PC	PC	PC	PC	DVD & VCR Players, Videogame systems, Digital Camcorders & Cameras, HDTV & TV Tuner sources, PCs	

GLYPH TECHNOLOGIES

TRIP2



Are you ready for performance beyond your expectations? Whether you're recording, mixing, editing or capturing — no matter the production environment — the Trip2 will reward you with a stress-free storage experience. Since Glyph puts its Oxford 911- based bridge board in the tray with every drive, you are assured that your data integrity and performance will be consistent wherever you take it.

Specifically designed for high-resolution audio, as well as the high-bandwidth requirements of video, Trip2 allocates audio files to multiple drives allowing you to achieve maximum track counts at today's high sample rates. Plus, a single FireWire Hot-Swap Tabletop receiver is included with the Trip2 so you can take your work on the road. Last, the Trip2 is ultra quiet, employing Glyph's QuietMetal which reduces noise from vibration. No matter the audio environment, the Trip2 offers options that go way beyond an off-the-shelf storage system.

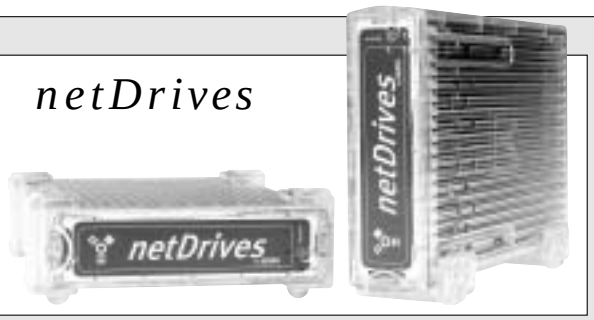
FEATURES

- The multiple ARM processors embedded in each Oxford 911 chip throughout the system (one processor for each drive), provide better than ever control and peak performance that will improve over time.
- For seamless integration into your studio choose from configurations with six hot-swappable drives, AIT back up, and DVD-RW, or 12 hot-swappable drives for over 2TB. Trip2 makes it easy for you to allocate files and use round robin features of DAWs to achieve 108 tracks of 192 kHz audio.
- Durable 3RU rack mountable enclosure
- QuietMetal construction for each drive isolates vibration eliminating external noise and enables ultra-quiet operation for your studio environment.
- Silent operation is enhanced by automatic thermal-sensing cooling to limit fan noise, prolong the life of your system, and protect your work.
- High performance 7200 RPM hard drives available in 40, 60 or 120GB capacities.
- Glyph S.M.A.R.T Manager Software to monitor your system and protect your data with predictive failure analysis to warn you of issues with your storage subsystem before it's too late

True FireWire Hot-Swappable Hard Drive Technology

- You can move projects from one room to another; you can track directly to it, or use a different drive for each client's work. You can send a drive home with a customer, or use one drive to back up the data on another one. The possibilities are endless with this receiver and tray design.
- What sets this system apart from all other "Hot-Swap FireWire" products is that the Glyph hot-swap interface operates according to the IEEE 1394 standard. The bridge board is built into the drive/tray. One of the benefits of FireWire is hot-swappability, which means you remove or insert the tray at the FireWire interface level.
- Removing a hot swap drive interrupts the data signal at the FireWire bus rather than on the ATA bus as with other hot swap systems, thus ensuring data integrity and consistency. Glyph's FireWire interconnectivity backplane assures you the best interface to maintain the integrity of your data.

netDrives



The netDrive FireWire hard drive is a value-priced, sleek portable storage solution for demanding environments. They deliver high performance with a dual port, standard 6-pin FireWire and USB 2.0 interface. Designed in a small, light package, it allows you to move your data files quickly, reliably and easily.

- Up to 180GB (FireWire-only models) or 120GB (FireWire/USB combo models).
- 7,200 rpm, ATA/100 drives with a 2MB cache and Oxford 911 chip for smooth, consistent data transfer.
- Small, durable, polycarbonate case runs quietly, is portable and easy to use anywhere.

GLYPH TECHNOLOGIES

TRIP2 TABLETOP

Hot-Swap FireWire Tabletop Receiver



With the Trip2 Tabletop, you can move projects from one room to another; can track directly to it, or use a different drive for each client's work. You can send a drive home with a customer, or use one drive to back up the data on another one. The possibilities are endless with this receiver and tray design. The Trip2 Tabletop accepts the same Glyph drive/tray technology as the Trip2. It includes a built-in power supply and standard dual 6-pin powered FireWire interface.
Imagine being able to record in the studio, pull the FireWire drive and take it home to mix the record there.

- Now Glyph's latest high-tech hotswap and bridging technology, first introduced in the 3U-rack Trip2 is available in the smaller, portable form factor of a tabletop drive
- What sets this system apart from all other "Hot-Swap FireWire" products is that the Glyph hot-swap interface operates according to the IEEE-1394a FireWire standard.
- The bridge board is built into the drive/tray. One of the benefits of FireWire is hot swappability, which means you remove or insert the tray at the FireWire interface level.

GLYPH QUIETMETAL (THE SANDWICH OF SILENCE)

When you're recording in the studio, any piece of gear that makes background noise is going to get in the way of your session. That's why some people try to silence their gear by putting noisy hard drive units in "isolation boxes" to muffle the noise of the drives and fans spinning. Unfortunately, "iso" boxes suffer from heat buildup. Even in the best boxes, it's nearly impossible to get total acoustic isolation and also get the heat out of the hard drive that's in the box.

Most of the noise in a hard drive system comes from the hard drive itself, and the fans used to circulate air through the enclosure. Noise is vibration, and it is the vibration of the hard drives and fans that is so objectionable in an audio recording studio.

So Glyph studied the situation, and decided to block the vibration of the hard drive and fan from getting to the chassis, where it could make its way into the studio. The clear place to do this was in the mounting hardware that holds the drives and fans within the chassis.

QuietMetal Tri-Laminate

Glyph took advantage of advanced composite metals technology, and incorporated a tri-laminate — two layers of metal skin, separated (and bonded) by a viscoelastic polymer core — into their systems to dampen vibration and control noise. The tri-laminate metal actually absorbs vibration before it can get to the unit chassis. Extensive research and testing demonstrated that careful application of QuietMetal would dramatically cut the noise level produced by the hard drives and fans.

Glyph uses QuietMetal for the internal drive frames and rear fan panels, making their products the quietest high-performance storage systems available today. The combination of viscoelastic damping and secure fastenings, along with our use of the latest in quiet drive mechanisms, virtually eliminates vibrational noise. This translates into a storage solution that is significantly quieter than the rest of the gear in your studio — which means your recording is quieter, too.

FIREWIRE FIXED-DRIVE TABLETOP

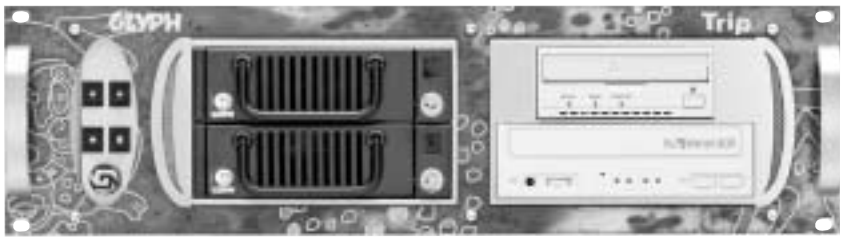


The FireWire Fixed-Drive Tabletop is the ultimate solution for your project and mobile studio. It includes a built-in power supply and a standard dual 6-pin powered FireWire interface, and it's built in a sturdy steel case. It features the latest technological advancements from Glyph, including QuietMetal to eliminate noise, and exclusive FireWire bridging technology. This high performance hard drive system is available in capacities up to 180GB so you can record, mix and edit multi-track audio. It will even handle high definition sample rates, and includes Glyph Audio Storage Toolkit formatting and partitioning software.

GLYPH TECHNOLOGIES

SCSI TRIP FAMILY

SCSI Drives



When you need a full solution in a single enclosure Glyph has the best Trip for you. The industry-standard, thousands of today's most successful audio and video projects were created with a Trip. They offer "Monster Truck" storage with removable technology and your choice of standard configurations. Power switch and SCSI ID selection switches are conveniently available on the front of the units and all Trip configurations include the necessary cables and termination.

SCSI provides significantly faster data transfer than Firewire and allows for the connection of up to 15 peripheral devices. The high speed and reliability of SCSI has made it the standard interface in audio and video production.

Ego Trip

The name says it all… For the professional who wants the biggest and the best. Two hot-swappable high-capacity 10,000 or 15,000 RPM hard disk drives for recording, mixing and editing. Also comes equipped with AIT-3 100GB native tape backup and 48X CD-R/RW with Glyph's own Firewire interface technology.

Project Trip

The Trip for the budget-conscious. Includes an AIT-1 35GB Tape Backup, two 18GB hot-swappable 10,000 RPM hard disk drives and a 48X CD-R/RW with Glyph's own Firewire interface technology.

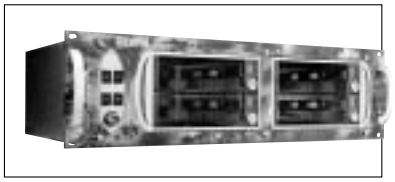
Power Trip

Ideal for high-speed and high-capacity backup. Equipped with a 50GB AIT-2, two 73GB hot-swappable 10,000 RPM hard drives, and a 48X CD-R/RW with Glyph's own Firewire interface technology.

Studio Trip

For efficient management of critical projects, the Studio Trip includes an AIT-1 35GB Tape Backup, two 36GB Removable Cheetahs and a Plextor 48X CD-R/RW.

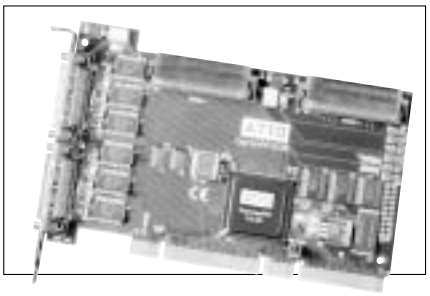
Naked Trip



This rack-mount enclosure is the ultimate in scalability, with 4 open hot-swap receivers available for drives in trays. We can configure with Naked Trip with Narrow, Ultra Wide or Ultra 160 receivers, it's your choice.

Note: Naked Enclosures do not include cables or terminators, that's why they're called "Naked".

Host Bus Adapters



Host bus adapters, or HBAs, are necessary when connecting a peripheral to a computer that doesn't have native support for that peripheral's interface. Glyph offers HBAs from ATTO and Adaptec Technology to support all major operating systems, with various connection interfaces and speeds.

ATTO ExpressPCI UL3D: Dual-channel Ultra3 SCSI Host Adapter.....**Call**

ATTO ExpressPCI UL3S: Single-channel Ultra3 SCSI Host Adapter.....**Call**

Adaptec FireConnect 4300:
3-port, 400 Mbit/sec FireWire/1394 host adapter for PCs and Macs.....**Call**

AUDIO STORAGE GUIDE

In multi-track audio there are generally numerous data files for each track, especially with punch-ins and edits. The rotational speed of the platters directly affects the number of audio tracks you can achieve because of the time taken for the drive to find and process the various files. Files are written to the different sectors across the platter, over the course of your project these files become non-contiguous, resulting in drive fragmentation. The heads move back and forth across the platters to write and read the scattered data. Because of the time it takes for the data to pass underneath the heads, drives with slower spindle speeds like 7,200 RPM take longer to access each file than drives with speeds of 10,000 and 15,000 RPM. Audio applications are very demanding on drives since small delays in the delivery of requested files results in play-back errors. Faster speeds yield extra audio tracks with more edits. This is also why that bargain price 5,400 RPM drive is useless for your DAW.

GLYPH TECHNOLOGIES

DOUBLE SPACE RACK ENCLOSURE

Double Space (2RU) Rack Mountable Systems



These are classics. Totally customizable, you can have any two devices configured however you need them. Whether you need dual hot-swappable hard drives, a single hard drive with a tape back up device, or even a single tape back up device with a media storage bay, this is the multi-functional, rack space saving classic.

Drives: Glyph-optimized for digital audio and video. Available capacities of 18GB, 36GB, 73GB, 146GB

Spindle Speeds: 7,200 rpm, 10,000 rpm, 15,000 rpm

- **Interfaces:** SCSI 3, Ultra160
- **Compatibility:** Macs and PCs with appropriate host bus adapter cards for your applications.

SCSI Tabletops

The total package of reliability and performance



Simply put, Glyph's stackable Tabletops deliver big with 10,000 RPM super fast drives that are perfect for hard disk audio recording. Glyph's stylish enclosures are as innovative as they are unusual. Built specifically for use in audio environments, they offer quiet fans, well-insulated drive housings and beefy power supplies to keep your SCSI drive spinning for years to come. Not only do you get reliable operation and loads of space for your precious data, but you also get one of the best names in the business standing behind your recording media.

All of Glyph's Tabletop SCSI hard drives are pre-tested and formatted, and include a high-quality 3' SCSI cable, active terminator and a power cable.

WILDFIRE

Fast, Reliable FireWire 40x CD Recorder



The WildFire is the ultimate portable, easy-to-use, rewritable CD recorder. Its speed and reliability make it an excellent addition to any studio. WildFire has the features you need to fully master your CDs. It has Disc-At-Once (DAO) technology and a 4MB buffer for smooth transfer of your audio files, and it's Red Book Audio compliant. This FireWire CD-R/RW ships with cable, Mac/PC software and media so you can start burning right away.

- A fast, reliable 3-in-1 CDR/W drive, it writes at 40x, re-writes at 12x, and reads at 40x
- Use as alternative to SCSI for a clean configuration; free up SCSI bus for other devices
- WildFire is a great alternative to a SCSI CD-R/RW. Because it's FireWire, it's simple to use and won't require a PCI card which uses up a PCI slot. FireWire's hot-pluggable dynamic reconfiguration means that you can re-cable your peripheral devices without re-booting.
- Utilizes the Oxford 911 bridging chip which is optimized for CD-R(RW) applications
- Disc At Once (DAO), Red Book Audio compliant so you can fully master your CDs
- 4 MB buffer assures smooth transfer of your audio data
- Burn-Proof technology eliminates buffer underrun errors in fast write modes, allows for multi-tasking
- Flash ROM allows for easy upgrade over the Internet
- Compatible with Macintosh & PCs (your PC may require a host bus adapter and software)
- Comes complete with Toast, FireWire cable, and CD media accessory pack

AIT DRIVES

High-performance Tape Back-up



As robust as today's hard drives may be they are still mechanical devices that will stop working one day. In addition, with the advent of 96kHz and higher sampling rates, audio files are larger then ever, which means we need more and more storage space. With this in mind, high capacity, reliable backup solutions are even more critical than ever. Tape is a natural backup method because of its long shelf life and durability. It also provides greater transfer speeds than previous methods while constantly offering higher storage capacities. Glyph offers AIT drives in FireWire and SCSI, tabletop and rack-mountable. The AIT line is available in capacities of 35GB, 50GB and 100GB per tape, utilizing AIT-1, AIT-2 and AIT-3 respectively. The AIT format was developed to handle large capacity requirements for data-intensive applications, and works exceptionally well for digital audio and video. It employs a helical scan recording style in which data tracks are written at an angle relative to the edge of the tape. This provides higher density recording, giving you lower media costs and increased efficiency by storing more data on a smaller amount of tape.

Why Backup?

Backing up and archiving your data is easy and extremely important. When you consider how much time you put into your work, and how large your files can be, regular preventative system maintenance is much less painful than re-creating a session or project. In digital audio and video production, you rely on your system to the point that data loss can be devastating. No matter how well you treat your system, you cannot guarantee your data will be safe if it exists in only one place.

A backup is a copy of your data in a separate place from the original, such as a hard drive, tape, DVD or CD. Many people backup their work to tape at the end of the session or on session breaks. Programs, such as Retrospect or Mezzo, allow you to keep track of what has been backed up, and what hasn't. At a glance you can see when the last backup occurred, and what files were copied. As you continue to record, edit or add new files to your working drive, continue to back up that data by running the backup program again.

You may also consider backing up your data on hot swappable hard drives. The working drive and backup drive can be the same capacity, making it easy to keep your files organized on both drives. Some people use the Trip2 as both working drives and backup drives. Record your 64 tracks of 96K 24bit audio to three drives at a time, then backup to the other three at the end of the session.

Tape is often used for longer term backup because it is more likely to survive an accidental drop than a hard drive. Recovering data from a hard drive is not easy, and is expensive.

Optical media may be the most robust for the long term archiving. Optical media, like CD or DVD, provides you more cost-effective options for restoring your data.

Like tape, hard disk drives are magnetic devices. Just as tape can exhibit dropouts or degaussing, a hard drive can crash or have bad sectors and irreplaceable data can be lost forever, unless it has been backed up. The best practice is to regularly back up your projects to tape, and over long periods of time archive your data with optical media such as DVD. Optical media offers a longer shelf life than tape and is usually easier to store on your shelves.

Glyph provides fast, large capacity backup and archiving systems with the option of a FireWire or SCSI interface. SCSI AIT3 offers capacities up to 100GB native, and the FireWire AIT1 and AIT2 tape back up is reliable and extremely easy to use.

AIT SPECIFICATIONS

AIT-1

- 35GB per tape
- Available in SCSI and FireWire
- 240MB/min sustained transfer rate
- SCSI version compatible with Retrospect and Mezzo
- FireWire version compatible with Retrospect

AIT-2

- 50GB per tape
- Available in SCSI and FireWire
- 360MB/min sustained transfer rate
- SCSI version compatible with Retrospect and Mezzo
- FireWire version compatible with Retrospect

AIT-3

- 100GB per tape
- Available in SCSI
- 720MB/min sustained transfer rate
- Compatible with Retrospect and Mezzo



B&H PHOTO - VIDEO - PRO AUDIO

PYRO DV RaidRT



PYRO DV RaidRT is hardware-based FireWire RAID array enclosure designed for DV content designers who edit, create and store large mission-critical video projects. It provides more reliable storage and faster speeds than a single hard drive alone.

PYRO DV RaidRT is a four-bay RAID enclosure that accommodates Ultra ATA 100 hard drives. These drives are converted to FireWire/IEEE 1394 using internal dual channel RAID controllers. PYRO DV RaidRT utilizes an Intel i960 RM RISC Microprocessor and also includes 64 MB SDRAM for cache memory. Once drives are installed, it supports RAID level 0, 1, 0+1, 3 or 5. Installed drives are hot swappable and one drive can also be used as a hot spare.

- Capture 48 Tracks of 24 BIT Audio
- Add your own IDE Hard Drives or ANY Size
- Fully Integrated HARDWARE RAID, No Software to Install
- Stripe Hard Drives for RAID 0, 0+1, 3 or RAID 5
- 4 Removable Hard Drives, Hot Swappable
- Serial communication port enables array controller operation via PC VT terminal controller

MEDEA AUDIORACK LP

Low Profile Rack Mountable
Storage Systems for Digital Audio Workstations



AudioRack LP (Low-Profile) storage systems from Medea are designed to support professional digital audio workstations (DAWs) at a fraction of the cost of competing audio storage systems. AudioRack LP supports high track count sessions in a compact 1U rack mount enclosure.

- Packs up to 300GB of high-performance audio storage in a single 1U enclosure that requires only 1.75-inches of rack space. Stack three AudioRack LPs. and over 1TB (1000 GB) of storage can be configured in only 7 inches.
- The 2-drive versions of AudioRack LP support the simultaneous playback of up to 24-tracks of audio. The 4-drive versions support 64-track sessions of 24-bit/96kHz audio and are the ideal storage solution for professional DAWs including Nuendo, Pro Tools and Tascam MX-2424.
- AudioRack LP is optimized with Medea's proprietary AST (Audio Stripe Technology) to support 64-track sessions in a single SCSI ID.
- AST enables independent seeks to the disk drives in AudioRack LP and dynamically distributes audio tracks evenly among the drives. The bottom line is that you can simply record/playback up to 64-tracks to/from a single AudioRack LP and eliminate forever the need to manually manage your audio media by juggling between multiple storage devices.
- AudioRack LP attaches to any DAW equipped with a SCSI host adapter. Simply attach and initialize the system as if it was a single disk drive.
- As your audio storage requirements grow, AudioRack LP systems can be daisy-chained together for storage capacities up to 4.5 TB (4500 GB)—over 65 hours of 64-track, 24-bit/96kHz sessions.
- AudioRack LP is ultra-quiet and won't interfere with your productions.
- Backed by a 5-year factory warranty

PORTABLE HARD DRIVES

Data Bank



Design by F.A Porsche, the Data Bank is the most compact hard disk ever. Weighing less than 5 oz. the Data Bank is barely larger than a credit card and thinner than a mobile phone 4.4 x 2.5 x 0.5˝). A masterpiece of understated elegance, the Data Bank offers 20GB of storage at fast FireWire and USB 2.0 speeds, in a sleek, ingot-shaped magnesium design.



- Hot-pluggable to easily share among cross-platform desktops, it features one FireWire and one USB 2.0 port
- USB 2.0 standard boasts transfer rates of up to 480Mbps while still maintaining backward compatibility with USB 1.1 devices. The blazing FireWire interface delivers transfer rates of up to 400Mbps.
- Offers true plug and play connectivity, without the need for driver or software installation for Windows XP and Mac OS X users.
- No power supply needed. Simply connect the drive to your computer and use the power supplied by the FireWire or USB cable.

PocketDrive



Designed for today's traveller, the LaCie PocketDrive family offers an ideal solution for data exchange, backup and archiving on the road. Slender and compact, the drives are easy to carry and feature a silicon bumper for maximum shock protection.
The PocketDrive U&I provides both USB 2.0 and Firewire (IEEE1394) interfaces in one drive. With two Firewire ports and one USB port, you can connect and disconnect any of the PocketDrives while the computer is running with immediate device recognition.



- PocketDrives weigh just 12½ ounces, yet they can easily store heavy graphic, audio and video files, and perform complete backups of most systems.
- PC and Mac compatible, the PocketDrive comes complete with USB 2.0 and FireWire cables, AC power adapter, and LaCie's Silverlining Pro drive management software.
- For those who don't require Firewire, the PocketDrive USB 2.0 delivers all the features and performance of the PocketDrive U&I, except Firewire connectivity. This streamlined version allows a more aggressive value without compromising performance.

Mobile Drive



The perfect companion for desktop and notebook computers, the LaCie Mobile Drive delivers up to 40GB storage capacity in a package that fits in your pocket. With a rugged aluminum design to reliably protect your data, and an ultra-thin, portable design to provide complete mobility, the Mobile Drive allows you to move freely and take your data with you wherever you go.

- Plug and play with USB 2.0 interface, this lightweight, portable drive boasts transfer rates of up to 480Mbits/sec. — forty times faster than the previous version — while maintaining convenient backward compatibility with USB 1.1 devices
- Uses the power supplied by the USB bus—so no AC power adapter needed—making it even easier to work when traveling.

D2 HARD DRIVES

Mobile & Desktop Hard Drives

LaCie's D2 series mobile and desktop hard drives are available in a variety of interfaces for easy connection to Macs and PCs. While USB is best-suited for personal backup and data sharing, Wide SCSI and Firewire are ideal for use with high-bandwidth applications, such as video and audio editing. (USB & Firewire drives are Mac OS X and Windows XP compatible).



- The D2 drives feature a sturdy aluminum case designed to enhance portability, durability, and appearance. Made of a metal alloy that dissipates heat and prolongs drive performance, the drives feature a slender, stylish case that allows you to arrange them vertically on the desktop, horizontally stacked in a desk rack, or mounted in 19″ racks.
- Available with up to 500GB of storage, a 250GB D2 drive can store the equivalent of a 385 CD-Juke-Box or 50,000 MP3 songs.
- Fan-free design ensures quiet operation
- The drives are hot-pluggable, allowing for connecting and disconnecting while the computer is running. They are also compatible with Mac and Windows enabling simple cross-platform data sharing and exchange among workstations.

D2 USB 2.0 & FIREWIRE 800 HARD DRIVE

- Equipped with the Firewire 800 standard—the fastest-ever, this drive is capable of reaching transfer speeds of up to 800Mbits/s for Firewire 800 users and up to 480 Mbits/s for USB 2.0 users. The d2's dual interface, 7200 rpm speed and 8MB buffer make it the fastest-possible storage solution for digital content creators.
- With the appropriate cable, the drives can be connected to any computer equipped with FireWire 800 (9-pin), FireWire 400 (6-pin), USB 2.0 or USB 1.1 ports, making it the most universal drive ever.
- Chained and striped in a FireWire 800 RAID 0 configuration, can reach unprecedented sustained transfer rates of up to 100MB/s, making them ideal for workstation and small-server environments.

D2 FIREWIRE HARD DRIVE

- Drives are hot-pluggable, allowing for connecting and disconnecting while the computer is running
- Up to 250GB of additional storage
- Transfer rates of up to 400Mbits/sec
- Automatic plug and play configuration for Mac and PC

D2 USB 2.0 HARD DRIVE

- Spacious storage capacity: up to 200GB of additional storage
- USB 2.0 delivers transfer rates of up to 480Mbits/sec
- Automatic plug and play configuration for Mac and PC

D2 ULTRA320 SCSI HARD DRIVE

- Fastest SCSI transfer rate: up to 320MB/sec
- Ultimate performance: up to 68MB/s sustained rate per drive
- Use with RAID 0/1 arrays, audio, high-speed servers and workstations
- Backwards compatible with Ultra160, Ultra2 and UltraWide SCSI interfaces. It it can even be chained with previous-generation SCSI drives without causing speed limitations.

D2 ACCESSORIES



LaCie Desk Rack
Conveniently stack up to four D2 drives in this separate rack and save valuable space on your desk. When holding three drives or less, this sturdy desk rack features an area to neatly store media such as CDs and DVDs **49.00**

LaCie Security Lock
Also available for use with the D2 design is a Kensington-type lock to which protect your equipment from theft or damage. This security lock can be quickly and easily installed and features a durable steel cable **19.00**





LaCie Rackmount Kit
Configure your LaCie D2 drives in standard 19″ racks with this optional rack-mounting kit. This kit allows for greater organization of storage peripherals and tidy consolidation of hardware. Kit includes two aluminum brackets and one aluminum junction piece.....**39.00**

Included LaCie Drive Stand
LaCie D2 drives ship with a sturdy metal drive stand that provides stability and enables upright desktop use. When positioned vertically, these slender drives occupy less surface area for more efficient use of desktop space.

d2 CD-RW DRIVE

High-Speed, Portable Firewire or USB CD-RW Drives



Quickly creating data, audio, photo and multi-media CDs has never been easier. Enclosed in a lightweight but sturdy aluminum case for complete mobility (AC powered) and reliability, the d2 FireWire or USB-equipped CD-RW drives allow you to quickly and easily record, write and play CDs. Use them to master your own music CDs, or store and archive your important data. They come complete with LaCie CD Utilities including recording software for pre-mastering, digital audio/video archival and storage, and creating backup copies.

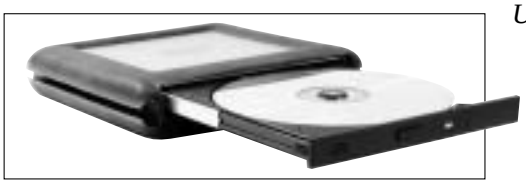
- Ships with recording software, so you'll be able to start burning immediately. The LaCie CD Utilities includes all of the necessary drivers and Toast for the Mac and Easy CD Creator for the PC, allowing you to create custom audio CDs and store files.
- A "plug and play" peripheral, the LaCie CD-RW Drive easily configures to Macs running OS 9.x and 10.x, as well as Windows 98SE, Me, 2000 and XP. No need for device IDs or terminators.
- Hot-pluggable FireWire technology lets you connect and disconnect your drive without shutting down your computer.
- Stackable desktop or rackmount design (optional rackmount kit and desktop rack)

POCKETCD-RW DRIVES

Super-Compact, Portable Firewire or USB CD-RW Drives



Designed for today's traveller, the LaCie's portable, lightweight USB or Firewire-equipped PocketCD-RW Drives are engineered with rugged durability, and designed to enable fast CD recording anywhere, anytime.



Use it to back up your projects while on the road, share information with clients, or shuttle data between home and the office. To guard against the bumps and jolts that come with life on the go, the PocketCD-RW is surrounded by a shock-resistant bumper for reliable data protection.

They Both Feature:

- True plug and play convenience—no driver or software installation required
- Store large audio and video files, as well as perform complete backup of most systems.
- Ideal match for Macintosh PowerBook or iBook computers equipped with iTunes, the drives enable simple data sharing among Macintosh workstations.
- Weighing only 25 ounces, the PocketCD-RW Drive is designed for travel. In addition there is no external power supply to carry and no AC adapter required.
- To guard against the bumps and jolts that come with life on the go, they are surrounded by a shock-resistant bumper for reliable data protection.
- Burn digital video and data files in less than four minutes for fast archival, backup and data exchange. Use iTunes to quickly compile audio and MP3 CDs.
- Featuring advanced FireWire or USB 2.0 technology, the drives can be connected and disconnected while the computer is running with immediate device recognition.

Firewire Version

- Offers extremely fast, reliable performance for users working with digital audio and video. The Firewire PocketCD-RW conveniently utilizes the Firewire cable for both power and data transfer—no external power supply needed.

USB 2.0 Version

- Equipped with the USB 2.0 interface, the USB PocketCD-RW drive delivers transfer rates of up to 480Mbits/sec. while conveniently maintaining backward compatibility with USB 1.1 devices.

D2 DVD-RW

USB 2.0/Firewire DVD Rewritable Drive



Encased in Lacie's d2 design, this DVD rewritable drive features Firewire and USB 2.0 connections and can be positioned horizontally on the desktop, stacked in a separate desk rack, or mounted in standard racks with the optional rackmount kit.

- Record digital audio, video and data or backup valuable data. Read/write to CD and DVD media. Utilize Easy CD Creator and Toast to make audio and MP3 compilations, create and playback your DVDs.
- Fully hot-pluggable, simply connect the drive and start creating. Drives can be shared between workstations for cost savings.
- Assess the needs of your project and then choose the type of media best suited to your needs. Offering the ability to write to DVD-R/RW and CD-R/RW media, allows for true project control.

DVD MULTIDRIVE

DVD-RAM, DVD-R, DVD-RW, CD-R, CD-RW All-in-one-Recorder



Record, archive and share digital content on durable, reliable CD and DVD media with DVD MultiDrives. Equipped with blazing write speeds, these drives allows you to record data, video and MP3s on DVD general use discs, and archive songs, MPEG-2 DVD video or documents. Large-capacity, double-sided media allows for up to 9.4GB of reliable storage space.

- DVD-RAM functionality provides random-access capabilities, similar to a hard drive, allowing for almost instant retrieval of specific files: an excellent solution for backup and fast data recovery.
- Edit and author video, store digital photos and music files, and archive large amounts of data The MultiDrive is capable of reading most CD and DVD formats, and is backward compatible with previous generations of DVD-RAM media
- Bundled with recording software for Mac and Windows including DVD-RAM Utilities, Sonic MyDVD Video Suite for video capture, editing and authoring, Roxio Toast Lite and Easy CD Creator Basic for CD and DVD recording, MusicMatch Jukebox Basic for audio CD compilation, and InterVideo WinDVD for DVD playback.

DVD INTERNAL RECORDER

Internal E-IDE/ATAPI High-Speed DVD Burner



Edit and author professional, interactive video DVDs, store music files and archive digital photos — all on reliable DVD media. Record data, video and MP3s on DVD general use discs and store up to 4.7GB of data — up to 1,000 songs, more than two hours of MPEG-2 DVD video, or as many as 500,000 documents.

- Store up to 1,000 MP3 files on a single DVD disc
- Buffer under run protection reduces failed burns
- Bundled software incorporates easy-to-use interfaces and a range of professional-grade tools, allowing you to create superior, high-quality DVDs to be used for storage, backup or playback
- Ideal for storage and backup of audio, video and data in consumer and corporate environments.

POCKETDVD-RW

Mobile DVD Recording



This lightweight, mobile DVD-RW drive offers enough capacity to store large graphic, audio and video files, as well as perform complete backup of most systems on reliable DVD media. An ideal companion for Apple Power Mac computers and PowerBook laptop computers, LaCie's PocketDVD-RW Drive delivers a complete, lightweight DVD recording and authoring solution.

- Record digital video, MP3s and data files on DVD general use discs. Offering generous storage capacity of up to 4.7GB, DVD general use discs are capable of storing as many as 1,000 MP3 songs, up to two hours of MPEG-2 DVD video, thousands of digital still images, and as many as 500,000 documents.
- Ships with full-featured DVD authoring software, as well as CD and DVD recording software for Mac. A wealth of professional-quality tools to burn audio or author exciting video DVDs.
- FireWire powered, the PocketDVD-RW Drive can be connected and disconnected while the computer is running with immediate device recognition. An external power supply is not needed.

PINNACLE SYSTEMS

INSTANT COPY

Perfect CD and DVD Duplication for Music



Making copies of your DVDs and CDs for personal use has never been faster or easier. Featuring a friendly interface and revolutionary duplication technology (advanced buffer underrun protection, predefined burning profiles, sophisticated read-error handling), Instant Copy burns perfect copies every time— and at unbeatable speeds!

Burn true 1:1 copies of DVD content, transfer DVD content to Video CD's and make perfect bit-for-bit copies on any CD media. (Doesn't copy CSS-encrypted discs.)

- Burn a 9 GB, dual layer DVD video onto a standard 4.7 GB recordable disc
- Transfer DVD content to VCDs and SVCDs that can be played on most DVD players
- Create perfect bit-for-bit copies of any CD media: audio, data, video and music

CLEAN & CLEAN PLUS

All-in-One Audio Restoration



The perfect digital tool for archiving and preserving your old vinyl LPs, cassettes, and analog media, CLEAN allows you to record your favorite albums, and automatically apply a number of effects and filters to remove all of the artifacts of age.

Hiss. Crackle. Scratches. Pops. These are the enemy. CLEAN uses advanced audio restoration technologies to scour your recordings and deliver results as crystal clear as the original recordings. CLEAN PLUS adds a specialized pre-amplifier to help you get the highest quality into your PC.

- Removes clicks, pops and background noise from any recording
- Fully automatic process for great results, quickly and easily
- Unique processing presets for LPs, tape and MP3 based music
- Includes CD burning software for easy cataloging on disc
- Complete parameter control over restoration process

MYMP3PRO *MP3 Jukebox Software*



The ultimate tool for cataloging and controlling your music, myMP3PRO combines high quality MP3 and MP3 Pro encoding with professional level audio editing, mixing, special effects and CD burning capabilities to deliver incredible control over your music library. Capture and store all of your favorite music into custom playlists, add custom EQ processing and other cool effects, then save your mixes to CD, or interface with your portable MP3 player for complete flexibility.

- myMP3PRO delivers everything you need for an amazing audio experience. Encode, edit, mix, alter, reorder and transfer your music. Then upload directly to your hardware MP3 Player, or DJ your own mixes live! Total MP3PRO support gives you the best compression without sacrificing sound clarity. Virtual effect slots allow for incredible sound processing, and integrated Surround-Sound gives you complete control over your music.
- myMP3PRO includes many features found only in high-end audio editing suites. Sophisticated EQ capabilities allow you to create custom playlists specifically for different speaker or headphone setups. Professional waveform editing tools allow you to adjust volume, phase, panning, dynamics and more!
- Packed with features to give you complete command of your audio library. Listen to, organize, burn and convert MP3 files. Fade in, fade out, crossfade, and normalize your tracks. Add audio effects in real time, or even create your own CD labels and covers with the built-in Label Editor. Whether you want to take your favorite music on the road with your MP3 player, or DJ your own parties, there's nothing like myMP3PRO for making and mixing MP3s!



TOAST 5 TITANIUM



CD and DVD-Recording Software for Mac

Toast 5 Titanium is a comprehensive, all-in-one DVD and CD burner software that enables you to create, organize, share and store all of your digital content (music, data, photos, and video) on CDs or DVDs. Turn scratchy LPs and cassettes into crystal clear CDs. Convert MP3s into professional audio CDs or burn them on MP3 discs that hold hours of music. Print custom labels and cases for your new CDs. The perfect companion to iMovie, iTunes, and DVD authoring software—Toast 5 Titanium burns fast, burns clean, burns on more drives, and burns in the background too.

- Automatically converts any QuickTime audio file to CD audio format, including WAV, MP3, and even audio tracks from QuickTime movies.
- Vary the gap time between songs, with a different pause between each song. You can also create MP3 CDs that hold hours of music.
- Design and print expressive CD labels and jewel case inserts with Magic Mouse Discus.
- Included Panic's Audion lets you organize, edit and get the most out of your music collection. Play your MP3 files with rich visualizations, organize them into playlists, or edit them using waveforms to turn your MP3s into personalized compositions.
- CD Spin Doctor turns scratchy records or cassette tapes into high fidelity CDs:
  - — Records from any analog source (phonograph, tape, or microphone)
  - — Eliminate noise, pops, hisses, and clicks
  - — Enhancement filters convert mono to stereo and boost Bass or Treble
  - — Wave form display with zoom in/out

Toast With Jam



Toast with Jam is the ultimate studio tool for arranging and mastering pro audio CDs on Mac OS X. In addition to the full CD and DVD burning power of Toast, this suite adds the award-winning BIAS Peak LE and Roxio's Jam 5.

Jam 5: Arrange your tracks and build your 100% Red Book-compliant audio CD. Create dynamic crossfades between tracks, edit PQ subcodes, set the gain for each channel of each track, cleanly trim your tracks, and more.

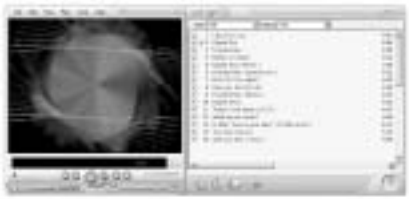
BIAS Peak LE VST: Use lighting-fast, nondestructive waveform editing tools to improve and add special effects to your music. Enhance vocals with concert reverb, soften dynamic peaks and create other effects with included VST plug-ins.

Easy CD & DVD Creator 6

A complete digital media suite, you can rip, play, edit, organize and burn your digital music with AudioCentral. Make DVD movies with professional transitions and animated menus with DVD Builder. Create personalized disc labels with Label Creator. Save large data projects to multiple discs with Creator Classic or copy personal discs with Disc Copier.



- AudioCentral software includes a player, media manager, ripper, tag and sound editor - integrated in a single application. Create greatest hits CDs, preserve your old LPs and cassettes in digital format, enhance your tracks with sound effects and more.



- Drag and drop files for burning to CD or DVD. Disc formatting and burning will happen automatically.
- Back up, share or archive your important files to CD or DVD, or large projects to multiple discs. Easily retrieve files or projects back to the original or new location on my Windows PC with Roxio Retrieve.
- Roxio Label Creator lets you make personalized CD labels, jewel case inserts and DVD case covers. Choose from dozens of professional designs for all occasions!


emagic
Logic 6


dp
dp


2.2
SONAR


PRO TOOLS
PRO TOOLS

B&H

PHOTO - VIDEO - PRO AUDIO

The ProfessionalAudio SourceBook

Section 11b

Integrated Systems

Cakewalk.....**892-903**

Digidesign.....**904-947**

Emagic.....**948-971**

MOTU.....**972-993**

Steinberg.....**994-1031**

Cool Breeze.....**1032-1033**

CAKEWALK

SONAR 2.2 & SONAR XL 2.2



Digital Multitrack Recording System

SONAR 2.2 is Cakewalk's flagship audio and MIDI production sofware for Windows Me/2000/XP. SONAR 2.2 offers unlimited MIDI and audio tracks, realtime 32-bit DirectX8 effects and DXi soft synths and powerful Acid-style loop construction all tied together with automated mixing of virtually every parameter. SONAR provides support for 24-bit/96kHz audio hardware using ASIO and Windows WDM drivers, offering latencies as low as 1.5ms. Almost all of SONAR recording, precision editing, mixing and routing functions can be accessed directly from a single window known as the Track View. This elegant and intuitive interface combined with SONAR's extensive built-in control surface support delivers an exceptional level of speed and efficiency that is sure to optimize your creative potential and workflow. SONAR 2.2 ships with a suite of essential DirectX8 effects plug-ins along with three powerful DXi soft synths including: Cyclone, an Acid-compatible Groove Sampler; as well as Virtual Sound Canvas 1.5 and DreamStation 2.0 synthesizers. Also available is SONAR XL 2.2 which adds an advanced DXi drum sampler workstation and 64-bit DirectX 8 audio-mastering effects to the plug-ins already included with SONAR 2.2.

FEATURES

- Integrated multitrack recording, editing, loop construction, automated mixing, and delivery of audio and MIDI

Audio Support

- Simultaneously record multiple audio tracks with support for multiple multi-channel hardware up to 24-bit/96 kHz.
- Unlimited audio tracks and real-time effects inserts, 16 auxiliary busses, and up to 64 virtual main outputs.
- WDM and ASIO driver compatibility (Windows 2000/XP) supports latency as low as 1.5ms and provides extremely fast audio processing and mixing including live input monitoring of effects and immediately responsive DXi soft synths.

MIDI Support

- Unlimited MIDI tracks and MIDI I/O ports
- Internal MIDI precision of 960 PPQN for rock-solid synchronization, recording and editing accuracy
- Real-time MIDI FX plug-ins for non-destructive MIDI editing.
- Precise multitrack Piano Roll, Notation, Event List and Drum Grid MIDI editing.

Loop Construction

- Support for beat slicing and import/export of ACIDized Wav file as well as live performance of loops using the included DXi soft Synth Cyclone.

Audio/MIDI Editing

- Non-destructive slip editing (trimming) of audio and MIDI data in the Track View
- Unlimited undo/redo with history for all editing functions.

Included Soft Synths and Effects Plug-ins

- Includes 3 DXi soft synths (4 in SONAR XL), 16 realtime DirectX audio effects (18 in SONAR XL) and 14 MIDI FX plug-ins.

Control Surface Support

- Direct support for the Mackie Control, CM Labs MotorMix, Tascam US-428 and US-224 as well as the Radikal Technologies SAC-2.2 provide dedicated real-time, hands-on control of your mix, automation, effects, and more.
- A Global Control plug-in features a learn-mode that allows you to quickly integrate other MIDI-compatible control surfaces.

Import/Export Audio

- Import multiple file formats: AIF, ASF, AU, AVI, BUN, MID, MP2, MP3, MPEG, MPG, SND, WAV, WRK
- Individual tracks or entire mixes including automation, plug-ins and DXi soft synths can be bounced to a new track within the current project or exported to disk
- Export as WAV, ACIDized WAV or export audio to AVI, Quicktime or MPEG video.
- Encode audio to MP3 with a 30 day trial version of Fraunhofer MP3 encoder, RealAudio G2, and Windows Media ASF.
- Import/Export of OMFI (Open Media Framework Interchange) and Broadcast Wave files — the film, video and audio industry's standards for saving and transferring digital post-production projects between different workstation platforms.

Synchronization

- Frame-accurate SMPTE synchronization with variable frame rates and auto-detection of incoming timecode
- Able to send/receive MIDI clock, receive MTC (MIDI TIME CODE) and send MMC (MIDI Machine Control) commands

SONAR 2.2 & SONAR XL 2.2

SONAR's streamlined Track View is the heart of your recording and editing environment. What's great about this view is that you can do all of your recording, arranging, mixing, and most (if not all) of your editing in one place. This really maximizes your creative workflow.

A number of tools are available for editing audio and MIDI clips including split, scrubbing, zooming, snap grid settings

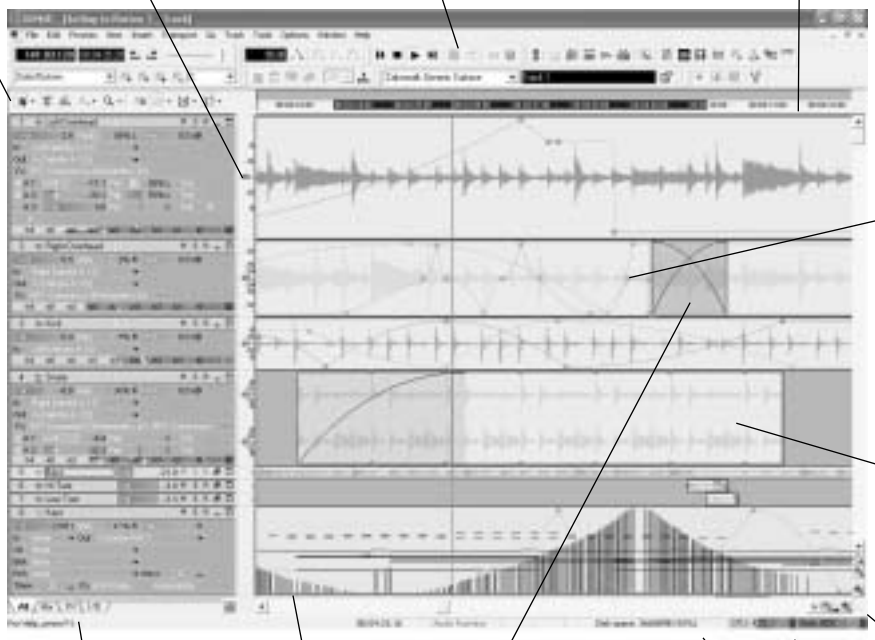
Audio Scale Ruler is a vertical zoom tool for audio clips that allows detailed editing of quiet passages

The customizable Toolbar ensures that the functions you need the most (transports, looping, tempo, markers, record mode, synchronization and more) are always a mouse click away. You can click on any toolbar and drag it to any location on the Track View.

The time ruler is switchable between Measures/Beats/Ticks, samples, Hrs/Mins/Secs/Frames. The shaded area represents loop and punch points

Track Pane

A dedicated channel strip for each audio and MIDI track allows you to control mixing, automation and effects without leaving the Track View.
Each channel strip can be individually resized allowing you bring into focus the specific track(s) you're working on.



The Clips Pane is where you record and arrange Audio, Groove and MIDI clips

Automation data including mixing and effects for Audio and Groove clips can be freely drawn and edited with sample accuracy using a choice of 4 shapes – linear, fast and slow curves as well as jump (instantaneous change).

Audio Track Controls Include:

Volume, Pan, Input Trim
Mute, Solo, Record arm
Input/output assignment
Effects inserts (insert, bypass)
Aux sends (level, pre/post assignment, pan, bypass)
Polarity invert
Mono/Stereo assignment
Input/Output Meter (switchable peak, RMS and peak + RMS with clip indicator and Hold function)

Beat–matched Audio Looping

Simply click and drag loops (Grooveclips or ACIDized .WAV files) where you want them and make tempo and pitch changes to audio in real-time. Click and drag the left or right edge of a groove-clip to have a loop repeat. You can even combine different loops on the same track.

MIDI Track Controls Include:

Volume, Pan
Velocity + (Velocity trim)
MIDI input/output channel assignment
MIDI port assignment
Bank and Program selection with patch lists provided for all major synths
MIDI effects inserts (insert)
General MIDI compatible
Chorus and Reverb send controls
Key + (transpose)
Time + (time offset)

These tabs allow you to quickly choose which items are to be displayed in the track pane (all parameters), Mix (volume, pan, trim), effects, I/O assignment

Crossfades can be created automatically between overlapping audio clips using one of 9 available crossfade presets.

Controller, RPN and NRPN data for MIDI clips can be edited in the Track View can be freely drawn using the same tools available to Audio clips
Controller data edited in the Piano Roll or Event List Views are also displayed with each MIDI clip

The Status Bar

Provides realtime display of the cursor location, whether the audio engine is running, how much disk space is available as well as global Mute, Solo and record status for the project.
The CPU and hard disk meters allow you to monitor the status of your system's available resources.

The zoom tools provide horizontal and vertical zoom down to the ample level allowing detailed audio editing directly from the Track View

Audio Loop Construction Toolkit

- Use the flexible beat slicing and time stretching/pitch shifting tools in the Loop Construction View and incorporate perfectly beat matched loops alongside your standard audio and MIDI tracks.
- Make tempo and pitch changes to audio in real-time.
- Combine an infinite number of different loops on the same track.
- The Loop Explorer allows you to quickly find, organize, and audition loops in real-time – even during playback.
- Import and export ACIDized WAV files, or create loops from any recorded audio, WAV, MP3, or AIFF file.
- Click and drag to lay down loops when and where you want them.
- Includes hundreds of license free loops.



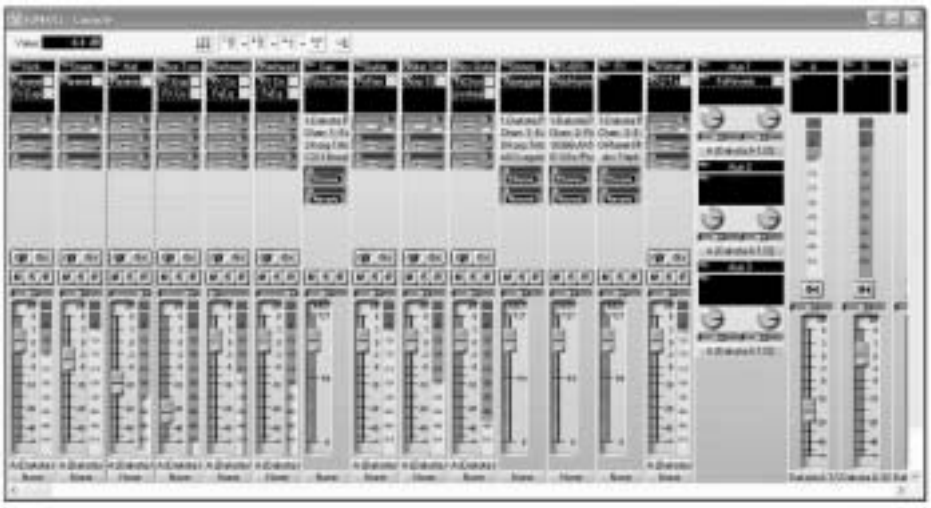
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Console View

Although you can control a complete mix directly from the Track View window, the Console View provides an alternative for mixing and automating audio and MIDI tracks using a familiar hardware-style mixing interface.

- Console View channels include audio, MIDI, auxes, virtual mains (master outputs).
- Each audio and MIDI track includes level, pan, mute, solo, record arm and comprehensive metering as well as access to plug-ins, auxes, busses and I/O routing.
- A pull down menu lets you to decide which audio and MIDI channels are to be displayed.
- Automation created in the Console View is automatically reflected as envelope automation in the Track View and vice-versa.
- Remote control of all mixing parameters are available via MIDI.



PLUG-INS

Fully-Automatable DirectX 8 Realtime Audio Effects

- Includes the following fully-editable, fully-automatable 32-bit DSP-FX audio plug-ins designed by Power Technology — StudioVerb, Parametric EQ, Delay, Chorus and Flange
- Automation can occur directly from the effect's interface, via MIDI or for an even higher level of precision, by drawing vector envelopes in the Track View
- Additional 32-bit, non-automatable effects include: EQ, Expander, Gate, Limiter, Tape Saturation, Pitch Shifting, and more
- Real-time, live input monitoring of effects is possible when using audio hardware that supports WDM drivers

Expand Your Suite Of Audio Processing Tools

- Hundreds of additional DirectX-compatible audio plug-ins are available from industry-leading third party developers including Antares and Waves







Alien Connections – ReValver – Virtual Guitar Amp Systems

- Combining a wide variety of realistic pre-amp models with EQ, power amps and cabinets with post- EQ, and effects that can be routed in virtually any order — provides nearly endless possibilities
- Automatable parameters allow you to switch or morph between different guitar tones within one track



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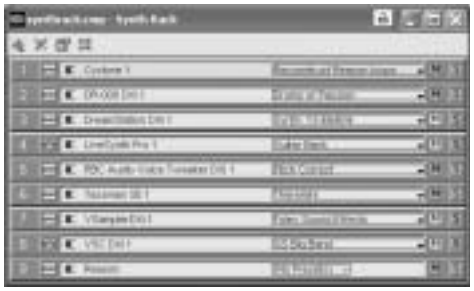
EQUIPMENT LEASING AVAILABLE

SONAR 2.2 & SONAR XL 2.2

DXi (DirectX Instruments)

- DXi is the open standard for soft synth plug-ins, based on Microsoft's DirectX technology
- DXi synths are CPU efficient with low latency — unlimited DXi synths can be loaded simultaneously (CPU dependent) with full automation and multiple output capabilities
- Three DXi instruments are included while additional DXi synths are available separately from leading developers including Native Instruments, IK Multimedia and others

Synth Rack



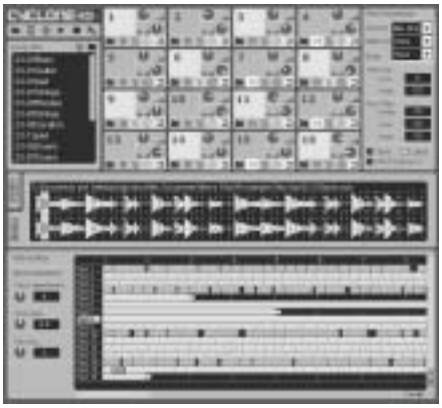
- The Synth Rack View allows you to easily insert instruments or ReWire devices into your project. It makes loading and working with synths in your project a snap – host Audio and MIDI tracks with proper I/O assignments are created automatically when an instrument is loaded
- View all of the loaded DXi synths at a glance along with their current presets

Audio Simulation – DreamStation 2.0 DXi Polyphonic Analog Synth



- Accurately reproduces analog synth sounds by modeling analog circuits and oscillators.
- Combines subtractive synthesis with FM synthesis.
- 3 oscillators, multimode self oscillating IIR filter, VCA, LFO, user EG for each voice.
- Waveforms include sine, triangle, sawtooth, pulse, square and noise.
- Filter types include 12dB/Octave Lowpass, Highpass, Bandpass, 24dB/Octave Lowpass and Formant.
- Up to 16 voices polyphony.
- Linear frequency modulation, Hard synchronization, Ring modulation, Distortion.
- Supports sample rates up to 96kHz.

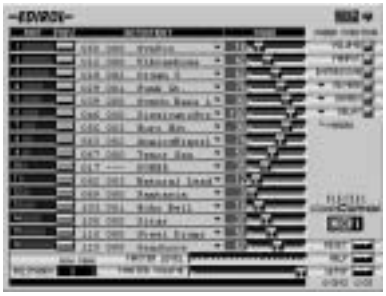
Cyclone DXi Groove Sampler



- Cyclone is a 16-part groove sampler, composition tool, and loop editor wrapped up in a single DXi synth – Load up to 16 loops (ACIDized .Wav or SONAR Groove Clip) or samples onto the 16 trigger pads and create loop-based compositions in real-time using any MIDI device, MIDI tracks, computer keyboard, or mouse.
- Samples will automatically match the tempo and pitch of your project.
- Rearrange, combine, slice, tweak, and customize samples to create new grooves.
- Control gain, pitch, and pan of individual loop slices.
- Fully adjustable loop points allow you to create unique polyrhythmic textures
- Supports multiple outputs & key mapping
- Export combined loops as a Groove clip that can be used in SONAR & other apps
- Snap to grid with assignable resolution.
- Slice preview

EDIROL – Virtual Sound Canvas 1.5 DXi

- Modeled after one of the world's most popular sound modules
- 16-part multi-timbral up to 128-voice polyphony
- GM2/GS compatible – 902 tones plus 26 drum sets
- Reverb, chorus and delay effects



ReWire 2.0 Support

- Patch Cakewalk's own Project5, as well as Propellerhead's Reason, ReBirth, and other ReWire 1.0 and 2.0 compatible devices right into the Synth Rack.
- Route ReWire channels into audio tracks and process with DirectX effects
- You can even use MIDI FX, automation, and sequencing capabilities to control Reason
- Add VST instruments and effects with an optional VST to DX converter.

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SONAR 2.2 & SONAR XL 2.2

MIDI Processing and Editing

- Quantize from a whole note to 32nd note triplets with multiple parameters including strength, swing, window and offset
- Groove Quantize functions include importing DNA grooves and extracting timing info from audio files
- Additional MIDI processing functions include transpose, slide, scale velocity and retrograde (reverse)

Multitrack Event List

- Provides a list of all selected events occurring in all selected tracks
- A display filter lets you to specify the data type(s) to be viewed and edited (notes, pitch bend, controller, etc...)

Multitrack Piano Roll

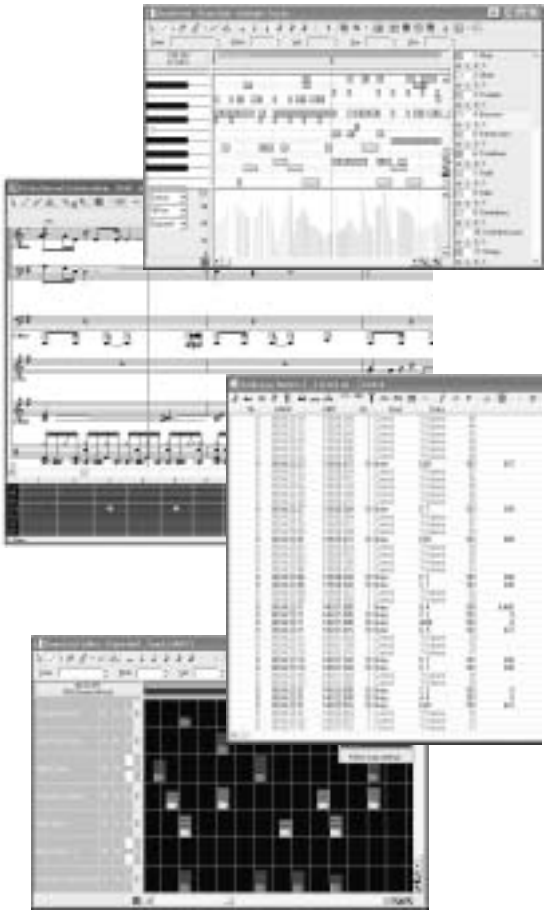
- The multitrack piano roll is a flexible and intuitive display that lets you select, and edit multiple tracks of MIDI notes and controller data simultaneously with a 64th note grid line resolution
- Includes monophonic and polyphonic Pattern Brush tool that allows you to paint complex MIDI parts

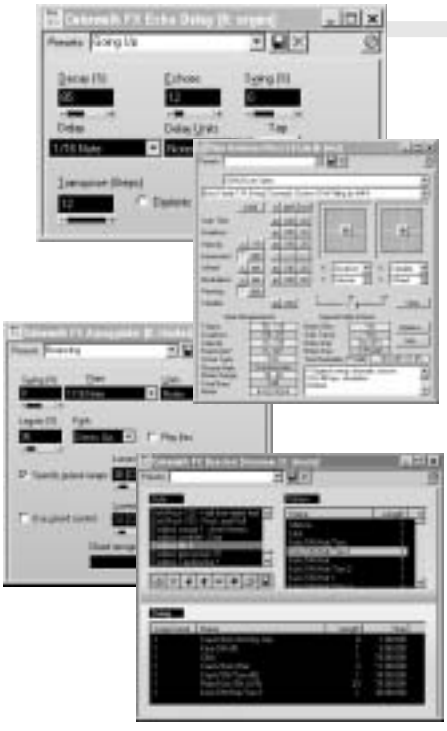
Drum Grid Editor

- Paint rhythmic patterns in a single mouse stroke using hundreds of included Smart Loops patterns, or create your own
- Custom drum maps – preview and create kits with real-time remapping across multiple MIDI devices, multiple outputs as well as DXi synths

Create Scores Fast

- View and edit MIDI events in standard notation, then print sheet music of complete arrangements or individual parts in sizable fonts
- Print 24 staves per page, along with lyrics, chord grids, percussion notation, dynamic markings, and enharmonic spellings
- Guitar tablature with customizable fretboard display
- A dedicated Lyrics view displays scrolling text during record or playback





MIDI FX plug-in library

- Non-destructively edit MIDI tracks with 14 included real-time MIDI FX
- Real-time MIDI plug-ins can be used on input, applied in as track inserts in the Track or Console view, or used as off-line commands

Session Drummer —

- Create and arrange drum patterns in real-time, selecting from a library of professional MIDI drum performances or your own MIDI drum tracks

NTONYX Style Enhancer Micro 2.0 Lite —

- Uses performance modeling in real-time to give MIDI recordings a more "human" feel

SlicyDrummer Lite—

- Instantly create drum loops by mixing and matching prerecorded patterns for each individual drum instrument

MusicLab Rhythm'n'Chords 2.0 Lite—

- Easily create MIDI rhythm guitar parts – Enter chord symbols, select a rhythm pattern and your guitar part is ready

MusicLab VeloMaster Lite —

- Real time MIDI dynamics processing of note velocity using an interactive graph.

MusicLab Looper —

- Create and edit MIDI loops – simply select the section a part of the track you want to loop, and drag it into the Looper window
- Quickly find and audition loops from your MIDI loop libraries.

MusicLab Fixed Length —

- Quickly set note lengths for your MIDI tracks – a compressor/expander for note duration

Plus the following additional plug-ins:
Arpeggiator, Quantize, Delay/Echo, MIDI Filter, Transpose, Velocity and Chord Analyzer

SONAR 2.2 & SONAR XL 2.2

SONAR XL

SONAR 2.0 XL provides all the capabilities of SONAR plus two 64-bit, fully-automatable DirectX 8 mastering effects, and an advanced DXi soft synth drum sampler.





- The Timeworks Equalizer is a fully-automatable, 64-bit mastering EQ with a integrated 30-band spectrum analyzer and phase meter. Both clean and vintage EQ algorithms, high and low shelf filters with resonance, 6 bands of bell filters, sweepable filter points, and adjustable Q
- The Timeworks CompressorX is a pristine sounding, versatile, 64-bit, fully-automatable mastering-quality compressor. It features hard or soft-knee compression, no-clip brick wall limiting, ultra-fast attack/release times, and analog-style metering
- The DR-008 is a powerful DXi drum sampler and synthesizer that provides advanced analog drum modeling (808, 909, etc...), flexible 32-bit sample playback (Wav, Aiff, LM4, etc...) along with pattern-based sequencing. Other features include 96 virtual drum pads for loading samplers and drum models, 8 stereo outputs, 8 aux outputs and a variety of editing functions for each pad. Six sample-based drum kits are included.
- Support for Digidesign audio hardware, including the Digi 001, 002, Mbox, and Pro Tools HD products using ASIO drivers under Windows XP.

Sonar 2.2 and XL 2.2 System Requirements	Minimum	Recommended
Operating System	Windows Me/2000/XP	2000/XP
Processor Speed	500 MHz	1.2 GHz
RAM	64 MB	256 MB
Hard disk space	100 MB for full program installation	
Hard disk type	Any	EIDE/Ultra DMA (7200 RPM) or SCSI Hard Drive
Graphics (resolution; color depth)	800 x 600; 256 colors	1024 x 768; 24-bit color
MIDI Interface	Windows-compatible	Windows-compatible
Sound Card [1]	Windows- compatible	WDM compatible [2]

1. Required for audio playback 2. Requires Windows Me/2000/XP

Pyro 2003 – Complete MP3 and CD Maker

Pyro provides an integrated environment that allows you to play, organize, edit, and restore your digital music, create audio and data CDs and download their music to a portable player. Pyro allows you to make MP3s, burn CDs, digitize and clean old LPs or cassettes, archive important data files and more — all within one centralized user interface.



- Quickly turn your CDs into MP3s, WAV or WMA files
- Quickly locate and organize all the music files on your PC.
- Create unlimited MP3s
- MP3 files open automatically in Pyro's unique preview display waveform within seconds
- Convert LPs and cassettes into CDs, MP3s, WAV, WMA files.
- Auto-recorder with timer for recording LPs, cassettes and internet broadcasts.
- Rip individual songs or entire albums
- Split long audio files into multiple tracks
- Fully-integrated 64-bit EQ as well as click and noise removal.
- Integrated audio editor with waveform preview: non-destructive audio editing; edit fades, crossfades & volume visually
- Automatic crossfades and volume normalization between overlapping tracks.
- Instantly download song titles and artist info from Gracenote/CDDB.
- Portable music player support
- CD labeler kit included
- Back up your valuable PC files to data CD

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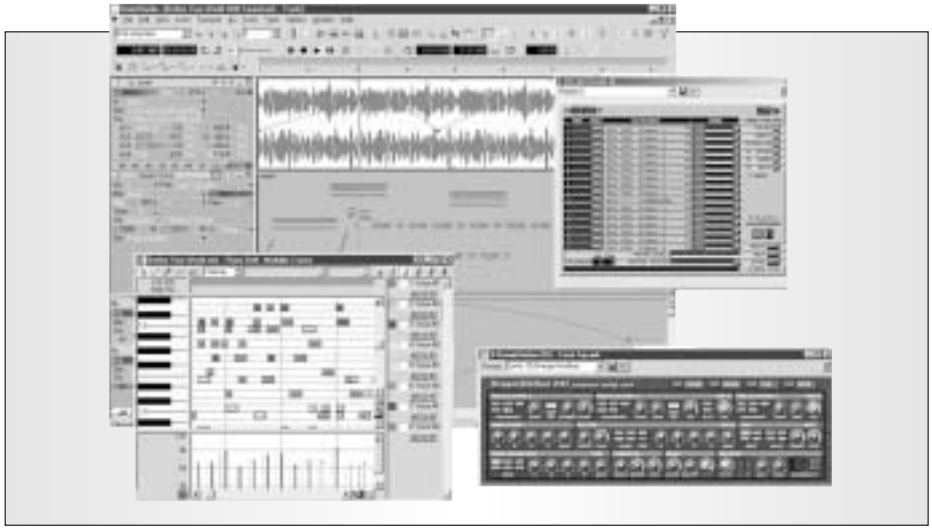
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HOME STUDIO 2004 & HOME STUDIO XL

Affordable Audio and MIDI Recording Software



Cakewalk's Home Studio 2004 and Home Studio XL are simple yet powerful recording tools, for the entry level musician and songwriter, based on Cakewalk's SONAR next-generation audio and MIDI technology. Home Studio 2004's intuitive Track View window lets you quickly record, edit, arrange, mix and automate unlimited audio and MIDI tracks in a single window. Acid style loop creation allows you to create and modify loops to match the tempo and pitch of any music project. A CD featuring an ACIDized loop library is included. Both programs include a number of DirectX audio effects and DXi Soft Synths, as well as MIDI effects with support for third party plug-ins. Both Home Studio 2002 and Home Studio XL are compatible with Windows 98/98SE/Me/2000/XP.



- Record and play back unlimited audio and MIDI tracks.
- Support for 16- and 24-bit audio up to 48kHz
- 960 PPQ MIDI resolution
- Real-time mixing console view
- 16 simultaneous real-time audio effects plus 16 simultaneous real-time MIDI effects
- 2 Auxiliary busses and 1 virtual Main Buss
- DirectX automation support
- Import .Wav, ACIDized .Wav and MP3 files and export to .Wav, MP3 and Real Audio G2
- Loop-based song construction tools with automatic tempo and key matching of Groove-Clips and ACIDized .Wav files
- Non-destructive slip-editing of audio and MIDI clips
- Unlimited number of edit undos
- Audio and MIDI envelope automation of volume, pan, aux sends and MIDI controller data
- Bounce down individual tracks or an entire mix with automation and effects
- Professional MIDI editing including multitrack piano roll view and SysEx editing.
- Integrated DXi Soft Synth Plug-ins include Virtual Sound Canvas and DreamStation with support for third-party plug-ins.
- Includes 6 realtime audio effects including Amp Sim Lite, Reverb, Chorus, Delay, Flange, EQ with support for hundreds of available third-party DirectX plug-ins.
- 5 realtime MIDI effects – Arpeggiator, Quantize, Delay/echo, Change Velocity, MIDI event filter, Session Drummer, NTONYX Syle Enhancer Micro 2.0 Lite and MusicLab VeloMaster Lite.
- Import and synchronize AVI video to music projects
- Compose, edit and print sheet music, along with chord symbols, guitar chord grids, lyrics, dynamic markings
- Remote Control (for MIDI Gear)
- Includes a complete library of ACIDized audio loops as well as Fruityloops Express loop generator.
- Burn CDs and create MP3s with trial version of Pyro.

Home Studio XL Adds—



- FXPansion DR008 SE Drum Sampler
- An extra CD containing over 600 loops from the Smart Loops collections, also including dance and techno loops from PowerFX and X-Mix
- AFX1 - Total Dynamics Processing - A complete home studio mastering suite: Compressor/gate to keep levels similar, a limiter to keep your volumes below a set threshold, and an expander/gate to make loud parts louder and soft parts softer.

SOFTWARE COMPARISON CHART

Cakewalk Product Comparison	SONAR 2.2 XL	SONAR 2.2	Home Studio XL	Home Studio 2004
Number of Audio Tracks	unlimited	unlimited	unlimited	unlimited
Number of MIDI Tracks	unlimited	unlimited	unlimited	unlimited
Number of real-time audio effects	unlimited	unlimited	16	16
Number of real-time MIDI effects	unlimited	unlimited	16	16
Control surface Support	Yes	Yes	Limited	Limited
Synth Rack	Yes	Yes	No	No
Multi-port Synths	Yes	Yes	No	No
DXi 2.0 support	Yes	Yes	No	No
Rewire 2.0 support	Yes	Yes	Yes	Yes
Advanced Project Management	Yes	Yes	No	No
Cyclone DXi	Yes	Yes	No	No
Yamaha OPT Panels Support	Yes	Yes	No	No
Multi-port drum editing	Yes	Yes	No	No
# of Aux Buses	16	16	2	2
Virtual Main Buses (Stereo Outputs)	64	64	1 at a time	1 at a time
Direct X Automation Support	Yes	Yes	Yes	Yes
Direct X Automatable Plug-Ins	Yes	Yes	No	No
Audio FX 1 Plug	Yes	Yes	Yes	No
Audio FX 2 Plug	Yes	Yes	No	No
Sonic Timeworks EQ	Yes	No	No	No
Sonic Timeworks Compressor X	Yes	No	No	No
Import ACID .wav files	Yes	Yes	Yes	Yes
Groove-Clip	Yes	Yes	Yes	Yes
Loop Construction View	Yes	Yes	Yes	Yes
Assorted ACID-format Loops	Yes	Yes	Yes	Yes
Unlimited Undo	Yes	Yes	Yes	Yes
Bounce Down Tracks	Yes	Yes	Yes	Yes
Audio Envelopes	Yes	Yes	Yes	Yes
MIDI Envelopes	Yes	Yes	Yes	Yes
SMPTE/MTC, MIDI sync, MMC	Yes	Yes	No	No
Professional MIDI Editing	Yes	Yes	Yes	Yes
Virtual Sound Canvas DXi Soft Synth	Yes	Yes	Yes	Yes
DreamStation DXi Soft Synth	Yes	Yes	Yes	Yes
Live Synth DXi	Trial	Trial	Trial	No
DR008 DXi Drum Sampler	Yes	No	Yes	No
Fruityloops 2.7 Express	Yes	Yes	No	No
FX Pad (DX8 plugin)	No	No	No	No
Real Time Audio Effects	18	18	6	6
Real Time MIDI Effects	15	15	5	5
Multiple Monitor Support	Yes	Yes	No	No
Studioware Panels	Yes	Yes	Yes	Yes
Multiple Sound Card Support	Yes	Yes	No	No
Lyrics and Markers View	Yes	Yes	Yes	Yes
Video	Yes	Yes	Yes	Yes
Compose and Print Sheet Music	Yes	Yes	Yes	Yes
Sysx Editing	Yes	Yes	Yes	Yes
Audio Support	up to 24-bit	up to 24-bit	up to 24-bit	up to 24-bit
Max Sample Rate	96 kHz	96 kHz	48 kHz	48 kHz
Remote Control (for MIDI Gear)	Yes	Yes	Yes	Yes
CAL Programming Language	Yes	Yes	No	No
Fraunhofer MP3 Export	limited trial	limited trial	limited trial	limited trial
Export to WAV, Real Audio G2	Yes	Yes	Yes	Yes

CAKEWALK

PROJECT5

Soft Synth Workstation



Project5 is a complete virtual studio that combines a suite of DXi compatible soft synths and samplers with sequencers, audio and MIDI effects and Acid-compatible audio looping tools — all in a single integrated production environment. The Project5 soft synths include a polyphonic analog-modeling synth, multi-format compatible professional sampler, drum synthesizer, sampling drum machine as well as the coveted Cyclone DXi groove sampler. The Project5 sequencers allow real-time and step-recording, as well as piano-roll style editing and step sequencing. Project 5's live performance audio engine provides you with realtime control over the soft synths and samplers with advanced, user-configurable synth layering and keyboard mapping, as well as the ability to trigger patterns in realtime. Project5 also allows you to customize and expand your studio's arsenal with support for additional soft synths, audio and MIDI effects, sample libraries. Everything in Project5 is automatable! That means that you can automate every mix, synth, and effect parameter. Make changes on screen or use learn mode to quickly integrate your favorite hardware surfaces and MIDI devices. Project 5 will work with any WDM, MME, or ASIO-compatible Windows hardware, and also sends MIDI Sync to outboard sequencers. In addition, Project5 instruments and effects will integrate directly into SONAR. Project5 comes packed with nearly 2 CDs of professional samples and ACID-format loops from Q-Up Arts, FXpansion Audio, ProSamples, and Smart Loops.



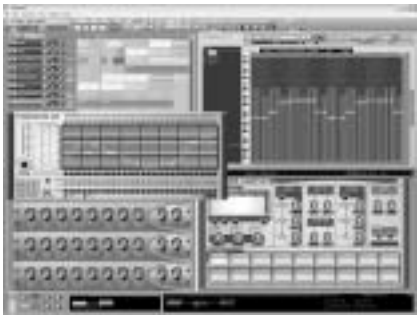
P-SEQ - Pattern Sequencer

- Easily create and edit music and automation using an intuitive "piano roll" style pattern sequencer that superimposes track automation with note data.
- Record using step and real-time modes
- Pattern preview, auto-looping, and a range of editing tools allow you to tweak each pattern to perfection.
- You can automate track and MIDI data as well as any DXi or audio effect parameter.
- The Pattern Bin lets you browse, organize, and audition multiple patterns. Patterns can be dragged to the Pattern Editor for additional editing.
- All data can be treated as a pattern and automation can be written to pattern independent of note data. Edited patterns can be sent to a track where you can click and drag to repeat patterns for the required duration, and cut and paste patterns to when and where you like.

SYNCHRON32 - Step Sequencer



- Analog hardware-style step-sequencing allows you to quickly generate patterns, trigger synths, and trigger patterns in real-time from MIDI devices.
- Provides 32 steps with control over pitch; rhythmic duration with shuffle; tempo sync; per-step flam, legato, bends; polyphonic mode; 32 banks for pattern presets.
- After creating patterns in SYNCHRON32, you can simply right click in the Track View and select from the list of preset patterns. Then click and drag to repeat patterns for the required duration, and cut and paste patterns wherever you like.



- ACID-compatible loop integration with automatic beat matching, flexible time stretching/pitch shifting, and support for multiple loops per track. Click and drag to roll out tracks using ACIDized loops that match your projects tempo and pitch.
- Advanced synth layering and keyboard mapping capabilities allow you to perform and record using multiple MIDI controllers routed to multiple instruments; or use multiple, layered instruments or split synths from one MIDI controller.
- The realtime live performance audio engine provides a continuous musical workflow, allowing changes to be made on the fly and never compromises playback.

CAKEWALK

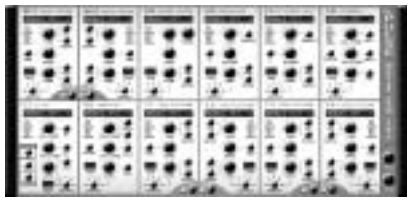
PROJECT5

PSYN — Virtual Analog Synth



- A fully automatable, 64-note polyphony, subtractive synthesizer with multiple filters, oscillators, envelope generators, and LFOs designed to give you access to classic 70s analog and contemporary electronic sounds. Other features include: a modulation matrix that provides tempo sync effects; portamento, ring modulation, FM synthesis and more.

nPulse — Analog Modular Drum Synthesizer



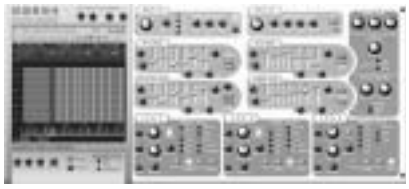
- 12-voice multi-timbral drum synth with one oscillator per voice combines the vintage-style electronic drum voices of classic TR-series drum machines, with the flexibility of modern day soft synths.
- 5 assignable outputs; multiple presets; sync effects; key-mapping.

Velocity — Drum Sampler



- The multi-timbral drum sampler provides 18 polyphonic voices with up to 32 velocity-layered samples and 5 stereo outputs.
- Precise sample editing with effects.
- Supports WAV, AIF, LM4, and proprietary format sample formats.

DS-864 — Digital Sampler



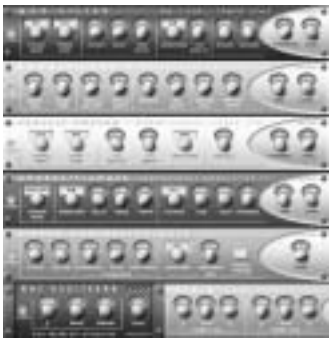
- Flexible and easy-to-use with support for up to 8 layers, 64 voice polyphony and 8 audio outputs.
- Supports key mapping and velocity zones
- Sample-level editing and effects for creation of new sounds.
- Two resonant filters with key tracking can be configured in series or parallel
- Four 6-stage envelopes with multiple slopes
- Support for Akai S5000/6000, Kurzweil, Sound Fonts 2, WAV, AIF, and proprietary sample formats.

Cyclone — Groove Sampler



- 16-part, ACID-compatible groove sampler, composition tool, and loop editor.
- Trigger loops in real-time following the tempo and pitch changes of a project.
- Edit individual slices of individual loops, and even combine parts of multiple loops.

Automatable Effects



- Access a rack worth of quality automatable DX audio effects including: Envelope/LFO Filter; Stereo Delay/Echo with tempo sync; Exciter; Classic Phaser; Studio Reverb; Stereo Chorus/Flanger, Stereo Graphic and Parametric EQ; and a Compressor/Gate.

Spectral Transformations



- Turn your audio inside out with six unique realtime effects integrated into one plug-in.
- Transposer – real-time pitch shifting produces major tonal changes ranging from high reedy sounds, to richer low sounds.
- Accumulator – glissando, delay, pitch bends and reverberation effects.
- Exaggerator sweeps through the peaks and valleys of spectral frequencies bands.
- Lo-Hi Filter removes frequencies above or below specified frequency levels.
- Tracer allows you to thin the sound until only a 'trace' of its former self remains.
- Shifter adds/subtracts the values to various frequency components.

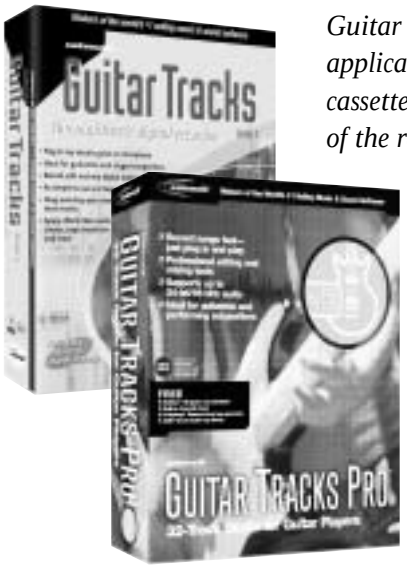
Expand and Customize

- Add DXi or VSTi synths to expand your studio. Use Project5 instruments and effects directly in SONAR.
- Edit and mix using the included audio & MIDI FX. You can also add third party MFX or DX effects as well as VST effects using the included VST/DX adapter.
- Integrate Project5 with a ReWire-compatible host like SONAR 2.x for advanced audio recording, editing, and mixing.
- Works with any WDM, DirectSound, or ASIO compatible Windows hardware.
- Sends MIDI Sync to outboard sequencers.

CAKEWALK

GUITAR TRACKS PRO & GUITAR TRACKS 2

Multitrack Recording Software For Guitar Players



Guitar Tracks Pro and Guitar Tracks 2 are straight forward digital audio recording and mixing applications for the Windows platform, that combine the ease of use of a portable multitrack cassette recorder with the power and flexibility of your PC. Each software package provides all of the recording, mixing, and editing tools the guitarist or singer/songwriter needs to achieve professional results. Guitar Tracks 2 features two track simultaneous record and eight track playback at 16-bit/48kHz. Guitar Tracks Pro features up to 32 simultaneous record/playback tracks with support for 24-bit/96kHz audio hardware. Both packages also feature guitar amp simulator plug-ins and other high-quality DirectX effects with support for third party plug-ins, a software-based drum loop generators and a chromatic-tuner. Whether you are a seasoned pro that needs a scratchpad for capturing ideas at the drop of a hat, or taking your first step towards computer-based recording, you will find that the no nonsense user interface and transparency of Guitar Tracks 2 and Guitar Tracks Pro will not stand between you and your music.

- Multitrack cassette recorder interface — just plug in your guitar or a mic into your sound card and start recording
- Record, edit and mix up to eight tracks of digital audio (32 tracks with Guitar Tracks Pro) and Mix down to stereo WAV files to burn CDs or make MP3s for the Internet (Trial version MP3 Encoder Included)
- Easy punch-ins, looping, and locate markers help the creative process keep moving
- The Edit View lets you move around song ideas with drag and drop ease
- Cut, copy and paste audio clips with the click of a mouse
- Digitally bounce audio for virtually unlimited tracks
- Includes stereo EQ, delay, flange, chorus, reverb and guitar amp simulation with two vintage amp models, British Overdrive and American Lead
- DirectX plug-in compatibility provides access to hundreds of 3rd party plug-ins
- Built-in chromatic tuner
- Includes Drag-and-Drop Drummer Lite – a simple tool that lets you construct pro-quality drum tracks from a library of digital audio drum loops
- The included SlowBlast guitar practice software lets you grab a guitar riff from an audio CD, slow it down, without changing its pitch and loop it until you learn it





Guitar Tracks Pro Adds –

- Record and playback up to 32 tracks of digital audio.
- 32-bit mixing and audio effects, and support for up to 24-bit/96 kHz audio
- Clip-based envelope mixing
- Dynamics processing including compressor, gate, expander
- ReValver SE guitar amp simulator lets you choose and tweak various pre-amps and power amps, 9-band EQ, Auto-Wah, and Stereo Reverb
- Fruityloops Express drum loop generator provides unlimited beat sequences with a variety of drum kits, audio samples, and drum patterns to use or modify.
- ACIDized audio loop library included

System Requirements

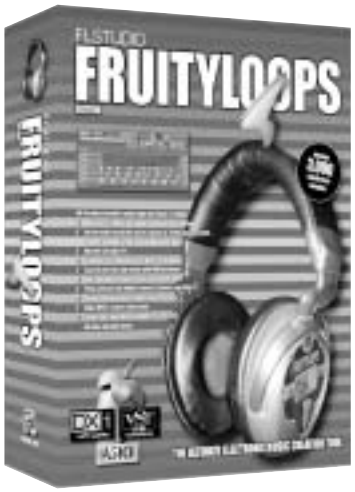
Windows 98, 98 SE, Me, 2000, XP
300 MHz processor; 64 MB RAM; 100 MB free hard disk space, CD-ROM drive and a Windows-compatible sound card

	Guitar Tracks Pro	Guitar Tracks 2
Maximum Number of Tracks	32	8
Highest Supported Audio Format	24-bit/96 kHz	16-bit/48 kHz
Multi-channel Audio Card support	Yes	Stereo only
Max # of Simultaneous Effects	16	8
ReValver SE Amp Simulator	Yes	No
Dynamics Processing	Yes	No
ACIDized Loop compatible	Yes	No
Chromatic Tuner	Yes	Yes
Fruityloops Express	Yes	No

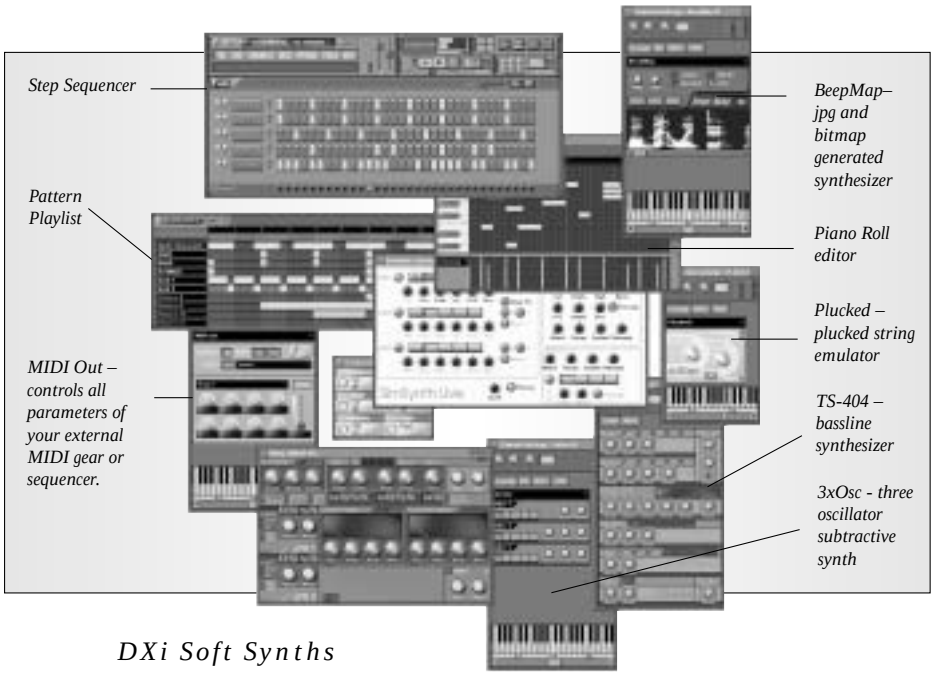
B&H PHOTO - VIDEO - PRO AUDIO
ONE HOUR FREE PARKING
AT 349 W. 34th STREET (with purchase of $100 or more)

FRUITYLOOPS 4

Music Creation Tool



FruityLoops is a hybrid loop creation and drum sequencing tool for Windows 2000 PCs, that combines loop generation, synth emulation, and pattern-based sequencing within a user-friendly photo-realistic interface. Features include an integrated BeatSlicer and Sample Browser, CD-quality stereo output, an integrated TS-404 bass line generator, a virtual keyboard, MIDI and DirectSound support, and individual panning/volume/pitch/mix controls. In addition to the built-in effects and sound generators, FruityLoops also provides support for DXi soft synths and DirectX audio effects. You can export samples to WAV, MP3, or MIDI files.



- Standalone and VSTi includes a wide variety of sound generators, sampled sounds, and emulations of vintage gear combined into one easy to use program
- CD quality 16Bit/44KHz stereo output
- Export loops and songs to .WAV, MP3, and MIDI
- Includes five polyphonic soft synths, over 3000 samples, and 250 loops right out of the box.
- Integrated TS-404 bass line generator
- Unlimited number of tracks and samples
- Pattern-based sequencing with 4 to 64 notes per pattern.
- Piano Roll view with chords, note lengths and slides
- External control from MIDI keyboard or PC keyboard.
- Integrated sample browser
- Control panning, volume, pitch, mix, and tonal characteristics for individual samples
- Playlist view for mixing several patterns
- Peak meters
- Skin support for custom graphics
- BeatSlicer support

Minimum System Requirements

Windows 95/98/Me/2000, Pentium class 200MHz processor, 32MB RAM, Windows-compatible audio hardware, CD-ROM drive

DXi Soft Synths

- Supports DX Instruments (DXi) from Cakewalk and includes the Dreamstation DXi soft synth.
- Additional software synths are available from leading manufacturers like Native Instruments, Edirol and many others.

Automation

- You can record movements you make on any wheel or slider in Fruityloops or on your external MIDI controller in realtime.
- Automation of main sliders, volume, effects, oscillators, resonance, cut-off, panning, mute, effects sends, tempo, and more.
- After recording you can edit it all in a handy, graphical event editor. You can also draw in automation data or use an
- LFO generator to automatically create fluent automation movements.

Create and Mix On The Fly

Create drum and melodic patterns using Fruityloops instruments and the included Piano Roll view. Then arrange your patterns using the Playlist view to create your own songs and mixes.

DirectX & VST Plug-in Support

- 16 auxiliary effects channels and one master effects channel is provided for patching DirectX audio effects.
- VST plug-ins are also supported including: 7 Band EQ, Bassboost, Overdrive, Ultrafunk Compressor, Delay, Fast LP, Filter, SmartElectronix Flanger, Free Filter, SmartElectronix Phaser, Ultrafunk Reverb, and more.

DIGIDESIGN

M-BOX

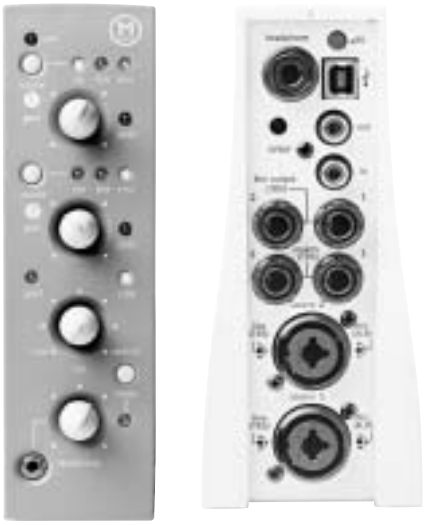
2 Channel USB-powered Pro Tools LE-based Micro Studio



Mbox is a compact and affordable 2-channel USB-powered audio interface for MacOS 9.1 and up, MacOS X 10.2.3 and up, and Windows XP, that's been specifically engineered to offer the home/project studio and the pro who needs a studio quality digital recording, mixing and editing environment that can be taken virtually anywhere. Mbox features a pristine 24-bit input-to-output signal path with two Focusrite designed (Green Range) mic preamps accessible via 1/4″ TRS / XLR analog inputs; two 1/4″ analog outputs as well as 1/8″ and 1//4″ headphone jacks with a dedicated volume control for monitoring. A 24-bit stereo S/PDIF digital I/O is also provided for transferring data to and from the Mbox entirely in the digital domain. Every Mbox includes ProTools LE software which features 32 audio tracks and 128 MIDI tracks along with a host of RTAS (Realtime) and AudioSuite (File-based) effects plug-ins.

FEATURES

- Two analog inputs featuring two Focusrite mic preamps with variable gain and switchable 48V phantom power.
- XLR, 1/4″ inputs using Neutrik combo connectors with separate source selection (MIC/LINE/INST) and gain controls with peak LEDs per channel.
- Two analog inserts, using 1/4″ TRS connectors, allow you to use outboard processors while recording to disk.
- 1/4″ TRS left and right outputs with 24-bit D-to-A converters.
- Support for 44.1 kHz and 48 kHz sample rates.
- 24-bit signal path from input to output
- 24-bit coaxial S/PDIF digital I/O allows you to work entirely within the digital domain.
- Zero-latency monitoring
- Inputs and outputs accept both balanced and unbalanced connections.
- 1/4″ and 1/8″ headphone outputs with dedicated volume control.
- 100% USB powered
- Mbox sessions can be migrated to another Pro Tools platform, from Digi 001 to Pro Tools|HD, for further manipulation.
- Includes a USB cable



Specifications

Mic pre-amp: > -120 dB EIN @ > 40 dB gain	
A/D Dynamic Range:	101 dB (A-weighted) 99 dB (unweighted)
Maximum Input:	+24 dBu
D/A Dynamic Range:	103 dB (A-weighted) 101 dB (unweighted)
Frequency Response:	20 - 20k Hz (±0.5dB)
Maximum Output:	+4.2 dBu

Software Supplied with System

- **Pro Tools LE** Software — 32 audio tracks, 128 MIDI tracks with powerful editing, mixing real-time plug-in support and automation.
- **DigiRack RTAS and AudioSuite Plug-Ins:** D-Verb, Dither, Dynamics II, EQ II, Invert/Duplicate, Mod Delay, Normalize/Gain Change, Reverse/DC Removal, Signal Generator, Time Comp-Exp/Pitch Shift, Trim
- Digidesign **ASIO Driver** for Windows XP allows you to use the Mbox with a wide range of popular third-party audio apps such as Cakewalk SONAR 2.2, Propellerhead Reason and Ableton Live.

DigiStudio

DigiStudio is featured component of ProTools software that provides a revolutionary way to collaborate with other Pro Tools users via the Internet. DigiStudio participants interact from any Net-connected location to exchange Pro Tools session media, from raw audio tracks to MIDI data to plug-in information to automation.

DIGIDESIGN

DIGI 001

PCI-based Pro Tools LE Music Production System



The Digi 001 is a computer-based 24-bit digital audio workstation system that offers a seamless hardware and software solution, on both the MacOS and Windows platforms, for project studios and production suites.

The Digi 001 system combines a rackmountable breakout interface and PCI card that together provide 18 channels of simultaneous analog and digital I/O, including two high-quality mic preamps and MIDI I/O. Also included is the latest version of Digidesign's powerful recording, mixing, automation and editing software, Pro Tools LE, which offers 32 tracks of audio, 128 MIDI tracks as well as real-time (RTAS) plug-in support.

FEATURES

18 Simultaneous Inputs And Outputs

- All inputs and outputs run at a 24-bit resolution at 44.1 or 48 kHz sample rates
- Eight channels of 1/4″ balanced analog inputs and outputs
- 2 mic preamps, with switchable phantom power, variable and a switchable 26dB pad are provided on the front panel.
- Eight channels of ADAT optical I/O (located on the PCI card)
- Coaxial S/PDIF digital I/O
- One MIDI In and one MIDI out port is provided for accessing an external control surface, keyboard controller, synth, sampler or other MIDI device.
- A pair of 1/4″ TRS stereo monitor outputs with a dedicated volume control run in parallel to the Main outputs (outputs one and two).
- Headphone output with separate volume control.
- 1/4″ footswitch port for QuickPunch recording control.
- Rack ears are provided for installing the I/O box in a standard 19″ rack.



Digi 001 Factory Bundle



- With over $2,400 worth of plug-ins from Digidesign and their Development Partners the Digi 001 Factory bundle is an ideal choice for rounding out your home and project studio production environment.

Plug-ins include:

- Digidesign's D-Fi, D-Verb and Maxim combining a unique blend of creative processing plug-ins, high-quality reverb and mastering tools.
- The classic analog sounds of the 1176, LA-2A compressors and the mooger-fooger Lowpass filter from Bomb Factory.
- Supertap, Metaflanger and Q10 EQ from Waves
- Native Instruments Dynamo 001 Edition analog synthesizer
- Also includes the iLok USB Smart Key which provides a portable, convenient way to store software authorizations.

DIGIDESIGN

DIGI 002

FireWire-based Pro Tools LE Music Production System



The Digidesign Digi 002 is a 24-bit/96kHz FireWire-based mini studio that combines a touch-sensitive control surface borrowed technology from Digidesign's Control|24; an audio interface with eighteen simultaneous audio ins and outs, including 4 mic preamps; MIDI I/O; along with the latest version of Pro Tools LE software that works with both Windows XP and Mac OS 9.x and Mac OS X 10.2.3 and higher. The Digi 002 delivers seamless integration between hardware and software as each and every adjustment of the control surface is reflected in the Pro Tools LE software interface — and vice versa. Adding even greater flexibility, the Digi 002 can be uncoupled from Pro Tools and used as a stand-alone mode as an 8x4x2 digital mixer with onboard EQ, dynamics, delay and reverb. The Digi 002 comes bundled with over $2200 worth of high-powered plug-ins from Digidesign, IK Multimedia, Native Instruments, and Waves.

FEATURES

- Audio, MIDI and control surface information is passed over a single IEEE 1394 FireWire cable.
- All audio is processed at 24-bit resolution with sample rates up to 96 kHz providing a dynamic range greater than 108 dB.

18 Simultaneous Channels of Inputs and Outputs

- 8 simultaneous 24-bit/96 kHz analog inputs: eight 1/4″ TRS inputs and four XLR mic pres with individual gain and high-pass filter; 48V phantom power enabled on channel pairs.
- Eight 1/4″ TRS analog outputs with 24-bit/96 kHz D-to-A converters
- Left and Right 1/4″ TRS Monitor outputs (+4 dBu) with dedicated volume control mirror Main outputs 1 and 2.
- 1/4″ TRS headphone output with dedicated volume control knob
- Unbalanced RCA –10 dBV fixed output
- Eight channels of ADAT optical I/O (up to 48kHz), switchable to 2 channels of optical S/PDIF I/O (up to 96 kHz).
- Two channels of coaxial S/PDIF digital I/O supporting up to 24-bit, 96 kHz audio
- Alternate RCA –10 dBV inputs for monitoring tape decks, CD players, etc.)

Control Surface

- The top panel is arranged into four sections: The Fader section includes eight channel strips; The Console/Channel View section provides controls for viewing and controlling inserts, plug-ins and sends; The Transport and Navigation controls access many on-screen navigation features.

Additional Features

- One MIDI In port and two MIDI Out ports, providing up to 16 MIDI input channels and 32 MIDI output channels can be used for adding a keyboard controller and/or external synths and samplers, etc.
- 1/4″ TRS Footswitch for QuickPunch I/O control.

Included Software

Includes the latest version of Pro Tools LE software as well as a number of high quality plug-ins, offering more power and flexibility than ever before including up to 32 tracks. The free plug-ins are valued at over $2,250 and include: SampleTank LE and Amplitube by IK Multimedia, D-Fi, D-fx and Maxim by Digidesign, PR0-52 by Native Instruments, Renaissance Collection by Waves

Stand Alone Mixing

- The Digi 002 can be used in stand-alone mode, (without your computer) as an 8x4x2 digital mixer with onboard effects (including dynamics processing, EQ, delay and reverb) and snapshots.
- 8 analog inputs including 4 mic preamps, with dedicated volume, pan, solo and mute controls.
- Dedicated three-band EQ available on input channels 1–8
- Dedicated Compressor/Limiter available on input channels 1–4
- Built-in high-quality Reverb and Delay
- Four sends on each input channel:
  Sends 1–2 dedicated to internal Delay and Reverb effects
  Sends 3–4 can be used for integrating external effects.
- Stereo Main and Monitor 1/4″ TRS (+4dBu) outputs plus an alternate Main Output pair (–10 dBV) that mirrors the Main Outputs.
- Stereo headphone output with level control
- 10 scribble strips for pan/volume and effects control, send levels, and track names
- Channel metering available using rotary encoder LED rings (in Meter mode)

DIGI 002



Mic Pre Controls including variable Gain and source select (Mic/Line/Inst)

Multi-purpose scribble strips for displaying track name, pan, send, fader, plug-in values and timeline position.

Multi-purpose, velocity-sensitive rotary encoders with LED position indicators.

Solo/Mute/Rec enable buttons

Eight 100mm motorized touch-sensitive faders

Headphone output with level control plus a separate Monitor output control.

Instantly switch between channel views – EQ, Dynamics, Inserts and Pan/Send.

Status indicators displays MIDI, FireWire and L/R output activity in realtime as well as the currently selected sample rate

Specialized function keys for Pro Tools software and standalone mode

Navigation buttons can be assigned for fader bank select, zoom in/out and nudge

Instantly switch between Plug-in, Mix and Edit windows. Play/Rec mode switches include Loop Play, Loop Record and Quick Punch

Transport controls – Play, Record, Stop, FF, Rew and Return To Zero



Two IEEE-1394 FireWire Ports — the first port connects the Digi 002 to the "Host" computer while the second port allows you to daisy-chain additional peripherals such as FireWire hard drive.

Eight simultaneous balanced inputs: four 1/4˝ TRS line / instrument inputs and four XLR mic inputs (channels 1 -4); and four 1/4˝ TRS line inputs (channels 5-8).

Digi 002 Rack — 2U FireWire Interface



Digi 002 Rack is Digidesign's first rackmount FireWire-based Pro Tools product. Designed with both recording and performing musicians in mind, Digi 002 Rack features analog, digital and MIDI I/O, 24-bit/96 kHz converters, and 32-track Pro Tools LE software to provide you with a creative playground like no other.

- FireWire connectivity — A single FireWire/IEEE-1394 cable conveys all digital audio, MIDI data, and Pro Tools data between Digi 002 Rack and your PC or Mac
- 24-bit/96 kHz fidelity
- 18 channels of audio I/O: 8 analog (with 4 mic preamps), 8 ADAT optical, and 2 S/PDIF
- Integrated MIDI I/O (16 channels in/32 channels out)
- 32-track Pro Tools LE 6.x software

Includes DigiRack plug-ins and the Digi 002pack v2.0: a cross-platform RTAS, AudioSuite plug-in bundle.

Digidesign Maxim and DiFi — World-class peak limiting and sound level-maximizing and Analog and retro signal processing in the digital domain

IK Multimedia SampleTank SE — Includes 64 factory preset instruments to enhance your compositions; and **AmpliTube** with over 1,200 amp configurations as well as stomp box and effects modeling.

DIGIDESIGN

PRO TOOLS LE



Host-Based Music Production Software

Pro Tools LE is the music production software that is bundled with Digidesign's host-based hardware audio interfaces. Using the host processing capabilities of today's super fast Mac and Windows computers Pro Tools LE allows you to record and play up to 32 mono digital audio tracks and up to 128 MIDI tracks with a host of non-linear, random-access, precision editing and mix automation functions all within a non-destructive environment. Pro Tools LE is essentially divided into two main working environments: The Edit window and the Mix window. The Edit Window provides a graphic overview of your song along a timeline that allows you to adjust every aspect of both audio and MIDI data simultaneously and at sample-level resolution. Using the Edit window you can trim (resize) waveforms, reprocess regions of audio, as well as rearrange and automate your song all within one streamlined and elegant interface. The Mix window's console-style graphic interface displays audio and MIDI channels in the same order that they appear in the Edit window. The Mix window not only provides access to standard volume, pan, mute and solo functions, it allows you to process your audio tracks with realtime (RTAS) effects plug-ins — up to five effects sends and five inserts per track, depending on available CPU resources. Both the Edit and Mix Windows provide a variety of methods for automating your mix to perfection. Audio Tracks can be recorded with selectable 16-bit or 24-bit resolution at sample rates up to 48kHz for the Mbox and Digi 001 and up to 96 kHz for the Digi 002.

FEATURES

Audio

- Simultaneously record and play back up to 32 tracks of audio at sample rates of up to 48 kHz (96 kHz with the DIGI002).
- 24-bit signal path from A-to-D to D-to-A.

MIDI

- Record and play up to 128 MIDI tracks
- MIDI events and sequences can be manipulated right alongside audio regions with sample level accuracy.

Audio Editing

- Most audio editing is non-destructive — whether cutting, pasting, trimming, separating, or clearing regions, you are only performing these functions on a map of the actual audio data. The source audio files remain untouched.

Multiple Undo

- Pro Tools can keep track of up to 16 of the last undo-able operations, allowing you to return to a previous editing state.

Plug-ins

- A number of high quality realtime (RTAS) and file based (AudioSuite) plug-ins, from dynamics to effects to sound design, are included with Pro Tools and many others are available from third party developers.
- Because of their seamless integration into the Pro Tools application, and their ability to be instantiated onto multiple tracks, software plug-ins often improve upon the capabilities of the hardware counterparts — customized settings are saved with your session and parameters can be automated.

Mix Automation

- Pro Tools LE provides dynamic automation of volume, pan, and mute controls for audio tracks and sends, MIDI tracks, and real-time plug-in parameters.
- You can write automation moves in real time during playback of your session.
- Edit breakpoint automation data, in the Edit Window, with the same techniques you use to edit audio and MIDI data.

Importing Audio

- ProTools LE can import the following Audio files types: AIFF; WAV; SD II; SD I
- MP3 import via optional software
- Sound Resource (AIFL—Macintosh only)
- WMA (Windows Media—Windows only)
- On Macintosh systems, you can import tracks from an audio CD — ideal for working with CD-based sample libraries.

Quicktime Video

- You can use Pro Tools to import QuickTime movies and audio, perform audio post tasks, and export the finished product as a new QuickTime movie.

Third Party Apps

- ProTools hardware and/or ProTools software can be integrated with popular audio production applications allow you to work in a hybrid environment, if you wish. These applications include MOTU's Digital Performer, Emagic Logic Platinum (Mac) and Cakewalk's SONAR 2.2 (Win).

PRO TOOLS LE

The Edit Window

The Edit window provides a timeline display of audio, as well as MIDI data and mixer automation for recording, editing and arranging tracks. As in the Mix window, each track has controls for record enable, solo, mute and automation mode.

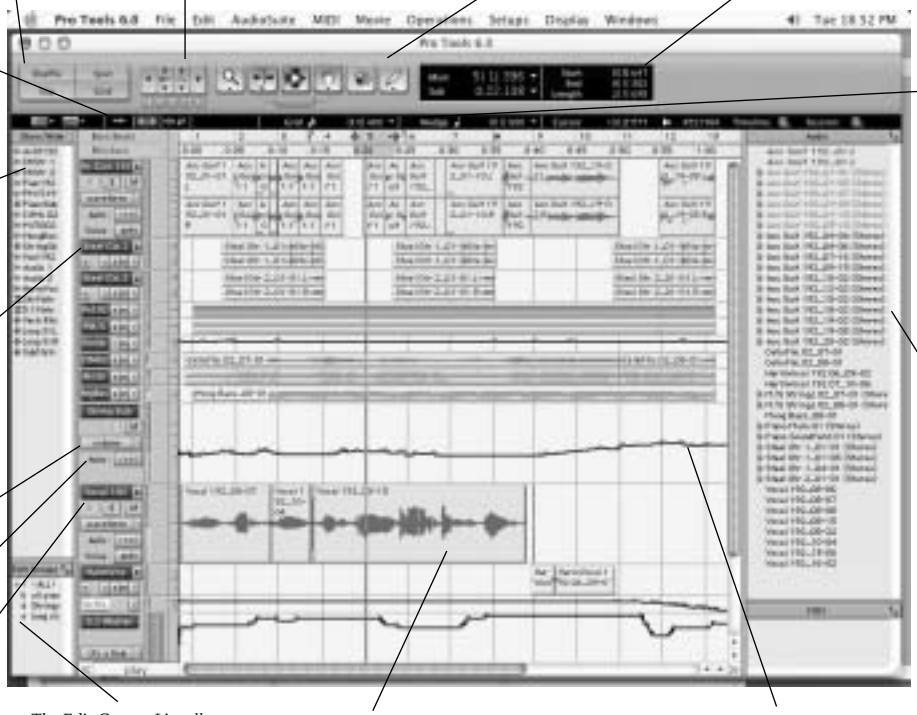
*The Edit Mode buttons affect the movement and placement of regions within the edit window. In **Shuffle** mode, regions automatically snap to one another along the timeline. In **Slip** mode, regions can be moved freely within a track or to other tracks. **Spot** mode allows you to place regions at precise locations based on any available time format. In **Grid** mode, regions and MIDI notes that are moved or inserted "snap" to a user-definable time grid.*

Multi-purpose editing tools include: the Magnifier (zoom) tool; the Trim tool for resizing regions; the Selector tool for defining regions and selecting playback positions; The Grabber tool for selecting regions for placement and editing; The Scrub tool; and the Pencil tool for destructively redrawing a waveform

The current position on the timeline is displayed by the Location indicators. The Event Edit defines the start, end and length of selected audio or MIDI data.

The Tab to Transients option lets you to navigate the cursor to just before the detected transient peak. Ideal for defining selections and play ranges, as well as start and end points for new regions, without having to zoom in on the waveform.

Zoom Buttons and five Zoom memory presets



Select the desired Grid and Nudge values for placement and editing MIDI and audio data. Nudge allows you to move regions, MIDI notes or automation breakpoints by precise increments using the + and - keys. You can also use Nudge to trim a region.

Show/Hide Tracks List

The Playlist Selector pop-up menu allows you to select a group of regions arranged on an audio or MIDI track. Each track has edit playlists and automation playlists.

Regions can be dragged from the Audio Regions list and placed anywhere on the timeline.

Track View selector let's you choose between a waveform and the different automation views available for each track

Automation mode selector

Track name
Track Rec/Solo/Mute buttons

The Edit Groups List allows you to create and organize groups of tracks (i.e all background vocals) for viewing and editing.

Audio Waveform – track heights can be individually resized

Dynamic Breakpoint Automation — Control volume, pan, and mute for audio tracks and sends, MIDI tracks, as well as real-time plug-in parameters. You can write automation moves in real time during playback of your session. You can also edit automation data with the same techniques you use to edit audio and MIDI data.

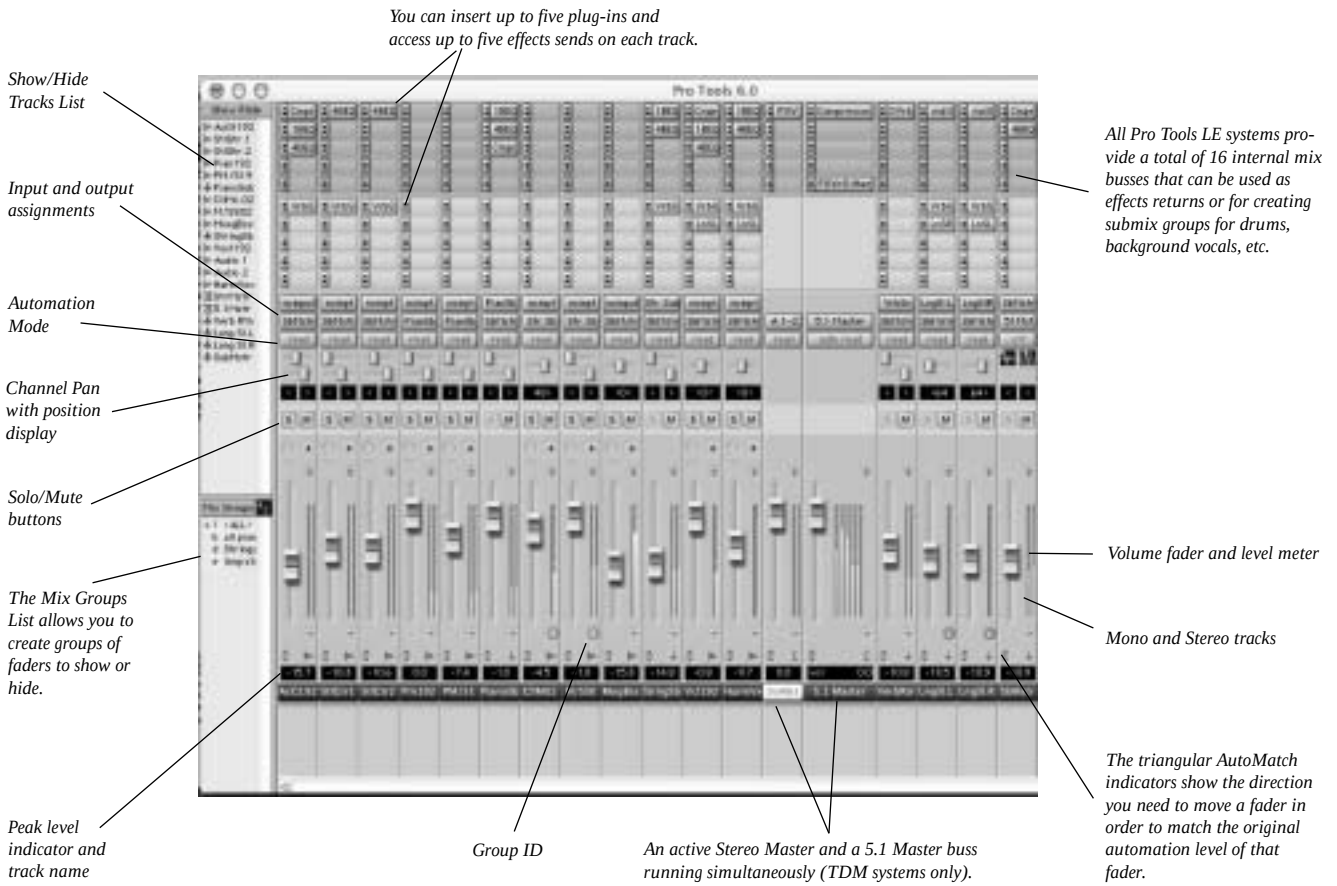
Four Recording Modes

- In normal Non-destructive Record mode, audio is recorded non-destructively, which means that if you record over a track's existing regions, the audio is not erased from your hard drive. Both the new and old audio files remain on your hard drive, available as regions from the Audio Regions List.
- In Destructive Record mode recording over existing regions replaces the original audio permanently.
- Loop Record allows you to record take after take while the same section of audio repeats. This is a convenient technique for quickly recording multiple takes of a part without losing spontaneity.
- QuickPunch allows you to instantaneously punch-in and out on a record-enabled audio track, non-destructively, by clicking the Transport's Record button.

PRO TOOLS LE *continued*

The Mix Window – Integrated Multi-Channel Mixing and Surround Panning

In the Mix window, tracks appear as mixer strips (or channel strips), with controls for inserts, sends, input and output assignments, volume, panning, record-enable, automation mode, and solo/mute. The following section explains each of these track controls.



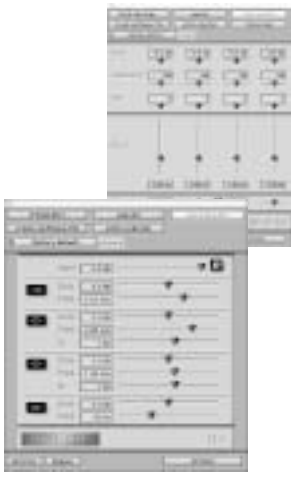
Plug-Ins

- A number of high quality plug-ins, from dynamics to effects to sound design, are included with Pro Tools and a many, many others are available from third party developers.
- Because of their seamless integration into the Pro Tools application, software plug-ins often improve upon the capabilities of the hardware counterparts — you are free to employ presets, customize settings, as well as automate a plug-in's parameters.

RTAS and AudioSuite Plugins

- RTAS (Real-Time AudioSuite) plug-ins are real-time processing effects that rely on the DPS provided by your computer. They can be assigned to multiple tracks as frequently as you like depending on the available CPU resources.
- AudioSuite plug-ins are non-real time, file-based processors. Once an AudioSuite plug-in is applied to an audio file or region, a new audio file is created, leaving the original unprocessed available as a back-up.



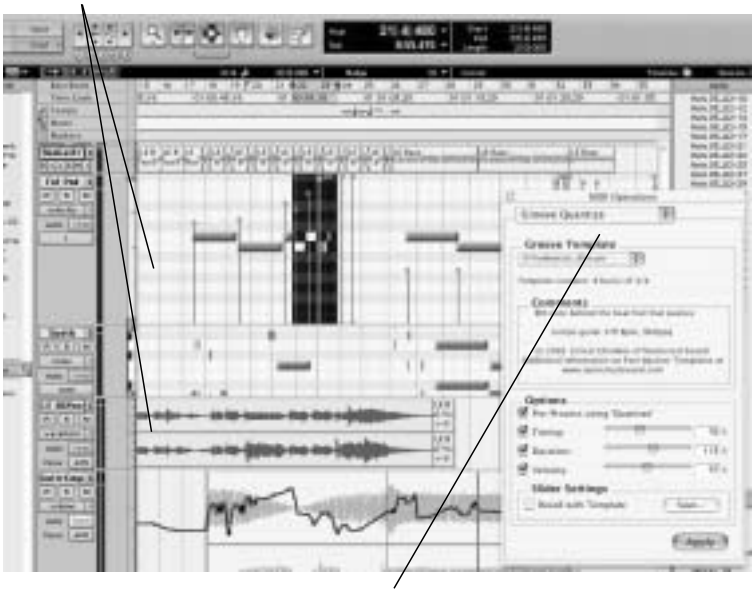


PRO TOOLS LE *continued*

MIDI Functions and Editing

- An integrated MIDI environment, with support for up to 128 MIDI tracks gives you full control over the MIDI components of your projects.
- Full support for Mac OS X's Core MIDI Services.
- MIDI Time Stamping (MTS) support provides sample-accurate MIDI with Pro Tools-compatible software synths and samplers, eliminating the possibility of compromised sync and drift issues.
- Up to sub-millisecond-accurate MIDI with Digidesign's MIDI I/O and other supported MTS-capable interfaces, rivals dedicated hardware sequencers.
- Import and export Standard MIDI Files
- A range of precision editing tools are provided for creating and editing MIDI notes and controller events within the ProTools Edit window including: the Pencil tool which can be used for inseting notes at specific pitches as well as adjusting a note's velocity and duration; as well as the Trimmer and Grabber tools for changing the start and end points of MIDI notes as well as their position along the timeline.
- There are also a number of "MIDI Operations" windows that allow you to transform groups of MIDI notes including: Quantize for correcting timing; Transpose, Change Velocity for making notes louder and softer; and Change Duration.
- The Restore Performance function allows you to return to the original state of a MIDI-based performance at any time. The Flatten Performance feature solidifies your edits and gives you a new, permanent basis from which to proceed.
- The MIDI Event List editor displays a detailed list of all Note and Controller events for a single MIDI track including each events start position, the event type and the event's duration. Events can be inserted, copied and pasted, selected, or deleted. A View Filter allows you to hide certain events for easier editing.

MIDI events and sequences can be edited and manipulated right along side audio.



The Groove Quantize feature incorporates intelligent Bar|Beat adjustment as well as support for the new 960 ppq DigiGroove templates.

Mixdown and Mastering

- In addition to providing an ideal recording, editing, and mixing environment, you are able to add the finishing touches to your projects through mastering.
- Editing capabilities and plug-ins used in conjunction with the Pro Tools software interface enable you to make the necessary fine adjustments to your final mix to achieve the perfect combination of warmth, presence, and balance.
- The real-time "bounce to disk" feature reads all automation information, allows incorporation of various outboard gear, and provides you with mono, stereo, or surround output of your final mix, which can then be handed over to your mastering expert of choice in a variety of digital audio file formats.
- Digidesign's MasterList CD, a software application specifically designed for mastering tasks, can be used to master a single piece or a collection of work with professional results.

POW-r Dithering



- Dithering is used to reduce quantization noise when mixing or fading low-level audio signals during word size reduction to 16-, 18-, or 20-bits.
- The POW-r Dither plug-in is an advanced type of dither that provides optimized word length reduction. It is designed for final-stage critical mixdown and mastering tasks where the highest possible fidelity is desired when reducing bit depth.

PRO TOOLS LE *continued*

Crossfades

- You can quickly and easily create crossfades between two adjacent audio regions to smoothing transitions between regions and prevent pops, clicks, or sudden changes in sound.
- The crossfade duration, position, and shape are all user-definable.
- The Fades dialog window allows you to select, view, and manipulate the curves used to perform the crossfade. Different volume curves can be assigned to the fade-out and fade-in portions of crossfades.
- You can also preview audition the crossfade from within the Fade dialog.



Strip Silence



- The Strip Silence window allows you to automatically divide a selected audio file or region by removing the areas of silence from a selection. This can be useful for quantizing individual audio events to musical values or cueing sound effects to SMPTE locations.
- **Strip Threshold** which sets the amplitude threshold for Strip Silence. Audio falling below this threshold is considered silence and removed.
- **Minimum Strip Duration** Sets the minimum duration (from 0 to 10,000 ms) that the material below the threshold must last to be considered silence.
- **Region Start Pad** and **Region End Pad** specifies a time value to be added to the beginning and end of each new region created with Strip Silence. This is useful for preserving nuances that fall below the threshold, such as the breath before a vocal phrase, or a finger slide on a guitar.

ReWire Support

- Support for Propellerhead's ReWire technology provides real-time streaming of 24-bit audio and MIDI between ReWire compatible software synthesizers, samplers, and other host-based instruments, with sample accurate synchronization and common transport functionality. ReWire-enabled instruments can be be directly routed into the Pro Tools mixer where they can be integrated with your session and access additional mixing and processing functions.

Digidesign Core Audio Driver (Mac OS X)

- The Digidesign Core Audio driver allows third-party applications supporting the Apple Core Audio standard to utilize a Pro Tools system for audio input and output. Depending on the hardware and application, you can record and playback multiple channels of 24-bit audio up to 96 kHz.
- A Digidesign MIDI I/O driver is available for users of the Digidesign MIDI I/O.

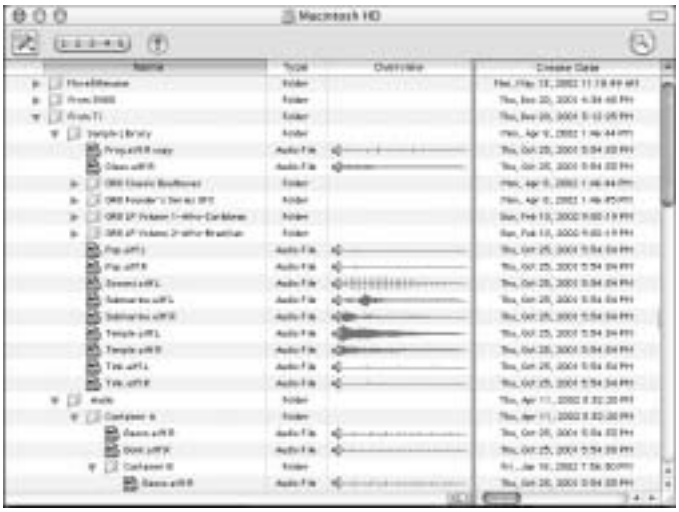
Digidesign ASIO Driver (Windows XP)

- The Digidesign ASIO Driver is a multi-channel, multimedia sound driver that allows third-party audio programs which support the ASIO standard to record and play back using Windows XP Professional or Home.

DigiBase File Management Utility

To help you organize larger Pro Tools sessions a file management utility called DigiBase is similar to Mac's Finder or Windows' Explorer, enabling you to manage all of the files on your system.

- DigiBase allows you to view complete file information including duration, time stamps, along with two user comments fields.
- Display layouts can be customized to focus exclusively on the information you need.
- You can also audition files and view their waveform thumbnails before importing into Pro Tools, regardless of file format or sample rate.
- When you've determined a selection, the one-step drag-and-drop process from DigiBase to your session timeline includes automatic file copy and conversion, guaranteeing compatibility with your project.
- Tasks such as copying, conversion, and fade creation are handled in the background so that you can continue to record and edit without interruption. A new Task Manager feature also enables users to monitor and manage background processes.



PRO TOOLS LE *continued*

DV Toolkit for Pro Tools LE

DV Toolkit is an optional bundle that adds desktop post production functionality to the Pro Tools LE system. Specialized functionality gives users tools to edit audio with timecode functionality; support for the import and export of media files with other applications including new AAF interchange with Avid Xpress DV; and the ability to replace bad on-camera dialog and repair/restore poor field audio.

- AAF and OMF Import and Export (with bundled DigiTranslator option)
- Timecode ruler in the Pro Tools LE interface for spot-to-timecode capabilities
- One-of-a kind lip sync capability when replacing noisy location dialog with Synchro Arts VocALign Project
- DINR AudioSuite Noise Reduction plug-in

Enhanced Windows XP Features

- Streamlined new look for the Mix, Edit, and Transport windows, the powerful DigiBase file management utility, enhanced MIDI functionality and Beat Detective features and more.

Windows Media Audio 9 Encoder

- Microsoft's Windows Media Format is the optimal digital media format available for streaming and download-and-play applications on PCs, set-top boxes, and portable devices. Soon, Pro Tools users will be able to take advantage of this powerful format using the new Pro Tools Windows Media Audio Encoder. The Windows Media Audio Encoder allows Pro Tools TDM and LE users to both import and export WMA files in the simple and familiar Pro Tools environment. Visit Microsoft's booth to see it in action and www.microsoft.com for more information on Windows Media.

Pro Tool's support for Windows Media 9 Series offers:

- 16-bit 24-bit audio encoding
- 44.1kHz, 48kHz, and 96kHz sampling rates
- 2-channel stereo, 5.1 and 7.1 multichannel configurations
- One-pass and Two-pass encoding
- Support for Windows Media Audio (WMA) 9 Lossless encoding.

ProTools LE Compatible Hardware Comparison

System Type	Mbox	Digi 001	Digi 002
Simultaneous Playback Tracks	32 (Pro Tools 6.0 or higher and Pro Tools 5.3.1 or higher on Windows)	32 (Pro Tools 6.0 or higher and Pro Tools 5.3.1 or higher on Windows)	32
Highest Supported Sample Rate	48kHz	48kHz	96kHz
Number of I/O Channels	up to 2 in/2 out	up to 18 in/18 out	18 in/18 out (up to 48 kHz) 10 in/10 out (at 96kHz)
A-to-D Conversion	24-bit	24-bit	24-bit
D-to-A Conversion	24-bit	24-bit	24-bit
Digital I/O	24-bit	24-bit	24-bit

Canopus — DV converter specialized for NLE

Convert your S-VHS, Hi8 and 8mm analog tapes to DV in one simple step using the ADVC-100. The converted DV streams are transferred to your PC or Mac via IEEE 1394 (i.Link, FireWire) and stored on your hard drive where they can be manipulated using your favorite photo or video editing applications. This device is ideal for all OHCI and DV-only capture cards for Macintosh or PC.

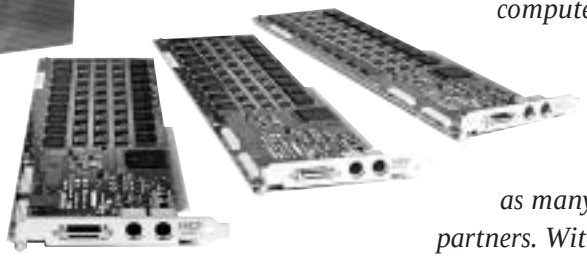


- Uses Canopus's award-winning DV CODEC Technology, which is found in AVID's Xpress DV software and the full line of Canopus DV capture cards.
- Locked Audio Support allows you to capture long clips w/perfect audio sync.
- Analog output of NTSC color bars for reference signal)
- 4-pin DV jack on front; 6-pin FireWire jack on back
- Analog video In/Out: NTSC (525/60), PAL (625/50) S-video and composite
- Analog input connector on front
- Analog output connector on back
- Includes AC adapter, DV cable (6 pin - 4 pin), Video cable (S-video/comp)

DIGIDESIGN

PRO TOOLS|HD





High-Definition Audio Production System

Pro Tools|HD is Digidesign's latest and most advanced digital audio workstation ever, with support for up to 24 high resolution audio tracks at 192 kHz sample rate and all of the processing power required for audio recording and playback, mixing, effects and automation being supplied via dedicated HD Process cards that reside in the PCI slots of your computer. Pro Tools|HD is available in three basic configurations, or Core systems, consisting of one to three HD Process cards (HD 1, HD 2, and HD 3), and packaged with the latest versions of Pro Tools TDM software along with a host of effects plug-ins from Digidesign, as well as many third party plugins from Digidesign's development partners. With twice the DSP-based mixing and plug-in processing power as previous Pro Tools systems, Pro Tools|HD delivers a substantial increase in guaranteed track count (up to 128), with up to 96 channels of I/O as well as increased routing and expansion flexibility. Digidesign and its development partners offer a variety of high resolution audio interfaces (at least one interface is required) and other in-demand peripherals, including control surfaces, MIDI and sync interfaces, software plug-ins, as well as support for a total of seven HD Process cards, allowing you to custom configure a fully integrated and professional production solution that suits your specific needs.

FEATURES

Modular Design – Expandable DSP Power and I/O Capacity

- Support for 24-bit audio with a sample rate up to 192 kHz.
- Guaranteed support for up to 128 simultaneous audio tracks with no stress on the computer (96 tracks with an HD 1 system).
- Pro Tools|HD's dedicated processing power and I/O capacity is based around a full length PCI card known as the HD Process card. The first card in the system is called the HD Core card. The HD Core card supports up to 96 tracks of direct-to-disk recording and playback, 32 channels of I/O and enough DSP power for mixing and plug-in processing.
- The HD system's modular hardware design supports up to 6 additional HD Process cards (for a total of 7) allowing you to dramatically increase your system's mixing and plug-in processing power as your needs grow. Each HD Process card provides up to 32 additional channels of I/O and you can expand your system to a total of 96 channels of I/O using multiple HD cards.

A Complete Professional Audio Solution

- There are three world-class, high-resolution audio interfaces available to the HD system: 192 I/O and 192 Digital I/O support sample rates up to 192 kHz and the 96 I/O which supports sample rates up to 96 kHz. Each interface offers a wide range of standard analog and digital connections allowing integration of the HD system into any professional working environment.
- A proprietary multi-pin DigiLink interface is used for connecting the HD Core and HD Process cards with an HD compatible audio interfaces. Interfaces can be connected up to 100 feet from your computer with sample rate support up to 96kHz. This allow you to keep your computer and hard drives nestled away in a machine room thus allowing a quieter control room.
- Digidesign offers a number of additional dedicated peripherals, including control surfaces, synchronization and MIDI interfaces, mic preamp and more, for enhancing Pro Tools' advanced recording, editing and mixing functions.

Core Systems

Pro Tools|HD 1 features the HD Core card, supporting up to 32 channels of I/O, 96 simultaneous audio tracks, and housing a host of powerful DSPs that provide the dedicated mixing and processing horsepower audio professionals have come to expect from Pro Tools.

Pro Tools|HD 2 includes the HD Core card and an additional HD Process card, offering more than double the mixing and processing power of HD 1 systems, support for 64 channels of I/O, and up to 128 simultaneous audio tracks.

Pro Tools|HD 3 features the HD Core card and two additional HD Process cards, supporting up to 96 channels of I/O and 128 simultaneous audio tracks. This configuration offers the ultimate in power and flexibility, enabling you to complete projects more quickly and efficiently than ever before.

- Each Core system requires at least one Pro Tools|HD audio interface — 192 I/O, 192 Digital I/O, or 96 I/O — for handling the input and output (I/O) of audio signals to and from your system.

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ONE HOUR FREE PARKING
AT 349 W. 34th STREET (with purchase of $100 or more)

PRO TOOLS|HD

TDM-Based Software

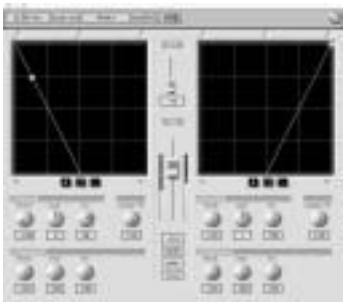


The Pro Tools TDM software included with every Pro Tools|HD system, features all of the recording, editing and mixing functions of the host-based Pro Tools LE software, plus adds a powerful set of additional features implemented to take full advantage of the HD systems dedicated DSP and enhanced I/O capabilities. For starters, because Pro Tools TDM software relies on the HD process card and the TDM bus and not your computer for processing tasks, you are guaranteed playback of up to 128 audio tracks when using an HD 2 system or higher (96 tracks are guaranteed with an HD 1). In addition, Digidesign's proprietary TDM plug-in architecture will allow you to instantiate numerous DSP intensive effects without affecting the load on your CPU. Pro Tools TDM software also supports complete surround sound capabilities allowing you to mix in every popular surround format available.

Four Types Of Plug-ins

- **TDM** plug-ins are exceptionally powerful realtime effects that utilize the DSP power provided by the HD Process cards in your Pro Tools system.of TDM technology rather than relying on the limited resources of the host computer. This affords much more processing power for DSP intensive plug-ins such as reverbs and surround processors.
- **HTDM** and **RTAS** plug-ins rely on your host computer's resources and are designed to run as efficiently as possible to offer the same and different processing options as TDM plug-ins. HTDM was designed as an optimum plug-in architecture for integrating realtime synthesis and sampling plug-ins within the Pro Tools environment.
- **AudioSuite** is Digidesign's proprietary non-real-time, file-based plug-in format. When an AudioSuite plug-in is applied to an audio region, a new audio file is created with the processing permanently intact while the original audio remains untouched on your disk drive.

Surround Mixing



- Pro Tools TDM software enables you to mix in every popular surround format — including LCRS, 5.1, 6.1, and 7.1. You can even work in several surround formats simultaneously.

Pro Tools|HD 1 systems provide recording and playback of 24-bit and 16-bit audio files with the following track counts:

- Up to 96 tracks at 44.1 kHz or 48 kHz
- Up to 48 tracks at 88.2 kHz or 96 kHz
- Up to 12 tracks at 176.4 kHz or 192 kHz

Pro Tools|HD 2 and HD 3 systems provide recording and playback of 24-bit and 16-bit audio files with the following track counts:

- Up to 128 tracks at 44.1 kHz or 48 kHz
- Up to 64 tracks at 88.2 kHz or 96 kHz
- Up to 24 tracks at 176.4 kHz or 192 kHz

Beat Detective



- Beat Detective automatically detects the tempo of a session and conforms an audio track or selection to that tempo by separating it into regions and aligning (quantize) it to the beats.
- Beat Detective lets you extract tempo and groove information, from audio performances, which can be applied to groove templates, called DigiGrooves. Beat Detective allows you to apply the attributes of these templates to other to audio selections. DigiGroove templates can also be used with Groove Quantize for MIDI data on all Pro Tools systems.

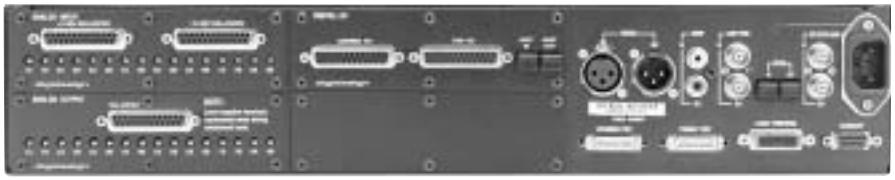
DIGIDESIGN

192 I/O

World-Class 24-Bit/192 kHz Multi-Channel Audio Interface



The 192 I/O is a 16 channel, high-definition, 24-bit/192 kHz, digital audio interface specifically designed to take full advantage of the increased fidelity and dynamic range capabilities available within the Pro Tools|HD environment. The 192 I/O features support for up to 16 simultaneous channels of inputs and outputs based on a wide range of available analog and digital interfaces including: 8 channels of high-definition, pristine quality analog I/O; 8 channels of AES/EBU; eight channels of TDIF; 16 channels of ADAT; and 2 additional channels of AES/EBU and S/PDIF digital I/O. Aside from its outstanding sonic specs, 192 I/O features the most flexible architecture ever offered in a Digidesign audio interface. With an I/O expansion bay, 192 I/O allows you to expand I/O options with one of three available expansion cards. Whether you want more analog inputs than outputs or more digital I/O. 192 I/O is the best sounding audio interface ever offered by Digidesign, rivaling interfaces costing more than twice its price.



FEATURES

Multi-Channel I/O

- 16 simultaneous discrete channels of high-definition I/O with Pro Tools|HD derived from 50 possible inputs and outputs.
- Eight channels of 24-bit A-to-D and D-to-A converters for superior analog input and output at sampling rates of 44.1, 48, 88.2, 96, 176.4, and 192 kHz.
- Eight channels of 24-bit AES/EBU digital I/O on a DB-25 connector with support for sampling rates of up to 192 kHz.
- An eight channel TDIF digital I/O port is provided on a DB–25 connector.
- Sixteen channels of optical I/O via two sets of eight channel ADAT Lightpipe I/O connectors. One optical pair can be switched to two channels of optical S/PDIF I/O.
- An additional AES/EBU digital I/O, using XLR conectors, is provided with support for sampling rates up to 96 kHz.
- 24-bit-capable coaxial S/PDIF digital I/O supporting sample rates of up to 96 kHz.
- Switchable, real-time sample rate conversion on the multi-channel digital inputs allows easy streaming of digital signals at any sample rate.
- Word Clock input and output for synchronizing 192 I/O with external Word Clock or 256x (Slave Clock) devices.

Metering

- 4-segment LED Meters to monitor input and output on each of the 16 channels.

Expansion

- Simultaneous use of up to eight 192 I/O units is supported, for a maximum of 96 channels of I/O at 96 kHz.
- An DigiLink Expansion Port is provided for direct connection of an additional 192 I/O or 96 I/O.
- Optional addition of cards to expand analog or digital I/O.
- A Legacy Port is included for Digidesign MIX-series audio interfaces.

Soft Clip Limiting

- When using the A-to-D converters, a switchable Soft-Clip Limiter algorithm allows you to achieve higher signal levels to disk helping eliminate clipping transients that can cause digital distortion while the same time offering the warmth and compression characteristics of analog tape saturation.
- With Soft Clip enabled, the 192 I/O supports an additional 4 dB of headroom by rounding off the top 4 dB to the clip point.

DigiLink Cables

- A 1.5´ DigiLink cable and 1.5´ BNC cable is included with each interface.



192 DIGITAL I/O

World-Class 24-Bit/192 kHz Multi-Channel Audio Interface



The 192 Digital I/O is a high-definition audio interface designed for those of you using a digital mixer or third party A-to-D converters with ProTools|HD. The 192 Digital I/O provides you with up to 16 channels of AES/EBU, TDIF, and ADAT I/O via digital I/O cards. The 192 Digital I/O employs the same comprehensive front panel status and metering capabilities, realtime sample rate conversion, expansion ports and additional 2 channel AES/EBU I/O (XLR), coaxial S/PDIF I/O and ADAT I/O that are available on the 192 I/O. And with two Multi-channel expansion bays still available, you are able to add more I/O down the road.



SAME FEATURES AS THE 192 I/O EXCEPT —

Multi-Channel I/O

- 16 discrete channels of digital input and output
- 16 channels of 24-bit AES/EBU I/O at sampling rates of up to 96 kHz in single-wire mode; or 8 channels at sampling rates of up to 192 kHz in dual-wire mode
- 16 channels of 24-bit TDIF I/O at sampling rates of up to 48 kHz
- 16 channels of 24-bit Optical (ADAT) I/O at sampling rates of up to 48 kHz

Comparison	96 I/O	192 I/O	192 Digital I/O
Multi-Channel I/O Cards:			
Max Simultaneous I/O	16	16	16
Analog Inputs	8 channels up to 192kHz (1/4″ TRS)	8 channels up to 192kHz (DB25)	–
Analog Outputs	8 up to 96kHz (1/4″ TRS)	8 up to 192kHz (DB25)	
AES/EBU I/O (DB25) AES/EBU I/O (DB25)	– –	8 channels up to 192kHz (single wire mode) 4 channels up to 192kHz (dual wire mode)	16 channels up to 96kHz (single wire mode) 8 channels up to 192kHz (dual wire mode)
ADAT Lightpipe I/O	8 channels up to 48kHz	8 channels up to 48kHz	8 channels up to 48kHz
T/DIF I/O	–	8 channels up to 48kHz	16 channels @ 48kHz
Available I/O Bays		1	2
EnclosureConnectors			
Coaxial S/PDIF I/O	2 channels up to 96kHz	2 channels up to 96kHz	2 channels up to 96kHz
AES/EBU I/O (XLR)	2 channels up to 96kHz	2 channels up to 96kHz	2 channels up to 96kHz
ADAT Lightpipe I/O	8 channels up to 48kHz switchable to 2 channel S/PDIF	8 channels up to 48kHz switchable to 2 channel S/PDIF	8 channels up to 48kHz switchable to 2 channel S/PDIF
DigiLink ports	1 primary, 1 expansion	1 primary, 1 expansion	1 primary, 1 expansion
LOOP SYNC In and Out	BNC	BNC	BNC
EXT. CLOCK In and Out	BNC supports Word Clock or 256x (Slave Clock) devices	BNC supports Word Clock or 256x (Slave Clock) devices	BNC supports Word Clock or 256x (Slave Clock) devices
AC Power	Auto Power-selecting (100V to 240V)	Auto Power-selecting (100V to 240V)	Auto Power-selecting (100V to 240V)

DIGIDESIGN

96 I/O

High-Quality, 24-Bit/96 kHz Multi-Channel Audio Interface



The 96 I/O offers Pro Tools users a high-resolution, multi-channel audio interface with excellent sonic performance. An affordable 16-channel audio interface for Pro Tools|HD, 96 I/O features a wealth of I/O options, including 8 channels of high-definition analog I/O, 8 channels of ADAT optical I/O, 2 channels of AES/EBU and S/PDIF I/O, and Word Clock I/O. Over the past few years, Pro Tools|24 MIX systems have offered professionals a comprehensive and completely flexible, proven solution for music, post production, broadcast and multimedia applications. Now, Digidesign has topped Pro Tools|24 MIX systems on every level with Pro Tools|HD: the new, high-definition Pro Tools digital audio workstation. With Pro Tools|HD, Digidesign brings professionals the best sound quality ever in a Pro Tools system, along with twice the power of previous MIX systems, increased expandability options and much more. Among the new audio interface options available for Pro Tools|HD, 96 I/O offers a high definition, multi-channel audio interface at a low price point.



FEATURES

- Eight channels of 24-bit analog inputs and outputs using balanced 1/4″ TRS connectors
- Eight channels of ADAT optical I/O switchable to a optical S/PDIF pair
- Stereo AES/EBU (XLR) and stereo S/PDIF (coaxial) digital I/O as well as Word Clock I/O (BNC)
- Support for 96, 88.2, 48 and 44.1 kHz sample rates conforming with the latest standards of professional digital audio production.
- All channels can be configured on a per pair basis allowing you to select up to 16 channels of simultaneous I/O from the 20 available I/O channels
- The Expansion Port on the rear panel allows you to use the supplied DigiLink cable to connect another 96 I/O for 16 more channels of high-definition input and output.
- Legacy Peripheral port supports connection of 888|24 I/O, 882|20 I/O, 1622 I/O, or 24-Bit ADAT Bridge I/O at sample rates up to 48kHz

Specifications	192 I/O	96 I/O
A/D		
Sample Rate:	44.1, 48, 88.2, 96, 176.4, 192 kHz ±10%	44.1, 48, 88.2, 96 kHz ±10%
Dynamic Range:	120 dB (A-weighted), 118 dB (unweighted); see Notes 1 and 2	115 dB (A-weighted), 112 dB (unweighted); see Notes 1 and 4
THD+N:	0.00035% (-109 dB); +21 dBu @ 997 Hz; see Note 1	0.0007% (–103 dB); +17 dBu @ 997 Hz; see Note 1
Frequency Response:	±0.05 dB @ +2 dBu, 20 Hz – 20 kHz; see Notes 1 and 3	±0.05 dB @ –2 dBu, 20 Hz – 20 kHz; see Notes 1 and 2
D/A		
Sample Rate:	44.1, 48, 88.2, 96, 176.4, 192 kHz ±10%	44.1, 48, 88.2, 96 kHz ±10%
Dynamic Range:	118 dB (A-weighted), 115 dB (unweighted); see Notes 1 and 4	114 dB (A-weighted), 112 dB (unweighted); see Notes 1 and 3
THD+N:	0.00056% (-105 dB); -1 ' @ 997 Hz; see Note 1	0.0013% (–97 dB); –1 dBFS @ 997 Hz; see Note 1
Frequency Response:	±0.05 dB, -20 dBFS, 20 Hz – 20 kHz; see Notes 1 and 3	+0.1/–0.2 dB; –20 dBFS, 20 Hz – 20 kHz; see Notes 1 and 2

Sample Rate = 48 kHz, Noise BW = 22 Hz – 20 kHz unless otherwise noted, Tambient = +25 C

Note 1: Measurement made using +4 dBu inputs or outputs

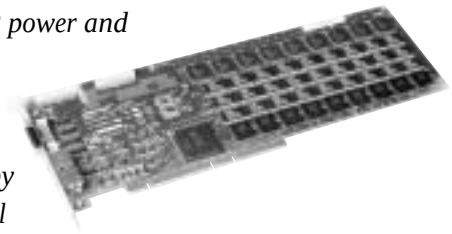
Note 2: ADC measured with analog input at -38 dBu @ 997 Hz

Note 3: Measured relative to level at 1 kHz

Note 4: Measured with digital input at -60 dBFS @ 997 Hz

PRO TOOLS HD ACCESSORIES

HD Process Cards



Boosting the DSP power and I/O capacity of your Pro Tools|HD system can be achieved by adding additional HD Process cards. Up to six cards may be added to an HD Core system (a total of 7 cards), each providing 32 more channels of I/O capacity (up to 96 channels total), and dramatically increasing the system's mixing and plug-in processing power.

DigiSnake Cables –
Customized Audio Cabling for HD Systems



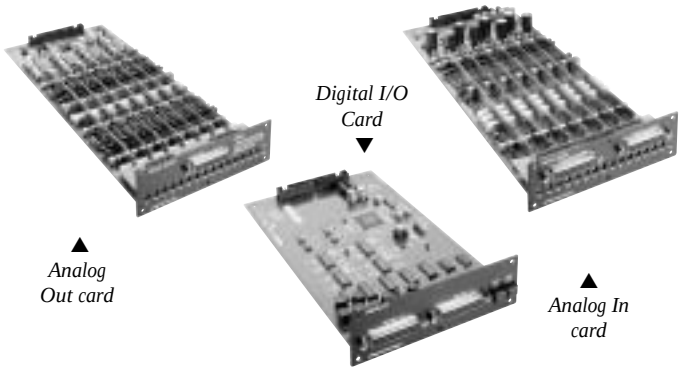
Digidesign offers an array of quality cabling options to ensure the fidelity of audio signals as they travel between the I/O components of your Pro Tools system. Uniquely designed for Digidesign gear, DigiSnakes are custom colored and labeled, feature a lifetime warranty, and are comprised of 110 ohm "digital grade" cable to accommodate either digital or analog signals. DigiSnakes make connecting your gear a quick, easy process, freeing you to spend your valuable time on your audio projects rather than the wires they travel across.

DigiSnakes are available in 4´, 12´ and 25´ options:

DB25-DB25 DigiSnake, 12´ **70.00**
DB25-DB25 DigiSnake, 25´ **95.00**
DB25-DB25 DigiSnake, 4´ **65.00**
DB25-DB25 TDIF Only DigiSnake, 12´ **150.00**
DB25-DB25R ProControl DigiSnake, 25´ **95.00**
DB25-TRS DigiSnake, 12´ **95.00**
DB25-TRS DigiSnake, 4´ **90.00**
DB25-TS +4 dBu/-10 dBV Padded DigiSnake, 12´ **115.00**
DB25-TS +4 dBu/-10 dBV Padded DigiSnake, 4´ **110.00**
DB25-XLR M+F AES/EBU DigiSnake, 12´ **95.00**
DB25-XLR M+F AES/EBU DigiSnake, 4´ **90.00**
DB25-XLRF DigiSnake, 12´ **95.00**
DB25-XLRF DigiSnake, 4´ **90.00**
DB25-XLRM DigiSnake, 12´ **95.00**
DB25-XLRM DigiSnake, 4´ **90.00**
ProControl DigiSnake Kit **345.00**

192 I/O Expansion Cards

Thinking of adding more analog inputs, outputs or digital connections to your 192 I/O? The 192 AD expansion card provides eight more channels of high definition analog input while the 192 DA card adds eight channels of analog output. To increase your digital I/O options, the 192 Digital card adds eight more channels of AES/EBU, TDIF and ADAT I/O.



DigiLink Cables



Lightweight DigiLink cables provide connectivity between HD audio interfaces and HD Core or HD Process cards, and are also used to daisy chain 192 I/O and 96 I/O units. DigiLink cables allow you to keep the computer in a soundproof location up to 100´ away from your audio interfaces.*

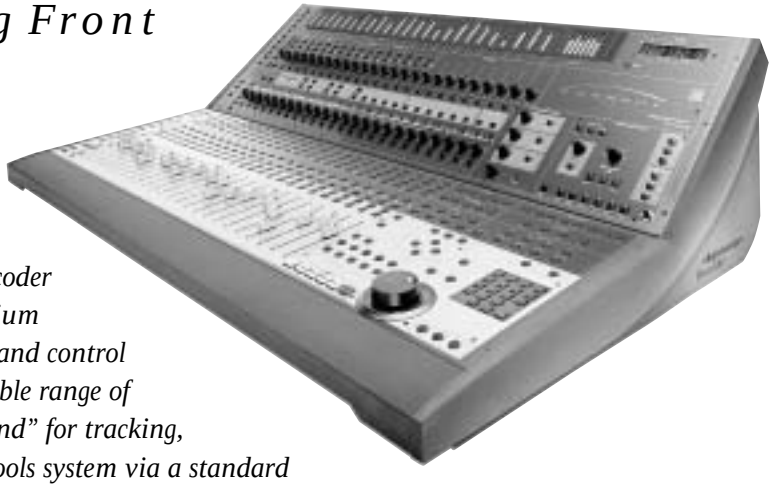
DigiLink Cable 1.5´ **50.00**
DigiLink Cable 100´ **295.00**
DigiLink Cable 12´ **65.00**
DigiLink Cable 25´ **95.00**
DigiLink Cable 50´ **150.00**

** 100´ DigiLink supports up to 96 kHz sample rates*

DIGIDESIGN

CONTROL|24

Control Surface and Analog Front End for Pro Tools TDM



Control|24 is a dedicated control surface, co-developed by Digidesign and Focusrite, that offers hands-on access to nearly every recording, mixing, automation and editing function in Pro Tools. Features include 24 channel strips, each featuring a dedicated rotary data encoder and touch-sensitive motorized fader, along with 16 premium microphone preamps, a comprehensive transport control and control room monitoring sections, and a line submixer. This flexible range of audio routing features make Control|24 an ideal "front-end" for tracking, monitoring or mixing. Control|24 connects to your Pro Tools system via a standard Ethernet connection which delivers maximum responsiveness and resolution. Whether you are new to Pro Tools or a seasoned pro, the extensive features and affordable price of Control|24 make for an extremely cost-effective, and powerful environment for recording, editing, and mixing audio and MIDI.

FEATURES

24 Channel Control Strips

- Each channel has a motorized, touch-sensitive fader and a multi-function knob (which defaults to pan)
- Dedicated EQ and Dynamics switches on every channel
- Dedicated illuminated switches for Mute, Solo, Record Arm, Channel Selects, and Automation Mode on every channel.

Recording Mix Automation

- Control|24 provides touch-sensitive access to the dynamic automation of faders, pans, fader mutes, send levels, send mutes and plug-in Parameters available to Pro Tools.
- MIDI volume and pan, as well as channel mute, can also be automated for additional mix capability.
- Control|24 replicates Pro Tools on-screen automation features, with channel-based and session-wide automation controls.
- AUTO switches found in each channel strip allow you to select the desired channel automation mode (Read, Write, Trim, Latch, Touch, and Off). The current automation state is indicated using the LED labels to the right of each AUTO switch.



16 Focusrite Class A Mic Preamps

- 16 high-quality preamplifiers accessible via XLR mic, 1/4″ line connectors, make an ideal analog front end for your Pro Tools system. Inputs 1 and two also feature D.I. (direct inject) inputs for plugging in your bass or guitar.
- The mic pre design, based on the same preamps found in the Platinum line of analog processors from Focusrite, delivers smooth sound across a wide frequency bandwidth and dynamic range with extremely low noise and distortion.
- The gain range for mic level signals is from 0 to +60 dBu (Mic). A 12 dB/octave high-pass filter at 75 Hz is also provided.
- Phantom power is switchable in two groups of eight channels from the rear panel.
- The Preamps are output interface easily with your Pro Tools interface via two streamlined 8 channel DB25 connectors.



24 stereo, 14-segment LED meters provide global pre-fader or postfader levels of individual channels.

Choose your input source, arm tracks, control input gain and the 12 dB/octave high-pass filter at 75 Hz for each input channel.

26, four-character scribble strips provide system feedback such as channel names and plug-in parameters.

CONTROL|24

Transports and Navigation

Group function buttons: Create; Enable; Edit; Delete; and Suspend.

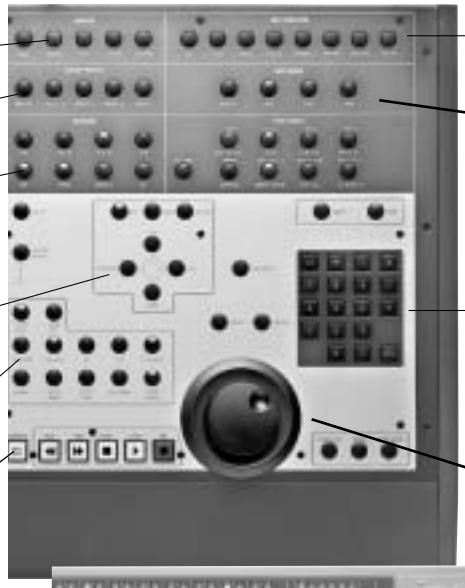
Store and recall five Zoom presets

You're a single button push away from the various Pro Tools windows: Mix, Edit, Transport, Plugins, etc...

Navigation and Zoom functions The UP, NEXT, DOWN, and PREVIOUS arrow buttons provide roughly the same functionality as the arrow keys on your computer keyboard as well as some additional features. In Zoom mode, the quadrant buttons are used to zoom in and out and adjust the perceived amplitude of the waveform display in the Edit window on-screen.

The Control Section can access special transport functions such as Loop modes, Quick Punch, Pre/Post Roll, as well as machine control status.

Standard transport controls and a talkback button



Direct access Pro Tools' most important editing functions: Cut, Copy, Paste, Delete, Separate, Capture, Duplicate and repeat

Edit Mode buttons: Shuffle, Slip, Spot and Grid

Edit Tools for audio and MIDI are all a button click away.

The Numeric Keypad mimics keypad operation in Pro Tools and is used for memory location store and recall, data entry of counter values or numerics, for completing an editing operation (using the ENTER key), and for Pro Tools Transport and Shuttle modes.

The Scrub/Shuttle wheel can be used for scrubbing and shuttling, to help define regions, and other edit operations.

Submixer / Monitor Section

The integrated submixer consists of eight stereo inputs that can serve a wide variety of purposes including: submixing returns from MDMs, routing synthesizers, samplers, or drum machines, or returning aux busses. You can also use the submixer to bring multiple input channels down to a stereo pair, to be routed into a pair of Pro Tools audio interface inputs.



High-resolution LED display for transport location at a glance

Comprehensive Control Room section capable of up to 5.1 surround monitoring with TDM system.

	ProControl	Control|24	Motor Mix
Touch-Sensitive Faders	Yes	Yes	No
# of Motorized Faders	8 – 48	24	8 – 32
Expandable	Up to 48 Faders plus Edit Pack	No	Up to 32 Faders
Built-In Mic Preamps	None	16	None
Surround Monitoring	Up to 5.1	Up to 5.1	None
Supported Systems	Pro Tools TDM only	Pro Tools TDM only	Pro Tools TDM & LE
Connection Type	Ethernet	Ethernet	MIDI

DigiSnake Kits for Control|24

Digidesign offers two optional, complete DigiSnake Kits for Control|24 that include seven custom-labeled, custom color-coded cable snakes — one with D-Sub to XLR/TRS connectors, and one with D-Sub to TRS connectors — allowing you to quickly and easily connect this versatile control surface to your Pro Tools system.

(D-Sub to XLR-TRS, or D-Sub to TRS)

Control|24 TRS DigiSnake Kit (Complete, TRS-only Custom Cable Kit Package for Control|24) **695.00**

Control|24 XLR/TRS DigiSnake Kit (Complete Custom Cable Kit Package for Control|24) **695.00**

DIGIDESIGN

PROCONTROL

Control Surface System For ProTools TDM



ProControl is Digidesign's most comprehensive control surface that adds a high-quality tactile mixing and editing controls to Pro Tools TDM systems. ProControl has the look and feel of a high-end mixing console, with dedicated channels strips, transport controls and a mains section that will allow you to exploit the extensive recording, editing, mixing, DSP processing and automation capabilities that Pro Tools has to offer. ProControl's eight channel strips are centered around exceptionally smooth DigiFaders: Digidesign's patented 100mm motorized digital faders which can be configured to control volume and send levels for mono, stereo and surround channels. ProControl also features a comprehensive analog monitoring section that supports stereo and surround sound monitoring with audio fed from your ProTools interfaces directly into ProControl's analog I/O section. ProControl's flexible, modular design allows you to integrate up to 48 physical faders using additional eight channel Fader Packs while the optional Edit Pack provides tactile surround mixing via two motorized DigiPanners as well as a color-coded QWERTY keyboard.

FEATURES

- The most comprehensive tactile access to all of Pro Tools editing, mixing and automation parameters.
- Eight bank-switchable channel strips expandable to a total of 48 channel strips with optional Fader Packs.
- The DSP Edit/Assign section allows you to view and edit plugins, sends, and inserts, or perform surround panning, using an intuitive array of rotary encoders and displays.
- Dedicated stereo and surround sound control room monitoring and routing section with onboard analog I/O for routing monitor signals between ProControl and ProTools|HD interfaces.
- Automation Switch Group lets you effortlessly switch between Write, Touch, Latch, Trim, and Read modes for fader levels, pans, sends, mutes, plug-in parameters, and more.
- High-quality, illuminated transport control switches, weighted Scrub/Shuttle Wheel
- Connectivity to Pro Tools TDM systems via high-speed Ethernet.
- Cross platform for Mac OS and Windows TDM systems



Faders and Channel Controls

- The are a total of eight, bank-switchable channel strips each with dedicated fader, multi-function rotary encoder and a variety of illuminated switches. The flexible mixing environment of Pro Tools lets you customize any channel strip for mono, stereo, or surround operation.
- 100 mm. touch-sensitive DigiFaders for controlling channel volume or send level.
- Data encoder with 15-LED indicator rings for pan, send and I/O control.
- Large, illuminated Solo and Mute buttons
- Illuminated Channel Select buttons for I/O assignment, automation, grouping and other channel-specific edit functions.
- Dedicated, illuminated switches for EQ and Dynamics editing/bypass control, insert assignment/bypass and record-ready states
- Each channel strip also features an 8-segment LED scribble strip for displaying channel name and headroom.
- Another 8-segment LED dot-style scribble strip displays parameter values of the data encoder data.

DigiFader Touch-Sensitive Moving Faders

- The patented DigiFader is a true digital fader that incorporates a sealed encoder, servo-controlled motor, and "flex-circuit." This unique design provides the feel, performance, and reliability of moving faders found on other high-end mixing consoles.
- Each 100mm DigiFader provides 10-bits of resolution or 1024 steps; Pro Tools then provides 24-bit interpolation of these values, to ensure accurate reproduction of fader moves.
- The use of advanced touch-sensitive circuitry allows you to means all you have to override servo control by merely touching the fader.

Meter Section

- Eight LED meters for Input/Track level indication
- Time Counter Display
- Six LED meters for master output level

PROCONTROL

Fader Pack



- Fader Pack features everything found on the fader section of the ProControl Main Unit.
- Up to five Fader Packs can be added to ProControl, offering simultaneous access to a total of 48 channel strips.

Main Section

- The Main section features standard transport controls; and a Scrub/Shuttle wheel; as well as Audition Mode switches that allow you to quickly listen to the selection start or end, with or without pre- and post-roll.
- Dedicated Window switches provide direct access to the desired Pro Tools window.
- Fourteen dedicated edit buttons are provided for accessing Pro Tools various editing Modes, Tools and Functions. Dedicated save and undo witches are also provided.
- The Channel Matrix is a multi-purpose channel status and navigation control section, consisting of 32 primary switches, that provides a number of unique capabilities to Pro Tools including: Bank and channel navigation; Shortcuts, commands, text, and data entry; Channel Solo, Mute, Record, and Select status; Plug-in parameter page selection; and Group selection, creation, and editing.
- An integrated Trackpad is provided for times that you require mouse like control on the Pro Tools graphical user interface.
- Two Bank Select switches along with a Nudge function switch allow you to easily scroll though banks of eight channels or select adjacent channels.
- Four Zoom/Select switches – labeled up, down, previous next – provide the same functionality as the arrow keys on your computer keyboard for zooming in on audi and MIDI regions as well as navigating through and selecting various functions.
- A Numeric Keypad is also provided for memory location store/recall, data entry of counter values and any other function that is available to your computer's keyboard.

Analog Monitor and Control Room Sections

- To harness the powerful and flexible routing capabilities of ProTools, ProControl provides a number of analog I/O connectors that can be connected directly to your ProTools|HD I/O interfaces and controlled via the dedicated monitoring section.
- Two eight-channel DB25 input connectors supply ProControl with Pro Tools audio interface output signals, and external signal busses.
- An eight-channel DB25 output connector supplies ProControl monitor outputs for main speakers, alternate speakers, cue mix systems, mastering decks, and other destinations.
- The Control Room Monitoring section supports stereo and surround monitoring with volume control, source and speaker selection, mute, dim, mono, talkback, listenback, and signal routing options.

DSP Edit/Assign Area



The DSP Edit/Assign section is dedicated to tapping in to Pro Tools' powerful DSP processing environment, by offering intuitive editing and total recall of all plug-in parameters. This area provides 26 illuminated switches plus eight data encoders and scribble strips for assigning and editing plug-ins.



Digidesign also offers an optional, complete cable kit for ProControl that includes 3 custom-labeled, color-coded cable snakes — allowing you to quickly and easily connect this world-class control surface to your Pro Tools system.

Edit Pack – *Dedicated Editing and Surround Mixing Unit for ProControl*



- Two DigiPanners — motorized, touch-sensitive panning joysticks based on patented DigiFader technology
- Twenty dedicated dual-function switches
- Eight high-resolution, 40-segment, dualchroma meters
- Three eight-character, seven-segment numeric displays for region or selection Start, End and Length values
- Color-coded, QWERTY keyboard with support for USB, ADB and PS-2 connections
- Ergonomic, two-button 1.5″ trackball
- Comprehensive Machine Control section with eight track-arming switches and complete Edit Mode control
- Host computer and ProControl connection via high-speed Ethernet

DIGIDESIGN

MOTOR MIX

Compact, Expandable Control Surface For Pro Tools TDM and Pro Tools LE Systems



Motor Mix is a compact and low-cost control surface that offers real time, hands-on control of any Pro Tools system including transport control, track arming, motorized faders, mute/solo switches, pan/send control, plug-in editing and more for Pro Tools and other computer-based digital workstations. Each unit features eight 100mm motorized faders, and eight rotary encoder knobs for controlling panning, EQ, auxiliary send levels, Plug-In settings and more. Nearly every aspect of a mix is accessible using Motor Mix including automation functions. A 2 line, 40 character LCD display reflects the Motor Mix's current operating status including rotary pot settings and effects parameters; channel assignments, levels and pan settings etc... Up to four Motor Mix units can be linked together, providing a total of 32 simultaneous faders. Motor Mix's small footprint, comprehensive feature set and expandability makes it a cost effective solution for smaller editing suites, home project studios and hybrid audio/video environments.

FEATURES

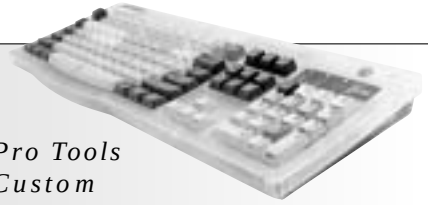
Eight Channel Strips

- Each of the eight channel strips incorporates a 100mm long-life Motion Sensing motorized fader as well as a high-quality, continuously variable rotary pot for control of panning, aux sends, dynamics, plug-ins and more. A detented rotary selector with push button, accompanied by a dual 7-segment LED display, provides control over the rotary pot functionality.
- Each channel also features dedicated, backlit Channel Select, Solo and Mute buttons.
- Four View buttons provide access to an unlimited number of mixer channels. Channels can be switched in banks of 8 channels or via group assignments up to 8 channels.
- Eight Multi Switches, with green LED status indicator, provide access to eight different channel functions including channel eq, dynamics, delay and more.
- Eight "Burn" Switches with red LED status indicator can be assigned to control channel audio recording, automation enables, or other commands you can embed into project files.

Transport Controls and Function Switches

- Eight System Keys, flanking the left side of the unit provide access to a number of functions including: automation enable and mode selection; group creation; plug-in assignment, control and compare; insert control; Pro Tools window navigation; edit tool selection; as well as Save and Undo and more
- The Eight System Keys on the right side of the unit provide access to a full range of transport controls including Play, Stop, Fast Forward and Rewind; input/output assignments; locate functions; monitor and audition functions for pre/post roll and in/out punch points; as well as enter and escape functions and more.

- A highly visible 40x2 dot LCD display provides a readout for channel assignments and rotary pot settings such as PAN, Aux Send Level, effect parameters, input/output assignments, channel level meter, and soft key control.
- MIDI In and Out connectors provide bi-directional communication between Motor Mix and your computer's MIDI interface.
- Expandable – link up to four units for a total of 32 fader tracks with Pro Tools. Motor Mix intelligently switches in banks up to 24 channels wide when using an expanded system.
- Small footprint (10½″ wide x 12½″ deep) allows MotorMix to fit in space-limited project studios.
- An attractive dual slope chassis design provides easy viewing with an ergonomic feel.

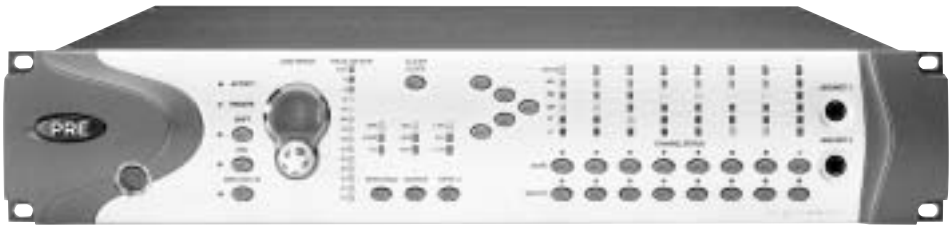


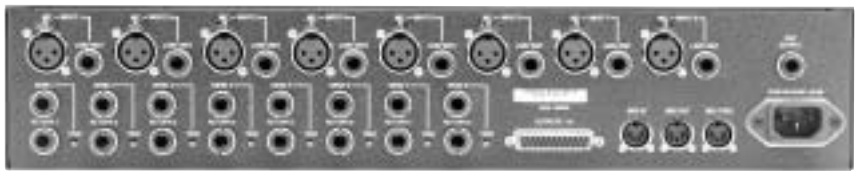
Pro Tools Custom Keyboard

This USB computer keyboard features all of the Pro Tools key commands and shortcuts printed directly on its color-coded key caps, along with all the conventional QWERTY keyboard letters and symbols.

PRE

Remote Controllable, World-Class 8-Channel Mic Preamp





PRE is an eight channel, remote controllable microphone preamplifier designed to complement the 192 kHz capabilities and world-class converters available to Pro Tools|HD systems. Featuring 8 discrete, matched-transistor, hybrid mic-preamp circuits, PRE can accept nearly any input signal, easily accommodating mic, line, and direct instrument (DI) level inputs on all eight channels. PRE's MIDI In, Out and Thru ports allow you to control PRE's front panel functions remotely via the Pro Tools software interface or Digidesign control surfaces. This allows you to place PRE anywhere in your studio (closer to the mic source) as well as instantly recall settings with your session. All of PRE's function are also available in stand-alone mode, making the same high-end sonic quality available for non-Pro Tools applications and all standard MIDI controllers.

FEATURES

Inputs and Outputs

- Eight high-performance, discrete matched transistor hybrid microphone pre-amplification circuits can accommodate nearly any input source.
- Each channel features support for XLR mic as well as 1/4″ direct instrument (DI) and line level inputs on the rear panel. Two of the 1/4″ line/inst inputs are also duplicated on the front panel for easy access.
- Input gain for each channel is variable from 0 dB to +69 dB, in approx. 3 dB steps.
- Switchable 75 Hz @ 18 dB/octave high pass filter, phase (polarity) reverse, 48 V (phantom power) and an 18 dB pad for each channel.
- Selectable impedance matching for optimizing mic, line or instrument inputs
- Separate 1/4″ send and return insert points are provided for each inputs allowing you to access your favorite outboard processor.
- All eight channels are output via DB25 connector. This allows easy, uncluttered interfacing between PRE and your Pro Tools interface or other mixer/recorder.

Remote Control

- MIDI In/Out/Thru ports accommodate remote operation via Pro Tools software and Digidesign control surfaces.
- Remote control via Pro Tools allows you to instantly recall all settings on a session by session basis and create mix templates with all routing and control.
- Remote operation also allows you to keep PRE near the audio source and away from the computer environment.

Additional Features

- A built in oscillator, with a 1/4″ output connector offers a convenient way of calibrating to other peripherals, such as the 96 I/O and 192 I/O audio interfaces.
- All features are also supported in stand-alone mode, including remote control via MIDI, making PRE an ideal analog front end for any recording system.
- Up to nine PREs are supported through software for up to 72 channels of remote-controlled Mic Preamp channels.

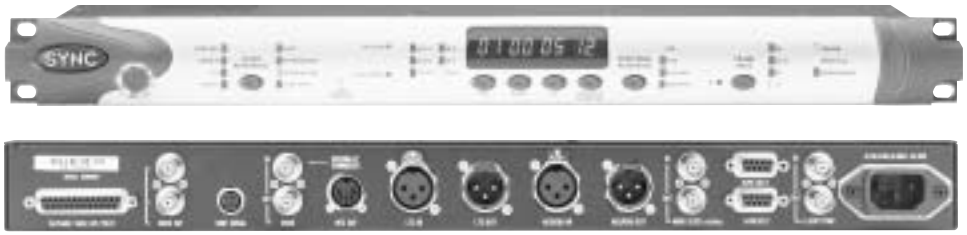
Specs

INPUT STAGE	
Circuit Topology:	Discrete Matched Transistor Hybrid
Input Gain:	–18 to +66 dB in 1 dB steps
Mic E.I.N (unweighted):	–127 dB (60dB gain) 150Ω source
THD+N (@ 1 kHz):	0.001% @ +10 dBu 0 dB gain
CMRR:	> 80 dB @ 60 Hz, > 72 dB @ 10 kHz
Crosstalk:	< –100 dB @ 40 dB gain, 1 kHz
Frequency Response:	(–3 dB) 5 Hz – 1 MHz
Frequency Response:	±0.5 dB @ 40 dB gain 10 Hz – 100 kHz
Max: Input Level:	(any type) +45 dBu
Input Impedance:	Mic 1.5 k; Line 10 k, Inst/D.I. 1.5 Mohms
OUTPUT STAGE	
Max Output:	+26 dBu, balanced 600 Ω, DC coupled
Output Impedance:	~50 ohms
Nominal Level:	+4 dBu
Trim level:	–10 dB

DIGIDESIGN

SYNC I/O

High-Quality, Versatile Synchronizer for Pro Tools



SYNC I/O is a multi-purpose synchronization device that supports all industry-standard clock sources and time code formats. Accommodating the high sample rates intrinsic to Pro Tools|HD with the utmost accuracy, SYNC I/O features near sample-accurate lock to time code or bi-phase/tach signals and a 192 kHz capable, high-fidelity, low-jitter Word Clock. AES/EBU clock I/O, video reference in/thru, video program in/out, and a host of other features make SYNC I/O the most consistent, dependable solution for Pro Tools|HD installations in commercial music and post production facilities.

FEATURES

- High-fidelity, low-jitter clock design provides maximum audio fidelity, even when locked to varying external timing sources (such as when generating Word Clock from free-running SMPTE time code).
- Extremely fast lock up time while maintaining near sample-accurate lock at sample rates up to 192 kHz.
- Word Clock I/O up to 192 kHz capable
- Time code formats supported include LTC, VITC, bi-phase and MTC out.
- Clock sources supported include video reference (blackburst), video signal, LTC resolve, AES, word, bi-phase, and internal.
- While locked to clock or positional reference formats, SYNC I/O simultaneously generates LTC, VITC, MTC, Super Clock, Word Clock and AES Null Clock.
- Supports direct translation of bi-phase and tach pulses to the various output formats, allowing time code/clock generation from mag machines and flatbed editors. The reference lock point for the signals can be set either from the front panel or from the SYNC I/O Setup application.
- Supports industry-standard pull-up/pull-down rates for film/video
- Two 9-pin ports allow dual device control when using Machine Control option in master or remote slave mode.
- Large, bright, and easy-to-read 7-segment time code display along with source and status lights, allowing quick and easy access to reader/generator functionality.

Local and Remote Modes

- Local Mode allows front panel control of all parameters, including clock references and time code reading or generation.
- Remote Mode allows you to control SYNC I/O's parameters via Pro Tools or the SYNC I/O Setup application. SYNC I/O switches to Remote Mode automatically when either of these applications are launched.
- Using auto-configuration, all settings can be saved and instantly recalled with each session.
- Add visual time code window burn to video signals passed through the video I/O. Positioning and placement of the window burn can be controlled from Pro Tools or SYNC I/O setup software.

DigiDrive FireWire 80—Pro Tools-compatible 80GB FireWire Hard Drive



Process, store and retrieve more information more affordably than ever before. The 80 GB DigiDrive FireWire 80 supports 24 tracks of 24-bit/44.1 kHz audio per drive — and may be connected to a second drive for a total of 48 tracks. It's as easy as hooking up a FireWire cable.

- Rapid throughput via FireWire connectivity
- Certified compatibility with Pro Tools (Mac OS and Windows)
- Designed for both removable rack-mounting (with optional rackmount kit) and interlocked stacking, the self-contained DigiDrive enclosure lets you choose the configuration that best fits your system. Plug & play makes setup simple, and FireWire drives can be hot-swapped without rebooting the host computer.
- Digidesign's exclusive QuietDrive technology dramatically reduces drive noise by up to 20 dB to a whisper. DigiDrive also keeps your data cool and safe with advanced thermal sensing circuitry that enhances cooling efficiency by controlling fan speed. There's even a built-in heat detection/indicator circuit to alert you to potential problems before either the drive or your critical data are threatened.

DIGIDRIVES

Storage Systems for Pro Tools



In a world crowded with generic hard drives, DigiDrives takes the guesswork out of finding the perfect mate for your Digidesign system. Optimized for certified compatibility with Pro Tools, DigiDrives deliver the speed, capacity, versatility and reliability to handle your most demanding projects. And, you can be sure that your investment in a DigiDrive will be a safe one. Dealing directly with Digidesign customer support including guaranteed next business day replacement in most locations will help to keep your sessions on track.

DigiDrive 36|10k and 73|10k

Designed for both removable rack-mounting and interlocked stacking, the fully self-contained DigiDrive enclosure allows you to choose the configuration that fits your system best. It also features Digidesign/Avid's exclusive QuietDrive technology, which dramatically reduces drive noise to a whisper (reduces drive noise by up to 20 dB), so you can listen to your audio instead of your gear.

- Fast 10,000 RPM hard drives (36 and 73 GB capacities) offer transfer rates of up to 160 MB/second, and super-fast seek time of 5.1 ms read and 5.2 ms write.
- The 36GB drive stores 6,800 track minutes of 16-bit/44.1kHz (Red Book) audio, which gives 280 minutes of 16-track recording. At 24-bit resolution, handles 4,400 track minutes, enough for 180 minutes of 24-track recording. (The 73GB doubles all of this).
- Certified compatibility with Pro Tools (Mac OS and Windows). DigiDrive Tuner application optimizes drives for Pro Tools
- Support Fast/Wide, LVD and Ultra160 SCSI standards
- Quick-Release rack-mount or interlocking vertical stacking
- Non-proprietary docking means audio-anywhere portability
- Built-in heat detection/indicator circuit alerts you to potential problems before either the drive or data are threatened.
- Advanced thermal sensing circuitry protects data integrity and also keep your data cool and safe by controlling fan speed.

DigiDrive MediaDock II

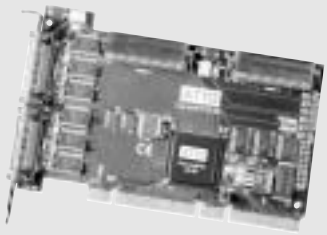


Maximize the efficiency and versatility of your Pro Tools production environment with the DigiDrive MediaDock line — a removable storage system for post production, professional and project studios. MediaDock provides an optimal solution for data transfer, portability, and backup, streamlining workflow while protecting your work. Products include MediaDock Shuttle|36, MediaDock Shuttle|73 and MediaDock II chassis.

- Incorporating high-performance LVD technology, MediaDock II's compact 2U chassis provides two bays that support DigiDrive MediaDock Shuttles as well as legacy and current LVD shuttles. Both bays have their own SCSI bus with two 68-pin connectors and front-mounted SCSI ID selection switches.
- Front-panel LEDs monitor power, temperature and shuttle functions, while a variable-speed, temperature-sensing fan and an internal auto-ranging power supply ensure that your projects stay up and running. Along with the included rack-mount kit, DigiDrive MediaDock II ships with either two 36 GB or two 73 GB DigiDrive MediaDock Shuttles.

SCSI|128 Kit

SCSI solution for Pro Tools|HD



- SCSI|128 offers 128 tracks of record and playback with Pro Tools|HD on both Mac and Windows when used with LVD drives.
- Built-in LVD support results in improved data integrity, and allows for cable lengths up to 12.5 meters per bus accommodating strategic placement of your LVD-supported SCSI drives.
- Includes ATTO UL3D Dual Channel Ultra3 SCSI Host Adapter and two cables, pre-tuned for optimal performance with Pro Tools|HD.
- UL3D's two external channels handle throughput up to 160 MB/second per channel.
- The SCSI|128 kit also supports up to 64 audio tracks on Pro Tools|24 MIX and d24 systems on both Mac and Windows.

DigiDrive RackMount Kit

- The 2RU DigiDrive Rackmount Kit allows you to quickly and easily slide in or remove up to 2 DigiDrive hard drive units, making it easier than ever for you to transfer your sessions back and forth between different Pro Tools systems.

MIDI I/O

Professional Multi-Port MIDI Interface



The MIDI I/O is a multi-port MIDI interface designed by Digidesign to deliver a completely robust and powerful MIDI solution for all Pro Tools systems. Equipped with ten MIDI input and output ports for a total of 160 channels, MIDI I/O connects to the computer and is powered via USB. MIDI I/O is housed in a rugged metal 19˝ chassis making it simple to plug and unplug cables when the unit is rack mounted. If you're looking for a way to bring unity to all your racks of synthesizers, samplers and other MIDI gear, MIDI I/O is the interface you've been waiting for.

- Ten simultaneous MIDI I/O ports supporting a total of 160 MIDI channels
- Up to four MIDI I/Os can be connected to your Pro Tools system simultaneously for a total of 40 I/O ports, providing access to a total of 640 MIDI channels.
- Programmable hardware thru mode patches any inputs to any combination of outputs
- Full OMS compatibility
- Connects to the computer via USB, providing an accurate and stable way of transferring multiple channels of MIDI data. The MIDI I/O also draws its power from the USB port – no wall wart, and no need to add another power strip just to plug in one more power cable.
- Output ports 9 and 10 mirrored on front and rear panel
- Digidesign Time-Stamping support delivers rock solid timing accuracy of less than one millisecond thus alleviating the inherent timing and jitter issues normally associated with recording and playing back MIDI information using a computer.

MASTERLIST CD 2.4

CD Mastering Software



MasterList CD 2.4 provides the final link between your digital audio workstation and professional CD masters. With MasterList CD 2.4, your Digidesign digital audio workstation is transformed into a cost-effective, professional CD mastering solution — all in one integrated system. Running stand-alone on any compatible Power Mac—no Digidesign hardware required—MasterList CD lets you easily produce Red-Book standard, CD-DA masters.

- Supports the full feature set of many SCSI-based CD recorders
- Supports 1x, 2x, 4x, 6x, 8x and 12x write speeds
- File support includes Sound Designer II and AIFF 16- and 24-bit, mono and stereo files, as well as split-stereo files from Pro Tools (24-bit files are dithered to 16-bits during disc writing).
- Make seamless transitions between tracks using editable, non-destructive, RAM-based crossfades.
- Up to 100 auto-locate points for auditioning between tracks
- Independent channel level adjustment from 96 dB to +12 dB in 0.1 dB steps.
- Easy level matching across the entire CD, provides location and level details of the audio peak within a selection
- Supports the full set of PQ subcodes, allowing the CD to be precisely defined as desired, whatever your needs.
- Individual track level cut and boost
- Definable crossfades between MasterList items
- Create up to 99 tracks per CD, with up to 100 index points per track. Index points can be manually entered or set using Sound Designer II markers or region boundaries from Sound Designer II and Pro Tools.
- Created CDs can be used to directly make the glass CD master during CD production, eliminating the additional expense and potential errors associated with conventional tape transfer and manual PQ subcode entry.

B&H PHOTO - VIDEO - PRO AUDIO

ONE HOUR FREE PARKING
AT 349 W. 34th STREET (with purchase of $100 or more)

MACHINE CONTROL

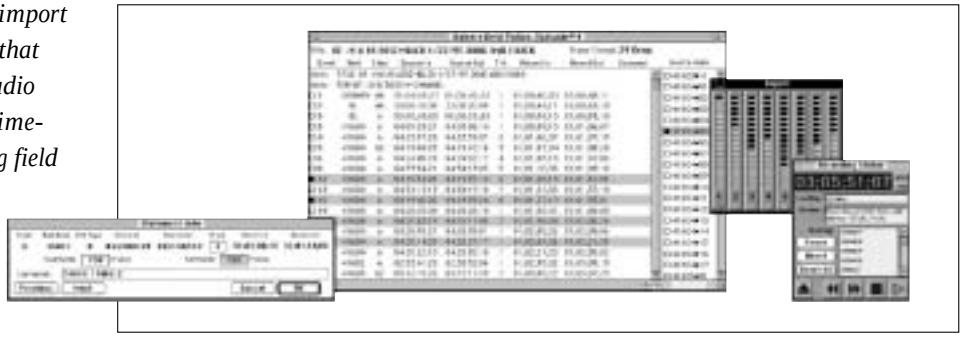
Integrated Machine Control Option for Pro Tools

Available for Mac and Windows-based systems, Machine Control enables easy integration and control of external audio/video equipment within the Pro Tools workstation environment. Machine Control lets you be more productive by allowing you to control external devices using quick, intuitive key commands directly from your computer keyboard. Features that you would expect to find in hardware-based systems, such as FF, REW, jog/shuttle, and so on, are easily accessible via the Pro Tools graphical interface.

- Slave TDM-equipped Pro Tools systems to external controllers via 9-pin Remote Mode for the trigger of playback, recording, and track arming from a remote machine.
- When used with Pro Tools 4.3 and higher, Machine Control also offers track arming with insert and assemble recording.
- Control your audio or video machine's transport operations within the Pro Tools environment: jog shuttle, FF, REW, etc.
- Transport counter keeps you informed of the machine's position at all times
- Status indication in the transport window provides user feedback for the various control modes. Even displays the difference between Shuttle mode and FF/REW modes.
- Scrub audio and video simultaneously from within ProTools. Check your audio and picture sync quickly and efficiently using the scrub-to-picture feature of Machine Control.
- Supports compatible non-linear video recorders/players
- Auto-detects many commonly used transports and presents the appropriate track configuration to the user for intuitive, integrated lay-backs to master tapes.
- Alerts Pro Tools if a machine's local/remote mode has been changed from its front panel, eliminating communication problems that may otherwise occur.

POSTCONFORM

EDL Import/Autoconform Software



PostConform is an EDL (Edit Decision List) import /autoconform software for Pro Tools systems that enables automatic capture and spotting of audio elements to picture. With PostConform, the time-consuming process of loading and conforming field audio and other audio elements can be accomplished automatically, greatly increasing productivity. PostConform automates the process of assembling an EDL into a Pro Tools session file, first by importing and sorting the EDL. Then, the user can select all or only certain source elements for loading and can specify additional record handles for future editing flexibility. At this point, the autoconform process begins. PostConform prompts the user to load successive source reels into a remotely controlled deck. Using Digidesign's optional Machine Control, PostConform automatically locates and records the sound elements selected by the user and assembles these elements into their proper position in the program — as specified by the EDL. If the picture editor makes changes to the program, PostConform can re-conform the loaded sound elements to a new EDL.

- Sources can be recorded and conformed to a new Pro Tools session or placed into the currently-open session via Apple Events
- Adjustable "checkerboard" feature allows you to organize assembled elements and to manage overlapping events
- Flexible sorting of EDL elements by Source In, Record In, or Reel Name
- Includes 9-pin serial machine control cable, but also requires Sync I/O or LTC-to-MTC converter (not included).
- Supports 9-pin and V-LAN compatible external video and audio transports and TimeLine Lynx I or II synchronizers for additional transports
- Supports CMX 3400-and 3600-series EDLs.

DIGITRANSLATOR 2.0

OMF Interchange Utility for Pro Tools

To facilitate the seamless exchange of sound and picture elements, Avid developed OMFI (Open Media Framework Interchange). Dozens of manufacturers support OMFI, and hundreds of Pro Tools customers rely on OMFI every day for file interchange with other workstations, such as Avid's Media Composer.

DigiTranslator 2.0 takes OMF Interchange to the next level by offering rock solid reliability and more control over the conversion process than ever before, all within the Pro Tools environment..

Packed with sophisticated conversion options, DigiTranslator 2.0 allows translation of clip-based volume data, a choice between rendered audio effects or untreated sources, and options for media copying and consolidation. The online tool tips provide constant feedback at every step of the conversion process. With DigiTranslator 2.0 fully integrated into Pro Tools, a previously unattainable degree of efficiency is now available to post professionals. Without launching a separate application, you can take advantage of DigiTranslator 2.0's many features directly, including the import and export of OMF media files and sequences. And, with DigiTranslator's thorough reporting of any translation errors, you always know what to tell the picture editor if a problem arises.

- Supports import and export of OMF media files and sequences directly into Pro Tools without launching a separate application.
- Translation of both clip-based and keyframe volume data into Pro Tools breakpoint automation playlists
- Reads and writes Broadcast WAV (.WAV) files, as well as AIFF and Sound Designer II OMF audio files
- Choice of importing either rendered audio effects or their untreated source files
- Integrated frame rate, sample rate and bit-depth conversion
- Embedded-media OMF sequences play back without time-consuming extractions
- Creates sample-accurate exports of Pro Tools audio and session files for Avid and other OMF 2.0-capable systems
- Supports import/export of 24-bit OMF files
- Engineered and tested to deliver reliable interchange with Avid picture workstations
- Simultaneously converts audio and/or video media data
- Pro Tools requires AVoption or AVoption|XL for import and playback of AVID video media.

Advanced Authoring Format (AAF) Support

Pro Tools 6.1 adds support for Advanced Authoring Format (AAF) via the DigiTranslator option. AAF is a media interchange file format that enables content creators to easily exchange sequence information, digital media, and metadata across platforms and between systems and applications. AAF simplifies project management, saves time, and preserves valuable metadata that in the past was typically lost when transferring program material between applications.

Digidesign will be the first audio company to bring this new level of interchange to the market. Additionally, Avid, a founding member of the AAF Association, will now support AAF interchange across the entire Avid and Digidesign product line.

DOLBY SURROUND TOOLS

Surround Encoding/Decoding TDM Plug-ins



Dolby Surround Tools makes surround mixing very efficient for Pro Tools users. D-A and A-D conversions are no longer needed due to the Dolby Surround Tools' encoding and decoding processes matching those of the industry standard Dolby Model SEU4 and SDU4 units. In addition, Dolby Surround Tools also supports the Pro Tools multi-channel mixer and sample rates up to 96 kHz.

While Dolby Surround Tools is not designed for the final mixing of matrix-encoded theatrical film soundtracks, it can be used to preview the results of the process (4-2-4 monitoring) on the discrete four-channel audio tracks (L, C, R, S) that will subsequently be combined with other elements during the final mixing of film soundtracks. This is a vital asset for any facility that regularly supplies four-channel sound effects and music recordings for theatrical film productions. (Dolby Laboratories continues to supply producers with dedicated film sound encoder systems for final film mastering.

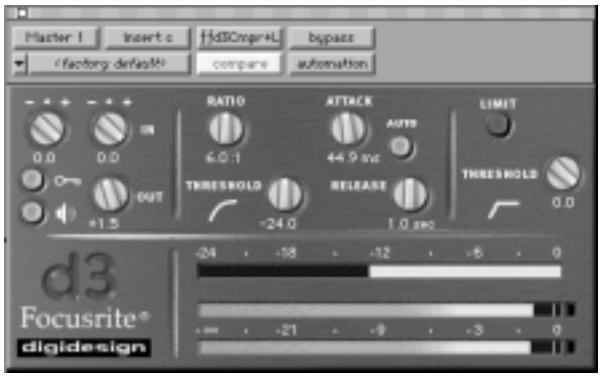
FOCUSRITE D2/D3 BUNDLE

Multiband EQ and Compressor (TDM, RTAS & AudioSuite)



Modeled after their respective Red Range hardware counterparts, the Focusrite D2/D3 bundle brings high quality equalization and professional dynamic control to Pro Tools.

Designed by Rupert Neve, the 24-bit D2 is for the professional who requires the highest levels of performance and maximum versatility in a software-based EQ providing multiple modular configurations. With superior resolution, intuitive graphics, dynamic automation and save/copy/paste options, the D2 takes full advantage of the power and capabilities of the Pro Tools TDM environment.



The D3 provides flexibility and versatility through separate insert configurations in addition to its highly impressive sonic characteristics. The D3 also functions in AudioSuite, providing file-based processing and conservation of DSP resources. Like the D2, the D3 takes takes full advantage of the power and capabilities of Pro Tools to offer dynamic automation, multiple inserts and save/copy/paste options.

Both the D2 and D3 interfaces closely emulate their respective sibling Red Range hardware units, with their red brushed-aluminum panel graphics, illuminated push buttons and rotary control knobs. A floating editor window supplies instant access to the various modules of the D2 - six band, four band, or dual/single band - or, in the case of the D3, one-button access to both compressor and limiter modules. A numeric display area is located below each rotary knob for continuous feedback and also allows numeric entry from the key-board, if desired. For precise processing, independent input and output level controls and high-resolution plasma-type meters are also provided. With the D2, offset level adjustment is accommodated by use of the left, right and link controls.

Red Range 2 Dual EQ

- Provides three different mono or stereo modules: six band, four band, and dual/single band. These versatile configurations allow DSP power to be used where it is needed most. Any band type (or types) can be accessed up to the available DSP power in the system.
- Includes high- and low-pass filters, high and low shelving filters and high-mid and low-mid peaking filters.
- Analog-style overload indicators track clipping and a highly accurate Cartesian graph displays EQ settings as they are being adjusted. When an offset exists for the left and right channels, the graph will display both channels simultaneously.

Red Range 3 Compressor/Limiter

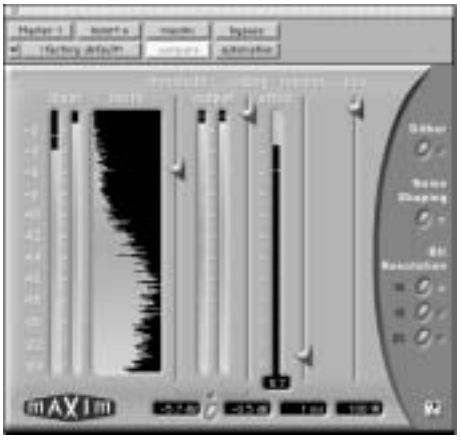
- Provides two mono or stereo configurations.
- The Compressor+Limiter supplies both compression and limiting at all times. Also allows users to select compression or limiting, conserving DSP power for other operations as necessary.
- One D3 plug-in can be used multiple times simultaneously, limited only by DSP capacity.
- Pro Tools supports side-chain processing with D3, accepting post-fader input from any track or bus to control dynamics parameters.
- All settings are automatically restored and recalled with the Pro Tools session.

Don't forget that, like other TDM Plug-Ins, you can run multiple D2 and D3 plug-ins up to the limits of your available DSP power. This means that, for the price of one plug-in bundle, you'll have access to multiple Focusrite processors, making this extremely cost effective.

MAXIM

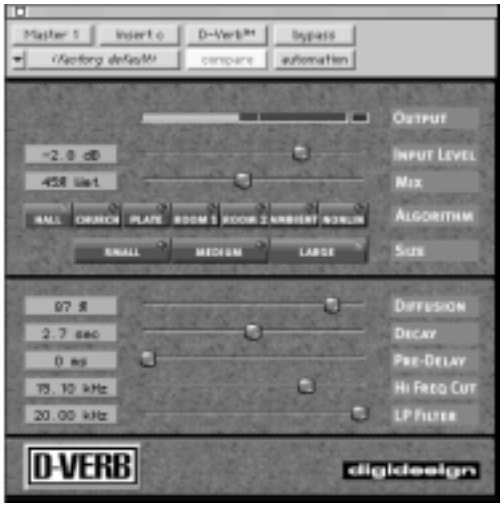
Peak Limiting and Sound Level Maximizer

Create professional-sounding master tracks for your Pro Tools sessions with Maxim. More than just a world-class peak limiter, Maxim optimizes the overall level of the audio input while preserving the integrity of the original sound. Its proprietary sound level maximizing technique distinguishes it from analog limiters and digital versions of them, which often add unwanted distortion or coloration to the audio. The result is transparent, or perfect, peak limiting. Maxim also offers built-in dithering, on-line help, and a full-color Histogram, making it an indispensable plug-in for mixing and mastering in the Pro Tools TDM-based, RTAS or AudioSuite environments.



- Rather than clipping peaks, Maxim uses an advanced proprietary technique that takes advantage of the Pro Tools non-linear environment to intelligently process the signal
- It looks ahead to anticipate peaks in the audio file and then reduces them —relative to the track's lower level audio — without changing the overall sound quality. The result is perfect peak limiting that preserves the sonic characteristics of the audio in a way not possible with any other limiters.
- In addition, Maxim automatically adjusts the over-all gain of the audio to meet the ceiling as limiting is applied. In other words, it combines advanced peak limiting (without clipping) and normalization
- Flexible enough to be used as a dynamics processor on any a channel or across an entire mix
- 16-, 18- or 20-bit Dithering improves audio performance and reduces quantization noise when mixing or fading low-level audio signals.
- Color Histogram helps you determine where the energy resides in a song, and provides valuable visual feedback for precise threshold adjustment. Maxim always provides a full graph reading,whether the playback period is 10 seconds or 10 minutes long.
- Display also includes plasma-type input and output meters, a threshold slider, output ceiling, and attenuation meter.
- Compatible with Mac OS, Windows 98, and Windows NT-based Pro Tools systems.

In addition to TDM, RTAS, and AudioSuite compatibility, Maxim, D-Verb and D-Fi all offer realtime, dynamic automation of virtually every parameter.



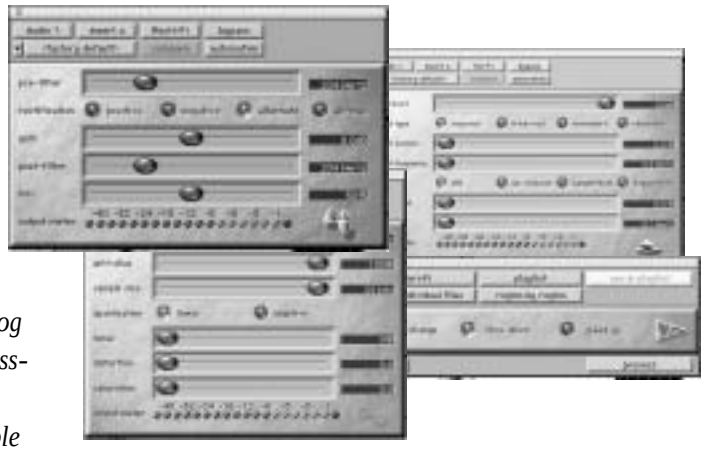
D-VERB *High-Quality Reverb*

D-Verb brings professional-quality reverb or ambience processing to your Windows or Mac-based Pro Tools session. From spacious halls to intimate rooms, D-Verb provides you with several different high-quality reverb algorithms to apply to single or multiple tracks of your session. Delivering real-time processing with automation of every parameter, D-Verb can be run on single or multiple tracks, auxiliary inputs and group masters, and in mono, stereo, or mono-to-stereo to create a stereo image from a monotrack. With an attractive interface, easy on-screen editing, superb audio quality, the ability to do the work of multiple stand-alone reverb processors, and your choice of TDM, RTAS or AudioSuite processing, D-Verb is the perfect addition to any Pro Tools system.

- D-Verb has seven reverb algorithm (Hall, Church, Plate, Room 1, Room 2, Ambience and Non-Linear) and has control over room size, diffusion, decay and pre-delay.
- Separate low-pass and high-cut filters make it easy to tailor the reverb sound to your needs. Reverb presets may be saved or recalled from disk for added flexibility.
- Multiple reverbs may be run simultaneously
- All-digital signal and mixing path
- Save, Copy, and Paste settings

D-Fi

Analog and Retro Processing in the Digital Domain



Retro instruments and processors are often used to create certain sonic signatures in contemporary music. D-Fi is a unique family of four (Lo-Fi, Sci-Fi, Recti-Fi and Vari-Fi) plug-ins that can provide similar sound design tools to Pro Tools users, without having to re-sample the audio files through an 8-bit sampler or run a file through a modular analog synth. While Lo-Fi provides bit-reduction for retro sound processing without the expense of retro equipment, Sci-Fi adds analog synth-type ring modulation, frequency modulation, and variable frequency resonators. Recti-Fi provides super- and sub-harmonic synthesis, and Vari-Fi allows processing of disk files to create the effect of audio changing speed to or from a complete stop.

While Pro Tools offers extremely high quality signal paths with extremely low distortion and noise artifacts, the D-Fi plug-ins alter this in the name of creativity. Lo-Fi diminishes the audio quality through bit-rate reduction, and pure unadulterated noise and distortion. Sci-Fi contorts the audio signals through ring modulators and resonators. Recti-Fi warps the signal through waveform rectification. Vari-Fi simulates turntables starting-up or slowing-down while playing back audio. All support MultiShell II for DSP sharing on TDM systems.

Lo-Fi

- Variable amplitude noise generator
- Sample rate reduction
- Sample size reduction
- Anti-aliasing control
- Soft clipping distortion and saturation

Recti-Fi

- Full wave rectifier
- Subharmonic synthesizer
- Pre-filter for adjusting effect frequency
- Post-filter for smoothing generated waveform

Sci-Fi

- Ring modulator
- Frequency modulation
- Positive and negative resonator
- Modulation control by LFO, envelope follower, sample-and-hold, and trigger-and-hold

Vari-Fi (AudioSuite Only)

- Speed up from a complete stop to normal speed, or slow down to a complete stop from normal speed

Digidesign and Digidesign Development Partners make a wide range of plug-ins for four different Digidesign Plug-In formats — Pro Tools|HD professional systems supports all four.

TDM

Digidesign's Time Division Multiplexing (TDM) technology is the foundation of the Pro Tools professional environment, providing 24-bit mixing and real-time digital signal processing (DSP) capabilities. TDM Plug-Ins use the dedicated hardware of TDM systems, which offer easy expandability — add as many TDM plug-ins as you want —up to the DSP capacity of your system, no matter how many edits or how much automation you have in a session. If you need more DSP, simply add another HD Process card.

Real-Time AudioSuite (RTAS)

Real-Time AudioSuite Plug-Ins are host-based processors, using the computer's processing power to do their job. Functionally, RTAS Plug-Ins offer many of the real-time benefits of TDM Plug-Ins. They are fully automatable, their parameters can change in real time, and their effects are not permanently written to the audio file. Since they are host-based, RTAS Plug-Ins require trade-offs between track and plug-in count, edit density and amount of mix automation in a session.

HTDM

Host Time Division Multiplexing (HTDM)represent a hybrid of TDM and RTAS technologies. HTDM Plug-Ins provide all the logistical functionality of standard TDM Plug-Ins, but, like RTAS Plug-Ins, they allow for all the processing to be done on the host. In so doing, HTDM Plug-Ins offer two primary benefits: The resources of the Pro Tools|HD Core and/or Process card remain available for other tasks; and, despite running on the host, HTDM Plug-Ins can be instantiated on any track, a characteristic previously confined to TDM Plug-Ins only.

AudioSuite

AudioSuite Plug-Ins provide file-based processing, meaning they process or alter the sound file and create a new file with the processed sound. The resulting effect is applied to the entire file. AudioSuite Plug-Ins are great for conserving DSP power and certain types of processing where there is no real-time benefit or application, such as normalization and noise reduction.

DIGIDESIGN

BRUNO & RESO

Cross-Synthesis Plug-In Duo for TDM Systems





Take your audio into creative new realms of sound with Bruno and Reso, two TDM plug-ins for Pro Tools. Both use a cross-synthesis technique to synthesize any existing audio in real-time to create some very unique and interesting results.
Both plug-ins can be played interactively via MIDI using an external controller. Play a chord progression or a scale on your keyboard, and Bruno or Reso will apply the performance to your audio. In addition, an on-screen keyboard allows you to latch the keys to create chords or play a scale right on the screen. Bruno & Reso is a seriously advanced piece of software, but it isn't rocket science to use. Just grab the knobs, start dialing and have fun. In the process, you'll take your audio to amazing new place.

- With up to 24 voices, Bruno & Reso allow you to create a deep layer of adjustable resonant tones. Shaping and modulation parameters can be adjusted with a wide selection of on-screen control knobs.
- Timbre, amplitude, pitch, stereo spread, low-pass filter, Q and follower characteristics can be fully automated. In addition, velocity and pitch bend parameters can be controlled by your keyboard. MIDI clock capabilities and an envelope follower let Bruno/Reso match the groove of the source material.

Bruno

Bruno uses time-slicing, a technique whereby timbres are extracted from the source audio during playback and crossfaded together. Crossfade lengths between these sections can be adjusted and a switch function produces additional textural changes. Switch can be triggered by threshold, MIDI beat clock or keyed to an external track. Further shaping and modulation is achieved through a wide variety of timbre, amplitude, pitch, voice and stereo spread parameters.

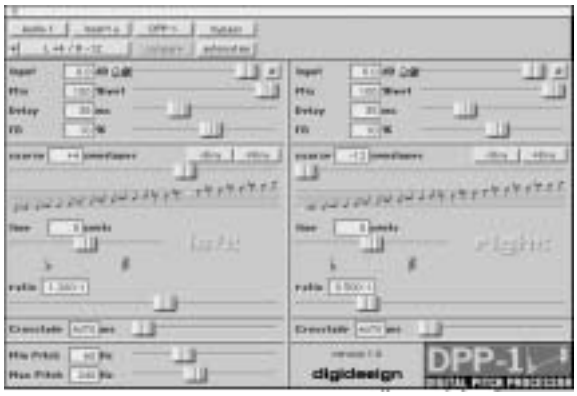
Ruso

Reso has similar features as Bruno, but uses a resonance generator to add harmonic overtones. Damping and other processing knobs help shape the sound. An all/odd harmonics toggle can be triggered from an external keyboard in the same manner as Bruno's switch control. At the end of the chain, a resonant filter modulates low-pass frequency sweeps with adjustable Q and follower knobs.

DPP-1

Pitch Processor for TDM Systems

The DPP-1 brings high-quality 24-bit pitch change and delay processing to the Pro Tools TDM environment. The DPP-1 can operate in mono or stereo modes, and supplies up to 4 octaves of high-quality stereo pitch transposition on single or multiple mixer channels. With easy on-screen editing and program storage, and the ability to run multiple pitch processors for the price of one plug-in, the DPP-1 can do the work of multiple standalone pitch processors — at a far lower cost.



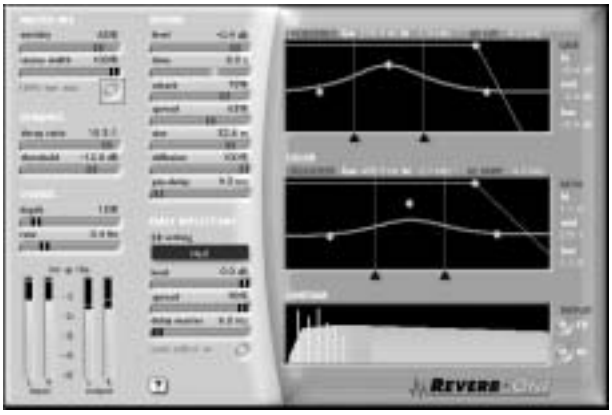
DPP-1

The DPP-1 supplies an intuitive user interface that utilizes musical staff notes and octave switches for up to 4 octaves of stereo pitch change, adjustable in semi-tones or cents. Clicking on a musical note value on the on-screen staff allows quick navigation to new pitches. In addition to pitch change, the DPP-1 supplies an additional delay line with up to 125 ms. of delay per channel, with positive or negative feedback. Signal presence and internal clipping metering is provided.

- Internal 24-bit processing and high-quality deglitching pitch change algorithm provides excellent sound quality
- Easy-to-use software interface makes operation straightforward
- Pro Tools TDM systems supply a versatile mixing and processing environment, letting you use the DPP-1 Pitch Processor on single or multiple channels, which can come from any input source: disk playback, live input, TDM-equipped SampleCell II Plus, etc.
- Libraries of parameter settings can be saved and loaded by using the Pro Tools Plug-In Librarian. With the additional DSP power available, you can run multiple DPP-1 Pitch Processors simultaneously.
- Independent processing control over left and right channels
- 4-octave pitch range
- Delay processing with feedback
- Automation features, including dynamic TDM Plug-In automation

REVERB ONE

World-Class Reverb Processing for TDM Systems



If you're looking for a reverberation processor of uncompromising sonic quality that's also easy to use, look no further than the Reverb One. A TDM plug-in, Reverb One is a reverb processor built from the ground up by Digidesign to satisfy demanding audio professionals. At every step in its development, Reverb One was subjected to critical listening tests, resulting in a TDM plug-in that can compete with any software or hardware reverb processor on the market today. Reverb One gives you complete control of the reverberant characteristics of your mix, and is equally at home in music, post production and sound design applications that require the utmost in clarity and precision.

➢ In addition to the exceptional quality of its reverberation, Reverb One provides a unique set of shaping tools that let you modify the details of a sonic space. You'll find standard controls for room reflections settings and decay time, as well as for multi-band equalization, reverb dynamics (for long tails or gated characters) and chorusing. With Reverb One, you get everything you need to control all aspects of reverb in a single plug-in window — and editing reverb characteristics is straightforward with its intuitive, editable graphic displays.

➢ For all the versatility of Reverb One, it is remarkably easy to use. Its interface conveys reverberation character at a glance, and a wealth of carefully designed presets gives you everything you need to get started. From natural-sounding halls, to vintage plates or that classic digital reverb setting, Reverb One allows you to customize your creative soundscape to anything you can imagine.

➢ As with any TDM plug-in, you can take advantage of the ability to automate any Reverb One parameter in real-time, and to recall all plug-in settings in Pro Tools. With its extraordinary sound quality and versatility, it's hard to believe that Reverb One only takes up a single DSP chip.

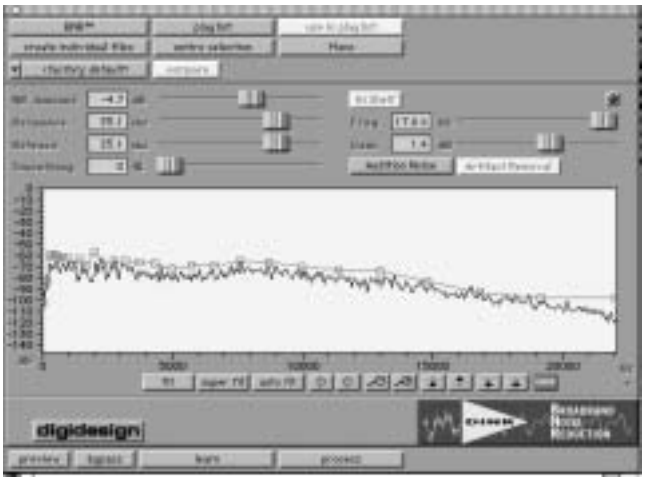
- Independently controllable reverb settings, including Level, Decay Time, Attack, Spread, Room Size, Diffusion, Pre-Delay
- Extensive library of reverb presets
- Early reflection controls, including Level, Delay, Spread, plus early reflection room presets
- Dynamics and Chorus sections for shaping reverb decay and optimizing reverb effects
- Multiband EQ with adjustable crossover points for shaping tone spectrum of both early reflections and reverb tail
- Reverb color controls allow independent adjustment of harmonic content and decay times over different frequency ranges
- Interactive early reflection and reverberation contour display

DIGIDESIGN

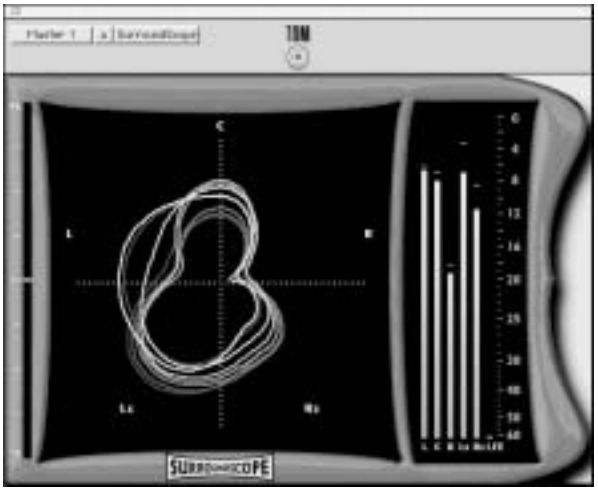
DINR

Intelligent Noise Reduction (TDM & AudioSuite)

DINR (Digidesign Intelligent Noise Reduction) is an award-winning plug-in for real-time, high-quality broadband noise reduction and hum removal. A software-based, digital signal processing module, DINR is designed to reduce the full spectrum of unwanted noise, from air-conditioner rumble to tape hiss to guitar-amp buzz, for more professional results.



- DINR reduces noise by intelligently subtracting the noise from the digital audio file. First, the noise signature is created by selecting and analyzing an example of the noise within the source material. Once the noise is analyzed, an on-screen slider can be used to apply from 0.1 dB to an infinite amount of noise reduction.
- Other parameter settings can also be adjusted on-the-fly to achieve maximum noise reduction while minimizing signal loss and artifact generation.
- And since DINR analyzes and subtracts noise entirely within the digital realm, the results can be virtually free of side-effects such as distortion, dynamic modulation and the undesired fluctuations in frequency response associated with conventional noise reduction systems.
- Parameter settings can be saved and recalled for use on other source audio files affected with similar noise. Plus, the noise signature of any device can be kept and used with other source material the device may affect.
- All noise signatures and parameter settings are automatically stored with your Pro Tools session for instant recall.
- High fidelity, real-time processing with TDM and file-based processing with AudioSuite
- Automation features including dynamic TDM plug-in automation
- Compatible with Pro Tools systems (4.1.1 or higher) running on Mac or Windows NT



SURROUNDSCOPE

Surround Metering (TDM & RTAS)

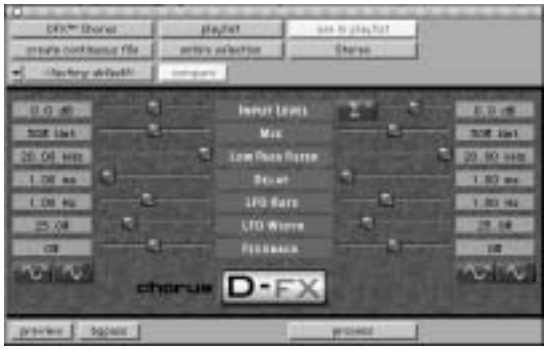
Now you can visualize the surround capabilities of Pro Tools with DigiDesign's SurroundScope plug-in. With its intuitive interface, you can accurately see what your listeners will hear — right from within Pro Tools. SurroundScope provides a graphical display of the signal level for each audio channel within the multi-channel sound field in a Pro Tools mix. The phase meter shows coherency of your signal across the full spectrum, from perfect mono to completely out of phase. In addition to its surround and phase displays, SurroundScope features a multi-channel level meter that supports up to eight channel tracks.

- Real-time, accurate display of stereo or multi-channel phase information and levels
- Support for all standard multi-channel formats: stereo, LCR, LCRS, Quad, 5.0, 5.1, 6.0, 6.1, 7.0 and 7.1
- Auto detection of a track's channel format
- Multi-dimensional display shows position of the audio signal within the multichannel track or surround mix
- Full-featured center-zero and Lissajous phase meter display
- Highly accurate level meters for each channel of a multi-channel

B&H PHOTO - VIDEO - PRO AUDIO

D-FX

AudioSuite Plug-In Package



The D-fx family of AudioSuite plug-ins offers a great selection of time-domain effects all in one affordable package and is a perfect addition to any system running Pro Tools software. D-fx includes several effects including Chorus, Flanger, Multi-Tap Delay and Ping-Pong Delay. D-fx also includes D-Verb for use in both RTAS and AudioSuite environments.

- The perfect addition to any system running Pro Tools, D-fx offers a great selection of time-domain effects in one affordable package
- Each D-fx plug-in is designed for intuitive use, with a consistent graphic interface for every effect. All parameters are displayed on one screen allowing instant adjustments.
- Since D-fx is completely integrated into the Pro Tools software, there's no transition time lost switching between effects processing and editing.
- In "stereo" mode, all D-fx plug-ins allow processing of two channels as left and right; or, combine the two input signals for a mono in/stereo out effect
- Use with Pro Tools systems for file-based effects processing
- Save and recall settings, using Pro Tools Plug-in Librarian and Settings pop-up menus

SOUND REPLACER



Drum Hit & Sound Replacement (AudioSuite)

SoundReplacer is definitely one of the most unique and innovative plug-ins on the market today. As a Drum Hit and Sound Replacement tool for Pro Tools, SoundReplacer allows you to replace or blend up to three different samples to three independent adjustable amplitude threshold zones with your existing audio file. For more than just repairing weak mixes, this AudioSuite plug-in has enormous creative potential for sound design and post production applications.

➢ SoundReplacer is a unique AudioSuite Plug-In that lets you replace or mix an audio event with new samples from your sound library or hard drive. It is the perfect solution for that all-too-familiar situation of the imperfect drum track. SoundReplacer allows you to retain the feel of the original performance by automatically adjusting the dynamics of the replacement sound to match the changing levels within the performance.

➢ What's more, SoundReplacer lets you replace or blend up to three different samples to three different adjustable amplitude threshold zones in the audio file. Variations in amplitude within the performance determine which sample is triggered at a given time. Performances can be further adjusted with a mix slider, dynamics slider and a peak align feature.

➢ Although perfect for repairing weak mixes, SoundReplacer also has enormous creative potential for sound design and post production applications. Sound effects designers can morph three different characteristics to a single source file, thereby adding a sense of realism to an audio track.

- Load your favorite samples from your sound library, hard drive, etc.
- Trigger up to three samples, each set to a customizable threshold zone
- Crossfade or hard shift between samples
- Set the amount of sample replacement using the adjustable mix slider
- Expand or contract a performance's dynamic range
- Ensure phase-accurate alignment using the Peak Align option

DIGIDESIGN

ACCESS VIRUS

World-Class DSP-based Synthesizer for Pro Tools TDM

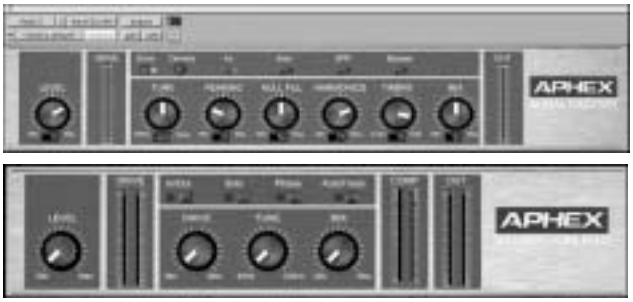


Based on the highly acclaimed and successful Access Virus hardware synthesizer, Access Virus is a TDM-based synthesizer plug-in that transforms your Pro Tools system into a luscious 20-voice synth at 48k and 10-voice synth at 96k on Pro Tools|HD – with up to eight different sounds per DSP chip. In addition to offering incredible warmth and quality, this "virtual analog" synthesizer also features perfect Pro Tools integration, extensive automation and total recall capabilities. MIDI and audio connections are a breeze to set up, allowing you to start playing Virus in seconds. It is also offers several user interface pages for different sets of parameters, so every single control can receive a dedicated knob or switch. Gone are the headaches associated with glaring at tiny LCD displays.

- Access Virus features three oscillators, two filters, three LFOs and two envelopes for every voice. It also provides built-in effects like chorus and delay for creating stunning pads, screaming leads, punchy bass sounds or whatever you might have in your mind.
- Includes more than 500 extremely variable and useful sounds -everything from Moog basses to analog drumkits
- Fully automated with frame accuracy for over 150 parameters
- Direct Pro Tools integration allows easy MIDI/audio routing and takes advantage of the powerful mixing features in Pro Tools
- Use input mode to modify single tracks or entire mixes with powerful resonant filters. Enable the vocoder and add the robotic sound to your recording using other tracks.
- Built-in arpeggiator syncs to MIDI beat clock and follows your song tempo at any time. (Ultra low latency for extremely fast MIDI-to-audio timing).
- Loads patches directly from the original Access Virus hardware—no need to reprogram all your existing sounds
- Runs on a dedicated DSP for true power-on-demand, up to eight multi-timbral synthesizers per DSP
- Up to 80 voices on a Pro Tools|24 MIXplus system
- Pro Tools plug-in library function allows easy patch management and exchange

APHEX BIG BOTTOM PRO / AURAL EXCITER

Bass and Harmonic-Generating, Detail Enhancing Bundle



Want to enhance the bass frequencies in your Pro Tools sessions? Aphex Systems' Big Bottom Pro is a powerful TDM plug-in for Pro Tools that will add all the excitement and punch you need to the lower frequencies in your mix. And Big Bottom Pro comes bundled with Aural Exciter giving you the same unique ability to increase presence and restore natural brightness and detail, without significant EQ. The Aural Exciter/Big Bottom Pro plug-in bundle supports Mac and Windows-based Pro Tools|HD systems at sample rates up to 192 kHz.

The Big Bottom circuit works on the bass end frequencies and adds low-end presence and punch without adding peak level. This lets you pack more bass into your sound without overloading recorders or blowing up speakers. The Big Bottom Pro plug-in is an enhanced, pro version of this powerful circuit. Used together with Aural Exciter, Big Bottom Pro completes the enhancement of the entire audio spectrum.

The hardware version of Aphex Aural Exciter, its patented circuitry helps recorded or amplified audio get closer to the natural acoustic sound — clean, open, present and detailed. The effect is achieved by adding musically and dynamically related harmonics to the input signal. This effect has been used on thousands of recordings, movies, commercials, broadcasts and live concerts.

- Brings instruments and vocals up "out of the mix"
- Patented Harmonic Generation Process can actually increase bandwidth without increasing level
- Increases presence, clarity, speech intelligibility, and detail without significantly increasing peak output
- Total flexibility in "tuning" the Aural Exciter to your needs

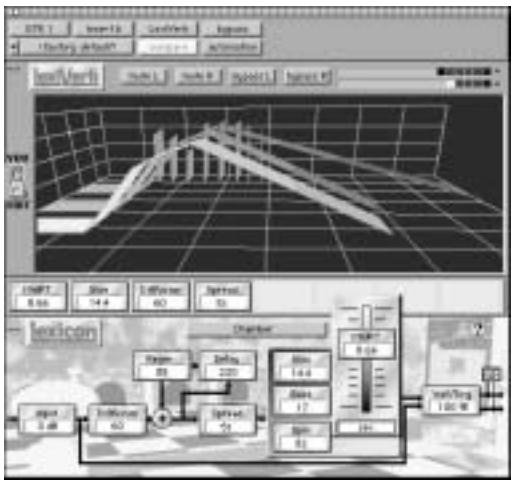
DRAWMER DYNAMICS

High Quality, Multi-Band Dynamics for Pro Tools TDM

The Drawmer Dynamics plug-in provides premium quality gating, compression and limiting to Pro Tools TDM users. Drawing on 20 years of experience in innovative signal processing, the Drawmer Dynamics plug-in offers the performance attributes as well as the look and feel of world-renowned Drawmer analog units.



- Based on Drawmer's legendary DS201 Noise Gate, the Gate/Compressor/Limiter plug-in is the ultimate processing tool for percussive and highly transient material.
- The Side-Chain Trigger can be set to key the gate from any audio track you choose, providing limitless processing possibilities. Key filters "tune" the gate to the required frequency to assist triggering, while the ultra-fast attack time adds punch. Audio is further enhanced by the compressor and is kept under precise control by the limiter.
- For less percussive signals like vocals, guitar, and piano, Drawmer's Program Adaptive Circuits can intelligently shape and enhance the signal, breathing life into dull tracks and gently controlling excessively dynamic passages.
- The Expander/Compressor/Limiter plug-in is based on the Drawmer DL241 Auto Compressor and DL251 Limiter. Auto gain adjusts the gain of the compressor when the threshold or ratio controls are adjusted.
- Drawmer Dynamics takes full advantage of the latest Pro Tools functionality including dynamic automation, total save and recall of all parameters and mapping to external control surfaces.



LEXICON LEXIVERB

Reverb Plug-in for Pro Tools TDM

LexiVerb combines world-class Lexicon reverb with a simple, yet powerful interface to create an indispensable tool for your Pro Tools system. Pro Tools automation is supported and LexiVerb parameter groupings, giving you total control over multiple parameters with a single fader. LexiVerb provides over 100 presets built from four powerful algorithms.
With the award-winning LexiVerb TDM Plug-In, no additional hardware is necessary to obtain the legendary Lexicon Sound — the sound that's heard on more than 80% of the world's most successful albums and soundtrack

- With LexiVerb, Pro Tools users can achieve that legendary Lexicon Sound with all the benefits of the TDM plug-in architecture. LexiVerb features four powerful, classic Lexicon reverb algorithms: Chamber, Plate, Inverse and Gate with 100 factory presets designed for any recording task.
- Each algorithm is displayed graphically in a three-dimensional space, allowing easy control over reverb characteristics.
- Low, mid and high frequency envelopes can be altered in real time by tweaking up to eight assignable faders.
- There are also customizable macro controls for assigning as many as four parameters to a single fader. The Macro editor allows you to group parameters together as well as adjust how they scale with one another.
- Several adjustable parameters reside within each LexiVerb algorithm. These include Bass, Diffusion (echo density), Delay, HF Cutoff, Input, MidRT (time), PreDelay, Regeneration (feedback), Size, Slope, Spin, Spread and Wet/Dry mix settings.
- The Spin setting is unique to LexiVerb. It sets the randomization rate of the reverb tail for a more natural sound without destabilizing the position of the instruments.

Line 6 — Amp Farm

Guitar Recording TDM Plug-In



The ultimate solution for guitarists and Pro Tools TDM users, Amp features Line 6's TubeTone technology, to bring you physical models of specific, sought-after classic amplifiers. Process live guitar inputs or tweak the amp setup for pre-recorded tracks. Amp Farm lets you adjust your amp tone right up to the final mix, with full automation of all controls, including Amp Model selection. No special pickups are required —just record your guitar output directly into Pro Tools and say goodbye to the hassles of working with traditional amplifiers (the hauling, the buzzing, the special miking techniques, etc.)

- Controls are simple and intuitive, making Amp Farm extremely easy to use.
- Your Amp Farm amp set-up can be changed at any time, even after tracks are recorded, with full automation. You can even switch amps on the fly, and everything is stored right in your Pro Tools session. A library of presets is also included.
- Line 6's TubeTone uses a 100% digital software technology to physically model the sonic properties of tubes and emulate the tone generating circuitry of classic amplifiers—add a whole collection of classic guitar amps to your studio — all programmable and automatable.
  - 1959 Fender Bassman, 1965 Fender Twin, 1964 Fender Deluxe, 1964 Marshall JTM 45
  - 1968 Marshall Plexi, 1990 Marshall JCM 800, 1960 Vox AC 30, 1963 Vox AC 30 with top-boost
  - 1994 Mesa Boogie Dual Rectifier Trem-overb combo, 1995 Mesa Boogie Dual Rectifier head
  - 1989 Soldano Super Lead Overdrive head, 1987 Soldano X88R Preamp, 1996 Matchless Chieftain



Line 6 — Echo Farm

Vintage Echo TDM Plug-In

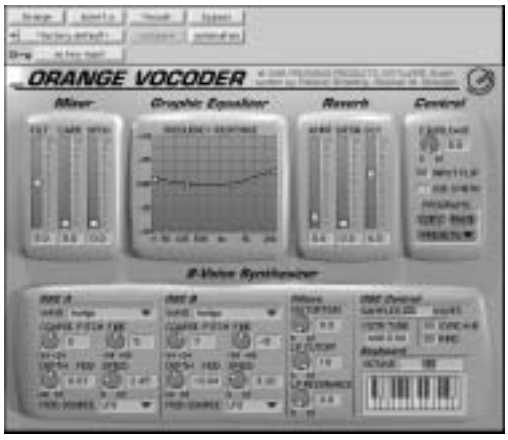
Line 6's Amp Farm revolutionized the way guitar is recorded. Now, the same patented modeling technology is available in the Echo Farm TDM plug-in. Echo Farm brings the sound of vintage echo effects units to your Pro Tools system—no need to wrestle with cantankerous vintage hardware to get classic tape echo, old school analog delay or filter-swept echo-phonic modulations, Echo Farm perfectly emulates the sonic properties of tubes, tape, and vintage echo unit electronics.

- Easy to use as a classic stomp box, with control of tape wow and flutter, filter-swept modulation, bit resolution, and more
- Just pull up Echo Farm to get these classic vintage tones with all the modern advantages, including extended delay time, bpm/note value delay time setting, tap tempo and complete automation.
- Echo Farm isn't limited to just guitar processing. Put any audio signal through it — vocals, synths, drum tracks— and add the classic sound of vintage echo to your mix.
- Add a whole collection of classic echo units to your studio — all programmable and automatable
- Set delay times by knob, milliseconds, or bpm/note value — or just click your mouse a few times on the tap tempo button.
- Defeatable Time Ramp gives smooth tape-style delay time changes complete with pitch smear
- Use Echo Farm as a guitar stomp box in front of Amp Farm, or as a processor for all your other audio track
- Models based on the following effects units:
  - Maestro EP-1 Tube and EP-3 Echoplex
  - Roland RE-101 Space Echo
  - Boss DM-2 Analog Delay
  - Electro-Harmonix Deluxe Memoryman
  - Dynamic Delay (ala TC Electronics 2290)
  - Lo Res Delay (variable from 24 to 6 bits)
  - Sweep Echo (filter-sweep delay)
  - Digital Delay with Modulation, Ping Pong
  - Reverse (xirdneH imiJ ekil tsuJ), and Auto-Volume Echo.

PROSONIQ — ORANGE VOCODER

Real-time Vocoder and Software Synthesizer (RTAS)

Tired of dull sounding tracks? Want to breathe some funk into your mix? Well, here's the perfect tool. Prosoniq's Orange Vocoder offers perhaps the fastest, most flexible and most transparent sounding simulation of an analog vocoder effect. Following a long tradition of German Vocoder effects, the Orange Vocoder is fully customizable and features an 8-voice virtual analog synthesizer unit, Freeform EQ and Filterbank Reverb all controllable from within one easy-to-use interface.



- Real-time Vocoder effect for the RTAS platform, uses either any input source/ audio track or the built in oscillator as carrier signal, and any audio track as modulator.
- Graphical control over every parameter, full use Pro Tools LE's, advanced automation capabilities, and an attractive user interface
- Supports both RTAS sidechain routing and Pro Tools automation
- Integrated 8-voice virtual analog synthesizer with 2 oscillators per voice featuring 10 basic waveforms and 7 sampled sounds. Voice detune, pitch LFO, 4-pole Lowpass filter with cutoff and resonance, oscillator hard sync, and ring modulator.
- Fully customizable Freeform EQ with max. 12 nodes, ranging from 20Hz to 20kHz and resolving ±30 dB FS
- Input Channel Flip: instantly exchange carrier and modulator signals without re-routing tracks.
- Post-Vocoder Filterbank Reverb Effect with Mix, Decay and Density
- Integrated preset program switch for easy parameter adjustment

SONIC SOLUTIONS — NoNoise

Powerful Noise Reduction/Audio Restoration Plug-in Suite

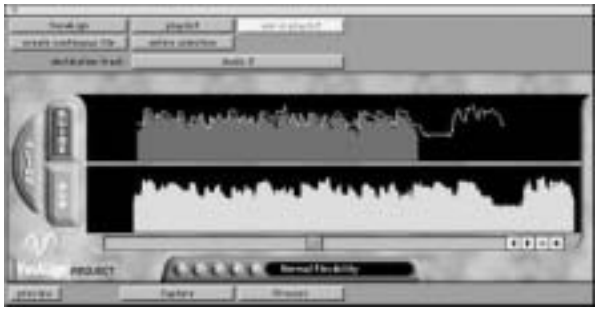


Sonic Solutions' Emmy Award-winning NoNOISE is the premier tool for restoring audio recordings. Its advanced processes isolate and eliminate audio artifacts such as hiss, scratches, hum, mechanical and impulsive noise, ambient and background noise — while preserving the integrity of the original recording. NoNOISE is not a single process or software module, but a set of powerful tools to remove bothersome noise without damage to the program material. Use it to restore old recordings, remove unwanted noises from field recordings, and repair audio materials that have suffered damage. Since it's introduction over 15 years ago, Sonic's NoNOISE technology has been used to restore hundreds of thousands of music, film, video recordings, as well as forensic recordings used by intelligence and police organizations around the world.

- NoNOISE is comprised of a set of powerful tools that address the full spectrum of noise issues, as well as offering both automatic and manual means of analysis and repair without damage to program material.
- Capabilities include frequency analysis, peak distortion removal, broadband denoising, click detection, production (automatic) and manual declicking, and decrackling.
- Ideal not only in the mastering and re-mastering stages of audio production, but also for many other audio treatments needed to restore, clean, or enhance recorded material.
- A variety of high-resolution filters (also implemented as TDM plug-ins) is included: Presence, Notch, High and Low Shelf, High and Low Pass, DC-Removal, Emphasis, De-emphasis and RIAA/No RIAA.
- While NoNOISE for Pro Tools does not include all of the application features found in the SonicStudio version, it is designed to strike an ideal balance between the workflows of both experienced Pro Tools users and SonicStudio users.

SYNCHRO ARTS — VocALign

Audio Alignment for Music and Post Production (AudioSuite)



VocALign is an amazing tool which reduces the time taken to re-record dialog or vocals —saving money by drastically reducing production and studio time. Artists can now concentrate on performance and not sync during overdubbing sessions. VocALign's unique ability to align two signals enables it to be used creatively to take guide or even "live" tracks and create performances with perfectly aligned overdubs.

This AudioSuite plug-in gives Pro Tools users perfectly aligned double-tracked vocals, tight backing vocals, easy re-grooving of recorded vocals for remixing and shorter overdub sessions. Users also have the ability to choose the best rhythm and pace for a specific vocal, or even lay down the required tempo pattern for the artist. For postproduction, VocALign will automatically edit a line of replacement dialog so that it aligns with the dialog recorded with the original film or video. The artist can concentrate on performance and the result is perfectly lip-sync'd dialog, or more convincing foreign language dubs and reduced studio time re-recording the dialog.

- Takes just seconds to automatically align two audio signals so that the timing of one matches the other
- Sophisticated pattern matching techniques reject guide track noise and work with different speakers, singers, words, instruments and sounds
- User interface allows adjustment of selected audio regions for processing, and the scrolling, scalable display allows visual confirmation of sync
- User-adjustable processing presets ensure optimal performance
- Pitch of aligned audio is unchanged

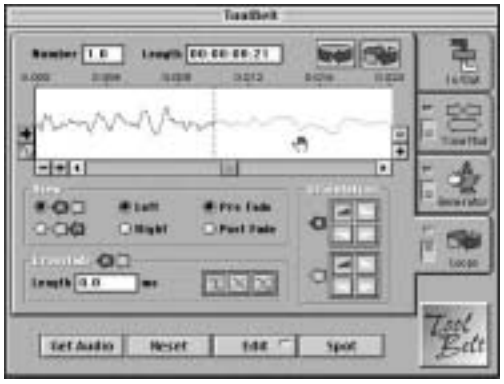
Artists can now concentrate on performance and not sync during overdubbing sessions

- Align a vocal or instrumental performance to a previous take
- Re-create a "live" vocal's timing in the overdub session
- Tighten up backing singers with one another or with the lead
- Tighten two instrumental performances
- Lock double tracked vocals together
- Remixers: completely change the timing of a vocal performance
- ADR: Lip-sync replacement dialog automatically

SYNCHRO ARTS — ToolBelt

High Speed Tools for Efficient Audio Production

A stand-alone program that both captures audio from Pro Tools and returns it with single keystrokes, ToolBelt provides five high-quality, non-real-time processes (Audio Generator, TimeMod, Power Looping, Reversing, Inverting) in a time-saving, production-efficient interface. In fact, many of these processes and features are simply not available in any other products.



ToolBelt's processes can be used separately or in combination. ToolBelt also provides keystrokes for auditioning selected processes or changes to signals before they are returned to Pro Tools.

Applications Include:

- Sound Effects: Create altered sounds and atmosphere loops quickly and easily
- Repair Work: Correct timing errors and channel inversion
- Dialog Editing: Make lengths fit; make background fills quickly; create special vocal effects
- Music Editing: Create precise musical loops and special effects; change tempos without pitch change

TOOLBELT

TOOLBELT FEATURES

- ToolBelt's unique Audio Generator can automatically extend very short pieces of atmospheric or back- ground sounds into long, natural sounding regions without sounding looped. It's great for making fills for dialog and effects editing.
- Looping function provides graphical displays and A/B pattern-making facilities
- TimeMod provides high-quality, mono and stereo time compression / expansion without pitch changes; maintains stereo phase and musical rhythm
- Stereo channels can be independently processed
- Audio playback from ToolBelt allows input and output auditioning
- Invert function (180° phase change)
- Built-in macros for seamless integration with Pro Tools
- Intuitive interface and simple operation
- Processed audio can be placed back, sample-accurate, on any session track
- Compatible with 8, 16 and 24-bit audio

SYNCHRO ARTS — TITAN

The Essential Aid for Conforming Soundtracks

TITAN is a must-have application designed for Pro Tools editors involved in audio post-production for film or video, who spend long hours manually making sync adjustments after conforming or processing audio, or have to recut and rename regions in continuous audio tracks that have been transferred into a Pro Tools session from another system.
TITAN is a multi-function program that creates new Pro Tools sessions in a fraction of the time normally required. This powerful application aids and speeds up the autoconforming process, creates sessions from EDL-based change lists, automatically performing many of the tedious and repetitive tasks normally associated with the conforming of audio.



TITAN is a stand-alone application that provides separate functions for processing a Pro Tools session file. An efficient user interface provides easy access to functions, track selection and processing options. The Flash Conform function instantly conforms soundtracks from an EDL and pre-loaded soundfiles, and can conform or reconform entire sessions in seconds— a big time saver

Fix Sync Function

The Fix Sync function automatically corrects the position of audio that is out of sync by up to ±1.3 second to sample level accuracy. Similarly, it can correct the sync of audio that has passed through signal processing devices and has been delayed by unknown amounts. TITAN virtually eliminates the need to adjust sync manually while listening for phasing or visually comparing waveforms with a worktrack.

- Can process 400 to 500 regions in a session in ten minutes. Manually, the same job could take up to a day
- Stereo regions are adjusted together to keep phase relationships intact
- Display shows size and number of sync errors
- Autonaming of processed sessions, tracks, and regions

Flash Cutter Function

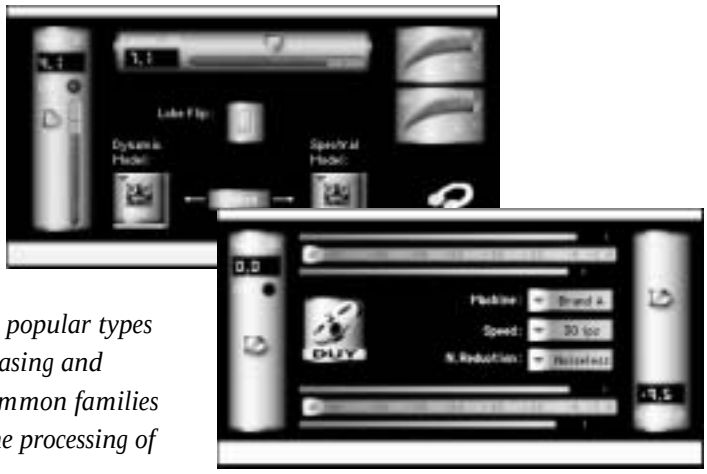
Flash Cutter eliminates the need to manually recut and rename edits in situations when tape is used to transfer audio from any editing system to a Pro Tools system and an OMF transfer is not a practical option. Once the audio on tape has been transferred into continuous tracks in a Pro Tools session, Flash Cutter uses a standard EDL from the first editing system to recreate and name the audio regions automatically.

- Automatically cuts and names hundreds of regions in seconds
- Regions, sessions, and tracks automatically named with user-selected information from EDL
- Handle adjustments allow extending edits if required

DIGIDESIGN

DUY – ANALOG BUNDLE (DAD VALVE & DAD TAPE)

Tape Saturation and Tape Emulation for TDM Systems



Now, in one bundle, you can get analog tape and real-valve sound, with their classic warmth, within the digital domain, and with characteristics and control impossible to achieve in conventional tape recorders or valve systems.

DaD Valve is a revolutionary approach to the simulation of the classical valve sound for digital audio systems. It features the most popular types of valves — triode, tetrode, and pentode — with a full range of biasing and operating levels. It features optimized responses for the 38 most common families of acoustic and electronic musical instruments. Also suitable for the processing of any kind of sound: music, effects and voices.

DaD Tape is a revolutionary approach to the simulation of the analog tape sound for digital audio systems. It features physical modeling of the four most representative tape recorders on the market: an old vintage machine with valve circuitry, a transistor based machine of the late 60's, an operational-amplifier based machine of the 70's and a machine of the latest generation. It also offers simulation of the three most common noise reduction systems, plus a proprietary noiseless-tape mode. Ideal for music, film soundtracks, audio post-production, mastering, broadcast and multimedia.

DaD Tape

- Especially suitable for percussive sounds, mastering of complete mixes.
- Round off peak transients and add warmth to electronic and acoustic instruments, either individually or by sections or sub-mixes.
- Independent input/output sliders and meter
- Switchable tape speed (7-1/2 /15/30 ips) for maximum control—even permits unusual combinations of tape recorders and speeds.
- User adjustable operating level
- Easy and intuitive user interface

DaD Valve

- Proprietary system based on physical modeling providing absolute and independent control over both spectral and dynamic responses, with a total of 1600 different configurations—Ideal for voices and sound effects, mastering of complete mixes
- Easy and intuitive user interface, easily updateable and expandable
- High signal to noise ratio due to high-quality internal resolution
- Processing of all kind of electronic and acoustic instruments, either individually or by sections or sub-mixes
- Suitable for all styles. It adds even harmonics to cold electronic sounds and restores the lost warmth to acoustic instruments



DUY EVERPACK

The Final Touch for RTAS-based Studios

The perfect solution for any M-Box or Digi001 user, the EverPack includes five excellent plug-ins for RTAS and Audiosuite: DUY Z-Room reverb, DaD Valve, DUY Shape, DUY Wide and Max DUY.

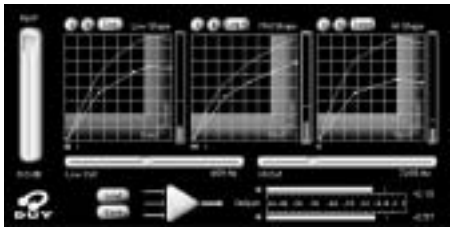
(The same as above except that it is compatible with RTAS, DUY DaD Valve is the valve emulation that gives you total control over the dynamic and spectral response of 1600 modeled combinations of sounds, introducing the typical non-linearities of tubes, and enabling you to warm up any of your sounds.)

B&H PHOTO - VIDEO - PRO AUDIO

ONE HOUR FREE PARKING
AT 349 W. 34th STREET (with purchase of $100 or more)

DUY — EverPack

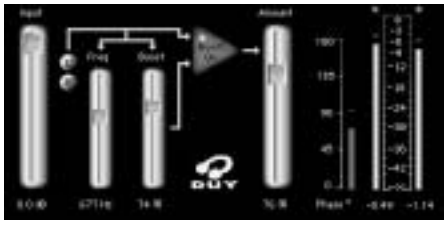
DUY Shape



Perfect for defining individual sounds of mastering entire mixes.Suitable for processing of any kind of sound: music, musical instruments, voices or effects. Processing of independent tracks or mastering of final mixes. Processing possibilities include dynamic enhancing, smooth equalizing, waveshaping compression, frequency enhancing, sound "revitalizer" for old mixes, user defined harmonic redistribution.

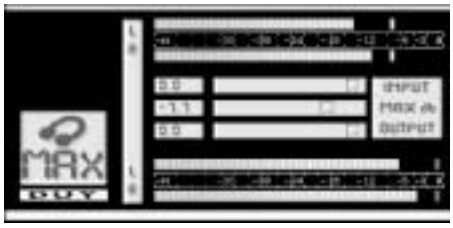
- ◆ FDWS (Frequency Dependent WaveShaping) algorithm processing
- ◆ Adjustable 3-band smooth filter with full audio range continuous crossover points
- ◆ High quality analog sound filters using modeling technology
- ◆ Three independent user-defined Shapers with virtually infinite resolution and accuracy. Each shaper has:
  - – Eight different Shaper curve types including Linear, Log and Cosine functions.
  - – Linear expand function to optimize dynamics
- ◆ A set of 5 factory presets for typical applications
- ◆ Simultaneous input and output plasma-style meters
- ◆ Mix switch for Low, Mid, Hi Shape or all (default) which allows easy independent adjustments

DUY Wide



- ◆ Spatial enhancing and widening of stereo image.
- ◆ Sound placement outside of physical stereo speaker locations.
- ◆ Individual per channel phase inverter
- ◆ Mono compatible.
- ◆ Doesn't add unwanted color to the signal
- ◆ Range of effect from subtle to dramatic
- ◆ High signal to noise ratio due to an internal resolution of 48 bits.
- ◆ Spatial enhancing of existing mixes, both independent stereo tracks and final mixes.
- ◆ Remastering, rebalancing and spatial enhancing of stereo and multichannel mixes.
- ◆ Complex spatial processing of pairs of channels in multichannel mixes.

MAX Duy



- ◆ Exclusive ILO (Intelligent Level Optimization) algorithm provides:
- - Seamless level maximizing.
- - Zero harmonic distortion even at low frequencies.
- - Unnecessary level scaling and limiting thus improving signal to noise ratio
- - Release free operation avoiding unwanted pumping effect.
- - Overall improvement in signal to noise ratio.
- ◆ Easy and intuitive user interface
- ◆ Maximum resolution peak level meters with hold and shift function.
- ◆ High signal to noise ratio due to an internal resolution of 48 bits.
- ◆ Sound level optimization for music, film soundtracks, broadcast and multimedia.
- ◆ Suitable for the processing of any kind of sound: music, musical instruments, voices or effects, for both individual tracks and final mixes.
- ◆ CD mastering

DUY Z-Room



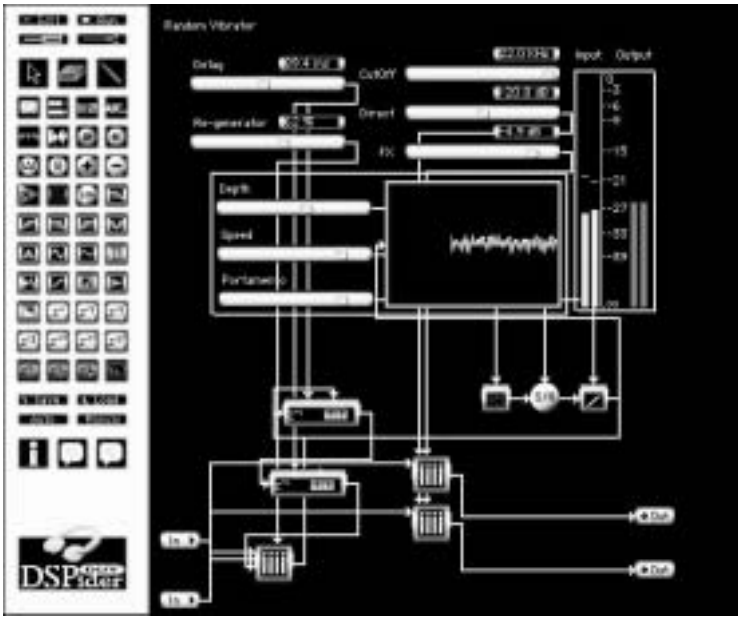
Z-Room is a high-quality reverb, it features excellent density and diffusion that provide an unparalleled sound quality. DUY Shape is a sound enhancer featuring 3-band smooth filters with full audio range continuous crossover points and three independent user-defined shapers with virtually infinite resolution accuracy. DUY Wide is a stereo and multichannel spatial enhancer, which allows widening of the stereo image, as well as sound placement outside the physical speaker locations. DUY Wide is mono-compatible. Max DUY is a sound level maximizer based on DUY's exclusive ILO algorithm, featuring seamless level maximizing, zero harmonic distortion even at low frequencies, and release-free operation.

DIGIDESIGN

DUY — DSPIDER

"The Plug-in Creator"

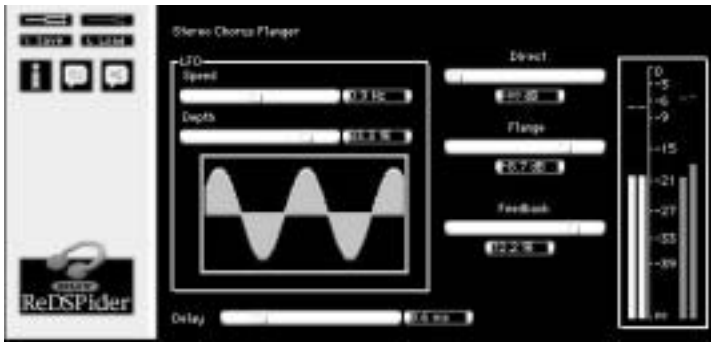
DSPider is a modular plug-in creator for TDM that consists of 40 modules that can be linked by the user in countless different ways, providing a unique and revolutionary approach to the creation of user-defined processors. DSPider allows the development of virtually any type of existing processing device, and provides a preset library with over 220 loadable and modifiable presets such as compressors, reverbs, equalizers, limiters, synths, noise reduction systems, 3D effects, de-essers, sound effects generators, etc. Applications include the whole spectrum of the audio world: post-production, music, broadcast, mastering, multimedia, education, research, etc.



- DSPider allows users to "build" their own plug-ins. As a result, a huge range of unique and exciting effects and processors can be created with just one plug-in.
- Instant assistance is provided by help balloons and "talking help". Electronic instructions for patches enable easy and intuitive use.
- 3rd party developers can create or modify patches, allowing the development of virtually any type of existing processing device, including those that don't currently exist in the marketplace. This makes possible the creation of a vast range of unique proprietary processors and signature sounds. A "Locked Patch" option allows developers to maintain the secrecy of their algorithms.
- Advanced Mode provides a large working surface and enables you to edit and run patches, whereas the Reader Mode has a reduced screen and is designed especially for users to run patches.
- Patch Manager lets users switch between patches automatically, allowing the optimization of DSP resources and a previously unavailable degree of control and flexibility.

The Modules:

DSPider´s 40-module list ranges from simple operators such as addition, subtraction, multiplication, shift, etc., to multi-functional modules like filters, oscillators, waveshapers, filters, multiple delays, envelope followers, pitch trackers, ramp generators, sample-and-hold, spectral shapers, etc. A flexible multi-reflexion chamber and multiple delay modules allow the creation of complex user-defined reverbs. By providing modules previously only available in modular synthesizers, DSPider permits exclusive processing capabilities and the creation of entirely new audio processors. Graphic modules like sliders, numeric readouts, plasma meter readers, scopes, etc., allow the creation of complex graphics. All modules can be easily programmed and patched with a simple "drag and drop" procedure.



DUY —REDSPIDER

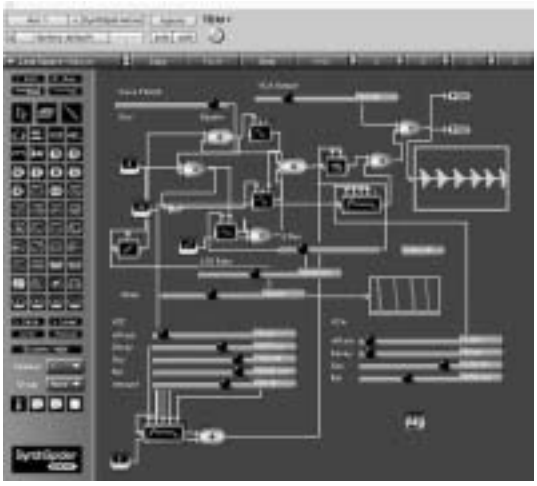
Plug-in Library for TDM

Based on the great success of DSPider, DUY's ReDSPider is the first plug-in library for the Digidesign TDM Bus. ReDSPider features over 220 plug-ins which include compressors, limiters, reverbs, noise gates, expanders, equalizers, 3D effects, synth sounds, sound enhancers, de-noisers and many more. You don't need DSPider to use ReDSPider! Supports sample rates up to 96 kHz

DUY — SYNTHSPIDER

Creating Sounds and FX

A modular TDM synthesizer software, SynthSpider links the 40 supplied modules, allowing you to create your own sounds and sequences. The high quality of DUY's algorithms is reflected in the power of the modules, which include filters, envelope followers, wave generators, and even a built-in analog-like sequencer, plus a "scanwave generator" that allows you to use external audio. Unlike other TDM synthesizers, SynthSpider's modularity makes it the most versatile tool of its kind. MIDI-compatible, SynthSpider includes the most powerful tools to make your sounds be totally unique. The perfect tool for sound designers, composers or sound effects creators, SynthSpider's applications include the whole spectrum of the audio world: post-production, music, broadcast, multimedia, education, research, etc.



FEATURES

- A large library of presets is provided with SynthSpider. A "normal" synthesizer would just include variations of an identical setup, but due to the modularity of SynthSpider, all these presets have a different internal structure and open a whole range of possibilities for the user, who can modify any of the given presets and save them as proprietary.
- The "Locked Patch" option allows developers to maintain the secrecy of their algorithms.
- Users can create or modify presets, allowing the development of virtually any type of sound, that doesn't currently exist in the marketplace. This makes possible the creation of a vast range of unique proprietary sounds.
- The Advanced Mode provides a large working surface and enables you to edit and run patches, whereas the Reader Mode has a reduced screen and is designed especially for users to run presets.
- The exclusive Patch Manager feature provides users the possibility to switch between presets automatically, allowing the optimization of DSP resources and a previously unavailable degree of control and flexibility.
- Instant assistance is provided by help balloons and "talking help". Electronic instructions for patches enable easy and intuitive use.

DUY GLOBAL TDM BUNDLE

The Perfect Software Package for any TDM Studio

DUY's Global Bundle contains six high-quality TDM plug-ins that will become the best solution for your ProTools-based studio:

DaD Valve & Tape ("DUY Analog Bundle"): analog tube and tape saturation and compression emulation

DUY ReDSPider: a package of over 200 different TDM plug-ins including compressors, reverbs, EQs, special effects and distortion.

DUY Shape: an adjustable 3-band waveshaper that will become the only tool you will ever want to use for high-precision sound definition. You can modify the curves to achieve expansion/compression of the signal levels. Perfect for defining individual sounds of mastering entire mixes.



Max DUY: Sound level maximizer, featuring DUY's exclusive ILO algorithm for the optimization of the level of the signal, the ease of use combines with the sound quality. Max DUY can also be used as a limiter.

DUY Wide: Spatial stereo enhancer. Features frequency compensation and includes a high resolution phase meter

EMAGIC

LOGIC 6 SERIES

Music Production Software For The Mac

Emagic's Logic 6 Series consists of a choice of three levels of digital audio and MIDI production software applications for the Mac OS— Logic Audio, Logic Gold and Logic Platinum. Each version of Logic has a virtually identical feature set combining support for high resolution 24-bit / 96kHz audio files (192kHz with Logic Platinum), and unlimited MIDI tracks along with a plethora of virtual plug-in effects and instruments along with powerful sample-accurate automation capabilities of just about every parameter. All versions include a myriad of editing capabilities for audio and MIDI as well as scoring. Support for a wide range of third party hardware and plug-ins as well as Logic's own plug-ins are just part of Emagic's commitment to providing their customers with cutting edge production tools. Logic Audio allows you work with up to 48 stereo or mono audio tracks, Logic Audio Gold provides 64 stereo or mono audio tracks and Logic Audio Platinum 5 provides 96 stereo or mono audio tracks. Whether you're a beginner, project studio or professional, there's a version of Logic with the right feature set and functionality to suit your needs.



They All Feature

- Optimized for Apple's G4 Velocity Engine including dual processor models.
- 960 ppqn timing resolution (1/3840 Note)
- Tempo resolution accurate to 1/10,000 bpm from 0.05 to 9,999 bpm
- Multiple undo/redo

Audio

- Ultra-precise 32-bit internal processing with support for 16 and 24-bit audio files up to a sample rate of 96kHz (192kHz with Logic Platinum)
- Support for MacAV, ASIO, Direct I/O, and Audiowerk under OS 9.x and Core Audio under OSX.
- The built-in Sample editor allows sample accurate non-destructive and DSP-based editing of mono and stereo audio files
- Entire mixes complete including live inputs, effects and automation can be bounced to disk in realtime
- The transparent POW-r dithering process assures that high resolution recordings retain their true sonic character even at 16Bit/44.1kHz CD resolution.
- Hardware independent audio scrubbing
- Hardware independent realtime sample-rate conversion.

MIDI

- Virtually unlimited MIDI tracks
- Non-destructive, realtime quantization, transposition, velocity, track delay, gating and more
- Multiple MIDI editing environments: Matrix Editor, Event List, Hyper Editor, Score Editor and Transform Window
- Ultra-precise MIDI timing using Emagic's AMT8 and Unitor8 MIDI interfaces
- OMS, MTP (OS9) & CoreMIDI (OSX) support
- A wide range of synchronization options including MTC, MMC and SMPTE ensures that Logic connects easily to both the digital and analog worlds

Effects and Virtual Synths

- Includes a wide variety of high end effect plug-ins as well as three soft synths
- Support is provided for Emagic's line of virtual instruments as well as third party VST2.0 plugins and virtual instruments under OS9 and Audio Units under OSX

Track-based Automation

- Track-based, sample-accurate automation is provided, with 32 Bit data resolution, for virtually every parameter including effect plug-ins and Audio Instruments

Logic Control Support

- Logic Control is a hardware control surface, co-designed by Emagic and Mackie Designs, that offers complete hands-on control over hundreds of MIDI and audio functions, including Logic's 32-Bit track automation system.

Screensets and Key Commands

- Instantly switch between 90 screensets, containing customized window configurations including window size, position and zoom settings, using your computer's keyboard numeric keypad you are able to
- The most important program functions and navigational macros can be quickly accessed by 100s of user-definable Key Commands and MIDI commands

Environment

- Provides extensive control and customization of your external MIDI set-up, audio routing, mixing and real-time effects
- Assign Logic's software-based processing devices (faders, arpeggiators, delays and more) to create unique controllers for your MIDI gear and audio tracks.
- Create a different environment for your MIDI instruments, audio tracks, audio instruments, etc... and store a Screenset for each one

LOGIC 6 SERIES

Arrange Window – Logic's Primary Workspace with an Emphasis on Realtime Composition

Transport controls
Cycle record/playback
Punch in/out recording
Track solo
Metronome and Sync options
Tempo and time signature

Track Information
Track number, Mute and Record buttons Track icon and name Current automation parameter, meter and mode

Markers allow you to quickly "jump" to any location in your songs

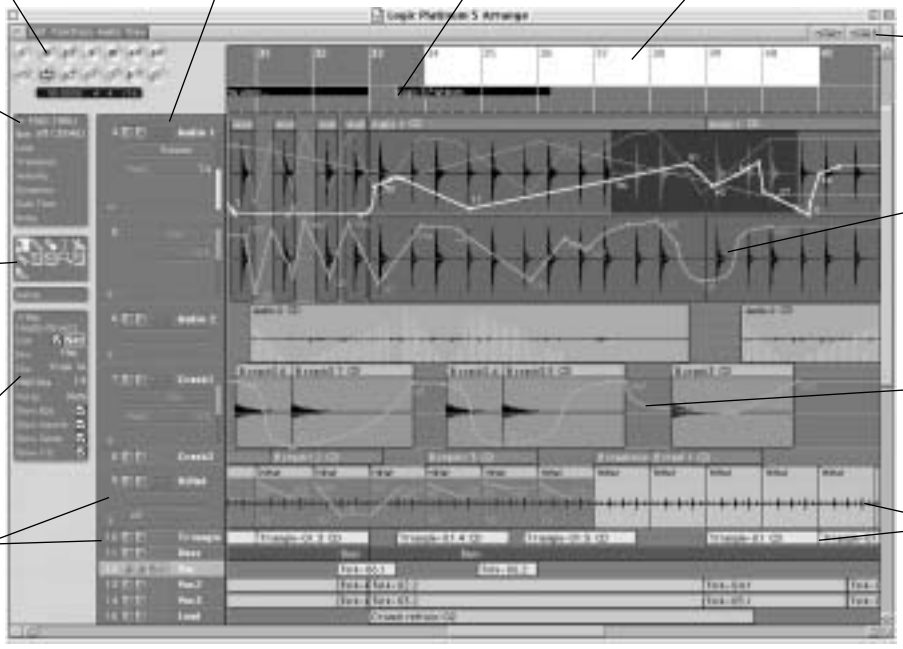
Click and drag your mouse from left to right along the timeline to select the desired bars for cycle record/playback

Individual Sequence parameters include realtime non-destructive control over quantization, transposition, velocity, dynamics (velocity compression), gate time, delay (by note value or ms.) as well as loop on and off

*The Tool Palette includes a pointer, pencil, eraser, scissor, glue, crossfade and other tools for editing audio regions and MIDI sequences – hit your keyboard's **esc** key and the Tool Palette will appear at your cursor position*

Track Parameters
Track name, icon, device (audio driver) or port (MIDI), MIDI Channel and more..

Audio and MIDI objects can all be individually resized, soloed, muted, looped and mixed in the Arrange window - all in real-time and without stopping the sequencer or your creativity



Zoom in and out of the Arrange window (horizontally and vertically) using the telescope tools

Color-coded automation data is displayed directly in the Arrange window as envelopes with break points that can be freely drawn, edited and scaled

Automation curves are freely adjustable between convex, concave, s-form or linear shapes

Because automation is track-based, it can be copied and moved independently of their associated MIDI sequences and audio regions

MIDI sequences and audio regions can be recorded, arranged and edited in a virtually identical fashion.

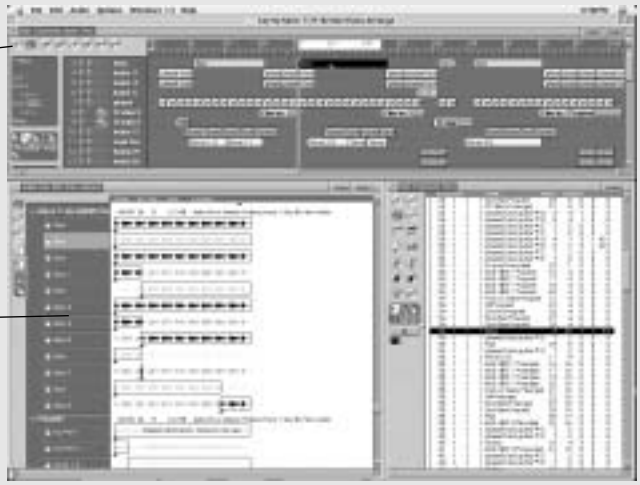
File Formats and Compatibility

- Mono, interleaved stereo and split (L/R) stereo audio files can be recorded, imported and exported using Sound Designer II, Aiff and Wave file formats
- REX2 (recycle) files can be imported into the arrange window and in the optional EXS24 mk2 software sampler
- The audio output of ReWire compatible software instruments, such as Propellerhead's Rebirth and Reason, can be streamed into Logic's mixer where they have full access to Logic's extensive array of effects and mixing facilities.
- Built-in Quicktime movie support allows you to efficiently view and synchronize video within your project. An ideal for scoring to picture without the need to synchronize to an external VTR.

Access up to 90 customized Screensets like this one with a single keystroke

Each editing window features a Catch and Link button that when activated allow multiple windows to interactively follow the currently active window's event selection as well as any movement along the timeline – Notice that the bass region selected in the arrange window is also highlighted in the Audio and Event List windows

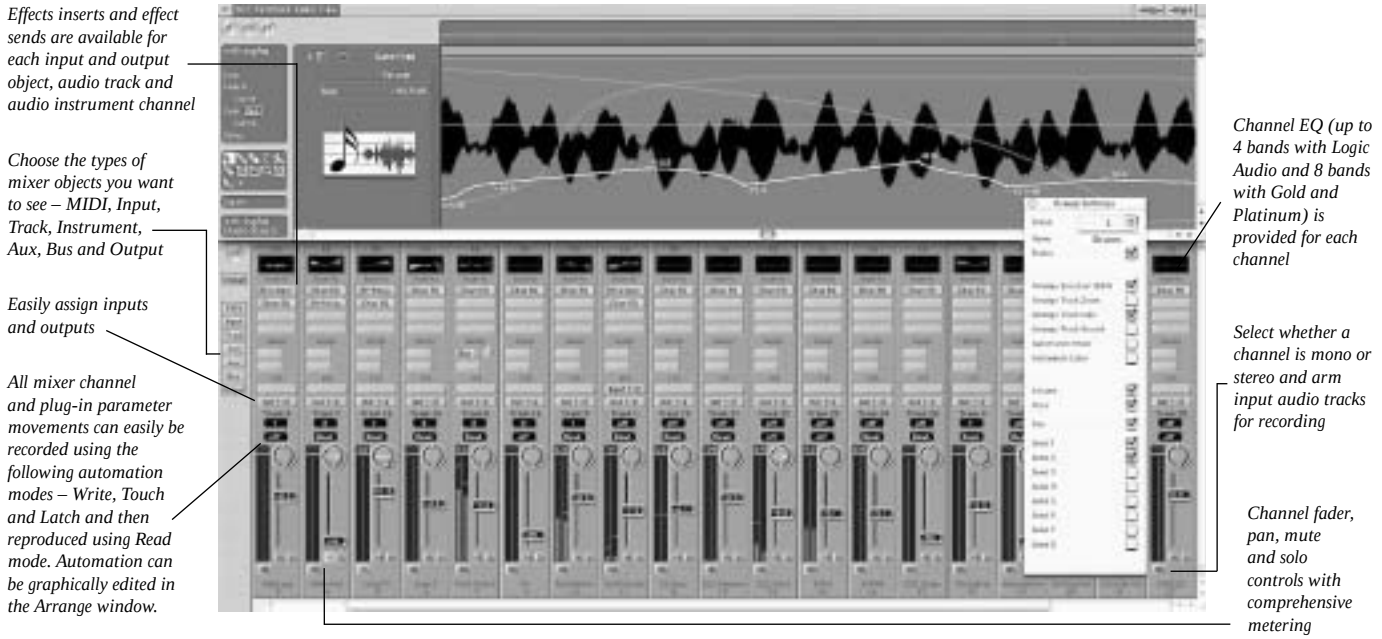
Audio Window
Audio files and their associated audio regions, recorded (or imported) into your session, are automatically listed in the Audio window where they are easily organized. Audio files and regions can be auditioned here and then dragged onto the desired track in the arrange window. Double clicking on an audio file or region will launch the selection in Logic's built-in Sample Editor



LOGIC 6 SERIES *continued*

Adaptive Track Mixer

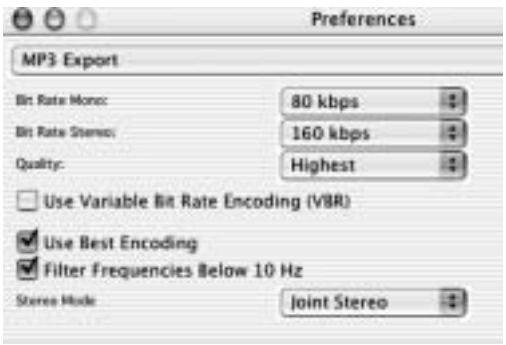
The Adaptive Track Mixer reflects the track layout of audio and MIDI channels shown in the Arrange window.



Customizable Audio Objects

- Audio Track Objects are used for record and playback of audio files.
- Audio Input Objects allow you to monitor live audio inputs with effects. Logic Gold and Platinum add the ability of pre-processing (recording) Input Objects with effects
- Bus objects (mono/stereo) are available for use as effects returns and for creating sub mix groups.
- Aux Input Objects (mono/stereo) are available for creating multiple outputs for virtual instruments and for adding functionality to the busses.
- Audio Instrument objects can be created for accessing virtual instruments.

Support of More Audio Formats (OS X only)



Logic 6 can import and export MP3 files. In addition, QuickTime audio formats can be imported and exported including: AAC, Qdesign Music 2, Qualcomm PureVoice, Law, Alaw and IMA formats.

The master output object features a Bounce to disk button that allows you to bounce any number of tracks, even an entire mix, including any automation and effects to a mono or stereo (single or split left and right) audio file.



POW-r Word Length Reduction

- POW-r is a high-end dithering algorithm designed by the POW-r consortium — a team of digital audio designers who come together to create one of the audio industry's most transparent and dynamically stable methods of Word Length Reduction.
- POW-r #1: uses a special dithering curve to minimize quantization noise.
- POW-r #2: (Noise Shaping): uses additional noise shaping over a wide frequency range which can extend the dynamic range by 5-10 dB.
- POW-r #3: (Noise Shaping): uses additional, optimized noise shaping which can extend the dynamic range by 20 dB within the 2-4kHz range - the range the human ear is most sensitive to.

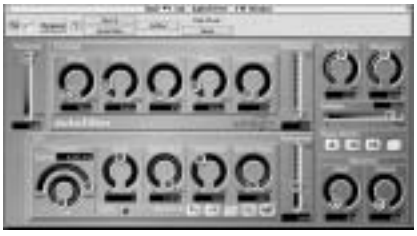
LOGIC 6 SERIES *continued*

Fully Automated Plug-in Effects System

- Dozens of effects plug-ins are included covering a wide range of precision and creative processing tools from the standard dynamics, reverbs, delays and EQs, to unique sound sculpting tools and mastering processors
- All plug-ins operate at 32-Bit resolution, and support sample rates from 11kHz to 200kHz, ensuring outstanding audio quality and making signal overloads virtually impossible
- Third party support is also provided for VST/VST2 (OS9 only) and Audio Units under OSX
- Plug-ins can be accessed in a number of ways including: Inserted directly on to live input channels; audio playback tracks; or virtual audio instruments; they can also be inserted on busses and aux channels and accessed via direct channel routing or via channel sends



- Sidechain Inputs (Gold and Platinum only) — Audio tracks, inputs and busses can be used as sidechain sources for sidechain capable plug-ins and audio instruments. For example, this allows the optional EVOC20 vocoder to process live audio inputs via the sidechain input





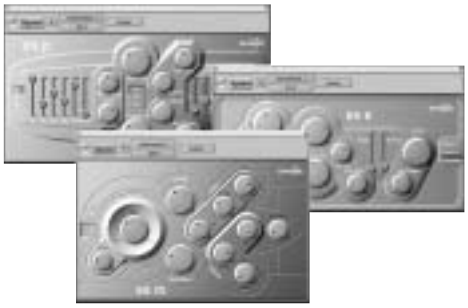
- The Plug-In I/O (Platinum only) allows you to insert the signals of an external processor into any audio object channel. A pop-up menu provides access to the I/O available in your audio hardware.
- Effects can be assigned to live input channels for monitoring purposes or for pre-processing.

Virtual Audio Instruments

- Soft synths and samplers including Emagic's optional ES1, ES2, EXS24, EVP88, EVD6 and EVB3 can be inserted into Audio Instrument channels and seamlessly integrated into Logic's mixer with access to effects and automation
- Third party support is provided for VST2 under 09 instruments and Audio Units under OSX
- Playback timing for all audio instruments is sample accurate
- Logic Audio supports up to 16 audio instruments, Logic Audio Gold supports up to 32 and Logic Audio Platinum supports up to 64
- Multiple Outputs for virtual instruments is also supported

Integrated Synthesizers

- Logic 5 incorporates three powerful subtractive analog software synths as standard – The ES M monophonic bass synthesizer, the ES P for the creation of polyphonic sounds and the ES E for ensemble sounds



REALTIME NATIVE PLUG-INS

	Audio	Gold	Platinum
Reverbs:			
3.0 Reverb	✓	✓	✓
AVerb	✓	✓	✓
SilverVerb	✓	✓	✓
GoldVerb		✓	✓
PlatinumVerb			✓
EnVerb			✓
EQ:			
DJ EQ	✓	✓	✓
Fat EQ		✓	✓
Hi and Low Shelving	✓	✓	✓
Parametric	✓	✓	✓
Silver EQ	✓	✓	✓
Hi and Low Cut Filter	✓	✓	✓
Hi and Low Pass Filter		✓	✓
Delays:			
Stereo Delay	✓	✓	✓
Sample Delay	✓	✓	✓
Tape Delay	✓	✓	✓
Dynamics:			
Silver Compressor	✓	✓	✓
Silver Gate	✓	✓	✓
Compressor		✓	✓
Expander		✓	✓
Limiter		✓	✓
Noise Gate		✓	✓
DeEsser			✓
Adaptive Limiter			✓
Multipressor			✓
Distortions:			
BitCrusher		✓	✓
Distortion	✓	✓	✓
Clip Distortion	✓	✓	✓
Overdrive	✓	✓	✓
Phase Distortion	✓	✓	✓
Modulation Effects:			
Chorus	✓	✓	✓
Flanger	✓	✓	✓
Modulation Delay	✓	✓	✓
Tremolo	✓	✓	✓
Ensemble		✓	✓
Phaser		✓	✓
Spread'er			✓
Enhancers:			
StereoSpread			✓
SubBass			✓
Exciter		✓	✓
Special:			
Volume	✓	✓	✓
Gain'er	✓	✓	✓
AutoFilter			✓
Enveloper			✓
Native I/O Insert			✓
Pitch ShifterII	✓	✓	✓
Spectral Gate			✓
Oscillator		✓	✓
DeNoiser	✓	✓	✓
Helper:			
Direction Mixer		✓	✓

LOGIC 6 SERIES *continued*

MIDI Editor Windows

Logic's linked editors provide the power to fine-tune your music at any level. View a composition in its entirety or select and make changes to the smallest detail with the Matrix, Event List, Score or Hyper Editors.

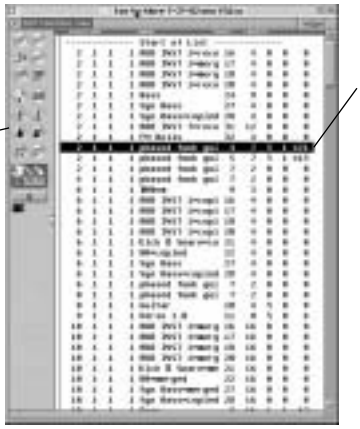
The Matrix Editor is a piano roll style editor that offers an intuitive approach to editing the pitch, placement and duration of MIDI events using a vertical piano keyboard and horizontal timeline axis
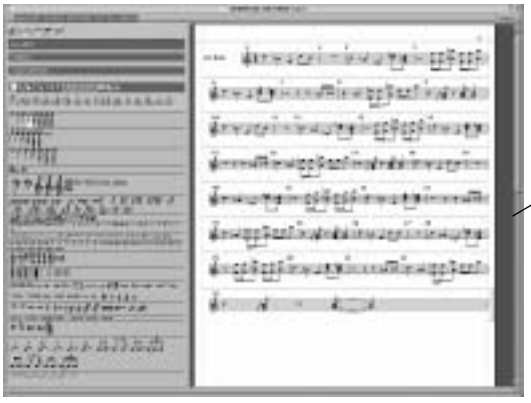


The velocity of each MIDI note is represented by the intensity of the events color (blue = 0, red = 127). A velocity tool allows you to adjust the velocity of selected notes.

Use Hyper draw to automate MIDI controller data

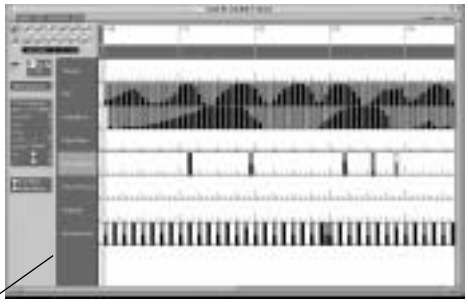
A filter allows you to choose the specific type of data you would like to edit.



The Event List Editor can be used to view and edit the position, track and length of sequences and audio regions in the arrange window as well as let you view and edit individual events (notes, controller data, pitch bend, program change) within MIDI sequences.

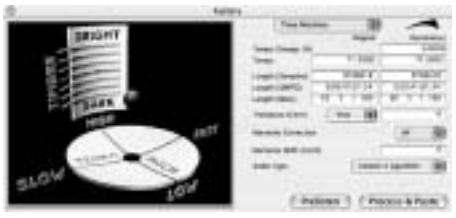


The Score Editor is a MIDI editor that displays traditional music notation. It also provides layout functions which make it possible to create professional score printouts quickly and easily.



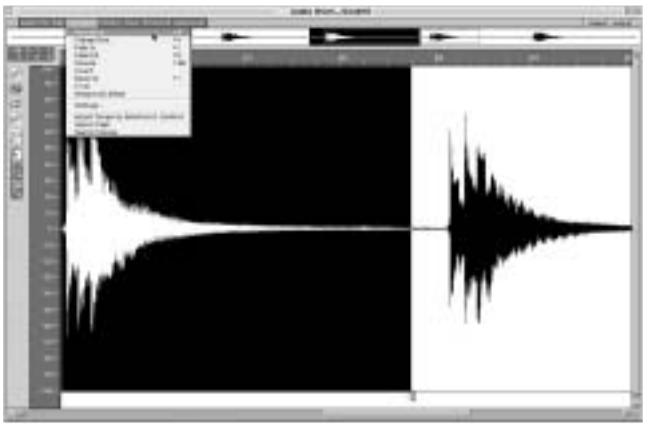
The Hyper Editor is the ideal Step Sequencer. You can insert and edit note and controller data quickly and easily using only a few simple tools. Event values are represented by vertical bars. Use the pencil tool to insert single events such as program changes or draw sweeping curves to control filter cutoff etc..

Sample Editor



The Sample Editor provides a wide range of sample accurate waveform editing capabilities from creating regions to performing DSP functions including Normalize, Trim, Silence, Reverse, Fade In/Out, Remove DC Offset, Time compression/ expansion and formant corrected pitch shifting using Time Factory II and much more.

The enhanced TimeMachine II time compression and expansion, available in Logic Gold and Platinum, provides five algorithms designed to work with specific source material – Monophonic, Pads, Beats, Version 5 and the default Any Material.



LOGIC 6 SERIES *continued*

DOT – DSP Optimization Technology

- Native support is provided for the latest Apple G4 Velocity Engine-equipped computers including multi-processor models. In some cases, these optimizations can increase the total number of tracks, effects and audio instruments that are available for simultaneous use.
- DOT can effectively double the amount of available CPU power, when DOT optimized plug-in modules are used in Logic. A number of Logic plug-ins are specifically optimized for the latest G4 processors.

Rock Solid MIDI Timing and Synchronization with AMT (Active MID Transmission)



The Logic Series incorporates a proprietary Emagic technology called AMT (Active MIDI Transmission). Using Emagic's Unitor8 or AMT8 MIDI interfaces with Logic ensures that MIDI timing is precise across all ports, even in the largest of MIDI systems. Literally hundreds of MIDI ports can be addressed, with MIDI data arriving simultaneously at each, no matter how dense the MIDI arrangement. If you need to synchronize with external sources, the entire Logic Series will send and receive MIDI Clock, MTC, MMC and word clock signals, making it ideal for film, TV and post-production facilities.

XSKey Expandable System Key – Copy Protection System



This programmable hardware key for the USB port includes the license for Logic 5, and temporary licenses for all Emagic software instruments.
Following the initial launch of Logic 5, you can try out all Emagic software instruments for a period of one month without any functional restrictions.

	Logic Audio	Gold	Platinum
Audio Files and Processing:			
Highest Supported Bit Depth / Sample Rate	24-bit/96kHz	24-bit/96kHz	24-bit/192kHz
Ultra-precise 32 Bit internal processing	✓	✓	✓
Export/Import Region Time Stamps	✓	✓	✓
Realtime Variable Pitch	–	–	✓
Software Sample Rate Conversion	–	–	✓
Arrange Window:			
Nondestructive Audio Editing	✓	✓	✓
Nondestructive Crossfades	✓	✓	✓
Variable fade curves for crosssfades	–	✓	✓
Time Stretching In Arrange	✓	✓	✓
Arrange Punch on the Fly		✓	✓
Max Number:			
Playback Tracks (single audio hardware)	48	64	128
Playback Tracks (multiple audio hardware)	N/A	128	256
Inserts per Track	4	8	15
Busses / Aux Channels	8 / 8	16 / 16	64 / 64
Audio Instruments	16	32	64
Multiple Outputs for Audio Instruments	8	16	64
Max. Simultaneous I/O	12	24	Unlimited
Number Of Included Plug-Ins	28	42	53
Multiple Undo/Redo Steps	3	30	Unlimited
Undo/Redo History	–	–	✓
Definable Key Commands	300	600	800
Side Chain for Plug-Ins	–	4	4
Hardware Support:			
ASIO Support (OS 9)	✓	✓	✓
Core Audio (OS X)	✓	✓	✓
Digidesign Direct I/O (OS 9)	✓	✓	✓
ProTools TDM (OS 9)	–	–	✓
ProToolsHD	–	–	✓*
Logic Control Support	✓	✓	✓
Mixing and Automation:			
32-bit Sample Accurate, Track-based Automation	✓	✓	✓
Freely Adjustable Automation Curves	–	–	✓
Adaptive Track Mixer for Audio & MIDI tracks	X	X	X
Software Monitoring	✓	✓	✓
Audio Input Objects monitoring with effects	✓	✓	✓
Pre-processing (recording) with effects	–	✓	✓
Bus and Input Objects as sidechain inputs	–	–	✓
Copy complete mixer configurations and their settings		✓	✓
Route track to bus / Route bus to bus	✓ / –	✓ / ✓	✓ / ✓
Tracks as side chain input source	–	✓	✓
Surround (including TDM systems)	–	–	✓
ReWire / Rewire2 support	✓	✓	✓
Track / Offline Bouncing with effects	✓	✓	✓
High-end POW-r dithering Algorithm	✓	✓	✓
File Support:			
MP3 import/export/convert (OS X only)	✓	✓	✓
REX2 File Support	✓	✓	✓
OMF (Avid/ProTools) import/export	–	–	✓
Open TL import/export	Import only	Import only	✓

**Requires PTHD extension*

LOGIC 6 SERIES *continued*

Emagic has released version 6 of the Logic Series. This update provides a new level of processor efficiency and extensive enhancements to the Arrange window that offers more editing and mixing power than ever before. In addition, Logic 6 debuts two new setup and file management solutions — the Setup Assistant and the Project Manager. Users will also benefit from an improved feature set for film scoring, a newly-developed superior sounding EQ and MP3 import/export functionality. Further optimization of the automation system round out this update.

Logic 6 Arrange and Adaptive Track Mixer Window Features

The "Freeze" button is a CPU resource maximizing feature available to all audio and audio instrument tracks. Activating the "Freeze" button automatically creates an offline bounce of an audio or audio instrument track and disables any of the plug-ins associated with that track. The unprocessed track is then replaced with the processed (or bounced) track thus freeing-up potentially very large amounts of CPU-power which can be used for other purposes. Tracks can be unfrozen at any time for further editing.

User-definable Arrange window Icons.

Logic 6 also supports Time Stretching directly in the Arrange window. This allows the length of an audio file or a loop to be quickly adapted to the chosen bar.

Sample accurate display resolution in the Arrange window.

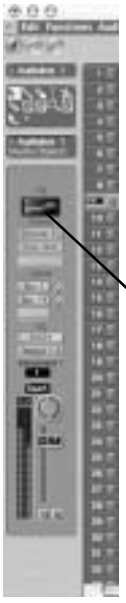
The Marquee Tool is an arrange window tool that allows region and sequence-independent selection and editing of data.



The Video Thumbnail track (Logic Platinum only) displays single frames of QuickTime video, horizontally in the Arrange Window. The higher the zoom resolution, the higher the number of thumbnails will be displayed. A DV-formatted QuickTime movie can be output via FireWire to a standard DV camera or FireWire conversion box allowing you to view the video on a standard TV or video monitor.

(Logic Platinum only) Each channel can now be assigned to one or more of 32 available Groups. Each group can be a mix group, an edit group, or both. This feature improves Logic Platinum's fundamental automation engine while at the same time accelerates workflow and affords users improved live editing possibilities. Such as, improved vocal editing, assigning level meters to a group, or muting drum sets.

The Offline Bounce feature allows users to bounce tracks faster than real-time or to bounce songs that would otherwise cause system overloads.



The Arrange Channel Strip

You can view and edit a tracks channel strip from within the Arrange window. Selecting the desired track in the Arrange window will display the corresponding channel strip in the parameter section. This allows you to edit and mix a track without ever leaving the Arrange window.

The Channel EQ



The Channel EQ is an intuitive 8-band EQ (4-band in Logic Audio) with fixed modes: a highpass filter, a low shelving EQ, four parametric EQ bands, a high shelving EQ and a lowpass filter. Each band is fully parametric. Each channel EQ has an optional "high resolution analyzer" graphic display mode that can be applied pre/post the EQ allowing you to analyze the signal before and after the EQ.

Expanded Control Surface Support



Logic 6 can be controlled by the following third-party hardware controller units: CM Automation Motormix, Mackie HUI and Baby HUI, Radikal Technologies SAC-2k and SAC 2.2, Yamaha DM2000, 02R96 and 01V96. These hardware controllers can be used in combination with each other or as part of an expandable Logic Control and Logic Control XT system.

LOGIC 6 SERIES *continued*

Quick Automation Access

- The integrated One Fader Automation facility allows the selected automation parameter to be controlled by any controller such as a Mod Wheel.

Improved Time Machine

- The Time Machine in Logic 6 features a new algorithm that provides enhanced sound quality and more accurate time stretching and pitch shifting.

The Setup Assistant (OSX Only)



- The Setup Assistant Starting allows the initial setup of Logic to be quicker and easier than ever.
- The Setup Assistant is an unique tool that guides users step-by-step — even the extreme novice — through the process of setting up and configuring Logic. In a simple and interactive Q & A session users learn how to easily create preferences, configure key settings and create their first template song in Logic.

Save As Project

- All media files can be saved in a new "project" file format, making archiving and transport of projects created in Logic both efficient and easy.

LOGIC PLATINUM ADDS –

Intelligent Project Manager

- The Project Manager consolidates all media files on local volumes by recognizing the dependencies between audio files, sampler instruments, settings and movies.
- The Project Manager allows you to rename files, add or edit comments and change references to one, or multiple song files.
- Extensive search facilities further enhance the usefulness of this tool which can be accessed directly from the arrange window.

Surround Sound

- Surround mixing is available in any of 12 formats (up to 7.1) for audio channels, busses and instruments. Individual formats can be assigned for each track, bus or instrument.
- Import and export of OMF (Avid / ProTools) files

Multiple Undo with History

- Easy to use multiple undo/redo system with a user definable number of maximum undo/redo steps
- The undo/redo history window gives access to a complete script of the song's evolutionary progress, allowing any step to be selected, and all steps up to that point to be reconstructed. This history can be stored, and reactivated the next time the song is launched.

Added Functionality For ProTools Users

PTHD Extension



The Emagic PTHD Extension connects Logic Platinum and Pro Tools HD hardware to form a uniquely powerful system, providing not only superlative audio quality and generous I/O capabilities, but also – via Digidesign's TDM II – access to an ever-increasing number of exciting DSP plug-ins.

- Enables Pro Tools HD audio hardware to run together with Logic Platinum 5.1.3 or higher on Mac OS 9.x
- Supports up to 128 tracks, 24Bit/192kHz, TDM II

ESB TDM



Provides a link between your TDM system, and your computer's previously unexploited native CPU processing resources.

- Allows Logic's native audio engine to be routed into the TDM mixer. This enables all Logic tracks, native and VST plug-ins, including Audio Instruments, to be used in conjunction with your TDM system DSPs.
- ESB TDM also enhances the functionality of the EXS24 by allowing it to be inserted into the Aux channels of Logic Platinum's TDM mixer.

Host TDM Enabler



- Allows you to insert ES1, ES2, EVP88, EVB3 and EVD6 in TDM Aux channels under any TDM/HTDM host software
- Software instruments, using your computer's CPU, can be processed directly with any of your TDM plug-ins.
- Can be used with Logic Platinum, Motu Digital Performer or Digidesign Pro Tools Software (running on Pro Tools 24 MIX, Pro Tools MIX or ProTools HD systems).
- Supports up to 32 channels (16 stereo channels) with sample rates up to 192 kHz.

EMAGIC

EXS-24 MKII

Emagic Xtreme Sampler 24-bit

The ESX-24MKII is professional 64-voice software sampler whose integration into Logic's mixing, sequencing and sample editing environments makes it an essential component for all Logic users. The ESX-24MKII provides sample-accurate playback timing, stunning audio quality up to 24-bit/96kHz and is fully accessible to all of the effects and automation capabilities available within your version of Logic. The EXS24MKII's Instrument Editor provides an ergonomic user interface that allows you to quickly and easily assign and edit individual samples for designing simple single sample instrument or complex multi-sample, multi-layered sample instrument. The advanced synthesis engine features an awesome sounding multimode filter section, fast envelopes and three tempo-syncable LFOs as well as an extensive range of modulation routing possibilities.



FEATURES

- Multiple instances of EXS 24, each with up to 64 simultaneous voices, are supported within each version of Logic at extremely low processor loads.
- Assign samples to up to 5 stereo pairs and 6 individual mono outputs per instance of the EXS-24mkII.
- The amount of available sample time is limited only to the amount of available RAM in your computer or by available hard disk space when using VSM (Virtual Sample Memory).
- User-defined hierarchical menu structure and intelligent Sample Instrument search facility with naming filter.

Perfect Integration With Logic

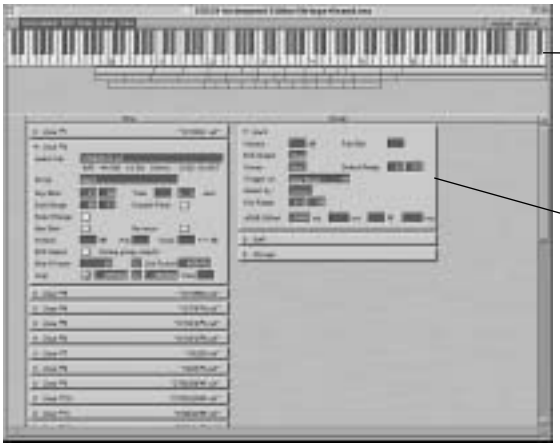
- Each EXS-24 instance is loaded as an Audio Instrument in Logic's 32-bit internal mixer where it has full access to all effects plug-ins and automation functions.
- Individual samples can be launched directly into Logic's Sample Editor for fine tuning sample and loop start and end points and for applying DSP processing.
- All Sample Instruments and their settings are automatically saved and loaded with your song.

Sample Format Compatibility

- Reads AIFF, WAV and SDII in 8 to 24 Bit depths and up to 96 kHz
- Converts AKAI S1000/S3000, SampleCell and SoundFont2 as well as Gigasampler format samples and programs
- Import ReCycled (.REX) files
- The imported samples can be in any bit resolution from 8 to 24 Bits
- A CD ROM with over 500MB of high quality sounds will be shipped to you after registering

The Instrument Editor uses Zones and Groups to control the playback characteristics of individual samples contained in a Sample Instrument. You can add as many Zones and Groups as you like to create anything from a single sample played across the entire keyboard to dense, multi-layered constructions comprised of numerous individual samples.

Each sample in a Sampler Instrument is loaded into its own Zone where you can define many of that sample's playback properties.
– Key Note as well as upper and lower key range (zone)
– Assign the Zone to a Group if desired
– Minimum and maximum Velocity ranges
– Volume and pan as well as a Scale function which balances the levels between the lower and higher notes in the keyboard range.
– Non-destructive reverse playback
– Sample tune/fine tune and pitch disable
– Sample start and end points as well as loop start and end points with fine tune and auto crossfade adjust.
– One Shot parameter plays a sample straight through — ideal for triggering drums sounds or sound effects
– The two "E" buttons launch a sample from its Zone directly into Logic's sample editor where you can graphically edit loop and sample start and end points as well as take advantage of the sample editor's powerful DSP functions.



This 9 1/2 Octave virtual keyboard allows you to trigger notes while you set up the zones and groups for the instrument. The key range for each Zone is graphically represented beneath the keyboard.

Groups offer several parameters for controlling multiple Zones simultaneously.
– Define the maximum number of voices – ideal for creating hihat mute groups or for voice limiting
– Volume and pan controls
– EXS Output lets you assign the output destination of an entire group (i.e. toms from a drum kit)
– Velocity range for dynamically switching samples
– High and Low key range
– Offset the ADSR of the group against the global ADSR in the plug-in window
– Offset the Cutoff and Resonance for each group ±50% from the main cutoff and resonance controls in the plug-in window.

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EXS-24 MKII

The Plug-in window gives you access to all of the sampler's synthesizer parameters such as filters, LFOs, envelopes and more.

The pitch of the Sample Instrument can be tuned ±2 octaves in semitone increments. Pitchbend has a selectable range of ±36 semitones. Glide and Pitcher sliders allow you to control portamento time. Random detune simulates the tuning drift of analog synthesizers.

Choose between Legato, Mono and Poly modes (up to 64 voices). Unison mode plays multiple voices when each key is triggered: In Poly mode, 2 voices per note; In Mono or Legato mode, you can adjust the number of voices per note up to 8.

Easily select and load an Instrument from the Sampler Instruments sub-folder using the pulldown menu. The + and - keys allow you to scroll through adjacent Sample Instruments.

The Multimode Filter features a cutoff frequency, resonance, drive (distortion) and key follow controls
A choice of three filter types: 12dB/oct High Pass and Band Pass filters as well a Low Pass filter with four steepness curves — 12dB/oct, 18dB/oct, 24dB/oct classic as When using the Low Pass Filter, a Fat switch helps preserve strong bass even with higher resonance settings. The filter section can be turned off to conserve polyphony.

"Hold" lets you define the modulation source assigned to control sustain "Xfade" allows you to crossfade between layered sample Zones with adjacent "Select Range" settings.



Control the global output level and set minimum and maximum velocities to dynamically controlling sound levels.

The Modulation Matrix supports up to 10 modulation routing configurations each with an independent source and destination. A fader lets you set the modulation depth with a positive or negative value. A second source can be added allowing you to manipulate the strength of the original source via an external controller (i.e. key velocity can be used to control LFO speed which in turn is controlling the filter's cutoff). Individual routings can be bypassed.

The Amplitude Envelope features control over attack, decay, sustain and release – the attack parameter is controllable via velocity

Three LFOs can each be assigned as a mod source in the Modulation Matrix. The Rate for each LFO can oscillate freely between 0 and 35 Hz, or can be tempo synchronized in values between 32 bars and 1/128 triplets. LFO one and two provide a choice of Triangle, rising and falling Sawtooth, Square up and Square down, a random stepped and smoothed random waveforms. LFO three always uses a triangular waveform. LFO one is a polyphonic LFO with key sync. This means that it is triggered individually for each note. An envelope generator lets you control the decay or delay of LFO one. LFO two and three are monophonic and runs continuously.

A dedicated ADSR envelope is provided for controlling the filter's cutoff frequency – the attack parameter is controllable via velocity

The decay time of the filter and amplitude envelopes can be shortened for higher notes to emulate the decay time response of acoustic instruments.
A convex or concave response curve can be applied to the filter and amplitude envelopes attack slope

EXSP24 – Emagic Xtreme Sample Player 24 Sample player for VST 2.0



The EXSP24 sample player runs on host applications capable of supporting VST 2.0 Instruments. It includes the intuitive user interface and many of the features which made the Emagic Xtreme Sampler 24 Bit (EXS24) so successful. As with the EXS24, the EXSP24 can play samples ranging from 8 to 24 Bit depths, any sample rate from 11 to 96kHz, in any combination. Sample import facilities will include support for WAV, AIFF, SDII and SoundFont2 across platforms. Parameter changes, such as enveloping, filtering and modulation can be saved as a preset. Samples will automatically load as part of saved song files. To edit samples and instruments, Logic Audio and an EXS24 are required.

LOGIC CONTROL • LOGIC CONTROL XT

Realtime Control Solution For The Entire Logic 5 Series



Logic Control is an ergonomic, expandable remote control surface, co-developed by Emagic and Mackie Designs, that provides the most comprehensive tactile control support available for the Logic 5 Series line of music production workstations. Logic Control offers realtime control over hundreds of editing, mixing and automation functions for MIDI and audio tracks, plug-ins and software instruments. Eight full-featured channel strips with 1024 step, ultra-fast motorized, touch-sensitive faders; infinitely variable Rotary V-Pots and dedicated channel function keys have direct access to any of Logic's channels using the onboard Bank Switching functions. Other features include dedicated transport controls, a Jog/Scrub wheel, navigational cursor keys, a myriad of automation and function buttons and a large backlit LCD display that all work together to ensure seamless interaction between Logic software and Logic Control and you. The Logic Control system can be extended to incorporate an unlimited number of physical channels by adding multiple 8 channel Logic Control XT expansion modules.

FEATURES

- Bi-directional communication – every change in Logic Control is reflected in Logic and vice-versa
- Convenient hands-on control of Logic 5's 32-bit track-based automation
- Total recall of faders and displays with song loading
- 100mm Penny & Giles ultra-fast 1024 step motorized faders (8 channel and 1 master) follow value changes immediately
- Each of the 8 channels also features an infinite stepped rotary V-Pot encoder with circular LED graphs and push button; Solo, Mute, Record and track Select buttons; and a signal present LED for audio or MIDI
- Large transport controls with status LEDs as well as a Scrub/Jog wheel and Scrub key with status LED
- 41 function keys with 23 status LEDs plus 8 additional, freely assignable function keys
- 2 x 55 character multifunctional backlit LCD displays parameter changes in realtime – Level meters for all channels can be activated, vertical or horizontal including overload display
- Control parameters such as volumes, EQs, MIDI, busses, plug-ins, slot selection, sends, Screensets and more
- Select, launch and edit sends, EQs, plugins and audio instruments (Emagic's as well as third party), without stopping the sequence

Fundamental Operation Modes

- Choose between Track View and Global View to decide whether the channel order in Logic Control reflects the track order in the Arrange window or, if Logic Control's display will reflect a specific type of channel (MIDI tracks, audio tracks, instruments, aux, busses, ins and outs)
- There are two additional modes selectable from within the Track and Global Views: In Multi Channel View, a specific parameter, such as panning, is displayed for all tracks. This parameter will be shown in the LCD and can be adjusted via the V-Pot. In Channel Strip View mode you can control a complete function group, such as EQ, for a selected channel.
- Each operating mode is immediately accessible with a single key stroke.
- Flip mode exchanges the functions of the motorized faders with the V-Pots allowing you to, for example, control EQ gain using the motorized faders.

Customizable

- Faders, knobs and switches are all easily assignable to any Logic function accessible as a Key Command using standard drag and drop methods within Logic's elegant Controller Assignment window

The Rear Panel

- MIDI in and out provides communication between Logic Control and Logic with true plug & play installation and allows future firmware updates
- Two assignable foot switch inputs can control, for example, Start/Stop and Punch In/Out and one assignable controller input for use as a volume or wah pedal
- International (100-250V) external power supply for standard power cords

Construction

- High-quality, durable components including a sturdy 1mm steel chassis and case

Third Party Compatibility

- The Logic Control System can control every OMS-compatible Mac OS 9 audio and MIDI application, that offers support of the Mackie Control or the Mackie HUI including Pro Tools (including Pro Tools LE and Pro Tools Free), Digital Performer and Nuendo. This is ideal if you need to switch between different host programs, and want a dedicated control surface at the heart of all of them.

LOGIC CONTROL

Large 55 x 2-digit multifunction backlit LCD display shows detailed information on all selected parameter names and values; track names and the contents of dialog boxes and can even be used for level metering – vertically, or horizontally with peak hold and overload indicator.

Six buttons, accompanied by a 2-digit, LED display allows you to select which parameter groups the Channel Strips are going to control – Track, Pan/Surround, EQ, Send, Plug-In, Instrument

10-digit, LED display, switchable between SMPTE or bar/beats/sub division/ticks, for song position information

Eight freely definable user keys

These eight buttons allow direct selection of specific mixer tracks such as audio tracks, MIDI tracks, inputs, busses etc.

*The **Modifier** buttons duplicate the Shift, Option, Control and Command keys of your computer keyboard*
*The **Automation** buttons activate the Read, Write, Touch, Latch, Trim and Group modes*
*Four **Utility** buttons provide direct access to saving your song, Undo as well as cancel or confirm in dialogs.*

Marker, Nudge, Cycle, Drop, Click and Solo buttons extend the functionality of the basic transport controls

Five Transport buttons with status LED for Forward, Rewind, Stop, Play, Record

The Jog/Scrub wheel provides precise location to any song position as well as audio scrubbing

V-Pot: A digital endless rotary knob with LED position indicators and push button allows parameter adjustments of pan, EQ, send levels, etc.

A signal present LED that shows if an audio signal is present

Four buttons with integrated colored LEDs control channel functions such as: Record, Solo, Mute and Channel Selection

The eight Channel Faders and the Master Fader are motorized 100mm touch-sensitive Penny & Giles faders with a 10-Bit (1024 step) resolution

Comfortable, durable wrist rest



The six buttons just above the master fader allow you to shift the mixer left or right one channel at time or in 8 channel banks; Flip the assignment if the channel fader and V-Pots and toggle the track view to access global view parameters.

Four cursor keys (up/down/left/right) allow you to quick navigate through plug-in slots and parameter pages. The Zoom button to switches the Navigation buttons to zoom controls



Logic Control XT –
Extension Unit For Logic Control

- Adds eight complete physical channel strips and an LCD to your Logic Control system using the same high-quality key components as the Logic Control
- Practically unlimited number can be added to your system allowing greater customization and extending the number of simultaneously adjustable parameters.
- Each Logic Control/XT combination forms a Control Surface Group (CSG). Each CSG can control different parameters; e.g. one group controls volume/pan and the second one controls an audio instrument.

Why MIDI and not Ethernet?

Each Logic Control or XT unit requires a discrete MIDI in and out connection. MIDI, as a protocol, was designed for realtime performances. It has, for many years now, proven to be accurate enough for that most timing-sensitive of MIDI events – note information. By comparison, the timing of controller events is far less critical. To move a fader to a target value, an amount of data equivalent to a single note event is all that is required. In conjunction with the fast Penny & Giles motorized fader units, there is more than enough head-room in MIDI's bandwidth to deal with the most complex mixes. MIDI, in comparison with Ethernet, provides simple connectivity and proven, stable low-latency performance.

EMAGIC

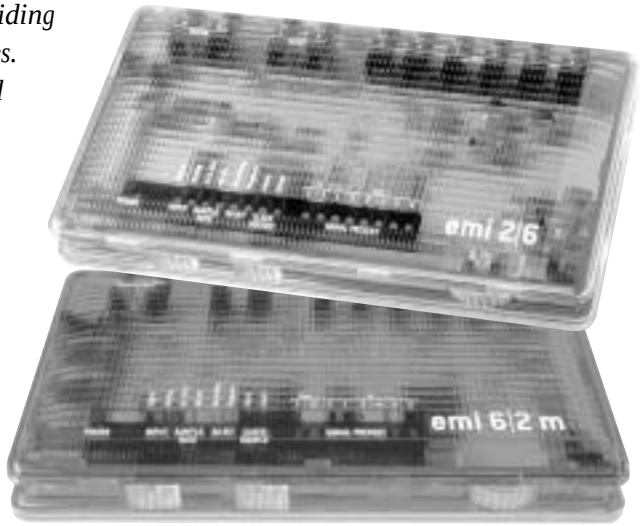
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COMPUTER AUDIO

960

24-bit / 96kHz Multi-channel USB Audio Interfaces

The EMI 2I6 and EMI 6|2 m are portable USB audio solutions providing professional 24 Bit audio quality at 44.1, 48 and 96 kHz sample rates. The EMI 2I6 features two analog inputs, six analog outputs and and S/PDIF I/O while the provides EMI 6|2 m six analog recording and two analog playback channels and two cinch sockets, which can be used as either S/PDIF digital I/O, or as MIDI in and outputs. Both units are extremely lightweight and compact — approximately the size of video cassette, and are powered directly via your computer's USB port — ideal for mobile recording with notebook computers. Both units also employ Emagic's low latency technology which guarantee monitoring of audio input signals with no perceptible delay as well as realtime performance of software synths. Both interfaces are compatible with Core Audio under Mac OS X as well as SoundManager and ASIO using Mac OS 9.1 and higher.



FEATURES

They Both Feature

- 24 bit A-to-D and D-to A converters
- Support is provided for 16 or 24 Bit audio resolutions at 44.1, 48 and 96 kHz sampling rates

EMI 2|6 Inputs and Outputs

- The EMI-2|6 features two analog inputs, six analog outputs using unbalanced RCA (phono) connectors – compatible with -10 dBV and +4 dBu audio levels
- Coaxial stereo S/PDIF digital I/O is also provided
- Also includes a stereo headphone jack, with independent volume control

EMI 6|2m Inputs and Outputs

- The EMI-6|2m features six analog inputs with and two analog outputs using unbalanced RCA (phono) connectors – compatible with -10 dBV and +4 dBu audio levels
- 2 cinch sockets can be used as either a coaxial S/PDIF digital I/O or as a 16 channel MIDI In/Out port

Visual Monitoring

- Status LEDs for all interface modes
- Signal presence LEDs for all inputs and outputs

Zero Latency Monitoring

- Latency-free hardware monitoring allows you to monitor the audio recording channels with no audible delay. The stereo monitor signal can be routed to the analog, digital or integrated headphone outputs.

Ideal for Virtual Instruments

- Emagic's advanced Low Latency technology ensures that software-based virtual instruments such as Emagic´s EVP88, EXS24, ES1 and VST2.0-compatible instruments can be played in real-time.

The USB Advantage

- The unit is automatically recognized by the computer, making installation simple and painless
- Because power is supplied via USB there is no need for additional cabling (an optional power supply is also available from emagic)
- Integrated USB HUB provides two additional USB ports
- Doesn't require any free PCI or other card slots – no need to open up your computer
- Hot plug capability means that the interface be connected and disconnected without turning off the computer

The EMI 2|6 – An Ideal Surround Sound Solution

The EMI 2I6 provides an affordable Surround mixing hardware solution via suitable audio software, such as Logic Audio Platinum. All Surround formats up to, and including 5.1, are supported. This makes the EMI 2I6 the ideal companion for the production of Surround sound audio mixes for DVD or film distribution.

System Requirements

Free USB port; CD-ROM or DVD drive

MACINTOSH:

Minimum: Mac OS 9.1

Recommended: Mac OS X 10.2.2 or higher
Compatible with Core Audio (Mac OS X), SoundManager, ASIO (Mac OS 9)

WINDOWS:

Minimum: Pentium or compatible with 233MHz and 64 MB of RAM under Windows 98 SE 3)

Recommended: Windows XP with Service Pack 1 (SP1)
Compatible with DirectSound, MME, ASIO and EASI applications

B&H PHOTO - VIDEO - PRO AUDIO

MT4 • AMT8 • UNITOR8 MkII

Multiport MIDI Interfaces For Windows and Mac



The MT4 is a portable USB-powered MIDI interface with two MIDI in and four MIDI out ports. The AMT8 and Unitor8 MkII are each 8 port, USB compatible MIDI interfaces. The Unitor8 MkII adds SMPTE I/O for synchronizing to external video and audio gear making it an ideal communications hub between your music computers, MIDI devices, tape and video machines and automated mixing devices. Both the MT8 and Unitor8 MkII benefit from Emagic's Active MIDI Transmission technology which provides a measurable, and more importantly audible improvement of MIDI timing. All three interfaces are networkable over USB and can be mixed and matched allowing up to 8 interfaces to be stacked together.

MT4 Features



- The ideal MIDI interface for USB-equipped laptop computers, or for smaller desktop installations.
- Two MIDI in ports and four MIDI out ports provide you with 32 input and 64 output MIDI channels.
- USB connection provides both superior MIDI timing and power. An LED confirms USB activity.
- Minimum size and weight
- Two slip-resistant rubber inserts bind the housing providing extra stability
- MIDI activity on each port is indicated by a dedicated LED.
- Software configurable Patch Mode allows the MT4 to be used as a MIDI patch bays with up to 32 patches that can be saved and recalled.
- Directly supported by Logic Audio and MicroLogic AV Version 4.01 and up and by SoundDiver 2.07 upwards
- Supplied drivers provide compatibility with OMS-compatible software

Active MIDI Transmission

A segment of the sequencing engine of AMT compatible software, (e.g. Logic Audio), is put into the MIDI interface, (AMT8 or Unitor8 MkII), where it takes care of sending the MIDI events to the individual outputs with the highest possible timing accuracy. At the same time, the transfer of data from the computer to the interface is optimized with extreme efficiency by sending notes in parcels during the pauses between the musical events. These parcels are then unpacked in the interface and sent to the individual outputs at precisely the required time.

AMT8 Features



- 8 MIDI In and outs with Activity LEDs
- Networkable (up to 8 units)
- Mac compatible (AMT, MTP emulation and OMS)
- Windows 95/98, Windows NT 4.0, Windows 2000 and Windows XP compatible (AMT, MME)
- MME compatible multi-client MIDI driver (multiple midi programs can use different MIDI ports simultaneously)
- Stand Alone Operation with 32 Patches
- Panic/Patch button
- USB connection (PC/Mac) as well as RS-422 for Macintosh and RS-232 for PC

Unitor8 MkII Adds–

- LTC Generator and Reader with 1/4″ I/O jacks and read/write status LEDs
- LTC Sync at 12-1000% original tape speed - forward and rewind
- VITC Generator and Reader with S-Video connectors and read/write status LEDs
- VITC-Sync during single frame advance forward/back
- Timecode Video Burn-In - also during LTC Sync
- Timecode Refresh function, adjustable Freewheeling, Jam Sync function, Adjustable MTC Full Frame Message
- Click input allows an audio trigger signal or foot switch to be the synchronization source
- Operating system can be updated via SysEx

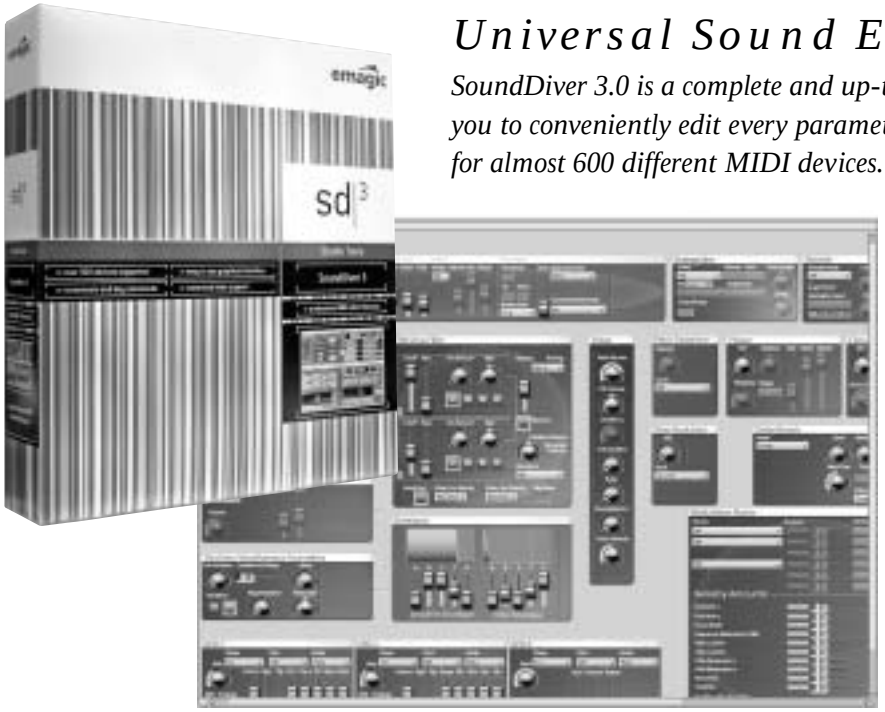
Features	MT4	AMT8	Unitor8
MIDI Input Ports	2	8	8
MIDI Output Ports	4	8	8
MIDI activity LEDs	•	•	•
USB Power	•		
Patch Mode	•	•	•
USB port	•	•	•
RS-232 port (Win)		•	•
RS-422 port (Mac)		•	•
AMT Compatible		•	•
Panic button		•	•
Click input			•
SMPTE I/O			•
OS update via MIDI			•

EMAGIC

SOUND DIVER 3

COMPUTER AUDIO

962



Universal Sound Editor

SoundDiver 3.0 is a complete and up-to-date MIDI Editor/Librarian that allows you to conveniently edit every parameter, and catalog, store and organize the patches for almost 600 different MIDI devices. SoundDiver's slick user interface features graphic representations of all editing parameters with switchable editing depths as well as photorealistic representations of your MIDI devices.

A user definable Key and/or remote MIDI command is available for almost every SoundDiver function. AutoLink offers advanced communication with Logic Audio that allows you to display proper patch names within in Logic as well as synchronized the switching of screensets. External hardware control surface support allows you to edit and control devices remotely via MIDI.

- Includes editors and librarians for almost 600 MIDI devices
- Once launched, SoundDiver automatically scans and identifies your devices and displays them onscreen, along with their parameters.
- The attractive 3-D look with photorealistic device icons and clearly laid-out user-interface allows you to easily edit every aspect of your setup with just a few mouse clicks.
- Access all the potential of your synthesizers and MIDI devices.
- Virtually unlimited undo and redo steps
- Create randomly generated sounds, copy or paste complete function blocks or all adjustments of a complex envelope.
- Freely definable key and MIDI commands and screensets help to speed up your work allowing you to be more productive and create sounds faster and easier.
- Refined help system offers context sensitive help and ToolTips.

Sound Management

- SoundDiver creates collections of SysEx data called Libraries, can contain programs, sounds, combinations, multi's, effects or other data independently of how they are managed in the respective hardware devices. This simplifies the task of sound management by allowing you to organize data in using a consist methodology within the software
- Libraries can contain compilations of sounds from different devices or can be specific to each.
- All data can be sorted freely or automatically by name, module, data type, date or comment.
- The library functions allow you to quickly find the right sounds; for example, by searching a specific sound category such as "pads" within your entire system. A new library can then be created from this result, with all the dependencies between source libraries intelligently managed.

Hardware Controllers

- Remotely control every MIDI device in your studio using a central hardware control surface such as the Logic Control.
- Free Controller Assignments currently allow virtually every parameter of every editor to be controlled by any MIDI message – an ideal solution for editing, or accessing the live performance potential, of your MIDI instruments and effects devices such as delays, filter banks, reverbs.
- MIDI controllers with the appropriate support will display the current parameter assignments. If equipped with motorized faders, they will even follow every move you make on screen.

AutoLink

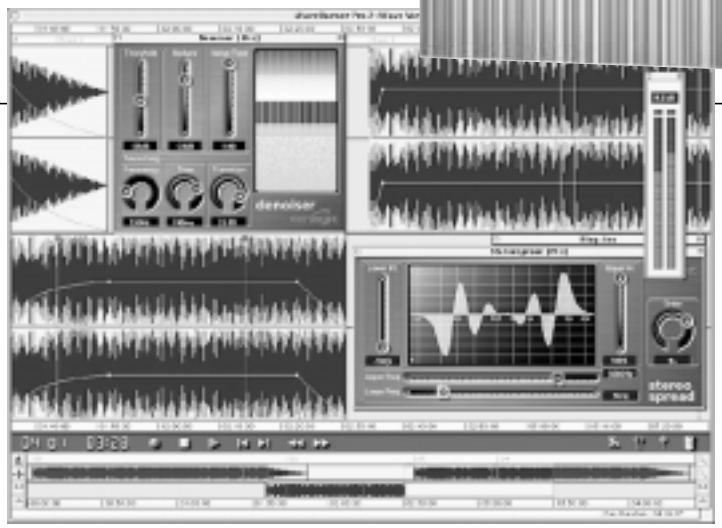
- AutoLink is an active connection between Logic and SoundDiver that allows you, for example, to display patches by name rather than as a list of program numbers.
- Opening a song transfers all included sounds and patches to your MIDI devices.
- Edits and parameter changes made in SoundDiver can be recorded in Logic.
- Screenset switching is synchronized between Logic and SoundDiver, making the two applications behave like a single program with all the inherent advantages.

WAVEBURNER PRO

Professional Red Book Audio-CD Mastering and Burning Software



WaveBurner Pro is Emagic's professional, award winning Red Book audio-CD mastering and burning software for the Mac OS. WaveBurner Pro allows you to quickly and intuitively assemble a playlist of tracks using non-destructive editing of regions and crossfades, plus the creation of break, track and index markers. WaveBurner Pro's comprehensive plug-in support, audio material can be further refined and processed in WaveBurner Pro. A host of mastering plug-ins from the Logic Series is included for applying to individual regions or to the final mix. Third Party VST plug-ins are also supported. Waveburner Pro is compatible with a wide range of internal ATAPI and external FireWire and SCSI CD burners with support for write speeds up to 32x.



FEATURES

- Create Red-Book-compatible CDs with support for full PQ editing from a variety of audio file formats
- CDTEXT support
- Supports Copy Prohibit, Pre-Emphasis, ISRC number and UPC/EAN code

Recording Function

- All processing is done at an internal 32-bit floating point resolution until the final mastering stage where the signal is converted to the 16-bit audio CD standard using the acclaimed 'POW-r' (Psychoacoustically Optimized Wordlength Reduction), dithering algorithm which uniquely reduces word length without adding noise or tonal coloration developed by the POW-R Consortium LLC
- Real-time conversion to 44.1kHz from high definition audio files up to 24 Bit/96 kHz

Editing

- Non-destructive editing within the Waveform Window of Tracks, Pauses, Indices, Level, Fades and Crossfades
- Create multiple tracks from one audio file or a single track from multiple audio files
- Crossfade support, including S-shape curves, allows you to make silky smooth transitions between audio regions
- Audio in track pauses
- First track number larger than 1 possible

Integrated Mastering Plug-Ins

- A suite of precision mastering plug-ins are provided for taking on critical mastering tasks. These include:
  - Fat EQ
  - Compressor/limiter
  - Multi-band compressor
  - StereoSpread
- A number of audio restoration tools are also provided for removing undesirable noise artifacts from cassette or vinyl recordings without destroying the top or bottom end of an audio file
- A virtually unlimited number of plug-ins can be inserted at one time, the total number depends on CPU performance
- Compatible with third party VST plug-ins

Supported Audio Hardware

- Audiowerk2 or Audiowerk8 SoundManager, ASIO, Direct I/O

Supported Data Formats

- Direct loading of WAV, AIFF, SDII (also split stereo) files is supported
- MP3 file import and conversion to CD audio is also supported
- Disc image export/import
- Export the Tracklist as text file

System Requirements

- PPC 604e/200 MHz, Mac OS 8.6 or higher, 96 MB RAM, QuickTime 4.x for MP3
- Supports all internal CD burner drives from Apple, and all SCSI or FireWire devices which support the Multi Media Command Standard (MMC or SCSI-3)
- Support is also available for many legacy SCSI CD recorders (independent of MMC)

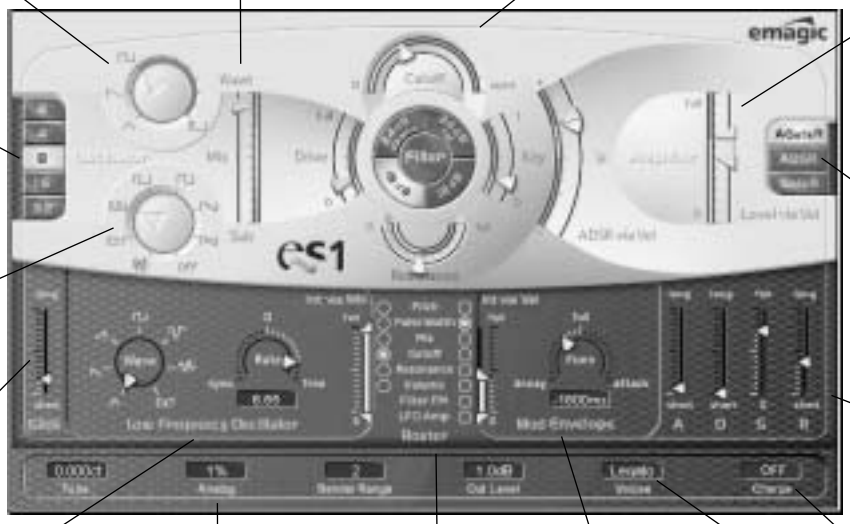
EMAGIC

ES-1 EMAGIC SYNTHESIZER ONE

Analog Modeling Synthesizer for Logic

The ES1 was Emagic's first native virtual instrument and since it's release, in 1999, it has become the staple analog modeling synthesizer for Logic users seeking the classic sounds of vintage subtractive synthesizers. Emagic's Synthesizer One features all of the essential elements that have given analog synthesizers such a lasting presence in modern music production – rich sounding oscillators, fat filters, flexible modulation possibilities and extremely fast envelopes. Playback timing for the ES1, as with all other virtual synths, is sample-accurate and the 32-bit sound engine ensures exceptional sound quality. The ES1 integrates seamlessly into Logic's digital mixing environment where it can be further processed by adding plug-ins. And what's more, every parameter can be fully automated. Includes dozens of preset instruments from bone rattling basses to lush pads and from scorching leads to exotic effects and percussion – all easily accessible using Logic's hierarchal menu system.





Each instrument uses a single oscillator with a choice of Triangle, Sawtooth and Pulse waveforms.

The main and sub oscillators are blended before being fed to the filter section

Low Pass Filter has four selectable slopes (12dB/18dB/24dB classic/24dB fat) and features variable drive(distortion) cutoff, key follow and resonance up to self-oscillation. The cutoff can be controlled via the ADSR or via your keyboard

The main and sub oscillators are blended before being fed to the filter section

This double slider provides control over minimum and maximum velocity values

The octave range for the main oscillator is selectable from 2 to 32 feet.

***AGateR** lets you control Attack via the ADSR and Release via note-off.*
***ADSR** controls volume via the ADSR*
***GateR** – same as above except that Attack is always zero*

The sub-oscillator follows the pitch of the main oscillator with a choice of 5 waveforms as well as noise. You can also assign an external audio source to the sub-oscillator

Classic ADSR envelope reacts with microsecond accuracy.

Glide controls the portamento control

Switchable chorus effect adds spaciousness and dimension to patches

The LFO's rate can be set to run freely or synchronized to your session's tempo. The LFO features a choice of Triangle, positive and negative Saw, Square, Sample and Hold and Smooth Random waveforms. You can also assign an external source as a modulator of the LFO.

The Analog parameter randomly adjusts the pitch and filters cutoff frequency for each note to replicate the character of classic polyphonic synths

The modulation router allows you to assign a target for the LFO and modulation envelope

The Mod Envelope allows you to assert velocity sensitive decay or attack attributes to a wide range of targets like filter cutoff, oscillator mix, pitch and more

Choose up to 16 note polyphony for each instrument plus legato mode

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MC
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AMERICAN EXPRESS Card

ES-2

Emagic Synthesizer 2 – Advanced Hybrid Modeling Synthesizer for Logic

The ES2 is a unique and powerful software instrument for Logic 5 users, that combines the warmth of subtractive synthesis found in classic analog synths and digital synthesis techniques such as Vector Synthesis, Wavetable Scanning and Frequency Modulation (FM) with a myriad of in-demand synthesis techniques. Each ES2 can be assigned up to 32 voices, each offering three especially flexible oscillators, two filters, extensive modulation possibilities. The Unison function allows you to create massive layered textures up to 32 voices deep. All of these features and more have been wrapped up in an innovative ergonomic user interface to provide you with a seemingly endless realm of sonic possibilities. Includes over 400 sounds created by renowned sound designers that will immediately unlock the awesome potential of the ES2 right out of the gate.

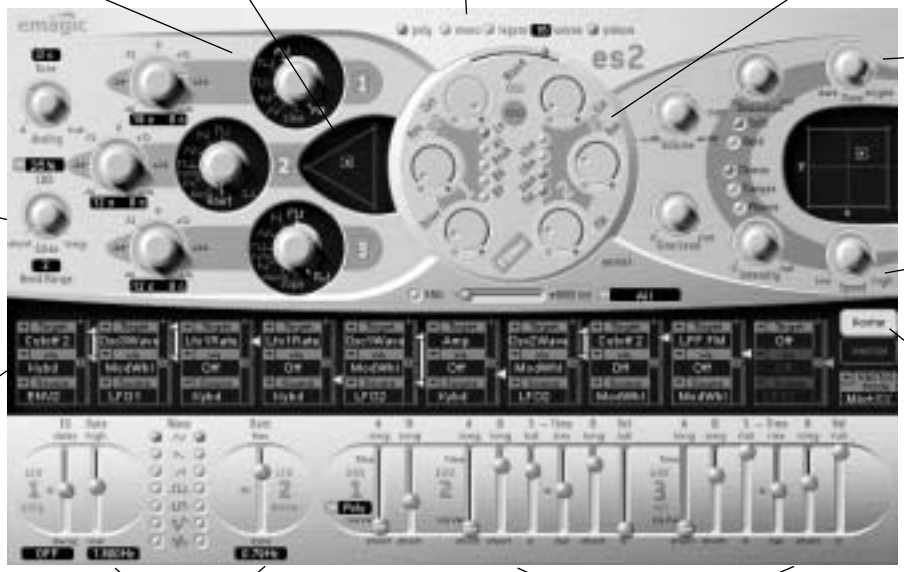


3 Oscillators *(Oscillator 2 and 3 can be synced to Oscillator 1) with a choice of classic analog waveforms (Triangle, Saw, etc.) and 100 digital waveforms as well as noise, FM and ring modulation*

Vector synthesis for dynamic oscillator mix and control of 2 further selectable parameters

Up to 32-note polyphony per ES2 instance (CPU dependent), selectable mono and legato modes as well as unison mode which allows you to layer all 32 notes into one huge patch

Two filters configurable to run in serial or parallel: *A Multimode Filter with a variable distortion circuit, (selectable between Band Pass, Band Reject, Peak, Hi Pass); and a Low Pass Low Pass Filter with selectable slope (12 / 18 / 24 dB) and a switchable fatness parameter for enhanced bass response, even at high resonance settings. Frequency modulation of the low pass filter via oscillator 1.A sine wave from oscillator 1 can be mixed with the filter output for increased low end.*



Built-in Distortion circuit switchable between hard and soft modes features its own tone control knob

Portamento control and pitch bend range

Built-in modulation effects (chorus, flanging and phasing) with independent Intensity and Speed controls, offers including.

10 freely assignable modulation paths, with independent control of modulation intensity, lets you route any of the 22 available modulation sources to any of the 34 available targets.

The Router display can be switched to graphically indicate LFO and envelope shapes or, when switched to "Vector", can be used for dynamic oscillator mix control and control of two freely selectable parameters over time.

Two independent LFOs, one monophonic LFO and one polyphonic LFO with envelope control each with selectable free or beat-synced modes and seven selectable waveforms

Three Envelopes – 2 ADSR envelopes with sustain time and 1 AD/AR envelope (selectable between monophonic or polyphonic) Attack times are controllable via velocity

EMAGIC

EVOC 20 – EMAGIC VOCODER 20 BAND

Professional Vocoders and Formant Filter Bank for Logic

The EVOC 20 package contains a suite of three software plug-ins that model a variety of analog vocoders that allow the sonic characteristic of the analysis signal to be printed to the synthesis signal. The first plug-in is the evoc20-PS. It uses a 16 voice, polyphonic synthesizer, playable in realtime over MIDI to emulate classic vocoder effects. The second plug-in, the evoc20-TO, uses a monophonic oscillator to track the pitch of the analysis (input) signal. Each plug-in supports up to 20 bands of filtering, formant stretch and shift high and low knobs that allow you to determine the part of the signal to effected. The The third unit is a formant filterbank that uses volume faders for controlling up to 20 filter bands available for the two filters. The elegant and intuitive graphical user interface of each plug-in begs to be tweaked and guarantees instant gratification.



EVOC20 PS - Polyphonic Synthesizer



- Combines a vocoder with a polyphonic synthesizer, playable in realtime via MIDI.
- Utilizes a 16 voice dual oscillator polyphonic synthesizer with 50 special waveforms, FM capabilities and colored noise whose signal can be articulated by any input signal, giving you classic vocoder sounds like talking robots, singing strings, percussive choirs and more.
- Up to 20 filter bands can be used for the articulation process
- The Resonance control adjusts the vocoder's sound characteristic
- Formants can be stretched and shifted or modulated by the syncable LFO
- The Unvoiced/Voiced Detector increases speech intelligibility by adjusting the reaction speed to sound changes in the analysis signal
- The Freeze button holds the latest sound characteristic of the analysis signal
- The Ensemble effect provides that familiar warm, swirling sound.
- Stereo spread control enhances the width of processed signals.

EVOC20 TO - Tracking Oscillator

- Similar architecture in most respects to the evoc20-PS (up to 20 filter bands, modulatable formants, controllable filter resonance for sharp or soft vocoder sounds, Freeze, adjustable reaction times and the U/V Detection circuit) except that the evoc20-TO uses a monophonic pitch tracking oscillator, instead of a polyphonic synthesizer to follow the pitch of the analysis signal
- The oscillator's waveform can be switched between sawtooth and 2 oscillator FM.
- Pitch tracking that can be limited to any musical scale
- Many surprising sounds can be attained when polyphonic input signals or drums are used as the input signal



EVOC20 FB - Filter Bank

- The input signal is run through two distinct, fully adjustable analog-style filter banks in parallel
- Up to 20 filter bands per filter can be blended manually or via a syncable LFO.
- Each discrete filter band features independent level controls with the option to dampen frequencies up to 100%, enabling precise and drastic sound sculpting.



- Filter resonance, adjustable filter slopes, and optional distortion provide an expansive range of tone colorings.
- A second LFO, syncable to song tempo, can be used to modulate the formants through the movement of filter bands. Formant movements can also be manually achieved.

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ONE HOUR FREE PARKING
AT 349 W. 34th STREET (with purchase of $100 or more)

EVP88 • EVP73

Emagic Vintage Piano 88



The EVP-88 is a virtual software instrument that digitally reproduces, the unique feel, warm tones and expressiveness of vintage Fender Rhodes, Wurlitzer and Hohner electric pianos with stunning accuracy. These highly sought after electric pianos include the Fender Rhodes Mark I of the Suitcase and Mark I & II of the Stage series, the Wurlitzer Electric Piano 200A and the Hohner Electra Piano, as well as many tasteful variations – there are 12 models in all. Emagic's native realtime tone generation ensures that each piano responds to every nuance of a player's performance with impressive realism. The EVP-88 also features a variety of authentic effects including EQ, Distortion, Phaser, Chorus and Tremolo. A group of realtime model parameters are also provided allowing you to adjust stretch tuning, control the volume of the bell and damper as well as control the decay and release time for the tines.

- Emagic's native tone generation captures the essence of 12 different vintage electro mechanical piano models
- Playable with full polyphony over a range of 88 notes, with 88 voices (CPU dependent).
- Integrates directly into Logic's digital mixer with sample accurate timing, access to all plug-in effects and automation with total recall.
- Tonal characteristics are easily modified using the variable Decay and Release EG, as well as volume controls for emphasizing Bell and Damper noise.
- Stretched Tuning Curve and Warmth controls add even more realism to the sound.
- Stereo spread can also be quickly modified.



Effects

- Two band (bass and treble) EQ controls based on Emagic's Fat EQ
- Drive (distortion) circuit with gain and tone controls allows you to add just the right amount of bite.
- The variable intensity Chorus adds a final high-quality shimmer
- The four step Phaser with color control and controls and the Tremolo with variable rate and intensity controls recreate classic effects that add movement to your mix. The Phaser and Tremolo also feature an adjustable stereophase control allowing you to add even greater width and depth to the effects.

EVP73 – Emagic Vintage Piano 73



- VST plug-in modeled after the Fender Rhodes Stage Piano Mk II electric piano
- Compatible with any host application capable of running VST 2.0 Instruments
- Provides seamless integration into the host application's mixer.
- 73-note polyphony and the same core engine, great sound and retro styling of the EVP88.
- The realtime native tone generation system allows harmonic interaction of generated notes in response to the users playing style.
- Includes the same decay and release envelope generator, bell and damper controls, stereo width and tremolo controls as the EVP-88

EVB3

Emagic Vintage B3

The EVB3 is an optional software instrument for Logic 5 that not only emulates the sound of the legendary Hammond with stunning authenticity, using Emagic's Component Modeling Technology, but improves upon it by offering an even broader range of musical and creative possibilities. Every nuance of the original, from the tone wheels and the drawbars to the range of Leslie speaker simulations, and other effects, have been faithfully captured and are easily controllable with a graphical user interface that replicates the layout of the original B3. Whether you're a jazz, gospel or rock organist, or a remixer/producer, you will find the responsiveness and seemingly limitless sonic palette of the EVB3 an indispensible component in your virtual instrument arsenal.



FEATURES

General

- 32 Bit Engine
- Up to 195 Voices of polyphony per EVB3 - (CPU dependent)
- Sample-accurate playback timing
- Fully automatable and supported by Logic Control

Tone Generation

- The very best organs and Leslie speaker cabinets were analyzed to provide Logic 5 users an authentic, realtime emulation of the electro mechanical tone generation of the Hammond B3 Tonewheel Organ including control over key click noises and tone wheel crosstalk attributed to the artifacts and colorations inherent in aged components. Other events such as the loudness robbing and tapering which typified the B3 can also be easily simulated.
- Each tone wheel is synthesized by Emagic's Component Modeling Technology.
- Rotor cabinet simulation allows you to choose between different cabinet types and horn combinations
- The virtual microphone for the horn can be freely placed at any distance or angle.

Intuitive User Interface

- Laid out like the original Hammond B3 organ, with two (Upper and Lower) keyboard manuals and a Bass pedal board.
- Both keyboards provide instant access to different drawbar combinations via 24 preset buttons, available to each manual.
- An additional octave above and below the five octaves of the original, allowing screaming highs and really fat bass sounds
- High-resolution drawbar control system provides over 250 million possible combinations
- The Upper and Lower Manual feature two sets of 9 drawbars each with 8 values.
- The Bass Pedal features 2 drawbars each with 8 values
- Percussion Decay, volume, upper manual level
- Envelope for bass pedals, upper and lower manuals
- You can morph between presets via MIDI
- All parameter modifications can be stored as Settings.

Effects

- Leslie Cabinet emulation with Chorale, Brake and Tremolo modes. Can be modulated via aftertouch, sustain pedal or modulation wheel
- 6 Types of Scanned Vibrato with adjustable Rate and Chorus controls.
- 3 band EQ based on Logic's Fat EQ and Reverb
- A Wah Wah effect with 6 classic filter types can be controlled via any MIDI controller. Even the peak can easily controlled via foot controller
- Realistic tube distortion with tone warp controls and 3 distortion types
- It is also possible to determine the order of the effects in the signal path



Effects Plug-ins

- The Leslie, Vibrato and Distortion effects are also included as separate plug-ins allowing audio tracks to be run through the effects in any order



EVD6

Emagic Virtual D6 Clavinet



The EVD6 is a software instrument for Logic 5, which uses Emagic's component modeling technology, powered by a 32-bit floating point processing engine, to flawlessly recreate the classic sounds of the ultra-funky Hohner D6 clavinet. The EVD6 sports a stylish retro-look and an intuitive graphic user interface that is packed with several enhancements that deliver sound creation and modulation possibilities that far exceed those available to the original stringed instruments. Each instance of the EVD6 provides up to 73 note polyphony depending on CPU resources. And like the other Emagic software instruments available to the Logic 5 series, the EVD6 provides sample-accurate playback timing; can be fully automated and is supported by Logic Control.

FEATURES

Tone Generation

- Authentic, realtime emulation of multiple clavinet models including the Hohner D6 with its characteristic overstrung design as well as a number of exotic stringed instrument sounds are represented in the included presets
- Component modeling delivers a flawless simulation of the electro mechanical mechanism of the original instruments.
- Key Click on key down/up with Intensity, Random and Velocity parameters emulates key strike and release noises caused by the aging of the strikers in the original
- As with the original D6, the combination of pick-ups can be freely selected from 4 configurations. The position of the pick-ups can also be changed in realtime or automated to create phaser-like sounds.
- Rocker switches are provided for the Filter section with Brilliant, Treble, Medium and Soft settings. An enhancement of the EVD6, not available on the original D6 is the ability to turn the filter section off.
- Sounds can be easily modified via high-resolution controls that allow precise adjustment of string Decay, Release, Damping, Tension Modulation, Stiffness, Inharmonicity and Pitch Fall
- Discrete Exciter with Shape and Brilliance controls
- Stereo Spread control for Pickup or Key
- Damper wheel control
- All parameter modifications can be stored as Settings.

Three Foot Pedal Effects

- Three integrated foot pedal effects include; Phaser with adjustable Rate and Intensity controls, Distortion with adjustable Tone and Gain controls, as well as a Wah effect with 6 classic vintage Wah Wah filter models
- The routing order of the effects in the signal path is freely configurable
- The Wah effect has adjustable Width and Depth controls for creating autowah effects or you can control the wah effect's cutoff frequency via a MIDI footpedal (or any other MIDI controller).

Wah Effect Plug-in

- A Wah Wah / distortion effect is also available as an independent plug-in that can be applied to other sound sources in Logic 5.

Emagic's Software Synthesizer Technology

- Emagic's broad range of software instruments were developed to take full advantage of the capabilities of native sound synthesis and processing.
- Each instrument delivers outstanding sonic quality with minimal latency and ergonomically designed user interfaces.
- Each software instrument integrates seamlessly within Logic 's internal digital mixer, where they can access additional plug-in effects as well as the 32-bit automation engine.
- The direct connection of software instruments within Logic's sequencing engine guarantees unsurpassed precision with sample-accurate timing that is superior to any external MIDI synthesizer.
- Another advantage of software instruments is that all effects, mix settings, instrument patches and edits are saved with your song file where and are instantly recalled with your song.
- Software instrument parameters can be easily controlled using external control surfaces including Emagic's Logic Control.

EMAGIC

SOFTWARE BUNDLES



Logic Audio Big Box

The Logic Audio Big Box includes Logic Audio 5 audio and MIDI production software, the critically acclaimed EVP73 Emagic Vintage Piano, the astoundingly fat ES1 Emagic Synthesizer, plus the super versatile, ultra low-latency EXSP24 Sample Player, which includes over 600 MB of high-quality samples ready to play. To sweeten the pot, the Big Box also comes with the in-demand Xtreme Analog sample CD containing over 200 fat sounds sampled from classic analog synths. With five individual world-beating Emagic products in one box, this jackpot is no gamble.

EMI 6|2 m Production Kit: The All-In-One Recording Solution



The EMI 6|2 m Production Kit is a bundle of six complete Emagic products in one package: the EMI 6|2 m USB/Hub/MIDI interface; the latest version of the award winning Logic Gold 5 recording software: and the Emagic instruments EVB3, EVP88 and the EXSP24 bundled with the Xtreme Digital Sample Library. The EMI 6|2 m is a USB interface that provides six 24 Bit analog inputs and two analog outputs, at sample rates up to 96 kHz, MIDI I/O, a stereo headphone jack and stereo S/PDIF digital I/O, with the latter doubling for MIDI I/O duties. The Logic Gold 5 professional music production software delivers up to 64 tracks of 24bit/96 kHz audio, 32 bit floating point internal processing, unlimited MIDI tracks and 34 integrated plug-ins with support for third party Audio Unit (Mac OS X) and VST 2.0 (Mac OS 9) plug-ins. The EVB3 is a software-based B3 that uncannily captures the essence of the original with unmatched realism. The EVP88 captures the sound of the legendary Fender Rhodes, Wurlitzer and Hohner electric pianos. The EXSP24 sample player is also included along with the Xtreme Digital sample collection.



Emagic Vintage Collection: Analog Legends for The Digital World

The Emagic Vintage Collection includes the successful. multiple award-winning EVP88, the EVD6 and the EVB3, all for a special price. With the Emagic Vintage Collection, the vintage myths of the Fender Rhodes, Wurlitzer and Hohner electric pianos, Hohner Clavinet and the legendary B3 comes to life in today's computers. Uniquely, these vintage software instruments do not just statically recreate an instrument's tonal characteristics, as would be the case with samples. Rather, they expressively react to the player's every nuance with impressive realism. The use of the Emagic Vintage Collection requires an installed copy of Logic Audio 5.3 (or higher) for Macintosh.

Emagic Synth Collection: A Fine Selection of Synthesis Techniques



The Emagic Synth Collection includes three multiple award-winning software synthesizers for Logic Users: the ES2, the EVOC20 and the ES1 — all in one premium, specially priced package. The ES2 is a brilliant and powerful sounding soft synth with a unique ergonomic user interface that draws from the past to redefine the future. Its combines an extraordinary synthesis engine that delivers exceptional tonal quality in all registers with a complex voice architecture that provides extensive sonic possibilities. Able to combine the full-bodied energy of analog systems with the shimmering detail of glistening digital tone generation, the ES2 owes its enormous sonic wealth to an unsurpassed palette of in-demand synthesis techniques. The EVOC20 package provides Logic users with three powerful tools for sound design: A classic polyphonic vocoder with built-in synthesizer, a formant filter bank, and a pitch tracking vocoder. Each plug-in provides maximum flexibility and features pristine 32 Bit sound quality. As a real synthesizer, the ES1 offers extremely flexible tone generation that puts the entire palette of analog sounds at your disposal: earth-shaking basses, rich pads and textures, screaming leads, ultra-sharp percussion and exotic effects.

SOFTWARE BUNDLES



TDM Collection

The TDM Collection combines three sought-after Emagic software products in one specially priced package: ESB TDM, Host TDM Enabler and PTHD Extension. Emagic's **ProTools HD Extension** (PTHD | Ext) for Logic Platinum (Mac) provides compatibility with Digidesign's ProTools HD system including the latest TDM II architecture, with support for sampling rates up to 192 kHz. The **Emagic System Bridge TDM** (ESB TDM) is the ideal link between the DSP world of the TDM system and the computer's native CPU processing. The ESB TDM allows you to incorporate native audio tracks, Logic's and VST plug-ins and Audio Instruments into Logic's TDM mixer. The ESB TDM also allows you to use the EXS24mkII directly in the Aux channels of Logic Audio Platinum's TDM mixer. The **Host TDM Enabler** allows you to insert the Emagic software instruments ES1, ES2, EVP88, EVB3 and EVD6 – available separately – into aux channels of the TDM mixer, including TDM-compatible hosts other than Logic Platinum, where their audio signals can be further processed with any of the TDM plug-ins.The TDM Collection requires an installed copy of Logic Platinum 5.5 (or higher) for Mac OS 9, apart from the Host TDM Enabler, which can also be used with other TDM-compatible host software.

EXS24 AND EXSP24 FORMAT CD ROMS

Xtreme Analog



The Xtreme Analog CD-ROM contains a vast array of expertly compiled multisampled analog synthesizer waveforms, many of which were generated in stacked or unison mode for maximum impact. These fat, rich analog oscillator waveforms and layers are used as the initial sound source and then shaped using the EXS24mkII's extensive range of powerful synthesis capabilities including PWM, Sync, FM, cross modulation, filters, envelopes and LFOs. The user-friendly interface of the EXS24 positively invites the intuitive editing of sounds with both individual waveforms and complete Instruments, further enhancing the creative potential of the Xtreme Analog sound library. The velocity-controllable sample start point modulation function of the EXS has been utilized to create a number of dramatic and extremely vivid sound structures. And for the traditionalists, you can check out the exceptionally authentic core sounds of several legendary synthesizers, including the Jupiter 8, Matrix 12 and Oberheim Xpander.

Xtreme Digital



As with Xtreme Analog, the most important goal with Xtreme Digital, was to create the fattest, deepest sounds possible, covering stacked sounds and complex sound variations. Xtreme Digital provides sounds of unparalleled richness, as most of the stereo multisamples equate to a sound comprised of four single sounds from an expensive digital synthesizer or workstation. The complexity and depth of the samples makes many of the sounds on the CD-ROM surprisingly light on RAM, while maintaining the highest possible quality. To create individual sounds from scratch, you will find basic waveforms in the Sampler Instrument/Initial Waveforms folder.

A special Highlight of the CD are the 12 Chromatic Kits. These are percussive, metallic synthesizer sounds that have been sampled chromatically. As you'd expect in a multisampled drum kit, each key in the 5 octave range features a different sound. In most cases, these sounds have a atonal characteristic, making them well-suited for drums and percussion.

Xtreme HipHop



With Xtreme HipHop, Emagic and King Tech have compiled the most comprehensive resource available for the types of sounds, loops and performances demanded by the the most discerning HipHop, R&B, and Pop producer. Xtreme HipHop contains hard hitting, street-ready sounds from some of Hip Hop's most noted producers and artists. In addition to the traditional rough and nasty street kicks, snares and hihats you would expect from these contributors, Xtreme HipHop offers a huge bank of bonus instrument files and sound bites including: Rhodes phrases, pads, scratches, turntable noises, wah guitars, vinyl hits and more. Every sample on the CD was hand picked and meticulously edited for maximum impact. They deliver fat sounds that have been processed under the advice of some of today's hottest mixers, producers, and engineers. Of course, by using the superb filtering, envelopes, and multi-layering capabilities of the EXS24 or EXSP24 you can contour the samples found on the Xtreme HipHop CDs even further.

MOTU

DIGITAL PERFORMER



Audio Workstation Software with MIDI Sequencing for MacOS

An integrated digital audio and MIDI sequencing production system, Digital Performer provides a comprehensive environment for editing, arranging, mixing, processing and mastering multitrack audio projects for a wide variety of applications. Digital Performer allows you to simultaneously record and playback multiple tracks of digital audio and MIDI data in a totally integrated, creative environment. It features dozens of real-time DSP-effects with easy to use graphical controls, complete automation and supports multiple processor computers. Extensive audio file editing is also included, from the usual cut, paste and copy tools to MOTU's PureDSP functions, providing independent control over the duration and pitch of audio files with exceptional sound quality. This allows for tempo-conforming drum loops, adding vocal harmonies or even gender-bending vocal tracks. Digital Performer's award-winning multitrack sequencer design, combined with non-destructive digital audio editing capabilities, provide you with unprecedented flexibility and control over the audio you create. And now, version 4 takes full advantage of OS X's MIDI and audio services, providing 100% compatibility and inter operability with OS X and all CoreMIDI- and CoreAudio-compatible software and hardware.

FEATURES

Audio

- Unlimited audio tracks (as many tracks as your CPU and hardware will allow) with support for high resolution 24-bit audio up to sample rates of 192kHz.
- Direct support is provided for MOTU's range of professional audio interfaces, such as the 2408MKIII, as well as a wide range of third party audio hardware using Core Audio drivers. ProTools HD support is expected in the near future as a free update.
- Audio tracks can be mono, stereo and surround (n-channel) tracks.
- The Bounce to Disk feature allows you to combine an unlimited number of audio tracks, along with effects and realtime automation, into a single mono, stereo or surround track. Need to hear 100 audio tracks at one time? Just bounce them down. Original tracks are always preserved, so you can go back and tweak them if needed.

MIDI

- Record and playback an unlimited number of MIDI tracks simultaneously.
- MIDI can be edited with a resolution of 1/10,000.000 PPQ (pulses per quarter).
- MIDI timing resolution is accurate within a single MIDI byte (under 1/3 of a ms) when used with a MOTU USB MIDI interface.
- Individually zoomable tracks, flexible window arrangement, and navigation tools allow for trouble-free manipulation within even the largest projects.

Effects

- Dozens of real-time 32-bit and 64-bit DSP-effects are provided with DP to meet the demands of today's audio production including EQs, dynamics processors, reverbs, modulation effects, delay, filters, preamp simulators, mastering plug-ins and more

Audio Editing

- View and edit all of your digital audio tracks in a single, intuitive window
- Audio editing is accurate to a single sample
- Everything from simple dialog editing to complex restructuring of large compositions is as easy as playing with Lego blocks

Automation

- Everything is automatable, including effects parameters with five advanced automation modes and sample accurate editing of automation data
- Support for control surfaces including Mackie Control, Mackie HUI, CM Labs MotorMix, Radikal Technologies SAC-2.2
- Save your fully automated mixdown for instant recall at any time, and then create an unlimited number of alternate mixdowns
- Virtually every parameter can be immediately accessed via customized key commands or MIDI controller. You can even save your key command settings and import them into a DP session at another studio.

How Many Audio Tracks?

The maximum number of tracks you can play back at a time depends primarily on your CPU speed, how fast your hard drive is and how much RAM you have. Based on a Dual 1.2Ghz G4 w/512MB RAM you can expect to get approximately 140-150 independent tracks of audio with 8-band EQ and dynamics on every track.

DIGITAL PERFORMER

Sequence Editor

Side-by-side display of MIDI and audio tracks — The Sequence Editor window provides combined viewing and editing of MIDI notes, audio soundbites (audio regions), audio automation and MIDI controller data in one window along a single timeline.



Open the various MIDI and audio editors and other windows with a single mouse click.

***Movie track** – a QuickTime movie track displays movie frames side by side with MIDI and audio data. DP intelligently displays more frames as you zoom in and fewer frames when you zoom out, so that frames are never obscured by overlapping one another.*

Zoom in to work on fine details or zoom out for a wide overview.

This expanded MIDI track uses a graphical Piano Roll style editor to display MIDI events and controller data using 'audio style' breakpoint automation.

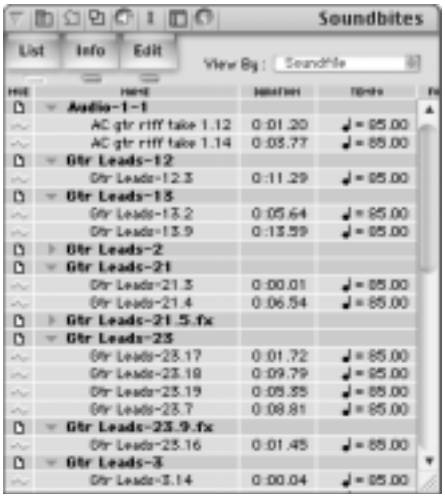
Independent vertical zooming – Both MIDI and audio tracks can be independently resized vertically. Many zoom shortcuts are provided, including the ability to enlarge one track and automatically scale all other tracks to fit in the window.

Record and insert volume and pan effects.

Scale individual audio track sizes as well as adjust the waveform height within each track.

Non-Destructive Audio Editing
Copy and paste regions to repeat verses or sound effects.
Graphically draw volume and pan automation curves.

Soundbites Window



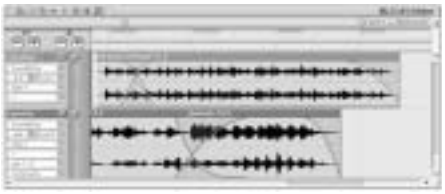
- "Drag and drop" audio regions from the soundbites window to any editing window for intuitive placement.
- Sort by any parameter (length, bit depth etc.) Trace the genealogy of the soundbite by viewing it's family tree

Waveform Editor

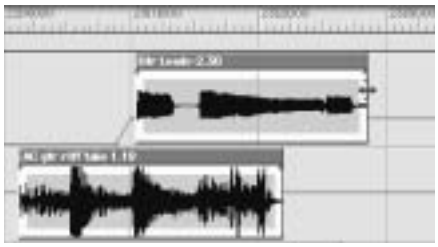


***The Tool Palette includes:** arrow cursor, pencil, waveform tool, waveform selector, rhythm brush (for painting percussive patterns in the Drum Editor), magnifying glass, loop insert, and scrub. These tools operate consistently across all applicable windows. If a tool does not apply to the currently active window, it grays out.*

- The fully-integrated waveform editor provides all of the tools you'd expect from a premium editor including:
  - – View and Edit mono and stereo audio files and create soundbite boundaries with single sample accuracy
  - – Use the pencil tool for removing clicks
  - – A loop tool lets you create perfect loops for your sampler as well as mute and scissor tools
- Many navigation tools are provided to help you work more efficiently like scrubbing, jumping to selection or loop boundaries, as well as user-definable zoom levels



Crossfades help you eliminate unwanted clicks and pops when overlapping soundbites and allow you to perform fade-ins and fade-outs of soundbites. Crossfades (of any length you specify) can also be set to be automatically applied any time you cut, copy, paste, splice and otherwise edit your audio files.



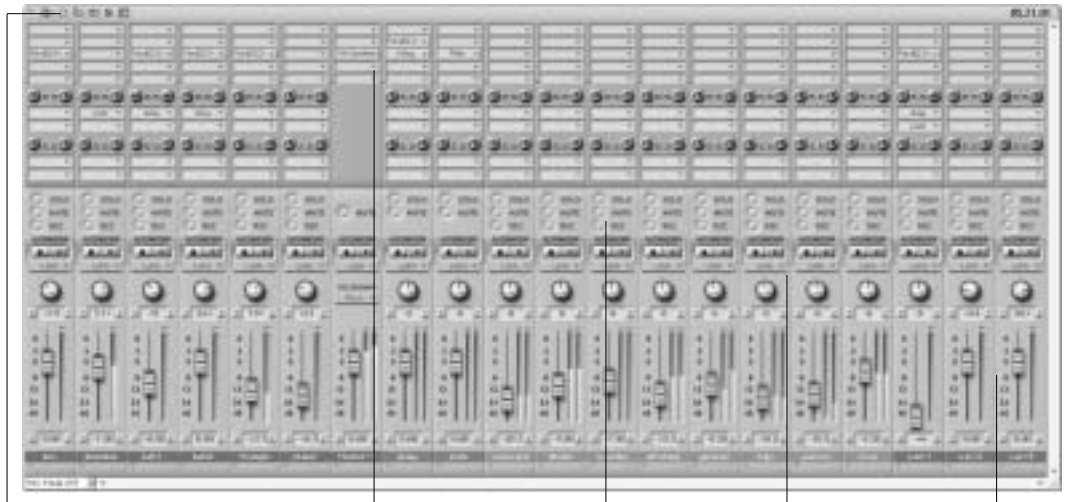
Re-size soundbites by dragging their edges. Split and trim soundbites to work with smaller regions.

MOTU

DIGITAL PERFORMER *continued*

DP's Mixing Board – a virtual mixing console on your computer screen

- More flexible than even the most expensive hardware consoles ever made
- Instantly create customized board layouts: drag track strips anywhere you like, and show or hide any combination of tracks - or even mixer sections (like the inserts section) - with a mouse click.
- Save and recall any number of custom Mixing Board configurations
- Assign tracks to plug-ins and MIDI effects processors for real-time output processing
- Faders can respond to control surfaces or any MIDI controller such as a volume slider or pedal, mod wheel or any data slider.
- View your MIDI and audio tracks in a single, unified mixer
- Save multiple effect presets as a single clipping and then drag and drop them onto any track in any mix
- Use Aux tracks to combine hardware instruments and virtual instruments into one mix



Create snapshots of the entire mixer - or any portion of it – anywhere in your sequence with the click of a button.

Configure up to twenty effects inserts per each audio channel, and the 32 stereo busses.

Solo, mute and record enable buttons

Automation mode and enable/disable buttons

Ride faders and knobs during playback to record automated mixes.

32-Bit and 64-Bit Fully Automatable Realtime DSP Effects

- 32-bit and 64-Bit realtime effects with easy to use intuitive graphical controls
- Effects can be inserted on a single channel or on a bus and accessed via a channel send
- Effects can be easily organized into folders
- Multiple plug-in windows and surround panners can be opened simultaneously
- Up to 20 effects can be inserted pre or post fader per channel
- The number of simultaneous effects is limited only to the speed of your computer
- Multi-processor support of MOTU's and third party plug-ins provides nearly twice as much processing power with dual processor CPUs
- Compatible with effects supporting mono, stereo or surround inputs and mono, stereo or surround outputs
- Sidechain inputs allow you to control an effect parameter from any audio signal in mixing environment by busing the audio to the sidechain input.



Included Effects

- 2, 4 and 8 band EQ
- PreAmp-1 tube-simulation and distortion plug-in
- 3 reverbs
- 2 noise gates including the MasterWorks Gate with real time look ahead gating
- 2 compressors
- A synth-style multimode filter
- Echo & delay effects including a surround delay
- Modulation effects including chorus, phaser, flanger, the Sonic Modulator and more

DIGITAL PERFORMER *continued*

Automation

- Everything in the mixing environment can be automated in real time, including effects parameters, track mutes/solos, effects bypassing, send levels, send mutes
- Automation can be recorded during playback, or you can draw and edit parameters in the audio graphic editor
- Support for three types of automation data: ramps, stare-step, one-shot (such as waveform type - sine, square, etc.)
- Automation data is always displayed using real-world values such as decibels and milliseconds
- Mute and Solo automation allows you to bring tracks in and out of the mix in real time as you listen
- Save your fully automated mixdown for instant recall at any time, and then create an unlimited number of alternate mixdowns
- Using beat/tempo based automation you can control plug-in effects to move in perfect time with your music, from LFO synchronization to filter sweeps that land on downbeats to multitap delays that create syncopated rhythms
- Sample-accurate 32-bit floating ramp automation insures smooth automation movements without unwanted artifacts or zipper noise
- Five advanced automation modes allow you to: Modify automation data you've already recorded; Scale values up or down while maintaining its current contour; Overwrite automation that's already there - but only after you punch in
- Effects automation can be controlled in realtime via MIDI allowing you to add seemingly complex sound design elements, such as sequenced filtering effects in minutes
- Effortlessly create fader automation groups with any fader as the master
- Support for control surfaces including Mackie Control, Mackie HUI, CM Labs MotorMix, Radikal Technologies SAC-2.2 and more

Automatic Audio Conversion

- DP can automatically convert audio data wherever necessary to conform to the current project's sample rate, sample format and tempo. This greatly streamlines the process of importing audio quickly into your projects.
- Automatic conversion options include:
  - Automatic sample format and sample rate conversion when importing audio
  - Automatic time-stretching of audio when placing soundbites into a track
  - Automatic time-stretching of audio when the tempo is changed
  - Automatic placement of imported audio files in a pre-designated folder, regardless of if format conversion took place or not

Acid File Import

- Acid WAV files can be imported using standard drag and drop. If the Acid file has a tempo and automatic tempo conversion is enabled, it will automatically conform to the sequence tempo when placed in a track.
- REX 2 file import allows you to conform Recycled audio files to your DP sessions tempo
- ReWire support allows you to integrate outputs from Propellerheads' ReBirth and Reason applications into DP
- Import and export SDII. Aiff and .Wav files

Time Compression/Expansion

- PureDSP (time compression/expansion) functions provide high quality, independent control over the duration and pitch of audio files allowing you to conform the tempo of drum loops to your project and add vocal harmonies or gender-bend effects to vocal tracks
- Both PureDSP and off-line audio processing occurs in the background so you can continue working
- The graphic time-stretching function allows you to grab the edge of a soundbite with the hand cursor and stretch it longer or shorter.

Sampler Integration

- The Samplers window lets you transfer samples between your project, supported sampler and the Mac's desktop (hard drive) via SCSI using simple drag and drop
- Imported samples are automatically converted into Sound Designer II files and back to the sampler's format when exported
- Using the integrated waveform editor, you can define a region of audio from your project, edit it, trim the edges with sample level precision, specify loop points, normalize, fade in & fade out and send it to your sampler - without ever leaving DP

POLAR

- POLAR is a unique, interactive, RAM-based loop recording environment that allows you to layer multiple passes of audio (limited only to the amount of available RAM) without having to stop recording.
- Use POLAR to layer vocal harmonies; generate mesmerizing polyrhythms; build an entire song or just plain jam.
- When going for that 'perfect take' POLAR can mute the previous take for you automatically so you can concentrate on the music not on playing engineer.



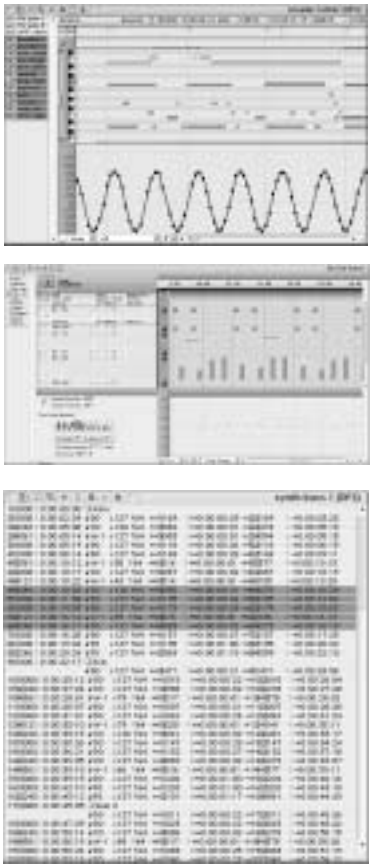
- Because POLAR records audio directly to RAM, you can overdub on the fly without affecting your disk tracks.

DIGITAL PERFORMER *continued*
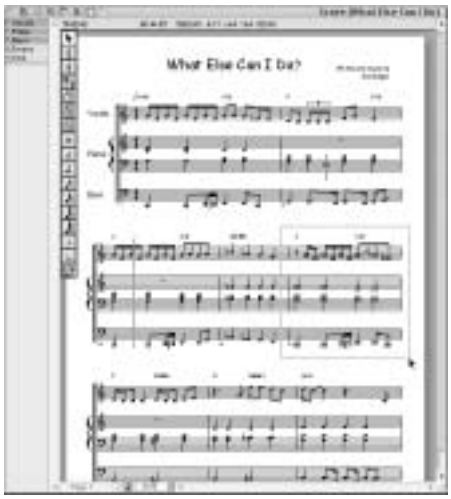
MIDI Editing

- DP provides a range of environments, besides the Sequence Editor, for editing MIDI data including a Graphic (piano style) Editor, an Event List editor, a dedicated Drum Editor and QuickScribe (notation). This combination of editors offers you the flexibility of fine-tuning single MIDI events, or make changes to entire sections
- Multiple MIDI tracks can be edited and displayed in one Graphic Editor window – notes for each track are displayed in the track's color. A track selector list allows you to show/hide tracks as desired
- Three freely switchable MIDI continuous controller editing modes provide flexible and intuitive automation of MIDI instruments

  1) DP's classic 'points' mode
  2) 'Vision-style' colored bars
  3) Audio-style breakpoint automation lines which intelligently interpolates CC data



QuickScribe Notation Window



- Print the whole score or individual parts
- Continuous scrolling moves the music under the wiper which stays fixed to the center of the window.
- Support for five different clefs: treble, bass, grand staff, alto and tenor clefs.
- You can choose a specific instrument transposition as well as a score transposition for each instrument.

Adjustable PPQ

The display resolution can be adjusted to 480, 960, 1920 or any number from 2 to 10,000 PPQ and each tick value can be displayed with up to four decimal places. For example, if you normally edit MIDI data at 480 PPQ, you can set your edit resolution to 480.000 for 1000 times more precision. With the resolution at its maximum value of 10000.0000, you can nudge a MIDI event by a little as one one-hundred millionth of a quarter note.

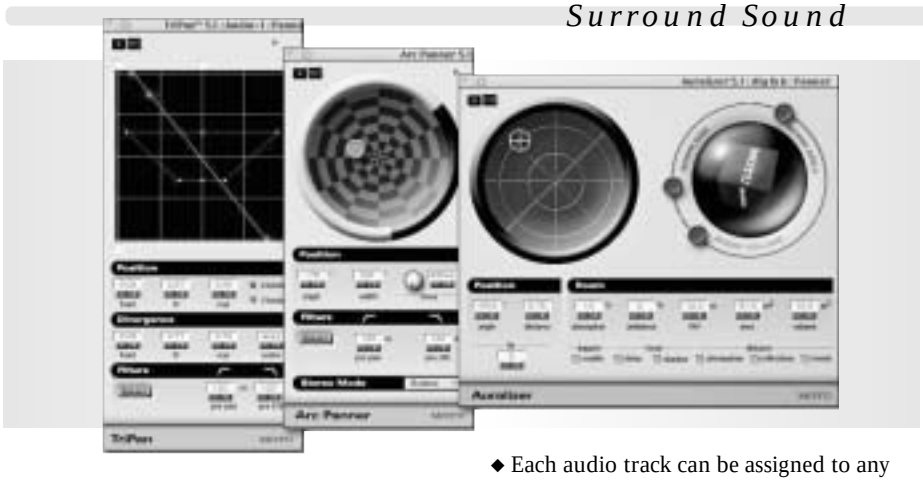
Non-Destructive MIDI Effects Plug-ins



- A variety of MIDI processing plug-ins can be can be inserted into the mixing board window and applied to your MIDI tracks non-destructively in real time
- Bundled MIDI plug-ins include:
  Arpeggiator; Echo; Remove Duplicates (gets rid of duplicate events on the same tick); Transpose; and Shift can all be applied non-destructively and in real-time
- Plug-in settings can be saved and recalled for use on other tracks or in other sessions
- Realtime MIDI Effects can be 'printed', (destructively applied) to the track. This allows you to add effects to certain portions of a track, instead of the entire track

DIGITAL PERFORMER *continued*

Surround Sound



- A complete environment is available for creating surround recordings from start to finish including support for quad, LCRS, 5.1, 6.1, 7.1 and 10.2 surround formats
- Four panner plug-ins are provided including a localizing room simulator – support is also provided for third party panners
- Each audio track can be assigned to any surround sound format and have its own, independent surround panner
- The desired surround panner can be controlled with the panning dish available for each track's channel strip in the Mixing Board. Panners can also be opened as separate windows (just like a plug-in)
- Panning movements are fully automatable
- MacOS Input Sprockets support allows you to connect a compatible USB joystick to your Power Macintosh and use it to control any MOTU or third-party surround panner
- Record, edit and apply effects to multi-channel tracks as easily as mono and stereo tracks – master the multichannel mix using a wide range of channel effects, including the MasterWorks Limiter and bounce the surround submix to disk or record the output of a multichannel mic setup or print a multichannel reverb
- The included Auralizer effect is a room simulator that allows you to localize a sound in space using psychoacoustic cues – design the size and absorptive characteristics of the space you wish to place your audio, then place a sound precisely in that space.

DIGITAL PERFORMER 4.0 AND MAC OS X SUPPORT

- Universal compatibility and interoperability with Mac OS X and all CoreAudio- and CoreMIDI-compatible software and hardware. No special drivers or wrappers — just install your software, plug in your hardware, and go.
- Operate multiple audio hardware systems simultaneously with Digital Performer — use your MOTU 828 FireWire audio interface together with your MOTU 2408mk3 PCI audio interface - or use any combination of CoreAudio-compatible interfaces.
- Mac OS X's Audio MIDI Setup utility delivers universal MIDI system management. DP4 shares a common studio setup with all of your Mac OS X MIDI applications.



- Easy device remapping from FreeMIDI to CoreMIDI when you open a project created in an earlier version of Digital Performer.
- Mac OS X's MIDI device patch lists and drum note names allow you to view the sounds in your MIDI synthesizers by name in Digital Performer's patch lists. Build drum kits with instrument names like "808 kick" and "sizzling hat" rather than note numbers. All of FreeMIDI's of patch lists have been ported to Mac OS X, where they can be used by any CoreMIDI-compatible software.
- Support for interapplication MIDI - Digital Performer can publish an unlimited number of MIDI inputs and outputs, allowing it to transmit and receive a virtually unlimited number of MIDI data streams to and from other CoreMIDI-compatible software.
- Mac OS X's MIDI Time Stamping achieves a 12th of a millisecond MIDI timing accuracy on input and a 3rd of a millisecond on output when used with a MIDI Time Stamping compatible hardware including MOTU's rack-mount MIDI interfaces.
- ReWire 2.0 support provides compatibility with Propellerhead Reason 2.0, Rebirth and all other Rewire applications. Direct MIDI I/O support allows you to sequence MIDI tracks in Digital Performer using your ReWire instruments as virtual synths.
- Enhanced QuickScribe notation transcription engine - display and print unquantized MIDI tracks as beautifully engraved music scores, instrument parts and lead sheets.

Digital Performer Version 4 features that are on the way - to be supplied as a free update soon:

- Support for Audio Units plug-ins for effects and virtual instrument.
- Support for Pro Tools hardware running under the Digidesign Audio Engine (DAE) including ProTools HD hardware.

MOTU

2408MK3

PCI-based Audio Workstation



A cost-effective solution for recording and editing digital audio without sacrificing the power and flexibility needed in today's project and commercial studio environments, the 2408MK3 has everything you need to turn your computer into a powerful 24-bit/96kHz digital audio workstation. The 2408MK3 provides 8 channels of pristine 96kHz analog recording and playback, combined with 24 channels of ADAT and Tascam digital I/O — the most ever offered in a single rack space audio interface. Connect up to four interfaces to the PCI-424 card and you have a system capable of 96 simultaneous active input and output connections at 96kHz. The 2408MK3 is ideal for both the computer-based studio with no mixing board and more elaborate studios built around a digital mixer of any size. For the computer-based studio, the bundled PCI-424 card also features CueMix DSP, a flexible DSP-driven mixing and monitoring matrix that eliminates the need for an external mixer or patch-bay. Connect all your studio gear, including synths, keyboards, and even effects processors. Then control it all from the desktop with no audible monitoring latency and no processor drain on your computer. With 24 channels of digital I/O, the 2408 is ideal for connecting your computer to a digital mixer. Both ADAT and TDIF digital connections are supported at up to 96kHz. And with built-in video and SMPTE synch, you can slave your entire workstation directly to video or SMPTE time code with sub-frame accuracy—without a dedicated synchronizer. The 2408mk3 is cross-platform compatibility with Mac, Windows 98SE/Me/2K/XP and most audio software and host-based effects via WDM/ASIO/Sound Manager drivers. Or use the included AudioDesk workstation software for Mac, with 24-bit recording/editing and 32-bit mixing/processing/mastering.

Inputs/Outputs

- The rear panel of the 2408MK3 rack-mount interface provides eight 1/4″ balanced TRS analog inputs and outputs, an extra pair of main analog outs (1/4″ TRS), three banks of 8-channel ADAT optical "lightpipe", three banks of 8-channel Tascam TDIF, RCA S/PDIF (with an extra S/PDIF output), Audio Wire and BNC word clock I/O.
- The extra stereo output has front panel volume control so you can plug powered monitors directly into the 2408mk3.
- You can choose any three banks (up to 24 channels) to be active at one time. This means you can hook up three ADATs, three DA-88s, and eight analog devices all at the same time and access any three banks - in any combination of formats. And you can freely switch formats at any time.

Expansion Capability

- A core 2408mk3 system provides 24 inputs and outputs, but it also offers the most expansion ever offered in a single PCI card. Up to four interfaces can be connected to the PCI-424 for up to 96 channels of simultaneous input and output at sample rates up to 96kHz. Think about the possibilities: you could connect twelve 8-channel ADAT or TDIF compatible devices including MDMs, digital mixers, and A-to-D converters to your computer, along with 32 channels of analog I/O, and simultaneously record and play as many tracks of audio as your computer allows.
- All four audio formats (analog, optical, TDIF and S/PDIF) support operation at 44.1, 48, 88.2 or 96 kHz. Each 2408mk3 provides up to 24 channels of simultaneous I/O at 44.1 or 48kHz or up to 12 channels —three 4-channel banks– of digital I/O at 88.2 or 96kHz.



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EQUIPMENT LEASING AVAILABLE

S/PDIF Input/Output

- The 2408MK3 provides S/PDIF digital I/O so you can exchange digital audio with a wide variety of other devices. But the 2408MK3 takes S/PDIF one step further than other systems by providing an extra S/PDIF stereo output of the main mix. You can use this extra stereo digital output for whatever you want. For example, you can leave it connected to a DAT machine so that at any time you can record a stereo mix of the project you are working on - without having to swap cables with another S/PDIF device that may be connected.

SMPTE & Word Clock

- The 2408MK3 is the first audio interface to provide on-board video and SMPTE time code synchronization features. This allows you to slave your 2408MK3 system to video, SMPTE time code or both - without a dedicated synchronizer. The PCI-424 card provides a DSP-driven phase-lock engine with sophisticated filtering that provides fast lockup times and sub-frame accuracy.
- S/PDIF jacks (RCA) on the rear panel can be switched via software to become a dedicated SMPTE LTC timecode I/O. However, because the 2408mk3's sync features are driven by the PCI–424's DSP, any analog input can be chosen for SMPTE input, and any active channel, digital or analog, can be chosen as a SMPTE time code output.
- Supplied software also provides a complete set of tools to generate SMPTE for striping, regenerating or slaving other devices to the computer. Synchronization features are cross-platform and compatible with all audio sequencer software.

Legacy I/O Support

- Users who already own a 2408MKII or other PCI-based MOTU recording system can connect their legacy MOTU hardware to the PCI–424 card of a 2408MK3 system.
- All legacy PCI-based MOTU interfaces are supported, including the original 2408, 2408MKII, 1296, 1224, 24I and 308. Users can mix and match legacy interfaces with the new 2408MK3 as they please.
- Legacy interfaces can also take full advantage of the PCI-424's CueMix DSP near-zero latency monitoring.

PCI-424 Card & Cuemix DSP



For the computer-based studio, the PCI–424 card features CueMix DSP, a flexible DSP-driven mixing and monitoring matrix that provides the same near-zero monitoring latency as today's latest digital mixers. CueMix DSP allows you to connect keyboards, synth modules, drum machines, and even effects processors and then monitor these live inputs with no audible delay and no processor drain on the host CPU.

The CueMix DSP engine resides on the PCI–424 card, so it works across all interfaces connected to the card. The CueMix Console software provides an on-screen mixer that gives users hands-on control of their monitor mix, regardless of what audio software they prefer to use. Digital Performer users have the additional option of controlling CueMix DSP directly within DP's mixing environment.

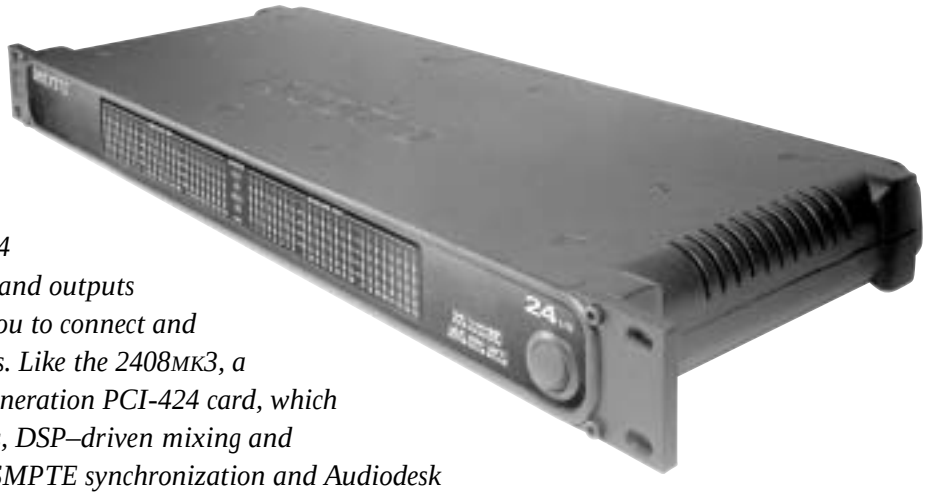
- The PCI-424 is expandable to 96 active inputs/outputs. It supports sample rates up to 192kHz and provides DSP-driven monitor mixing across 96 inputs @96kHz with zero host buffer latency.
- PCI-424 resolves directly to SMPTE time code via any available analog input with sub-frame accuracy. No separate SMPTE synchronizer required. Also provides word clock and ADAT Sync to achieve sample-accurate digital transfers between digital recorders and the computer.
- Cuemix DSP allows you to mix multiple channels into multiple mix busses or directly assigned outputs. Manage your monitor mix with the Cuemix Console.
- Cuemix Console provides independent level and pan controls for each input on every mix. And with CuemixDSP you aren't limited to four stereo mixes. You can configure a unique mix for every physical stereo output you have available. This means you can have up to 48 separate stereo mixes on a 96 output system.

Additional Features

- Includes a setup wizard for both Mac and PC based systems that will have you up and running in minutes. The Setup wizard is an interactive, stand-alone director movie that helps you connect everything together. It even configures the 2408MK3 driver for you when you're done!
- Switchable input levels (+4/ –10dB) for each analog input pair via software.
- 2408MK3 connects to the PCI-424 card (installs in the computer), via 6-pin IEEE 1394 cable. The PCI-424 supports cable lengths of up to 50 feet. Long cable runs are often necessary in studios where the computer is housed in a separate room than the rack–mounted 2408MK3 audio interface.
- Dedicated front-panel five-segment level meters for every analog input and output.
- Separate front-panel volume knobs provide independent volume control for the front-panel headphone jack and the rear-panel main outputs.
- Across-the-board software compatibility — the 2408MK3 ships with a complete set of drivers for Windows 98SE/Me/2K/XP and Macintosh and is compatible with virtually all audio software on both platforms.
- Core 2408MK3 system all includes AudioDesk, MOTU's sample–accurate workstation software for Macintosh with 24-bit recording/editing and 32-bit automated mixing, processing and mastering.

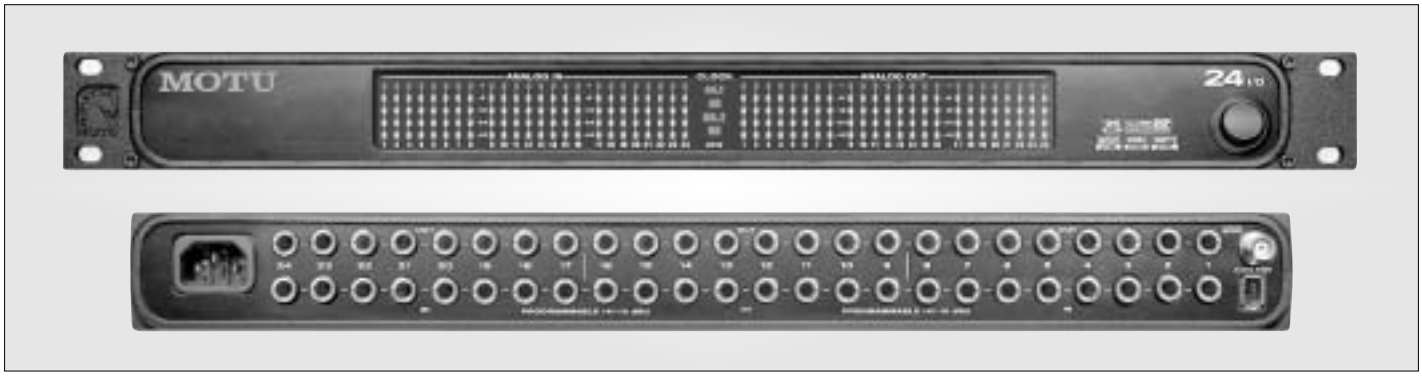
MOTU

24I/O



The 24I/O is a single rack-space 96kHz audio interface for Macintosh and Windows systems that fulfills the promise of host-based hard disk recording: to record, edit, mix process, and master multitrack recording projects entirely inside your computer. It provides 24 balanced 1/4˝ (TRS) 96kHz analog inputs and outputs in a single rack-space enclosure, allowing you to connect and record from 24 simultaneous analog sources. Like the 2408MK3, a 24i/o core system includes MOTU's next generation PCI-424 card, which provides expansion up to 96 inputs/outputs, DSP–driven mixing and monitoring, legacy I/O support, on-board SMPTE synchronization and Audiodesk workstation software for Macintosh.

The 24I/O rack interface which is also available separately as an expansion I/O for 24i/o or 2408mk3 systems provides 5-segment front-panel meters for every input and output and software-switchable (-10/+4) analog input levels.



Metering

The 24I/O's front panel is essentially a dedicated meter bridge for your hard disk recording system. Audio activity for every input and output is represented by its own five-segment LED bar graph.

Front Panel Power Switch

Useful in the unlikely event you wish to turn off your 24I/O. An interesting fact about MOTU Audio interfaces is that they are hot swappable. This means you can power off, plug-in, add and remove interfaces without turning your computer off or restarting. In fact, if you already have a 2408MK3 or other PCI-424 system, adding a 24I/O is as easy as plugging in an AudioWire cable.

24-bit analog TRS I/O

The 24 analog inputs and outputs are professional-grade 1/4˝ TRS (balanced/unbalanced) connectors operating at either -10dB or +4dB. The input level can be switched via software in banks of eight inputs. In addition, CumixDSP provides continuously variable input trim with up to 12dB boost per channel.

Word Clock I/O

The 24I/O audio interface provides industry standard word clock sync on a single BNC connector that can be programmed for input or output via software. When operating as a word clock input, the word connector allows the 24I/O to synchronize smoothly with today's digital audio studio. When operating as a word clock output, the word connector can be used as a master clock to drive other digital devices in your studio, such as a digital mixer.

AudioWire

Uses the second generation AudioWire introduced with the 2408MK3. It is a high-bandwidth digital audio format capable of carrying up to 24 channels of 96 kHz digital audio in and out of the computer at the same time. AudioWire can readily handle the 24 simultaneous inputs and outputs of the 24I/O audio interface.

Power Supply

Comes with an internal power supply and standard detachable IEC power cable. No wall warts or line lumps.

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HD192

High-Definition 192kHz Audio Interface for Mac and PC



Designed for those who demand the very best audio quality available, the HD192 is equipped with the latest generation 24-bit, enhanced multi-bit 128x oversampling 192kHz converters to achieve a remarkable A-weighted signal-to-noise ratio of 120 dB with balanced XLR connectors throughout. Ideal for surround applications, the HD192's 12 inputs and 12 outputs can support two simultaneous 5.1 mixes. And the HD192 is full of advanced features, like AES/EBU I/O with sample rate conversion both in and out.
The PCI-424 card introduced with the 2408mk3 is 192-kHz ready. Just connect the HD192 expansion interface to record 192-kHz audio. The HD192 can be purchased as an expander for an existing PCI-424-based MOTU Audio System, or as a core system which includes the PCI-424 card and Audiodesk workstation software.

Highest-Quality A/D Converters

- The most important link in the digital recording chain is the A/D converter. It is the critical component that converts your analog input signal into digital information. The higher the quality of the A/D converter, the better your recordings will sound. The 192kHz converters in the HD192 are the same ones used in more expensive HD systems. With 120dB A-weighted dynamic range and a THD+N of 0.0005%, the HD192 is simply the best sounding front end you can buy for your digital audio workstation.

Comprehensive Metering

- The HD192's front panel provides dedicated 19-segment level meters for all 12 inputs and outputs. Two over meters are included, a temporary over meter and an over/hold meter that is resettable via software. Comprehensive clock status LEDs are also included so you can see at a glance the current sampling rate and clock source.

Front Panel Power Switch

- Useful in the unlikely event you wish to turn off your HD192. An interesting fact about MOTU Audio interfaces is that they are hot swappable. This means you can power off, plug-in, add and remove interfaces without turning your computer off or restarting.

Inputs/Outputs

- All 12 XLR inputs and outputs on the HD192 rear panel support operation at 44.1, 48, 88.2, 96, 176.4 and 192 kHz and use professional-grade XLR connectors operating at +4 dB.

AES/EBU

- The HD192 features stereo AES/EBU digital I/O with built in sample rate conversion. This gives you the ability to integrate external digital audio sources, no matter what the sample rate, while recording audio at higher sample rates on the analog I/O of the HD192. For example, you can still take advantage of the 96-kHz digital I/O on your favorite outboard reverb, while recording audio at 192-kHz.

Word Clock I/O

- An industry-standard word clock input and output allow the HD192 to synchronize smoothly with today's digital audio studio. The word clock output can be used as a master clock to drive other digital devices in your studio, such as a digital mixer. The AES digital I/O can be driven separately via a dedicated word clock input. You can even slave to a 48kHz clock source, even when the HD192 is operating at 192-kHz.

Audio Wire

- The HD192 uses the same second generation Audio Wire as the 2408MK3. It is a high-bandwidth digital audio format capable of carrying up to 24 channels of 96-kHz or 12 channels of 192-kHz digital audio in and out of the computer at the same time.

Power Supply

- Like all MOTU hardware products, the HD192 comes with an internal power supply and standard detachable IEC power cable. No wall warts or line lumps.

MOTU

828MKII

COMPUTER AUDIO

982

Firewire Audio Interface with ADAT Digital I/O, Mic Preamps and Sync



An affordable, FireWire -based digital audio hard disk recording system for Mac (OS 9, OS X) and Windows (ME, 2000, XP), the 828MKII takes professional audio recording a leap forward in portability and plug-and-play operation, With a total of 20 simultaneous inputs and 22 outputs, two built-in mic pre-amps, zero latency monitoring, and the ability to synchronize to anything via ADAT sync, the 828 provides a complete studio in a box. Also included with the 828MKII is the acclaimed AudioDesk workstation software for Macintosh as well as ASIO, CoreAudio, Sound Manager, GSIF, and WDM drivers for compatibility with your favorite software.

FEATURES

- Eight 1/4″ TRS bal/unbal analog inputs and outputs using 24-bit / 96kHz converters. Input level can be switched between +4/-10 for each input via software.
- Two 24-bit 96kHz analog inputs, on the front panel, using 1/4″ / XLR combo Neutrik connectors, accept either high impedance quarter-inch inputs or low impedance XLR microphone inputs. The microphone preamplifers provide plenty of gain and individually switchable phantom power and variable trim.
- Dual 1/4″ Sends allow you to insert a compressor or effects processor into the two front panel input channels.
- Dedicated 1/4″ TRS main outputs can be connected directly to your powered monitor system.
- Front panel 1/4″ headphone jack
- A dual function knob on the front panel provides separate volume control over main outputs and headphones.
- Eight channels of 24-bit ADAT lightpipe I/O (4 channels at 96kHz) — switchable to stereo optical S/PDIF.
- Stereo 24-bit / 96kHz coaxial S/PDIF I/O
- Dual FireWire ports – one to connect the 828mkII to your computer and a second port to daisy-chain more devices, including additional 828, 828mkII and 896 interfaces without the need for a FireWire hub.
- The CueMix DSP mixer controls, on the front panel, allow you to create up to four stereo monitor mixes with 20 inputs each. Mix to any four output pairs you wish. Because the mixing takes place in the 828mkII itself, your computer's processor is free for other tasks.
- The backlit LCD display provides an intuitive interface to configure your entire system. It even allows you to use the 828mkII as a stand-alone mixer.
- Front panel signal present LEDs for the analog, S/PDIF and optical I/O, as well as tach, lock status and clock rate.
- ADAT 9-pin sync input jack allows sample accurate audio transfers with Alesis ADATs, or any device that supports ADAT sync
- MIDI I/O is provided for accessing a control surface or synth.
- The dedicated 1/4″ SMPTE I/O allows the 828mkII to resolve directly to an external analog tape deck or video source.
- Word clock input and output (BNC) provides synchronization with a wide variety of digital devices or a centralized word clock source.
- A pedal input allows you to plug-in a footswitch for hands-free punch-in recording (software auto-senses pedal polarity for easy setup
- A built in international power supply uses a standard IEC cable. No noisy wall warts or line lumps to keep track of.

Bundled Software

Drivers

Includes drivers (ASIO, WDM, GSIF, CoreAudio, Sound Manager) for all popular Mac and Windows audio software.

AudioDesk

AudioDesk gives you all the advanced features you need to record, edit, mix, process and master an audio project in your Macintosh. It even includes MOTU's ground-breaking PureDSP time-stretching and pitch-shifting technology and dozens of real-time, 32-bit effects plug-ins.

CueMix Plus - Monitoring Control Panel

The 828 gives you the ability to select an input via software and mix it with the main outputs inside the 828mkII to create a no latency monitor mix. Because the mix does not have to travel back to the computer, the result is delay-less patch-thru. This mix is sent out the independent main+monitor outs and headphones. You can adjust the patch thru volume relative to the main mix with the convenient front panel monitor level knob.

896

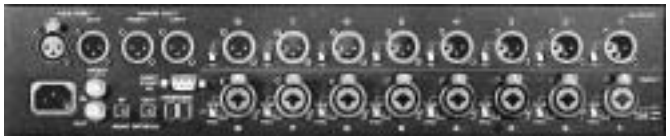
96kHz Firewire Audio Interface

The 896 is a 96kHz Firewire audio interface for Macintosh and Windows. It provides everything you need to make great recordings, all in a durable 2U rack mountable package.



The 896 is expandable. Should your needs grow in the future, simply add additional 896 or 828 interfaces for more I/O.

The 896 is the perfect companion for your laptop computer. With eight built-in mic pre-amps with individually switchable phantom power and no-latency CueMix Plus monitoring, you don't need a mixer for portable recording. Included with the 896 is the acclaimed AudioDesk workstation software for Macintosh. Of course, the 896 ships with Sound Manager, ASIO, GSIF and WDM drivers for compatibility with your favorite software.



FEATURES

AES/EBU Digital I/O

- AES/EBU digital in/out provides digital I/O at any supported sample rate up to 96kHz. Sample rate conversion is supplied on either input or output. For example, it can convert 44.1 kHz audio input to 96 kHz or it can convert 96 kHz audio to 48 kHz.
- In addition, the AES/EBU OUT can resolve to the AES/EBU input while converting, with the input serving as a clock source only (not as an input source).

Analog Inputs

- Each input is equipped with a 24-bit 64x oversampling A/D converter, built-in pre-amp, front-panel trim knob, front-panel 48 volt phantom power switch and a rear-panel 3-way input level switch with three settings: MIC, LINE and +4/FIXED.
- The MIC and LINE settings provide front-panel trim, while the +4/FIXED setting bypasses the trim circuit for unattenuated signals up to approximately +18dBu.
- Together, the MIC and LINE settings provide a trim range of approximately 55dB with some overlap. The LINE setting can accommodate a -10 or +4 input signal.
- MIC setting can be used for mics or unamplified instrument pickups (guitars, etc.).

Analog Outputs

- Eight analog outputs on XLR connectors. each equipped with 24-bit 128x "enhanced multibit" D/A converters and their own switch for either +4/-10dB operation.

Main Outputs

- These two extra XLR outputs can be connected directly to a set of powered monitors. A front panel volume knob is provided for convenient hands-on volume control.
- The main outputs mirror the signal on outputs 1 and 2. In addition, they sum these signals with up to two channels of live input from a pair of inputs via the 896's no-latency monitoring bus. You may choose any two analog or AES/EBU inputs (via software) as the monitor inputs. A separate knob is provided on the front panel for convenient hands-on adjustment of the live input level, relative to the main mix.

Word Clock I/O

- I/O allows the 896 to resolve word clock via a standard synchronizer such as the MOTU MIDI Timepiece AV. Lock to SMPTE time code, blackburst, video or other external clock sources. Slave to word clock at half- or double speed. Send word clock at half speed when running at 88.2 or 96 kHz.

Firewire Interface

- Up to four interfaces can be daisy-chained (connected one to another) on a single FireWire bus (without a hub), providing 72 channels of I/O at 44.1 or 48 kHz. At high sample rates (88.2 or 96 kHz), the ADAT optical channels are disabled and two or three interfaces can supply analog and AES/EBU I/O for 20 to 30 channels of I/O, respectively (depending on system performance).
- A MOTU 828 FireWire audio interface can also be connected to the second FireWire jack of a MOTU 896. In addition, multiple 828s can be mixed and matched with multiple 896s using standard FireWire hubs

Additional Features

- Front panel volume control of monitor input level on main outs
- Front panel headphone jack with independent volume control
- Extensive multi-segment front panel metering for all I/O and clock modes
- Software-configurable peak and clip hold times
- Trim controls for all inputs on front panel
- Front panel footswitch/punch jack

MOTU

AUDIODESK

Multitrack Audio Recording Software — Bundled with all MOTU Audio Interfaces



AudioDesk is a full-featured audio workstation software package for Mac. This is not a watered-down "recording-only" version of another company's software. Instead, the software provides all of the high-end features you'd expect in a serious workstation application, like 24-bit recording and real-time, 32-bit effects processing.

The software includes multi-track waveform editing, sample-accurate placement of audio, a complete virtual mixing environment with up to 64 stereo busses, automated mixing, graphic editing of mix automation, scrubbing, trimming, spotting, crossfades, support for third-party effects plug-ins (in the MOTU Audio System and Adobe Premier formats), unlimited digital track bouncing (including effects and automation), and much more.

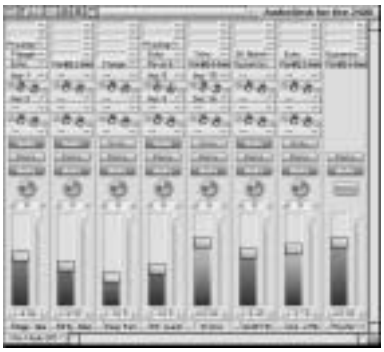
FEATURES

Effects

- Real-time effects all of which are processed with 32-bit, floating-point calculations.
- Because the effects are host-based, the faster your Mac, the more simultaneous effects you can use.
- An entire rack of outboard effects gear right on your Mac —
  - Up to 40 bands of parametric EQ per track
  - Dynamics Processing (expander, compressor, limiter, gate)
  - eVerb reverb plug-in (includes acoustic modeling)
  - PreAmp-1 tube pre-amp simulator (simulates tube warmth and presence, as well as wild distortion effects)
  - Chorus
  - Echo
  - Flange
  - Autopan
  - Tremolo, and more
- More real time effects are available from 3rd party companies including Anatares, Kind Of Loud, Waves, TC Works, Native Instruments. The Adobe Premiere plug in format is also supported

Non-Destructive Editing

- View and edit all of your digital audio tracks in a single, intuitive window
- Graphically draw volume automation curves and pan sounds from left to right
- Select only the tracks you want to see or show them all. Zoom in to work on fine details or zoom out for a wide overview
- Everything from simple dialog editing to structuring complex musical arrangements is as easy as playing with Lego blocks
- Re-size soundbites (audio regions) by dragging their edges
- Scrub digital audio to find exact start and end points for edits
- Split and trim soundbites to work with smaller regions
- Strip Silence works just like a noise gate with adjustable threshold, attack & release
- Crossfade lets you create seamless transitions between overlapping soundbites — eliminating unwanted clicks and pops
- Copy and paste regions to repeat verses or sound effects
- "Drag and drop" audio regions from the soundbites window to any editing window for intuitive placement
- Even import material from your favorite sample CD-ROMs, then use any of AudioDesk's editing functions to customize them for your music





AUDIODESK

Automation

- Mixing Board window gives you a virtual mixing console on your computer screen.
- Ride faders and knobs during playback to record automated mixes.
- Create snapshots of the entire mixer - or any portion of it - anywhere in your sequence with the click of a button.
- Save your fully automated mixdown for instant recall at any time, and then create an unlimited number of alternate mixdowns - all available from this menu for instant recall.
- Easily create fader automation groups with any fader as the master.
- Each track has its own solo, mute and automation enable/disable buttons.
- Instantly create customized board layouts — drag track strips anywhere you like, and show or hide any combination of tracks - or even mixer sections (like the inserts section) - with a mouse click.
- Any number of custom Mixing Board configurations can be saved and recalled at any time.
- Assign tracks to plug-ins for real-time output processing.
- Faders can respond to any MIDI controller such as a volume slider or pedal, modulation wheel or any data slider.

Transport/MMC

- Control any MMC-compatible (MIDI Machine Control) recorder including Tascam DTRS, Alesis ADAT and others.
- MMC devices will chase AudioDesk's transports with frame-accurate timing. Set auto punch-in and preroll times.

OMF Export To Pro Tools

- Built in OMF integration tools allow you to seamlessly export your project to a ProTools environment.
- Besides copying audio data the OMF Export translates critical information such as session offset, audio region placement, and crossfades (either rendered as audio regions or transferred as editable crossfades) is preserved.

Audio Region Layering

- The Audio Layering feature lets you record into a track as many times as you want and then have complete, intuitive graphic control over all of the separate takes to build the perfect performance.
- What you see in the Audio Graphic Editor is always exactly what you will hear.
- Soundbite layering commands give you all the control you need to manage even the most complex multilayered tracks.

Drag And Drop Ease

- AudioDesk provides drag-and-drop consistency and ease throughout
- Add markers to identify SMPTE hit points — grab a marker from the marker well in the main tracks window and drop it right on the time ruler.
- Add a sound effect or other bit of audio by dragging it from the Soundbites list and dropping it in the main waveform display.
- Quickly build a playlist of audio regions by dropping them in an event list.

MOTU HARDWARE AT A GLANCE

	828MKII	896	HD192	2408mk3	24I/O
Analog Inputs (channels)	8	8	12	8	24
Analog Outputs (channels)	8	8	12	8	24
S/PDIF RCA (channels)	2	-	-	2	-
S/PDIF Optical (channels)	2	-	-	-	-
AES/EBU (channels)	-	2	2	-	-
Tascam TDIF (channels)	-	-	-	24	-
ADAT Lightpipe (channels)	8	8	-	24	-
Sampling Rates	44.1, 48, 88.2, 96K	44.1, 48, 88.2, 96K	44.1, 48, 88.2, 96, 176.4, 192K	44.1, 48, 88.2, 96K	44.1, 48, 88.2, 96K
Protocol	FireWire	FireWire	AudioWire	AudioWire	AudioWire
Rack Spaces	1U	2U	2U	1U	1U
Expandable	yes	yes	yes	yes	yes

AudioDesk is bundled free with all MOTU Audio Interfaces

MOTU

MIDI INTERFACES

MOTU manufactures a wide range of MIDI interfaces for both Mac and Windows compatible computers to meet the requirements of any MIDI Setup. Whether you are looking for a basic 2x2 interface or the ability to integrate video, hard disk and tape based recording systems, MOTU has got you covered. And now, MOTU's award-winning line of MIDI interfaces are available for those of you with USB equipped Macs and soon PC's. Featuring easy to use plug-n-play setup as well as hot swap capabilities that allow you to plug and unplug your interface even when your computer is turned on. For even greater flexibility, you can mix and match any interfaces and even use multiple interfaces connected to a standard USB hub. From the affordable FastLane to the top of the line MIDI Timepiece AV, MOTU has a USB interface for every application and price range.

FASTLANE USB

2x2 MIDI Interface for Mac & Windows



A professional MIDI interface for the home studio, FastLane installs in seconds and opens your computer to a world of music software. And FastLane's unique advantage: a MIDI thru button allows you to play to your gear even when your computer is turned off-without having to disconnect cables. No other USB interface offers this feature for this price.

FastLane takes full advantage of USB, giving you increased MIDI throughput, better timing accuracy, support for "hot-swapping" and plug-and-play expansion. Need to connect another sound module or synth? No problem. Just add another FastLane via a standard USB hub. Or mix and match FastLane with another multi-port USB interface. FastLane is specially engineered to operate with multiple USB devices, so your software will never lose contact with it, regardless of how many times you restart your computer or how many other USB devices you connect.

- Two 16 channel MIDI I/O ports provide 32 simultaneous channels of MIDI inputs and outputs for USB-equipped Mac and Windows 98/ME computers.
- USB allows hot swapping and plug-and-play expansion — add another FastLane to a standard USB hub or mix and match interfaces to suite your needs.
- Connects any MIDI (Musical Instrument Digital Interface) device, such as a synth or sampler, to the computer via USB.
- MIDI Thru button splits MIDI signals from each input to both outputs simultaneously allowing you to play a MIDI sound module from a MIDI keyboard or other controller even when the computer is turned off.
- Powered by its connection to USB — no power supply or batteries are needed.
- Four status LEDs indicate MIDI activity.
- Compatible with both FreeMIDI and OMS on the Mac
- Available in graphite, charcoal, indigo, red and sage
- Includes a USB cable and easy-to-install software drivers on CD-ROM.

MICRO LITE *5-in/5-out MIDI Interface*



The Micro Lite is a professional MIDI interface that provides portable, plug-and-play connectivity to any USB-equipped Mac or Windows computer. Features high-speed MIDI throughput, sub-millisecond timing accuracy, support for "hot-swapping" and plug-and-play expansion. And, powered by the USB port — no wall wart or AC plug required.

- Five MIDI inputs and outputs provide 80 MIDI channels. Add additional MIDI ports by plugging in another MOTU MIDI interface.
- Includes drivers for Mac OS9, Mac OS X and Windows

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POCKET EXPRESS

Single 2x4 MIDI/SMPTE Interface

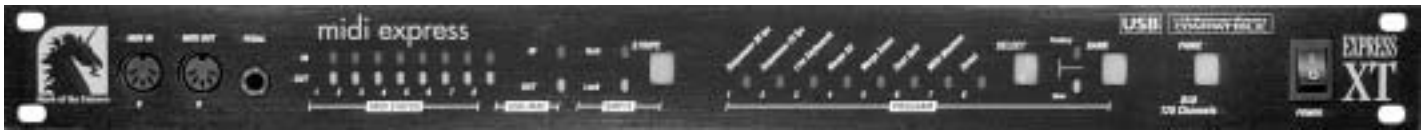


This 2 in, 4 out 32-channel MIDI/SMPTE interface works with either your Macintosh or Windows computer at a great price. Installs in seconds and opens your computer to a world of music software. Fully independent, paired outputs provide 32 separate MIDI channels. And the Pocket Express's unique advantage in its price range: a computer bypass button allows you to play your MIDI gear even when the computer is turned off.

Compatible with all Mac and Windows MIDI software. Includes full SMPTE synchronization capabilities in all formats, including jam-sync for trouble-free synchronization. And another Pocket Express unique advantage at its price: front panel SMPTE controls for fast, convenient striping and lockup.

MIDI EXPRESS XT

8-in/8-out MIDI/SMPTE Interface/Patchbay/Merger



The MIDI Express XT USB is a multiport MIDI interface and SMPTE synchronizer for Mac and Windows. Featuring 128 channels, the Express XT is ideal for medium-sized MIDI setups and essential tasks such as MIDI sequencing, multimedia production, and sound management with patch/editor librarian software.

- 8 MIDI inputs and 9 MIDI outputs provide 128 MIDI channels via USB
- Sixteen convenient one-touch front-panel presets (8 factory and 8 user programmable) with memory backup for plug and play performance.
- Converts LTC to MIDI Time Code (MTC) allowing you to sync your computer or any other devices to SMPTE time code.
- Converts and stripes SMPTE in all formats (24, 25, 29.97 drop/non-drop, and 30 fps). SMPTE jam sync with adjustable freewheeling for drop-out free sync.
- Full support for MIDI Machine Control—serves as either an MMC master or slave; distributes transport commands to all other MMC devices.
- Pedal input for momentary foot switch or audio click-to-MIDI conversion.
- Internal power supply (no wall-warts)
- 8x8 MIDI merge matrix works as a standalone MIDI patchbay/merger processor. Bundled software console provides comprehensive routing, merging, and muting on your Mac or PC.

—Route any inputs to any outputs
—Merge any inputs to any outputs
—Mute any MIDI data on any cable
—Re-channelize on input and output
—Recalls setups via MIDI patch change

MICRO EXPRESS

4-in/6-out MIDI Interface w/Sync



The Micro Express is a mini version of the MIDI Express XT. It has all of the same features as the Express XT - except that it is smaller and even more affordable. The Micro Express is a half-rack unit with 4 inputs and 6 outputs for 96 MIDI channels.

MOTU

MIDI TIMEPIECE AV

8x8 MIDI Interface/Synchronizer/Patchbay/Merger



Combine MIDI with ADATs, ProTools, or video. The MIDI Timepiece AV takes the world-renowned MIDI Express XT and adds synchronization that you really need, like video genlock, ADAT sync, and word clock sync. Even Digidesign superclock! Slave ADATs, Digidesign hardware or word-clock compatible devices to SMTPE and video. Drive your ADATs from your sequencer without an Alesis BRC. Or control everything from an MMC controller like JL Cooper's CuePoint. The MTP AV pulls it all together for stable, centralized, hassle-free synchronization and MIDI networking. For Macintosh, Windows, and stand-alone setups.

The MTP AV is also an 8x8 merger and router. You can merge any combinations of the eight inputs simultaneously to any combination of the eight outputs. Because the MTP AV's internal memory can be programmed from the front panel, the MTP AV is the perfect choice for live performance applications where fast and flexible MIDI patching is essential. You can take the MTP AV on the road because it can operate with or without a computer. The MTP AV can be set to change scenes in response to patch changes, allowing remote control from any MIDI controller.

Takes full advantage of USB, giving you increased MIDI throughput, better timing accuracy, support for "hot-swapping" and plug-and-play expansion. Need to connect another sound module or synth? No problem. Just add another MIDI interface via a standard USB hub.

FEATURES

- 8 MIDI inputs, 8 MIDI outputs (128 MIDI channels)
- Networking offers up to 512 MIDI channels
- Support for Mac and Windows in one unit - it can even network Mac & PC to each other
- Video sync input - genlocks to video or blackburst
- ADAT sync output - drives ADATs without a BRC
- Word clock sync output
- Digidesign "superclock" output -drives Pro Tools
- MIDI Machine Control - controls everything, even ADATs, from an MMC sequencer or LRC
- Stripes frame-locked LTC onto video
- Syncs ADAT with Pro Tools
- Slaves ADAT, Pro Tools and your computer to video
- Provides 0.1% pullup/pulldown for digital audio with video

MIDI EXPRESS 128

8-in/8-out MIDI Interface



Built from the same technology found in MOTU's flagship MIDI Timepiece, the MIDI Express 128 is a professional MIDI interface that provides plug-and-play connectivity to any USB-equipped Mac or Windows computer. Taking full advantage of USB, MIDI Express 128 gives you high-speed MIDI throughput, sub-millisecond timing accuracy, support for "hot-swapping" and plug-and-play expansion. Need to connect another sound module or synth? No problem. Just add another MOTU USB MIDI interface via any available USB port.

- Eight independent MIDI inputs and outputs for a total of 128 MIDI channels.
- Expansion is simple. Add additional MIDI ports by plugging in another MOTU MIDI interface.
- Includes drivers for Mac OS9, Mac OS X and Windows Me/2000/XP.
- Bus-powered from USB connection. No additional power necessary, and no wall wart or AC plug.

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ONE HOUR FREE PARKING
AT 349 W. 34th STREET (with purchase of $100 or more)

DIGITAL TIMEPIECE

Digital Synchronization Hub



The Digital Timepiece synchronizes digital audio equipment with knife-edge precision: MOTU 2408MK3, Digidesign ProTools systems (without a VSD or SDD), Alesis ADAT (without a BRC), Tascam DA-88 (without an SY-88 or RC-848), ADAT and DA-88 compatibles, SONY 9-pin video decks, word clock, S/PDIF DAT decks, Digidesign Audiomedia II & III, NTSC/PAL video (all formats), computer software and digital audio workstations - including support for SMPTE time code, VITC, MIDI Time Code and MIDI Machine Control.

Until now, phase-locked synchronization between these systems has been difficult or impossible. The Digital Timepiece is the first synchronizer to lock them all together with a stable, high-resolution time base - with no dithering, rounding, or software delays. Choose any one device as the master; all the rest slave precisely and reliably. And drive everything from a MMC controller, computer sequencer or digital audio workstation.

FEATURES

- Universal synchronizer for SMPTE, digital audio, and video systems
- Supplies all components necessary for accurate, stable synchronization: address (SMPTE time code location and audio sample number), time base (word clock), and machine control (transport and cueing)
- Choose any source as the time base master, and the Digital Timepiece continuously and simultaneously generates all other synchronization formats to drive all connected devices in perfect phase lock
- Sample accurate phase-lock during playback for all word clock devices, including ProTools
- Sample accurate locating for all devices that support it, including ADATs, DA-88s and ADAT/DA-88 compatibles
- Front panel selection of any time base includes ADAT, DA-88, MTC, SMPTE (LTC), Video (VITC), word clock (1x), Digidesign "superclock" (Word Word 256x), S/PDIF, Control Track, or Internal
- Generates extremely stable, high-resolution time base with custom-designed VLSI technology and a proprietary high-frequency phase engine
- Fast lock-up time
- No dithering, rounding, or software delays for pristine sound
- Eliminates the need for expensive synchronization add-on equipment, such as the Digidesign SMPTE Slave Driver, Video Slave Driver, Alesis BRC, Tascam SY-88 card, and more
- Supports MIDI Machine Control - serves as either an MMC master or slave, distributes MMC transport commands from a sequencer, or any MMC transport controller to all other devices, including SONY 9-pin VCRs, ADAT and DA-88
- Proprietary control track sync format provides address, time base, and machine control - allows multiple DTP's to be synchronized together, also allows support for future devices
- 44.1 or 48 kHz samples rates with 0.1% pull-up

Video Features

- Supports SONY 9-pin
- Video sync generator
- Reads and writes VITC
- SMPTE burn-in and status display
- Sequencer-triggered streamers with full-screen punch
- 0.1% pull-up/pull-down at either 44.1 or 48 KHz for drop-frame conversion

Supported Devices

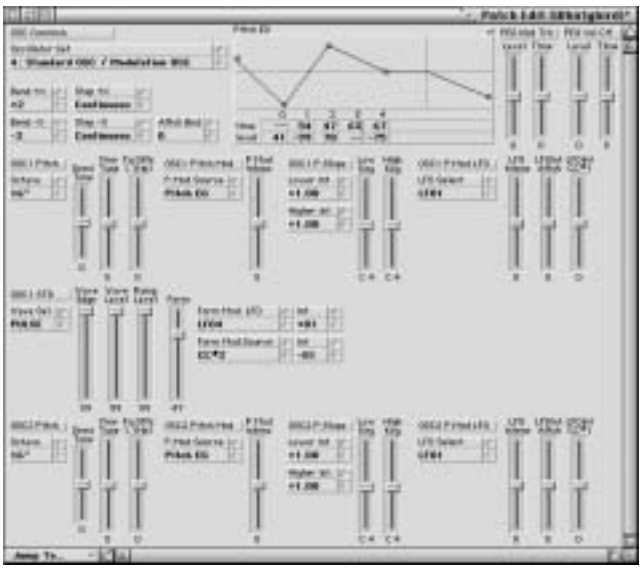
- Alesis ADAT or any other ADAT-sync compatible device (without Alesis BRC)
- Tascam DA-88 and DA-38 (without an SY-88 sync card or RC-848 controller)
- Digidesign hard disk recording systems such as Pro Tools III and Pro Tools Project (without a SMPTE Slave Driver or Video Slave Driver)
- Any digital audio recording system that can slave to (and, optionally, generate) standard word clock
- S/PDIF-compatible audio devices, such as DAT decks and hard disk recording cards like Digidesign's Audiomedia III - includes S/PDIF thru button on front panel.
- SONY 9-pin compatible video decks
- MIDI Machine Control (MMC) devices
- Computer software or any hardware system that can slave (or generate) SMPTE Time Code (LTC or VITC in all formats, including 29.97 drop and non-drop)
- Video (VITC and Video Black)
- Any devices that slave to (or generate) MIDI Time Code (MTC)

MOTU

UNISYN

MIDI Device Editor/Librarian

The industry's most popular editor/librarian, Unisyn provides the most comprehensive sound management features available on the Macintosh and Windows, including seamless integration with Performer. You can modify a sound in Unisyn using graphic envelope controls and faders, while getting instant feedback within the context of your music as Performer plays the sequence. Generate entire banks of new sounds with a click of the mouse using Blend, Randomize, and Copy/Paste Parameter features. Unisyn can even share bank names with Performer and other FreeMIDI-compatible software for accurate pop-up sound lists. Unisyn can store thousands of sounds at your fingertips and recall them instantly using database-style search criteria, such as "plucked electric bass" with "bright stereo flange". Frustrated because you can't recreate the settings in your gear for last month's project? Unisyn can do it with a few clicks of the mouse.



FEATURES

- Includes over 40 new devices such as the Korg Triton series, Proteus 2000/2500, Waldorf synths and Roland modules like the powerful XV-5080.
- Digital Performer users will feel right at home with Unisyn's new interface. Editors have been updated to provide a clearer, more easy to use layout of parameters.
- Unlimited undo allows you to concentrate on your editing, not on the software.
- Intelligently manages removable media
- Nearly everything is remembered between sessions. You automatically pickup right where you left off.
- 26 window sets provide a quick and flexible way to manage window layouts.
- Maintains links from parent files (for example, performances) with their associated 'children' (for example, patches) to make managing your data a breeze.
- 'Compare' feature displays which parameters have changed and what the different values are. Compare is "live," keeping track of your changes as you edit a patch.
- Most profiles use a modular bank upload/download system - which speeds up your workflow by only downloading the data that is required.
- Existing profiles have been updated to take advantage of Unisyn improvements such as parent/child relationships, card/cartridge management and hierarchical patch editor structure.
- Unisyn's blend & mingle and randomize features are combined into a full-blown patch generator that generates hundreds of patches at the click of a button (unrestricted by the bank size of the device)
- Snapshots group together all the individual elements that used by your devices so everything can be restored perfectly, even if your MIDI setup changes. All the required data resides inside a snapshot.
- Improved pasting options allow you to paste sections or groups of data from one patch to another to greatly speed sound design.
- All-purpose documents (formerly known as libraries) can contain a mixed combination of patches, banks and snapshots, consisting of any type of data. The number of items in the document is unlimited.
- Unisyn allows you to assign any number of keywords to a patch so you can locate patches quickly and easily.

FreeMIDI

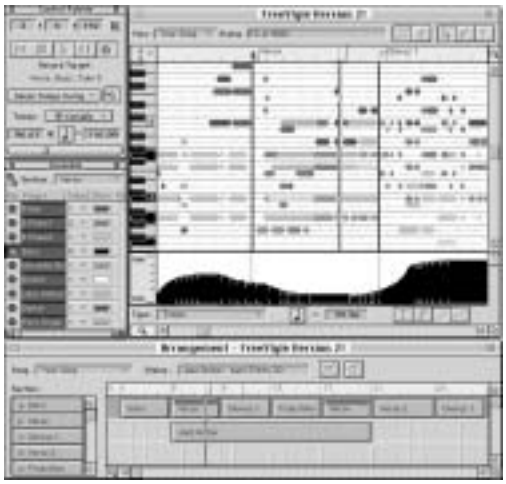
FreeMIDI is a complete MIDI operating system for Macintosh. It ships free and is automatically installed with all MOTU audio software products. FreeMIDI is also supported by many third party developers. (FreeMIDI developer kits can be downloaded from MOTU's website).

FreeMIDI automatically detects what type of MIDI interface is connected to the Mac's serial or USB ports, automatically detects what MIDI devices are connected to interface (it "knows" over 200 types of devices), and provides a graphical representation of their MIDI studio.

FreeMIDI also provides pop-up sound lists for over 100 popular MIDI synthesizers-as well as generic support for any General MIDI device. It even includes advanced features such as inter-application communication and multiple application real-time synchronization.

FREESTYLE

Trackless Sequencing and Instant Notation Printing Software



A trackless sequencer with instant music notation, Freestyle is powerful, yet easy to use so you have the freedom to concentrate on what is most important—your music. Create compositions intuitively using ensembles, players, takes and arrangements. Get inspired with FreeStyle's dozens of drum riffs. Notate your performances as you play. See your music on screen exactly as it will print. Get started quickly with built-in support for dozens of popular MIDI instruments, including any General MIDI device. FreeStyle provides both tape recorder and drum machine style composing.

FEATURES

- Event List window displays any type of MIDI data, from notes to system exclusive data to Registered and Non-Registered Parameter Numbers. Show or hide any combination of MIDI data; Edit any data parameter numerically; Scrub over the list of data with the playback wiper; Fly through consecutive edits with the arrow keys.
- Sense Tempo feature lets you sit down in front of your keyboard and just start playing. No robotic metronome clicks. No countoff. Just you and your inspiration. FreeStyle records your every nuance - both the notes you play and the tempos at which you play them - at the highest resolution available today in any sequencer.
- FreeStyle also has a host of beat adjustment features to produce optimum transcription and musical editing

—With the Identify Beats feature, just click on notes that occur on downbeats in the graphic editor or notation window.

—Adjust Beats feature lets you drag barlines and beats in FreeStyle's graphic editor to match the notes below, scrubbing the music as you drag so you can easily zero in on the correct downbeat.

—With Record Beats you can simply tap along with your rubato performance to tell FreeStyle where the beats are.

— After using any combination of these beat adjustment features, you can choose to listen to your music with its original tempo and feel, or you can temporarily switch to "Constant" tempo mode to hear it at a perfectly straight tempo.

- Change meter and key signature anywhere in your music. If you're in a hurry, just click on a barline to insert a meter or key change there. If you want to map out a whole section, use the Change Meter or Change Key windows.
- Draw, record and edit as many tempo changes as you like
- Add, duplicate, name and delete as many different tempo maps as you want for a section - lets you freely experiment with different tempos
- "% Variable" tempo mode: Speed up and slow down your music with the tempo slider, even if it already has an elaborate tempo map
- Create tempo changes by moving the tempo slider
- Assign more than one sound to a player - even from different MIDI instruments
- Override FreeStyle's dynamic channel assignment feature for complete control over your MIDI gear
- Support for QuickTime Musical Instruments allows you to play music on your Mac without a MIDI instrument
- MIDI Monitor window tells you what's going on with your MIDI gear - and with FreeStyle's MIDI output - at one glance
- Visual Step Record lets you rapidly enter and visually verify note data
- Choose Notes command lets you select and deselect notes based on their location, pitch, duration and velocity
- Name-able takes helps you keep track of multiple takes for a player
- Note spelling algorithms give you improved automated transcription so you spend less time fiddling with accidentals, while manual note spelling adjustment gives you complete control over note spellings
- Score transposition–display a player in any key you wish in your scores
- Recording transposition–choose any transposition you wish while recording a player– play in any key and have FreeStyle transpose what you play on the fly
- Tweak Notes (velocity, duration, scaling) feature lets you make notes longer, shorter, louder, faster - and other effects - in one convenient step, while the music plays
- Duplicate Take command copies the entire contents of a take in one easy step
- Switch Staff command moves notes from one staff to another on a grand staff in one easy step
- Change a player's sound in mid-performance - even to an entirely different MIDI instrument
- Expanded MIDI file support—reads and writes Type 0 and Type 1 MIDI files
- SMPTE synchronization lets you slave FreeStyle to your tape deck and record vocals and other audio tracks in perfect sync. Or lock FreeStyle to film or video. All frame rates are supported. FreeStyle also transmits MIDI Time Code (MTC), which allows you to synchronize it with popular hard disk recorders

MOTU

MACHFIVE

Universal Sampler Plug-In



A universal sampler plug-in for Mac and Windows, MachFive supports all major plug-in formats on both platforms (MAS, VST, RTAS, HTDM, Audio Units and DXi) and imports all major audio file, sample and soundbank formats including AKAI, Kurzweil, Roland, EMU, SampleCell, GIGA, EXS24, HALion, Creamware, WAV, ACID, AIFF, SDII, REX and more. MachFive also offers many advanced sampling features, including support for 192kHz and the ability to import and playback multi-channel samples in surround.

MachFive operates inside application such as Digital Performer, Pro Tools, Logic Audio, Cubase, Sonar and virtually any major application for Mac and Windows. This allows MachFive to be employed as a flexible, state-of-the-art sound source directly within their projects and then save all MachFive settings with the project for instant and total recall. Since all MachFive settings are saved with their host application session, you enjoy the highest degree of convenience and speed because there is no separate application or associated documents to manage.

FEATURES

As a plug-in, MachFive displays all editing and performance parameters in one window, showing users everything in one glance, without the need to flip through different pages or dig through menus.

With unprecedented compatibility and inter-operability, MachFive is a truly universal software sampler. It supports every major production platform on Mac and Windows (MAS, VST, Audio Units, HTDM, RTAS and DXi), so you can effortlessly move from one platform to another —or collaborate with those who use different audio software.

For example, a user might compose and track a project in Digital Performer or Logic and then move to Pro Tools for mixing. The user would simply save a MachFive performance (a snapshot of all MachFive settings) in DP or Logic and then load it into MachFive running in Pro Tools. All settings are exactly preserved, and MachFive is ready to go.

Compatible with all Sample Libraries

- MachFive includes UVI-Xtract, an import utility that allows users to audition and load programs and samples from every major sampler format, including Akai, Roland, Kurzweil, SampleCell, HALion, EXS24, Gigasampler and others.
- UVI-Extract even allows users to insert sample content CD-ROMs from Roland, EMU, AKAI and other formats into their computer — discs that otherwise cannot be mounted on their computer desktop — and then convert programs (or even the entire disc) in just a few clicks.
- MachFive helps users consolidate their current sound libraries into a unified MachFive library, instantly accessible from within their host application.

Flexible Multi-Timbral Performance

- Open as many MachFive plug-ins as the host CPU permits. Each plug-in instance of MachFive provides 16 parts (separate instruments). Each part has its own unique audio output assignment, volume, pan, etc. Each part can receive MIDI data from any channel and send its output to unique audio outputs (depending on the host audio software).
- Simple stacks can be created by assigning two or more parts to the same MIDI channel. However, MachFive also provides Expert Mode, which lets you create complex layers and splits that can be dynamically triggered and modified on the fly. Expert Mode was designed for demanding live performance situations.

MACHFIVE

Intelligent File Management

The most critical feature users need from a sampler is easy access to their sounds. MachFive offers unprecedented sound bank management, helping users concentrate on their music — not file handling chores on their hard disk. MachFive always remembers where sounds are located, and it has been optimized for browsing and loading libraries. Even multi-gigabyte libraries are quickly and efficiently scanned.

Powerful Synthesis Engine

The central section of the MachFive window shows the filter, three envelopes and every modulation tool. Six filter algorithms are provided and everything can be controlled and automated by MIDI. Modulation options are provided at every stage of the synth section. The award-winning UVI-Engine that powers MachFive delivers unlimited polyphony and ultra-low latency.

Multichannel Waveform Editor

MachFive accepts audio samples in all formats from mono to 5.1 surround. The waveform display allows truncating, normalizing, fading and many other destructive DSP audio manipulations — all in real time. Users can even set the crossfade of a loop in real time while listening to their full mix.

24-bit 192kHz Audio

MachFive takes advantage of high definition audio interfaces like the MOTU HD192, allowing users to build their sound libraries at any sample rate up to 192kHz. MachFive also provides on-the-fly downsampling, allowing users to build a high-resolution sample library and use it at any sample rate that their current project calls for, without the need for lengthy sample rate conversion operations.

Surround-capable Sampler

MachFive is a true surround sampler. Users can play and transpose 5.1 audio files in real time, add multi-channel effects and route multi-channel presets to their host software's mixer with multiple outputs (if the host software supports these surround features).

Drag & Drop Import of Samples

Users can simply drag samples from their computer desktop or host application to the MachFive keyboard. Users can even drag multiple samples in one step to map into layers or splits, chromatically, on white keys only, according to their name, or even according to their pitch, which MachFive can determine automatically. Users can also audition samples when importing them, listening to each note as they stretch the sample over a range of keys.

Integrated Multi-Effects

Each multi-timbral part can have up to four unique effects, for up to 64 effects per performance with instant recall, including reverb, tempo-synced delay, tremolo, chorus (and other modulation effects), filter, BitCrusher, and others. All effects settings are saved for total recall. Users can also save effects with each preset for permanent storage in their sound library. The effect section allows users to leave a part's effects in place while they audition other parts.

Analyze and Tune

MachFive is a one-stop sound design factory, with tools such as a graphic spectrum analyzer and a built-in tuner with graphic display. When importing samples, you can even ask it to analyze the sample, determine its pitch and assign it to the appropriate keymap pitch.

Modulation and Tempo Sync

Four LFOs are available per preset: two that can be applied across a preset's entire keygroup, plus two additional LFOs that can be applied individually to each sample. Each LFO can be routed to an assortment of destinations including filter frequency, filter resonance, drive, pitch, pan and amplitude. MachFive can sync both the LFO and effects parameters to sequence tempo.

MachFive Specifications

Compatible Audio Formats	
AIFF:	mono, interleaved (2, 4, 6 channels)
AIFF:	Separate files: 2 (stereo), 4 (quad), and 6 (5.1) channels
SDII:	mono, stereo interleaved
SDII:	Separate files: 2 (stereo), 4 (quad) and 6 (5.1) channels
WAVE:	mono, stereo interleaved
WAVE:	Separate files: 2 (stereo), 4 (quad) and 6 (5.1) channels

Soundbank Import Formats
Akai S1000 / S3000
Akai S5000 / S6000
Akai MPC2000 / MPC3000
Roland S7xx
EMU III / ESI / IV / EOS
Tascam GigaSampler/GigaStudio
Emagic EXS24
Digidesign SampleCell
Creamware Pulsar STS

Sample Import Formats
Kurzweil K2xxx
Akai MPC .snd
Creamware Pulsar .s
WAV/ACID
AIFF
SDII
REX

Channel Formats
mono to stereo
mono to quad
mono to 5.1
stereo
stereo to quad
stereo to 5.1
quad to 5.1

Plug-in Formats
MAS, VST (Mac & PC), RTAS, HTDM, Audio Units, DXi

STEINBERG

CUBASE SL • CUBASE SX



Professional Music Creation and Production Software

The Cubase legacy continues with Cubase SL and Cubase SX — the most advanced and powerful versions of Steinberg's renowned Virtual Recording Studio software to date. Designed from the ground up to take advantage of the latest breakthroughs in processor and operation systems technologies including WindowsXP and Mac OSX, both Cubase SL and Cubase SX provide the potential for recording and playing back an unlimited number audio tracks (up to 96kHz) and MIDI tracks and feature a comprehensive suite of advanced editing facilities, realtime effects, mixing and automation facilities. Both applications support realtime VST 2.0 effects plug-ins and virtual instruments, and low-latency, multi-channel ASIO 2.0 compatible audio hardware allowing you to outfit your computer-based recording studio to suits your specific requirements. VST System Link functionality allows several computers to be linked together and actually perform as one fully integrated system. Cubase SX adds comprehensive 5.1 surround sound support, complete score layout features and a wider range of automation modes.

THEY BOTH FEATURE

Audio

- Native audio – no expensive additional outboard equipment is required to record and playback audio. Plus ASIO soundcard support for low latencies.
- Up to 200 or more simultaneous playback of Audio Channels (cpu dependent)
- Recording multiple channels of audio simultaneously.
- Support for 16-Bit, 24-Bit and 32-Bit float audio files at sample rates of 44.1 kHz, 48 kHz, 88.2 kHz, and 96 kHz.
- Record of AIFF, WAVE and Broadcast WAVE files, (Sound Designer II for Mac)
- Non-destructive Part Editor
- Destructive rendering of virtual effect processors into parts from the Project Window (with process history).
- Very quick and efficient parts bouncing
- Simple 'Drag-and-Drop' functionality means allows objects to be moved quickly, not just within projects, but also between multiple projects.

Mixing and Automation

- Adaptive and configurable track mixer selectable (wide/narrow) views.
- Up to 64 Group Channels with access to the same audio effects and EQ as normal audio tracks.
- Sample accurate automation with automation tracks for every audio, group track and plug-in.

VST Effects and Instruments

- Includes a wide range range of professional real-time VST effects and (VSTi) virtual instruments.
- Supports virtual instruments with multiple outputs (VST 2.0 standard)

Sample Editing

- Destructive Sample Editor with offline process history.
- Sample accurate zoom and in-place editing of audio directly in the Project window.
- Automatic tempo matching of audio loops with integrated Slice and Stretch functions.

MIDI

- Unlimited MIDI tracks and multiple MIDI recording modes: Cycle, Mix, Overdub, Step, Punch.
- Sample-accurate timing for MIDI-events. For ease of use the displayed musical resolution is set to 480ppq. If you need a higher resolution for your editing needs you can simply switch to sample resolution.
- A wide range of MIDI editors are provided including: Key (piano keyboard); Drum; Event List; SysEx and Logical. Cubase SX adds professional Score editing.
- Supports high-precision MIDI timing when using an LTB compatible MIDI interface such as Steinberg's Midex 8.
- A Tempo Track allows you to control song tempo using a graphic editing interface.
- Sends MIDI clock and send/receive MIDI timecode (MTC).
- Extensive control surface support including Steinberg's own Houston, Mackie Control, Yamaha Digital Mixers, Radikal Technologies SAC-2.2 and more.

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CUBASE SL • CUBASE SX

The Project Window

The Project Window – provides an overview of the entire project area and allows real-time recording, playback placement, editing and automation of audio and MIDI. You can move, nudge, fade in, fade out, change volume, or crossfade parts directly from the Project Window with great ease and efficiency.

Point and click on these icons to access the five most common editing windows within a project – the inspector pane, project overview, object parameter list, audio pool window and project console window

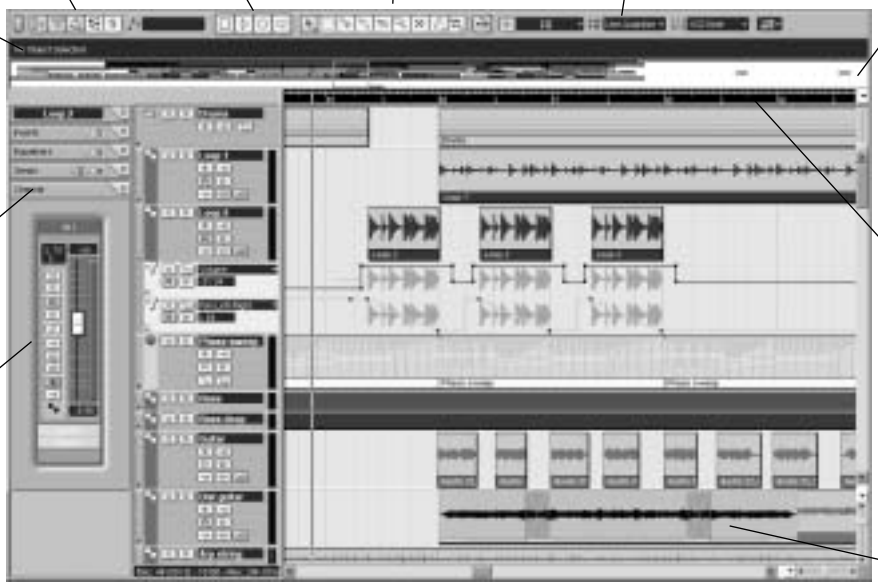
These tools allow you to edit and arrange single or multiple objects within the project window:
- *The Arrow Tool – Sizing, Time Stretch, Sizing Moves content*
- *Range Tool (both A and B selection)*
- *The Pencil Tool lets you draw controller and automation data (freehand or grid based) using 5 Modes – Draw, Line, Sine, Triangle, Square*
- *Scrub Tool has 2 Modes – tape-style Scrub and position Play*
- *Other tools include – • Glue Tool • Erase Tool • Magnifier Tool • Mute Tool*

Project Overview
Provides quick and easy access to any location within the project window – the smart cursor allows you to click and drag on the bottom of the overview to navigate to the specific area of the project you want to access or click and drag at the top of the overview to zoom select a new zoom value of the project window.

The most common Transport Controls – stop, play, record and loop

The Grid and Quantize is adjustable down to single sample resolution
There are 5 Grid modes to choose from – Grid, Event, Magnetic, Cursor and Shuffle

All Event based information such as Part Name, Start and End Points, Length, Offset, Snap, Volume and Fade-info is shown within the object parameter 'Infoline'.



Track Inspector
Provides instant access to any parameter of the currently selected audio or MIDI track.

Timeline Ruler
Option Positions and lengths displayed as:
Bars+Beats – Bars|beats|sixteenth notes and ticks (120 per 1/16 note)
Seconds – Hours, minutes, seconds and milliseconds
SMPTE Frame (Hours, minutes, seconds & frames) – 24, 25, 29.97, 30 fps as well as 29.97 and 30 dfps
Samples – Samples

Each component of a channel's mixer is displayed as a folder that can be expanded to reveal that components parameters – add insert-effects, adjust aux sends, pan and volume and more, without calling up the mixer

Non-destructive, editable crossfades

Supported Audio Formats

- Import of AIFF, AIFC, WAVE, Broadcast WAVE, WMA, MP3, REX files as well as SDII on Mac.
- Export audio to MP3, RealAudio G2, AIFF, Broadcast Wave and WAVE, as well as Windows Media Audio and Real Audio V5 on PC and SDII on Mac.
- Import/export files with sample rates of up to 96kHz depending on format.

Supported Video Formats

- MOV, QT, AVI, MPG, MPEG

Video Functions

As working in sync with the picture is an essential demand for film scoring, Cubase does not only allow playback of video files within a video window, it also provides you with a thumbnail video track for fast and easy composing to the picture and placing musical events at takes and frames.

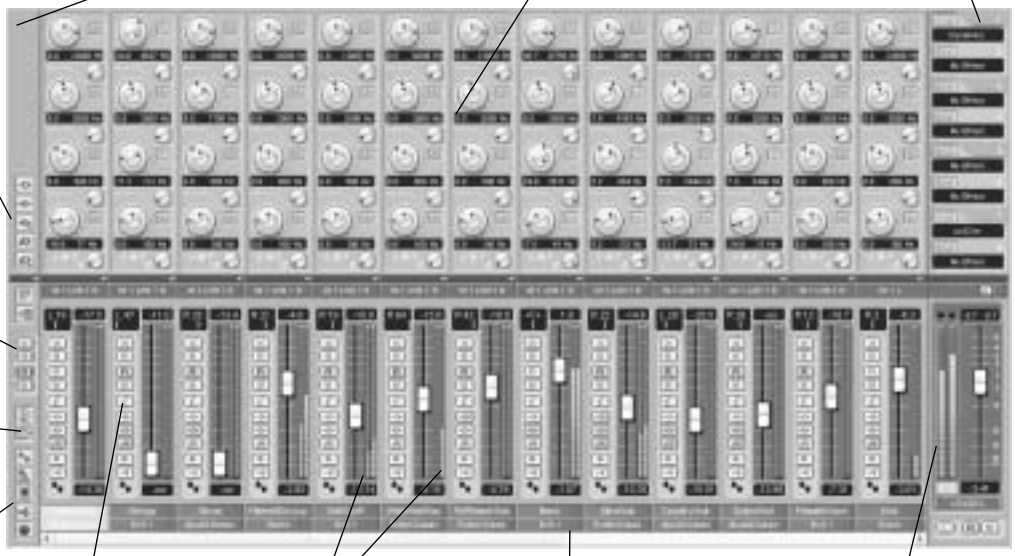
Unlimited Undo/Redo with Offline Process History

Off-line processing of individual audio files allows effects to be 'stamped' onto them layer after layer. The off-line process history, allows you to jump back to any individual process such as the reverb and edit it, remove it or replace it with another effect.



CUBASE SL • CUBASE SX *continued*

Mixer Channels



EQ, Insert, Aux sections of Channel strip can be bypassed fast and individually

Each channel within the mixer has 4-bands of parametric EQ, 8 insert points for plug-in effects and 8 aux sends

The Master Section features 8 insert slots plus dithering using Apogee's UV-22 HR algorithm

The extended view adds a top zone to the standard size mixer that adds an upper zone where you can switch between views for EQ and aux-send or insert effects per channel or globally.

Global Solo/Mute and automation Read/Write buttons

Complete channel setups can be copied and pasted between channels or even saved to disk and loaded into new projects – a great way to create a custom library of multiple effects or virtual instruments with effects.

Select the type of channels you want to view in the mixing console – audio, group, virtual instrument, ReWire and MIDI channels

Click the Edit button to open the Channel Settings window (detailed below)

Mixer channels can be switched between mono or stereo

Any combination of mixer channels can be freely grouped or sent to one (or more) of up to 64 sub group channels.

The response time of the meters is adjustable (fast/slow/hold)

Channel Settings Window



4-band fully parametric EQ with variable frequency (20 to 20k Hz), gain (±24db) and Q per band–
Bands 1 and 4 are switchable between Shelf, HPF, LPF and Bell
Bands 2 and 3 has a Bell curve

These 8 Aux Sends can be routed to the FX rack, Groups or Bus outs.

Pan control

Mute/Solo

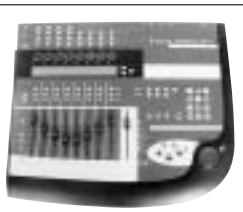
Read/Write Automation

Ultra-smooth volume faders with 15-bit resolution and 2 decimal precision

Up to 8 effects inserts are available for each audio channel

Save and load EQ presets

Hardware Control Surface Support



- Of course, Cubase is fully compatible with Steinberg's Houston controller surface which allows you to grab hold of the virtual world of Cubase with physical knobs, buttons and faders.
- Features such as transport control, scrub, fast forward, rewind, track muting, track arming, control over plug-in parameters and full moving-fader control over your mix are just a few of the possibilities.
- A wide number of 3rd Party remote controllers are supported including devices made by Mackie, JL Cooper, CM Automation, Radikal Technologies, Roland, Tascam, and Yamaha.

B&H PHOTO - VIDEO - PRO AUDIO

ONE HOUR FREE PARKING
AT 349 W. 34th STREET (with purchase of $100 or more)

CUBASE SL • CUBASE SX *continued*

VST Realtime Effects Processors

- Steinberg's Virtual Studio Technology (VST) interface allows the seamless integration of virtual effect processors and instruments into your digital audio environment. These processors run the gamut from software emulations of classic hardware effect units and instruments to never-heard-before creative processors. All functions of a VST effect processor or instrument are directly controllable and automatable.
- A complete suite of Steinberg VST effect processors is included with Cubase SL and SX and because VST is an open standard, new virtual effect processors and virtual instruments are constantly being developed by Steinberg as well as numerous third party companies including Antares, Waves, Waldorf, Native Instruments, IK Multimedia and many more.
- DirextX plug-in support is also available on the PC

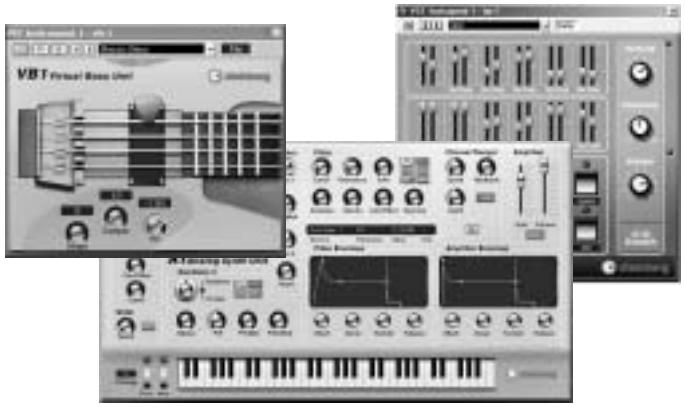
Included VST Effects Processors

Multitap Delay	Flanger	Reverb
Classic Delay	Phaser	Reverb 32
Dynamics	Rotary	Vocoder
DeEsser by SPL	Chorus	Phatsync
Overdrive	Metalizer	Bitcrusher
QuadraFuzz	Transformer	Ring modulation
Chopper	Grungalizer	SMPTE Generator
datube	Symphonic	6 to 2 Mixer

VSTi (VST Instruments)

- VST Instruments are software synthesizers and sampler that are loaded into Cubase just like standard VST plugins.
- VST instruments can be triggered via MIDI, just like their hardware counterparts, and their audio outputs appear on separate channels in the Mixer, allowing you to add effects or EQ, just as with audio tracks.

Included VST Instruments



- Three VST instruments are included and numerous others can be purchased separately from Steinberg and third party manufacturers.
  A1 – Analog Synthesizer Unit developed by Waldorf
  VB1 – Bass Emulation Unit
  LM7 – 24- Bit Drum Sampler Unit



MIDI Plug-in Effects



- A number of realtime MIDI effect plug-ins are included for transforming and generating MIDI events in a variety of ways.
- MIDI effects can be applied in real time to the MIDI data played back from the track or can be inserted while playing live on your MIDI keyboard - an inspiring performance tool
- Just like any other parameter in the Cubase mixer, MIDI plug-ins can also be fully automated.
- MIDI Plug-ins Effects Include: Arpache (Arpeggiator), AutoPan, Chorder (create chords from single keys), MIDI Echo, Microtuner, Quantize, Step Designer (step sequencer), MIDI Compressor, and many more.

ReWire 2.0 Support

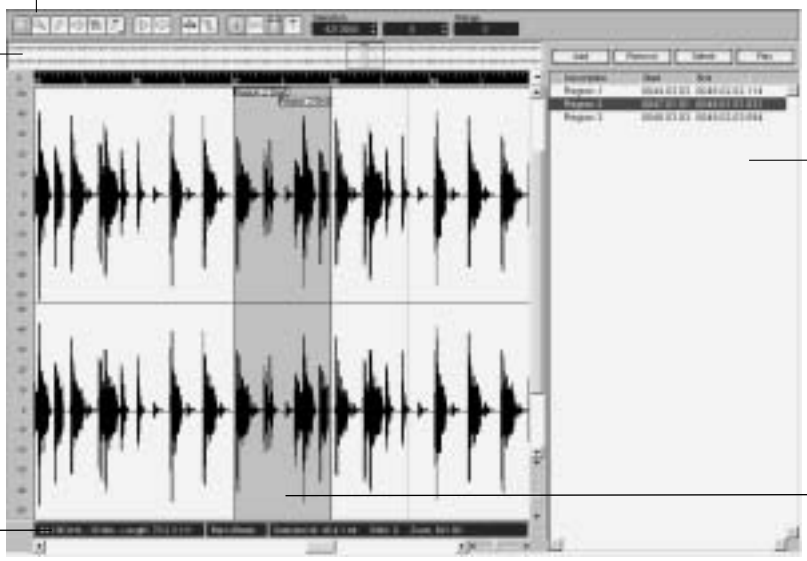
Stream up to 64 audio channels between ReWire compatible applications, such as Propellerhead's Reason and Rebirth, and Cubase with sample accurate synchronization. Audio outputs from the ReWire compatible application are connected via Rewire to the VST mixer, allowing VST Effects, EQ and Dynamics to be applied in real time. MIDI-streaming between applications — the ReWire application shows up in Cubase as a MIDI out port, ready to trigger. Use Cubase to control the ReWire apps transports.

Sample Editor

Even though you can do most of your editing directly within the Project window, an integrated Sample Editor is included for editing mono and stereo audio files. The Sample Editor allows viewing and manipulating of audio data at the Audio Clip level. Functions such as cutting and pasting, removing or drawing audio data is fully "non-destructive", in the sense that you can undo changes or revert to the original versions at any point, using the Offline Process History. Any audio clip can be permanently edited and changed with full backup, plus multi-level undo and redo. A number of integrated non-destructive audio processes are provided for optimizing and even resynthesizing audio files: Acoustic Stamp, Crossfade, Envelope, Fade-Ins and Outs, Gain Change, Normalize, Phase Reverse, Pitch Shift and Timestretch, DC Offset removal, Reverse and Stereo Flip.

Sample Editor tools include: the selection tool for creating regions within an audio clip; the magnifier tool; The Pencil tool allows you to to manually edit out a spike or click; and the Scrub tool which allows you to locate positions in the audio by playing forwards or backwards at any speed.

The Thumbnail display provides an overview of the whole Clip.



All regions created from an audio clip are shown with their name, start and end positions in the Regions List display. You can audition regions and drag and drop them directly from this list into the Project window.

Regions can also be exported as audio files

The Info Line shows all relevant information about the edited Audio Clip.

Regions allow you to define important sections within an Audio Clip. Regions can be dragged into the Project window from the Editor or the Pool.

THE LOOP EDITOR

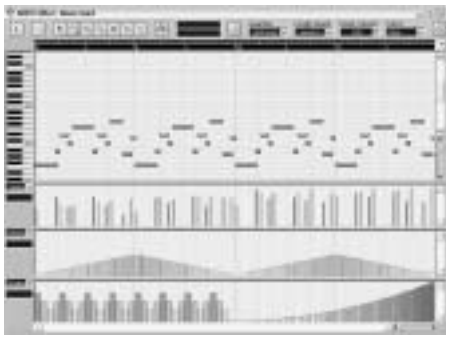


- The Loop Editor allows you to automatically match the of an audio phrase or loop to your song's tempo by applying time compression and expansion to short audio-segments contained with the audio file.
- Audio segments can be created automatically or by manually — the phrase or loop is analyzed and then segments are generated using the audio file's transients as markers.
- You can then make tempo changes to your song in real-time and your audio playback will follow.
- Use odd meters and loops of unlimited length when mixing.
- You can combine a number of loops with different feels and tempi and conform them to your song's tempo.
- Match the groove of MIDI tracks to looped audio or vice versa. Quantize the groove of looped audio.
- Re-arrange grooves by exchanging single samples inside your loop.
- The Loop Editor is also provides an easy auto cutting and match quantizing of single words from vocals takes.

CUBASE SL • CUBASE SX *continued*

MIDI Editing

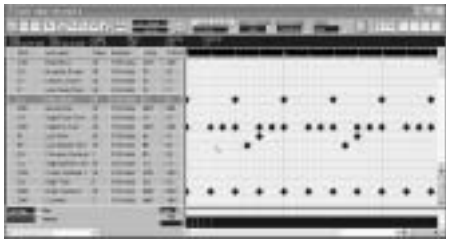
Key Editor



- The Key Editor / Piano-roll Editor features many standard functions for manipulating MIDI note and event data. You can Draw, Move, Copy, Delete, Mute, re-Size, Transpose and change Velocity for note-events.
- The Multi-Lane Controller Editing functions allow simultaneous editing of various controller data sets at the same time.
- The Mathematical Curve Function (such as parabola, sine, triangle, square) allows you to draw precise Controller data curves quickly and easily.
- Step Recording allows recording events off-line by inputting notes via MIDI-in at the start position and automatically stepping these to the next Grid point (e.g. 1/16th). This allows the easy creation of musical lines that are difficult or impossible to play.

Drum Editor

- The Drum Editor features standard editing functions (Draw, Move, Copy, Delete, Mute, Size, Transpose note-events, Velocity and Controllers) as well as Drum Map support allows you to assign sound names (eg. kick, snare) to notes.
- Each sound or notepitch can have its own settings for Quantize, Output and Channel.
- The Drum Editor also provides access to the same Multi Lane Controller Editing, Mathematical Curve and Step Recording functions found in the Key editor.

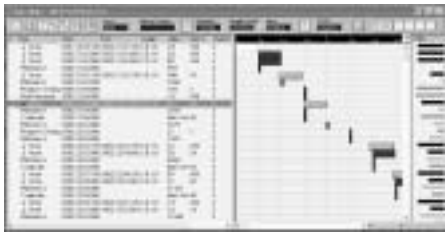


Logical Editor

- The Logical Editor allows you to find certain MIDI events, based on defined criteria and conditions. You can then manage and edit this data using the list of available actions.
- Applications range from simple MIDI event filtering to complex tasks such as changing the scale of a piece of music from minor to arabian.

List Editor

- The List Editor allows all data (MIDI, Audio events, Automation, etc.) to be edited numerically.
- It provides an overview of the whole project so that you can easily browse through all of the project's tracks. This allows very precise control when adjusting the 'fine detail' like sample-accurate positioning of multiple data types.
- The event display shows the events graphically while the value display shows the "value" of each event, allowing for easy viewing and graphical editing.
- The information available in the List Editor Includes: Event Type; Start and End position; Length; Data 1 and Data 2 (eg. pitch and modulation); Channel and Comment.
- Single and multiple events can be muted and unmuted using the dedicated Mute Tool.



Steinberg Technology

ASIO

- ASIO (Audio Stream Input/Output) is a high performance, low latency audio driver architecture that forms the backbone of Steinberg's Virtual Studio Technology. ASIO was developed to deliver a truly professional audio recording solution — one that supports variable bit depths and sample rates, multi-channel I/O and synchronization — all within a native computer environment.
- ASIO is a supported standard by many leading audio hardware manufacturers including MOTU, M-Audio, Tascam, Edirol, Steinberg and many more.

VST System Link

- VST System Link is a platform independent communication protocol that provides sample accurate networking of several computers running VST System Link-compatible host applications including Cubase SX, Cubase SL and Nuendo. It enables synchronization, transport control and audio data between two or more workstations over standard digital audio cabling systems such as ADAT, TDIF, AES/EBU or S/PDIF.
- Expandability, Flexibility and connectivity for your virtual studio — exploit the DSP resources of multiple computers within a single project.

Accurate MIDI Timing with LTB

- LTB (Linear Time Base) is a multi-channel communication protocol which offers sub-millisecond MIDI timing accuracy.
- LTB is a MIDI Time Stamping technique that bypasses the computer's operating system and thus significantly reduces latency.
- LTB is utilized by the MIDEX series of hardware MIDI USB interfaces to offer the lowest latency of any MIDI interface available.
- Supported by Cubase SL and SX as well as Nuendo.

CUBASE SL • CUBASE SX *continued*

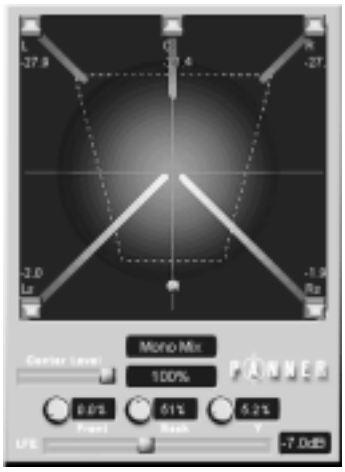
Cubase SX Adds-

TrueTape Recording Technology Tape Saturation



- TrueTape is a unique Steinberg technology that emulates the behavior of a professional analog tape recorder.
- While digital audio recording has a number of benefits, some may perceive digital sound to be somewhat "sterile" and "cold" compared to high quality analog recordings. The TrueTape algorithm remedies this by recreating the desirable warmth and compression sounds of analog tape saturation at the recording stage using Cubase's high resolution 32-bit floating point format.
- The Drive control allows you to adjust the amount of tape saturation effect to your liking.

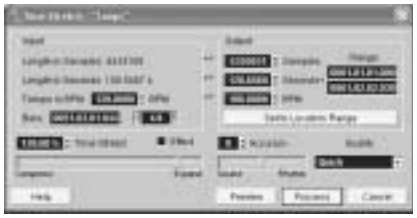
Surround Sound



- Cubase SX supports up to six-channel surround sound with a configurable speaker set-up - allowing the user to adjust to his specific studio configuration.
- Supported surround formats in the Master Setup include: Stereo; Quadro; LRCS; Standard 3/2; and 5.1

Advanced Time Stretching

- The Time Stretch function allows you to change the length and "tempo" of the selected audio, without affecting its pitch.
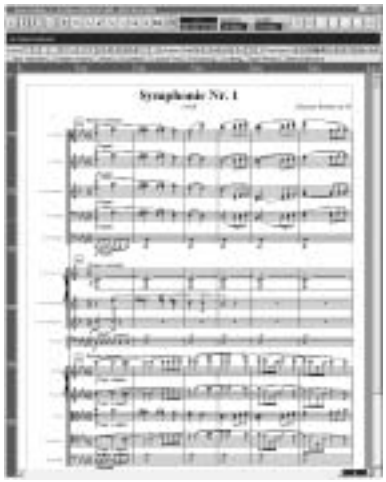


- There are five different Quality modes available for time stretch: Quick, Standard, High, MPEX, and Drum.
- The MPEX Quality mode is based on Prosoniq's (Minimum Perceived Loss Time Compression/Expansion) algorithm which uses an artificial neural network for time series prediction in the scale space domain to achieve high end time and pitch scaling.
- Drum mode is a special algorithm developed by Spectral Design, optimized for processing rhythmic material.



Apogee UV22HR Dithering

- The UV22 HR is a high quality dithering plug-in based on an advanced word length reduction algorithm developed by Apogee.
- You can use the UV22 HR plug-in whenever you need to take your audio from the 32-bit float resolution within Cubase to lower bit resolution, especially when exporting/converting to 16-bit resolution for audio CD burning.

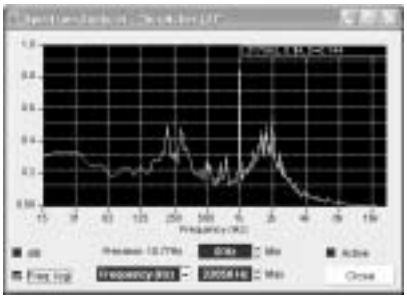
Complete Scoring



- Cubase SX includes all of the scoring and playback features of Cubase Score VST, which for many years has been one of the industry standard software applications used for film soundtrack production and music education. This is because of its perfect integration of score writing, sequencing and music production.
- 32 staves per page
- 8 voice polyphonic
- Page layout and editing
- Drum notation & guitar tablature
- Over 100 symbols

Spectrum Analyzer

- The Spectrum Analyzer function analyzes the selected audio, computes the average level distribution over the frequency range and displays this as a two-dimensional graph, with frequency on the x-axis and level on the y-axis.



CUBASE SL • CUBASE SX *continued*

Features	*Cubase SL*	*Cubase SX*
Interactive, real-time graphic editing	✓	✓
Unlimited Undo and Redo functions	✓	✓
VST System Link	✓	✓
Best compatibility with Houston controller	✓	✓
Video track with Thumbnail Preview	✓	✓
Various Project Templates included	✓	✓
Cubase VST 5.x song import	✓	✓
Native audio – no expensive additional outboard equipment required	✓	✓
Simultaneous playback of hundreds of audio tracks *	✓	✓
Adaptive and configurable track mixer	Wide and narrow view	Wide, narrow and freely configurable extended view
Supports 6 channels surround sound	–	✓
Effect Insert Slots per Channel	5	8
VST Instrument Slots	16	32
Includes VST Instruments such as A1 Virtual-analog Synth Unit powered by Waldorf	✓	✓
Includes professional real-time VST effects	✓	Plus DeEsser by SPL
Sample accurate automation with automation tracks for every audio, group track and plug in	✓	✓
Automation modes	Touch Fader	Touch Fader, X-Over and Autolatch
Dithering	Apogee UV22	Apogee UV22HR
TrueTape recording technology for tape saturation	–	✓
Up to 64 sub-groups with the same EQ and effect features as a normal audio channel	✓	✓
Offline Process History	For built-in audio processes	For built-in audio processes and all plug-ins
Very quick and efficient parts bouncing	✓	✓
Huge selection of integrated audio processes	✓	✓
Spectral Analysis	–	✓
Statistic Functions	–	✓
Audio/Tempo analysis and Audio/Groove analysis	✓	✓
Slice & Stretch for audio loops, automatic tempo matching	✓	✓
Pitch-shifting / time stretching algorithm	3 different modes	Plus MPEX
CD audio grabbing	✓	✓
ReWire 2 support	✓	✓
Full MP3 Fraunhofer Encoding	Available as an upgrade	✓
Recording modes: Mix, Overwrite, Normal, Merge	✓	✓
Editors: Key (piano keyboard), Drum, List, SysEx, Score, Logical	✓	✓
Professional Score layout and printing functions	–	✓
Supports LTB for high-precision MIDI timing	✓	✓
Soft Quantize	✓	✓
Supports Steinberg MIDI plug-ins	✓	✓

**CPU dependent*

STEINBERG

HOUSTON

COMPUTER AUDIO

1002

Motorized Control Surface for Cubase and Nuendo

A MIDI/USB remote controller designed specifically for the VST audio engine, Houston provides access to potentially every parameter available in VST windows. It features a total of nine 100mm touch sensitive motorized faders, eight rotary encoders with LED position indicators and a matrix of buttons bringing total hands-on mixing within Nuendo or Cubase, without having to use the mouse or PC keyboard. Additionally it offers a large LCD display, transport controls, and a jog & scrub wheel. A numeric keypad is available for entering values and for selecting setups and marker positions. The ultimate controller for Steinberg's virtual studio, Houston is also extremely fast and flexible with all basic parameters displayed at the push of a single button.



FEATURES

- Houston uses an all-for-one or one-for-all layout meaning that both the display and related rotary encoders are showing the same parameter for all eight channels or eight related parameters for one channel.
- Parameters for VST audio effects and VST Instruments can be displayed, edited and automated from the Houston's controls.
- A compact desktop unit (same width as a standard PC keyboard), Houston connects to the host computer via MIDI or USB.
- Illuminated function matrix clearly indicates the current display and all basic VST parameters. If a set of related parameters require more than a set of eight controls, the page up and page down buttons let you jump to the additional parameters. And since you always stay in the context of the selected function group, it isn't necessary to repeatedly press buttons to scroll through all available parameters. With two button pushes you gain access to virtually all VST function groups.
- Houston supports the mixer views and windows sets of both Cubase and Nuendo. It is possible to open and close windows completely remotely, and to change the currently addressed set of eight channels—and these can be any of the VST audio channels regardless of whether they are audio, group or synthesizer channels.
- Song and edit buttons to easily save a song or revert to the last version, undo/redo the last action, confirm, cancel and deny actions in open dialogs and windows.
- Number pad for general purposes plus buttons for quickly accessing windows sets, marker positions, etc. For even greater comfort, the ZAP button lets you toggle between the two last functions.
- Fader section with nine motorized faders. A number of option buttons let you switch between the different target faders within the host application and select fader sets. You can also jump back and forth between the available sets of eight channels each.
- Transport Controls and Jog Dial for full and easy navigation within your song.
- Rear panel with MIDI trio and USB port. Houston lets you freely choose whether it should communicate with your host application via MIDI or USB.

What are the differences between Cubase SX and Nuendo?

Cubase SX is clearly a music creation and production program, with audio and MIDI features specifically designed for these activities such as the Drum Editor, Score Editor and a newly designed Audio Loop Editor. Multi-lane Controller Editing is provided offering complete control over the playback characteristics of all MIDI and VST-I equipment used. MIDI plug-ins and enhanced MIDI processing functions are now standard and we have built in the support for loading up to 16 Virtual Studio Instruments simultaneously. Extensive groove and swing functions are designed specifically for easy handling of rhythmic musical elements and excellent automation control is built in so you can create the perfect final mix.

Nuendo, on the other hand is designed as a media-production system. It can handle post-production, DVD authoring, music recording and many other types of specialist production tasks. It features tools such as Sony 9-pin support, OMF-Import, replace audio in video file, get audio from video file, Premiere EDL import and independent project pools, to name just a few. Other features are the Matrix Encoder/ Decoder plug-in as well as support for Dolby and DTS Encoders.

STEINBERG

MIDEX 3 / MIDEX8

USB MIDI Interface with LTB Technology



The Midex 3 and Midex 8 are hardware MIDI solutions for those who place emphasis on absolutely accurate timing. Designed for VST-based studios who use external sound generators to supplement their range of virtual instruments, the portable Midex 3 has one MIDI in and three for 48 MIDI channels, while the Midex 8 with its eight MIDI inputs and eight MIDI outputs allow up to 128 MIDI channels to be addressed. Both don't need an external power supply, as their interface's power is supplied by the USB bus.



Numbered MIDI port LEDs provide a clear overview over your MIDI set-up. Midex 3 and Midex 8's comprehensive range of functions is rounded off by the relay-controlled MIDI Thru function as well as an integrated cable tester. Several Midex 3s or Midex 8s can be combined to build larger MIDI systems.

- Several Midex 3s or 8s can be combined to build larger MIDI systems
- LTB protocol – Linear Time Base prevents audible timing lapses and guarantees accurate sending of MIDI events to within a single millisecond
- Powered by USB – no external power supply needed (Midex 8 has a connection for external power unit – just in case)
- They are automatically detected once they are connected to the computer. They can also be connected to a computer which is already switched on thanks to USB
- Incorporate MIDI Thru function
- They include USB cable and driver CD with online manual in 6 languages
- Numbered MIDI port LEDs for a clear overview of your music project

Midex with LTB Technology — Superior timing and more than just an interface!

Today's complex operating systems and computer configuration can lead to small timing inaccuracies in MIDI event sends—making precise MIDI timing difficult. This can be up to 5 ms in extreme cases – small but audible. Depending on musical taste or the user's individual expectations, these inaccuracies can compromise the quality of songs. Additional delays caused by sound generators are also increasingly being seen as unacceptable among producers of today's dance music or any other musical styles where maximum 'tightness' is deemed particularly important. In the age of virtual instruments, precision MIDI timing is essential because VST instruments operate with sample accurate timing. When they are used together with hardware sound modules, timing inaccuracies can become painfully apparent.

The solution is Steinberg's LTB technology, a high-speed MIDI protocol which gets around an operating system's limitations in a simple yet elegant way. With LTB, every MIDI event is sent to the MIDI interface ahead of time, together with information describing the exact point of time when it is to be sent out from the interface. Here it is stored and released from every Midex port to the respective sound generator at precisely the right time. This guarantees a timing accuracy for single MIDI events of under a single millisecond - on every port. This is made possible by a micro sequencer built into all Midex devices. It offers a timing accuracy which was previously only possible when working with old analog or hardware MIDI sequencers. LTB is incorporated into all Steinberg Midex series MIDI interfaces as well as Nuendo and the latest versions of Cubase.

- Numbered MIDI port LEDs for a clear overview of your music project
- Integrated MIDI cable test function
- Steinberg's latest drivers offer multi client capability, allowing you to address single Midex inputs and outputs with different programs. This lets you combine stand alone applications such as Reaktor and Cubase with your MIDI hardware – even if one program uses LTB while the others do not.
- They offer secure and reliable use with Macintosh OS as well as Windows 98SE, ME, 2000 and XP. A special driver optimization gets around limitations in certain Windows versions which allowed only 10 MIDI ports to be used simultaneously.

Midex 3 Only

- 3 separate MIDI outputs and one MIDI input – up to 48 different MIDI channels can be addressed per Midex 3
- Sturdy Macrolon housing

Midex 8 Only

- 8 separate MIDI outputs and 8 MIDI inputs –up to 128 different MIDI channels can be addressed per Midex 8
- Easy access to extra MIDI input and output on the Midex 8's front side
- Fits into every professional 19″ rack – or can also be used simply as desktop unit

STEINBERG

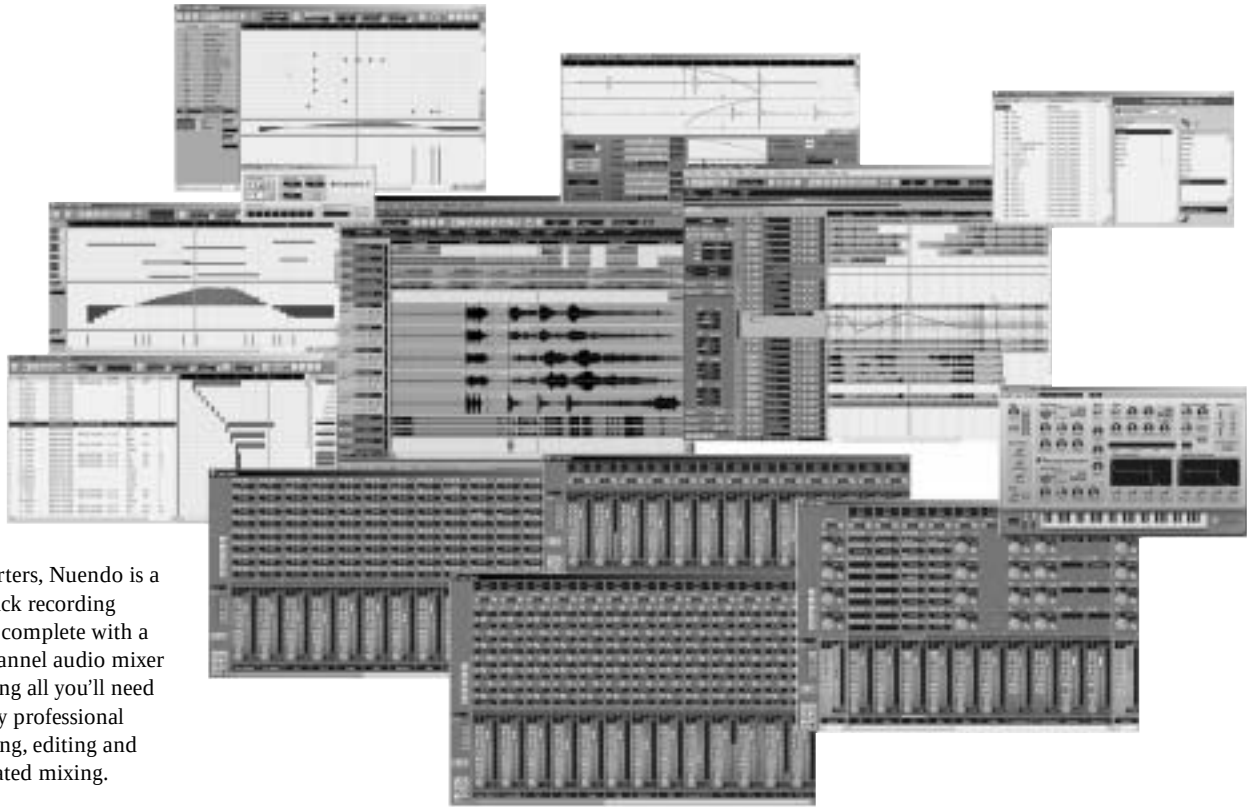
NUENDO 2.0

Media Production System



Nuendo 2.0 is a high-end digital audio workstation and media production system designed specifically for the demands of modern multimedia productions. Regardless of whether you are working on a music or post production project, in broadcasting or video—Nuendo 2.0 offers a solution which will perform to the highest expectations—and more cost-effective than any other system.

Created to handle any kind of job within the media industry, all recording, editing and mixing functions were developed from scratch, allowing Nuendo to offer the most advanced studio technology. And Nuendo's extensive interface, import and export functions guarantee perfect integration into every studio environment regardless of genre—composition, film, broadcast, music, postproduction, surround, games and multimedia.



For starters, Nuendo is a 200-track recording facility complete with a 200-channel audio mixer including all you'll need for fully professional recording, editing and automated mixing.

As Nuendo supports mono, stereo or surround format (up to eight discrete channels deep) your creativity is never inhibited. From input to final mix, Nuendo 2.0 is all about surround, with multi-channel architecture through the entire signal path. Every input, audio track, effect, group and output now offers up to 12 discrete channels, ready for full-scale 5.1, 7.1, or even 10.2 productions. To make routing in the project even more transparent, you can customize multichannel input/output configurations and switch between them with a single keystroke. Several input and output buses can be utilized at the same time, with any type of configuration possible - mono, stereo or any of a wide range of surround formats - and any track can be routed to and from any of these buses. Nuendo 2.0 even allows switching between multiple monitoring configurations (speaker arrangements) and can simulate a wide variety of end user monitoring environments.

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ONE HOUR FREE PARKING
AT 349 W. 34th STREET (with purchase of $100 or more)

NUENDO 2.0

Nuendo 2.0 offers complete flexibility in mixing and signal routing. Its 32-bit floating point mixer features multiple multichannel input and output buses. This allows for recording in either split or interleaved surround audio file formats and makes managing a surround project easier than ever before - Nuendo always keeps track of which part of the signal chain is multichannel and which is simply stereo or mono. When routing signals, Nuendo even adds effect return channels that allow effects to be added to the input signal while recording, as well as full delay compensation throughout the signal path.

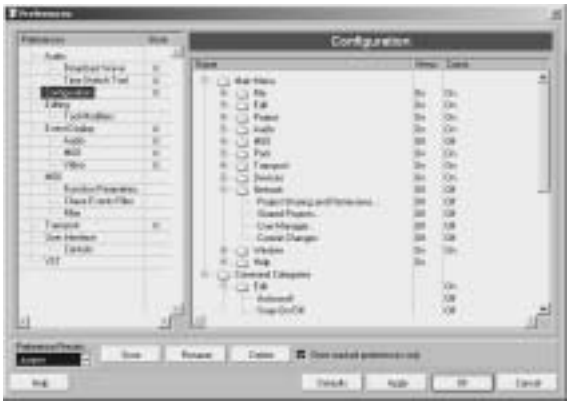
Of course, Nuendo incorporates VST System Link—Steinberg's revolutionary technology that allows users to increase the realtime processing and mixing power for any project by simply adding as many Macs, PCs, desktops or laptops to the system as are needed. But Nuendo 2.0 doesn't stop there. An array of powerful new networking capabilities allows the transferring of tracks and events in a network over TCP/IP LAN. Each user in a network gets his user ID and read/write permissions for each TCP/IP project. The standard network access built into Nuendo 2.0 supports recording to, as well as copying and playing back from dedicated servers or other workstations: all that's needed is TCP/IP LAN. Combining separate Nuendo 2.0 projects is no problem, either - a Merge Project function fuses tracks from one project into another.

Nuendo 2.0 also allows the flawless import of Cubase SX projects, and support is provided for a huge range of export formats in order to exchange projects with systems from other manufacturers. Nuendo 2.0 can also import numerous file formats such as AES31, Open TL 3.0 and OMF. Finally, Nuendo 2.0 allows you to set up a project the way you want it. If there is no need for MIDI functions in a particular mix, for example, all MIDI controls can be hidden for each track to suit your personal way of working. You can hide functions you don't use from Nuendo's menus - making for a better overview and faster response - and you can even define which buttons and controls you want to see on your tracks, and in which order.

Versatile and Flexible

Set up a project the way you want it. All program menus are user-configurable, enabling you to hide features you don't currently need for a specific kind of production. For example, if you don't need MIDI functions for a 10.2 surround mix, you can hide all MIDI menu entries, disable the corresponding key commands and assign these preferences to a template. Do you really need to have video-related menu entries if you are using Nuendo for music production? Simply hide them and design the interface that you need yourself. All hidden features are still waiting in the background in case you will need them in the future.

But Nuendo does not stop there. You can also configure the controls for each track to suit your personal way of working. If, for example, you are engaged in audio recordings, Nuendo can instantly be adapted as a pure audio tracker with only a record and a monitoring button on each track. After finishing your recordings you can "unhide" additional functionality as it is needed or switch to a template which immediately gives you access to all track controls again.



User Presets

Nuendo 2.0 allows for the storing and recall of user preferences presets. This allows you to recall different system settings for various working situations but also take your preferred settings with you and apply them to another system when you are working on a Nuendo system at another location.

Define Your Project

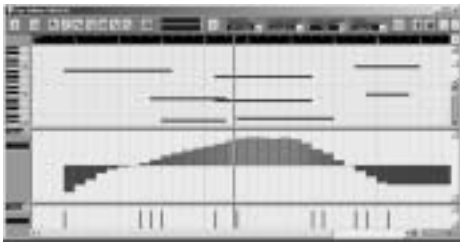
Nuendo 2.0 offers a wide range of timeline formats for just about any situation in audio production, including samples, beats/bars, seconds, feet:frame rates of 16mm and 35mm as well as a freely definable frame rate. Using multiple ruler tracks with various timeline formats, you have the correct position in overview for any of your audio, video or MIDI events. Also, the new mixer allows you to show or hide track classes and channel strip sections to adapt to your visual needs.

NUENDO 2.0 *continued*

Recording

- Pristine 32-bit audio quality with sample rates of up to 192 kHz will capture every nuance of your source audio. Processing audio internally at 32-bit floating point gives you a theoretical headroom of several hundred dB, making it practically impossible to introduce digital clipping.
- Offers a choice of a variety of record formats, including Wave, Broadcast Wave, AIFF and Wave64, for long duration recordings.
- A variety of record modes suit any approach, with the stacked record mode allowing the compiling of takes directly in the Project Window.
- Innovative multi-channel architecture lets you record not only in mono, stereo and 5.1 but in all common surround formats up to 10.2. Record your surround material either as individual files or interleaved as one file, in the destination folder of your choice.
- Recording to timecode is easy. The online recording function puts Nuendo in record mode as soon as a valid timecode signal is received.

MIDI Features



Extensive capabilities, making it easy to set up and configure MIDI devices for use with Nuendo.

- The large range of MIDI editors (Drum, Logical, SysEx and List) make editing MIDI information more intuitive than ever before.
- MIDI plug-ins such as Quantizer, MIDI Echo and Compressor ensure more creative possibilities and flexibility, and are all directly reachable from the Project Window.
- The Nuendo automation system also extends to all MIDI parameters, letting you draw MIDI automation data with any of the mathematical shape tools. Multiple controller lanes within the editors give you a clear oversight over your MIDI data, as all data can now be displayed on the same page.

Mixing



From input to final mix, Nuendo 2.0 is all about surround, with multichannel architecture through the entire signal path. Every input, audio track, effect, group and output offers up to 12 discrete channels, ready for full-scale 5.1, 7.1, or even 10.2 productions. To make routing in the project even more transparent, you can customize multichannel input/output configurations and switch between them with a single keystroke.

- Several input and output busses can also be utilized at the same time, with any type of configuration possible - mono, stereo or any of a wide range of surround formats - and any track can be routed to and from any of these busses. You can even switch between multiple monitoring configurations (speaker arrangements) and simulate a wide variety of end user monitoring environments.
- The mixer itself is user-configurable, so you can choose between a variety of display options. Above the normal mixer with the fader section you can display either insert effects, effect sends, EQs or an additional view with input and output settings including gain change and phase shift per channel. The mixer is also available in a narrow view for simultaneous display of many channels.
- Nuendo 2.0 offers the most flexible mixing experience in digital media production. Its 32-bit floating point mixer features multiple multichannel input and output busses. This allows for recording in either split or interleaved surround audio file formats and makes managing a surround project easier than ever before — and Nuendo keeps track of which part of the signal chain is multichannel and which is simply stereo or mono.
- Provides complete flexibility in signal routing to and from virtual effects for each and every track. It even adds effect return channels which feature the full parameter set of regular channels and of course the complete mixer has full delay compensation throughout the complete signal path.

Networking

With VST System Link and the option to integrate IP network collaboration, Nuendo 2.0 features a huge range of functions dedicated to maximize computer processing potential and manpower for your project. Using any of the two technologies alone or combining them has can vastly increase efficiency in your project workflow.

—With VST System Link, not only can CPU intensive tasks of larger projects like effect processing, video or virtual instruments playback be outsourced to selected exclusive computers, but large projects can be allocated to various computers as well. This permits simultaneous playback of hundreds of audio and MIDI tracks from a range of different computers in sample-accurate synchronization.

—Unique TCP/IP functionality allows true networking on any kind of project, from post-production to music, from multimedia to game sound design. This networking technology supports the full range of editing tools over a network, on projects, audio, MIDI and even video tracks. It allows users to connect multiple Nuendo workstation computers via standard LAN network cards. An entire project or just selected tracks can be opened on multiple computers for other members of the network to access and edit.

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NUENDO 2.0 *continued*

Supersonic Editing

Nuendo 2.0 has one of the most powerful editing engines around, with unlimited undo/redo as standard. Not only can you undo cuts, fades or other basic edits, but also bouncing, offline processing and removal of complete tracks can be undone. The Edit History window lists all actions made on your project down to every single event, ready to undo or redo to the point where you want to work from.

Every single audio file shown in the pool or used in the project window has its own offline process history. All processing carried out on a file can be removed, changed or replaced by another process or plug-in, no matter how much processing has taken place in the meantime. All later processes will be reprocessed automatically for you in the blink of an eye.

Supports various kinds of editing styles, depending on your work style. Traditional cuts, nudging, trims and fades can be made using a variety of tools: object based, range based or using a jog wheel from any of the supported remote controllers. Mouse, keyboard or remote control can be used separately or combined for optimal workflow speed.



The Nuendo Project Window gives you a complete, transparent overview of your project, with immediate access to track parameters, views "zoomable" right down to sample level and much more.

Allows scrubbing of all audio at the same time as well as scrubbing the video track. Various forward and backward scrubbing speeds are supported, and can be executed from a specific wheel on the Nuendo transport bar or by the jog or shuttle wheel of a supported remote controller or mixing desk. Fast locating, trimming and scrubbing of video and audio could not be easier.

You can also do all editing within the integrated sample editor. This opens an additional range of features for an even faster workflow: easy creation of regions, bouncing these into a library, creating processed regions or tuning sync points by scrubbing the audio with them. Nuendo offers all this plus a smart hitpoint detection feature for fast locating of audio peaks, adapting grooves and creating groove templates in the sample editor which, of course, supports full drag and drop to tie up with the Project Window.

Surround Sound

Offers everything needed for modern surround productions. Every input channel, audio track, effect, group track and output section of Nuendo offers up to 12 discrete channels, ready for full-scale 5.1, 7.1, or even 10.2 productions. To finalize your production, optional encoders for Dolby Digital and DTS are available. Nuendo lets you simulate a wide variety of end user monitoring environments before you start encoding to these industry standards, to ensure that your mix is always delivered with the highest compatibility to all kinds of end formats.

Plug-Ins

To help you get the best out of your project, a huge array of virtual effects, ranging from standard dynamic processing and filtering to creative modulation effects or restoration processors, is supplied.To help you get the best out of your project, a huge array of virtual effects, ranging from standard dynamic processing and filtering to creative modulation effects or restoration processors, is supplied. Or choose from a wide range of third-party VST or DirectX plug-ins. All plug-ins can be used both online or offline, and as many instances can be loaded as your computer can handle.



If you are using MIDI, Nuendo also offers the latest Steinberg technology to create sounds hitherto unheard of. Arpeggiators, chord processors and many other plug-ins are included that manipulate the dynamics, pitch and time elements of MIDI events, as well as three unique virtual instruments to create warm analogue layers or play back powerful drum samples. The integration of the VST interface opens Nuendo up for additional software samplers like the award-winning HALion Sampler, synthesizers or many other virtual instruments.

Applications

Multimedia: The creative process for music and sound design is supported by integrated sound modules and many features for automatic creation of impressive audio collages. The large set of tools to synthesize and match sound to any kind of visual make Nuendo a real time-saver when creating audio for your project. The integrated video player supports all important multimedia files and lets you keep the focus on your product while you are adding audio information.

Radio Broadcast: Design expressive commercials or jingles, create radio news stories or work on multi-part radio. Nuendo is equipped for all the above and more. The Nuendo project concept means that both pitch and time elements of your projects can freely be manipulated, and guarantees intuitive, single-layer editing, fast browsing of sound archives and concise arranging of content.

NUENDO 2.0 *continued*

General

- Support of ASIO, MME, DirectX, mLAN and CoreAudio
- Optimized multithreading: unlimited amount of processors supported, processor load is dynamically spread over all processors
- Support of Intel Hyperthreading Technology
- Timeline Formats: Timecode, Feet:Frames (16mm, 35mm), Seconds, Samples, Bars and Beats
- Multiple timelines showing various formats at once
- 1 user definable frame rate possible
- Extensive event display possibilities
- Complete user configurability: unused menu entries can be hidden, key commands can be disabled, preferences, key commands and the menu structure can be stored and recalled
- Toggle alternate key command sets
- User configurable project templates store windows settings, layouts, track heights/sizes, project and channel settings, plug-ins w/parameters as well as folders and files in the pool
- Easily configurable window layouts, track controls, toolbar and track inspector
- 2nd track list for fixed tracks

Import Formats

- Audio: import of AIFF, AIFC, Wave, Broadcast Wave, Wave 64, MPEG-2 and 3, Dolby Digital AC-3 (with optional Nuendo Encoder) and Ogg Vorbis. WMA, WMA Pro, WMV, WMV Pro (PC only), REX I & II files, SD2, up to 384 kHz depending on the file format
- AES31 • Cubase SX project • Open TL 3.0
- OMF 1 & 2, embedded audio or file references.
- Premiere Generic EDL
- Ability to convert interleaved multichannel file into multiple mono files on import
- CD audio grabbing (selection inside track possible)
- MIDI File import
- MPEG, AVI, DV AVI, WMV/WMV Pro (PC only) and QT movies video import
- Extract audio from video file
- Nuendo tracks including media files, mixer channel settings & automation

Export Formats

- Mixdown to AIFF, Broadcast Wave, Wave, Wave 64, MP3, MP3 Pro, RealAudio G2, Windows Media, SD2 (Mac), Ogg Vorbis, Dolby Digital AC-3 and DTS (optional encoder)
- Supported export sampling frequencies up to 32-bit/384 kHz (Broadcast Wave, Wave, AIFF)
- Audio mixdown to a mono or multi-channel file up to 12 channels
- AES31 • Open TL 3.0 • MIDI File
- OMF 1 & 2, embedded audio or file references.
- Track sheet printing, smart track sheet functions for better overview
- Real-time mixdown option
- Nuendo tracks including media files, mixer channel settings & automation

Synchronization

- VST System Link for sample accurate sync, audio and MIDI transport between multiple Nuendo systems
- Sync to MTC, ASIO Positioning Protocol or send MTC and MIDI clock
- Send 9-pin & MMC machine control
- SMPTE Generator plug-in
- Fast locate and sync lock up
- Sample accurate sync with ASIO 2.0
- MMC slave with track arming support

Recording/Playback

- Record 16/24/32-bit at up to 192 kHz depending on the audio card
- Recording of multiple channels of audio simultaneously per track, split or interleaved
- Record of AIFF, Wave, Broadcast Wave and Wave 64 for long duration recordings.
- Jog and scrub of the complete project
- Project shuttling with fixed speeds, custom editable key commands, and lock to zoom
- Linear Record modes: Normal, Merge, Replace
- Cycle record modes: Mix (MIDI), Overwrite (MIDI), Keep Last, Stacked lanes
- Online recording: record starts when a valid timecode signal is received
- Separate pre roll / post roll settings and separate activity displays in transport
- Multiple user-definable record destinations

Editing

- Real-time non-destructive crossfades
- Advanced crossfade editor with pre & post roll
- Auto-fade and auto-crossfades with definable fade times for smooth transitions
- Sample editor with region list and sync point editing. Drag and drop regions from the sample editor to the project window
- Unlimited undo/redo with edit history list (includes zoom undo/redo)
- Visible second tracklist with fixed tracks
- Timestretch tool allows fast snap-based stretching of material
- Preset curve tools for automation editing
- Various options to lock events in size, fade-length, position and others
- Folder tracks for group-based overview
- Vertical view option for recording takes in the project; browser view allows all data (events, automation, etc.) to be edited numerically
- Edit mode for syncing events & fades to the picture
- Sync points available in events and regions, s-points scrub the audio
- Detect & strip silence
- Hit point detection for loop editing and cue splitting
- Autoselect event under cursor on selected track(s)
- Zoom presets in the project window and part editor
- Analysis: statistics, FFT based spectral analyzer
- Automation data move with the audio events
- Range selection drawn on playback

File Management

- Pool with logical structure and extensive file information
- Import and export of pool files
- Advanced search field with preview option
- Advanced file converter
- Offline processing from the pool
- Minimize function to delete silence & unused material from audio files
- Archive and backup functions
- Libraries can be created including files, file references, regions, subfolders and sync points

NUENDO 2.0 *continued*

Mixing

- Configurable mixer, showing selected channels and channel strip sections at wish
- Up to 4 mixers available at the same time, for more overview for groups, VSTi's etc.
- Unlimited amount of channels, inputs and outputs, effect returns, VSTi & group channels (depending on your hardware)
- VST Connection window: ASIO & master bus routing with preset management
- Up to 12 speaker channels available for inputs, audio tracks, effects, groups and outputs
- Multiple output configurations for mono, stereo and surround formats at the same time
- Plug-in delay compensation throughout the complete signal path
- Interleaved surround recording and playback
- Switchable dual mono panner, combined panner or stereo balance on stereo tracks
- Flexible effect or dry recording from any physical input to any audio track
- 8 inserts on inputs channels, audio channels, effect return channels, group channels, VSTi, ReWire and output channels with global and individual soft-bypass
- 8 auxiliary sends per input, audio, VSTi and group channel with up to 12 channels
- Phase invert and gain on all channels
- Access to internal effect plug-ins or external effects patched via the audio hardware
- Automation tracks for each audio track, group channel and for plug-ins
- Waveform display visible on automation background
- Automation modes Touch, Autolatch and X-Over for all parameters, Overwrite and Trim for volume
- Copy and save/load channel settings is possible

Remote Control

- Steinberg Houston, JL Cooper MCS-3000 and CS-10, Mackie Control, HUI and Baby HUI, Roland MCR-8, Yamaha 01V, DM2000, O2R96, CM Automation Motormix, Radikal SAC-2K, Tascam US-428 and US-224

Video

- Video track with thumbnail preview
- Video playback with QuickTime, DirectShow, DirectX or Video for Windows
- Audio extraction from video files, replace audio in video files
- Full screen video option

Processing

- Integrated processes: Acoustic Stamp, Envelope, Fade In/Out, Gain, Merge Clipboard, Noise Gate, Normalize, Phase Reverse, Pitch-shift, Remove DC Offset, Reverse, Silence, Stereo Flip, Time Stretch, Resampling
- Process history with the ability to modify, disable/enable or replace previous processes. Offline history can be saved as a batch process
- VST and DX plug-ins can be processed offline

Plug-ins

- Real-time support for VST and DirectX plug-ins (may also be applied offline)
- Bundled stereo plug-ins include: Flanger, Phaser, Overdrive, Chorus, Symphonic, Reverb A/B, QuadraFuzz, SPL DeEsser, Double Delay, ModDelay, Dynamics, Magneto, DaTube, Chopper, Transformer, Metalizer, Rotary, Vocoder, StepFilter, Bitcrusher, Ringmodulator, Grungelizer, MIDI Gate, UV22 HR (Apogee), MultibandCompressor, Test Generator, DeNoiser, DeClicker, Q, Nuendo EQ2
- Surround plug-ins include: SurroundPan, MatrixDecoder/Encoder, Mix8To2, Mix6To2, MultiScope, MixerDelay (distance compensation)

Surround

- Mixer is fully multi-channel based, in every aspect, up to 12 speaker channels
- Plug-in based surround panning
- Surround encoding: downmix plug-in for fast monitoring in other multi-channel formats
- Submixer and distance compensation plug-in
- Matrix encoder/decoder for LCRS or 3/2 encoding
- Optional Dolby Digital/DTS encoding/decoding

Network

- VST System Link for sample accurate sync, audio and MIDI transport between systems
- Peer to peer collaboration with track-locking over TCP/IP LAN
- Hierarchical sharing of complete projects down to single tracks using TCP/IP
- Standard network access allows Nuendo to record, play back and copy files from dedicated servers, other workstations or library servers. A regular TCP/IP LAN is sufficient

MIDI/Music

- Tempo-lock option is available on tracks to allow events to maintain their relative bar and beat position as the tempo changes
- MIDI Device Manager for naming and hiding MIDI devices
- 64 VST instrument slots
- MIDI, ReWire and VSTi channels available in the mixer
- User definable PPQ display resolution
- Key /List/ Logical /Drum editors
- MIDI step recording
- Multiple controller lanes in MIDI editors
- Beat calculator • Rewire 2
- Audio & MIDI metronome

SYSTEM REQUIREMENTS



Macintosh - Minimum

- Power Mac G4, 384 MB RAM, Mac OS X
- USB port required
- Supports ASIO 2 specification for high-end multichannel audio
- Supports Mac OS X specifications for compatible soundcards

Macintosh - Recommended

- Power Mac G4 733MHz or faster, 512MB RAM, Mac OS X, v10.2



PC - Minimum

- Pentium /Athlon 650MHz, 256MB RAM
- Windows 2000, Windows XP
- USB Port required
- Supports ASIO 2 specification for high-end multichannel audio
- Supports Windows MME / DirectSound specification for standard soundcards

PC - Recommended

- Pentium/Athlon 1.4GHz or faster, 512MB RAM

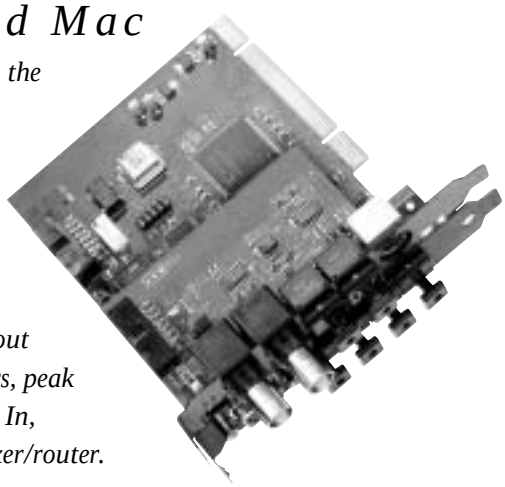
STEINBERG

NUENDO 96/52 DSP

COMPUTER AUDIO

1010

24-Bit/96kHz I/O Audio Card for PC and Mac



Representing the perfect blend of hardware design and ASIO driver implementation, the Nuendo 96/52 DSP delivers an incredible 26 audio inputs/outputs from a single slot PCI card at up to 24-bit/96kHz resolution (12 channel 24-bit/96kHz record/playback on ADAT optical) with unparalleled connectivity (with other digital audio components), complete synchronization possibilities, zero latency monitoring and reduced CPU load. ASIO-poured in hardware means that performance levels previously only achievable with very heavy CPU loading are now possible without exhausting processor power just to get multi-channel audio in and out of the computer in a timely fashion. DSP processing allows freely scalable level meters, peak and RMS calculation in hardware. Card features 3 x ADAT optical I/O, ADAT-Sync In, S/PDIF I/O, word clock I/O, 2 MIDI I/Os and TotalMix, a DSP-based real-time mixer/router.

FEATURES

High Performance

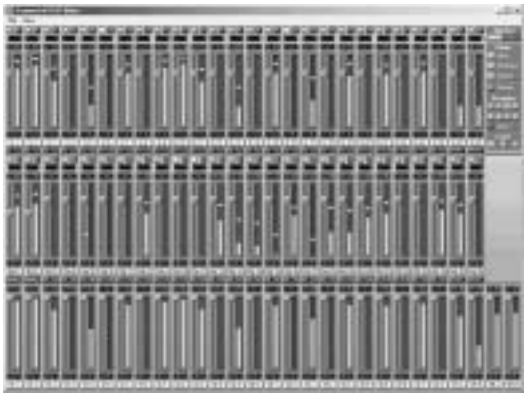
Normally it is the audio software drivers that allow audio hardware for the PCI bus to work with all applications that are using the PC for audio I/O. However, all driver actions are CPU actions, resulting in lost performance that could be better used by the host application. When the Nuendo 96/52 DSP is used with its ASIO driver, the complete data transfer is performed by the smart custom programmed logic of the audio card to the PCI bus. The CPU is simply not used.

Super Fast

The 96/52DSP card is a PCI bus-master device that works exclusively in 32-bit transfer mode. The card moves audio data directly to and from the ASIO host application's memory using the fastest PCI bursts. Nothing could be more efficient. There are no DMA transfers taking place, no multiple 16-bit data movements, and no data bit shuffling.

Auto Sync

With AutoSync activated, the card continuously searches for a valid input signal to lock to. This produces an instantaneous 'record on the fly' performance without having to wait to resynchronize to the input signal. Systems of up to 3 cards are supported and maintaining sync between them is as simple as connecting them all to a common device.



TotalMix Mixer

Any input and output can be routed and mixed to any output. Limited only by the available outputs of the I/O-box, up to 14 independent submixes are available. Internally, the mixer works with 40 bit wordlength. For level alterations there are 65536 steps that allow for a virtually continuous change between 6 dB gain and maximum attenuation. In unity gain setting, the mixer is even bit-transparent and can pass the input signal unaltered.

TotalMix not only leads to perfect and complete ASIO direct monitoring, but is also useful without ASIO. Thanks to a proprietary mixer surface, submix and Zero Latency Monitoring it can be used with all audio applications. An external mixer thus becomes unnecessary many cases.

Unique Features

- Working with multiple digital sources brings its own problems. All devices must be synced precisely or otherwise dropouts and crackles can occur. Ensuring that a system is correctly synchronized is made more transparent with the SyncCheck feature. The card can check all its inputs simultaneously and give an exact read out of the sync quality in the settings dialog.
- SyncAlign features guarantees absolute data alignment across all channels. It cannot scramble channel assignments, and ensures that all ports are started with precise sample alignment. SyncAlign also compensates for any time differences when starting recording while playing.
- Secure BIOS Technology allows to perform hardware updates via software/ driver without any risk. And thanks to its unique 'Zero CPU Load' technology, guarantees highest performance and lowest latency on both notebooks and desktops!
- Change latency on the fly. No manual reset, no re-boot - simply continue when using Cubase or Nuendo.
- Zero Latency Monitoring (ZLM) brings real tape machine feeling to the PC. At Punch-in the corresponding track is switched into bypass directly in the hardware, at Punch-out it switches back to playback. Thus the PC behaves exactly like a 'normal' tape machine.

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EQUIPMENT LEASING AVAILABLE

NUENDO AUDIOLINK 96 SERIES

Versatile 24-bit/96kHz Audio Interface Components

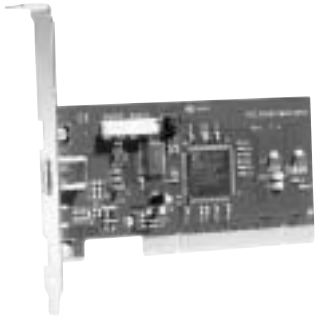
Developed from the technology originally created for the Nuendo 96/52 card, the Nuendo AudioLink 96 Series (consisting of two interface cards, two breakout boxes and the TotalMix software) offers the same low latency, AISO performance and high channel counts that has made the Nuendo 96/52 the audio card of choice for professionals. However, the increased flexibility which the components in the AudioLink 96 series offer, enable users to build a system which meets the specific demands of their current working environment. Studio or location, laptop or desktop, Mac or PC, there is a combination of card, interface box and driver to fit all needs.

Like the Nuendo 96/52, Nuendo AudioLink 96 components will shape a system that supplies 24-bit/96kHz audio; multichannel digital and analog I/O; ADAT optical, S/PDIF, ADAT Sync and Word Clock inputs; and latencies down to 1.5ms. But they also add MIDI I/O and Total Mix software, making the system especially valuable to the user on the move who now has one less box to carry around with them.

And Nuendo AudioLink 96 isn't just a self-contained construction kit for audio interfacing. The ADAT connectors on the Nuendo Digiset and Multiset I/O boxes enable you to hook up to all existing Nuendo converter racks.

COMPONENTS

Nuendo Audiolink 96 PCI



The Nuendo Audiolink 96 PCI is a more traditional PCI card for Mac and Window computers with the same functionality as the Audiolink 96 Mobile. Connects to the Nuendo Audiolink 96 Digiset or Multiset via a special 15´ breakout cable and features the low latency and Zero CPU performance of the Nuendo 96/52 DSP card.

Nuendo Audiolink 96 Mobile



This is an exclusive PCMCIA type II card that fits into all laptop computers for the most compact and powerful Nuendo system on the move. Connects to the Nuendo Audiolink 96 Digiset or Multiset via a special 15-ft. breakout cable. Features the low latency and Zero CPU performance of the Nuendo 96/52 DSP card.



Nuendo Audiolink 96 Digiset

This 9.5˝ breakout box gives you all the connectivity of the Nuendo 96/52 DSP PCI card (3 x ADAT optical I/O for 24 channels each way, S/PDIF coaxial for stereo I/O, ADAT-sync input for sample accurate synchronization, and Word Clock I/O) plus stereo line out for monitoring and two MIDI inputs and outputs. Ideal for those who want to do editing and/or MIDI recording on the move, but then needs to connect into 24 channels of ADAT I/O on returning to the studio for audio recording/mixdown.



Nuendo Audiolink 96 Multiset

Used in conjunction with the Audiolink 96 PCI or Mobile cards, this breakout box connects to external equipment with 8 analog inputs/outputs (balanced, 24-bit/96 kHz), one ADAT optical I/O for 8 additional channels each way, S/PDIF I/O, Word Clock I/O and ADAT-sync input. Plus, there is one MIDI I/O and a separate analog line output. Perfect for those who to do 8-channel recording as well as MIDI/audio editing on the move, but then connect into an additional 8 channels of ADAT I/O on returning to their project studio for recording/mixdown.

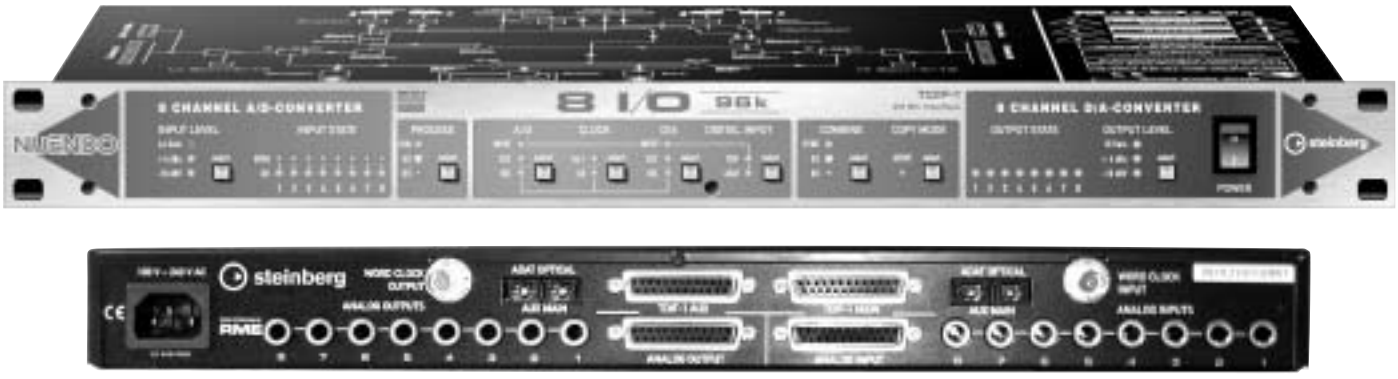
STEINBERG

NUENDO 8•I/O 96K

COMPUTER AUDIO

1012

8-Channel 24-bit A-D/D-A Converter



Perfect for those working at the cutting-edge of audio recording for formats like DVD-Audio, the Nuendo 8•I/O 96k uses the very latest 24-bit converters with 128x over-sampling to attain a real dynamic range of over 110dB at a sampling rate of up to 96k. The analog input circuitry uses a fully servo-balanced and completely symmetrical audio path. The flexible gain architecture allows input sensitivities of low gain, -10 dBV and +4 dBu and output sensitivities of high gain, +4 dBu and -10dBV.
With its intelligent clock control an internal clock is provided in 44.1 up to 96 kHz, but you also can use an external clock. And to ensure a perfect synchronization the Nuendo 8 I/O Sync Check immediately shows sync conflicts in your system and the Nuendo 8 I/O SyncAlign guarantees absolute data alignment across all digital outputs.

FEATURES

- 96k converters feature 8 channels of A/D conversion that can be output via both 24-bit ADAT optical and TDIF-1 interfaces and 8 channels of D/A conversion driven from either the 24-bit ADAT or the TDIF-1 interfaces, all in a single rack space.
- To make the best of the excellent dynamic range of the A/D converters, the input level switch steps through the AD8-I/O's three input sensitivities: Low Gain, +4dBu, or -10dBV. Similarly, the output level can be independently selected from High Gain, +4dBu, or -10dBV.
- Inputs can be accurately calibrated to the audio source using the signal present & signal clip indicators provided. Each analog output has its own 'Signal Present' LED that works in an analog way (brighter at higher levels). Pushing OUTPUT LEVEL switches between High Gain, +4dBu or -10dBV as the selected analog output level.
- Nuendo SyncAlign guarantees absolute data alignment across all channels. It ensures that all ports are started with precise sample alignment, and it also compensates for any time differences when starting recording while playing.
- All digital inputs/outputs operate over the full 24-bit range, and the Nuendo 8•I/O 96k can also be used as a a digital patch bay and format converter.
- Intelligent Clock Control (ICC) provides functionality not found on any other converter. For example, the AD converter clock can be derived from the internal clock at 44.1, 48 or 96kHz or externally from the BNC connector or from any of the digital inputs. The DA have the same range of options.
- The larger the system, the greater the chance sync isn't maintained across all digital and word clock inputs. And if all devices aren't synced precisely, dropouts and crackles can occur. To ensure that a system is is correctly synchronized and working at the best possible audio fidelity, the 8•I/O 96k incorporates SyncCheck to check all its inputs simultaneously and give to give an exact read out of the sync quality in the settings dialog.



Nuendo Studio System One 96k —The Complete Audio Production Solution

Nuendo Studio System One is a complete production solution designed for the audio production process, whether for music, post production or Surround Sound authoring for film, TV and DVD projects. Available for Mac or PC, the bundle includes Nuendo 2.0 software, the Nuendo Surround Edition, Nuendo PCI 96/52DSP card and the Nuendo 8•I/O.

STEINBERG

NUENDO DD8

Digital Format and Rate Converter

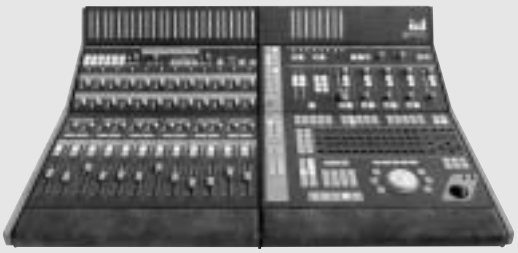
The Nuendo DD 8 converter is a compact 1U 19˝ rack-mounting device featuring digital interfaces in the following formats: AES/EBU, ADAT, TDIF. With its 8 channels of AES/EBU input and output it serves as a perfect AES/EBU front end for Nuendo users. The DD8 can also be employed as a digital patchbay for splitting digital signals and as a sample rate/digital format converter. Because it offers several combinations of the basic functionalities it can be a real lifesaver in studio environments, where many problems related to digital signals have to be solved every day.





- Switchable 24-bit sample rate converters (SRC) allow for both highest quality sample rate conversion and clock unlinking of all AES/EBU inputs.
- All of the Nuendo DD8's digital I/Os support 96kHz/24-bit. As ADAT optical and TDIF are usually restricted to 48kHz, the unique DS mode (double speed) allows two channels to be used for the transmission of one channel's data. The algorithm used is compatible to S/MUX and Double Wide, and is implemented in the Nuendo 96/52 DSP audio card. (Makes the Nuendo DD8 ideal when Nuendo 96/52 DSP owners need real time AES digital conversion).
- The Nuendo DD8 is a reference quality 8 channel format converter featuring exceptional features like Intelligent Clock Control (ICC), SyncCheck, SyncAlign, Bitclock PLL, digital patchbay functions, and active jitter reduction through SD-PLL and 96kHz/24bit sample rate conversion.
- AES/EBU to ADAT/TDIF and ADAT/TDIF to AES/EBU work both entirely independently and intelligently coupled.
- Dedicated LEDs display the state of the incoming and outgoing signals and all operations running inside the device.
- 8 channel AES/EBU (4 x stereo, XLR) to ADAT optical/TDIF-1 converter
- Sample rate conversion 96 kHz/24 bit switchable for the AES/EBU inputs
- 8 channel ADAT optical/TDIF to AES/EBU (4 x stereo, XLR) converter
- ADAT optical inputs, 24 bit, using bitclock PLL for sample accurate lock
- Clock unlinking (re-synch) by SRC
- ADAT optical outputs, 24 bit, fully compatible with all devices using this interface
- TDIF-1 interface, 24 bit, low jitter PLL, Emphasis support, DA-38, DA-88, DA-98 compatible
- Copy Mode for copying data to an output of the same format (ADAT to ADAT) etc.)
- Active jitter reduction by Serial Double PLL and SRC
- Digital patchbay for copying, duplicating and distributing the digital input signals
- ❖ SyncCheck, unequalled technology for checking clock synchronicity
- S/PDIF mode switchable for the second ADAT optical output

New! ID — Remote Control for Nuendo



Designed and handcrafted by a team of audio professionals to meet the demand for total control of Nuendo with a familiar "console"-like feeling and much more, Steinberg's ID (Input Device) is far more than another generic controller device. ID consists of 4 logical sections dedicated to various aspects of recording, editing, mixing and control room monitoring:

- Fader-section offers 24-channels for direct access. LCDs show track names and status of channel parameters: solo, cut, track- arming, direct channel-strip access and automation-status. 24 touch sensitive motorized faders and encoders control levels.
- Encoder-section features a complete channel strip layout with 24 endless encoders, 24 two line displays and 24 level indicators for direct access to any channel parameter.
- Controlling VST Plug-ins is supported by 24 endless encoders in conjunction with 48 lines of display and 12 additional Favorite-User-Function encoders with displays.
- Edit-Section's integrated Matrix ASCII keyboard and the trackball allow computer and software operation without mouse and keyboard. Also has switches and status displays for nearly every important edit function.

STEINBERG

NUENDO TIME BASE

COMPUTER AUDIO

1014

VST System Synchronizer



Nuendo Time Base is the first synchronizer supporting Steinberg's VST System Link technology. This makes it the perfect solution for sample accurate cueing of tape-based audio and video machines from Nuendo audio production systems. All functions can be controlled and edited from within the Nuendo software. The outstanding feature set of Nuendo Time Base makes it the most versatile and precise synchronizer available today. Nuendo Time Base can also be used as a standalone unit. Any Nuendo system can be used as a 9-pin slave —just hit "Start" and Nuendo will immediately follow every command with the Virtual Machine Option.

FEATURES

- Synchronization of digital audio and video systems (hard disk recorders, mixers, etc.) via house sync (Blackburst), AES-EBU, LTC, VITC, with analog/digital audio and video tape machines, sequencers, etc. with separate inputs for video sync and VITC R/W
- Read, generate and regenerate timecode (LTC, VITC, MTC and VST System Link)
- Burn visible timecode readout into video picture (2 different sizes, 4 display styles, freely positionable)
- Supports all sample rates from 16 to 192 kHz incl. NTSC Pull up/down
- 9-pin interface provides integration of machine control for decks like Betacam, Tascam DA98, etc.
- Ergonomic practical operation using 4 key pads and text display to show all functions and conditions plus software control, guarantees the user a full overview and intuitive usability
- Optional: Virtual 9-pin Machine. Emulation of 4 different 9-pin machines, allows use of Nuendo as a slave machine on professional 9-pin editors and (console integrated) controllers.

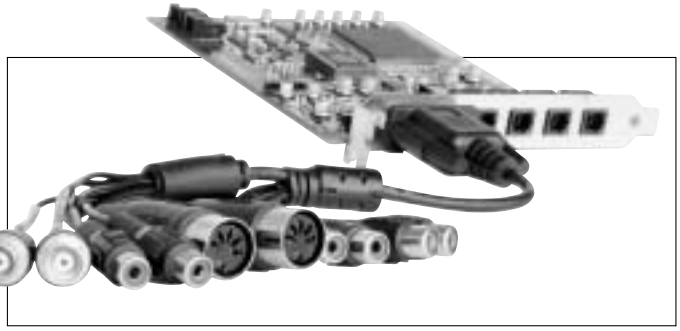
PRACTICAL SETUPS:

1. House sync for all digital devices: with its 4 WordClock outputs, 1 AES/EBU output and one of the best clocks available it is the perfect clock master in any studio with sample rates up to 192 kHz.
2. The Time Base can lock to a Blackburst signal. WordClock, AES/EBU and the timecode generator are video locked.
3. System Link generator: any timecode source - LTC, VITC, 9-pin - will be converted to system link data. By connecting the Nuendo audio card to the AES/EBU connections on the Time Base, Nuendo systems will slave sample accurate to the incoming timecode and machine control. Time Base can be run as a Virtual Machine and outputs the necessary System Link information (+ LTC, VITC inserter) to the connected Nuendo system(s), by running the virtual machine. The whole system including the timecode generator is video locked as long as Blackburst is fed to the Time Base.
4. The 9-pin Machine or the Virtual Machine can be controlled via MMC in parallel to the controls from the Nuendo system.*
5. By using the optional Virtual 9-pin Machine option (VTB), Nuendo – using VST System Link – behaves like a 9-pin Machine to the controlling system. Four different 9-pin emulations are available. (BVW 75, DVW500, 3348, TM). Depending on the chosen emulation, up to 64 tracks (track ready) commands will be supported.*
6. Separate from the video sync loop thru connectors, the Time Base offers a video in-/output which is used to read VITC, insert a VITC signal from the internal VITC generator and a switchable burn-in window. The video signal can be fed to the Time Base for timecode insertion on video playback.
7. NTSC Pull up/down: the Time Base can provide slower or faster clock signals (approx. 0.1%) for NTSC pull up/downs. All devices will receive the changed clock speed.
8. Additionally, the GPI port assures that standard GPI devices used in the studio are controlled by Nuendo via the Time Base, like red-light.



VSL-2020

Ultra-Low Latency ASIO Audio Card for Playing VST Instrument Live



The VSL-2020 provides 32 ADAT channels for pristine transmission of audio and MIDI data in 24-bit/96kHz quality to and from the other computers in your VST System Link network. Flawless support of ASIO 2.0 ensures stable, low-latency multi-channel network operation with other computers running Nuendo or Cubase. The VSL-2020 card is bundled with V-STACK (see below) so you can increase the available music-making power in your studio.

With the VSL-2020 card and V-STACK software you can turn your PC into a live virtual instrument station. The VSL 2020's ultra low latency architecture guarantees dynamic, live play of VST instruments at latencies as low as 32 samples (less than 1ms). V-STACK software hosts up to 16 VST instruments, and provides 8 send effects, 5 insert effect slots and 4 master bus effects for your VSTi sounds. All this in immaculate 32-bit audio quality.

- 32 ADAT channels - 2 optical ADAT I/O with S/MUX support
- S/PDIF I/O - supports AES/EBU and consumer S/PDIF
- Analog I/O - unbalanced (-10dBV) Stereo RCA I/O
- SuperClock (FS 256) and WordClock I/O with BNC connector
- Standard MIDI 5-pole I/O
- 24-bit with 32, 44.1, 48, 88.2 or 96 kHz
- Bundled with V-STACK VST System Link software

V-STACK

VST System Link/Live Performance Software



Since Steinberg introduced the VST 2.0 plug-in format a few years ago, there has been a rapid development of VST Instruments - software synthesizers and other sound sources played and controlled from within a host application. There is now a huge number of VST Instruments available, ranging from simple synthesizers and drum machines to exact software replicas of vintage synths and extremely advanced sound modules with no equivalent in hardware. VST Instruments can often be more flexible than hardware synthesizers, allowing for total recall and full automation of all parameters, patching and mixing in the digital domain, graphic interfaces and solutions that wouldn't be possible in hardware, etc. However, any computer can only play so many VST Instruments at a time - and typically, the more advanced a VST Instrument is, the more processing power it requires. Enter Steinberg's VST System Link. . .

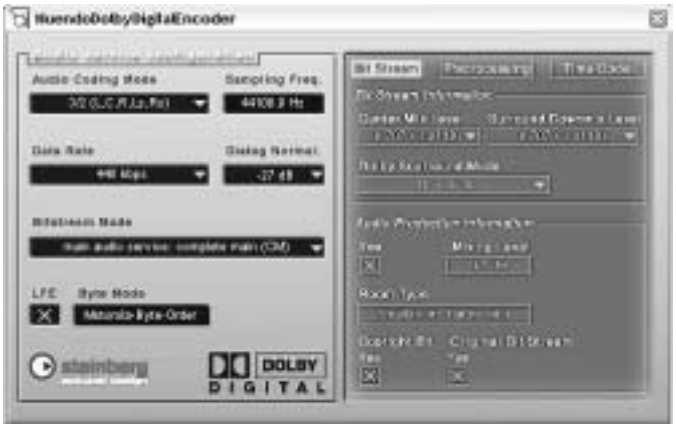
This revolutionary system makes it possible to have several computers working together as one large system, with no other requirements than ASIO compatible audio interfaces with digital audio connections. Computers connected via VST System Link will freely exchange audio and MIDI data, all in perfect sync, creating a digital audio network system. This provides an excellent solution to the problem of CPU-hungry VST Instruments: let one computer play audio and MIDI tracks and dedicate another computer to running VST Instruments only, taking full advantage of all available processor power!

V-STACK is the perfect application for this: a stand-alone VST Instrument host, supporting up to 16 VST Instruments with full mixing capabilities and VST effect support. The VST Instruments are controlled via MIDI over VST System Link or from a regular MIDI interface (e.g. for live use) and the audio can be freely routed to any outputs on your audio interface. Since there is no transport, editing, event handling, etc. all processing power can be focused on VST Instruments and effects!

STEINBERG

NUENDO SURROUND ENCODING

Until recently, the only way to encode audio for surround sound was by using expensive hardware systems which required lots of effort to get the audio in the right form to play into the encoding hardware and then to pass on the encoded audio to the authoring stage of the project. However, with Nuendo the whole process of encoding 5.1 Surround Sound—via DTS or Dolby Digital—can now be integrated into computer-based production systems where the processed files are ready for use in the authoring process.



Nuendo Dolby Digital (AC-3) Encoder

The Nuendo Dolby Digital Encoder software plug-in allows Nuendo projects to be encoded into Dolby Digital—the number one choice for delivering surround sound to millions of home theater system around the world. The multi-channel audio format for DVD discs, Dolby Digital (also known as AC-3) is based on an algorithm that takes advantage of auditory masking and both intra- and inter-channel redundancy. This technique guarantees both outstanding audio quality and bandwidth efficiency. Licensed by Dolby Laboratories, the Nuendo Dolby Digital Encoder comprises the full functionality of the original Dolby hardware. Imagine the sheer convenience of having the encoding system on the same platform as the program which is used to create and process the audio.

Nunedo DTS Encoder

A rival format to Dolby Digital, DTS (Digital Theater Systems) provides a similarly epic sonic experience in DVDs as well as DVD-Audio and SACDs where the increased fidelity of the rear speaker pair means that full bandwidth can be placed in all speakers in the surround field. Specifically licensed by Steinberg to allow Nuendo users access to the next stage in the surround sound production process, the implementation is simplicity itself with the smart integration of the DTS file format in Nuendo's export dialogue. The necessary routines are all automated so that the user merely needs to define which 6 files should be DTS encoded and the resulting encoded files are placed on the desktop ready for use in the authoring stage.

Urban Atmospheres Surround Library



This new surround library provides urban acoustic 5.1 environments recorded in immaculate quality. Spanning 95 scenes on 9 DVDs, Urban Atmospheres are constructed to ensure maximum versatility in building urban soundscapes for film, TV, radio, theater or music.

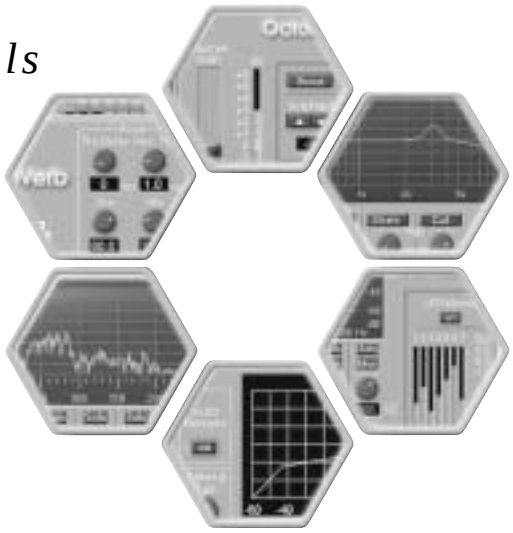
Urban Atmospheres are a collection of high-end sound environments professionally recorded in 5.1 surround that offer unequalled spatial characteristics and pristine sound quality. The SPL Atmos System was used to record 25 indoor and outdoor locations, spanning 95 scenes. These astoundingly authentic recordings were compiled onto 9 DVDs in AIFF format, ready for immediate use in any pro audio system (Windows or Macintosh).

Urban Atmospheres are perfectly suited for any use in sound-tracking or dubbing, be it for film, TV, theater, music or radio plays, or anywhere you need to create a spatially realistic acoustic environment. And because the unique construction kits containing background atmospheric recordings with no conspicuous audio events are combinable with the included foreground events, you can mold the material to suit any requirements or even build your own, even more complex acoustic environments with no additional software required.

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NUENDO SURROUND EFFECTS PROCESSING

Nuendo Surround Edition— Eight-channel Real-time Surround Tools



The Nuendo Surround Edition is a collection of 6 surround real-time effect plug-ins designed for daily use in media production work. Offering up to eight channels of compression, equalization, loudness maximization, reverberation, and LFE management, the Surround Edition is the perfect choice for anyone with multi-channel mixing needs. Since the requirements of a surround environment differ quite significantly from the classic two channel world, Steinberg Spectral Design has developed this collection specifically as a tool for the specialist fields of film, theater, DVD and modern music production. In fact these state-of-the-are plug-ins provide all the processing and "finishing" tools needed to produce mixes for both 5.1 and 7.1 productions.

OctoQ: A 7-band EQ with 3 parametric variable-Q mid bands and low and high shelving and cut filters which works at double the selected sample rate

OctoMaxx: A soft/hard gain maximizer with up to 4 linking groups, input/output level meters for each channel and Possible Gain and Achieved Gain meters

Octoverb: A reverb/ambiance plug-in with two independent room models and intelligent adjustment of the reverb tail according to the room size. Also features independent Level, PreDelay, High and Low Damp controls for each channel

OctoComp: A state-of-the-art compressor with auditionable independent sidechain filter for high and low frequencies, continuous blend between RMS and Peak characteristics, an autorelease function and up to 4 'linking' groups with individual settings

LFE Splitter: A 24dB brickwall filter with low cut filters for each of up to 7 full range channels from which it generates one LFE signal with selectable attenuation steps

LFE Combiner: With adjustable low frequency filters and solo mode for each channel and a monitor mode for the LFE signal



The native version of the highly regarded TC-quality reverb expanded to fill the 5.1 space with dense and entirely convincing studio quality reverbs, TC Surround Reverb is the ultimate surround reverb for use with Nuendo—and is indispensable to anyone mixing for film, home theater or DVD.

TC Surround Reverb

The graphic user interface provides intuitive control of all the main parameters for reverb design covering room shape and size through to diffusion and separate controls for early reflections and reverb tail. There are three independent tail controls for low, mid and high band with a sweepable band crossover and and an additional EQ stage with low and high shelf filters, giving an unprecedented sonic flexibility and realism to surround productions.

As Nuendo already takes care of positioning tracks in the 5.1 surround field, TC Surround Verb receives the signal already panned, so instead of it's own panner, it has an innovative surround metering which allows you to keep track of the audio's position in the virtual room. Dry and wet signals are displayed in two different colors, providing maximum visual feedback as to what is happening to the source signals and the effected components.

In the Space Editor you can set up size and shape of the room. This is the first aspect of any reverberation characteristic, where the early reflections from walls, floors and ceilings make a large contribution to the overall sound. The Time Editor allows you to process the reverb tail in three distinct frequency ranges, the PreDelays are separately definable for early reflections and tail, and the Filter Section gives an individual tone for both early reflections and tail with high shelving filters.

STEINBERG

WAVELAB 4.0

COMPUTER AUDIO

1018



The Standard in Digital Audio Editing

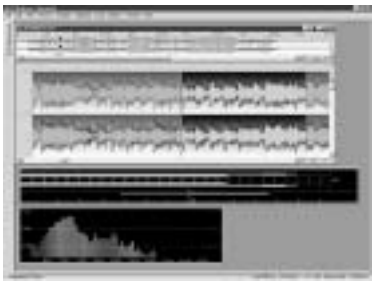
Professional audio editing and mastering software for Windows, WaveLab 4.0 can handle almost any audio editing task—be it sound design, high-quality mastering, internet audio, multimedia applications, and preparing audio for broadcasting or computer telephony. Combining easy-to-use stereo editors with a wealth of mastering features, WaveLab 4.0 offers powerful editing tools and an array of new, high-quality virtual effect processors. With its unique Audio Montage window, WaveLab 4.0 offers a revolutionary, non-destructive editing concept whose flexibility and performance no other software solution can match. Create Audio Montages and carry out precision edits in real-time. Use fades, crossfades, volume envelopes or effects without having to wait. The full implementation of multitasking allows editing and processing in the background during playback. You can even save or record while working on a different file. WaveLab 4.0 also features drag & drop and a customizable user interface, and you can freely scale all windows with a mouse. Comprehensive CD burning capabilities, real-time audio file analysis and batch processors further add to the functional depth of WaveLab 4.0. Plus, Steinberg's latest version also supports samplers, ASIO, WDM, a vast range of file formats as well as file resolutions of up to 32-bit IEEE/192 kHz, and real-time input and output monitoring.

FEATURES

Highest Quality Sound

WaveLab 4.0 offers outstanding sound quality. The internal 32-bit floating processing depth with a sample rate of up to 192 kHz offers superb audio clarity. Apogee's renowned professional UV22HR dithering algorithm converts your audio into other bit rates at the very last step in the mastering process.

Powerful Real-Time Analysis Tools



WaveLab 4.0 features a broad palette of professional analysis tools. Thanks to newly developed high-precision level indicators, you can visually monitor playback audio input and even each single sample position - in real-time, of course. Also included are a new version of the Peak/VU Meter (now free-floating), a 60-band spectrum analyzer and Phasescope, a correlation meter with integrated goniometer. With the FFT meter, you can also, for example, test rooms or PA systems.

Audio Montage



The Audio Montage window is the heart of WaveLab 4.0. Here, audio files and sections of audio are defined, arranged and edited. Open as many clips on as many tracks as you want simultaneously in the audio montage and place, move, cut, fade or crossfade them any way you like. Crossfades, panorama, volume edits and effect send levels are calculated in real-time. You can change the "rubber band" curves with the mouse: hear and see the changes you make straight away - without creating fade files that have to be calculated first. The Audio Montage is absolutely non-destructive, as the clips are only references to the original audio files. This gives you the flexibility of being able to move, cut or delete clips. Each clip can be processed using up to ten VST effects. The sum of all the tracks then passes through the Master Section, where another eight real-time effects (VST, WaveLab, Direct X) are available. Unlimited undo/redo lets you take back any edits or changes you have made.

Integrated Mastering Studio

WaveLab 4.0 offers everything you need to give your audio the finishing touches it deserves: a graphic, editable, fully parametric equalizer, a top level reverb algorithm, comprehensive dynamics section as well as special high-end loudness optimization algorithms. Also built-in are two plugins for restoration of audio recordings, Declicker and Denoiser. Using virtual effect processors in the Master Section couldn't be easier - you can drag & drop the plug-ins around and place them into the desired position in the processing chain. To help you compare results of different effect combinations easily, you can open several instances of WaveLab 4.0 at the same time.

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EQUIPMENT LEASING AVAILABLE

WAVELAB 4.0

Burn Your CDs

WaveLab 4.0 makes burning CD masters easy. You can set track and index markers directly in the waveform, including extremely accurate settings for codes and pauses. The integrated Label Editor lets you make your own CD labels and covers. The backup tool lets you archive your audio files to hard drives, CDs and JAZ. Also features audio-in-pause function for creating hidden tracks, 1:1 CD copying, ISO import and export, PQ sheet import, export and printing.

Transparent File Management

With its integrated audio database and batch processors, WaveLab 4.0 takes care of routine tasks. Organizing sound archives or databases is simple. These versatile tools can edit and convert audio files automatically. Using them couldn't be simpler - just select your audio files, choose the editing functions and set your destination folder. Meta Leveler can be used during batch processing to compensate undesired volume fluctuations.

Audio file formats

Supported file formats include WAV, AIFF, Ensoniq Paris 24-bit, RAW (8/16/20/24 bit), AU, Sound Designer II, ulaw, MP3, Sun/Java and many more…

Additional Functions

- ASIO, WDM and MME driver support
- Analyze audio signals in real-time: Level/Pan, Phase, Spectrum Analyzer, FFT Meter, Wave Meter, Bit Meter
- Full real-time input and output monitoring
- High-quality virtual effect processors Q (parametric 4-band mastering EQ), Multiband Compressor, Spectralizer (Enhancer), Denoiser, Declicker, Voice Attenuator, 192 kHz Resampler
- Backup function, data CD burning, CD copy, CD label creation
- OSQ (Original Sound Quality) - lossless audio file compression
- Save several audio files in the background while you continue working
- Apogee UV22HR Dithering
- Automatic pitch recognition and pitch adjustment
- Autosplit function splits audio files into segments with a range of different criteria

Applications

Mastering

WaveLab 4.0 offers a range of features which are indispensable for mastering: editing, arranging CD tracks, sound processing with internal EQs, dynamics and effects, automatic volume adjustment, UV22HR dithering and noise shaping, CDR writing, tools for problem analysis and comparing files, and a raft of real-time analysis tools (FFT, VU/Pan, Bit, Spectrum etc.)

Broadcast

Ideally suited for use in everyday broadcast audio, Wavelab includes a variety of solutions, such as: easy-to-use speech cutting tools, simple, graphic arrangement of audio files for radio, automatic ducking and automatic real-time crossfades.

Multimedia

Import and export MP3 files, while intelligent functions prepare your files for multimedia applications. These include the audio database and the batch processor. WaveLab 4.0 also features integrated dithering to word lengths between 8 and 24 Bits as well as sample rate conversion (5 to 192 kHz).

Sound Design for Samplers

Communicating directly using MIDI or SCSI, sample editing is a joy. Sound processing with internal EQs, dynamics and effects, volume optimization, distortion control, high quality time stretching and pitch shifting with pitch detection, auto-split functions and professional loop creation features. All supported by crossfade-looper and wave EQ.



WAVELAB ESSENTIAL

Digital Audio Editing For Project Studios

WaveLab Essential offers a range of indispensable digital audio editing features, from superior CD writing functions to comprehensive mastering possibilities. Based on award-winning WaveLab 4.0 technology, WaveLab Essential delivers outstanding quality for your project studio - and all this performance at an unbeatable price.

	WaveLab 4.0	WaveLab Essential
File Formats Supported	full set	slightly reduced set
Sample Rate	192 kHz	96 kHz
Audio Montage	fully featured	2 tracks, reduced feature set
Number of Slots	8	4
Dithering	internal+uv22/hr	internal
Audio CD Burning	x (enhanced)	x
CD Copy	x	
Professional CD Report Generator	x	
Basic Audio CD	x	
Recording Functionality	full	medium
Batch Processing	full, with specialized plugins	file encoding
Automated File Cutting System	x	

STEINBERG

MASTERING EDITION By Spectral Design

COMPUTER AUDIO

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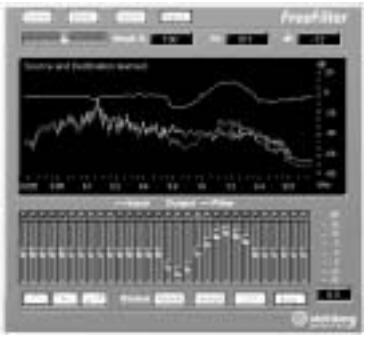


The Complete VST Mastering Solution

Every recording deserves to be mastered. Without mastering, a mix can sound dull and lifeless. Single tracks sometimes do not fit together with other tracks because of differences in the dynamics or the frequency characteristics.
Designed to meet today's ever-increasing demands in music production, the Steinberg Mastering Edition is a set of six high-quality plug-ins specially designed to help you achieve the most efficient and effective results. Compatible with a wide range of host-based audio applications, together they will satisfy all your mastering needs. So whatever the situation, the Steinberg Mastering Edition gives you the tools needed - in the quality you expect of Steinberg audio software.

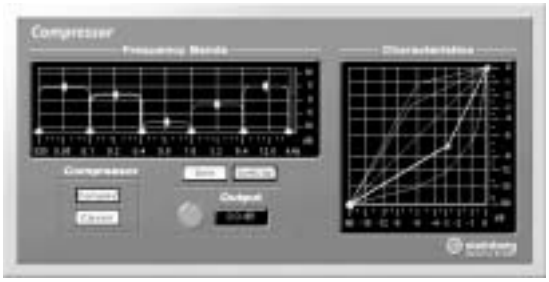
The ME Loudness Maximizer sets the standard in loudness optimization. It is the first dynamic processor whose algorithms were modeled with the express purpose of achieving the optimal effective loudness. Even with a mix where the loudness has already been pushed to the limit, the ME Loudness Maximizer can increase the subjective loudness of the audio material without losing "punch" or causing distortion. The intelligent algorithm analyzes the audio material and automatically adjusts all parameters accordingly. The preset functions allow definition of standard working situations, making this high-quality tool very user-friendly. The ideal tool for complex applications such as mixing, mastering, post-production, or broadcasting, the ME Loudness Maximizer guarantees outstanding results—no pumping, no clicks, and no unwanted artifacts or other side effects.





The ME FreeFilter is a linear phase real-time 1/3 octave equalizer, whose up to 30 bands of equalization allow very delicate adjustments of critical frequencies. The uniqueness of ME FreeFilter lies in its ability to conduct an extremely detailed analysis of audio sequences and use this data in a completely new way. The ME FreeFilter can "learn" the frequency spectrum of a piece of music and "copy" it to another. The Morph function allows real-time scalable mixing between the two frequency spectra. This means that different music tracks can quickly and easily be made to sound more alike. A large display shows the frequencies of both "learned" audio signals (and the differences between them) with pinpoint accuracy.

The ME Compressor is a multiband compressor, which divides the audio signal into five independent frequency bands. A line showing compression characteristics is available for each band. This allows highly flexible compression of varying frequency ranges, music signals and even single instruments, within the final mix. The compression characteristics are displayed on a logarithmic scale, making editing more intuitive and allowing highly delicate adjustments. The solo function allows monitoring and adjustment of each band individually. Has two modes:
Classic mode - Standard compressor with fixed Attack and Release times
Complex mode - A new approach to audio compression. Adaptive control produces a cleanly compressed signal without the "pumping" produced by many other compressors. The softclip function gives the output signal that certain sharpness.



STEINBERG

MASTERING EDITION By Spectral Design

ME PhaseScope and ME SpectroGraph



The ME PhaseScope and ME SpectroGraph round off the Steinberg Mastering Edition with two display-components indispensable for mastering work. The ME PhaseScope is a Goniometer with integrated correlation meter. The ME SpectroGraph displays a spectral plot as a real-time sonogram, which gives precise verifications of even the smallest detail in the recording.





VOICEMACHINE

Real-time Vocal Pitch Transformer

Consisting of two real-time voice effect tools, VoiceMachine is the ultimate application for redesigning the voice—whether by creating completely new sound files or altering the pitch leaving the natural character untouched. The VM Generator allows you to manipulate a vocal track as if it were an instrument. You can control the pitch and formant characteristics of up to 4 additional voices in realtime, simply by triggering them via MIDI. This lets you create lush vocal arrangements from a single lead vocal track. The VM Processor lets you change the melody or correct and smooth the intonation of a voice while maintaining its natural character. Their VST 2.0 interface allows all parameters to be addressed via definable MIDI controllers.

VM Generator



- ◆ Up to 4 voice harmonies can be triggered via MIDI in realtime allowing you to create a backing choir from a single voice in minutes, reducing recording time to a minimum
- ◆ Each voice features independent control of pitch change and voice character as well as:

– An LFO that provides vibrato simulation with a choice of three waveforms (Sine, Saw and Triangle) as well as adjustable Rate, Depth and Delay parameters

– Adjustable Gain, Pan, Formant and Fine Tune sliders

VM Processor



- ◆ The VM Processor features an intonation correcting algorithm that allows you to fix any tuning problems in your vocal recordings
- ◆ The portamento control can be used to smooth the transition between notes as well as create interesting effects at more extreme settings
- ◆ The VM Processor allows you to either change the melody or simply correct the intonation by changing the pitch of a voice without changing the voices original character
- ◆ Independent realtime control of pitch change and formant shifting voice character delivers natural sound pitch shifting
- ◆ Pitch/Character link function for “classic” pitch shift effects
- ◆ Independent controls are provided for the pitch shift and character profile to allow natural pitch shifting or used interactively to create a range of effects such as simulating other singing voices, and even imprinting the character of the male voice onto a female vocalist.
- ◆ Real-time natural pitch shifting (no ‘singing rodent’ effect)
- ◆ Independent control of pitch change and voice character
- ◆ LFO with different waveforms and delay for vibrato simulation
- ◆ Smoothing intonation problems in vocal recordings

STEINBERG

GRM TOOLS

Real-Time Sound Manipulation Plug-In Collections



GRM Tools is collection of eight unique plug-ins for VST by the Musical Research Group (GRM) at the Institute National de l'Audiovisuel in Paris. Famous for their research into sound processing tools, the possibilities offered by the GRM Tools collection goes far beyond ordinary filter, auto panning or chorus effects. High quality algorithms offer unprecedented flexibility and open up new dimensions in audio manipulation.

All 8 Plug-Ins offer the same interactive integrated preset handling which allows settings to be assigned to one button in order to be reselected with a single click. The key lies in the delay parameter which defines the time frame in milliseconds during which the settings from the one preset blend over into the other. Complex parameter mixes allowing sensational sound movements are created in no time at all!

GRM Tools are indispensable for sound design, dance, electronic and experimental music creation, film and TV, even jingles and radio effects.

- Parameter interpolation and settings management
- Slight to extreme manipulation of the audio source
- Time relevant parameters can be synced to tempo (VST 2.0 compatible host software)

Volume 1

Shuffling

Shuffling is a random micro-splicing and shuffling algorithm. Depending on the selected memory sizes and density, Shuffling introduces bouncing of signal fragments at definable points in the signal's actual timing, whilst preserving the overall continuity of its time sequence. This leads to spectacular panning, panorama width and stereo movement effects.

Comb Filters

Comb Filters is a bank of 5 comb filters with resonance up to self-oscillation. The comb filter amplifies the signal at a given frequency and at all harmonic frequencies (integer multiples of the fundamental) of that frequency. Harmless drum loops grow into exciting new groove happenings by emphasizing certain frequencies or morphing classical instruments into abstract fx tunes.

Pitch Accum

PitchAccum is an algorithm which combines two transposers with their feedback delay device. Transposition is switchable on a periodic or random basis. Harmonizer effects and spectacular pitch related creations can be generated in seconds thanks to the interactive window.

Band Pass

Band Pass consists of a combination of two filters: a high-pass filter and a low-pass filter which together form a variable width band-pass or band-reject filter. The cut-off frequencies can be set individually. This allows creations ranging from "telephone"-effects to filter blends of all kinds - even separately for each channel in the stereo version!

Volume 2

Freeze

Freeze lets you select a 3-second sample from an audio source and create up to 32 loops within that 3-second sample. Treat your loops as you wish.

Reson

Reson consists of up to 128 resonant high-pass, low-pass or band-reject filters, each of which resonates a single frequency and produces an ensemble of very soft resonances in an input signal.

Delays

Delays features a group of up to 128 variable delays, each controlled in its amplitude and timing.

Doppler

Doppler simulates the effect of a sound moving towards or away from you, similar to an ambulance passing by.

◀ *Delays*



Doppler ▶

WARP VST



Virtual Guitar Amp Collection

Imagine a big guitar sound with all its pure emotion and raw energy. Most guitarists envision the classic guitar set-up - usually involving glowing valves and a wall of cabinets. Now you can get that sound on your computer.
Warp VST, Steinberg is a VST 2.0 guitar amp simulator plug-in that allows you to combine one of three legendary amp models with any one of three speaker cabinet models. The sound characteristics of amps and three speaker cabinets have been authentically recreated in software, using Hughes & Kettner's groundbreaking DSM (Dynamic Sector Modelng) technology. When combined with an ASIO I/O solution, Warp VST is practically latency-free allowing you to laydown your guitar in real-time processed with your favorite classic guitar amp combination.

FEATURES

- With a few mouse clicks you can combine any off the 3 virtual amp models with any of the 3 virtual speaker cabinet models
- Each amp and cabinet model reproduces the same sound altering components of the hardware originals
- Full automation capabilities
- Gives you total control over a guitar sound even after the guitarist has left the studio.
- Extremely low latency thanks to Steinberg's ASIO technology. When combined with the AudioLink I/O, Warp VST turns your computer into a powerful guitar amplifier rig with an imperceivable delay between plucking the strings and hearing the signal.
- VST recording systems such as Cubase or Nuendo will allow you to monitor the signal in realtime through Warp VST while recording the untreated signal. This provides you with the flexibility of deciding on your amp/cabinet combination and their settings during mixdown.

Dynamic Sector Modeling (DSM)

- Developed by Hughes & Kettner, DSM represents a breakthrough in amplifier technology that allows an exact recreation of the components and materials of three of the most sought after guitar amplifiers
- Each amp offers its own individual dynamics and distinctive characteristics, selectable with a single mouse click.
- Combines the user friendliness and flexibility of digital technology with the sound quality of a leading analog amplifier.

Three Virtual Amps

- The legendary Jazz Chorus clean sound
- The dynamics of a 60s Plexi Tube Head
- The modern HiGain Chunk-Sound of the Rectifier Era
- Each amp model has the exact gain structure and EQ controls as the originals

Three Virtual Speaker Cabinets

- Combo (12″ speaker, open housing)
- British (4x12″ speakers from the 80s)
- Greenback (4x12″ speakers that support those powerful riffs)



Guitars and Computers

Please note when recording a guitar: Although an electric guitar has a large dynamic range, the output is relatively weak. And since standard soundcards don't have a high-ohm guitar input, you don't get good-sounding results when connecting a guitar directly to a soundcard.
To get the best sound possible, you need to convert the guitar signal in high quality. This can be done with a preamp which brings the guitar signal up to "line in" strength. You can also use an effect pedal that is switched to bypass mode. Some I/O solutions also have instrument inputs with which you can get very good results, such as MindPrint EnVoice, Aardvark Interfaces, Edirol USB interfaces. Important: The guitar signal has to be recorded in your computer as cleanly as possible. Please adjust your recording level accordingly to avoid distortion.

STEINBERG

HALion 2.0



32-bit Virtual VST Sampler

What's your dream sampler? A sampler that is seamlessly integrated into your computer. No rack space, no MIDI or audio cabling mess, no set up problems. Total recall inside each song you do. Up to 16 instruments at a time, assignable to 4 stereo, 4 mono and 5.1 surround outputs – giving you a total of 18 outputs. Easiest handling thanks to drag & drop. Pristine sound quality achieved through 32-bit/96kHz support. Advanced memory management, handling sounds of up to 20GB on any computer system and sample libraries in any popular format including AKAI, EMU, E-magic ESX-24 and Tascam GigaStudio.
So stop dreaming— it's here and it's called HALion 2.0—never before has a software sampler been more intuitive and productive.

FEATURES

- Automation and control effects via MIDI controller data or simply tuning the samples to the right key
- Waveloop editor shows how a modern software sampler should perform. A sophisticated crossfade function helps to set the optimum loop
- Zero crossings are automatically identified and the non-destructive editing saves faulty edit steps. Samples can be pitched, modulated or reversed.
- Creating complete programs and layers in the keyzone window is fast and easy with HALion 2.0's intuitive editing facilities
- Global presets for all envelopes and modulation settings for quick loading of individual settings for each program
- Loading samples into HALion 2.0 is a simple drag and drop-job directly to the keyzone window. The samples appear graphically on the screen. Via resizing and moving the keyzones, velocity and layers are set.
- 18 freely assignable outputs per instance (4x stereo/ 4x mono, 5.1 Surround)
- Pristine sound quality is achieved through • 8-/16-/24-/32-bit file support
- Various filter types (Notch/Hi-Pass/Lo-Pass/Band-Pass) with selectable 12 dB or 24 dB slopes as well as new Waldorf filters and "fatness" function provides opportunities for breathtaking sound experiences.
- Filter and settings can be made individually for each sample (and/or globally for the whole program)
- 16-way multi-timbral with an individual editor for program selection, 128 programs per instance, unlimited layers per program
- 256 Voices per instance, several instances loadable simultaneously
- Full individual parameter set for each sample, ideal for drum sounds
- Extensive, chainable modulation routing capabilities
- 2 synchronizable envelopes with up to 32 freely editable points and 2 LFO´s
- Virtual keyboard for auditioning samples
- WAV, AIFF, AKAI (S1000, S2000, S3000), Roland, and EMU (3/3X, ESI/4/4K, E64/E6400, ESynth and Ultra) import functionality, plus SoundFonts 2.x, GIGA, LM•4, LM•4 MarkII script import, SDII (Mac only) and REX file formats, ensures HALion 2.0 is optimally equipped for the world of sampling CDs.



Modulation/Tune Page for various settings

- Seamless integration within your VST 2.0 compatible host application including; sample-accurate playback timing, complete automation capabilities and extremely low latency thanks to Steinberg´s industry standard ASIO technology. All this makes HALion 2.0 the perfect sampler for optimizing your workflow within the VST system.



The Macro Page

HALion 2.0



Waveloop Page

- Streaming of samples from disc: GBs of sounds independent of RAM limitations
- MegaTrigg function to activate samples via controllers or keys and trigger them with NoteOff or Pedal command for perfect GIGA import
- Unprecedented navigation system (diagonal using a virtual "Navigation-ball") and drag and drop makes working with your samples faster and easier than ever.
- HALion 2.0 includes 4 CDs with over 1.6GB of sample content by sample experts Wizoo and E-Lab:

➢ XXL Acoustic Piano,
➢ XXL Nylon Guitar • XXL Bass
➢ XXL Drums & Percussion
➢ Clavinets ➢ Clean Guitar
➢ Electric Pianos ➢ Electronic Drums
➢ Synth Basses ➢ Analog Chords
➢ Digital Decays ➢ Pads
➢ House Organ ➢ Osc Toolbox
➢ Erazor FX ➢ Organs
➢ DrumTools
➢ LoopTools 65 bpm - 170 bpm
➢ MusicLoops 90 bpm - 170 bpm
➢ SoundTools Samples
➢ SoundTools Vox

HALion STRING EDITION Vol.1

Symphonic String Orchestra Instrument And Library



The ultimate symphonic strings collection for computer-based music production, the 5GB HALion String Edition Vol. 1 gives you a top quality orchestral string section (8 double basses, 10 cellos, 12 violas and 16 violins), with outstanding playability, a unique, warm character and a natural ambience. A specially-developed VST- and DirectX- compatible player is bundled to provides a direct connection to production systems like Cubase SX/SL and Nuendo. Musician and composer, Claudius Bruese, who helped in the development of the legendary VST piano, THE GRAND, also played a significant part in the production of the HALion String Edition. This software gives the precision in reproducing acoustic instruments, the superior sound quality and clarity, as well as the playing characteristics users around the world have come to expect of products bearing his name.

- 8 CDs with over 5 GBs of sampled string instruments of superior sound
- Complete orchestral strings - 8 double basses, 10 cellos, 12 violas and 16 violins
- Suits all musical styles from classical to contemporary; playing styles such as legato, tremolo, pizzicato or spiccato
- Expressiveness, full dynamic range, up and down bowing
- Natural concert hall ambience, true key release action
- Each section is provided in two versions you can combine freely with other sections.
- Top recording techniques with unobtrusive, natural ambience deliver outstanding sound quality with a warm character, well defined tones and balanced bow sounds
- A multitude of presets make arranging and combining sections a breeze.
- Specially developed VST- and DXi-compatible HALion String Player allows full control over articulations such as crescendo, bowing or portamento.
- Additional HALion programs allow creative editing and limitless combination with other sounds in HALion.

STEINBERG

LM•4 MARKII

32-bit VST Drum Module —The Ultimate Drum Machine

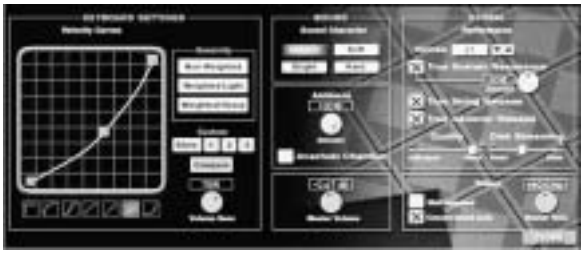


The LM•4 MarkII, a VST plug-in, recreates another of the classic beat-boxes of the early 80's. Heard on many a hit record of that period, the original was way too expensive for most musicians. Today, Steinberg delivers a virtual recreation at a fraction of the price. This was also achieved while combining intuitive handling and maximum performance yet keeping CPU load to a minimum. In addition, the LM• 4 MarkII comes with a huge range of high-quality 24-bit drum and percussion sounds that cover the most diverse music styles, from Latin and Rock, to House, Electro and Drum'n'Bass. The multiple audio outputs are routed directly into the VST system's audio mixer, ready for EQ and effects. Functions such as Drag & Drop and velocity switching make creating realistic and dynamic grooves a breeze. This makes the LM•4 MarkII the most user-friendly drum module around.

- 32-bit drum module with 18 pads per set
- ADSR envelope and reverse mode per pad
- Precisely scalable volume envelope with Attack, Decay, Sustain and Release (logarithmic, exponential, linear)
- BitCrusher for creating the time-honored vintage sound (powerful lo-fi effects)
- 12 outputs (3 stereo and 6 mono assignable) for each instance of LM•4 MarkII you have loaded in your virtual rack
- Velocity, pitch and panning for each instrument (automatable)
- Create your own sets
- The complete drum module and all settings can be saved with the song (Cubase VST)

THE GRAND

Virtual Concert Grand Piano





The challenge was to develop a virtual concert piano equal in quality to an original acoustic piano, even including the "unmatchable" playing characteristics. The result, The Grand - a top-quality virtual concert piano with a breathtakingly rich and realistic sound, whose tonal precision and outstanding dynamic qualities rival those of a real, quality acoustic concert grand piano.
The Grand is not a digital piano, nor is it a ROM player that reels off piano sounds. Rather it is a perfectly simulated grand concert piano based on the sounds from a top-flight acoustic grand piano—guaranteeing you powerful bass, crystal clear mid-range and brilliant treble. Easily record The Grand in the highest quality without the hassle of tuning and miking—and with perfect integration into the VST system.

- Reproduces the thrilling experience that is a large, grand piano. There are no loops or audio compression. Absolutely realistic, authentic sound based on an intelligent VSTi engine optimized for a concert piano
- Various performance optimization features to get the appropriate balance between sound quality and system performance
- Playable with any MIDI keyboard
- Authentic functionality of sustain and sostenuto pedals
- Complex variation of different sound characteristics such as dynamic response, dying out of notes, hammer action and string resonance
- Free variation of acoustic ambiance thanks to complete separation of piano and room
- The Grand allows you tailor it perfectly to your needs: freely programmable velocity curves, four master tembres (Natural, Soft, Bright, Hard), and well tempered or concert grand tuning.

STEINBERG

MODEL•E

Classic Virtual Analog Synthesizer



The legendary analog sounds —without the restrictions of a hardware synth. Load the Model•E into your VST-compatible host application and put 16 synths at your disposal. Create sounds that combine the best qualities of analog sound synthesis with amazing timing accuracy, deep basses, sweeping filters, three heavy-duty oscillators. Model•E is triggered through MIDI tracks, allowing you to automate every parameter without MIDI cables or hardware interfaces. The virtual outputs appear directly in the Cubase VST Mixer, so you can treat its sounds with virtual effects and EQs.

- Exclusive sound banks created by synth gurus Hubertus MaaB and Wolfram Franke
- Make your own sound banks. Model•E sounds richly analog, but offers sample accurate timing for dynamite precision.
- Up to 64 Voices per instance
- 8 outputs per instrument
- 128 memory slots per instrument
- 16 multi-timbral parts
- All Model•E settings are automatically saved with your song files and the final mix can be streamed to an audio file

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PLEX *Restructuring Synthesizer by Wolfgang Palm*



PLEX is a synthesizer based on an absolutely novel synthesizing process. This process offers you almost limitless possibilities in varying and creating sounds. The original sound is split into four components, lower spectrum, higher spectrum, filter characteristics and amplitude envelope, using a new audio analysis technique. Each component can then be replaced and combined with other elements. An example would be combining the sounds of a trombone and a sitar with the envelope of gong. Just drag the Base part onto the trombone, the Top part onto the sitar and the Filter part on the gong. Doesn't just sound great, it's also surprisingly simple. It's not only the sounds you can create with PLEX that are revolutionary, the innovative and clearly structured user interface also sets new standards in intuitiveness and ease-of-use.

- Developed by synth legend Wolfgang Palm in cooperation with Steinberg, PLEX is based on a new synthesizing technique that gives musicians almost limitless possibilities in varying and creating unique new sounds.
- Independent manipulation of four integrated sound components: upper and lower frequency spectra, filter characteristics and amplitude envelope, while the information contained in each of the components is strictly separated from that contained in other components.
- Each component can then be replaced and combined with components from other original sounds. This means that, for example, the lower spectrum, or "Base" component contains neither amplitude envelope information nor filter characteristics. Replace any component you wish and still produce a "natural"-sounding result.
- Real-time swapping between components of 33 sound sources
- Up to 64 voices, 300 presets included
- 97 special pre-analyzed sound sources - acoustic, synthetic and filter characteristics
- Modulation section with stereo delay and flanger effects
- 3 ADSRs, Global Pitch LFO as well as 3 LFOs that can be synchronized (besides presets such as sine, square, sawtooth and random, Plex also allows you to create 16-step sequences)
- All parameters are MIDI controllable

STEINBERG

PPG WAVE 2.V

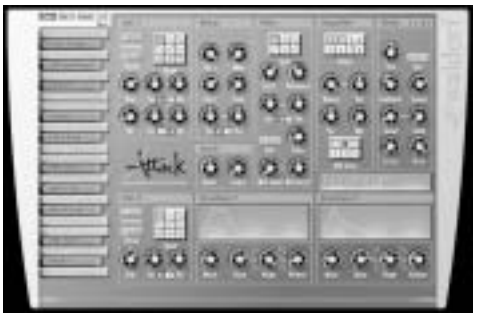


In the 80s, the PPG synthesizers and their wavetable synthesis technology could be heard on countless records. Their distinctive sound was an important influence in many music styles. With the PPG Wave 2.V, Waldorf and Steinberg offer an authentic recreation of the legendary PPG Wave 2.3 synthesizer as a VST instrument. That classy PPG sound is now more affordable than ever and can be seamlessly integrated into today's computer-based production environments. The photo-realistic recreation of the original PPG look is combined with a modern audio engine with superb audio qualities, simple user interface and full MIDI automation.

- Using wavetable synthesis is easy because it isn't much different from normal subtractive synthesis. Creating new sounds is as easy as replacing one wavetable with another. No other form of synthesis offers such a simple way of creating entirely new sounds.
- Almost 2000 waveforms (32 wavetables each with 64waveforms) recreate the exclusive sound of this classic synthesizer—with all the advantages that an integrated software production environment such as Cubase VST and Nuendo have to offer.
- Includes over 700 pre-programmed sounds
- Outputs appear in the VST mixer, which means that it can be routed through effects and EQ's.
- MIDI tracks are sent directly to the PPG Wave 2.V, making cables and MIDI interface completely unnecessary.
- Graphic display for envelope generators and Cutoff/Emphasis allows quick and simple sound editing
- Low Pass filter with Emphasis control
- Built-in Arpeggiator
- Two oscillators per voice
- The number of voices (up to 64) is limited only by CPU power. Up to 8 instances of PPG Wave 2.V can be used at once (and each unit is 8-way multi-timbral), giving you more than enough creative power.
- Realtime MIDI control and automation of all parameters

Continually setting new standards in sound quality and functional extras, Waldorf has had a decisive influence over the development of electronic music instruments, and created new sounds which have in turn, helped shape modern music. For over 20 years now, Waldorf has been developing innovative instruments such as microWAVE, Q, the avante-garde WAVE and the classic analog Pulse, allowing you to realize your musical ideas without any limitations.

WALDORF ATTACK



VST Percussion Synthesizer

A true percussion synthesizer, Attack seamlessly integrates the classic visceral 80s analog sounds and the electronic club drums of the 90s into the VST environment of the 21st century. Attack has 24 sounds available per percussion set, spread over two octaves. It requires neither MIDI nor audio cabling, however, all parameters are addressable via MIDI controller, and it offers absolutely sample accurate timing. Treat your ideas to the Attack's sublime sounds. Your groove deserves it.

- Two oscillators, each with nine waveforms (triangle, sine, square, sawtooth, sample and hold, noise, hihat closed, hihat opened, crash), represent the sources.
- Two integrated and synchronizable Modulation Delays for the stereo outs
- All parameters addressable via MIDI controller
- Percussive bass and lead sound– polyphonic playable
- Eight audio outputs (2 stereo, 4 mono)
- Twelve sounds of a drum set can be played melodically. With this you can create tom fills, conga grooves or other percussions in no time. You can even play bass, melody or sequencer lines.
- FM and Ring Modulation expand the sound spectrum with a metallic component and FX sounds:

– 'Crack'-module for authentic analog hand claps

– 6 filter types with resonance up to self-oscillation and overdrive up +52dB

– 2 envelopes for pitch, FM amount, filter cutoff and loudness

B&H PHOTO - VIDEO - PRO AUDIO

STEINBERG

VIRTUAL GUITARIST

VST Rhythm Guitarist

It's always the same problem: Try calling a professional guitarist in the middle of the night when you need a slick 12 string accompaniment for a ballad or a hard metal groove for an advert soundtrack. Introducing Virtual Guitarist - a perfect rhythm guitarist who plays both acoustic and electric guitar, including all keys and difficult chords, never gets impatient, follows the tempo of your song, sounds better, always plays 100% tight, and is always completely in tune. Of course, the Virtual Guitarist doesn't need to set up his equipment or be miked up, and integrates perfectly into your VST system.



- Acoustic and electric guitar performances by 27 different players with 8 parts per player
- Includes a variety a popular guitar sounds, phrasing and styles including Spanish, Steel String, Resonator, Clean Strat, Wah, power chords and ultra metal
- Intelligent fret noise for 100% authentic guitar sound
- Follows the tempo of your song
- Playable through MIDI, this VSTi creates authentic, professional guitar tracks for all styles—from country to metal.
- Flexible real-time variation (sound, phrasing) with keyboard, mod wheel, Aftertouch, Velocity
- Variable playing - Shuffle (groove), dynamic, timing (tight/loose), long chords, syncopated rhythms

VIRTUAL GUITARIST "ELECTRIC EDITION"



29 different players, variety of guitars/amps



VST Rhythm Guitarist and Effect Sound

Virtual Guitarist "Electric Edition" is the perfect rhythm guitarist for anyone needing electric guitar tracks. This VST instrument is based on real recordings, yet is as flexible as a synthesizer. Featuring 29 new all electric players from the 50s to the present day (recorded by top guitarist Thomas Blug), Virtual Guitarist "Electric Edition" oozes authentic sound character while offering unrivaled playability and outstanding tempo flexibility. Also includes a fully featured multi-effect board, useable both as a separate plug-in that you can use on any audio signal or integrated in the VSTi. Select a player, some chords and it plays perfect tracks. Liven songs up with options like syncopation, phrasing variations or long chords. Change shuffle or dynamics, use track doubling or be creative with the supplied effects.

- Vast range of electric guitar styles for all kinds of music
- Variable playing - shuffle, dynamics, timing, syncopation, long chords
- Stereo width, track doubling and other sound shaping options
- Flexible real-time variation (keyboard, mod-wheel, aftertouch, velocity
- Effects integrated into VSTi or separately usable as VST plug-in
- Effects: Wah, AutoFilter, Chorus, Flanger, Phaser, Delay, Reverb

STEINBERG

GROOVE AGENT

COMPUTER AUDIO

1030

VST Virtual Drummer



Groove Agent is a stunning VSTi that provides you with ready-to-go drum rhythms in only a few mouse clicks. Groove Agent can play the hottest, most popular and influential styles from the past 50 years of music history - inside your VST host application. Based on quality drum samples most of which were recorded especially to analog tape, Groove Agent puts a top studio drummer at your fingertips.
Creating your own drum track couldn't be easier - just choose a music style, a drum kit and you're off. Set the amount of drum room ambience, play fills and half-time breaks. Groove Agent can play at over two dozen different complexity levels, so you can create the right mood with your rhythm track simply by moving a slider. Adjust volume, ambience, decay, tuning and velocity response individually for each instrument.

- Over 50 musical styles (each with their own drum style patterns), programmed by top Swedish musicians
- Combine any style with any drum kit - instantly
- Four unique drum kits plus percussion and numerous electronic drum sounds
- Separate dry and wet (ambience) samples and 4 stereo outputs for total flexibility
- User-configurable drum kits, including tweaking of sounds
- 24-bit drum sounds, most of them recorded via analog tape
- Up to two dozen complexity levels in each style, including fills and half tempo feel variations
- Combine any musical style with any preset kit
- Fine-tune the overall feel with the Shuffle and Humanize controls
- Save snapshots of your tweaks for easy recall
- Anyone who can't do without their own drum sounds but wants Groove Agent to play their "live" drums for them can simply record the MIDI output to a GM-compatible drum track and then use their drum sample player of choice.





D'COTA

Multiple Synthesis VST Instrument

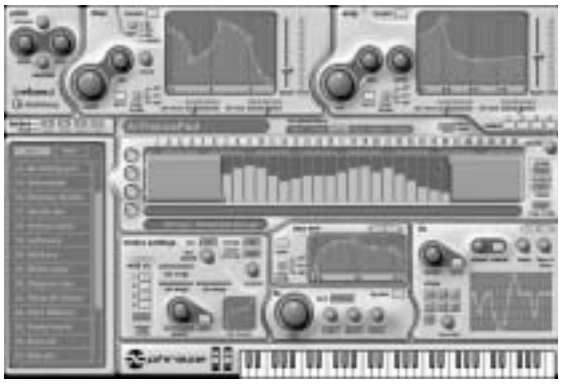
Imagine the best synthesizer you can and then multiply that by three. Why? D'cota combines the power of three types of sound synthesis in the compact editing environment of one synthesizer. All this in an unprecedented sound quality for all three synthesis types, thanks to the aliasing-free sound engine. Each of the three synthesis types offered by D'cota has access to an arsenal of four freely assignable ADSR envelopes and two assignable LFOs. The LFOs can also be synchronized to the song tempo. The effect section adds the simultaneous power of delays, distortion units and modulation effects. This exceptional synthesizer opens new possibilities to create extraordinary sounds.

- 3 different synthesis: Advanced Analog, Spectrum and Wave Impulse
- 8 times multi timbral with up to 128 voices, 4 stereo outputs
- Advanced Random Mode
- Superb sound quality (aliasing free oscillators)
- Multi FX unit with Distortion, Modulation and Delay section
- Ready to start sound preset included

B&H PHOTO - VIDEO - PRO AUDIO
EQUIPMENT LEASING AVAILABLE

XPHRAZE

VST Phrase Synthesizer



Designed for computer-based musicians, arrangers and sound designers, Xphraze brings together two worlds that have been separated for too long: powerful sound shaping and real-time polyphonic phrase creation.
Easy to use, Xphraze lets you create rave chords, complex polyphonic soundscapes, bass and lead riffs, sweeping pads and tempo-synced sound effects. Simply drag a phrase from the pool, tweak the groove and sound, play a note or a chord and listen as Xphraze textures kick in and move perfectly in sync with the song. You can even drag in your own samples and let Xphraze do it's magic to them.
But don't stop there -- you can also use real-time controllers and other sound-mangling tools to change the phrasing or sonic character in ways that were previously unimaginable.
At the heart of this sample-based instrument is a flexible multi-timbral synthesizer engine, offering extensive sound-shaping and real-time modulation possibilities. Powered by an integrated polyphonic phrase generator, Xphraze is capable of conjuring up any kind of rhythmical phrases like polyphonic chord patterns, drum grooves, bass lines and even tempo-synced synth textures ranging from pads to arpeggios.

FEATURES

- 4-part multi-timbral polyphonic phrase synthesizer, full loop and tempo-sync capability (phrase oscillators, envelopes and LFOs).
- Each Timbre consists of: Phrase Oscillator (32 cells) with individual note resolution, amp section and multimode filter (low-pass, high-pass, band-pass and notch filters with 12-72 dB slopes) with freeform loopable envelopes, two additional freeform loopable auxiliary envelopes, two freeform LFOs, arpeggiator, unison mode.
- Each Phrase Oscillator contains individual waveform per cell, plus a maximum of six assignable cell parameters, such as pitch, cutoff, resonance, pan, level and more. Cell parameter curves are freely mouse-editable.
- Unlimited, freely assignable modulation potential with full morph capability (one controller controls multiple parameters at a time, which can be adjusted individually).
- Phrase playback can be fully adjusted for each timbre: play and loop range (start/end) can be adjusted on-screen, switchable trigger/mute per cell, adjustable cell crossfade/duration.
- Performance controllable stereo insert effects for each timbre from 24 effect types, plus two master effects from six types.
- Sample-accurate audio engine with 32-bit floating-point resolution, 192kHz sample rate, and 1024 voices/notes (256 per timbre).
- Four assignable stereo outputs with vector synthesis -- output mix can be controlled via any assignable modulation source.
- Supports VST Instruments, DXi and Audio Units; compatible with Windows (98 and higher), Mac OS 9 / OS X
- Includes a pool of 256 preset phrases and 256 user phrases.
- 500 MB Wave ROM with more than 200 waveforms.
- Imports AIFF and Wave samples, automap function for multisamples
- Integrated mouse-controllable keyboard and wheels.

BPM Force



A software bundled designed for dance producers, BPM Force includes Cubase SL with three Waldorf instruments and filters to offer everything dance music producers need to make those crowd move.

Cubase SL includes the A1 synth for hammering bass lines. Attack delivers the beats - this percussion synthesizer integrates the analog drum sounds of the 80s and the massive club drums of the 90s in your VST System. PPG Wave 2.V, the second coming of a synth legend as a VST instrument, offers everything from luscious synth pads to manic techno sounds. Last but not least is D- Pole, with its 5 filter types that can morph any run-of-the- mill acoustic drum loop into a heavy-duty dance groove. Available for PC and Mac OS X/9.

COOL BREEZE SYSTEMS

COOL SCHOOL INTERACTUS CD-ROMs

Software and Hardware Media Systems for the Creative Mind

Cool Breeze's CSI (Cool School Interactus) series of CD-ROMs provide an easy-to-use and intuitive learning environment. The structure of the CSi CD-ROMs allows the user to go at his own pace. A flexible interface encourages each user to customize his learning path depending on his personal needs and desires. The novice through the veteran will find great benefit in using these CDs. Made up of system designers, engineers, acousticians, programmers, musicians, composers and graphic artists, Cool Breeze offers over 20 years of combined experience in digital audio operations, training, sluggin' brews and systems design.

Vol. 1.1 Pro Tools Basics



Provides Pro Tools basics and a broad fundamental education on the operation of Digidesign digital audio workstations. It's beneficial to anyone from the complete novice to the partially educated, to the old-school linear pro. This CD features many Digidesign products, and covers hardware/software installation procedures and detailed system configuration options. Content covers basic concepts, operation and functionality of Pro Tools, Sound Designer II, and SampleCell II, also sound fundamentals, computer basics, MIDI, and digital audio, Further education includes modules on System Upkeep and Synchronization.

Vol. 2.1 Pro Tools Tips and Plug-ins



Features Pro Tools 4 software from Digidesign and selected 3rd-party Development Partners. It includes screen element definitions with click-state pop-up simulations for Pro Tools 4 and plug-ins from Antares, Digidesign, Focusrite, and Waves. There is a massive glossary of DAW related terms, including DAE errors and over 50 movie tutorials that include basic functionality, production techniques, special short cuts, key commands, and before and after audio examples. Over 150 quick keys and shortcuts are covered. No special hardware needed, just a compatible Mac/Windows machine. Pro Tools and CSi can be open at the same time.

Additionally, Vol. 1.1 and 2.1 include the "CSi AutoPlay" which launches CSi into an "autopilot" mode. Sit back, crack open a cold one, and impress your friends as you soak in the info. The "AutoPlayer" can be disengaged and resumed at any time.

Vol. 3 Desktop Audio



A powerful guide covering the basics of sound production on a computer, whether you're into music, post-production or multimedia. Learn the many concepts, elements, products and possibilities - and gain the knowledge to put together the system that's right for you. Topics covered: goals (where the authors help you state your mission, and explain & categorize the different types of software and hardware), sound, computers & operating systems from the audio perspective, MIDI, digital audio, DAWs, plug-ins, synchronization, system set-up & upkeep, even studio furniture, and much, much more. Includes a searchable glossary of over 1,200 digital audio related terms, and is packed with swingin' music, top-notch graphics, excellent interactive examples, flow charts, pictures, animations, before and after audio examples, video, diagrams, and info-packed movie tutorials!

Vol. 4 Logic Audio Training



Whether you're driving with Silver, Gold or Platinum, this multimedia CD demystifies and opens up the vast capabilities, power and flexibility of Logic Audio 4. Get hip to the basic concepts, operational ins & outs, and new features. Vol. 4 combines the Cool Breeze vibe with the practical expertise of Logic Audio pros. CSi's tutorials include basic functionality, and real-world production techniques. Topics covered: Logic Audio applications, system requirements, hardware & software configurations & set-up, concepts of Logic Song, Audio Objects, Instruments, Ports, Screen sets and much more. Includes a searchable glossary of over 1,200 digital audio related terms, over 40 movie tutorials, loads of graphics, music, flowcharts, video, before-and-after audio samples, and effective interactive examples (software simulations with pop-up and clickstate definitions).

COOL BREEZE SYSTEMS

COOL SCHOOL INTERACTUS CD-ROMs

Vol. 5 Pro Tools 5



Whether you're driving Pro Tools FREE, Mbox, Digi001, Digi003, or a big bad phat TDM rig, this CD will help you get the most out of your Pro Tools system. Sit down with a DAW pro and learn about the new features available in Pro Tools 5. This CD includes screen element definitions with click-state pop-up simulations for the Edit Window, Mix Window, Transport and Menus. There is a massive glossary of DAW related terms, including DAE errors and over 50 movie tutorials which include not only basic functionality, but production techniques, special short cuts, key commands, and before and after audio examples. Movie Tutorials include the Smart Tool, Beat Detective, MIDI Controls, Loop Record and more.

Vol. 6 Digital Performer V3



This Digital Performer training CD-ROM covers the core operations of DP version 3. Sit down with a DAW pro and learn concepts, production techniques and the latest features from Digital Performer 3. This CD-ROM includes screen element definitions with click-state pop-up simulations for the Transport, Mixer, Sequence Editor, MIDI Graphic Editor and Menus. There is a massive glossary of DAW. related terms, and over 40 movie tutorials. The Getting Started topic will take you through set-up, first record, edit and mix. The CSi movie tutorials are designed to include not only basic functionality, but production techniques, special short cuts, key commands, and before and after audio examples.

For Vol. 5 and 6, no special hardware needed, just a compatible Mac/Windows machine. If you own a Pro Tools setup or Digital Performer rig, CSi can be open at the same time. View a movie tutorial in CSi then instantly go to the software and try it yourself. Additionally, Vol. 5 and 6 includes the "CSi AutoPlayer" which launches CSi into an "autopilot" mode. Sit back, relax, drive with no hands and impress your friends as you soak in the info. The "AutoPlayer" can be disengaged and resumed at any time.

Vol. 7 Cubase SX



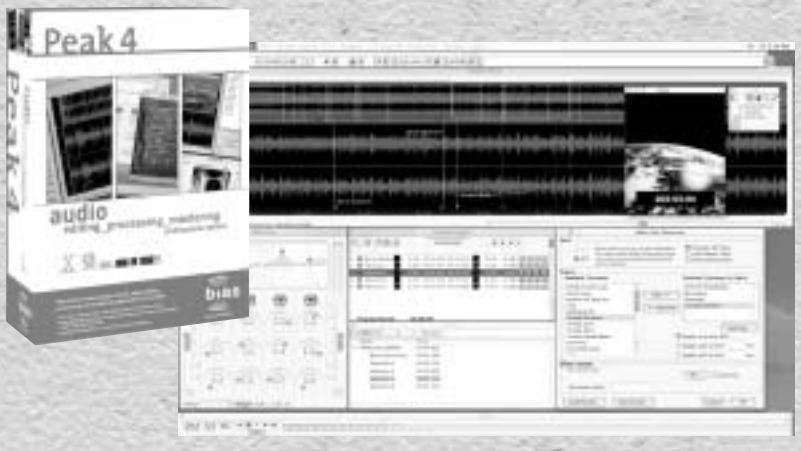
Covers key operational techniques and the cool features of Cubase. If you prefer the "Show me" style of learning, then CSi is for you. Sit down with a DAW. pro and learn concepts, production techniques and the new options available in Cubase SX. This CD-ROM is for Mac and Windows users, and covers the Project Window, Transport Panel, Key Editor, Drum Editor, Track Mixer, Sample Editor, Pool, and Menus. The Getting Started topic will take you through set-up, first record, edit and mix. The CSi movie tutorials are designed to include not only basic functionality, but production techniques, special short cuts, key commands, and before and after audio examples. No special hardware needed, just a compatible Mac/Windows computer.

Vol. 8 - Pro Tools 6



Covers key operational techniques and the cool features of Pro Tools 6. Sit down with a DAW pro and learn concepts, production techniques and the new options (DigiBase, new MIDI options, Beat Detective, DigiGroove templates, and importing session data are just of the few) available in Pro Tools 6. The Introduction topic will take you through set-up, first record, edit and mix. The tutorials are designed to include not only basic functionality, but production techniques, special short cuts, key commands, and before and after audio examples. No special hardware needed, just a compatible Mac/Windows computer. This CD adds to the information covered on Pro Tools Basics, Pro Tools Tips and Plug-ins and Pro Tools 5.

In addition to the hours of movie tutorials, Vol. 7 and 8 include a massive DAW related glossary and utilize the CSi MovieTutorial interface to provide an easy to use and intuitive learning environment. Plus, the new CSi MT interface includes a title bar that allows you to easily position the interface on your screen. If you're lucky enough to have a big fat monitor or maybe two, you can smoothly switch between Cubase SX or Pro Tools 6 and the CSi tutorials.


Peak 4
audio






ACOUSTIC ESSENTIALS


STREET BEATS
by boogie bell


HIP-HOP/R&B

B&H

PHOTO - VIDEO - PRO AUDIO

The ProfessionalAudio SourceBook

Section 11c

Computer Software

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ABLETON

LIVE 2

Audio Sequencer and Multi-Track Recorder



Live 2 is an audio sequencer that you can play like an instrument. Whether on its own or with other musicians or DJs, live on stage or when remixing in the studio, all you need is Live 2 and a Mac (OS 9.1 or OS X.1.5 or higher) or Windows PC (98/2000/XP). Live 2 allows you to bring together audio material from various sources. All the samples used are adjusted to the tempo of the song in real time and without changing the pitch. Even long pieces with tempo variations play in perfect sync. Samples can be played back and new ones can be recorded using the mouse, computer keyboard or MIDI notes. Drag-and-drop can be used to put together any sequences of effects for sound editing - all this while the music continues to play. Everything you do during the session is recorded and can be post-edited in detail, including all the automation of the mixer and effects.

FEATURES

Truly Elastic Tempo and Pitch Manipulation

- Record, import and arrange multiple audio loops into complete songs—all in real time— regardless of pitch or tempo.
- Synchronize loops and recordings of any length at any time.
- Time stretching can be applied to loops or to any audio material using time stretching algorithms optimized for rhythmic and melodic material.
- Live also allows key (pitch) changes independent of tempo throughout the production process — match the key of a bassline to a vocal without losing sync.
- Audio phrases, loops and performances can be quantized in real-time

Linear Tape-style Recording

- Live 2 offers a full blown multi-track recording and editing suite offering extensive real-time effects, dynamic parameter automation and unlimited undo.
- Unlooped, unstretched audio can be captured using linear tape-style recording in conjunction with loop-based manipulation and playback.
- Record unlimited takes of audio on an unlimited number of tracks directly into the Arranger or Session view.

MIDI Control

- Sample clips can be assigned to MIDI notes and controller messages using any of the four selectable clip launch modes: Trigger, Gate, Toggler or Repeat.
- The mixer and effect controls can be assigned arbitrary MIDI notes. Support is provided for the common standards for incremental controllers and MIDI devices with motorized faders.

Audio Interfaces

- Tracks can be individually routed to the outputs of any ASIO or CoreAudio compliant device and inputs can be routed to any Live track. DirectX for PCs and Sound Manager for Macs are also supported.

Seamless ReWire Integration

- As a ReWire client, you can route the outputs of Live into your DAW software (Logic, Cubase, Sonar, Digital Performer and Pro Tools).
- As a host, the outputs of programs such as Reason can be routed into Live's mixer.

Tempo Control

- You can set or tap the tempo of audio recordings completely independent of pitch at any time during recording, performance or playback.
- You can also record and edit tempo as a continuous curve.

Must Have Live Performance Features

- Privately audition audio from disk while the main mix continues have already made Live as adept at live performance as it is in the studio.
- The ability to treat long audio files as elastic, combined with the DJ-style crossfader, make Live 2 an ideal tool for cutting-edge DJs as well as other performance artists.
- Refined MIDI and computer key mapping of loops and phrases for playing sessions.
- Assignable scene advance and track launch buttons help navigate large sets via a MIDI controller box. Transport and tempo controls can all be freely assigned to MIDI controllers.

Elegant, Powerful and Simple

- Combining audio from diverse sources is a simple matter of drag-and-drop and/or real-time recording.
- Full-screen mode allows you to take advantage of all available screen real estate.

Synchronization

- Slaves to MIDI clock or MIDI time-code. This means, when the drum machine slows down, Live also plays audio files slower. It does this without altering the pitch or the groove. Live can also be a MIDI clock source.

LIVE 2

SESSION VIEW

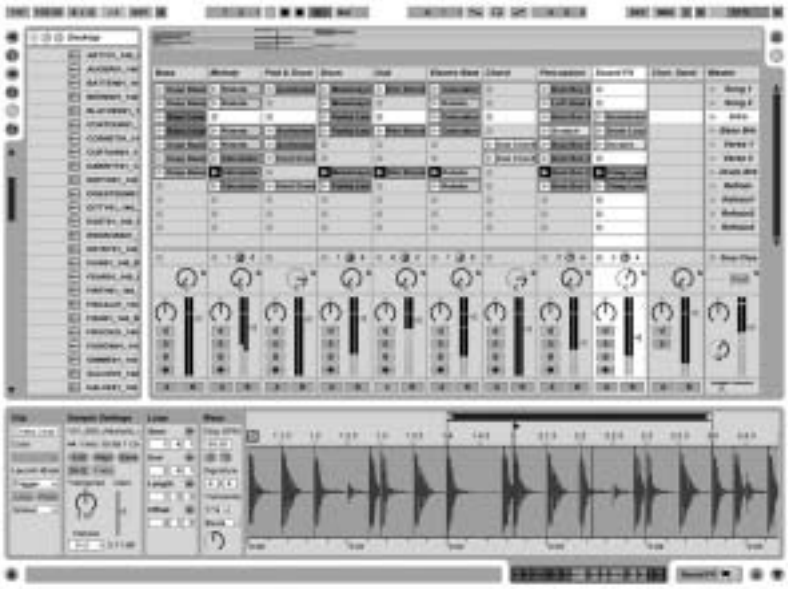
Start your loops and sessions at any tempo, then change tempo later – even as playback continues. The music never stops as you play, record, resample and process your music.

Tap Tempo to sync Live to musicians or the DJ.

Browse your hard drive for samples, Live's built-in effects and VST plug-ins.

Drag & Drop audio clips into your session while the music keeps playing – from short loops to complete songs. All audio automatically stretches to play in sync with the current song tempo. Audition clips from your hard disk via the private headphone send.

Real-time quantization insures that loops are always triggered in sync. Transpose and detune clips without affecting their tempo.



Live captures your jam. Improvise with total freedom, then edit to perfection later.

Launch clips using the mouse, the keyboard or MIDI notes. Operate all effect and mixer parameters via any MIDI controller.

Each row of clips is a scene. Launch all clips in a scene at once or start them independently.

Control up to 4 effect send channels for each track.

Change the playback region of oneshots, loop position, length and downbeat of looping clips – all while the music keeps playing.

Choose among Live's real-time stretching modes for "clean" stretching of all types of audio.

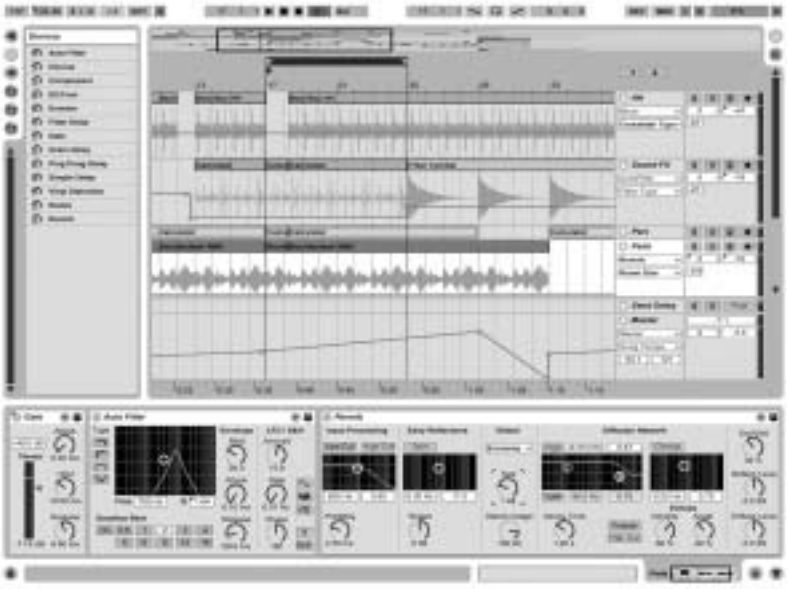
Use multiple inputs and outputs from your ASIO or CoreAudio interface to route audio to and from Live. Connect Live to ReWire master or slave programs to exchange audio and sync.

ARRANGE VIEW

Set, tap or change tempo before, during or after recording. All clips automatically play in sync with the current project tempo.

Live comes with a range of high-impact effects such as: reverb, filter delay, ping pong delay, erosion, gate, grain delay, EQ and many more.

Drag & Drop internal and VST effects into chains for tracks, sends and master in real-time. All effect and mixer parameters are fully automated.



Your entire arrangement is displayed in an overview for quick navigation.

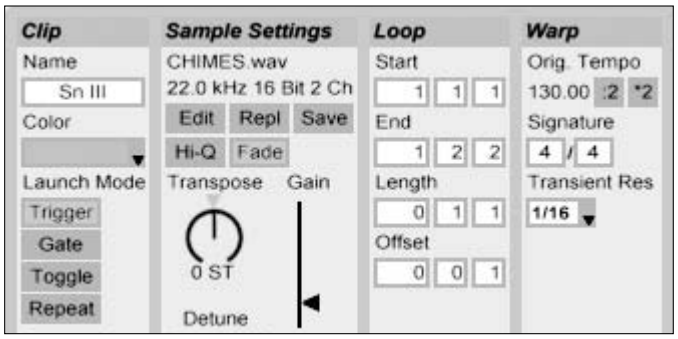
Live can capture every one of your actions in a timeline arranger. Record unlimited takes on unlimited tracks. Edit clips and breakpoint-envelopes, add and remove materials. Render your work after it's finished.

Scale tempo over time to conform scores to film and TV.

LIVE 2 *continued*

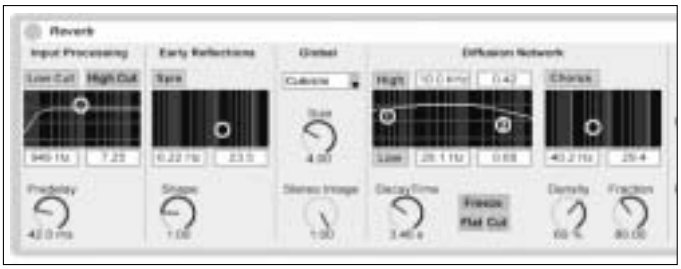
Playing and Recording Samples

- Live 2 lets you compile live sets from any audio clips, ie: loops, phrases, entire tracks. All the clips play in-sync with each other, irrespective of the original tempo. You can "fire off" clips live, using the mouse, assignable computer keys, or MIDI Notes. Real-time quantization can be used to prevent rhythmical error. Any clip can play at any time - you are not constrained to a fixed arrangement.
- Since audio is played directly from disk, there is space enough even for large live sets.
- Recording new clips is as easy as playing existing ones: Play a MIDI note to start the recording; play the note again to jump to playback mode - without any interruption. This way, you can capture loops or phrases from external sources and directly work them into your performance, without stopping the music. Thanks to the real-time quantization, you will always get perfectly cut loops.
- Live can also use it's own Master output as a signal source for recording, allowing you to resample your session while it's going on.



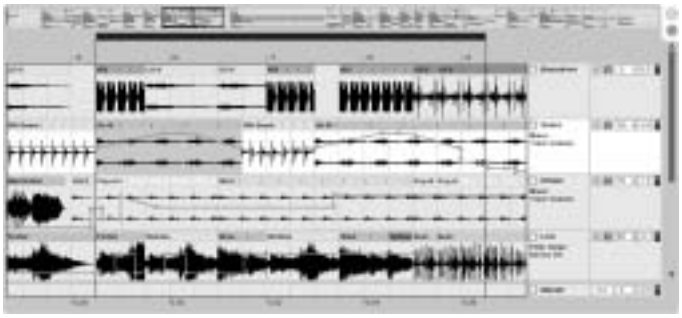
Effects

- Every track, each of four send-channels and the master, can have a chain of effects. Any number of effects can be pasted in and moved around using drag-and-drop, while the music is playing.
- Live comes with a special range of high-impact effects, suitable for subtle and drastic manipulations. Live's effects are controlled by intuitive 2-D displays, which are ideal for live articulation.
- VST plug-ins are also supported. A VST plug-in can have its own editor window, but you can also use an integrated panel with a 2-D display to control the plug-in.
- Live's mixer and all of its effect parameters can be mapped to arbitrary MIDI controllers, and are fully automated - including every VST plug-in parameter.



Browsing Samples and Effects

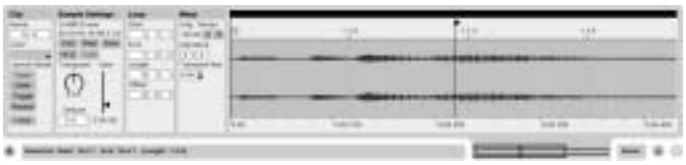
- Samples can be sought in Live's built-in browser, and privately auditioned by headphone, without the public or other musicians hearing them. There are three browsers for samples and Live Sets, one for the Live effects, and one for VST plug-ins.



Arranging Pieces

- Every action carried out in a session can be recorded. The session's protocol is graphically displayed in Live's Arranger view and can be edited, down to the smallest detail. Every movement of the controls is displayed and edited as an envelope curve. Just click on a control to view and edit its automation.
- Editing audio is extremely fast. Snap to Grid does away with having to zoom in on wave patterns to find musical counting intervals.
- Live offers you the ability to improvise arrangements, and to develop improvisations into pieces later on. And should you decide the previous version was better after all, you can undo as many edit steps as you like.

Warping Time



- Live's Time-Warping-Engine stretches and shrinks audio clips as they're being read from the hard disk. The pitch remains unaltered and can be adjusted independently. All clips are automatically synchronized to your session-tempo or to an external sync-source.
- The Time-Warping-Engine is controlled by "warp markers", which can be positioned freely. A "warp-marker" attaches a position in a sample to a particular time in the song. During play-back, the Warping-Engine stretches and shrinks audio clips so that each marker is reached at a precise moment, irrespective of the song-tempo or whether Live is being driven by external synchronization.
- If you only use short loops you won't need warp markers. If, however, you want to work a ten-minute piece into a set, then just position a few markers to line-up the rhythms. The more tempo irregularities there are in the take, the more markers you'll need. With warp markers, you can get even the sloppiest recordings to jive perfectly.

MOOG MODULAR V

Emulation of the Moog Modular System 3C

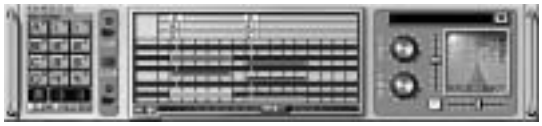
Arturia's Modular system is based on a set of algorithms created by Arturia's sound engineers and gathered under the name TAE - True Analog Emulation. TAE avoids aliasing; it also emulates the typical instability of hardware oscillators and brings the warm sound of the vintage synths. In addition to the replication of all the functions of the original, the Arturia's Modular system comes loaded with more than 400 presets made by a selection of sound designers.

- Stereo synthesizer
- 9 oscillators (921 Mg series)
- 2 dedicated LFOs
- 3 filter slots can be chosen among classic Moog 904 series (low-pass 24 dB, high-pass 24 dB, filter coupler), and an additional 20 dB multimode filter (7 modes)
- 1 noise generator (white and pink — 923 Moog module)
- 6 ADSR auxiliary envelopes based on the 911 Moog module
- 2 VCAs each with their own envelopes, and a panorama knob for natural stereo effects
- 16 auxiliary VCAs with modulation inputs
- Sequencer based on the 960 Moog module with 3x8 steps
- One filter bank (14 bands with their own bandwidth)
- One stereo delay and one chorus
- Mono/polyphonic (up to 64 voices)
- 96 kHz sampling rate support



- Can be used stand-alone or as a virtual instrument in VST, DXi, MAS, RTAS (Windows, Mac OS 9/X), and HTDM (Mac OS 9/X) formats.

STORM 2 *Music Creation Software for Mac and Windows*



Shadow: "analog"-sounding polyphonic synthesizer based on subtractive synthesis

Storm 2 is an all-in-on software package for music composition that accurately emulates the components of an entire recording studio. Storm 2 is comprised of a plethora of modular and realtime sequencers, synthesizers, sample players, drum machines, mixers and effects processors that can be selected and then incorporated into a studio right before your eyes. Once the studio has been created, you can create musical pieces in a fast, easy and extremely efficient manner.
Storm permits the import and export of all popular music file formats including MP3, AIFF and WAV, and it supports MIDI.

Composition

- Program up to 64 patterns for each module
- Songs can be up to 400 bars long
- Record automation moves using the onscreen knobs. Effects sends and returns can be automated.
- Real time synthesis and EFFECTS, a user-friendly interface and a fully automated studio provide you with an unprecedented comfort and facility of composition.
- Record output as a stereo .Wav, .Aif or MP3 files
- Import ACIDized WAV files
- Unlimited undo for all functions
- The COMPOSITION WIZARD helps you quickly understand the general use of the program and the different sides of creating a song. Storm can even help you learn how to compose in different music styles.

The Hall: Share The Experience

The HALL is a set of interactive tools that allows on-line collaboration with others. The Hall offers file sharing and downloading, chat rooms, newswires, tips and tutorials.

Customize Your Studio

- Choose from 13 virtual instruments: Arsenic, Bass52, Equinoxe, EZTrack, H3O+, Hork, Meteor, Orpheus, Psion, Puma, Scratch, Shadow and Tsunami
- 10 different effects modules including: Chorus, Compressor, Distortion, Dual Delays, Flanger, LPFilter, Reverb, Ring Mod, Seq Filter and Vocoder
- Storm 2 allows you to select the specific virtual instruments and effects you wish to use
- Simply drag and drop modules and in a few seconds your studio is ready

ANTARES

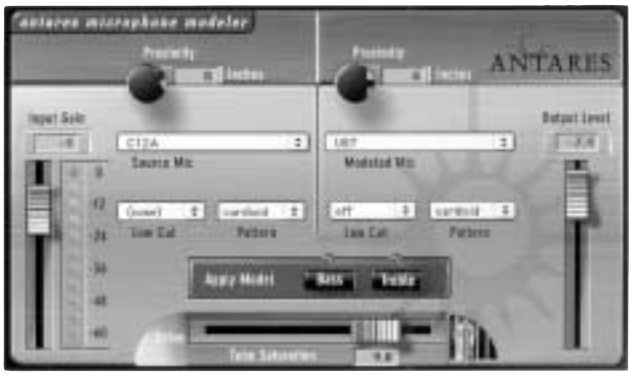
MICROPHONE MODELER

Using Antares' patented Spectral Shaping Tool technology, the Microphone Modeler can create precise digital models of a wide variety of classic and modern microphones. Simply tell it what microphone you are actually using and what microphone you'd like it to sound like. The models not only reproduce the sonic characteristics of each microphone but also gives you control of each mic's specific options including Low Cut Filter settings, windscreen on or off and proximity effect. Available as a plug-in for VST, MAS and RTAS on the Mac, DirectX for PC and TDM.

- Proprietary DSP-based acoustic modeling transforms any reasonable quality mic to sound like any of a ever-growing list of desirable mics
- Mic Models even reproduce the effect of each mic's specific options - windscreen (on/off), Low-Cut Filters, polar pattern and proximity effects
- Create hybrid mics that combine the bass response of one mic with the treble response of another
- Variable classic tube saturation distortion control
- Completely automatable when used with compatible host applications
- Use during mixdown to change the mic on an already recorded track
- Not to be underestimated are the limitless creative possibilities made available - select a Source Mic that doesn't match your physical mic, overdrive the Tube Saturation, over-emphasize proximity effect, etc...
- Incredibly simple to use - simply select the mic you're using and the mic you want it to sound like
- An extensive collection of digital models of are included and additional models can be down-loaded from Antares' web site

ANTARES MICROPHONE MODELER

VST/MAS	RTAS	TDM
(for MAC).....**259.95**	(for PC/MAC) ..**274.95**	(for MAC).....**399.95**



Signal Flow Of Mic Modeler

- The Input Gain fader and Level Meter are used for setting the input level of the audio to be processed
- The Source Mic section lets you select the mic that was actually used to record the audio (with or without windscreen) as well as the state of its various parameters — Low Cut Filter, Polar Pattern and Proximity
- The Modeled Mic section allows you to select the mic whose sound you would like to model (with or without windscreen) as well as the state of its various parameters — Low Cut Filter, Polar Pattern and Proximity
- The Preserve Source controls let you create hybrid mics that combine the bass characteristics of one mic and the treble characteristics of another. Incoming audio is split into its bass and treble ranges, one of the two frequency ranges can then be assigned to the Source Mic while the other can be assigned to any of the available mic models
- Tube Saturation adds a model of analog tube saturation (distortion) often found in high-quality tube mic preamps
- The Output Gain fader allows you to attenuate the output signal if necessary



Tube Classic Analog Tube Modeling Plug-in

Antares Tube is an affordable multi-format plug-in designed to model a wide range of analog tube effects, based on technology from Antares' multi-award-winning Microphone Modeler plug-in. Tube is designed to be extremely easy to use and is so DSP efficient that even a modest native system will support dozens of instances. Tube provides two basic tube models. The **blue tube** adds the warmth of a classic tube preamp to vocals, acoustic guitar, horns, strings, synth pads, in fact pretty much anything. The **orange tube** provides the deep, warm distortion of an overdriven tube amplifier. It can be used in moderation to impart a subtle effect to almost any track, or cranked up on electric guitar, bass, synths, even voice for more dramatic distortion effects. Tube also includes a unique **OmniTube** function that inserts a compressor into the signal path before the tube model. This compressor is set to compress the signal and then apply sufficient makeup gain to ensure that Tube's Drive control can drive the entire signal above the clipping level. After the tube effect is applied to the entire signal, an inverse gain function restores the signal's original dynamics.

Available for RTAS (Mac OS 9 and OS X and PC), VST (Mac OS 9 and OS X and PC), DirectX and MAS (Mac OS 9 and OS X).

B&H PHOTO - VIDEO - PRO AUDIO

AUTO-TUNE

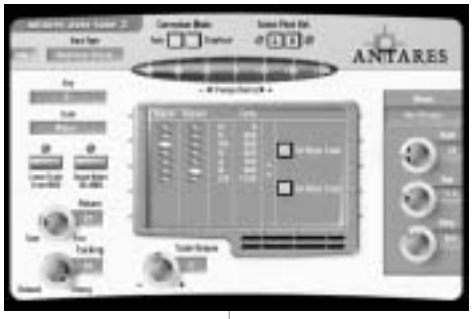
Intonation Correcting Plug-In

Auto-Tune is the multi-platform pitch detection and correction plug-in for Mac and PC considered to be the "Holy Grail of recording" by Recording magazine. Auto-Tune allows you to correct pitch and intonation problems on voice and solo instruments without distortion or artifacts. Auto-Tune is 88.2/96kHz compatible and works not only on mono sound files but also provides phase-coherent processing of stereo files. Two modes of operation include Automatic where pitch is continuously compared to a user selected scale and Graphical mode offering more precise control allows you to draw specific target pitches. Available for TDM/AudioSuite, VST, MAS, RTAS or as a stand alone application on the Mac and DirectX on the PC.

- In Automatic Mode the input pitch is continuously compared to a user-selected Scale and then corrected to the Scale note closest to the detected input pitch
- Graphical Mode displays the pitch envelope of the audio to be corrected along with a selection of graphical tools that allow precise manipulation of the performance's pitch
- An Input Type control allows you to choose an appropriate input source algorithm: Soprano Voice, Alto/Tenor Voice, Baritone/Bass Voice, Instrument, and Bass Instrument, ensuring the fastest and most accurate pitch detection and correction
- Bass Mode allows pitch detection/correction down to 25Hz to accommodate low frequency instruments such as fretless bass

Automatic Mode

- Select any key and scale - minor, major or chromatic as well as 26 historical and microtonal scales
- The Retune knob controls how rapidly (in milliseconds) the pitch correction is applied to the incoming sound
- The Tracking knob controls how much variation in the incoming waveform is allowed — useful with difficult to track sounds, such as a breathy or growling voice
- Individual notes can be removed from a scale or bypassed. Bypassed notes passed along without correction. You can apply pitch correction to a single note in a scale by bypassing all of the notes in the scale except for the problem note



- Scales can be de-tuned to any desired pitch
- The Vibrato section allows you to control the depth, rate and onset delay of a performance
- The Change meter shows how much the pitch is being changed, measured in cents (100 cents = one semitone)
- The Target Noted Via MIDI button allows Auto-Tune to track pitch in real-time via a MIDI keyboard or sequencer track
- The Learn Scale From MIDI function allows you to play a line from a MIDI keyboard or sequencer and let Auto-Tune 3 construct a custom scale containing only those notes that appear in the line

Graphical Mode

- Graphic mode begins with Tracking mode where an amplitude waveform overview and a pitch graph referenced to the scale selected in Automatic Mode are created across a timeline delegated by the length of the sound file. Once tracking is completed and Tracking mode is disengaged, an amplitude waveform and a detected pitch are displayed
- The next step is Correct Pitch mode which provides a number of graphic tools that allow you to actually draw your target pitches across the zoomable timeline. After editing, play the sound file and Auto-Tune will recognize the changes



- Horizontal line and curve tools are provided as well as cut, copy, paste and undo functions
- Includes Retune and Tracking functions identical to that in Automatic mode
- A memory (RAM) buffer allows you to allocate the necessary amount of time to process the sound file in realtime
- Auto-Curve function creates an editable curve directly over the detected pitch curve. This allows very precise pitch correction without having to draw each line and curve freehand.

ANTARES AUTO TONE

DIRECT X
for PC **199.95**

RTAS
for MAC **274.95**

VST/MAS
for MAC **259.95**

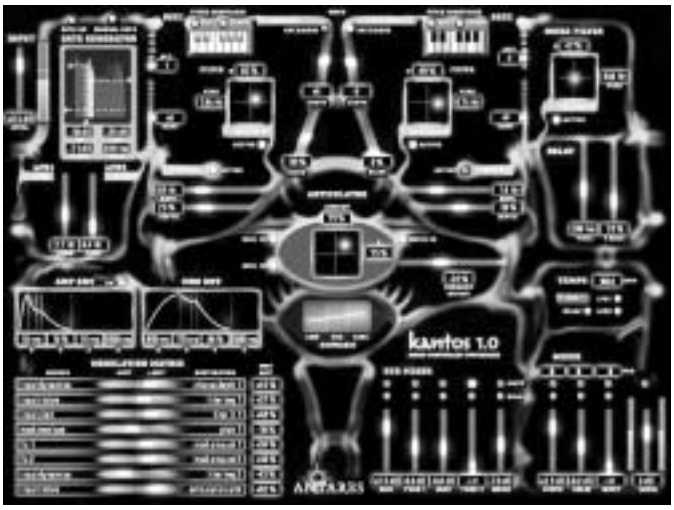
TDM/AudioSuite
for MAC **399.95**

ANTARES

KANTOS 1.0

Audio Controlled Synthesizer

From the company that revolutionized vocal intonation processing comes Kantos, a software-based synthesizer that liberates you from the tyranny of MIDI, keyboards, controllers, or anything that stands between you and the music you hear in your mind. Whether you are looking for an alternative to conventional controllers, or are looking to produce electronic music with a level of dynamic expression that wasn't possible till now, Kantos will literally change the way you make music. Designed to be patched as an insert plug-in within your digital audio software, Kantos uses the pitch, harmonic content and formant characteristics of an incoming audio signal—instead of using a MIDI keyboard to control synthesis.



FEATURES

Audio Controlled Synthesis

- The pitch, dynamics, harmonic content and formant characteristics of the incoming audio is analyzed and then extracted – live and in real time
- This information is then used to control the sound engine in unique ways never before possible with a conventional MIDI synth
- For reliable pitch detection, a pitched monophonic audio source such as the human voice or a musical instrument should be used – unison sections can also be used
- A lot of interesting results can be had with polyphonic audio signals (particularly with rhythmic input) except that you won't get predictable pitches from unpitched input.
- You are able to generate harmonies with predictable pitch control by processing or instantiating the software for each individual note in the harmony

Sound Engine

- The sound engine includes a combination of traditional synthesizer functions as well as many unique functions to for controlling the audio including – Wavetable oscillators, Pitch constraint and quantization control, Noise source, Resonant multimode filter, Timbral articulator, Envelope generators, LFOs, Modulation matrix, Gate generator, Delay line and a Mixer.

Gate Generator

- The Gate Generator gives you control over the triggering and duration of the envelopes as well as (optionally) retriggering the oscillators' wavetables
- A real time display of the input signal's amplitude helps you set the independent Gate On and Gate Off levels as well the minimum Gate Hold time
- A variable-threshold noise gate is also provided to help clean up any unwanted noise in the control audio

Wavetables Oscillators

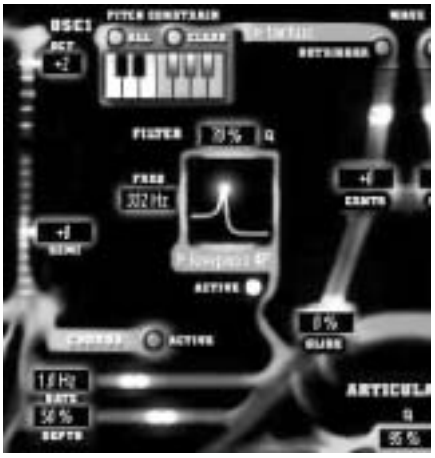
- 2 flexible wavetable oscillators are provided each with its own independent multimode resonant filter and chorus generator
- A pitch constrain keyboard, for each wavetable allows you to limit the notes that the oscillator can play
- Includes a variety of traditional - and not so traditional - wavetables (more can be downloaded). You can also create your own custom wavetables using an audio or sample editor

Delay and Tempo Control

- A delay line with variable feedback is included for generating rhythmic effects – delay time can be set in absolute time or, via the Tap Tempo function in BPM.

Articulator

- The heart of Kantos, the Articulator module, takes (with great precision) the harmonic content and formant information from the input signal, and dynamically applies it to the output of the oscillators and noise generator. In addition, you can warp and mutate the audio signal in a variety of ways:



—The X/Y-axis Amount and Q (resonance) controls defines the amount and character of the harmonic processing

—Formant Offset parameter shifts the inputs detected formants up or down in frequency

—The Emphasis section lets you tailor the sounds overall harmonic balance

KANTOS 1.0

Modulation Matrix

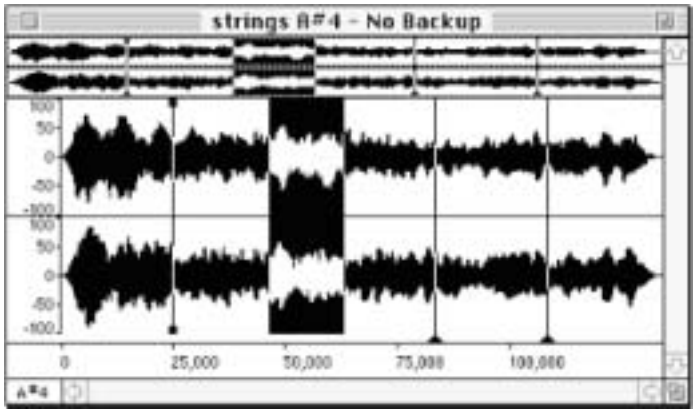


- A flexible modulation matrix provides an almost limitless variety of dynamic effects
- Any source can control multiple destinations and any destination can be controlled by multiple sources – Almost every variable parameter is available as a destination
- Two multi-waveform LFOs, with manual and Tap Tempo rate control, as well as two ADSR Envelope Generators are available as additional modulation sources.

Mixers

- A Submixer controls the internal sound sources. In addition to the wavetable oscillators and noise generator, two additional sine wave oscillators track the fundamental frequencies of their respective wavetable oscillators
- The Main Mixer offers control over the main synth output, the delay line return, as well as the original unprocessed input audio and if instantiated as a mono in/stereo out plug-in, each of the main mixer inputs also include a pan control

INFINITY *DSP Sample Looping Tools for Mac*



Imagine taking a raw sample, and then, with just a few mouse clicks, creating a loop so perfect that you can't tell when it started or where it repeats. Imagine doing that with sounds that are impossible to loop smoothly using your current sample editor, like string orchestras, choirs, fat analog synths, chorused guitars, and so on. Infinity lets you do it, quickly and easily.
Infinity is a powerful Mac-based DSP tool kit designed to be used in conjunction with your current sample editor. It uses unique DSP technologies to automate and improve the looping process, taking sample editing to a new level of quality, speed, and simplicity. It allows you to loop everything from solo woodwinds to huge synth stacks with dramatic improvements in quality and speed. This means that you don't have to put up with rhythmic pulsing or lifeless sustains any longer. Infinity preserves the "liveness" of your samples because it has a looping tool appropriate for every type of sound. Using Infinity's unique DSP technologies, beginners can get results which are superior to professionals using normal sample editors. For professionals, Infinity provides a level of quality, speed, and simplicity far beyond what's possible with conventional sample editing tools.

- Full support for 24 bit files
- Support for 88.2 kHz and 96 kHz playback via the Sound Manager
- Full 32-bit floating point audio processing
- Edit functions: Cut, Copy, Paste, Clear Unselected, Clear, Mix, Invert, Reverse, Silence, Create/Delete Loop Points, Select Loop Points, Select All, Set MIDI Note, Auto Zero, Show Clipboard
- Unlimited Undo/Redo nesting
- Special Edit functions: Real Time Loop Adjust, Variable Edit Blend Time and Taper, Switchable Window Modes, Enable/Disable Clipboard, Enable/Disable Undo
- Real Time Loop Adjust lets you move the loop points during real-time playback, so you can locate the best loop points by ear.
- Support for the AIFC file format
- Rotated Sums Looper lets you randomly layer multiple copies of the loop data to "homogenize" the sound while preserving your original loop points. Perfect for improving those problem loops on CD-ROMs and for special effects.
- Crossfade Looper with Smart Auto-Scan is an enhanced crossfade tool with an intelligent loop point scanner that lets you find the best loop point automatically.
- SPR Looper automatically creates seamless loops in chorused and ensemble sounds while preserving the exact sound color and stereo image of your sample. The resulting loop data can be automatically merged with the attack portion of your sample.
- Synthesis Looper lets you control the frequency domain sidelobe energy of the loop, and selectively smooth out lumpy sounding frequencies in chorused and ensemble sounds. The new sound is more stable and loop repetitions are less noticeable.
- Freeze Looper creates perfect loops in sounds like winds, brass, and other solo sounds that have a clear harmonic series.

ANTARES

JVP • MDT

DSP Plug-In Processors For Pro Tools TDM

The JVP (Voice Processor) and MDT (Multiband Dynamics Tool) are high resolution DSP plug-in processors designed for music production, mastering, post production and broadcast applications within the Pro Tools TDM environment. JVP provides four essential processors for achieving the highest degree of sonic detail and richness from vocals and solo instruments, sound effects, stereo mixes — you name it. MDT can be used to emulate just about any existing dynamics processor and can also be customized to bring out unique sonic characteristics from your audio. The buttons, sliders and settings menus for both plug-ins are completely automatable under Pro Tools 4.0 or later and are compatible with all Pro Tools TDM hardware running on a Power PC computer.

JVP

FOUR FUNCTION VOICE PROCESSOR

- 4 of the most useful signal processing functions in one easy to use plug-in running simultaneously on a single DSP chip — de-esser, compressor with a downward expanding gate, parametric EQ, and multi-tap delay
- Brings out detail and adds sparkle to vocals, instruments, sound effects, even entire mixes
- Processors can be used alone or together in any combination
- Simple to use single window interface with Bypass and Display toggle switches for each of the 4 processors
- High resolution In/Out level meters with 1 sample over clipping indicators
- 24-bit I/O and 56-bit internal accumulator processing resolution

De-Esser



- Gain reduction meter
- Threshold control with post-filter level meter
- Variable high pass filter
- Compressor with variable ratio, attack and release

Compressor/Expander/Gate



- Gain Reduction meter
- Expander/Gate with variable threshold and ratio
- Compressor with variable threshold, ratio, attack release and hard to soft knee settings

Parametric EQ



- 3 fully parametric EQ points with selectable HP, BP, LP, shelving filters
- 10Hz to 20kHz frequency, ±24 dB gain and .1 to 4.0 oct bandwidth controls
- 140 dB dynamic range

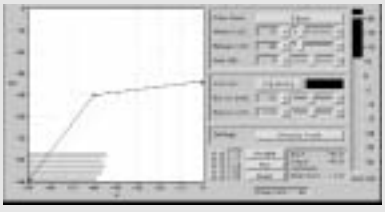
Multi-Tap Delay



- Up to 6 delay taps with L-R pan control for each tap
- Dedicated delay and feedback gain control
- Variable low pass filter and FX/Dry mix
- DSP card dependent delay time and feedback

MDT MULTIBAND DYNAMICS TOOLS

- Provides one, three or five individually adjustable spectral bands of dynamics
- Easy to use graphic interface
- Not only emulates a wide variety of existing compressors, limiters, expanders and gates including dynamic EQs, de-essers and spectral enhancers but also allows you to create your own unique single and multiband processors
- Clip Sentry is a high-speed digital limiter designed to reduce gain only when clipping is about to occur
- "Look ahead" peak averaging scans up to 1024 samples of audio data before it is processed, compares it to currently processed the data and then, uses both numbers to derive the final output



- In Multiband mode, Continuously Variable Offset updates the input levels for each spectral band continuously, ensuring that tonal balance will be achieved and that the same amount of gain reduction will be applied to each input
- 24-bit I/O and 56-bit internal processing resolution
- Greater than 144 dB dynamic range with imperceptible passband coloration

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ONE HOUR FREE PARKING
AT 349 W. 34th STREET (with purchase of $100 or more)

VJ 2.2



The Visual Synthesize for Musicians

If you're familiar with traditional multimedia think of ArKaos VJ as the first real-time multimedia program offering an innovative way to edit and manipulate a visual performance as it's running. If you're musically oriented think of ArKaos VJ as a 'synthesizer for pictures' – a program that brings to pictures and movies the same fresh attitude that samplers brought to audio.

Why just play music when you can immerse your audience in complete audio-visual experiences? ArKaos VJ does for images and video what samplers and synths do for music—and it does it in real time. That means that DJs, VJs, performing musicians and multimedia producers now have the ability to easily create and project dazzling interactive imagery to take live performances, dance events, videos and other media productions to new levels of creative expression.

Animate Your Presentations:

Displaying, projecting and recording dazzling visuals to accompany your music is just the beginning. With ArKaos VJ you can play live video sessions, record your own visual performance files and synchronize digital imaging automatically. And VJ 2.2 takes the concept even farther enabling you to create multimedia presentations using slide-show images and QuickTime movies, triggered by notes that can play music or audio samples like voice-overs. Animation can be synced automatically to MIDI Time Clock or to audio via advanced beat-matching technology, or you simply press the appropriate keys on the keyboard and you can play any part of your creation at any time and in any order. Program your show in advance, run live or just let ArKaos VJ generate the whole visual show.

FEATURES

- Adds dimension to concerts, dance clubs, raves, music videos and more
- JPEG, BMP, PICT still image support
- QuickTimeTM, AVI and Flash SWF video and animation support
- More than 60 customizable real-time visual effects
- Visuals sync to audio via powerful automatic beat-matching technology
- Allows any MIDI keyboard or device to control imagery and effects
- Integrates with any MIDI sequencer
- Supports multi-layering and chaining of effects
- Built-in visual recorder to create videos and visual remixes
- Multiple keyboard mapping to trigger images and effects
- Enables full-screen and multiple screen usage for performance events
- Import QuickTime movies, flash animations and still images, or grab live video
- Drag and drop images, clips onto keys of the virtual computer and/or MIDI keyboard
- Free ArKaos Visualizer software brings visuals to MP3 players

Here's how it works...

1. Import images, QuickTime movies or grab live signal.

2. Drag and drop the item onto the virtual computer or MIDI keyboard.

3. Select the required visual effects from the window.

4. Drag and drop the effect onto the keyboard

5. Record live performances, or let the beat matching artificial intelligence take care of it for you



BIAS

PEAK 4



Audio Editing/Processing/Mastering Software for Mac OS X

Peak 4 is an award-winning digital audio editing application that is fully "carbonized" (OS X native). Sporting a streamlined "Aquafied" user interface, and many other enhancements Peak 4 also runs just fine under Mac OS 8.6 through 9.2.

It takes full advantage of OS X's advanced capabilities, including protected memory space for enhanced system stability; the ultra-streamlined, elegant "Aqua" user interface; and CoreAudio, the audio I/O component of OS X that supports multichannel, multi-client hardware, and beyond 24-bit/96kHz resolution.

Peak 4 supports your Mac's built-in audio as well as a broad range of third-party hardware via Sound Manager or the powerful CoreAudio standard. (Translation: it works perfectly with all of the best-sounding audio hardware.)

Version 4.0 features include VXT plug-in management via a fully integrated version of BIAS vbox—the most advanced plug-in matrix control environment available. The Cursor Palette with an integrated tempo calculator, plus the ability to sort markers by time or name in the expandable contents window. Also includes dozens of free plug-ins, including BIAS Freq—a mastering-quality 4-band paragraphic EQ.

FEATURES

- Fastest and most intuitive waveform editing available on any platform
- Displays multiple document windows simultaneously
- Unlimited Undo/Redo with graphic history
- User definable display size and color
- Record audio directly or import all QuickTime compatible audio formats
- Supports built-in Mac audio I/O, plus most ASIO-, Sound Manager- and CoreAudio-compatible audio hardware
- Full DSP toolset, including Fade In/Out, Normalize, Gain Envelope, Reverse, Invert, Rappify, ultra-accurate Pow-r dithering, Convolve, Reverse Boomerang, Change Sample Rate, Mix, Change Pitch, Panner, Amplitude Fit, Find Peak, & more
- Playlist with automated Redbook (Disk-at-once/DAO) CD burning
- Easily create, edit, read & write region, loop & reference markers
- Sophisticated playlist featuring advanced crossfades & DSP
- Advanced looping tools — including the highly acclaimed Loop Surfer, LoopTuner, Crossfade Loop, & Guess Tempo
- 32-bit; sample rates up to 10Mhz
- Realtime VST effects support
- Advanced VST effects routing matrix via integrated Vbox support (includes Vbox SE)
- Automated Repair Clicks & Remove DC Offset
- Convolve function - instantly apply recorded room ambiance to any audio document
- Read/writes multiple file formats — even MP3 (has exceptionally fast, high-quality, on-board MP3 encoding)
- Full featured batch processing options — including Export Regions & much more
- Change Duration DSP function allows high quality time compression/expansion without affecting pitch
- Threshold function automatically divides files into discrete regions based on amplitude
- Fully customizable toolbar and keyboard shortcuts
- Trigger playback and cueing of up to 10 audio documents using Mac keyboard
- Recording notepad and timer
- Advanced QuickTime movie and DV clip support features better-than-frame accurate synchronization of audio and movie playback, even during selection and scrubbing
- Supports all SMDI hardware samplers
- Supports DV standard (16-bit/48kHz) without additional hardware
- Includes BIAS Freq, a mastering-quality 4-band paragraphic EQ that's fully native for OS X (also OS 8.6 and up). BIAS Freq brings high-end audio equalization to Peak, with -18dB to +18dB gain values, 0.1 to 10 Q values, sweepable 20Hz to 20kHz frequency values, high- and low-cut filters, individual band bypass buttons, 24dB stereo input/output meters, and a super-fast Carbon Event driven interface. (For more demanding applications, plus more extensive filtering and bandwidth control options, the 10-band BIAS SuperFreq is an advanced VST plug-in, available as a download-only purchase from the BIAS Web site.
- Bundled with more than 25 VST plug-ins including Reverb, Compressor, Limiter, De-Esser, Dynamics, Distortion, Stereo Imager, Degrader, Overdrive, Resonant Filter, Stereo Simulator, Sub Bass Synthesizer, Delay, Leslie Simulator, Pitch Tracker and Vocoder
- Also bundled with Roxio Toast Lite—the industry-standard CD burning software, and Waves AudioTrack, a multi-processor plug-in with 4-band fully paragraphic EQ, compressor/expander and gate.

PEAK 4

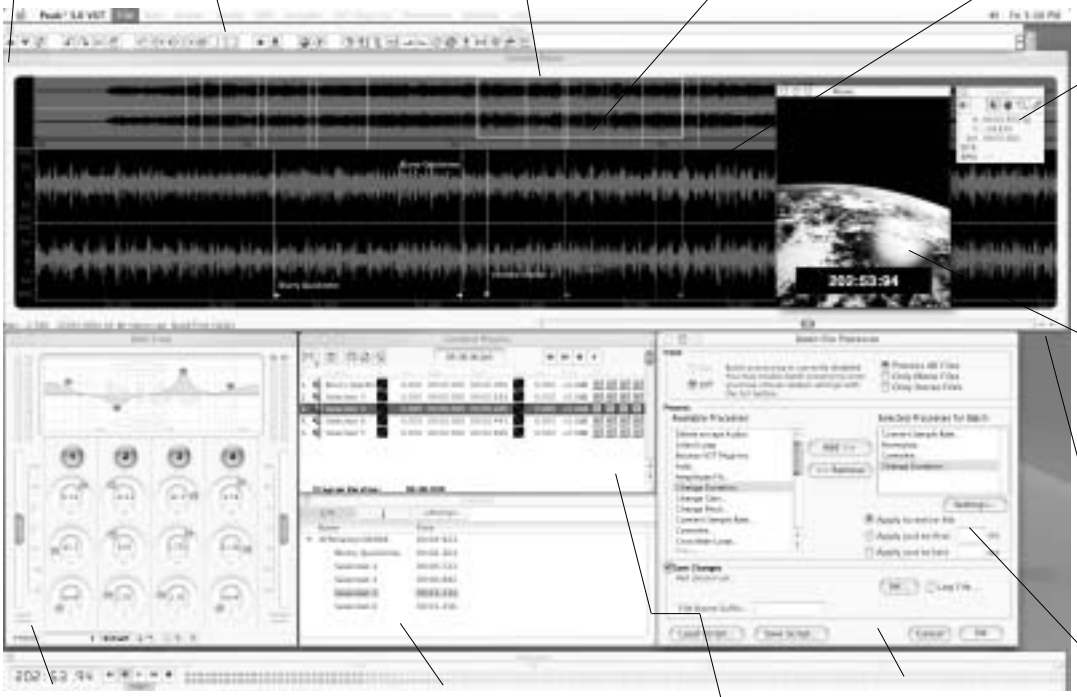
Customizable toolbar and user-definable key commands let you personalize Peak to work the way you want

Open an unlimited number of audio documents—with full cut and paste between documents, plus independent edit histories and waveform display options.

Reference markers let you locate pops, clicks, or other points simply by using the Tab key, or typing the first few letters of a marker's name. Region markers can designate areas, such as a discrete CD tracks or portions to export to a new document. Loop markers set loop begin/end points, and can be used with Peak's Guess Tempo & Loop Surfer to extract perfect loops instantly, with ease and precision.

Peak's DSP menu contains great features such as change pitch, change duration, convolve, repair clicks, change sample rate, fade-in/out, crossfade lop, normalize, mix, dither, reverse and much more.

Overview window displays the entire audio document as well as the detail area you're working in. Or if you wish, disable it for maximum waveform views.



The cursor palette includes a selection cursor, pencil tool, magnifying glass, blend on/off. It also displays selection duration in seconds, SMPTE/EBU frames, bars/beats, etc. Shows X/Y coordinates, BPM and DTR (distance to nearest reference marker).

Import QuickTime movies and digital video (DV) clips for audio sweetening: Movies appear in a sizeable window, and if you wish, the movie's audio can open a standard Peak audio document. Peak maintains better-than-frame-accurate sync, letting you play the movie while making audio selections or while scrubbing the audio file.

Edit window can be sized so your monitor can accommodate many files at once—or zoomed to show enormous waveforms. Edit windows can be stacked, tiled, and custom-colorized.

Save hours using Peak's batch processor to perform repetitive tasks in the background, while you do something else. Virtually any action, including plug-in signal processing, can be applied to dozens, hundreds, or even thousands of files with only a few mouse clicks.

Includes BIAS Freq—a mastering-quality 4-band paragraphic. Since Peak supports real-time VST plug-ins (like Freq), you can tweak and audition audio effects "live" without having to process files first.

Contents window is a quick summary of all regions, loops, and markers in all open audio documents. Use it to quickly access desired playlist elements.

Transport window with playback controls, as well as a location counter and super-responsive, high-precision VU meters.

Peak's playlist allows you to pull together regions from multiple audio documents — and then apply crossfades, plug-ins, gain changes, set gap times, apply DSP effects, and burn directly to CD, or bounce as a new audio document in a variety of formats (including a Roxio Jam Image).

Also Available. . .

Peak LE:
Provides pro-caliber editing, processing and power at a price anyone can afford. Whether burning audio directly to CD-R, or encoding mp3 audio files for the Internet, Peak LE is a great way to discover Peak power basics. Has many of Peak's essential features including playlist direct CD burning, and real-time VST plug-in support. And when you're ready, you can upgrade to the full version of Peak.

Peak TDM:
Offering a powerful mastering environment for your TDM plug-ins, Peak TDM takes full advantage of the processing power of the Pro Tools system, lets you chain multiple TDM plug-ins together, adding countless effects in real-time to your stereo mix.

Peak DV:
This special edition of BIAS Peak is optimized for users of Final Cut Pro, iMovie, iDVD, Premiere, DVD Studio Pro, and other video or DVD authoring applications. Peak DV has everything you need to edit audio for your video projects, including lightning-fast audio editing and processing, support for 16-bit/48kHz files, VST plug-in support, QuickTime movie support with full chase lock - even during audio selection, advanced marker and region implementation, batch file processing, unlimited undo/redo with edit histories, built-in DSP processing including a high-quality time compress/expand algorithm, gain envelope, multiple simultaneous audio document display, and more! Peak DV also supports Core Audio, Sound Manger and ASIO.

BIAS

DECK 3.5



Advanced Digital Audio Workstation

Looking for a high-performance multitrack digital audio workstation? Nothing else touches Deck 3.5's combination of features, speed, compatibility, and value—it even supports Mac OS X and OMF file importing—making it more compelling than ever. Deck 3.5 has a huge feature set, with virtually everything you'd expect in a professional DAW, at a fraction of the cost. For less than you might spend on one competing system, you could equip an entire team with Deck 3.5 workstations — including hardware and computers — without sacrificing quality or capabilities.

Perfect for video editing, audio post-production, and DVD authoring studios, as well as radio stations, music studios, and sound designers. This makes it a great complement to Final Cut Pro, DVD Studio Pro, iDVD, Avid Express, Premiere, and other video editing or DVD authoring systems.

Deck 3.5 imports, exports, and accurately syncs to QuickTime movies. Scrubbing and spotting audio events is astoundingly easy and fast. Unlike budget versions of other audio tools, you get true timecode synchronization plus a SMPTE/EBU timescale display. Deck 3.5 is also fully surround-sound capable, with built-in 5.1 surround panning.

FEATURES

- Record, mix, play back, & process up to 64 simultaneous tracks
- Up to 999 virtual tracks
- Realtime mixing console, with dedicated faders, pans, & other controls
- Automated level mixing with high-resolution moving faders
- "Rubberband-style" breakpoint automation
- Mixer states "snapshot-style" scene automation
- Clip-based automation: Copy, Paste, Drag & Drop automation envelopes follow audio regions as they are moved
- Advanced realtime signal processing with VST plug-in support
- Built-in realtime DSP effects including Chorus, EQ, Delay, & Multitap
- Super-fast spotting of audio clips to picture
- Supports up to 16-bit/48kHz audio (with compatible 48kHz hardware).
- SMPTE/EBU timecode-fluent for editing, spotting, external chase-lock synchronization
- Slaves with subframe accuracy to external video tape recorders & other transports — ideal for layback & other assembly
- Import QT movies & synchronize audio-to-picture with better-than-frame accuracy
- Scrub QT movie with full audio chase
- ADR (automated dialog replacement) capable
- Monitor previously recorded tracks while recording new tracks (optionally in sync to a QuickTime movie)
- Discontiguous clip selection
- Clip nudging & trimming — with better-than-subframe accuracy
- Master fader with stereo effects bus
- Sync to external MIDI sequencers (via third-party CoreMIDI drivers)
- Non-destructive punch-in & punch-out
- Supports most ASIO (OS 9), Sound Manager, and CoreAudio (OS X) compatible audio hardware
- Supports multiple file formats including AIFF, SDII, .WAV, .SND, & QuickTime
- Export sessions as QuickTime movie with multiple format options
- Allows external waveform editor such as Peak to be launched from session
- Library window keeps track of all the audio documents used in each session. Clean sessions of unused files, locate files on your hard drive, or view where files occur in session. Quickly and easily drag and drop files from the library into desired track/time in the tracks window.
- Master output offers a master stereo fader and precision metering. It also features eight mono (four left and four right) and four stereo effects inserts for applying mastering effects and other processors to an entire mix.
- Includes over 25 free VST audio effect plug-ins (same ones bundled with Peak), including Freq 4-band paragraphic EQ, Vbox SE , Bias Peak LE and Roxio Toast Lite CD burning software.

Deck LE

Very powerful with all of Deck's essential tools, Deck LE gives you everything you need for quick, easy, pro quality multitrack recording and editing, all at an incredibly affordable price. Record, edit, and play back up to 8 tracks of audio.

Deck LE supports ASIO-compatible audio cards, offering a greater choice of hardware options than any other competing product. Deck still offers direct support for the Korg 1212 I/O card, as well as any Sound Manager-compatible card. You can use Deck LE with your Mac's built-in audio connections, with no extra hardware needed. Deck LE also integrates with the Tascam US-428 DAW Controller, with full MIDI Map support.

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DECK 3.5

Control allow you to zoom in or out, and adjust track height (with zoom memories). Location displays show current location, plus start, end, and duration of selections.

The tracks window is used for arranging and editing audio clips and crossfades, using up to 64 simultaneous tracks for recording and playback (or up to 999 virtual tracks). You can also individually mute, solo, name, and arm-for-record each track within this window, without having to use the mixer.

Import QuickTime movies and DV clips, then sync them to your session with better-than-frame accuracy. You can scrub frame-by-frame through a movie with full audio chase. Alternately, once you select and audio region, you can easily spot it to a picture by scrubbing the movie, releasing the mouse button, and you're done.

In addition to moving fader and mixer states automation, Deck has "rubberband style" breakpoint automation. Even better, it's clip based—so your automation levels follow any clips that you might choose to move.



Deck 3.5 supports discontiguous clip selection – a real time-saving feature that lets you select individual regions in different tracks as a group.

Need to repeatedly paste a drum loop, room tone, or some other portion of audio? Don't bother pasting over and over—with Deck 3.5 you simply tell define how many times you'd like to drop the loop into a track and Deck takes care of the rest, automatically.

Realtime mixer offers a fast and intuitive control surface, with individual channel faders, pans, mute, solo and record controls – plus user-definable input/output assignments and renamable track labels. You can expand or contract the mixer window to view as few or as many tracks as you like.

Each channel supports up to four VST effects inserts—or a virtually unlimited number of effects using Vbox SE (included).

Transport window features location times plus tape deck-style button (Play, REC, FF, REW). Pitch control and scrub control sliders—as well as automated punch-in/punch-out, and looping controls—further your playback and recording options.

In addition to moving fader and breakpoint automation, Deck 3.5 offers "snapshot" Mixer State automation—for instant scene recall.

CPU meter keeps track of all system resources, including CPU, RAM and hard disk usage and availability.

Deck 3.5 gives you advanced moving fader automation—to store and recall all your moves with smooth-sounding precision. You can also update or overwrite automation on the fly, just by grabbing a fader (using your mouse or external control surface).

Advanced Audio-for-Picture Tool

Deck 3.5 is a must-have for film, video, video game, Web, and other media applications. In addition to import/export QuickTime movies, with better-than- frame-accurate synchronization, Deck 3.5 supports importing OMF sessions from video editing and authoring applications such as Final Cut Pro, Avid, and others. Sessions from OMF-capable DAWs–including Pro Tools, Digital Performer and Nuendo can also be imported. Deck is also ideal for spotting effects, editing dialogue, processing soundtracks, and recording ADR voiceovers.

The Most Affordable Solution for 5.1 Surround Mixing

Deck 3.5 now offers built-in 5.1-channel surround mixing, making it the most affordable high-performance, surround-capable DAW on the market. While this will have most obvious appeal to audio-for-picture professionals, Deck 3.5 is also perfect for music producers - whether they're repurposing stereo media for surround mixes, or crafting multichannel audio mixes from the ground up.

The surround interface includes a 360-degree surround mixer, a "virtual joystick" with a choice of three optional panning algorithms: Radius, Variable-Width, and Room Ambiance. Each channel in the mixer displays its own 360-degree panning window - and just like Deck's conventional stereo channels, each also supports up to four simultaneous VST signal-processing plug-ins. The master output window features a smooth fader for all of the six surround outputs (including the subwoofer/LFE channel), plus individual meters. Additionally, the interface offers complete control over output assignments for each of the six channels (requires ASIO or CoreAudio audio hardware with at least 6 output channels, and drivers).

BIAS

VBOX



Multi-effects Control Environment for VST Plug-ins (Mac OS 9/OS X or Windows

Plug-ins changed signal processing forever, bringing advanced effects to the desktop. Now, Vbox is changing plug-ins as you know them, bringing unprecedented creative command over all your signal processing options — and unleashing a universe of yet-to-be discovered sounds. Vbox is the ultimate multi-effects control environment for VST plug-ins, operating under Mac OS 9/X or Windows. It works either free-standing or integrated into any VST host program — including BIAS Peak and Deck, Steinberg Cubase, Emagic Logic, and others — where you can process and save any audio file or even record, process, and save a live feed. Unique effects matrix lets you combine, route, and mix up to hundreds of plug-ins per channel. The result is literally millions of sonic combinations — and unrivaled creative potential.

FEATURES

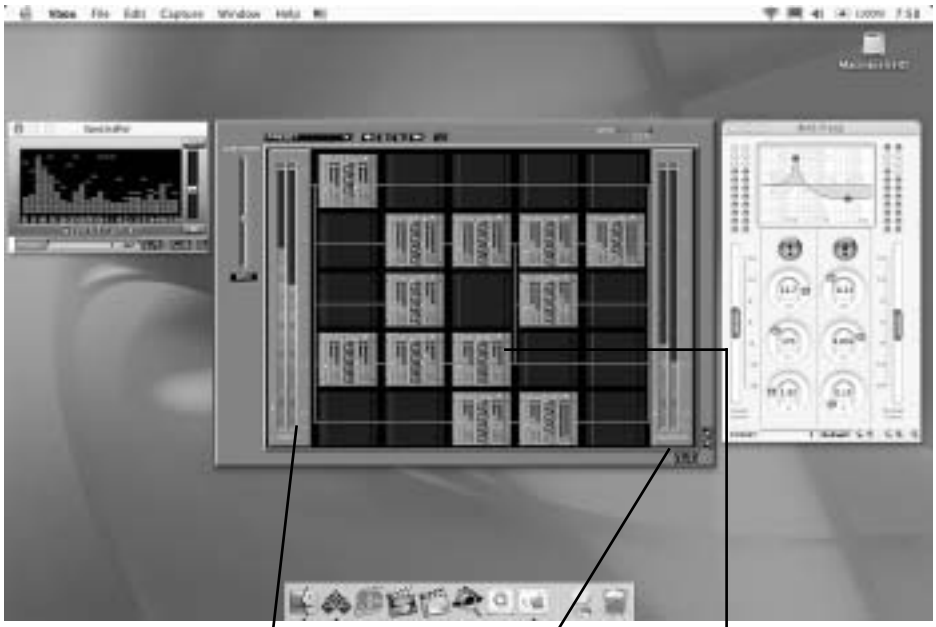
- Create, manage, store & recall a virtually unlimited number of VST plug-ins — all in a fast, intuitive, fun workspace
- Unique effects matrix lets you combine, route & mix multiple individual effects, in parallel, series, & series-parallel combinations — for millions of sonic permutations
- Hot-swap plug-ins instantly
- Comes with dozens of free VST plug-ins and works with the hundreds of VST 1.x compatible plug-ins already on the market
- Integrates seamlessly into VST host apps
- Use it as a stand-alone signal processor for AIFF, SDII, & .WAV audio files
- Process & record live feeds or CD playback — perfect for DJ's & live performance
- Easy muting, soloing, & bypassing of any individual plug-in — plus full control of input & output gain of each plug-in — directly in the vbox environment
- Accurate stereo metering at each stage
- Works with Sound Manager, ASIO (OS 9), & CoreAudio (OS X), compatible hardware
- Plug-in parameters for any plug-in in the Vbox matrix can be accessed by clicking the edit button on each active plug-in module—instantly revealing all the standard parameter controls and user interface of each individual plug-in.
- Parameters for multiple plug-ins can be opened simultaneously (only one plug-in at a time with Vbox SE.) All parameters can be adjusted and auditioned in real-time without stopping audio playback.

Master wet/dry mix control adjusts the relative balance between processed and unprocessed signals.

Toggle back and forth between settings with the A/B compare switch—a great decision-making tool. You can also store and recall a virtually unlimited number of effects combinations or individual effects settings.

Each plug-in assigned in the effects matrix has its own independent input/output faders, precision meters, Solo/Mute/Bypass/Edit buttons, and a signal flow indicator. You have unparalleled control over you sound.

Keep an eye on your computer's CPU usage with the CPU meter—so you'll always know how many plug-ins may be assigned.



Convenient master input and output faders - plus high-precision meters at both input and output stages - gives you complete control over master levels.

Vbox's unique effects matrix lets you assign and arrange a virtually unlimited number of VST plug-ins. The matrix default size is 5˝x 5˝, but you can custom-size it to suit your liking.

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ONE HOUR FREE PARKING
AT 349 W. 34th STREET (with purchase of $100 or more)

BIAS Studio Software Bundles

BIAS Studio takes your project from concept to completion with recording, mixing, editing, and processing in one awesome bundle. The ultimate desktop solution for your studio, BIAS Studio includes Peak, Deck, and Vbox for a fraction of the price that they cost separately. Also available in an LE version.

For those who want studio power together with laptop flexibility for field and live performance recordings, the Powerbook editions of BIAS Studio include the Digigram VXpocket PCMCIA card—the most portable, professional laptop audio I/O card available— to provide completely mobile audio workstations. The VXpocket gives you two 24-bit analog converters, S/PDIF I/O SMPTE/EBU timecode input, plus I/O breakout cable and supports sample rates up to 48kHz.

BIAS Studio LE: Includes Peak LE , Deck LE, Roxio Toast Lite and Waves Audio Track

BIAS Studio: Includes Peak 4, Deck 3.5, Vbox, and Roxio Toast Lite and Waves Audio Track

BIAS Studio LE Powerbook Edition: Same as above plus Digigram VXpocket PCMCIA audio I/O card

BIAS Studio Powerbook Edition: Same as above plus Digigram VXpocket PCMCIA audio I/O card

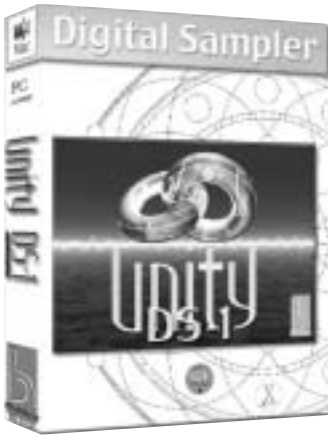
SOUND SOAP *Audio Cleaning Software*



Sound Soap is an easy-to-use, professional-quality noise reduction software designed for videographers, multimedia & web developers, music & audio enthusiasts & others. By simply adjusting just two knobs, anyone can remove unwanted hiss, room noise, rumble, electrical hum, and other background noise from almost any digital media file . These include digital video (DV), PowerPoint, Flash and digital audio workstation tracks, as well as cassette or other analog tape recordings that have been transferred to a computer.
Sound Soap works by removing "broadband" noise (such as room noise, tape hiss, road noise from a moving vehicle, or noise from air conditioners). It also removes low-frequency rumble, as well as hum - the 50 and 60Hz electrical "buzzing" sound common in many audio recordings. It can even learn the difference between the noise and the desired audio - achieving amazingly effective results. With moderate or subtle noise problems, Sound Soap can make the difference between a project that sounds amateurish and one that sounds professional. Sound Soap can even salvage otherwise unusable audio files.

- Use as a standalone application—works with any QuickTime compatible audio or video file (AVI, DV, .WAV, SDII, AIFF, MP3, MP4 & many more) or as a fully integrated plug-in with any compatible DirectX or VST host program — including BIAS Peak, Vegas Video, Cubase & others.
- Remove hiss, hum & buzzing, rumble & most other noise — without harming the audio you want to preserve. Hum removal is optimized for 50 or 60Hz
- One-step Learn Noise button automatically reduces noise (in under three seconds, SoundSoap will profile any noise in your signal and adjust the Noise Tuner and Noise Reduction knobs to their ideal settings), or fine tune the intuitive controls for even more professional results — in just seconds.
- Wash Window lets you listen and watch. The left side of the display shows the unprocessed signal, while the right side shows it after it's been cleaned.
- Preserve voice button helps remove noise (and other audio) outside of the range of the human voice.
- Noise Only mode lets you listen only to the noise being removed — and is ideal for fine-tuning SoundSoap's controls.
- Noise Tuner control helps you "tune in" the hiss and other background noise you want to remove — rather like tuning in a radio station (but one you don't want to hear).

UNITY DS-1



Digital Sampler

With the Unity DS-1 you can turn your computer into a full featured digital sampler, allowing you to recreate the sounds of acoustic instruments or any other audio source. All of the features of dedicated hardware samplers are here— 16 part multi-timbrality, 64 voice polyphony, a complete synth engine and the ability to create layers and splits—all with full MIDI control. Compatible with Akai, SampleCell, .WAV and .AIFF and CD audio files that can all be edited using the integrated audio editor.

Import more of your sample libraries and load them faster than anything else. Unity DS-1 was the first on the scene with the G4 and Dual Processor support for an unequaled 256 possible stereo voices. No other sampler even comes close. An on-board digital audio editor, two stereo effects processors, and the ability to play huge samples from RAM or stream from disk within Performer, ProTools, Logic, or Cubase give you flexibility for all your gigs.

FEATURES

General

- Sampling synthesizer in software, no additional hardware required
- Outstanding 32-bit/96kHz sound quality
- Up to 256 note polyphony
- Stereo signal path throughout
- FreeMIDI, PC300, OMS, & CoreMIDI compatible
- MAS, VST, RTAS, DirectConnect, & ReWire support, ProTools HD compatible
- Over 1GB of content included

User Interface

- 100 percent programmable
- Drag and drop sample importing, exporting & keymap placement
- Selectable graphic or slider views for envelope modulators
- Parameter values displayed in hertz, db, or milliseconds where appropriate

Sampling

- Imports 24-bit Sound Designer I & II, AIFF, CD-Audio, and WAVE files
- Imports and maps SampleCell II instruments, DLS, and SoundFont 2.0 automatically
- Option to automatically split samples
- Selectable interpolation: Linear, Quadratic, Lagrange or none
- Built-in digital audio editor includes stereo record function with level controls
- DSP functions include Normalize, Gain, Fade, Reverse, Crossfade Loop, EQ & more

Multisamples

- Up to 128 samples per MIDI note selectable via velocity or controller cross-switching
- Per-sample volume, tune, pan, sample start, FX send, envelope and mute grouping

Effects

- Two stereo insert effects processors per program
- Two additional stereo global (parallel) effects processors
- Parametric and shelf equalizers
- Multiple stereo reverb and reflection algorithms
- Stereo delays, chorus, phaser, and flange
- Overdrive, distortion, and degrade effects
- Delay times can be synced to MIDI clock

MIDI

- Control of layers, splits, and arpeggiator
- 16 simultaneous MIDI channel multitimbral
- Real-time control of all parameters simultaneously via NRPN's
- On-screen keyboard, no MIDI controller required
- Full simultaneous MIDI application compatibility
- Mac version includes QuickTime, OMS, FreeMIDI drivers, & native Logic support

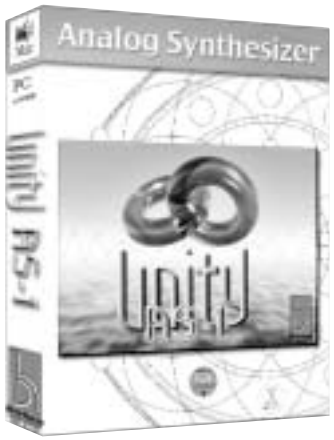
Output

- Supports output directly to digital audio file on disk in 8, 16, or 24-bit format
- Audio output: SoundManager, Direct Sound, MAS 2.0, ASIO, & Digidesign DirectConnect & Direct I/O (Mac only)

Unity DS-1 was designed by musicians, for musicians. That's why it gives you layers, splits, and full MIDI continuous control. It also gives you speed, with lightning fast note - on response time and the ability to load huge samples in seconds instead of minutes. So whether you use Unity DS-1 as a multitimbral sound module with your favorite MIDI application or a live performance synthesizer, it has power to spare. Because Unity DS-1 has the heart of a full blown analog synthesizer, your sampled instruments can be performed with a high degree of expression and/or manipulated beyond recognition to create new sounds of your own. Use the OscillatorStretch to adjust your loops to the perfect tempo. Then you can even write the sampler's complete multitimbral stereo output to disk as an audio file for use in your favorite digital audio applications.

UNITY AS-1

Digital Software Synthesizer



The moment you hear the warm, fat analog sounds of Unity AS-1 you'll know that this is no ordinary analog synthesizer. With over 1,500 different sounds that range from deep bass to deep space, Unity AS-1 blends the very best features of vintage analog analog synthesis and the latest computer DSP algorithms. Practically unlimited modulation routings and LFO's make it easy to bring vintage patches back to life. It's easier than ever to create amazing new soundscapes of your own with stunning 32-bit, 96kHz sound quality.

Play the Unity AS-1 "live", just like any other synthesizer with layers, splits and full MIDI continuous control, or use it as a multitimbral synth running alongside your favorite MIDI application. Extensive MIDI implementation allows real-time control of almost every parameter and easy integration into existing MIDI setups. The Unity AS-1 comes with over 100 editable parameters, 3 oscillators, 2 filters, 2 insert and 2 global effects, ring modulator, plus dozens of LFO's and envelopes. You can even record your performances as audio files for use in your favorite digital audio applications. Play AS-1 with just a laptop, or plug it into Performer, ProTools, Logic and Cubase.

FEATURES

General Features

- Compatible with Mac OS 8.6 through 9.2 and 10.1 or higher
- No additional hardware required
- Outstanding 32-bit/96kHz sound quality
- Up to 128 note polyphony (CPU dependent)
- True phased locked stereo signal path throughout
- Realtime MIDI input/performance
- Includes over 1500 classic analog sounds
- Record and save performances as AIFF or WAV
- FreeMIDI, PC300, OMS, & CoreMIDI compatible
- MAS, VST, RTAS, DirectConnect, & ReWire support, ProTools HD compatible
- Graphical editing of configuration

MIDI Plug-Ins

- Arpeggiator Chord, Arpeggiator Played, Channelize, Chord, Key & Transpose
- Full control of layers, splits and arpeggiator
- 128 simultaneous MIDI channel multitimbral
- Tempo control and note division
- Programmable note input & chords listing

Effects Plug-Ins

- Two insert (serial) effect processors per voice
- Two additional stereo global (parallel) effects processors
- Delay times can be synchronized to MIDI clock
- Full parametric and shelf EQs
- Effect algorithms include: Chorus, Compressor, Degrade,Dual Chorus, Dual Delay, Dynamic Filter, Flange, Overdrive, Parametric EQ, Phaser, Pitch Bend, Pitch Shift, Quad Delay, Reflection, Reverb, Shelf EQ, Single Delay, Stereo Chorus, Stereo Compressor, Stereo Delay, Stereo Flange, Stereo Phaser & Stereo Pitch Shift

Synthesis

- 3 stereo oscillators per voice
- 2 stereo filters per voice
- 8-octave range per oscillator
- Random pitch per oscillator
- Frequency modulation from any oscillator or filter
- Ring modulator (input from oscillators, external audio, filters input and output; output to filters)

AS-1 & DS-1 BOTH FEATURE

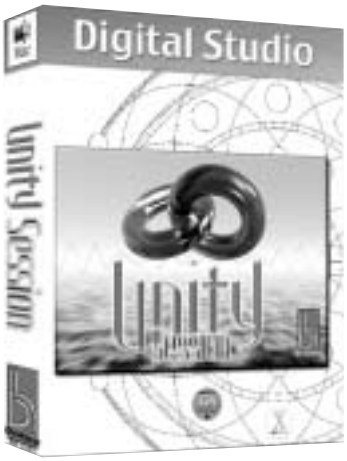
Modulation

- Modulators and routings limited only by CPU power
- Most parameters available as a modulation source and/or destination
- 6-stage envelope modulators (delay, attack, decay, sustain, sustain decay, release)
- Selectable linear or exponential curves for envelopes
- Six low frequency oscillator ("LFO") waveform shapes
- LFO's can be synchronized to MIDI clock
- Four continuous MIDI controller modulation sources are available simultaneously

Filters

- Two assignable stereo filters per voice
- 13 filter types, including 4-pole resonant lowpass, highpass, bandpass, allpass, notch & state-variable
- Multiple inputs allow parallel and/or serial filtering
- Per-filter cutoff, resonance, cutoff modulation source/amount and overdrive controls

UNITY SESSION



Sampler, Synthesizer and Physical Modeling with Integrated Mixer

Unity Session combines the functionality of hardware samplers, analog synthesizers, and physically modeled instruments into one fully integrated environment. The program brings together Unity DS-1 and Unity AS-1 by allowing both programs to be supported within the same mixer and editor applications.

All of the libraries, banks and programs are now selected and assigned within the same mixer application. The Unity Editor will allow the user to create and/or edit DS-1 and AS-1 banks, as well as opening and converting GigaSampler, Unity 2, Retro 2, Sound Font 2.0, Sample Cell II and DLS files. In addition to the sample formats available for editing through the mixer, Akai S-1000/S-3000 and Roland S-760/S-770 formatted samples can easily be converted and brought into Unity Session.

Also included in Unity Session, is a module for physical modeling. There are four physical models including Clarinet, Flute, Hammered String and Bowed String. The user can go in and physically modify the characteristics of the instrument. Finally, all of the components of the program can be combined to create an instrument. The newly created instruments are then saved in a standard bank and can be selected within an application just like any program.

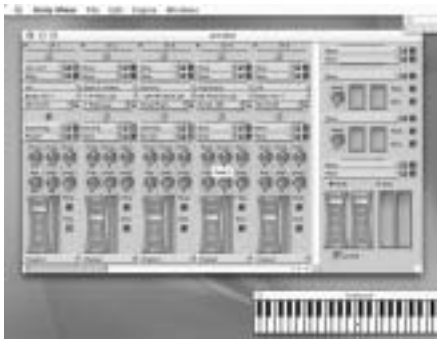
FEATURES

General Features

- Compatible with Mac OS 8.6 through 9.2 and 10.1 or higher
- Outstanding 32-bit, 96 Khz sound quality
- Up to 256 note polyphony (CPU dependent)
- FreeMIDI, PC300, OMS, & OSX CoreMIDI compatible
- MAS, VST, RTAS, DirectConnect, & ReWire support, Pro Tools HD compatible

Unity Mixer

- Assign 2 MIDI and 2 Audio FX per channel
- Assign 2 Send FX and 2 Global FX on Master outs



Synthesis

- 3 stereo oscillators per voice
- 2 stereo filters per voice
- 8-octave range per oscillator
- Frequency modulation from any oscillator to filter

Physical Models

- Clarinet, Flute, Hammered String and Bowed String

Sampling

- True stereo signal path
- Selectable interpolation: Linear, Quadratic and Lagrange
- Option to automatically split samples (for Audio CD's)
- Built-in sample editor which includes stereo record function with level controls
- DSP functions include Normalize, Gain, EQ and more

Multisamples

- Up to 128 samples per MIDI note selectable via velocity or controller cross-switching
- Per-sample volume, tune, pan, sample start, FX send, envelope and mute grouping

Effects

- 2 assignable MIDI effects per program
- 2 assignable Audio effects per program

Instruments

- Play samplers, synthesizers and physical models at the same time in user defined 'Splits & Layers'

Content Included

- Over 2GBs of content including Black & Whites pianos, Pop Drums, and new Orchestral Strings library, 1500 analog synth patches, and Osmosis for converting Roland and AKAI libraries.

Filters

- 2 assignable stereo filters per voice
- 13 filter types, including 4-pole resonant lowpass, highpass, bandpass, allpass, notch & state-variable

FileTypes

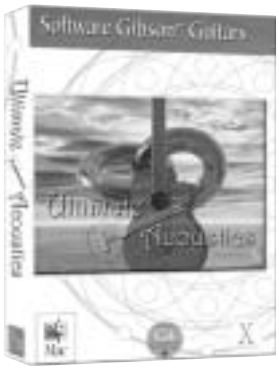
- Plays 24-bit Gigasampler, Retro AS-1, Unity DS-1, Sound Designer II, AIFF, SampleCell II, DLS, SoundFont 2.0 and WAVE files

UNITY-BASED SOUND MODULES

Each of the Unity sample libraries include their own sample playback engine based on DS-1 software sampler technology. Each library has a user interface, features and controller capabilities customized for that library. For example, Unity Discrete Drums allows you to mix the wet/dry mix for the samples using a MIDI sequencer or your keyboard's mod wheel, while Temper Tantrum employs Oscillator Stretch which allows you to instantly match any loop to your songs tempo.

Ultimate Acoustics

Gibson Guitars
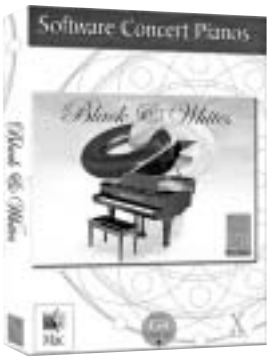


Ultimate Acoustics is a collection of vintage acoustic guitar and bass samples performed by Bryan Blumer and produced by David Das. All samples were sampled in 24-bit, 44.1 kHz resolution, some in mono and some in stereo, to give you full fidelity. Each guitars folder contains a few different playing styles including the following:

Singles banks contain samples of single notes, picked or plucked. Open and fretted notes are mapped across the keyboard to simulate how a real guitar would sound.

Chords banks contain samples of strummed major and minor chords plus several alterations. Each program is mapped out from C3 to B4 and broken down into a down-strum and an up-strum for a realistic sounding rhythm guitar. Extras banks contain useful extra samples such as mutes, harmonics, and harmonic chords for each guitar.

Black & Whites

Software Piano Module

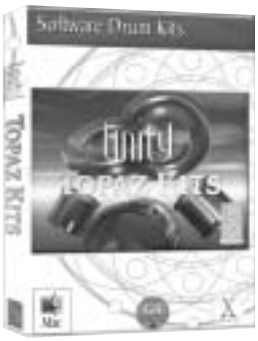


Black & Whites is a software piano module for your Mac or PC. It brings the legendary 7′ and 9′ Steinway Pianos to your computer. Based on technology developed for the award winning Unity DS-1 Sampler, Black & Whites uses the same playback engine to reproduce the professional quality sound found in Unity DS-1. Over 330 MB of samples are included, with instrument sizes ranging from 3 MB to 44 MB in size. Over 1400 individual programs cover everything from sonic purity to pure insanity.

Black & Whites was sampled in stereo and many of the models utilize 4 velocity layers. There are also Rhodes Electric Piano samples with many variations. So if you've been waiting for sampled pianos that can stand on their own three legs, listen to Black & Whites.

Unity Topaz Kits

Drum Kits



Unity Topaz Kits is a collection of multisampled drums based on the SonicEmulations Gigasampler format title, Topaz Studio Kits. Each of the drum kits included was sampled while played with various types of sticks and brushes, providing more variety and realism. The kits were created by well known session drummer Zach Danziger using both Yamaha and Gretsch acoustic drum sets. The result is a spectacular collection of high quality drum samples that you can use standalone with the Unity Player engine included, or within your favorite sequencer.

Unity Discrete Drums

Studio Drum Kits



The drum samples in the Unity Discrete Drums library were taken from the Discrete Drums Series One Multitrack 12 CD set. All of the samples are 24-bit audio. The CD allows the users to control the level of the Room tracks, just as in the Discrete Drums libraries. However, in instead of using individual audio tracks, the Unity version utilizes Unity banks.

Each drum set contains three banks - Dry (natural drum set), Room (just the room tracks) and Mix (links to the other two banks and has several preset mixes of the room level).

Tempo Tantrum

Breakbeats Loop Module



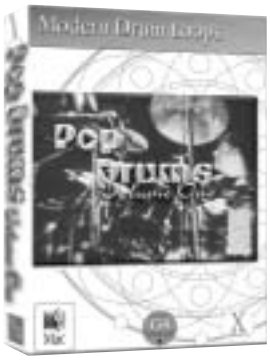
Tempo Tantrum delivers over 240 MB of breakbeats, drum loops, and bass lines straight to your computer. Creative freedom is yours with complete control over the bpm of each loop. Using amazing Unity DS-1 Oscillator Stretch technology, you can instantly adjust the tempo and feel of these breakbeats. You can even change the tempo in realtime with almost any MIDI controller.

With Tempo Tantrum, any drum loop or bass line can be instantly matched with any other loop in the collection. This flexibility allows you to get the most out of this sample library, since you can easily create new combinations of loops, tempos, and effects as needed.

UNITY-BASED SOUND MODULES

Pop Drums Vol. 1

Contemporary Drum Loops



Pop Drums Vol. 1 is the answer to any musician out there looking for a wide range of quality sounding drum loops and bass lines. With over 270 MB of content, you'll be sure to find just what you are looking for from Latin and Soul to Rock and the Blues. Over 270 MB of drum loops and bass lines that can be sped up or slowed down in realtime without changing pitch. Several different programs per bank for different tempos, effects, and pitch-shifting. You can also use the Pop Drums content in Phrazer.

Harry Sharpe Guitars

Guitar Loop Module



This collection contains over 420 MB of multisampled and multilayered Fender Telecaster libraries & loops to choose from including Rock, Funk, Blues, and even a few Country styles. Add studio recorded rhythms, leads, and riffs to your songs without ever setting up an amplifier. Three different pick-up settings give you distorted, clean, and licks in between. Several different programs per bank for different tempos, effects, and pitch-shifting. Loops are also compatible with Phrazer.

Unity Synth Expander 1

Comprised of six plug-ins, Unity Synth Expander 1 contains 500 sounds that can be used in the as a plug-in or in stand-alone mode. The synthesis package can be brought into Unity Session, Unity DS-1, Unity AS-1, or other sequencers. Includes full editing capabilities as well as plug-in support for Pro Tools, Digital Performer, Cubase and Logic.



Unity Expander 1 includes the following:

FM-1: 8 operator FM synthesis

BR-1: Brass physical model

EP-1: Electric Piano physical model

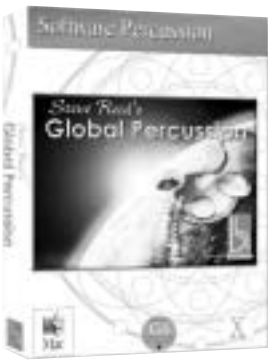
GL-1: Glottal/Vocal physical model

VS-1: Vector synth

WS-1: Wave sequencer synth

Steve Reid's Global Percussion

Percussion Module



World-renowned percussionist, Steve Reid, combines the high-quality samples of his vast percussion instrument collection with the powerful Unity DS-1 Digital Sampler Engine.

Steve Reid's Global Percussion contains pristine samples of exotic percussion instruments from all over the world. No additional hardware is required to play these sounds. The instruments are sampled in the Unity DS-1 format, as well as in Voodoo format.

Miroslav Vitous MINI

Symphonic Orchestral Samples



The Miroslav Mini is a collection of world class symphonic orchestra samples in native Unity DS-1 format. The library is a compilation of orchestral instruments and ensembles from the larger Miroslav Master Set. The Mini Library comes complete with over 25 individual instruments representing the strings, brass, and woodwinds families. The realism and expression of these instruments is unparalleled among orchestral sample libraries, and recreates the symphonic experience with stunning clarity and depth.

Tubes, Tines & Transistors

Vintage Keyboard Module



"Tubes, Tines & Transistors" is a dynamic collection of organ, electric piano, analog synthesizer, and related keyboard instrument sounds that can be played directly from your Mac or PC.s

Using amazing Unity DS-1 sampling technology, you get instant access to over 110 banks and 3,500 instrument programs! The sounds of actual vintage organs and classic analog synths such as Arps, Moogs, Sequentials, and Oberheims have been digitally captured and brought to your desktop.

RETRO AS-1

Software Synthesizer



Using DSP technology, Retro AS-1 is a full-featured software synthesizer which can recreate the sounds of many of the classic analog synthesizers of the past, or create amazing new soundscapes from scratch, with full programmability. Retro AS-1 is fully polyphonic and multi-timbral, with fast response time. Extensive MIDI implementation allows real-time control of all parameters and easy integration into existing MIDI setups. Sounds can be played live just like any other musical instrument with layers, splits, and full MIDI continuous control. Or used as a multi-timbral sound module running behind another MIDI application (such as a sequencer) on the same computer. A full-featured arpeggiator can generate analog-style sequences. Additionally, you can have Retro AS-1 write the synthesizer's complete multi-timbral stereo output to disk, and stored as an audio file for use in another digital audio application.

- 24-bit, 96 kHz sound quality
- 9 waveform: Saw, Pulse, Triangle, Sine, Sine squared, Glottal, Noise (white, pink, red)
- 13 filter types: Includes 4-pole resonant lowpass, highpass, bandpass, allpass, notch, and state-variable
- Up to 3 oscillators per voice not including low frequency oscillators
- 8-octave range per oscillator
- Syncs any oscillator to any other oscillator or filter
- Multiple inputs allow parallel and/or serial filtering and filters can be modulated by oscillators ("poly mod")
- Frequency modulation from any oscillator or filter
- Over 100 parameters per program
- On-screen keyboard
- No MIDI controller needed to trigger sound
- Supports output directly to digital audio file on disk in 8,16, or 24-bit format
- Automatic publishing of OMS and FreeMIDI names



VOODOO *Digital Drum Machine*

Voodoo turns your computer into a programmable "beat box" that plays samples and standard MIDI files. It contains over 20 different "Themes" so you can customize the program to your liking. All functionality is accomplished within one window where you can drag the samples you want from your hard disk onto the drum pads. By clicking on the drum pad you can then view its individual control parameters and its sample waveform. When your finished making the adjustments that you want, exporting the samples is as simple as dragging to your hard disk. So when you're ready to lock down a groove, Voodoo gives you both the simplicity and flexibility to do what's most important: Make music.

- Outstanding 24-bit, 96 kHz sound quality
- Up to 64 note polyphony (CPU dependent)
- Drag and drop sample importing and Standard MIDI file importing
- On-screen keyboard
- Parameter values displayed in hertz or ms
- Use standalone or play Voodoo from your MIDI sequencer

Sampling

- Imports AIFF, Unity DS-1 and SDII files.
- DSP functions include Gain, Shelf EQ, Parametric EQ, Reverse, Flange and Delay
- Up to 4 samples per "pad" selectable via velocity or controller switching
- Per-sample volume, tune, pan, sample start, FX send, envelope and mute grouping

Oscillators

- Single stereo oscillator per pad
- 8-octave range per oscillator
- Course and fine tuning

Filters, Effects & LFO

- Single filters per pad with variable cutoff, resonance, and Overdrive controls
- Filter types: lowpass, highpass, and bandpass
- Envelope and velocity control of filter cutoff
- Effects: Delay, Reverb, Reflection
- The LFO is routable to pitch, filter, or amplitude and has access to four wave-forms, as well as delay, ramp, and speed controls

Patterns

- Event list editing, quantize, shift, pad note, velocity and length edit functions.
- Each pattern has # repeats, tempo, pan, volume and FX 1/2 offsets
- Import and arrange multiple patterns

Output

- Supports output directly to digital audio file on disk in 8,16, or 24-bit format
- ReWire support for Cubase, Digidesign Direct I/O, ASIO and MAS support

MIDI

- Real-time control of all parameters via MIDI
- OMS, FreeMIDI, Logic & serial port support
- Sync to MIDI clock

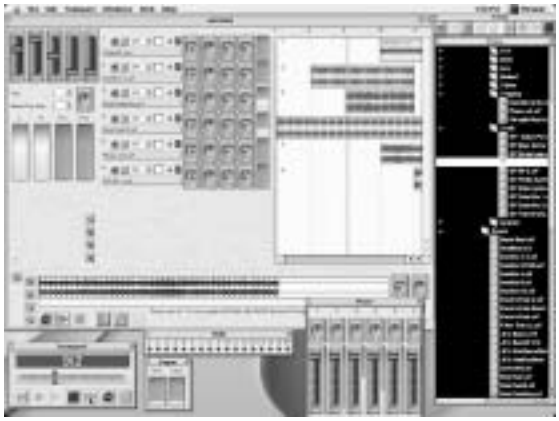
PHRAZER



Loop Composition Software

Take loop composition to the next level. Powerful pitch and tempo matching algorithms, built-in sample editing, mixing, and on-board digital effects provide total control of your sound. Integrate the power of multi-track digital audio and MIDI into one of the most comprehensive and easy to use loop-based production systems ever.

Phrazer is a powerful loop-based music production studio that makes it easy to take audio loops of different tempo and pitch and combine them to easily create music. Phrazer includes built in effects and an audio waveform editor. It imports Acid loops and songs, as well as WAV, AIFF and MP3 files with the ability to automatically detect the "hit points" or transients in the waveform and insert markers at those points which allow for Phrazer's tempo adjustments. Phrazer also has real-time MIDI performance features - you can assign a MIDI note number to each of the Phrazer audio tracks and then gate that track on or off using the MIDI keyboard. You can also assign QWERTY keys to each of the tracks for the true portable DJ set-up!



FEATURES

- Over 900 MB loops included
- Ability to create a master project folder that contains not only the sequence, but duplicates all samples used
- Compatible with most standard sequencers including ProTools, Digital Performer, Logic and Cubase
- Powerful pitch and tempo matching algorithms easily match up different audio loops
- Built-in sample editor with DSP for fine tuning audio files and customizing loops for changing tempo without changing pitch
- Roland PC 300 MIDI keyboard support
- Multitrack audio editing and playback
- Import MP3's, ACID I and II files, Unity DS-1 samples, AIFF, Sound Designer II, WAV files and CD-Audio
- Tempo calculation to set tempo based on sample length
- Assign multiple loops and DSP effects to each track. This makes it easy to keep track of your song since you can have all of your drum loops on one track, all of your bass loops on another track etc. Then you can drop insert effects right onto your track and automate them along with pan, volume and more.
- One shot tracks retrigger a sample each time a key is pressed for stuttering or scratch effects.
- Optimization for Apple's G4 for maximum performance
- Stream audio from hard disk or load audio into RAM
- Use ASIO, DirectConnect, Direct I/O, MAS 2.0, ReWire or Sound Manager for output
- Sync to MIDI clock for integration with studio hardware and software
- Unlimited tracks available (CPU and RAM dependent)
- QuickTime movie support
- Two send effects available per track including: Phrase, Flange, Chorus, Degrade, Distortion, Filter, Dynamic filter, Compressor, Delay with MIDI sync, Shelf/Parametric EQ and Reverb
- Hide editor view in main window
- Plug-in effects processor
- 24-bit support for sequencer rendering and record to disk

Phrazer LE

An alternative to Phrazer, the LE version is perfect for a beginner, or for those who don't need all of the features that come with the full version. Users have the ability to create a soundtrack to a home movie, use it live as part of your rig or make full songs using built-in FX, advanced editing features and the 500 MB of loops included! Three primary differences between the LE and full version:



- Amount of content (500 MB vs. 900 MB with full version)
- Sequencer support only in full version • LE version is limited to 8 tracks

PHRAZER



Phrazer Discrete Drums

Studio Drum Loops

This content CD contains 280 MB of Phrazerized drum loops taken from the Discrete Drums Pro Series Rock/Alternative collection. Phrazer Discrete Drums provides both 16-bit stereo and 24-bit mono loops.

The stereo loops contain the entire drum kit for those of you don't want to fuss with balancing the individual mics. The result is 24 Phrazer Discrete Drums songs broken down into Verse, Chorus, Fills, etc. But if you want to break it down even further, the other half of the content CD provides the 24-bit mono loops. Here in the Phrazer Discrete Drums "Parts folder", you can adjust the amount of Overhead and Room to your liking.

AfroCuban Percussion

Tropical Rhythm Module



AfroCuban Percussion contains 130 MB of quality sounding Caribbean-style instruments and rhythms. You will find Mambo, Merengue, and many other loops in this tropical library. The loops are "Phrazerized" so all you have to do is bring them into Phrazer and your on your way to a creating a fantastic island beat.



Underground Beatz Vol. 1

The library is a "best of" collection of the Cologne Cyclez I and II and the Hamburg Loopz 2 libraries from Wizoo. A must- have for those into the techno, electronic sounds of the European underground, Cologne Cyclez provides classic 133 bpm loops—perfect for the electronic DJ. The Hamburg Loopz 2 content contain many unusual and experimental patterns shaped by effects, distortion and analog synths.

Underground Beatz Vol. 2



Over 300 MB of quality sounding loop staken from Wizoo's Rude Loopz and Lofi Junkiez 2 content titles. Lofi Junkiez presents you with electronic cologne funk, Trip hop, Hip hop & Breakbeat drum loops from the European underground scene. You will surely be inspired by the loops brought to you by Cologne underground DJ's, with special guest, Craig Anderton.

The Rude Loopz content features a drummer recorded live to analog tape using vintage tube and solid state equipment. All of the loops in this folder have two 2-bar loops, each played and processed differently.

Phrazer Smart Loops

AfroCuban Percussion

These three Phrazer titles come to you courtesy of Smart Loops. All of the original loops were created specifically for use as loops, which allows you to mix and match them up freely. The three "phrazerized" titles based on the Smart Loops versions including Bass Guitars, Drums & Percussion, and Electric Guitars. You can mix and match the three titles as needed and bring them into Phrazer, Phrazer LE, Unity DS-1, Unity Session, or any of your favorite sequencers! No additional hardware required. Free demo version of Unity Session included with each title.



Bass Guitars

- Over 950 "phrazerized" loops
- A variety of styles including Rock, Pop, Funk, Blues and Heavy Rock
- Includes 4-string finger and pick, 4-string slap, 5-string with finger, 5-string slap and Fretless Bass
- Most bass loops recorded without any effects, giving you complete flexibility

Drums & Percussion

- 670 "phrazerized" loops in a variety of styles including Afro Cuban, Latin, Rock, Pop, Funk, Blues and Heavy Rock
- Create your own unique 2 and 4 bar riffs
- Instruments including Congas, Bongos, Triangles, Wood blocks, and many more
- Most loops recorded without any effects, giving you complete flexibility

Electric Guitars

- Over 950 "phrazerized" Loops
- A variety of styles including Rock, Pop, Funk, Blues and Heavy Rock
- Includes Guitar (clean), Guitar (Wah pedal) and Guitar (Heavy distortion)
- Most loops recorded without any effects, so you can easily add your own

BOMB FACTORY

'CLASSIC' PLUG-INS

Bomb Factory makes photo-realistic plug-ins modeled after analog devices built by other manufacturers. One look at Bomb Factory's software and you'll see the difference. There aren't any little sliders to click on. And no long columns of numbers to type in. Instead you get photorealistic representations or professional studio equipment.
The plug-ins are developed with the cooperation and direct involvement of these manufacturers and other recording engineers to ensure the plug-ins are true to the original devices. For instance, Bob Moog was involved in creating the Moogerfooger Lowpass Filter and Ring Modulator plug-ins. (They even have his signature).

Classic Compressors

Bomb Factory's Classic Compressors look, sound, and work just like the real thing. Whether you're just learning to use compression or a seasoned pro polishing a final mix, you'll love these time proven designs.

Teletronics LA-2A



The LA-2A tube compressor's delivers more than just "warmth" – it also has charm.

Teletronics LA-2A / Urei 1176

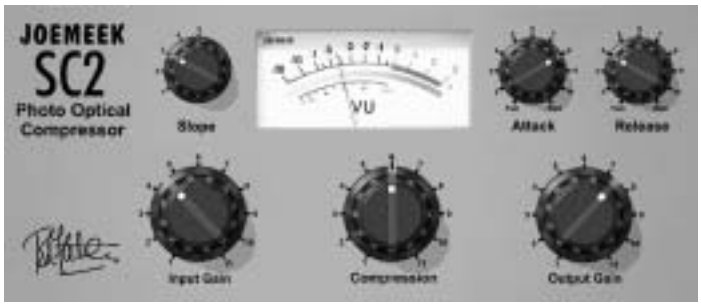
Meticulously crafted digital versions of the Teletronics LA-2A tube compressor/limiter and the Urei 1176—the most popular vintage compressors used in top pro studios—Bomb Factory's proprietary modeling technology captures every tube, transformer and transistor of the originals! Plus the software adds sidechain support and perfect stereo tracking—two features not available in the "vintage" domain.

Urei 1176



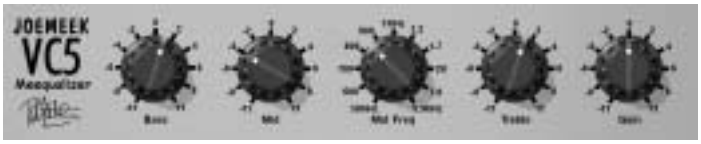
The 1176's classic compression action works magic on any voice or instrument.

JOEMEEK Compressor



Add a little magic to your mixes with the JOEMEEK SC2 Compressor.

JOEMEEK Meequalizer



The JOEMEEK Meequalizer offers simple controls and incredibly warm, musical results.

JOEMEEK SC2 Compressor

Legendary producer Joe Meek used to say: "*If it sounds right, it is right.*" Nowhere is this more apparent than in Joe Meek's masterful use of non-linear, sometimes severe compression in his productions.

Designed purely as an effects compressor, the JOEMEEK SC2 change the way the ear perceives sound. It changes the clarity, balance and even rhythmic feel of music. And by offering sounds no other compressor would dare, give your tracks an undeniable edge. In use by top producers the world over, JOEMEEK Compression is the secret weapon that gives your sound the character and excitement it deserves.

JOEMEEK VC5 Meequalizer

Picture this: drums in the spare bedroom. Microphones, cables, and recording gear strewn about the living room. A familiar scene, especially to legendary producer Joe Meek in 1962 as he prepared to record yet another chart-topping hit. Among countless other achievements, Joe Meek built custom gear to get the sounds in his head onto tape. Once device was a treble and bass circuit with a sweepable mid control.

The JOEMEEK Meequalizer re-creates the exact circuit used by Joe Meek. A great-sounding, easy to use equalizer. Try it on mono or stereo tracks! Don't forget, it was designed by a studio legend.

BOMB FACTORY

'CLASSIC' PLUG-INS

moogerfooger

Authentic analog designs from electronic music legend Bob Moog, built with amazing Bomb Factory digital technology! Moogerfoogers combine the best features of audio effects and vintage synth modules, opening new frontiers for sonic exploration. All are available separately, or together in a specially-priced bundle.

Lowpass Filter

The moogerfooger Lowpass Filter features a 2-pole/4-pole variable resonance filter with envelope follower. Use it to achieve classic '60s and '70s sounds on bass and electric guitar, or dial in warm, fat analog resonance on any instrument.

Analog Delay

The moogerfooger Analog Delay is simply the coolest (and warmest sounding) delay in the digital domain. Due to component shortages, fewer than 1,000 "real world" units will be made. Bomb Factory snagged serial #1 and meticulously modeled every detail of Dr. Moog's classic analog design. Wait until you hear it!

Ring Modulator

The moogerfooger Ring Modulator provides a wide-range carrier oscillator and dual sine/square waveform LFO. Add motion to rhythm tracks and achieve radical lo-fidelity textures — you set the limits!

12-Stage Phaser

The moogerfooger 12-Stage Phaser combines a switchable 6- or 12-stage phaser with a wide-ranging variable LFO. Start with subtle tremolo or radical modulation effects, then crank the distortion and resonant filters for unbelievable new tones — all featuring classic MOOG sound.

Lowpass Filter



Analog Delay



Ring Modulator



12-Stage Phaser



Who is Bomb Factory?

Assemble a team with a proven history of developing award-winning, legendary audio products. More than 100 years combined experience designing real-world equipment. Add in a range of recording studio experience. People who've produced, engineered, and performed on countless recordings. The result?

Digital audio tools that don't sound like they limped out of some computer programmer's cubicle. Analog meters with proper ballistics. Faders with correct logarithmic tapers. Fliters that sweep, resonate and oscillate. And knobs that actually turn in the direction you want them to go. Combine that core competence with some of the best creative thinking in the industry. Computer models which incorporate physical simulations of motors and pulleys. Distortion that growls the right way when you crank it up. On-screen control with scratches and dents, casting shadows that move as you turn the knob. Bomb Factory doesn't make "plug-ins". They make tools that let you get the job done. They sound great, look great, and work the way you'd expect a professional piece of gear to behave.

BOMB FACTORY

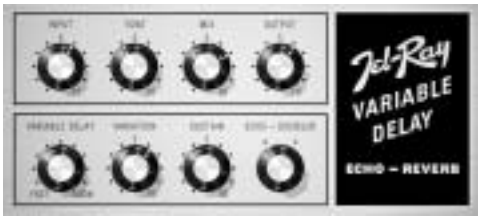
'CLASSIC' PLUG-INS



SansAmp PSA-1

The SansAmp PSA-1 provides the widest range of amplifier, harmonic generation, cabinet simulation and equalization tone shaping options available. It appears on literally thousands of recordings, used on everything from guitar and bass to drums and harmonica!
All 49 original SansAmp presets — featuring Marshall, Mesa Boogie, Hiwatt, Fender Lead, Rhythm and Bass; and Ampeg SVT sounds — are faithfully and authentically recreated. Explore dozens of new presets contributed by well-known artists and producers, then create your own unique tone!

Tel-Ray Variable Delay



In the 1960s, a small company experimented with electronics and technology. When they came up with something great, they would Tell Ray (the boss). One invention involved a tuna can, a motor, and a few tablespoons of cancer-causing oil. A technology, they were sure, that would be of great interest to companies like IBM and NASA. Though it never made it to the moon, the killer technology that gives Tel-Ray its unique sound was licensed by most every major guitar amp manufacturer! Ad-N-Echo to any voice or instrument using the Tel-Ray Variable Delay. It provides lush delay, amazing echo, and warms up your tracks and mixes.



Voce Spin

From Voce, classic effects that sound great on any instrument. Voce Spin provides the most accurate simulation of the well-loved rotating speaker. It's a favorite of producers, guitarists, and, of course, organ players. Fifteen classic recording setups feature different speaker cabinets, varying microphone placement — even the "Memphis" sound with the lower drum's slow motor unplugged! No belts to tighten, not tubes to wear out—just choose a preset and get the sound of a professionally-miked rotating speaker; instantly. Or customize exactly the sound you want using over a dozen fully automatic parameters. Voce Chorus/Vibrato recreates the B-3 Organ's mechanical scanner vibrato. Three settings of Chorus and three settings of Vibrato on one cool knob. Fun and easy to use, it's a classic effect used for over sixty years. Talk about vintage!

Fairchild 660 Compressor

Designed in the early 1950s, the Fairchild 660 is a variable-mu tube limiter.
It features an unusual form of vacuum tube that is capable of changing its gain dynamically. The result? In addition to featuring a tube audio stage like an LA-2A, the Fairchild actually achieves gain reduction through the use of tubes!
Bomb Factory's no-compromise replica captures every detail of this legendary piece of studio gear. But unlike the original, Bomb Factory's version is affordable enough that real musicians can afford to own one. Run a dozen or more at once — even in stereo — using any Pro Tools or MOTU system.



Pultec EQP-1A Equalizer

The Pultec EQP-1A provides smooth, sweet EQ and an extremely high-quality tube audio signal path.
And like most studio classics, it's incredibly easy to use. Grab a knob, give it a twist, and it just sounds great. Every time.
Bomb Factory's award-winning digital modeling expertise — combined with pro studio engineering chops — delivers the sounds you know and love.Use it anywhere you'd use the original: on individual tracks, critical vocals, or even across a stereo mix for mastering applications.



B&H PHOTO - VIDEO - PRO AUDIO
EQUIPMENT LEASING AVAILABLE

'CLASSIC' PLUG-INS

BF Essentials

A full suite of time-saving, trouble-solving studio essentials– for under $100.

BF Meter Bridge

The BF Essential Meter Bridge provides Bomb Factory's best-of-breed analog metering on any channel, using minimal DSP resources. Avoid flickering green lines and blinky fake LEDs, and enjoy the ease of use afforded by a needle, a big meter, and the only company that bothers to pay attention to the decades-old standard for meter ballistics.
Select RMS or Peak metering, and calibrate instantly for useful viewing at any signal level, just like a pro tape machine.





BF Essential Tuner

Until now, plug-ins could emulate every guitar amp and speaker cabinet known to man, but couldn't tune a guitar. Problem solved with the BF Essential Tuner.
The BF Essential Tuner uses the power of TDM and modern host CPU's to track better and tune faster than any of those battery-eating handheld gizmos. It works great on bass, electric guitar, vintage synthesizers– you name it.

BF Clip Remover

The BF Essential Clip Remover repairs clipped audio recordings. That red light no longer means a lost take! You'll be amazed how quickly this essential tool can repair clipped recordings. Best of all, it's way faster and more accurate than using the pencil tool. Set your levels very carefully. But when you blow it... try the BF Essential Clip Remover.

BF Essential Correlation Meter



Solve tracking and mix problems, and troubleshoot phase coherency using the BF Essential Correlation Meter. It works on stereo tracks or stereo submixes. Stop phase problems before they start!

BF Essential Noise Meter

The BF Essential Noise Meter is three meters in one!

- Set to "A" it's an A-weighted noise meter
- Set to "R-D" it's a Robinson-Dadson equal-loudness meter
- Set to "None" it's a VU meter with 100dB of visual range!

◆ Mono and stereo versions included.

Funk Logic Mastererizer



A mastering tool of unprecedented quality for MOTU Digital Performer, Digi 001, Pro Tools MIX, and Pro Tools HD users. Developed in partnership with Funk Logic, the Mastererizer high-quality potentiometers — constant impedance audiophile-grade components produce no crackles or noise. Hard knee or soft knee? It's a tough question to answer. The Funk Logic Mastererizer solves the problem by not having a compression section at all. It's fully automatic and simply "does the right thing."

◆ The Mastererizer delivers Bass Authority, which allows it to go from subsonic woofer-shaking to bass tones without adjustments in gain on the track.

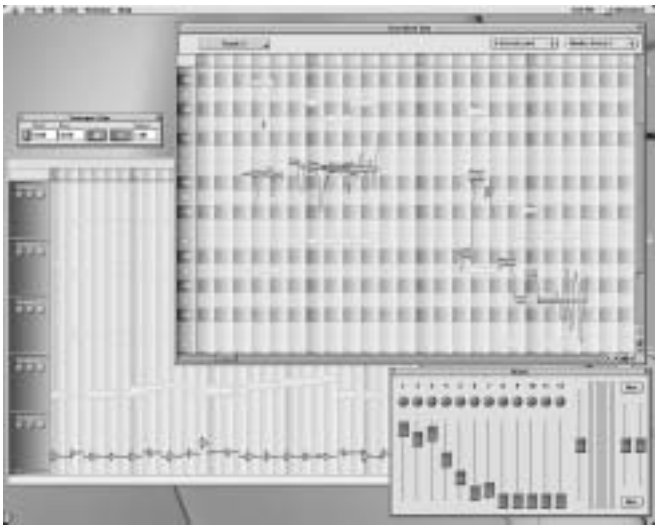
◆ Mastering-quality UV Meters: Superfast slew rates offer the benefit of Peak Metering with the convenience of RMS math. Plus you can calibrate -12, -18, or -24 equal to -0dBSPF.

◆ Hundreds of presets for every type of music: Finding the sound you want is easy, as nearly every possible combination of knob settings has been replicated in the preset menu.

MELODYNE

New Dimension in Audio Recording

Melodyne offers a completely new approach to the handling of audio material. It analyzes the pitch and time of monophonic audio files (for example, from, singers, wind or string instruments) and offers the opportunity to change whole melodies in a way only previously possible at MIDI-Level. Melodyne is able to change the musical parameters of voices or instruments without any actual influence on the character of the recording. Melodyne extracts the pitch and the rhythm, and provides an area in which to play with the audio material. Any change in this material is interpreted by the software in a musically intelligent manner and the result will always sound natural. Working with Melodyne is as easy as editing notes with MIDI.



FEATURES

- Melodyne detects the melody of any mono audio recording and extracts its notes. The tempo, pitch, formant, quantization and time position of these notes, can all be edited as easily as if it were MIDI data.
- Pitch Shifting and Time Stretching are automatically treated as completely independent parameters
- Detected notes can be moved arbitrarily with the mouse in pitch or time, or can snap to any musical scale.
- The tempo of an arrangement can be changed arbitrarily while playing, in a tempo range of ±50% changes without any noticable artifacts.
- You can create new melodies and correct intonation problems with a mouse click — Pitch shifting and formant correction by more than an octave can be achieved without affecting sound character and velocity.
- Control over formant (resonance volume) allows you to transform a trombone into a trumpet or a tenor into a soprano voice — important for preserving the quality of an audio file that is being pitch-shifted.
- Copy and paste a melody numerous times to create lush harmonies and use formant correction to give each harmony the appropriate timber for its range.
- A new melody can be derived from single notes or a new arrangement can be created from single tracks via copy and paste. Differences in beat will be corrected automatically.
- Variations in intonation can be achieved by an increase or decrease in phrasing or vibrato.
- The rhythm of an audio file is recognized, as each given note "knows" its place within a beat. Notes can be quantified automatically, and they can also be synchronized to a reference melody with a mouse click.
- When applying pitch changes to notes, even with extreme pitch changes, the internal phrasing of a note, e.g. vibrato, and the transition between notes will retain its musical musically intact. So between the notes of a melody it will hardly be perceived that the melody was not played that way originally. Melodies can be rearranged by copying and pasting notes, or multiple voicings can be realized by copying a melody to new tracks.
- Flexible time-stretching allows acceleration and deceleration of a single note or of a whole arrangement through the concept of local sound. There are no limits to deceleration, and a standing sound will be the ultimate result when velocity is zero.
- Supports up to 24 tracks in real time.
- Single tracks can be recorded within Melodyne, or can be imported from other recording programs.
- VST2 and Audio Unit plug-ins can be loaded, plus single tracks can be routed to an external mixer via ASIO or Core Audio.
- The mixer section features an EQ section with support for submixing-groups.
- MIDI integration allows you to use a MIDI keyboard or controller to edit and mix.
- Audio can be routed to your DAW software via ReWire or you can save tracks as single audio files for processing in other recording programs.
- The detected melodies can be exported as a MIDI-File to be viewed in a notation program or to be used in a sequencer software.

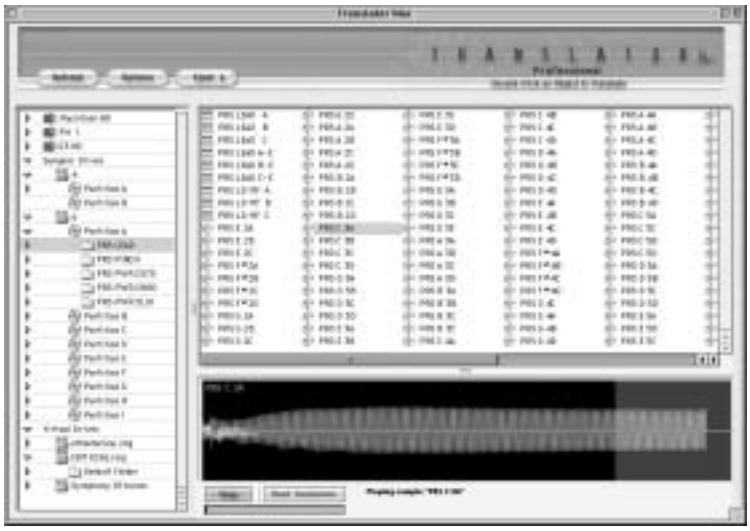
System Requirements:

- PowerMac, Mac OS 9 or X and 128 MB free RAM (256 or more recommended)
- Windows 98, ME, 2000 and XP with PII400 class processor and 128 MB of free RAM (1GHz and 256 MB recommended).
- Supports DirectX, SoundManager, ASIO2, and CoreAudio for sound I/O and many audio formats (WAV, AIFF, SD2, SND, AU).

CHICKEN SYSTEMS

TRANSLATOR

Universal Sample Convertor For Mac and PC



Translator is an easy to use sample format convertor that will take almost any major professional disk and file formats and allow you to translate between them. That means reading, interpreting, and writing between different systems. Translator uses an intuitive "Windows Explorer-style" interface and conventional drag and drop methods to translate between just about every professional sampler format available. A wave player allows you to audition individual samples. Translator converts samples and their associated programs quickly and accurately with a minimum of fuss. You can even create disk images, based on proprietary sample formats, and burn them to CD using your CD burning software. Whether you are working in multiple sampler formats or migrating from a hardware to software based sampling, Translator offers you a flexible and hassle-free solution.

FEATURES

Reads and Writes ALL Popular Sample Formats

- Convert one Sample, one Preset, one Bank, or an entire CD-ROM from any professional format into any other.
- Translator allows your computer to mount your sampler's proprietary formatted ATAPI, SCSI and USB drives, including fixed, CD-ROM and removable drives.
- Program parameter information is translated as well as the samples and structure - wavedata, keymaps, looping, effects etc...
- Supports 8-bit, 12-bit, 13-bit, 16-bit, 24-bit, and 32-bit files

Unique Two-Way Interface

- Translator's exclusive "Windows's Explorer-style" Two-Way Interface makes sample conversion simple: Select what you want to translate on the right; Drag it over to the left; Decide what you want to translate to and you're done (In fact, you can pre-select the destination format.
- You can drag and drop any level of file - a single sample; a program, instrument, or preset; banks, volumes, or whole directories; or even entire disks.

Wave Player

- The built-in Wave Player allows you to audition individual samples from within Translator. This works on samples from any format and any disk.
- The full sample is displayed along with the loop area (if there is one).

Translation Compatibility

Akai	read/write
Roland	read/write
Emu	read/write
Kurzweil	read only
Ensoniq	read/write
EXS-24	read/write
Giga	read/write
HALion	write only
SampleCell	read/write
SampleTank	write only
Reaktor	read/write
WAV/AIFF/SND/KSF/S	read/write
Pulsar	read/write
MESA	read/write
Triton	write only
XV-5080	write only
ASR-X	write only
SoundFont	read/write

Virtual Drives

- Translator allows you to make Virtual Drives, which are essentially large single files, or data images, that appear as SCSI-ATAPI Drives formatted with whatever proprietary format you want (Akai, Roland, Ensoniq, Emu).
- You can use a Virtual Drive to compile a disc image in any proprietary format and then burn it onto a CD-ROM, using any CD Burning program that will recognize raw disc images (most do).

Translation Special Editions

- If you only need to translate samples and programs into only one specific format, you might want to consider the Special Edition of Translator, which will translate from all available sources to one specific destination.
- Special Editions currently available are the Translator EXS-24 Edition, HALion Edition, SampleTank Edition, and SampleCell Edition.
- Each Special Edition version is dual-platform - for one price you can use the program either on Windows or Mac.

EASTWEST / QUANTUM LEAP

SYMPHONIC ORCHESTRA



Symphonic Orchestra was recorded where orchestral instruments sound their best - in a concert hall. Listen to the beauty of the critically-acclaimed Symphonic Orchestra and state-of-the-art $125 million concert hall that were captured, using custom built recording equipment, by audio legend - Prof. Keith O. Johnson - recipient of two Grammys, plus eight additional Grammy nominations for his 90-plus classical recordings.
Available in four versions (Strings, Woodwinds, Brass and Percussion, or as a complete collection), these are the first 24-bit orchestral sample libraries to include three simultaneous stereo mic setups (close, stage and hall), so users can mix together any combination of mic positions to control tonality and ambience. A major benefit of this recording philosophy is the tonal control the three simultaneous stereo tracks provide. Users can literally alter the tone and ambience of any instrument or section by incrementally moving out from the close mics, to the stage mics, to the hall mics (an audio zoom) in real-time!

FEATURES

- The "Virtual repositioning of the listener" in a concert hall environment offers a surprising amount of tonal control, virtually eliminating the need for EQ; and the ambient control, using any combination of the three stereo tracks (the close, stage and hall mic setups), eliminates the need for artificial reverb– which usually sounds artificial.
- Create the sound "you" want. With other orchestral libraries, you get what they give you, and apart from adding reverb and EQ, you have virtually no control over the sound of the instruments. The control of tonality and ambience in the Symphonic Orchestras enable each user to "design" the sound of the orchestra to suit the work that is being created— no other collection offers this!
- Say goodbye to lifeless, dry studio recordings.Use the stage mics for a typical big Hollywood sound; boost certain instruments by adding a hint of the close mics; add in the hall surround mics to create a surround sound mix, or to add natural hall reverb.
- In addition to this unprecedented control of tonality and ambience; the real ambience of the hall, combined with the automatic sample switching of up and down strokes that is built into the Native Instruments audio engine and interface (supplied with each volume of the library), eliminates the "machine gun" effect when playing fast repetitions.

Each volume comes as a virtual instrument which includes a high-performance Native Instruments interface powered by the efficient Kontakt audio engine (including Direct-from-Disk streaming) for Mac and PC.



- All of the players were recorded in position. All of the instruments and sections were chromatically sampled, with multiple dynamics, with extremely dynamic and expressive articulations, and although the library is huge, it has been programmed to be "extremely easy to use".
- Each virtual instrument combines intuitive handling with excellent sound quality, and functions as a plug-in instrument, without the need for a sampler, in hosts such as Cubase VST, Digital Performer, Logic, Nuendo, Pro-Tools, Sonar etc.
- Many sound shaping tools are built in to the interface. Powerful multi-mode filters, envelopes, and LFOs give a wealth of creative possibilities. Integrated high-quality reverb, chorus, and delay effects are also provided.
- Direct from Disk playback easily handles large samples beyond available RAM (which is assignable).
- Professional sound quality with advanced 32-bit processing; eight-part multitimbrality and 256 voice polyphony allow for lush arrangements. The clean and intuitive interface never lets the technology get in the way of your creativity. All important instrument parameters can be immediately adjusted without the need to page, scroll, or fumble through a manual.
- Supports VST 2.0, DXi, Soundmanager, ASIO, FreeMIDI, OMS, Audio Units, Core Audio, RTAS. Sound libraries can also be loaded into other Native Instruments samplers and KONTAKT (v1.3 and up) for more flexibility and more options for working with the material.

B&H PHOTO - VIDEO - PRO AUDIO

STORMDRUM



Stormdrum features 4 world-class percussionists and a lunatic producer with enough taiko drums to fill an 18 wheeler. Recorded at the Record Plant, Studio A, this library contains an amazing 4.5 GBs of loops and multi-samples. There are 1000's of completely original, evolving drum beds, featuring some of the largest and most impressive drums on the planet. Multi-sampled drum kits and percussion with up to 20-way velocity-switching, organic tech percussion, metal shop, big hits.

- Stormdrum includes intense multi-sampled acoustic drumkits. Each drum has up to 24-way velocity switching, with left and right hand samples. There are subtle ambient versions and huge sounding versions as well.
- There are 18 snares. These have left and right hand samples from the inside and outside of the drum, as well as rimshots. A typical hi-hat contains 200 samples. There are 20 kick drums, 10 cymbals and 8 tom sets.
- Includes amazing new tech drums based on acoustic samples that have been manipulated and programmed with 8-way velocity switching for an organic-tech feel. There are also 100 metal shop samples and a large collection of monstrous hits.
- Also includes a breathtaking set of action toms for film work.
- Has an impressive collection of multi-sampled percussion. Typically, one drum is comprised of over 100 samples. The drums are, taikos, turkish bass drums, chilean bass drum, darubuka with rattle, cuban bongos, large african djembe, lion drum, belly drum, 2 talking drums, large bodhran, dumbek, shakers, talking coconut, chun-mo and frog.

Stormdrum Loops

As an example: Centrifuse contains 15 different 4 bar loops, including an intro, ending and remix.

Action Beds

Centrifuse, Chain Reaction, Daredevils, Grand Theft, Kingpin, Manhunt, Prowler, Speed Freak, The Getaway, Violence

Big Beats

Basilica, Big Trouble, Bigshot, Conquest, Coot in the Shoot, Martyred Revolutionary, Red Leppelin, Righteous Hombre, Skywalker, Slippery Sam, Sloppy Jose, Spin It, Thrillsport, Up River

Ethnic Chase

Bazzar Chase, Camel Racing, Caravan, Carnival, Dark Towers, Horses, Imperial Alajah, Kvrumpaniil, Meca, Ongo Bongo, Opium Market, Panaang, Turbo Turban

Fastbreaks

Blackjack, Diamond Dog, Digital Bandito, Dingo, Dr. Schwarzbaum, Fasttrack, Gauntlet, Gravy Train, H763, Harry Nipfuls, Hornet, Knobby, Marge Da Barge, Momentum, Thorazine, Through the Ringer

Film Tech

Aang Djun, Admiral Synapse, Avalanche, Breeders, Countdown, Dark Ringer, Footsteps, Gobstopper, Graveyard Saviour, Green Goblin, Leppelin Island, Machine Chant, Magic Descent, Mean Machine, Mogaete, Petraglyph, Smack, Wall People

Thunderous

12 Shooter, Ape City, Around the Island, Assasins, Black Horse, Chariots, Clone Army, Deep Space, Dragon Beat, Min City, No Way Out, Pharoah, Pit of Fire, Re-entry, Sandstorm, Scottish War Drums, Seismic Cannon, Stormfront, Warhammer, Witness

No Sampler Required!

Like the Symphonic Orchestra (previous page) the Stormdrum library includes NI's (Native Instruments') stunning state-of-the-art plug-in audio engine/interface specifically designed for rhythmic loop playback, manipulation, and mayhem.

The convenient one-screen interface is based on (and provides all the sound manipulation tools of) NI's INTAKT sampler, and features tremendous sound shaping abilities without disrupting the creative flow, including a first-class multimode filter, an envelope follower, two LFOs, effects, and more.

The loops have been set up so that they can be automatically synced to tempo, either via the Beat Machine or the Time Machine.

The Beat Machine has loops divided into their individual hits (slices) which means they can be played at a wide range of tempos without any change in pitch. It also allows the user to give each hit individual settings for pitch, playback direction, pitch envelope, amp envelope, distortion, delay, and more; a MIDI file of each loop can be easily exported so that groove, accent, and feel can be manipulated in any host sequencer. Time Machine time-stretches or compresses loops in real time.

Not only can you sync, stretch, and pitch the loops, but also filter, effect and modulate them as well! A powerful sound-shaping filter, flexible envelopes, syncable LFOs, and distortion, syncable delay, and lo-fi effects can inject even more energy and individuality into the loops.

Supports VST 2.0, DXi, Soundmanager, ASIO, FreeMIDI, OMS, Audio Units, Core Audio, RTAS. Sound libraries can also be loaded into other Native Instruments samplers and KONTAKT (v1.3 and up) for more flexibility and more options for working with the material.

EASTWEST/QUANTUM LEAP

AFROLATIN SLAM



This library is a "must have" for sound designers, music producers, musicians and DJs, who would like to inject an authentic Afrolatin feel into their projects. Over a period of two years, Francis and Makende, together with some very special friends, have put together this eclectic selection of rhythms, loops and hits. Most of the rhythms were recorded at Sultan Sound studio in London, but some were recorded in the field, in Africa and South America, preserving the original local ambience.

Instruments included are : Djembes, Congas, Bongos, Tam Tam, Dondo (talking drum) Timba, Surdo, Rebolo, Cuica, Pandeiro, Berimbão, Timbales, Tongue Drum, Sanza, Cavaquinho, Repico, Tamborim, bells, Shakers and many other more obscure instruments.

- Over 300 loops and samples, comprising more than 600 MBs of sounds. The rhythm loops range from frenetic West African Kpalogo, to pulsating Afro-latin grooves.
- All loops are classified by region, tempo in BPM and key, if a melodic instrument. Most of the 'studio' loops start with a full versions of the loop and are then broken down into individual elements or 'construction kits'.

CONTENTS:

African Section:
Sounds from the Congo, Côte D'Ivoire, Ghana, Nigeria and more, including Afrobeat, Afrofunk, Juju, Afrolatin Fusion, Afrobossa and more.

South American Section:
Batucada, Berimbao, Caxixi, Candomblé, Chôro, Forró, Frevo, Makumba, Maracatu, Pandeiro, Salsa, Samba, Carribean soca.

Instruments & Individual Hits Section:
African Percussion, Balafon, Brazilian Percussion, Kalimba, Molo, Toy Percussion.

Misc Loops & Hits Section:
Classic Break Beats, Funky Dread Drums, Vintage Breakbeat, Funky Jazz Breaks, Garage beats, Dark Drum'n Bass, Vintage Drum n' Bass, and more!

NU JOINTS



Nu Jointz is the long awaited sample library by Vinnie Zummo who created the multi-key award-winning NY Cutz series. This New York City joint is filled with 64 thumpin' construction kitz, over 200 drum loopz in every tip including tiny-fi, mid-fi, lo- fi & hi-fi akoustik drumz, strait ahead hip hop MPC groovez, slammin' R&B, funk, & strait and wak drum machine loopz.

There are hundreds of new fonky buttah loopz including fuzz bass, akoustik bass, stank piano, frequency loopz, old recordz, lounge recordz, over 60 introz & breakz, over 90 individual drum samplez, over 90 skank guitar lix, over 60 fonky trombone lix, muted trumpet lix, a brand new tip, the totally madd Bass In Da' Room Next Door (just what it soundz like!) plus lotz more insanity in that bitin' NY Cutz flavah than we have room to mention here. Over 150 minutez of new flavah. If you're tired of those lame ass kokonut loopz that are out there and want your trax to be keepin' it tite, check out Nu Jointz. and bring some NYC stylin' to your next track.

KONCEPT & FUNCTION



Koncept & Funktion is an underground drum & bass sample library produced by David & Nic Higham in the UK. It features over 500 MB of brand new D+B material including 20 construction kits, plus a huge selection of driving Beats & Loops, Bass Loops, Drum Kit Sounds, Dark FX, Hits FX, Industrial FX, Loop FX, Noise FX, Space FX, Sweep FX, Liquid Funk Loops, Organ Loops, Pads, Pad Loops, Pad Sweeps, Synth Loops, Female Vocals, Spoken Vocals, and Vocal FX.

You´ll feel like you´ve died and gone to heaven when you get your hands on this creative and highly useable drum & Bass toolkit, which provides everything you´ll need to start exploring the genre. You will find yourself creating great tracks after loading only a few patches and you´ll be hard pressed to tear yourself away from your sampler at all! Even if you´re not interested in drum & bass, there´s enough variety and flexibility to make this collection worth investigating for all kinds of music. D & B producers will find plenty to keep them happy and the interface is so user-friendly and powerful and gives so much scope for experimentation that you´re likely to find themselves conjuring up a masterpiece before you know it.

B&H PHOTO - VIDEO - PRO AUDIO
ONE HOUR FREE PARKING
AT 349 W. 34th STREET (with purchase of $100 or more)

THE OPERATING TABLE



A stunning and amazingly flexible library of breaks with irresistible feel and compelling usability, from Emre Ramazanoglu (Return To The Planet OF The Breaks). All the breaks derive their feel and sonic basis from live drumming technique (though some are creatively processed), and were performed by one of four featured drummers (Emre Ramazanoglu or Dave Westlake, with contributions from Jack Yglesias and Steve Ashmore). The recordings were made in London at Metropolis, Padded Cell, Walworth Cage, Triangle, and Line of Flight.

'The Operating Table' was conceived as an alternative to genre-based sample CD's, offering sample-accurate live drum loops with an equal emphasis on diversity of sound and quality of performance. The emphasis has been on creating unique-sounding breaks through extremely creative processing techniques, but with the real feel of a performance to them.

The performances range from four bars to extremely long, and each changes and evolves, providing multiple loop variations. Real kits are the foundation of the sound of each break, although some incorporate additional live triggered analog drum machines or software triggered sounds. Each bar of each break has been designed to loop perfectly as a 1 bar break (sample accurate BPM), or as part of the whole evolving sequence.

PERCUSSIVE ADVENTURES 2

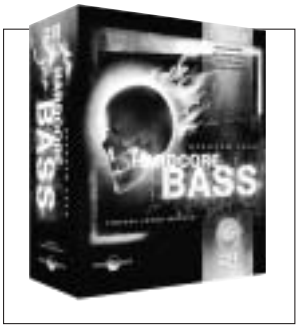


Producers Kurt Wortman and Tony Humecke have gone all out to offer the maximum flexibility. For example, multiple alternate mixes of each loop are presented to enable users to customize the loops to work with their track, or use as building-blocks for new compositions. Includes 5.3 GBs of multi-layered percussion compositions for multi-media productions. Large percussive soundscapes are included covering most styles of music.

WIRED



Wired is a massive and exclusive sound archive from Zero-G that provides every constituent you need to make the perfect trance track. This huge collection includes just about every 4/4 percussion combination that you can imagine. There are also great acid lines, full of energy and vibe, and easy to edit. The pads, pad loops, synth lines and bass loops will help you muster your last ounce of euphoria. There are also loads of effect stabs and snare rolls plus a huge selection of vital drumkit sounds.



HARDCORE BASS

From Quantum Leap producers Nick Phoenix and Pierre Martin, Hardcore BASS includes 2.2 GBs of mix dominating, teeth shattering, hardcore electric bass recorded through massive ampeg bass rigs with the volume turned up to 11! All samples are taken from musicman stingray & lakland 5594 5 string basses using different pick-up and amp settings. Every conceivable articulation of picked and fingered, muted, open, scraped, and mutilated bass - 6,600 multi-samples!

- Advanced programming includes keyswitching, release samples, automatic alternation of up and down picking, chromatic slides, legatos, effects and stacato stopping.
- No sampler required. Includes a custom high-performance Native Instruments audio engine and interface for Mac and PC
- The audio engine functions as a plug-in instrument, without the need for a sampler, in Cubase , Digital Performer, Logic, Nuendo, Pro-Tools, Sonar etc.
- Powerful multi-mode filters, envelopes, and LFOs give a wealth of creative possibilities. Integrated high-quality reverb, chorus, and delay effects are also provided.
- Direct from Disk playback easily handles large samples beyond available RAM (which is assignable).
- Advanced 32-bit processing; eight-part multitimbrality and 256 voice polyphony allow for lush arrangements. Parameters can be immediately adjusted without having to page, scroll, or fumble through a manual.

AMPLITUBE



Physically Modeled Guitar Array

AmpliTube is a guitar amp modeling plug-in for VST and Pro Tools (RTAS and HTDM) users (Mac or Windows). AmpliTube comprises three separate modules — an Amp module, a Stomp Boxes module and a Post FX module. The Amp module offers you numerous models of both modern and vintage PreAmps, EQs, power amps and speaker cabinets as well as mic emulation to produce a whopping 1260 amp combinations. The Stomp Box module features models of five of the most popular effects including Wah-Wah, Delay, Flanger, Chorus and Overdrive. Post FX include a 3-band parametric EQ, stereo reverb and stereo delay. AmpliTubes' highly innovative approach to amp modeling and the availability of all the guitar chain effects, makes it the ultimate software for any guitarist who needs the widest range of guitar sounds from clean to heavily distorted, from dry to fully effected, from rock to jazz.

Amplifier Module

- The heart of the software, the Amp module has been designed using sophisticated modeling techniques, based on vintage and modern must-have amps including Marshall, Fender, Vox, Mesaboogie and many more. Its 32-bit floating point processing ensures for the most accurate amp emulations and offers an incredible realism ranging from solid state to true tube amps modeling. Mic modeling (condenser/dynamic), position and distance, as well as Tremolo and Spring Reverb are also included.
- AmpliTube comes with 200 presets for all the classic amp and guitarist sounds you've ever wanted. Fully editable. Craft your sound, save it and get it back with a single click ! No more hours wasted trying to remember how you created that once-in-a-lifetime sound!
- Play it live! Using low latency card AmpliTube can be played live with the same feeling of having a big stack one foot away!
- AmpliTube offers the best sounds of and hardware or software amp simulator on the market. A-B comparisons show a realism and a dynamic which cannot be found anywhere else - plus an unmatched frequency response. It will be just like recording the real thing!
- Play AmpliTube live with full MIDI control (on Windows-ASIO cards). Simply download the free RT Player LE (by DSound) and you'll be able open AmpliTube VST and control it using a standard MIDI foot controller, including the possibility of switching presets!



Freely combine the 7 different preamp models, 5 EQs, 4 power amps, 9 cabinets and 2 mic models in order to craft amp sounds ranging from the emulation of the most acclaimed modern and vintage amps plus being able to build completely new configurations. The combination of these settings offers a total of 1,260 different amp configurations.

Pre-amp Models:
Solid state clean, Vintage clean, Tube clean, British crunch, Modern hi-gain, Solid state lead, Fuzz

EQ Models:
Tube american, Tube american 2, British class A, Tube british, Tube british 2

Power Amp Models:
Solid state 30W, Solid state 100W, Tube 100W, Tube 50W

Cabinet Models:
- Small combo
- Open back 1x12″,
- Open back 1x12″II
- Vintage open 4x10″
- Modern closed 4x10″
- Vintage closed 4x12″
- Modern closed 4x12″
- British 2x12″
- No speaker

Post FX Module



The Post FX module includes 3 stereo post-effects with 3-band parametric EQ, Stereo Delay and Stereo Reverb, bringing to 10 the number of different effects included in AmpliTube.

Stomp Box Module



The Stomp-box module offers the emulation of 5 effects including Wah-Wah, Delay, Chorus, Flanger and Overdrive modeled after some of the great classic guitar effects.

T-RackS



All-in-One Analog Modeled Plug-in Dedicated to Mastering

If you record music with digital gear, you know how challenging it can be to get that warm, broad, analog sound. T-RackS has everything you need to create superb, tube-toned masters on your desktop. Its rich, warm sound starts with algorithms based on true analog circuitry. Its familiar interface makes it easy to control five powerful processors: a state-of-the-art EQ, a classic tube modeled compressor, a multi-band master limiter, soft-clipping output stage, and a complete mastering suite. Enhances mix frequencies, stereo images, dynamic range and gives your audio a seamless, top notch sound. With separate processors, astonishing warmth and ease-of-use, you'll be putting a polish on your tracks you may have never thought possible. All this, while adding the beautiful warmth and space of the tube devices it emulates. No other plug-in offers this kind of dedicated environment. Also available as a standalone mastering suite (T-RackS 24).

FEATURES

4 sound processors: with its great sounding, analog-modeled EQ Compressor, Limiter and the Soft-clipping output stage, T-RackS includes everything you need to create final masters. And not only does it sound analog, it acts analog! T-RackS is so easy to use because its interface resembles a typical rack of hardware with analog controls.

Equalizer

- State-of-the-art six band parametric EQ modeled on classic top quality analog gear
- 4th order parametric, High and Low pass filters
- Parametric Low-shelving control
- Parametric Lo-mid control with Low or High "Q"
- Parametric Hi-mid control with Low or High "Q"
- Parametric High-shelving control

Limiter

- Multiband master stereo limiter
- Ultra fast peak detection
- Overload for accurate control of limiting and saturation
- Soft clipping feature for warm analog emulation
- Comprehensive gain reduction VU
- Very accurate peak LED indicators at the outputs

Compressor

- Classic tube stereo compressor/ leveler designed for mastering
- Vintage-style compression unit
- Classic warm "in-your-face" sound of old tube mastering levelers
- Very-soft knee compression with no threshold point
- Stereo-image width control
- Mastering-oriented variable time constants
- Comprehensive and accurate gain-reduction VU
- Ratio control

Soft-Clipping Stage

- Astoundingly musical and smooth performance
- Variable clipping shape (from hard clip to ultra-soft) for complete control and precision
- Easily increase gain with no compression or limiting artifacts
- Create warm, saturated mastering effects

Professional Metering and Monitoring

- Responsive analog VU's
- Ultra-fast and accurate pop-up digital peak meter with hardware-like performance
- x10 switch on the peak meter makes very easy to evaluate the level in the "critical zone"
- "Over" LED precisely shows when the master has "overs"
- Mono, stereo and difference monitoring to check the master for mono-compatibility and stereo image coherence, like in high-end consoles

Additional Features

- High quality real time preview and processing
- Supports 24-bit AIFF, Wav and SDII files for importing and exporting.
- High Order 32-bit internal resolution: with 32-Bit floating point resolution, T-RackS uses the best possible method to physically model analog components, resulting in an amazingly real tube sound. Only without the noise!
- 32-bit floating point internal processing for amazing audio resolution and true analog simulation
- Analog modeled animated interface
- Time-line markers to facilitate the immediate recall/play of mixes
- Control snapshots to recall entire set-ups with just one click
- Graphical built-in help, extensive manual and tutorial
- Presets included
- Drag-and-drop text parameter list can easily change the default architecture of T-RackS to achieve the finest special settings
- Compatible with all popular audio cards
- 8 amazing "vintage" skins
- Fade-in/fade out, loop and scrollable preview

IK MULTIMEDIA

GROOVEMAKER



Loop Remixing Software

GrooveMaker is revolutionary music software for creating hypnotic, non-stop electronic dance tracks, in real-time, with professional results. With GrooveMaker you have access to a completely new way of making music with loops. And "random mix generator" technology lets you automatically remix the included loops in virtually limitless ways.

Everything you do in GrooveMaker is done in real-time, without ever having to stop the music. It's like working with a real instrument! You can add, remove, and replace single or multiple loops as you listen, so there's no stopping your creative flow. Remixing is a breeze and a totally new experience. This make the composition as easy as drag and drop always with immediate editing possibilities. It puts you at the controls of the music. It's your mix, and it all happens in real-time. Just pick a drum groove. Layer on some loops. Change the tempo while the music never stops. Then save it all as a new remix!

Import and match any audio file instantaneously (including WAV, AIFF, MP3) and export your mix using many different audio formats ranging from hi-quality CD to internet publishing, including MP3!

GrooveMaker comes with hundreds of professional drum grooves, synth pads, sound effects and ambient loops that keep the music going all night. Experiment with million possible groove combinations in every style. Whatever style turns you on (Techno, Jungle, Progressive, Dub, Acid, Ambient, Eurohouse, Drum 'n' Bass, etc.) it's all possible with thanks to the huge range of add-on professional loop CDs available.

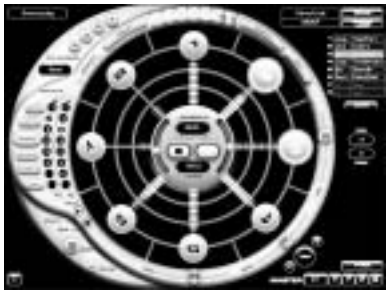
Virtual Synth Arpeggiator

A virtual synth arpeggiator is included to give your performance that rave impact! With 128 arpeggio presets and 150 expandable synth sounds and "smart note" indicators, it's the easyiest way to play along GrooveMaker grooves. Up to 99 arpeggios can be marked and recalled on-the-fly with a single click.

V-MIX

GrooveMaker also contains a virtual DJ offering the quickest way to produce not only a groove but a complete remix album with just one click!
V-MIX instantly assembles 4 random mixes with a simple click -that's it! Every time you click, another random song set is generated. Get hours and hours of instant remixes and never hear the same thing twice. When you hear something you particularly like, save and edit it in the GrooveMaker - or just let it groove all night long.

GrooveMaker



GrooveMaker makes it simple to create instant high-quality dance music. You can play, mix and randomize your own grooves on the fly, with instantaneous control over 8 stereo loop tracks. Manipulate the instrumental loops in real-time! Separate volume, pan, solo and mute controls are at your disposal, giving you an intuitive environment to create stunning grooves.

You can even add your own samples! Shift the tempos (+/-40%) just like a DJ spinning live. With the randomization feature, you'll discover endless loop combinations and keep your sound fresh. Every combination you hear can be saved on the fly, or marked for instant recall.

Assemble your marked grooves in the GrooveSequencer to create full-length remixes! Compose visually thanks to the visual pattern-based composition system integrated within GrooveMaker. This make the composition as easy as drag and drop always with immediate editing possibilities.



LoopMaker

With LoopMaker you can load and resync virtually any audio file (including WAV, AIFF, MP3, QDesign, µLaw, aLaw) with full control over duration, speed, loop-in/loop-out point and offset.

Everything is made in real time and modification are immediately applied while the music is playing in order to immediately resync any loop. It's even easier importing loop from sampling CDs! Make professional remixes in a snap or exchange audio material within different library styles to produce astonishing hybrid mixes adding your personal touch. An unlimited number of loops can now be imported within each of GrooveMaker song putting no limits to your creativity. You can also record loops directly into your computer's audio input using your microphone or any other external devices. Recorded loops can be manipulated the same way of imported loops having them immediately playing within your groove session.

B&H PHOTO - VIDEO - PRO AUDIO

GROOVEMAKER

TRANCE GRID
Trance Loop Library



Trance Grid pushes serious groovemaking to the limit. 2 mega-songs, of more than 120 loops each and with separate loops elements (BDs, snares and hi-hats), are able to produce a zillion numbers of trance grooves with the most complex and intricate rhythmic joints. The right fuel for your lead track.

STREET VOICES
Hip-Hop/Rap/R'N'B/Soul Loop Library



Maximum vocals, hip-hop and rap. An explosive collection dedicated to the voices of the street. Groove to the interaction of extremes, from the ultimate urban styles to the smoothest R'n'B. These loud and luscious sounds combine in an explosion of 250 key and tempo-tuned, mixable loops, pre-grouped into 4 songs.

HOUSE PARTY
House Loop Library



Pure house of the latest trend that will shake your speakers. Dynamite sounds and exploding drum loops to explore the latest dance floor contamination.Over 250 loops, pre-grouped in 4 songs. Basses, BDs, Effects, Loops, Synth Lines, Pads and Percussion all made with massive state-of-the-art gear. BPM range 120-127.

AXE'
Brazilian Percussion Loop Library



Capture the allure and mystique of Brazilian rhythms with this special library of all the "batucada" instruments, mixable in millions of complex, pure carioca rhythmic textures. Over 250 loops, totally recombinable. 10 percussion banks. BPM range 120.

TEKNOSTORM
Techno Loop Library



The raw energy of early techno, the monobeats of yore with a touch of eerie cyberfunk. It will hypnotize you. Over 250 loops, pre-grouped in 4 songs. Basses, BDs, Effects, Loops, Synth Lines, Pads and Percussion, all made with massive state-of-the-art gear. BPM range 140-180.

DRUMBASSTIC
Drum 'N' Bass Loop Library



Nocturnal and soulful psycho-trance, enormous environments and irresistible tribal grooves. Incredible collection of radical soundscapes. Over 250 loops, pre-grouped in 4 songs. Basses, BDs, Effects, Loops, Synth Lines, Pads and Percussion, made with state-of-the-art gear. BPM range 70-180.

CONTAMINATED
Ambient/Dub Loop Library



Metal dub, cool and aggressive uptempo beats. Features weird and chemically altered sounds. It's an enormous collection of phreaked out bass lines and alien noises. Over 250 loops, pre-grouped in 4 songs. Basses, BDs, Effects, Loops, Synth Lines, Pads and Percussion, all made with massive state-of-the-art gear. BPM range 85-135.

EARTH BEAT
Percussion Loops



Percussive loops from around the world. African, asian and latin percussions tempo-matched and fully recombinable.

STUN (Ilio Entertainments)
Hardcore/Techno Loop Library



A collection of dangerous, in-your-face technophobia. Tons of noise, super fast beats and other assorted chaos from world's sound best name ILIO Entertainments. Over 250 loops, pre-grouped in 4 songs. Basses, BDs, Effects, Loops, Synth Lines, Pads and Percussion, made with massive state-of-the-art gear. BPM range 80-150.

IK MULTIMEDIA

SAMPLETANK 2



Professional Plug-in Sound Module

A professional software sound module plug-in that integrates seamlessly into your favorite VST, RTAS or MAS compatible host application, SampleTank 2 combines a powerful sample-based synth engine with hundreds of top-class multi-sampled instruments. SampleTank 2 offers near-instant access to entire instruments each with up to four realtime effects and four sound shaping tools specifically tailored to enhance each instrument. Each instance of the plug-in is capable of producing 16 multitimbral parts and and multiple instances of the plug-in can be loaded into your sequencer. Additional features include the ability to convert AKAI sound libraries as well as access to an ever growing range of Sampletank sound libraries. A unique search engine ensures that you'll find the sounds you're looking for quickly and easily while the RAM doubling capability allows you to load twice as many samples into memory than other hardware or software based samplers. Sampletank 2 is available in two versions: SampleTank 2 XL with more than 650 top-quality sounds on six CDs and SampleTank 2 L with 200 sounds on 2 CDs.

FEATURES

- 32-bit floating point processing with an ultra-accurate resampling algorithm make SampleTank 2 the best sounding sample-based module ever.
- Perfect integration within your sequencing environment with sample accurate timing, one-click project recall and digital track bounce—no hardware or driver problems and need to have an extra application running simultaneously.
- Uncomparable sound organization and super-fast sound loading lets you surf hundreds of high-quality sounds and play them in a fraction of the usual time. The intuitive layout of the sounds in categorized, alphabetical folders (which reflects the folder organization you have on your HD) lets you get the sound you want instantly.
- Each module (instance of the plug-in) allows you to access up to 16 multitimbral parts with up to 128 note polyphony and 4 stereo (8 mono) outputs
- Compatible with the AKAI S1000/S3000 sound library—the world's biggest. Using the included converter with full sound-program conversion and parameters link SampleTank 2 can easily import those sounds.
- Multiple SampleTank 2 modules can be instantiated per session, with support for a total of 64 separate outputs, limited only to your host software application's capacity and available CPU resources.
- 4 specifically tailored dynamic sound parameters and 4 effects with up to 5 parameters per effect, are automatically loaded with each instrument.
- Multi-effects include: Compressor, Equalizer, Reverb, Ambience, Reverb Delay, Delay, Filter, Wah-Wah, Chorus, Flanger, Autopan, Tremolo, Rotary Speaker, Lo-Fi, Distortion, Phonograph, Slicer, AM and FM modulation.

4 years of work and hundreds of superb musicians and instruments were required to produce the astonishing sounds included in SampleTank 2. Recordings have been made around the world: russian strings, perfectly tuned german grand-pianos, Italian cathedral organs, tons of vintage electric instruments including electric pianos and organs, rare guitars, basses and drums. Accurately edited, programmed and digitally crafted in thousands of studio hours to obtain a realism and playability you thought it was not possible. Plus, a variety you won't find in any comparable product.

RAM Doubling

- SampleTank 2 uses an exclusive method called 2Pack, which allows the playing of huge sample banks using only half the amount of RAM normally required. The result is huge sound capacity. You will hear more realistic sounds, programmed using more notes and full natural decays.

Sounds

- They include an extra-large, ready to play collection of sounds, comparable to top-class sound libraries. Superb acoustic grand pianos, guitars, basses, strings, vocals, orchestral sounds, brass sections, woods, winds, drums, percussions and loops, etc. - all made with the finest samples and accurately programmed.

Sequencer Compatibility

- Cubase VST (PC version 3.7.2 or Mac version 4.1.1 or later)
- Logic Audio (PC and Mac version 4.5 or later)
- MOTU Digital Performer 3.01 or later with MOTU Audio System 2.33 or later
- Cakewalk SONAR (using VST-DX Adapter from) plus any other sequencer (including n-Track, Orion, FruityLoops) compatible with VST/MAS instruments

SAMPLETANK 2

Conveniences

- Many times after playing a riff, you stop and look for another sound. Maybe similar to the one you started with, but different. You might even come back to the original in the end. But trying out new sounds is something that you do every single song, every single track. Many times. With Sampletank's Search Function, you can type in "lead" and a list appears before you of all lead sounds in the module. Synths, guitars, orchestral instruments. Every lead sound you have. Cruise through different palettes with one click.
- Each instrument in Sampletank automatically brings the tools you will need for it to you. You don't have to go through endless menus trying to find a way to shape your sound. Sampletank knows what you will need and intelligently gathers them for you. A synth lead has a certain set of tools needed. Acoustic orchestras need other tools. It's not just easier, it's faster.

SampleTank 2 XL Include:

- 22 acoustic and electric basses
- 25 brasses including sax, trombon, trumpet, brass solo and section
- 53 drum kits, acoustic and electronic
- 39 acoustic and electric guitar
- 10 ethnic instruments
- 24 loop banks of D'n'B, House, Techno, Trance in construction kits with more than 500 loops
- 20 full orchestra sounds including string sections
- 15 acoustic and electric organs including B3, church, cathedral
- 29 percussion kits including Brazilian, Latin, African, Asian
- 20 pianos including concert grand, electric, electronic
- 148 synth sounds including bass, drums, lead, pad, percussions
- 31 strings including violins, violas, cellos, basses solo and sections
- 13 vocals including male and female choirs
- 4 woodwinds including bassoon, clarinet, flute, oboe
- Expand your sound arsenal with an additional 15 Sampletank native libraries

– Intuitive sound organization and super-fast sound loading

– Surf hundreds of high-quality sounds, categorized by user defined folders, with a few mouse clicks and play them in a fraction of the usual time.

Search Function

Simply type in "lead" and a list appears before you of all lead sounds in the module, synths, guitars, orchestral instruments – every lead sound you have

The name of the current instrument along with a description of the four associated realtime control parameters

The four most important parameters for each instrument are automatically loaded to these controller knobs when an instrument is loaded. Each knob is controllable in realtime and automatable via MIDI.



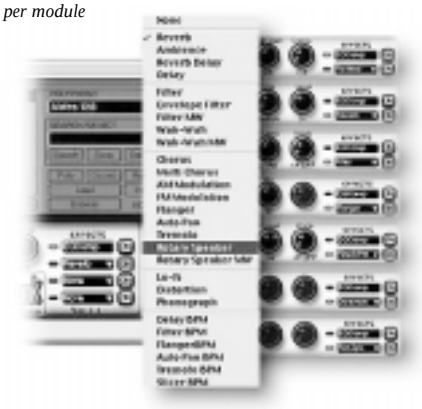
16 part multitimbrality and 4 stereo (8 mono) outputs per module

– 4 realtime effects slots with a choice of 28 high quality DSP effects including rich reverbs and beat matched delays, filter, wah-wah, chorus, AM and FM modulation, flanger, autopan, tremolo, rotary speaker, lo-fi, distortion, phonograph and slicer.

– Up to five realtime parameters are provided for each effect. These parameters can also be controlled in realtime and automated via MIDI

– Compression and EQ are permanently assigned to the first slot.

– Instruments come ready to play with inspiring realtime effects tailored to that particular instrument – just like a hardware sound module.



STRETCH Virtual Synth Engine

The advanced STRETCH Synth Engine allows you to instantly sync any loop to your sessions tempo or trigger complex harmonies from a single sampled vocal phrase or horn line, and each note will play in time.

- STRETCH is a new technology that uses a technique called Time Resynthesis to play adjacent samples in tune without changing the tempo. This provides full control over the tempo and pitch of the samples.
- Traditional samplers or sample-based sound modules produce adjacent notes by "resampling" the sample contained in the key note. To stay in tune, this old-style technology plays samples faster or slower, which is the reason many natural instruments loose their realism when played by a sampler or a sample-based sound module.
- Plus STRETCH expands sonic capabilities with full-control of an instrument's basic frequency distribution to tailor tone and push the limits of creative sound designing to an entire new level.
- Sync any loop tempo to a sequencer's tempo with one click. Full control over the loop's tempo and pitch make it easy to create the ultimate groove.
- Create rich harmonic parts and soaring choirs in a snap from a single looped phrase or drop in a sample of your singer and instantly turn it into a choir part.

SAMPLETANK 2

Plug-in Sound Modules

The range of SampleTank 2 sound modules combines a stadium full of world-class samples from the finest soundware houses, with SampleTank LE software for easy, affordable access to the world's most powerful virtual sound modules. A fully working SampleTank with 4-part multitimbral (instead of 16), 4 effects per voice selectable among 20 DSP effects and single stereo output, SampleTank LE works both as plug-in for VST/MAS/RTAS or SampleTank add-on sounds.



Loop Soup!

Multisamples

The ultimate dance loop library, this double CD set contains the loops from 'Skip To My Loops' by Norman Cook, 'Remix!' by Megabass, 'Kleptomania!' by Coldcut, and 'Pascal Gabriel's Dance Samples' plus all the loops from Now 1 and Now 2 by loads of the other AMG producers. All loops are tempo-grouped for easy sampling. Ideal set for anyone getting into sampling for the first time.



Terminal Head

Dance Sample and Loops

The world's first futurefunk sample CD. There are no rules. There are no BPMs. There are no keys. There is only the need for PuSHing the boundaries of people's perception into ways of creating music for tomorrow that fully embraces all technologies and forges solid bonds with all that is known to funk and groove. Totally unique, this CD has material born out of some very serious experimentation.

Brit Horns

Multisamples



Chromatically sampled Horn Sections and solo instruments from the legendary Thoms, Barnacle & Brooks brass section! This horn CD features trumpet, sax, and trombone - full sections in different combinations plus solos - all performed by three of England's finest horn players.

Earth Beat

Ethnic Percussion Loops



Percussive loops from around the world. African, Asian and Latin percussions tempo-matched and fully recombinable.

Trance Grid

Dance Loops



A very energetic trance loops library in construction kits with separate rhythmic elements which can be played as single instruments, including Basses, BDs, Snares, Hi-Hats, Effects, Synth lines and many more, or combined together. With 8 different sound and loop banks (over 250MB of samples), it generates zillion of trance grooves with the most intricate rhythmic combinations.

Tony Mason Drum Loops



Specially recorded 'black' grooves with attitude recorded in the UK and Caribbean. Tony Mason has produced a CD full of freshly recorded dance grooves with a classic feel. Many of the loops are presented in a number of variations for more flexibility when sampling. The whole CD was made using classic analog processing and recording equipment.

Features all the grooves sampled for easy loading plus 'chopped up' versions that come with associated MIDI Files and allow easy editing of tempos and patterns, switching sounds, editing beats, or removing elements. In all there are 382 associated MIDI files. (A MIDI File disk is available in DOS or Mac format free with purchase.)

AXE' *Brazilian Percussion Loops*



With over 550 MB of sounds and loops, AXE' captures the allure and mystique of typical Brazilian rhythms and sound, containing all the instruments of the Batucada set such as Surdo, Tamborim, Pandeiru, Berimbau, Repinique, Guiro, Agogo and many others.

AXE' is available in construction kits, a powerful tool to combine all sounds and loops, share the same tempo among them, and create in real time thousands of different grooves. Recycled files are included for maximum control and easy editing, giving you the ability to change tempo without pitch variation, or to quantize the resulting grooves, achieving a huge amount of different possibilities for every requirement.

SAMPLETANK 2

Masterbits Vintage Voltage ST



A great collection of analog synth sounds, 180 ready to play ST sound sets including Leads, Basses, Pads, SFX etc. taken from famous vintage synthesizers. Includes accurately sampled sounds from Roland Jupiter 8, Roland MKS 70, Planet S, Oberheim Xpander, Studio Electronics SE 1, Roland Juno 60 and 106, Oscar, Deep Base Nine, Moog Prodigy, Yamaha CS 15, Teisco 60F, Roland SH 101, Wasp, Roland MKS 80, Oberheim Matrix 12, Korg Polysix, Monopoly, Korg Delta, Kawai SX 350.

Masterbits MelloZone ST



This collection of Mellotron sounds, is based on over 800 carefully selected single recordings in half tone steps, produced by Mellotron specialist Klaus Hoffmann-Hook. Just like the original, the 26 "FRAMES" were converted into ready to play Sampletank sets. Each frame set contains 23/35 multi samples, which means one sample per loop to completely avoid unwanted side effects and to secure an absolutely authentic Mellotron sound. The center are the legendary choir sounds with 10 ST-Instruments and the well known string sounds that have reappeared as hits in the charts. Includes classics like various flute & brass sound sets as well.



Guitar Collection

A fantastic selection of both acoustic & electric guitars including nylon & steel string acoustics, 12 string and Dobros as well as Pedal Steel, Ebowed Guitars, Clean and Distorted Electric Guitars, Overdriven & Rotary Guitars plus many others. Perfect for a wide variety of musical styles from Rock to Jazz to Country & Blues, the Guitar Collection is a must for anyone looking to play realistic multisampled guitar tones!



Piano Collection

A fine selection of Sonic Reality's acoustic and electric pianos featuring both Yamaha and Steinway concert grands as well as uprights and baby grands too.

With acoustic pianos ranging from 32 to over 90 megs in size, the Piano Collection's value is hard to beat. In addition, there are extremely realistic classic Rhodes and Wurly EPs as well as FM pianos, Clav D6 and more.

Phil Gould on Drums



This massive collection of fresh beats includes a vast range of feels and styles - from recordings made as far back as 1984 right up to the present day! Ex-Level 42 drummer Phil Gould includes hundreds of specially recorded loops and excellent D&B loops. Loops range from 70 bpm right up to 165 bpm plus there's a selection of high quality single hits too.

Symphony Strings



Ideal for classical music and film soundtracks, as well as pop, rock, jazz and dance. Its organic quality makes it possible to add depth and realism for MIDI orchestration. Since it is made up of playable instruments only (no phrases or loops) it is a library that can be used over and over in unlimited musical situations. A Euphonix board and special outboard gear along with Neuman mics were used to capture the silky sound of strings in a medium ambient room. The engineers worked with artists such as David Bowie and Elton John to name a few.



Sample Messiah

This rapturous collection of dance samples in the classic format has everything you need for House, Electro, Detroit, Experimental, Techno, Trance and many more dance genres! DJ Harsh, The Notzel Brothers, Ars Larsen & Lars Boske pool their vast sample resources to create the ultimate, modern Euro dance sample CD. Capturing the essence of the classic dance sample collections with the sounds of today - this collection is a unique resource for anyone making modern music.

Special Collection! Sound Modules Series



A complete collection of 9 titles from the best soundware houses at a killer price. Includes AMG's Brit Horns and Loop Soup, IK's AXE' and Trance Grid, Masterbits Mellozone and Vintage Voltage, Sonic Reality Piano Collection, Symphony Strings, and OmniSynth

IK MULTIMEDIA

SONIC SYNTH by SONIC REALITY



Designed for a wide variety of musicians from beginners to professional keyboard players, Sonic Synth is a workstation-like sound module packed with one of the most diverse collections of high-quality playable instruments available anywhere, at any price. A unique tool, Sonic Synth offers a huge range of immediate playable instruments, suitable for many musical styles such as rock, pop, jazz, classical, dance, R&B, hip hop, country, film soundtracks and more.

Sonic Synth features stunning acoustic and electric pianos and other staple sounds for songwriting and producing. For moody pads, vocal textures and orchestral soundscapes Sonic Synth also provides a vast selection of inspirational content.

Whether you mainly play pianos and organs or do full orchestrations, funky guitar, bass and drum tracks or ambient music with the need for a touch of the ethereal, Sonic Synth has a great selection of inspiring sound content to choose from FAST!

- Sonic Synth excels in depth and realism. These virtual instruments were designed to easily sit in the mix without the need for much external processing. Very convincing virtual band tracks can be made using the deep basses, punchy drums, liquid guitars and classic keyboards with loads of character.
- A patch in Sonic Synth could have it's origins from Three Dog Night's Oberheim 4 voice, Larry Carlton's Arp Quadra or Joni Mitchell's Mellotron Mk 5, the Beach Boy's B3 organ (with a white Leslie 122 cabinet), Pink Floyd's Prophet 10 with patches that were used for The Wall, Mellotrons and Chamberlins that belong to Tom Waits, Neil Finn/Crowded House, Tom Petty and more.
- The combination of great studio rooms, experienced engineers, top notch recording gear and select choice instruments makes Sonic Synth a unique product packed with highly usable sounds.
- And, more importantly, all Sonic Synth ST content was programmed in native Sampletank format and designed to take advantage of all of Sampletank's special features from scratch, including internal effects programming, knob assignments, search functions and the loads of great internal synth programming and processing. This gives the Sampletank user a lot more expression over just importing basic multi-samples from AKAI format CD ROMS.
- Sonic Synth's sound set is over 2GBs which allows for less compromises in workstation-type sounds such as pianos, strings, organs, guitars and drums.
- Sonic Synths includes 532 sounds organized by 13 categories in 3 CDs:
  - SR Pianos 22
  - SR Guitars 34
  - SR Chromatic 22
  - SR Strings 20
  - SR Organs 34
  - SR Voices 5
  - SR Bass 21
  - SR Drums 50
  - SR Synths Pads 119
  - SR Ethnic 35
  - SR Brass-Winds 40
  - SR Synths Leads & Basses 54
  - SR Synths Misc Keys & FX 76



Acoustic Drum

Sonic Reality's I-Map acoustic drum kits were recorded in some of LA's best studios. Featuring multisampled hits from Ludwig, Gretch, Yamaha, DW and Pearl Kits, this collection will add the depth and realism of live studio drums to your sequences!



Bass Collection

Sonic Reality's Bass Collection features Fender P and Jazz Basses, Rickenbacker 4001 Bass, Hofner Beatle Bass, Lakeland Fretless Bass, Upright and Acoustic Basses and more. All multisampled playable instruments with finger, picked, harmonics, slaps, pulls, taps, slides and more!

Coming in 3rd Quarter 2003. . .

The eXXential Series

The eXXential Series is a new collection of SampleTank 2 sounds from IK Multimedia. These top-quality libraries offer instrument-dedicated series, which enter into the deep and distinctive characteristics of each instrument with a huge collection of sounds.

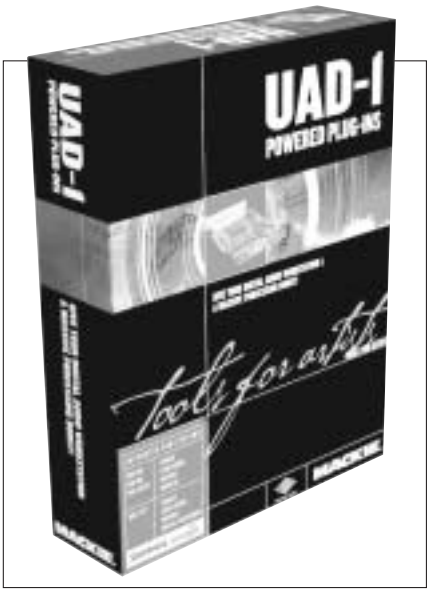
eXXential Drums– A powerful kit of both vintage and modern drums recorded at 24-bit/96kHz with a pure, unprocessed amazing ambience

eXXential Pianos– A collection of chromatically sampled Steinway and Fender Rhodes offer incredible realism and playability

eXXential Bass– One of the best and most complete collections of electric bass around. Sampled at 24-bit/96kHz with tons of variations

eXXential Strings– A pristine collection of 16 violins, 8 violas, 4 cellos and 4 basses in ensemble and solo, with multiple variations including pizicato, staccato, vibrato chromatically sampled. Performed with passion by the Moscow Session Orchestra.

UAD-1 POWERED PLUG-INS



Powered Plug-Ins For Your Native DAW

Powered Plug-Ins are a suite of professional, DSP-intensive plug-ins "powered" by the UAD-1 — a revolutionary PCI card that brings ultra-high computational processing to host-based systems utilizing a single proprietary high-speed DSP processor. The end result is a quantum leap in performance giving native workstation users the experience of using professional quality plug-ins with multiple instances unencumbered by the limitations of the host processor. The Powered Plug-Ins bundle feature Universal Audio's high quality stereo reverbs, Dreamverb and RealVerb Pro; the CS-1 Channel Strip which includes the EX-1 EQ/Compressor, DM-1 modulated delay module, and RS-1 Room Simulator as well as dead-on emulations of the legendary 1176LN and LA-2A Vintage Compressors and Pultec EQP-1A EQ. Also included is Nigel – a powerful amp/cabinet modeler and multi-effect guitar processor.

FEATURES

- The UAD-1 Powered Plug-Ins package includes the UAD-1 DSP card and a comprehensive package of ultra high quality plug-ins, including RealVerb Pro, the 1176LN and Teletronix LA-2A Vintage Compressors, the Nigel Guitar Processor, and the CS-1 Channel Strip.
- Compatible with your native DAW system using VST, MAS, Direct X architectures – combine the advantages of Powered Plug-ins with using the dedicated UAD-1 DSP card while continuing to use your favorite native plug-ins.
- Supports up to 32-bit / 192kHz audio
- Running intensive processors such as reverb and compression on the UAD-1 provides your host processor with additional capacity for more tracks and more instances of your favorite native plug-ins
- UltraDither (proprietary hardware dithering process) provides maximum signal quality at every step with no impact on processing capacity
- Artifact free smoothing allows you to adjust parameters in real-time with no zipper noise, pops, or clicks.
- Every parameter of every plug-in can be automated.
- Multi-card support allows you to use multiple UAD-1 cards in your system.
- 32-bit floating point processor matches the data format of the host application for a distortion free, ultra high-resolution signal path with maximum efficiency
- Non-partitioned DSP chip gives each plug-in the total capacity of the system when needed, giving the plug-ins a quality and complexity not available on native systems.

CS-1 Channel Strip

- Provides all the tools needed for recording and mixing in one simple plug-in or, you can load each plug-in separately
- Unique smoothing algorithm for all controls and DSP conserving bypass switches



Plug-in Modules

EX-1 EQ/Compressor:

- Five parametric bands with individual bypass switch, Hi/Lo shelving, peak/notch or high/low pass settings.
- Ultra-smooth compression with variable ratio, threshold, attack and release controls as well as gain reduction metering.

- **DM-1 Delay Modulator (mono or stereo:**
- Up to 300ms maximum delay per channel
- Selectable modulation waveforms with adjustable phase, including quadrature, in-phase and out of phase
- Mode selector providing popular forms of Chorus, Flanging, and Echo in one plug-in

- **DM-1L:** Same as DM-1 plus up to 2400ms delay and link button for enhanced control

- **RS-1 Reflection Engine:**
- Multiple room shapes
- Adjustable room size from 1 to 99 meters
- Forward and reverse gates
- Echo, ping-pong, and multi-tap delay

UAD-1 POWERED PLUG-INS

The Teletronix LA-2A Leveling Amplifier



- A digital copy of the original opto-electrical attenuator circuit responsible for the incredibly rich and magic sound found on countless classic recordings
- Upholds all the classic, analog characteristics of the original with unprecedented clarity and sound quality – including how each and every component interacts under a full range of studio conditions
- Identical look, controls, and operation of its analog cousin
- Lag-free, distortion-free optical attenuator emulation
- Frequency-dependent compression behavior
- Input gain and peak reduction adjustment knobs
- Limit/Compress function switch
- +4, -10, or gain reduction metering
- Mono or stereo operation

The 1176LN Limiting Amplifier



- The classic analog processor originally designed and built during the 1960's is now available as a plug-in for your DAW
- The first processor to use a field effect transistor (FET) as a voltage controlled variable resistor – the key to the processor's unique "warm" character
- Ultra-fast attack time
- Supports the popular "All Buttons Mode", just like the original
- Input & output gain adjustment knobs with adjustable attack and release settings
- Adjustable program-dependent compression behavior
- Custom output transformer emulation design
- Mono or stereo operation
- Flexible VU metering modes
- Detailed, precise screen shot provides a realistic analog look and feel

Legendary Pultec EQ



The Pultec EQP-1A Program Equalizer plug-in is a faithful electronic reproduction of the classic hardware equalizer. The Pultec EQ has long been a choice of recording and mastering engineers for its ability to bring out individual frequency ranges without significantly altering other frequencies. In addition, the Pultec is one of those magical pieces of gear that makes audio sound better - just by passing through it. The sophisticated modeling technology used in the Pultec EQP-1A plug-in captures both of these key characteristics.

- Legendary Pultec Program EQ
- Unique simultaneous boost and cut
- Smooth, sweet top end
- Dial in dangerous amounts of boost with incredibly musical results
- Modeled after a highly prized "golden" Pultec
- Models accurate to within unit-to-unit variation

Nigel— Guitar Effects

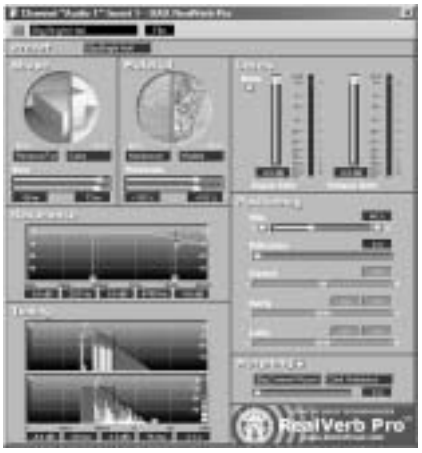
- The latest generation of guitar processing technology integrated into a complete multi-effects plug-in
- Uses an exclusive component modeling technology and intuitive design to deliver a complete pallet of guitar tones along with most every effect a guitar player might need, all with minimal latency and no load on your host computer's CPU



- The Preflex advanced guitar amp modeling technology delivers a wide range of highly playable classic amp tones from the "Clean & Warm" California tube sound to more metal soaked "British" tones as well as a number of original timbres not possible on any other guitar system
- PreFlex modeling also offers continuously variable morphing between any two amp presets bringing creative guitar processing to an even higher level
- Gate/Compressor for noise and dynamics control
- Phasor capable of modern and classic sounds such as those produced by the Mutron Bi-Phase, Small Stone and MXR series of phasers
- Mod Filter capable of wah, auto-wah, and envelope follower effects, modeled after the Mutron III and other popular filters
- Tremolo with Classic, Shimmer, VariTrem, and Fade modes
- Fade-in for swells and reverse tape effects
- Modulated Delay capable of chorus, flange and vibrato
- Echo Delay up to 1200ms
- Unlimited presets for total recall
- 24-bit, 96kHz support
- Each of Nigel's component plug-ins can be loaded individually

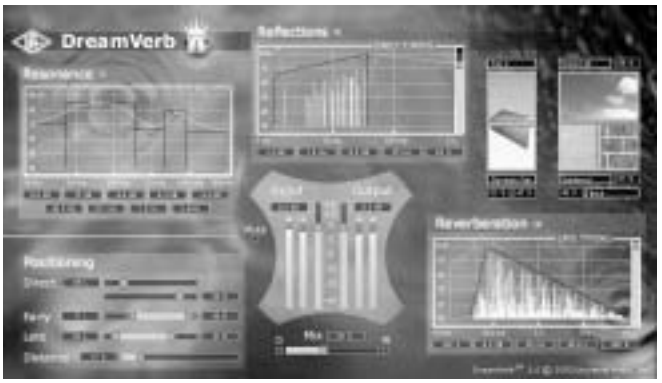
UAD-1 POWERED PLUG-INS

RealVerb Pro



- Unique set of algorithms let you design the perfect room just as you hear it with graphic menu of room shapes, and adjustable settings for wall material as well.
- Provides the trademark, distortion-free, smoothing diffusion control and ultra-long reverb tail
- Morph between different presets
- Adjust relative thickness of materials
- Blend between room shapes and sizes in realtime
- Adjust room size from 1-99 meters
- Run eight instances simultaneously using a single UAD-1 card
- Control intensity and timing of early reflections and late-field reverberation
- Comprehensive interface for in-depth parameters editing

Dreamverb – Flagship Stereo Reverb



- Drawing on the flexibility of RealVerb Pro, its intuitive and powerful interface allows you to create an acoustic space from an extensive list of different materials and room shapes. These acoustic spaces can be customized to great detail by blending or "morphing" different room shapes and surface materials, adjusting the effect of the room materials on the sound space, and even varying the density of the air to simulate different ambient situations.
- Dreamverb also features a flexible 5-band active EQ and unique level ramping for the early and late reflections for sounds ranging from ultra-realistic dynamic room simulations to lush reverb effects.

UAD version 3.0 supports the use of up to four UAD-1 cards inside a single PC, so you can purchase an additional card and add I/O capabilities at the same time.

Cambridge EQ (Available Separately)



The Cambridge EQ features five bands of fully parametric EQ, as well as high and low cut filters with a wide variety of filter types and curves, and switchable shelving filters for each EQ band.

- Complex lattice filters and a special algorithm deliver a warm analog sound without oversampling.
- A/B function allows for quick comparison of two different settings.
- A graphic display of the EQ curve provides “edit handles” for click and drag control of the EQ parameters, plus editable text displays for parameter values.

NEW! UAD-8 I/O

UAD-1 Card with ADAT I/O

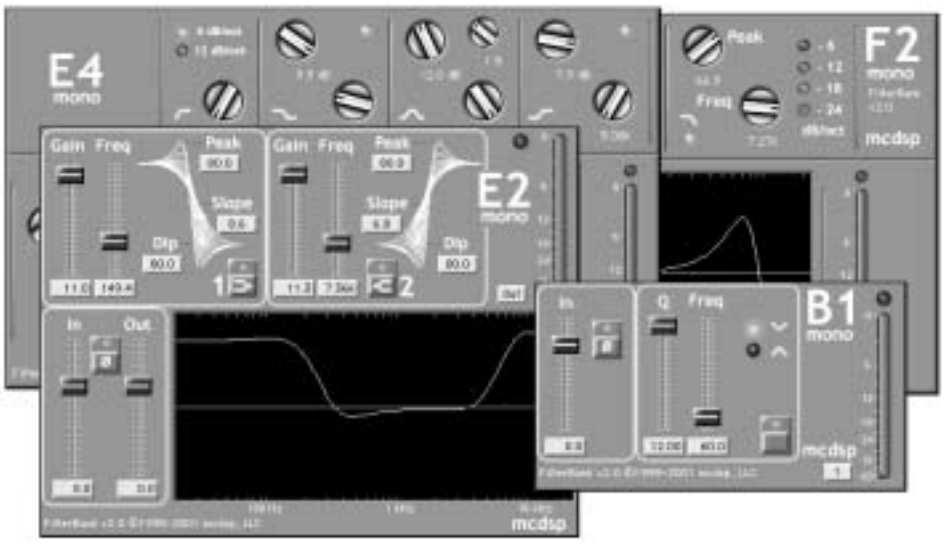
Combining the dedicated DSP capabilities of the UAD-1 card with an ADAT optical I/O, the UAD-8 allows you to interface directly with external devices which support ADAT I/O without the need to purchase a separate PCI card for your PC.

- Supports sample rates ranging from 44.1kHz to 192kHz.
- A stand-alone host application is included which allows you to access all the Universal Audio plug-ins without using a third party VST host application. This allows you to turn an ordinary PC into a standalone digital effects processor, capable of 192kHz sampling rates and able to run ultra-low latency effects for any digital interface using the ADAT optical interface.

McDSP

FilterBank

High-End EQ/Filter Plug-in



FilterBank is high-end EQ/Filter plug-in designed to emulate the sound of vintage and modern equalizers/ filters. But FilterBank isn't just another great sounding EQ - it's every EQ ever made in one plug-in. FilterBank does this through its unique design philosophy: give the user maximum control. All critical equalizer and filter design parameters are totally user adjustable. FilterBank can emulate any EQ: tube driven or solid-state, vintage or modern, fat or thin. Or create a distinct custom EQ. FilterBank is 20 plug-ins consisting of 2, 4, and 6 band configurations of EQs and filters, in mono and stereo versions.

FEATURES

Total Control Over All Critical EQ and Filter Design Parameters

- High and Low Shelving EQ: Independent control of frequency, gain, peak, slope, and dip. No other EQ gives you so much control and flexibility of shelving equalization.
- Parametric EQ: Frequency control range that covers nearly the entire audible spectrum (20 - 21000 Hz). The parametric bell is up to 5 octaves wide! Gain is +/- 12 dB. Constant-Q technology closely models analog parametric responses.
- Low and High Pass Filters: Adjustable slope and resonant peak greater than 24 dB.
- Band Pass and Band Reject Filters: Frequency control range that covers nearly the entire audible spectrum (20Hz-21kHz). The Q range is 0.2 - 12.0.

Additional Features

- Double precision arithmetic keeps the noise floor at the -138 dB level one would expect from the 24-bit TDM bus.
- Analog saturation modeling is provided in each EQ/filter band to prevent digital clipping.
- Performance: Up to 54 EQ/filter bands per HD dsp. Up to 48 EQ/filter bands per MIX dsp. FilterBank runs on any MIX system dsp type (SRAM, DRAM, no RAM). d|24 systems run up to 24 EQ/filter bands per dsp.
- ProControl, Control 24, and HUI compliant; control automation (PT 4.0 or higher); TDM, AudioSuite, and RTAS versions; supports d|24, MIX, HD, and host-based ProTools systems.
- FilterBank can emulate any EQ: tube driven or solid state, vintage or modern. Or FilterBank can be used to create a distinct custom EQ. And unlike the high-end outboard gear it emulates, FilterBank runs up to 48 bands of EQ/filters per MIX dsp chip, vs. to 4 to 6 bands per hardware box.

P-S-D (Peak-Slope-Dip) Control

P-S-D control allows you to adjust the critical response characteristics of low and high shelving EQ. These parameters replicate the sound of nearly every shelving EQ ever made.

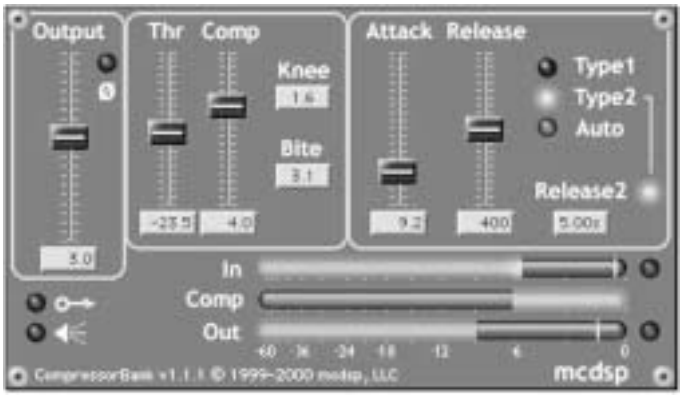
Peak: The peak control adjusts the amount of added punch in the shelved portion of the response. The peak in the shelved portion of the response gives the shelving equalizer additional brightness or emphasis near the shelved band frequency. The effective gain at the shelved band frequency can exceed the range of the gain control when used at its maximum setting. Peak and dip controls are interactive - as the peak control is increased, the overall dip in the shelved response is decreased, and vice versa.

Slope: The slope control adjusts the gradient (slope) of the shelved response. The more gentle the transition between the shelved and non-shelved bands, the smoother the equalizer sounds. Steeper slopes can give the shelving equalizer more definition and clarity.

Dip: Adjusts the amount of warmth added in the non-shelved portion of the response. The dip in the non-shelved response reduces the amplitude of signals whose frequencies are just outside of the shelved portion of the response. Dip and peak controls are interactive

CompressorBank

The Complete Compressor



CompressorBank gives the user complete control of dynamic compression. Common controls such as Output (make-up gain), Threshold, Compression (Ratio), Attack, and Release are provided. Non-standard Knee and Bite controls allow unique and unprecedented articulation of compression characteristics. Multiple peak detection circuit types provide flexibility only previously achieved by owning different compression units. Side chain support, side chain or in-line pre-filtering, and a static/dynamic EQ section (post compression) round out the signal processing capabilities of an already very capable compressor. CompressorBank is 8 plug-Ins: basic compression (CB1), compression with pre-filtering (CB2), and compression with pre-filtering and static/dynamic EQ (CB3), and modeled compression (CB4) - all in mono and stereo versions.

Side Chain Support

Provided in all CompressorBank configurations. Side chain pre-filtering (high/ low pass, and band pass, and parametric EQ) is also available. The pre-filter can also be placed in-line and applied to the compressed signal.

Compression Curve Modeling

In addition to the standard Threshold and Compression (Ratio) controls, the actual shape and response of the compression curve can be adjusted with the Knee and Bite controls. Knee softens the compression curve, creating a smoother response. Bite gives the compressor the ability to allow signal transients to pass uncompressed, while the overall compression response is unchanged. These controls allow the user to emulate responses of their favorite vintage gear.

Processing Specifications:

- Double precision arithmetic for 48-bit processing at side chain pre-filter, compression and active EQ stages
- Analog saturation modeling on each band prevents digital clipping.

Dynamic EQ

Using the same technology as FilterBank (including analog saturation modeling), this static/dynamic parametric EQ section can be applied to the compressor output, making possible effects such as signal enhancement, noise reduction, and harmonic distortion.

Multiple Peak Detection Circuits

Every compressor uses a model to detect and track signal peaks and then apply dynamic compression. Variation of basic models are available in CompressorBank:

– Type1: pure peak detection

– Type2: pure peak detection combined with adaptive release times, plus a secondary release control

– Auto: signal levels are automatically tracked

Compatibility

ProControl, Control 24, and HUI compliant; control automation (PT 4.0 or higher); TDM, AudioSuite, and RTAS versions; supports d|24, MIX, HD, and host-based ProTools systems.

The Classics

McDSP has developed emulations of the Teletronix LA2A, the dbx 165, the Neve 2254E/33609, the UREI 1176LN (blackface, including the '44' setting), and other classic high-end compressors and limiters.

Performance:

Up to 18 mono (or 14 stereo) channels of compression on a single HD dsp. Up to 15 mono (or 12 stereo) channels of compression on a MIX dsp (any type - SRAM, DRAM, no RAM). d|24 systems can run up to 8 mono (or 5 stereo) channels of compression on a single dsp.

MC2000

The Complete Multi-band Compressor

The MC2000 gives the user complete control of multi-band dynamic compression. Common controls such as Output (make-up gain), Threshold, Compression (Ratio), Attack, and Release are provided. Non-standard Knee and Bite controls allow unique and unprecedented articulation of compression characteristics. Multiple peak detection circuit types provide flexibility only previously achieved by owning different compression units. Steep 24 dB/oct crossover filters, taken from the award-winning FilterBank plug-in, minimize signal leakage into adjacent compression bands.

McDSP

ANALOG CHANNEL

The Complete Analog Experience

When the sound of digital won't cut it, Analog Channel transforms mixes into analog heaven. The most sought after high-end tape machines and analog consoles on the planet are an insert away. Analog Channel comes with several settings files based on a variety of tape machines and consoles, including emulations of the most popular playback heads used on the best analog 'reproducers' in the world. Plus, you can create your own distinct custom analog system. Analog Channel includes two devices: the AC1 for emulating analog mixing console channels, and the AC2, which emulates tape and tape machines.

AC1– Analog Console Emulation

Pure digital tracks can suffer from harsh digital clipping. AC1 acts as a 'digital pre-amp' allowing even excessive amounts of gain without digital distortion. A drive control determines where the audio 'sits' in the saturation/compression region, and even the attack and release times of the distortion character are adjustable. And AC1 is the most efficient McDSP plug-in ever - up to 24 mono channels or 16 stereo channels on a single MIX dsp chip. Place it across the entire mix to create a custom analog console.

AC1– Analog Tape Machine Emulation

Includes standard tape deck parameters such as bias, playback speed, and IEC1/2 equalization, plus controls that are well beyond the limits of 'reel-world' devices, like adjustable low frequency roll off and head bump - independent of playback speed. Select from several playback head types, and even control the rate at which tape saturation is disengaged. Modern and vintage tape formulations are available. And instead of frequency sweeps and test tones, the playback head and tape saturation responses are displayed in realtime - updated with every change.

FEATURES

Power and Flexibility

Four different configurations are available - Analog Channel Emulation (mono and stereo), and Analog Tape Emulation (mono and stereo). AC1 - control drive, attack, and release of amplifier characteristics;
AC2 - control every aspect of analog tape 'reproducers', as well as parameters not found in the original devices.

Performance

- **AC1:** Up to 28 mono (17 stereo) channels on a single HD dsp. Up to 24 mono (15 stereo) channels on a single MIX dsp chip.
- **AC2:** Up to 9 mono (4 stereo) on a single HD dsp. Up to 8 mono (4 stereo) on a single MIX dsp chip.
- Analog Channel can run on any MIX system dsp type (SRAM, DRAM, no RAM).

Additional Features

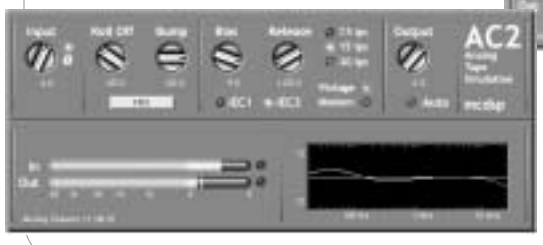
- Double Precision Arithmetic keeps the noise floor at the -138 dB level one would expect from the 24-bit TDM bus.
- Analog Saturation Modeling: at playback head and tape saturation stages in the AC2

Compatibility

ProControl, Control 24, and HUI compliant; control automation (PT 4.0 or higher); TDM, AudioSuite, and RTAS versions;
supports d/24, MIX, HD, and host-based ProTools systems.

The AC1 plug-in allows the user to design channel amplifiers such as those found on 'classic' analog mixing consoles. Parameters can be saved in a preset or as part of a session, and are completely automatable.





Instead of frequency sweeps and test tones, the setup of the AC2 playback head response and tape response are updated with every control change. Time consuming calibration and maintenance are no longer required.

AC1 - ANALOG CONSOLE EMULATION

- Input: +/- 24 dB, with phase control
- Drive: +/- 12 dB
- Adjustable saturation region
- Attack: 0.01 msec to 10 msec
- Release: 10.0 msec to 1000 msec
- Output: +/- 24 dB
- Auto Output Control, L/R Input and Output link
- Simultaneous input/output metering

AC2 - ANALOG TAPE MACHINE EMULATION

- Input: +/- 24 dB, w/ phase control
- Roll Off Frequency: 20 - 100 Hz
- Bass Bump: 0 - 100 % (up to 6 dB maximum)
- Selectable playback head types
- Bias: +/- 12 dB
- Release: 10.0 msec to 1000 msec
- Output: +/- 24 dB
- Vintage and Modern tape formulation modes
- Auto Output Control, L/R Input and Output link
- Simultaneous input/output metering
- Playback head and tape saturation response display

B&H PHOTO - VIDEO - PRO AUDIO

SYNTHESIZER ONE

High-end Software Synth



Synthesizer One combines many sound creation techniques into a single software plug-in including wavetable, FM, AM, Ring modulation, additive and subtractive synthesis methods. External audio can be processed and combined with synthesized sounds. Oscillator, filter, LFO, and envelope signals can be routed to multiple modulation targets. Two wavetable oscillators (three waves per voice) allow unprecedented manipulation of waveform data. Two multi-mode filters simultaneously output high pass, low pass, band pass, and band stop filtered signals. Three LFOs per voice with dedicated multi-mode ASR envelopes. The LFOs can use the same waves as the OSCs for the ultimate modulations. The LFOs can be synced to MIDI beat clock or internal clock. Multiple LFO loop modes are available, as well as key tracking. Three ADSTAR envelopes can be used as s source for any modulation target. The second attack (post key up) can be enabled or disabled. Effects include wave shaping, EQ (high pass, low shelf, high shelf), and a modulatable delay line capable of chorus, flange, vibrato, pitch warping, doubling, and echo effects.

FEATURES

Wave Editing

- The vast Synthesizer One sound pallete starts with its editable wavetables. This allows you to create unique waveforms. The waveform editing facilities are built into the Synthesizer One interface and Wave Edit control page. Several waveform functions, filters and edit modes catalyze the creative process. Mouse drawn waveforms and functions can be created. Up to 8 edited wavetables can be saved per setting file for an unlimited range of sounds.
- A modular design allows oscillator, filter, LFO, envelope, and audio signals to be routed to a variety of modulation targets. Many synthesis parameters have a dedicated menu of modulation sources on each control page for optimal editing speed. A 4-point modulation matrix is provided for additional modulation routing.
- A set of Quick Pages gives access to basic parameters, while the complete set of controls are available on the Synth Pages. MIDI controller assignment is programmable and saved as a preference inside the Synthesizer One. 4-point MIDI controller modulation matrix is available per patch.

Two Wavetable Oscillators

- 3 wave sources with independent waveform, level, phase, octave, PW and PWM control (note PWM available for ANY waveform - the only synth on the planet to offer this feature)
- Many preset waveforms and up to 8 editable waves are available per wave source
- FM and FM velocity controls for each oscillator
- Separate Ring Oscillator with selectable inputs (OSC, LFO, ADSTAR envelopes, etc.)
- Separate noise source

2 Multi-mode Filters

- High Pass, Low Pass, Band Pass, Band Stop with frequency and resonance control, 12 dB/oct per filter
- 2 frequency and 1 resonance modulation source per filter
- Simultaneous output of all filter types
- Filters operate in series, parallel, and other routing configurations

3 Low Frequency Oscillators

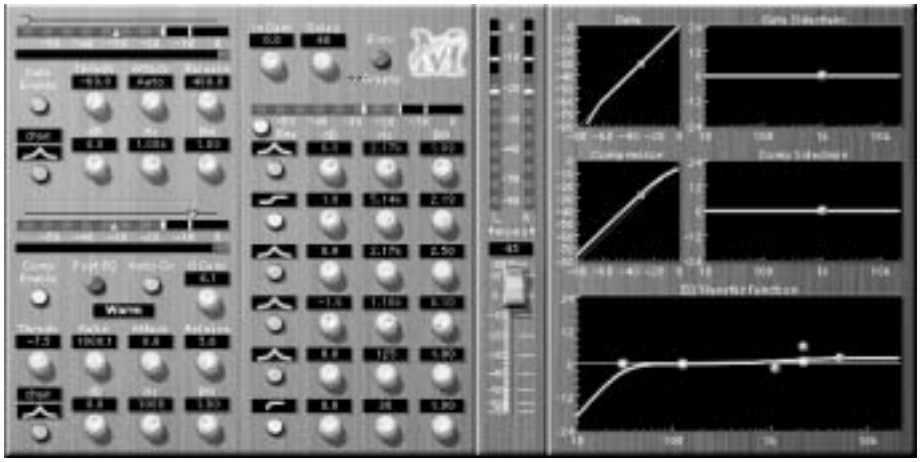
- Use same waves as available to oscillators
- Syncable to MIDI beat clock
- Multi-mode ASR envelope per LFO

Additional Features

- 3 ADSTAR Envelopes: Attack, Decay, Sustain, Time, Attack2, Release
- Every modulation target (PWM, VCF FM, etc.) can select from a popup menu of available modulation sources (LFO, ADSTAR envelope, OSC, etc.)
- User definable arpeggiations and sequences saved per preset
- Audio can be mixed, processed, and used as modulation source
- Double precision arithmetic for 48-bit processing
- Analog saturation modeling
- Requires MIX or HD TDM system
- Page table support for ProControl, Control 24, and other control surfaces

CHANNELSTRIP

World-Class Audio Engineering Tools



As the name implies, ChannelStrip replicates the audio processing section of a digital console – powerful dynamics and EQ integrated in a single, fully automatable plug-in for ProTools and MAS audio environments. Developed as a way to marry the sound and feel of a world-class mixing console with the flexibility of digital audio workstations, it offers exceptional audio quality, along with incredible DSP and CPU efficiency. The comprehensive user interface allows you to work with your DAW as efficiently and interactively as you would with a dedicated mixing console.
Also available with the same high-resolution algorithms but with a slightly scaled-down feature set. This means that ChannelStrip|SP will use less DSP and run more efficiently.

FEATURES

Transform your digital audio workstation into a world class mixing console. ChannelStrip provides all of the critical channel processing features you would expect to find on a top-of-the-line mixing console in a single mono or stereo plug-in. Processing functions include:

- Input level control
- Expander/Gate with filtered sidechain
- Compressor with filtered sidechain
- 6 band Parametric EQ
- Phase Invert
- Channel delay
- Advanced metering

Filter Type	Frequency Range	Boost/Cut	Bandwidth
Peaking/Parametric	20Hz-22kHz	±24dB / ±12dB (SP)	0.1-2.5 oct
High Cut	20Hz-22kHz	12dB/octave w/3dB	
Low Cut	20Hz-22kHz	12dB/octave w/3dB	
Low Shelf	determined by BW	+12dB/-24dB ±12dB (SP)	
High Shelf	determined by BW	+12dB/-24dB ±12dB (SP)	
Bandpass (CS only)	20Hz-22kHz	6dB/octave	0.1-2.5 oct

ChannelStrip TDM, RTAS and AS (includes ChannelStrip|SP).....599.00
ChannelStrip MAS, RTAS and AS (includes ChannelStrip|SP) 299.00
ChannelStrip|SP TDM, RTAS and AS 299.00
ChannelStrip|SP MAS, RTAS and AS.....159.00

	ChannelStrip	ChannelStrip|SP
EQ Section		
Total Bands	8 (2 on sidechain)	4
Selectable Filter Types	6	5
Compressor Section		
Detector Level Meter	Y	N
Character	smooth, warm, fast	warm
Ratio Control	1:1 - 1000:1	1:1-8:1
Attack Control	0-500ms	0-500ms
Release Control	5ms-5sec	5ms-5sec
Post EQ option	Y	N
Sidechain	2 bands EQ	N
Delay section	255 samples	N
Phase Invert	Y	Y
Peak Meter (.9 sec/20dB) w/ Clip Indicator	Y	Y
Dynamics Metering on Gate Knee	Y	N
Dynamics Metering on Compressor	Y	N
Gain Reduction Meter (0-60dB dynamic range)	Y	Y
Peak , RMS, VU Output Meter (IEEE standard 300ms RMS level)	Y	Y
Automation	Y	Y
Input Gain	+24dB	+24dB
Gate/Expander Section		
Detector Level Meter	Y	N
Threshold Control	Y	Y
Attack Control	max 100ms	N
Release Control	5ms-5sec	80, 400 or 1200 ms

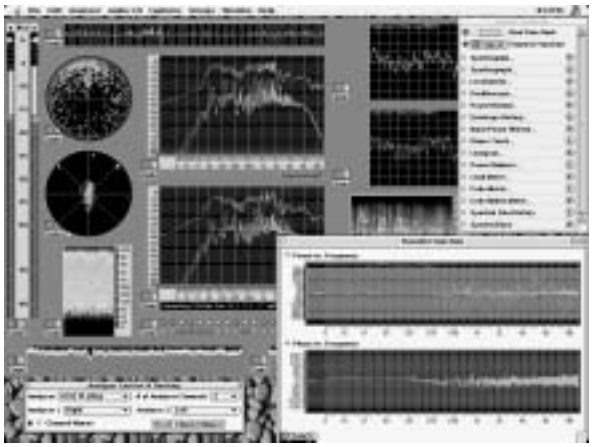
SPECTRAFOO • SPECTRAFOO COMPLETE

Visual Audio Monitoring System

SpectraFoo provides standards-based level metering, high-speed, high-resolution spectral analysis, the unique Phase Torch, correlation metering, triggerable waveform display, power balancing and a variety of power, envelope and spectral histories and phase analysis on any number of input or output channels. Create as many copies of each type of instrument as you like. Every copy is independently assignable, movable, sizeable and configurable. This allows you to create the exact metering set (or sets) you need.

Both absolutely essential for broadcast and critical mixing & mastering, the SpectraFoo Complete steps up with a Transfer function, signal generator, 24-bit audio capture function and three sample code metering tools (a bit-range meter, bit-matrix meter and bit-scope) to the fundamental set of software-based metering and analysis instruments.

SpectraFoo Complete



FEATURES

- The software automatically maintains a snapshot of the current system state, so every time you start working with the system it is configured exactly as you left it. It also provides a variety of sophisticated configuration management features to maximize your effectiveness while minimizing the time you spend tweaking the controls.
- Provides a complete parameter storage and retrieval library mechanism for all instruments in the system.
- Create named configurations for each of the various instruments and instantly recall them with a single mouse click
- Modifications of parameter libraries can be saved over the originals or cloned to create new libraries
- Comprehensive set of instrument presets
- Create a complete snapshot (meter positions, visibility, configuration, channel routing, etc.) of the entire state of the system and save it as a preset document. Or assign it to a hot key for instant one-key access to various metering configurations
- Flexible linking system lets you keep all of the parameters associated with a group of instruments linked. Use to link the setting on a stereo pair (or surround group) of instruments as well as synchronizing the settings of different types of meters that are all on the same channel.

SpectraFoo

- Lissajous phase scopes (X-Y, vectorscope and stereo field display modes)
- Full featured audio frequency oscilloscopes
- Power balance meters
- Full featured level meters with physical unit calibration
- Full featured audio spectrum analyzers
- Spectragram spectral history meters
- Correlation meters
- Correlation history meters
- Phase Torch frequency-sensitive phase meters
- Envelope history meters
- Band Limited history meters

SpectraFoo Complete— Step-up Features

If your only use of the software is for metering your mix, then SpectraFoo will give what you need. However, if you want to be able to measure your acoustic environment or verify the behavior of your external digital devices, then you need SpectraFoo Complete.

More Instruments:

- Three different kinds of bit meters for monitoring the low level audio bitstreams

Transfer Function Measurement System

- Allows direct measurement of both acoustic and equipment transfer functions (both frequency and phase)
- Use for equipment verification and test
- Can be used to “shoot” your control room for acoustic and electronic correction
- Provides full featured impulse and time-delay measurement tools for time aligning the transfer-function measurements
- Time-delay tools can be used for time aligning multiple audio tracks

World-class Signal Generator:

- Provides high resolution 24-bit distortion free signal generation
- Up to 9 simultaneous sine sweeps
- Pink and White noise generation
- Burst Generation
- FFT synchronized sine generation
- Direct generation to audio I/O, captures and files

Capture and Static Analysis system

- Allows you to record directly into SpectraFoo
- Provides loop playback
- Allows you to open soundbites in SpectraFoo for detailed analysis

METRIC HALO

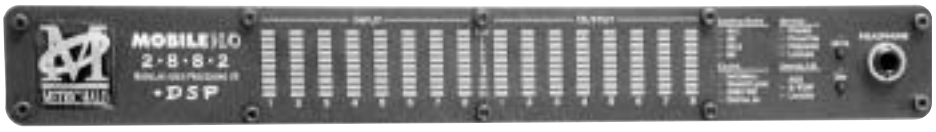
MOBILE I/O 2882 • MOBILE I/O 2882+DSP

Multi-Channel IEEE1394 Audio Interfaces



Designed to fit under a laptop (or rackmounted) the Mobile I/O 2882 is an affordable, portable high-quality, multi-format audio I/O for professional and multimedia applications.

Connecting to your computer via IEEE1394 (Firewire), the Mobile I/O 2882 line has 8 channels of analog input and output, 8 channels of ADAT lightpipe I/O, as well as AES/EBU and S/PDIF digital I/O with SRC. It supports all of the standard professional and multimedia sound driver APIs for both Mac and Windows including ASIO, Sound Manager, and Wave Drivers. Each input channel has individual phantom power, and all are fully matrixable. Includes powerful multitrack recording and mixing software. Exactly the same, the Mobile I/O +DSP has an extra DSP chip onboard to capitalize on the additional processing power of Metric Halo software and third party plug-ins.



- Portable audio solution that provides studio quality audio interfacing to any computer with a FireWire port
- Small 1RU high form factor is compatible in size with portable computers and computer bags – rack ears are included
- Full 10 segment metering for all input and output channels
- Low-power design – can be directly powered by the FireWire bus when used with desktop computers, or via an external DC power source for portable computers, or with an external battery pack.
- Multiple boxes may be chained to make larger systems. Box types may be mixed and matched to provide user configurable I/O. All boxes include programmable logic and a fully field-upgradable audio engine implementation.
- Completely silent operation – NO FAN!

Analog I/O

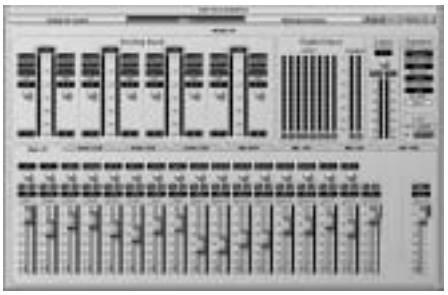
- Eight balanced analog inputs with high quality 24-bit/96kHz converters:
  Four XLR mic inputs with discrete mic preamps
  Four 1/4″ TRS line level inputs
- Eight 1/4″ TRS analog outputs using high quality 24 bit/96kHz D-to-A converters
- High quality 24 bit/96kHz 1/4″ stereo headphone/monitor output from DSP

Digital I/O

- 8 channels of ADAT optical I/O
- Stereo switchable AES/EBU (XLR) and S/PDIF (RCA) I/O with Sample Rate Conversion from 48 to 96 kHz on the fly.
- Word Clock I/O (2x BNC), switchable between 1x & 256x Clock, allows professional synchronization of multiple boxes to external studio clock.



- Built in matrix/routing mixer for low-latency foldback, monitoring, and routing
- Full cross-point routing between any two channels on any number of boxes



Also Available
ULN-2 Ultra Low Noise I/O

- The ULN-2 is a two channel interface that delivers the power, portability and sheer sound quality of Mobile I/O in an even smaller, lighter package.
- Mobile I/O ULN-2 supports simultaneous input and output of analog balanced & unbalanced (Mic, Line and Instrument), S/PDIF, AES, and IEEE 1394.
- Balanced analog inserts are also provided for each input.

- Mobile I/O 2882 and ULN-2 are both Macintosh OS 9 and X; ASIO and CoreAudio compatible.

FM7



The lively and punchy FM sounds were a breakthrough in sound synthesis when they were first introduced in the early 80's. Today, their special aesthetics are still highly appreciated by musicians and producers worldwide. FM sounds complement the sounds of analog and virtual analog synthesizers very well.

The FM7 not on reproduces the sounds of these machines exactly-right down to the smallest detail, but takes the legendary sound of FM synthesis to a new level. The FM7 reads the complete sound library of the classic DX synths (DX7, DX7-II, DX11, TX81Z, DX21, DX27, DX100, TX802, plus adds distortion and filter operators, extensive modulation capabilities, a comprehensive effects section, audio input and much more to the traditional FM architecture. The user interface of the FM7 makes it comfortable and easy to explore the fascinating new possibilities of FM – Native style.

FEATURES

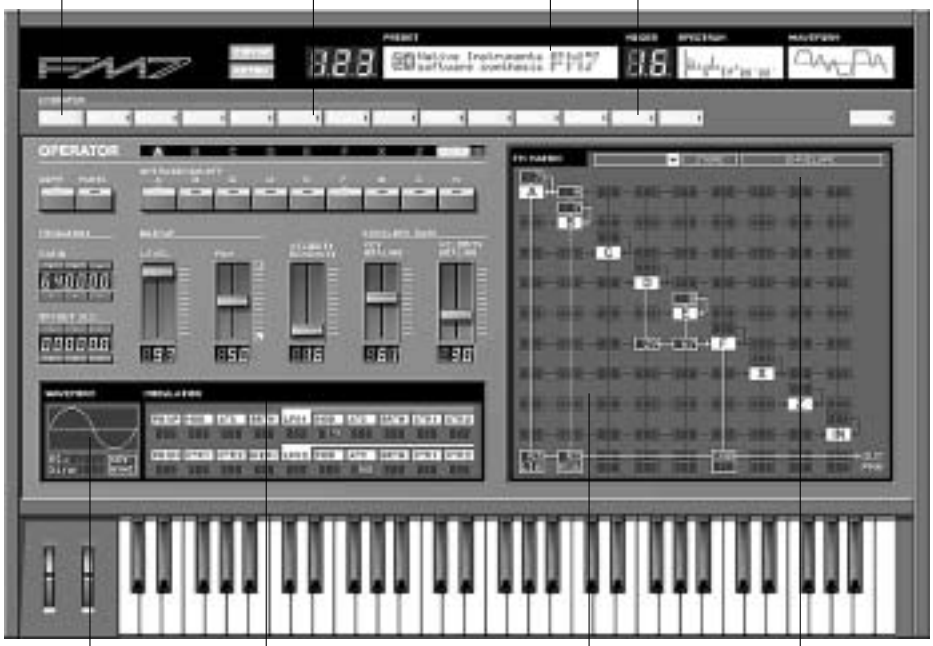
- The FM7 is more than an emulation. Enjoy the authenticity of the classic presets, and then transform them into something completely new. For example, each of the operators of the FM7 offers many waveforms besides sine.
- Additionally, a distortion operator with noise and an analog filter operator complement the waveform operators and increase the sonic range of the FM7, far beyond classic FM synthesis. Unlike the classic, FM routing can be set freely – you are not limited to preset algorithms. The effect section offers high quality stereo chorus, flanging and delay effects.
- Advanced graphical editors offer intuitive access to FM synthesis including a powerful analog-style Easy Edit page.
- While retaining compatibility to DX-Series instruments, it expands the sound repertoire and vastly improves the ease of use.
- With hardware synthesizers you have to choose between rackmount or keyboard version—now you get both plus a full graphical editor—at the press of a button.
- Includes a library of 256 top quality presets putting a wide selection of standard and unique new sounds at your fingertips. Go straight to making music without having to program any sounds.
- To make programming easier, FM7 has a page of dedicated analog-style controllers. Here, one knob can change the sound in drastic but intuitive ways, for example by turning up the Brightness, overall Decay Time, or Effect Depth.

Six identical Waveform Operators (the basis of FM synthesis), each sensitive to velocity for highly expansive control over the sound. The Operator is controlled with 32-bit accuracy and its frequency can be set with 6 digit precision.

FM7 has two additional Operators. X Operator provides distortion and noise for adding grunge to the sounds. Z Operator contains analog style filters to make the sounds warm and fat as desired.

Includes a library of 256 top quality presets, putting a wide selection of standard and unique new sounds at your fingertips.

To make programming easier, the FM7 has a page of dedicated analog-style controllers. Here, one knob can change the sound in drastic ways



Each of the Operators can have a different digital waveform. There are 32 different waveforms including Square and Sawtooth, giving each Operator a huge sonic potential.

Has a huge modulation matrix where the two LFOs and the performance controllers can be routed to control all the operators and the pitch, for expressive playing.

In the FM Matrix, the signal flow between the Operators is displayed and edited. Each operator can have a different stereo position to give a lively, spacious sound.

Each operator has its own Envelope. The Envelopes are very powerful with up to 32 stages, an adjustable curvature in each stage and sustain looping. They can also be synchronized to the song tempo.

NATIVE INSTRUMENTS

PRO-53



The Pro-53 is an incredibly faithful reproduction—in both sound and look—of one of the most popular analog synthesizers of the 80's. A virtual instrument fashioned after the Sequential Circuits Prophet -5, the Pro-53 combines the sonic properties of the unique originals with the practical requirements of the present day.
But the Pro-53 also adds 21st century technology to the immortal sound of the analog era—no fixed limit to the number of voices, many times more preset memories, velocity sensitivity, MIDI automation of all available parameters and the option to run several Pro-53's in parallel. And with the PRO-53, the third generation of the virtual analog classic is here. The synthesizer emulation has been vastly improved by a new oscillator technology that offers an even warmer and more brilliant sound. In addition the sonic possibilities have been expanded even further with new features like a high-pass filter mode and an invertible filter envelope. The control surface was also reworked and now shines in a whole new light.

FEATURES

- Complete analog voice architecture with comprehensive sound shaping facilities
- Accurately recreates the unique sound, panel design and synthesis architecture of the original:

—2 oscillators per voice

—Choice of pulse, triangle and saw-tooth waveforms

—Oscillator detune and synchronization

—24dB low-pass filter with resonance and self oscillation,plus high-pass filter mode

—ADSR envelopes for amplitude and filter, invertible filter envelope

—Low Frequency Oscillator with numerous modulation options, LFO-envelope retrigger function

- Almost 600 classic and modern analog sounds cover any eventuality., 64 of which were specially programmed by the legendary John Bowen, one of the creators of the Prophet-5 synthesizer.
- An optional 64 new sounds can be downloaded direct from NI for under $20. The sound set was created by Peter Krischker and contains sequencer sounds, pads, sweeps, effects, basses and lead sounds. It is suitable for musical styles like Trance, Electro, Techno, Ambient and Pop.
- 512 preset memories are available where you can store your own sound creations. The Pro-52 displays a name for each of the presets, unlike the original which only displayed the preset number.



Large knobs and switches provide easy, ergonomic operation and convenient editing

- Has an audio input for processing external audio signals through its filters and effects.
- Adjust the amount of inaccuracy and detuning inherent in the sound circuitry
- Integrates perfectly into any computer-based virtual studio. Supports VST, DXi, MAS and DirectConnect interfaces. Can be used as a sequencer plug-in or as a stand-alone synthesizer.
- The Pro-53 can import Prophet-5 SysEx sound data, which means it can read and accurately reproduce all the existing sound libraries made for the original. The subtle deviations from perfect behavior which are typical of analog circuits and crucial to the sound, can now be fine-tuned.
- A built-in effects unit adds multi-echo and ultra-fat chorus and flanging effects. This makes the sound of the Pro-53 'complete' for both live performance and studio production. The included preset sounds make extensive use of the effects.
- Realtime control and automation of all parameters using MIDI controllers Plus the MIDI-implementation is much more versatile. With MIDI-learn the control elements can be operated by hand by assigning them to the desired controllers, for example the knobs and sliders on a MIDI controller box. The set-up connections can be saved in a controller-map. Several pre-configured controller-maps come with the software and can be loaded directly into the PRO-53.

B4 ORGAN



The B4 is a complete virtual tonewheel organ, capable of reproducing in authentic detail the sound of the legendary B3 organ and rotating speaker cabinet. A state-of-the-art MIDI-automable audio engine transparently operating behind the photo-realistic graphics, the B4 can be used as a stand alone application using ASIO or VST 2.0, MAS, Direct Connect or DXi plug-in.
The B4 also adds flexibility in sound generation and seamlessly integrates into the recording environment, making it possibly the best organ since the original. Native Instruments carried out a painstaking analysis of the electro-mechanical sound generation and tube circuitry of the best organ and speaker cabinet combinations. The resulting software model, based on the same NSP-technology as NI's award-winning Reaktor system, produces sound virtually indistinguishable from the original hardware.

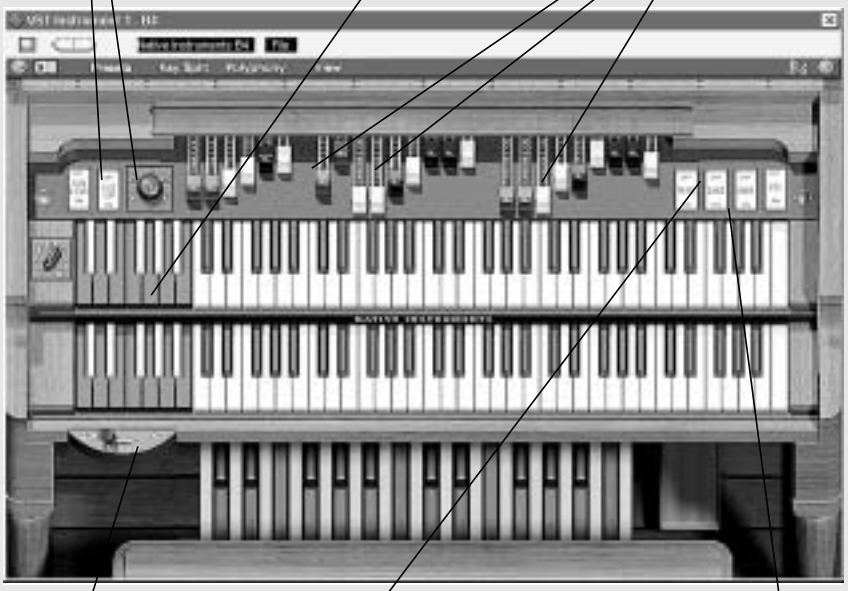
FEATURES

- Unlike sample-based instruments, the B4 accurately models subtle details such as harmonic foldback, drawbar crosstalk and loudness robbing. The result is a warm and powerful sound over the entire range of the keyboard, complete with the subtle tonal shades that improve playability and musicality.
- Capable of all the variations one expects from this type of organ, and is suitable for any flavor of music, be it rock, jazz or house.
- A set of knobs on the "rear panel" gives access to numerous adjustment parameters, which allow the sound to be tailored for particular applications.
- You can even use the B4 to process other sounds. Just plug it in as a VST Insert Effect and apply any combination of Scanner Vibrato, Tube Overdrive and Rotating Speaker effect to your tracks.
- Full MIDI control and automation—each and every parameter is mapped to MIDI controllers so that all settings and their changes can be easily automated and manipulated in realtime.
- The B4 has two views, one of which shows the two manuals, the bass pedals and other knobs relevant for playing. The second view skips the manuals and pedals but gives access to extended settings.
- With the optional B4 Tonewheel Set Vintage Collection you can replace the B3-modelled tonewheels with those of a Vox Continental, a Farfisa Compact, or a Harmonium to give your B4 a totally new character. Theset also permits the master-tuning to be altered, and to "age" the sound in six different levels, from "well matured" to "far beyond repair."

Patented Scanner Chorus/Vibrato effect can be switched on and off separately for the two manuals. The rotary switch chooses one of 10 Banks of Presets, containing the famous music sounds.

12 inverse colored keys are used to select presets not only for the Drawbars but to give total recall for all settings affecting the organ, amplifier and rotating speaker.

Nine drawbars per manual, another six drawbars for the bass pedal channel. The drawbars allow a huge number of different timbres to be selected.



Rotating speaker speeds can be realistically sped up and slowed down. The rotation speeds can be adjusted with complete flexibility on the virtual rear panel.

Percussion Circuit can be activated to amplify one of the Upper Drawer tones and lets it decay over time—essential for Jazz and other styles. Any of the nine drawbars can be selected for the effect, and speed of decay and percussion volume are adjustable.

The built-in Tube Amplifier can be operated in clean or overdrive mode. The rotating speaker simulation provides that lush, swirling spatial sound experience. The amount of both effects can be adjusted.

VOKATOR



Vocoding in a New Dimension

Vokator provides a revolutionary suite of vocoding tools with unrivalled flexibility and outstanding sound quality. Powered by an extremely high-resolution FFT spectral engine, Vokator sets the standard for vocoding transparency, detail, and smoothness. In addition to an exceptional vocoder, Vokator boasts a sophisticated synthesizer and a granular sampler: a virtual sound-fusion laboratory. With advanced features and multiple modes of operation, Vokator opens vast new worlds of sound-design possibilities. In addition to external audio sources, you can also use internally-generated signals from the synthesized/sampler and the audio file player. So Vovator places both an effects plug-in and a vocoder-based synthesizer at your disposal. Each input section has a delay unit, a dynamic processor and several special effects in the spectral domain.

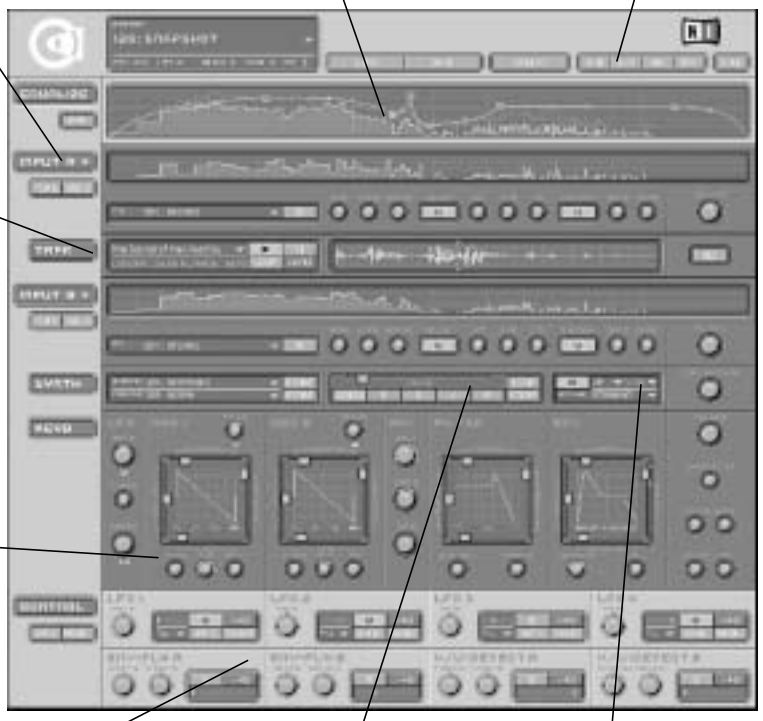
FEATURES

- Vokator's FFT spectral engine is a milestone in vocoder development. Previous vocoders used eight, sixteen, twenty, or even thirty-two frequency bands – the more frequency bands, the smoother and creamier the effect. Unlike its predecessors, Vokator's advanced engine uses 1024 bands. In addition to its stunningly transparent, full-resolution operation, Vokator's bands can even be grouped together for a convincing vintage-vocoder emulation. To ensure optimal dynamic range across the frequency spectrum, an integrated frequency-domain compressor balances the levels of all bands.
- A pair of vocoding channels can either be spectrally combined or played independently. Channel A can be switched between an integrated file player or external input A, while Channel B can be set to a full featured synthesizer, a time-stretching granular sampler, or external input B. For a traditional vocoder effect, the frequency spectrum of the external input would be controlled by the synthesizer, but Vokator's capabilities are much more advanced.
- Vokator is both an effect plug-in and a vocoder-based synthesizer. Its synthesizer features an advanced dual-oscillator design with dynamic preset morphing controlled by the modulation wheel. Vokator's granular sampler can independently control a sample's pitch and time. A full range of modulators – step sequencers, envelope followers, LFOs, and more – can be easily routed to nearly any of Vokator's parameters.

Each input section has a delay unit, a dynamic processor, and several special modulation effects in the spectral domain.

The graphic displays show the spectral composition of the audio signals. The main display also serves as a graphic EQ for drawing in precise filter curves for the output signal.

Each of the two inputs can act as carrier or modulator. In mix mode, both inputs work as carrier and modulator.



A MIDI-controllable audio file player, with loop function and waveform display, can stream large audio files from the hard disk.

Editing is straightforward with graphic display of envelopes, and waveforms. The synth's two oscillators can be replaced with a re-synthesizing granular sampler which allows start, loop, pitch, and speed modulation.

LFOs, envelope followers, voiced/unvoiced detectors, and step sequencers can modulate nearly any parameters. The yellow ring around a knob indicates the value of the respective modulation.

The morph function opens a new sonic dimension by allowing smooth transitions between synthesizer presets. The morphing is controllable in realtime by the modulation wheel.

The integrated arpeggiator provides different styles, directions and articulations to trigger the synth section.

SPEKTRAL DELAY



Effects Software

Spektral Delay differs from other effects in that it uses real-time FFT (Fast Fourier Transformation) to split up each channel of a stereo signal in up to 160 separately modifiable frequency bands (even up to 1024 bands internally). The level, delay time and feedback amount for each of this bands can be set separately. Additionally, various modulation effects can be applied to the signal in the frequency domain, which allows for even further sound manipulation.

And while Spektral Delay offers exceptionally great possibilities for creative sound design—from subtle corrections and rich effects to the total alienation of the input signal—it is still easy to use since all parameters are controlled intuitively and comfortably from the graphical user interface. Stand-alone or as a plug-in, it offers professional sound quality and can be fully controlled by MIDI.

FEATURES

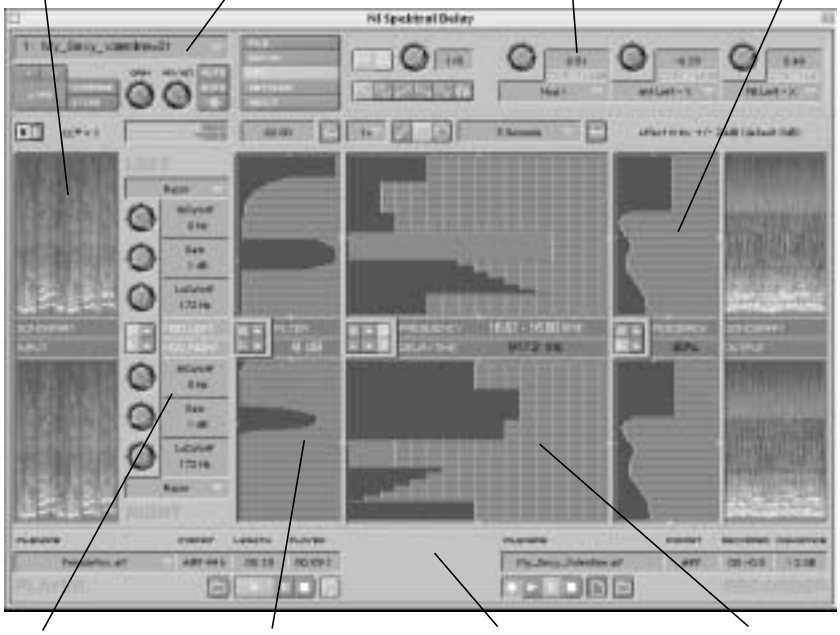
- Intuitive Edit Graphs let you design delay times and feedback levels for each frequency band: each of the up to 160 bands per side can be delayed and fed back separately.
- Delay times can be aligned with the rhythm via a selectable tempo grid. With up to 12-seconds delay, Spektral's range covers everything from subtle coloring to rhythmic flowing of the partial tones to dense, atmospheric sound textures.
- Modulation effects (expandable by plug-ins) are applied in the frequency domain of the signal and offer many unusual processing options – such as rotation of the frequency bands or reverb simulation by 'smearing' the amplitude courses of the bands—a big factor in Spektral's unique sound possibilities
- The amplitudes of the bands are also controlled intuitively by drawing in the Edit Graph with the mouse. Unlike conventional filters, the filter curves resulting from this can have any shape— design any filter response imaginable, from extreme settings to smooth sweeps. The filter curves can be drawn commonly or separately for the two channels of a stereo signal.
- All parameters such as filter, delay and feedback settings can be modulated with an integrated LFO or via MIDI. This makes it possible to create very organic and lively effects which can easily be played live.
- Includes many presets for different applications (e.g. guitar, vocals, drums, ambiences, reverberations, special effects). Runs standalone (Mac/Win) with MME, Direct Sound, Sound Manager and ASIO or as a plug-in with a VST-or DirectX-compatible host.

Real-time sonograms display the spectra of the audio signal at both the input and output stages, allowing you to visualize the frequency content of the sound and to see exactly how the processing has effected your sound.

Master Control Panel gives you direct access to presets as well as controls for freezing the spectrum in the delay buffer, controlling the gain of the processed audio, and mixing between the wet and dry signals.

An LFO section with sine, triangle, square, random, saw up, and saw down waveforms can be routed to any of the six edit graphs for very creative modulation possibilities and can be synced to an external tempo

Feedback Matrix editor gives you the power to apply a different feedback ratio for each frequency range, allowing you to generate creative ambiences, echoes, flanges, and phasing effects.



Input Modulation modules offer a wide range of unique processing algorithms, such as Deterioration, Foam, Sponge, Imprint, Horse Tail, Jello Mold, Razor, and Smear.

Input Filter section is controlled by an easy to use edit graph. Simply draw a gesture into the filter matrix with the pencil tool and you immediately effect the input signal with these filters.

Stand-alone version offers an integrated sound file playback and record deck, letting you feed SPEKTRAL with sound files of your choice and record your processing directly to disk.

Delay Matrix allows up to 12 sec. of delay for each frequency band. Modulate the gesture with the LFO or the automation tool, and all movements can be recorded and controlled by MIDI.

REAKTOR 4



Generate the Future of Sound

REAKTOR 4 is a complete sound design studio with almost endless possibilities for producing and performing music. Use it as a synthesizer, sampler, effects processor, groovebox, and much more. High quality signal processing algorithms and the 32-bit floating point precision of the audio engine guarantees sound quality that meets the highest demands. The steadily growing REAKTOR libraries contain hundreds of different instruments with thousands of presets for a wide range of sound generation and audio processing applications. All instruments can be modified freely, and even building your own instruments from scratch is possible. Runs stand-alone or as a plug-in for Windows and Mac with minimal latency and very precise MIDI timing.

FEATURES

The Library

REAKTOR 4 offers a huge and growing library with staggering instruments, music machines and effect tools. Explore analog modelled, additive, FM, and unique synthesizers. Groove with real-time time compression and expansion. Mix with compressors, EQs, pitch shifters, filtered delays, distortion, and up to sixteen channels of surround. Everything is immediately at your fingertips, from authentic emulations of classic gear to creations never heard before.

The REAKTOR Principle

REAKTOR 4 offers over 200 basic modules which are the source material for creating any sound generator or effect-processor you can think of: Numerous “analog” and sampling oscillators, various filters and envelopes, shapers, delays, sequencers and much more. You can interconnect all the modules freely in the structure window.

Macros are used as “shortcuts” to quickly recall a set of modules or structures and allows a more clear and organized layout to your designs. You can even have macros within macros. First-time users can be designing their own creations in no time.

Instruments are the next level up consisting mainly of macros, but also modules as well. It has its own MIDI parameters, program memory, and a separate panel window for the controllers. Several instruments can be combined together in a “multi-timbral” like environment giving you the power to create entire songs with REAKTOR 4.

Newest Modules

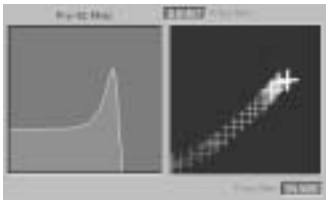
REAKTOR 4’s latest modules cover everything from analog-modelled filters to innovative graphical control elements, from multi-tap delays to hard disk recorders and players, from granular synthesis to matrix mixers.

XY Control and Realtime Display

The multifaceted XY objects turn REAKTOR 4 into a powerhouse for musical and visual performance. Visualize your sound or plug in oscillators just to see what patterns emerge. The XY objects can be resized as big as you like. By moving the mouse within the XY control area, you can control two parameters at once, while an oscilloscope, waveform, or pattern is displayed underneath.

Audio- and Event-Array

Draw your own envelopes, sequences, waveforms, and more. The Audio Array can import any sound file and cycle through it at any frequency. Create oscillators out of audio files. The Event array can be used to draw in envelopes with an unlimited number of curves.



Analog Modelled Filters

Taken straight from the acclaimed Pro-52, the Pro-52 filter module gives you the warmest analog sound, but under complete control. The Ladder filter module is an authentic emulation of the classic Moog filter.

Grain-Cloud

The Grain Cloud module gives you the power of granular synthesis with the ease of use, sound quality, and control that REAKTOR is famous for. Built-in randomized jitter lets you create delicate clouds of sound or complete sample shredding. Coupled with the advanced control elements that REAKTOR 4 offers, this module is a monster.

Hard Disk-Recording and Playback

Record your inspiration directly to hard disk, or stream files of unlimited length through REAKTOR 4 with the tape deck modules.

8-Tap-Delay and 8-Channel-Scanning-Mixer

The 8-Tap Delay module lets you create surround delays and reverb early reflections. In combination with the Scanning Mixer you can make effects like the classic Scanner Vibrato as used in the B4. The Scanning Mixer module takes a floating point input between 0 and 8 to route audio dynamically from its eight inputs with crossfading. The Multi Tap Delay lets you set eight individual delay times and send each delay tap out of its own output. It’s possible to create surround reverbs and delays of incredible realism and complexity.

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ONE HOUR FREE PARKING
AT 349 W. 34th STREET (with purchase of $100 or more)

REAKTOR 4

User Interface

User interface features drag & drop functionality, professional metering, enhanced toolbar, the floating inspector, visual samplers and analog modelled filters. All knobs, sliders, buttons, and toolbar offer great clarity. REAKTOR 4 also lets you customize your environment by choosing any color for the Panel background and the Selection color.

Drag & Drop

Drag & Drop functionality makes it easy to get audio files, instruments, and macros into REAKTOR 4. Download some new macros from the Internet, drag them into REAKTOR 4, then start to make music. You also have the ability to insert multiple audio files into a sampler module. Getting ready for a live performance? Drag 128 audio files into a Beat Loop module and you're ready to roll in seconds.

Professional Metering

Level Meters have two resolution modes (coarse and fine) and are as accurate and as fast any any digital metering. All Level Meters can also have a value readout to accurately monitor your audio levels, or to act as numerical displays for any object.

Enhanced Toolbar

Toolbar provides very accurate input and output metering, and can also be undocked and used as a floating palette. Toolbar lets you select sample rates up to 96 kHz, type in a tempo, and assign MIDI controllers to any panel element. It also displays CPU load, manages snapshots, polyphony, and more.

Floating Inspector

All parameters in REAKTOR 4 are set with the floating Properties Inspector. You can modify several modules one after the other, even if they are not related.

Visual Sampler and AKAI Import

All Sampler Modules can display their audio data in the Panel Window. Any compatible sound file can be dragged and dropped into the sampler's Property Inspector for instant use, and AKAI sample data can be imported from AKAI-formatted CDs.

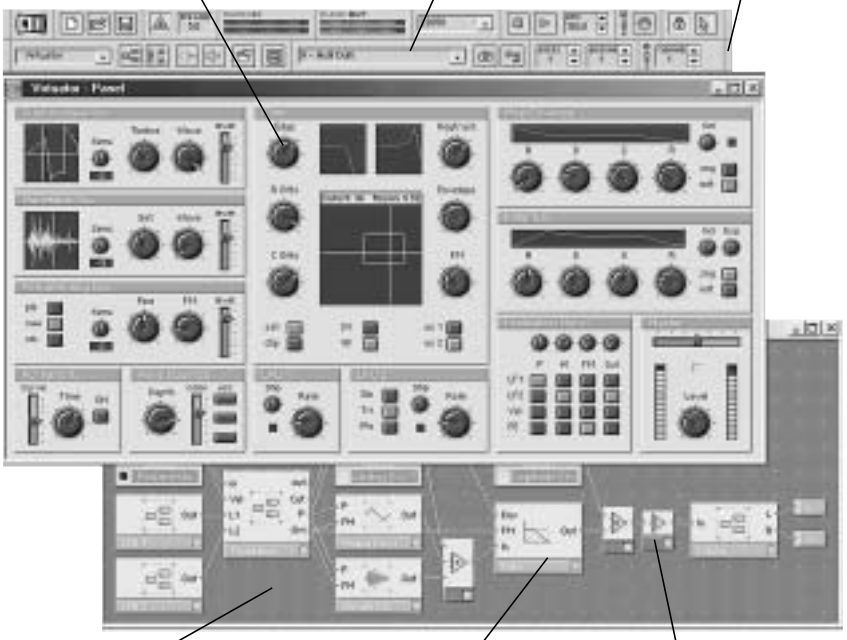
The main views of REAKTOR 4 are the panel and the property windows, and they are separated to make it easy to design, build, and perform. Convert your computer into a synthesizer, sampler, groovebox and effects processor. Build your own instruments and discover the treasures in the huge instrument libraries.

The panel is the user interface for an instrument. It can be freely designed. So it is possible to exchange knobs for faders, change the size and position of panel elements and add new buttons, knobs, lamps and more. All panel elements can be assigned to any MIDI controller. Property windows give access to the specific parameters of instruments, macros or panel control elements. A panel elements look, range of values, MIDI controller information and more can be set by double-clicking on the element.

Control elements such as faders, knobs, buttons, and switches are used to control the instruments. Filter curves, envelope displays, level meters and oscilloscopes show you what is going on.

Snapshots are saved settings. Since the positions of all parameters in an instrument are saved in a snapshot, it's possible to save a sound, an effect setting, or even a drum or sequence pattern.

Toolbars show the most important parameters of REAKTOR 4 and give access to frequently used functions (input/output levels, MIDI activity and CPU load are also monitored here).Snapshots can be stored, modified or recalled from the toolbars.



The structure shows the internal machinery of an instrument. Here, macro and modules—filters, oscillators, LFOs, envelopes, etc.—can be placed and connected together.

Macros encapsulate a combination of modules, their wiring and the associated panel controls. Macros make it possible to organize instruments in a hierarchical way. REAKTOR 4 offers a comprehensive library with premade macros to quickly build up new instruments.

Modules represent the elementary processing functions and panel elements. REAKTOR 4 comes with more than 200 modules: oscillators, samplers, filters, envelopes, shapers, delays, step sequencers and much more.

NATIVE INSTRUMENTS

REAKTOR SESSION

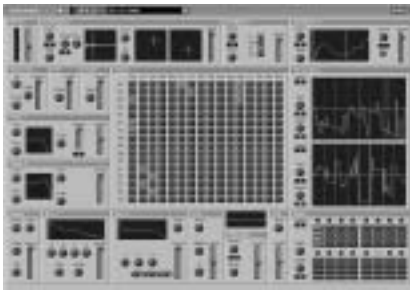


Access to the Future of Sound

REAKTOR SESSION is your key to a world of exceptional software instruments. A vast library of synthesizers and resynthesizers, samplers and beat-loopers, groove and drum machines, surround and innovative effects, and sequencers can be instantly integrated into your studio. Each instrument includes a wide range of preset sounds for instant use, while convenient patch management lets you create and save your own sounds.
REAKTOR SESSION lets you freely design your own studio from pre-built instruments. Multiple instruments can easily be connected together in the streamlined structure editor. Mix sequencers, synthesizers, samplers, and effects. It lets you effortlessly realize your studio ideas with its intuitive user interface. A built-in instrument/sample browser and convenient features such as prelisten, replace, and insert make the selection, and addition of instruments easy.

FEATURES

- The highly-optimized DSP algorithms of the 32-bit engine and up to 192 kHz sample rate support ensures unsurpassed sound quality with very little CPU usage.
- All user interface objects can be fully controlled over MIDI via the MIDI learn function. After the activation of MIDI learn for the desired parameter, move any hardware controller, such as a knob on a MIDI controller, and that hardware control will automatically be as-signed to the desired parameter. Naturally, all MIDI settings are saved with the instrument.
- Runs on Mac and Windows with minimal latency and excellent timing, and smoothly integrates into the virtual studio as a VST 2.0 or DXi plug-in. Even the most complex REAKTOR creations appear as one plug-in under VST and DXi, and multiple instances can be opened simultaneously. As a stand-alone application, supports ASIO, DirectConnect, and MAS.



Matrix Modular: One example out of several hundred instruments which can be easily loaded into SESSION

REAKTOR Instruments in Your Studio

In the Studio Panel you can control the instruments you have loaded. You can select, change, and create sounds, effects settings, and sequencer patterns. You can add and remove instruments, or simply rearrange their order. The instrument info display show MIDI activity and allows you to recall and save presets, mute and solo the instrument, select the MIDI channel, and choose general instrument settings.

SYNTHESIZERS

The REAKTOR SESSION library offers a world-renowned collection of synthesizers from analog-modeling to FM, from wavetable to additive. Each synthesizer includes many presets for an astonishing range of sounds right out of the box. High quality oscillators, filters, and envelopes produce warmly authentic “analog“ bass, lead, and string sounds. FM, ring-modulation, and hard-sync synthesizers produce a broad scope of organically complex and metallic sounds.

SAMPLERS AND TRANSFORMERS

Included sampler and transformer machines enable everything from quick and convenient sampling to a full granular workout. Samples can even be used as waveforms, or as the basis of FM synthesis and waveshaping. Several resynthesis instruments in the library allow you to vary speed and pitch independent of another, and even to freeze the sound in time. Realtime timestretching and pitch shifting let you perform live intuitively and quickly. A built-in loop recorded lets you easily create new loops to use in your sequencer program, or to put back through REAKTOR SESSION’s samplers and effects.

SEQUENCERS, DRUM & LOOP MACHINES

Many instruments in the library combine synthesis and sampling with sequencing for a variety of sequenced synths, beat boxes, and loop manipulators. A global tempo control ensures that multiple sequencers and even loop playback will always run in sync. Graphical note editors and random pattern generators create inspired tools which can be used for the creation of new loops, or can be played live.

EFFECTS

SESSION can be used as an extremely flexible effects processor. Process sounds in VST or DXi, from the audio inputs, or from the internal instruments. The effect library covers a full range of delay, reverb and modulation effects, filters, distortions, and dynamics processors, panning and surround effects, and granular pitch and vocoder effects. All effects can be combined for the creation of completely new real-time processing chains.

MIXER

A comprehensive selection of mixers and routers allow you to mix the input signals and instruments together, and to combine insert and send effects into the signal flow.

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NATIVE INSTRUMENTS

ABSYNTH



The benchmark for soft-synths, the powerful semi-modular architecture and clean interface of ABSYNTH make it easy to sculpt everything from organic textures to rhythmic madness to vintage sounds. ABSYNTH´s unique strengths lie in its multiple synthesis techniques combined with the most flexible envelope control ever. Forget everything you know about modulation - with ABSYNTH you can create everything from timbrally morphing percussive loops to physical modeling textures to time-evolving soundscapes. Any parameter adjusted in ABSYNTH is immediately heard without clicks or glitches. To taste ABSYNTH's power, draw the shape of your LFO with the mouse while twelve 68-point rhythmic envelopes twist your sound as you perform on a keyboard or MIDI sliderbox.

Patching and Waveforms

Flexible patching scheme makes it easy to combine subtractive, FM, AM, ring modulation, and waveshaping. Offers 6 oscillators, 4 filters, 3 ring modulators, a waveshaper and a delay processor per voice along with unprecedented graphical modulation capabilities. Start creating sounds immediately in the Patch Window - no need to create your own structures.
Model your own waveforms. Draw waves directly or use the spectrum mode to trim individual harmonics. Mix, modulate, filter or fractalize waves and hear the results while you edit.

Envelopes and LFOs

Over a dozen graphical envelopes with up to 68 breakpoints each. Create rhythmical envelopes with an integrated tempo-based function generator. Control almost any parameter with an envelope and edit multiple envelopes at the same time. LFOs can modulate pitch, amplitude, filter, FM depth, pan, and delay time.

Effects, MIDI, and Recording

Modulated delay effects can be tightly integrated into each patch. Delay times can be as short as one sample and as long as 10 seconds. And nearly every parameter in ABSYNTH can be modulated by MIDI continuous controllers, after touch, velocity, and note number.

Sounds

Comes with over 700 sounds – from classic resonances to acoustic emulations, from poignant pads to powerful percussion.

Plug It In

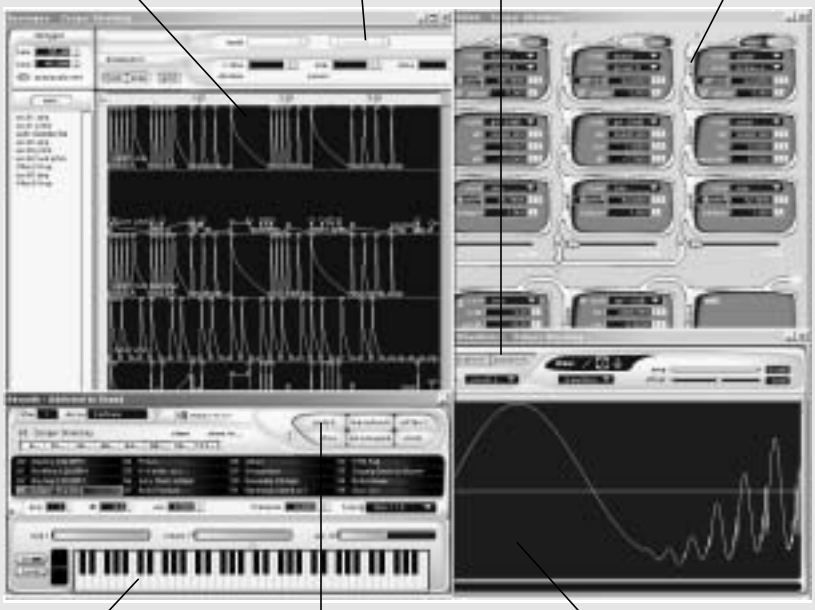
Run up to eight instances of ABSYNTH in any VST 2.0 compatible host. Create massive rhythm ensembles or ambient textures with hundreds of oscillators. ABSYNTH also supports MAS, ASIO and DirectConnect to fit into any system.

With up to 68 breakpoints per envelope and four modes (loop, retrigger, sustain, release), you can sculpt how your sound evolves over time and view multiple envelopes at the same time.

Build rhythms using ABSYNTH's envelopes. Using the pulse function generator, you can specify a tempo to create percussive patterns. Control anything with the envelopes – from amplitude to pitch to FM parameters to filters.

Spectral Waveform Editor lets you specify the harmonics and ABSYNTH does the rest – in real-time. Hear how your waveform sounds as you make harmonic gestures, or tweak the spectral view to fine tune your shape.

Choose between different synthesis methods and construct your sound by adding a wide selection of filters, wave shapers, and effects. Create a patch with up to six oscillators, four filters, three ring modulators, a wave shaper and master effects.



Compact yet informative VST window gives you a clear overview of the most important parameters. By clicking on any of the edit buttons in the top right, you can instantly edit any ABSYNTH parameter.

Use any waveform (even ones you've drawn yourself) or a sample and hold function as an LFO. Type in the frequency with four-decimal-place precision for locked tempo synchronization. Control effects, filters, FM parameters, oscillators, or any combination.

Draw in your own waveform and hear the results in real-time. Convenient editing tools let you draw parabolic shapes or free gestures. The waveform you draw can be used as an oscillator, an LFO, or a wave shaping function.

NATIVE INSTRUMENTS

BATTERY



Drum Sampler

Breathe life into your drum samples. BATTERY boasts separate sound parameters for 54 instruments, powerful modulation capabilities and unbeatable ease of use. It fulfills even the highest demands regarding precision, sound quality and flexibility, plus it is fast, effective, intelligent and convenient. It has sample-accurate timing, internal 32-bit resolution, up to 128 velocity layers per instrument and can play samples at any pitch

BATTERY has access to all the biggest sound libraries because it is compatible with AKAI, SF2, LM4, AIFF, WAV and MAP formats. It also comes with 30 high quality soundsets.

BATTERY runs either stand-alone or as a VST plug-in on Mac and Windows. It can be played and modulated via MIDI, offers full VST automation and supports all other common interfaces as well. Up to 32 outputs are available which can be configured flexibly as mono or stereo outs.

FEATURES

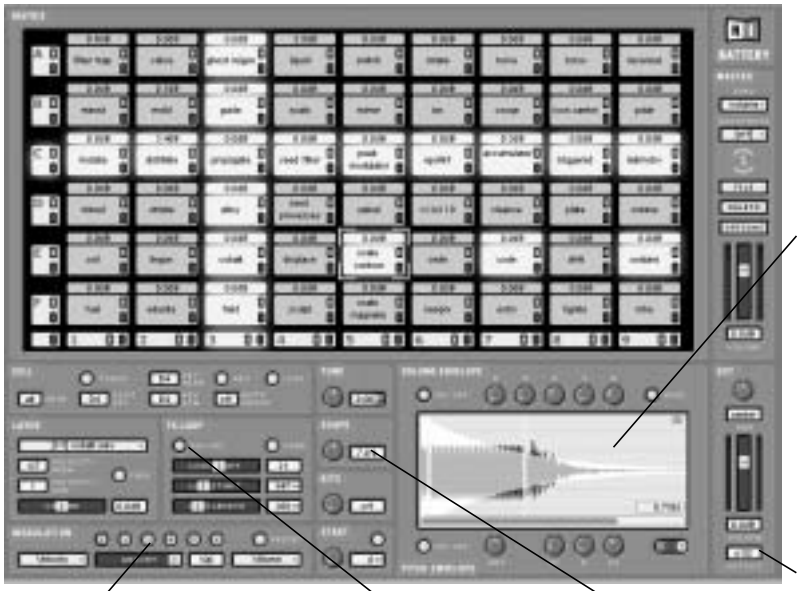
Instruments and Samples

- BATTERY can play 54 instruments at a time. The matrix and it's intuitive drag & drop functionality are a big time saver when arranging samples. Sounds from different soundsets can be easily compared and combined into new soundsets without the hassle of readjusting any parameters.
- Each instrument offers a complete set of sound parameters and can use up to 128 velocity-switchable sample layers. Samples may be loaded into the matrix from the desktop or using a dialog box with automatic sample preview function. BATTERY accepts samples of any resolution from 8 to 32 bits, and reads AKAI, SF2, AIFF, WAV and MAP formats. Up to three banks in LM4 format can be imported and used at a time, AKAI sounds can be read directly from CD.

Sound Shaping

Each of the 54 instruments offers a volume envelope (AHDSR), a pitch envelope (DBD), a bit reduction effect and a shaper for controlling the dynamics. All parameter adjustments are calculated and visualized in the wave display in realtime. Each instrument receives MIDI data on a selected MIDI channel and the pitch can be transposed according to the MIDI notes if desired - useful for basses, for example. Every instrument boasts six modulation inputs for which many useful and drum-oriented settings have already been configured as presets.

Samples can be selected either across rows and columns or independently. In addition, individual sounds can be grouped and controlled together. When grouped, all slot settings can be changed simultaneously.



All parameter adjustments, such as tuning, bit-reduction, waveshaping, volume envelope and pitch envelope are calculated in realtime and immediately shown in the wave display.

Every Cell can be routed to an individual output. Battery offers up to 32 outputs, to allow a studio-like workflow.

Every cell offers six modulators that can be routed to many parameters simultaneously. Twist a sample from mild to wild. The included presets give many examples of modulations that can be applied to a wide variety of sample material.

Each instrument has an FX Loop that can be activated to repeat a chosen area of the sample a given number of times. This is very useful for creating modern and innovative rhythmic sound effects, akin to extreme time stretching or granular synthesis. Create everything from repeated syncopation to machine-gun effects.

Tuning, shaping, quantizing will warp the sound to make it stick out from the mix or blend it in perfectly.

KONTAKT



Advanced Software Sampler

One of the most powerful sampler ever created, KONTAKT fuses an innovative design with an advanced sampling engine. The result is an inspiringly fast and intuitively flexible sampler with exceptional sound quality. In addition to supporting all the standard sample playback and manipulation abilities of its hardware and software predecessors, KONTAKT adds several technologies to give sampling a new dimension. Realtime time-stretching and resynthesis, graphical breakpoint envelopes, an integrated loop editor, analog-modelled filters, visually displayed modulation, and breathtaking efficiency create a sampler with the power to realize the most ingenious ideas. An outstanding sample library containing more than three gbs of sounds in various styles and categories is also included.

FEATURES

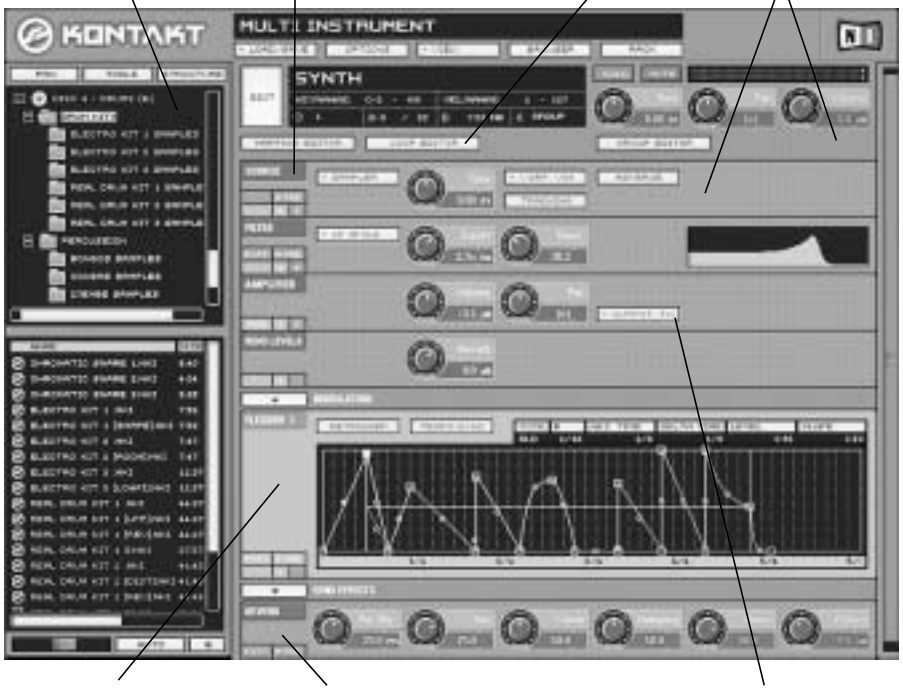
- Advanced design and dynamic resource allocation ensure that the audio engine always runs at optimal efficiency, for up to 256 stereo voices per instance on a standard computer. Filters, effects, and modulations can easily be added with one click.
- Has two special playback modes: Time- Machine mode allows for independent manipulation of length, pitch, and formant of each sample in realtime. Tone-Machine mode imprints a playable pitch onto the sample, regardless of the original pitch (or lack thereof).
- Offers 14 filters from analog lowpass and highpass to exotic sound design filters.
- A broad range of insert and send effects, including EQs, distortion, delays, and reverbs is also at your disposal.
- Nearly all parameters can be dynamically modulated by LFOs, breakpoint envelopes, step-modulators, or via MIDI. Time-based modulation can be synced to song tempo. Dynamic modulation means each parameter displays both current and set position.
- A drag and drop file-browser makes it quick to set up multisamples. Samples can be trimmed and looped within the main sampler window. The built-in graphical loop-editor features seamless looping.
- KONTAKT can import all popular sample formats, plus it comes with 3GB of top-quality samples, including percussion, acoustic pianos, basses, guitars and vintage instruments. The library makes extensive use of KONTAKT's dynamic modulation abilities for patches that sound truly alive.

Supporting drag and drop of multiple samples, the built-in file browser includes a Favorites function for quick access to commonly used folders.

Samples can be played back normally or with an advanced granular resynthesis algorithm for independent control over length, pitch, and formant.

Quickly and visually set sample start, end, and loop points with the integrated loop editor. The loop editor supports up to eight loop points for evolving sounds.

Filters, effects, and modulation are added as needed. Never juggle windows again, as each new element appears in a scrolling "rack" display.



Nearly all of KONTAKT's parameters can be modulated with LFOs, 32-stage breakpoint envelopes, step-modulators, envelope followers, and more. All time-based modulators can be synced to tempo.

In addition to the filter section, each sound can have an insert effect, including EQ, waveshaper, lowfi, stereo enhancer, distortion, or any of KONTAKT's seventeen filters. A variety of send effects are also available, including stereo chorus, delay, flanger, phaser, and reverb.

Each plug-in instance supports up to 32 outputs in supporting host applications, for a nearly limitless number of virtual outputs. The standalone version also supports up to 32 hardware outputs with an appropriate soundcard.

INTAKT



Shuffle the Groove

Intakt is a state-of-the-art sampler specifically designed for rhythmic loop playback, manipulation, and mayhem. Intakt's convenient one-screen interface offers tremendous sound shaping abilities without disrupting the creative flow. Using multiple algorithms, Intakt automatically syncs to tempo, and an outstanding library of loops from Zero-G and East West provides ample source material for any musical style. Intakt also provides a wealth of sound shaping options, including a first-class multimode filter, an envelope follower, two LFOs, effects, and more.



High-end filters and effects

FEATURES

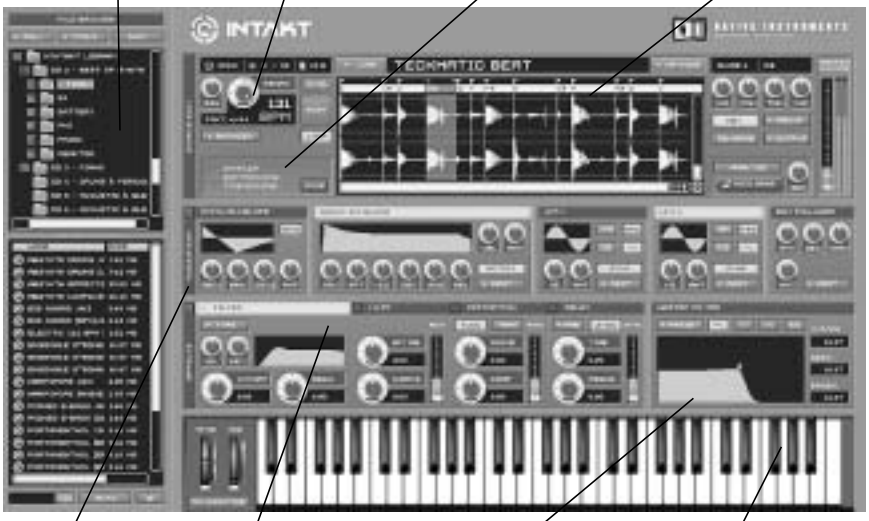
- Intakt features three sample playback algorithms to ensure maximum sound quality for a variety of source material.
- Using an advanced peak-detection algorithm, the Beat Machine divides the imported audio into individual hits. Each hit can have individual settings for pitch, playback direction, pitch envelope, amp envelope, distortion, delay, and more. A MIDI file can be easily exported so that groove, accent, and feel can be manipulated in any host sequencer. The Beat Machine can directly open REX files which already contain marker information.
- Time Machine stretches or compresses the sound in realtime. By first analyzing the source sound, Time Machine dynamically adapts itself to a wide variety of source sounds, from sustained vocals to polyphonic percussion.
- Sampler mode plays back the source sound like a standard sampler, by linking pitch with time. Intakt uses a sophisticated interpolation to ensure the highest sound quality, even with extreme pitch shifts.
- Intakt syncs, stretches, pitches, filters, effects and modulates as well. A powerful sound-shaping filter, flexible envelopes, syncable LFOs, and distortion, delay, and lo-fi effects inject energy and life into any loop.
- Even though Intakt ships with enough loops to produce hundreds of killer tracks, it also imports WAV, AIFF, REX1 and REX2 files, Gigasampler, AKAI, HAlion, EXS, Battery, and Kontakt instruments.

Supports Drag and Drop of samples, instruments. Also includes a Favorites function for quick access to commonly used folders.

Intakt easily syncs loops to any tempo. The groove can also be modified with a Shuffle/Swing function.

Beat Machine analyzes the audio and divides it into distinct hits. Time Machine dynamically time-stretches or compresses the sound, while Sampler links pitch and time.

The pitch, volume, pan, modulation, and effects setting of each slice can be adjusted independently.



Intakt features a comprehensive modulation section with a pitch envelope, AHDSR envelope, an envelope follower, and two LFOs with multiple waveforms.

A filter with 6 different types (2 pole LP, 4 pole LP, 6 pole LP, 2 pole HP, 2 pole band reject, and 2 pole BP), lo-fi, distortion and tempo-synced delay can be flexibly routed.

The Master Filter with lowpass, highpass, bandpass or 3-band EQ puts the finishing touches on the master output.

The keyboard displays the actual sample-mapping layout and makes it easy to audition sounds without a MIDI keyboard. The Input Quantize function ensures that live playing is always intakt.

KOMPAKT



Affordable Software Sampler

KOMPAKT is a streamlined, professional-quality sampler that provides access to literally thousands of sample-based instruments and libraries. Powered by the efficient KONTAKT engine, KOMPAKT combines intuitive handling and a capable architecture with excellent sound quality. The factory library contains hundreds of detailed and convincing pianos, basses, loops, drums, guitars, strings, and more from the outstanding East West and Zero-G libraries. KOMPAKT is the quickest way to build a production-ready instrument collection, and it can also import KONTAKT, GigaSampler, HALion, EXS, Akai, and other common formats.

FEATURES

Professional Sound Shaping

KOMPAKT makes no compromises playing and performing sample-based instruments. Powerful multi-mode filters, envelopes, and LFOs give a wealth of creative possibilities. Integrated high-quality reverb, chorus, and delay effects give each sample life and depth.

KOMPAKT offers professional details allowing for convincing realism of acoustic samples. Direct from Disk (with free downloadble extension) playback lets samples be as large as available hard drive space. Eight-part multitimbrality and 256 voice polyphony allow lush arrangements for top productions.

Outstanding Library

KOMPAKT features an exceptional selection of sounds. Shipping on three CDs, the 2GB library includes acoustic and electric pianos, basses, drums, loops, and guitars. A wide range of orchestral strings and solo instruments mix perfectly with choirs, brass, and percussion. Synths are covered with vintage leads, pads, basses, and much more.

Power and Simplicity

KOMPAKT's clean and intuitive interface doesn't let technology get in the way of creativity. An integrated browser with full drag and drop support makes it easy to find the perfect sound. All important instrument parameters can be immediately adjusted without the need to page,scroll,or fumble through a manual. KOMPAKT puts the world's finest sample-based instruments at your fingertips.

Integrated file browser supports Drag and Drop loading of instruments, samples, multis and includes a Favorites function for quick access to commonly used folders

8 individual instruments with different MIDI, output, and voice settings are combined in a single multi.

Adjustable velocity response accommodates various playing styles and controllers. Microtuning presets tune each instrument to unconventional scales.

This filter has 6 different modes (2-pole LP, 4-pole LP, 6-pole LP, 2-pole HP, 2-pole band reject and 2-pole BP) that can be set individually for each group.



Comprehensive modulation section features three AHDSR envelopes and 4 LFOs (Volume, Pan, Tune, and Filter). All LFO's have a choice of 3 shapes.

Three different effects – chorus, reverb, and delay –offer individual settings for each instrument. The effects include presets for easy access to pre-programmed sounds.

The Master Filter features lowpass, highpass, bandpass or 3-band EQ to finalize the master output.

The keyboard displays the actual sample-mapping layout and makes it easy to audition sounds without a MIDI keyboard.

NATIVE INSTRUMENTS

COMPARISON CHART

	KONTAKT 1.2	KOMPAKT 1.0	INTAKT 1.0
Max. Polyphony/Stereo Voices	256	256	128
Multitimbrality	16 Instruments	8 Instruments	1 Instrument
Groups per Instrument	Up to 4096 (user defined/edited)	Up to 4096 (playback only)	Up to 4096 (playback only)
MIDI Channels	1-16	1-16	1-16
Individual Outputs in Standalone Mode	16	16	16
Individual Outputs in Plug-In Mode	32 mono/ 16 stereo	16 stereo	32 mono
Max. Sample Frequency (host dependent)	192 kHz	192 kHz	192 kHz
Sample Bitrate	8-32 Bit	8-32 Bit	8-32 Bit
Virtual Keyboard	No	Yes	Yes
MIDI Input Quantize	No	No	Yes
MIDI Latch	No	No	Yes
		Sampler Engine	
Time Machine	Yes	No	Yes
Tone Machine	Yes	No	No
Beat Machine	No	No	Yes
Automatic Loop Synchronization	No	No	Yes
Direct From Disk Playback	Yes	Yes	No
Key Switch: Playback/Editing	Yes/Yes	Yes/No	Yes/No
Release Sample Support: Playback/Editing	Yes/Yes	Yes/No	Yes/No
Cycle-Round-Robin: Playback/Editing	Yes/Yes	Yes/No	Yes/No
Group Fading - Playback/Editing	Yes/Yes	Yes/No	Yes/No
Groups - Playback/Editing	Yes/Yes	Yes/Limited	Yes/Limited
		Sample Handling	
Graphic Mapping Editor	Yes	No	No
Keyboard Mapping Editor	No	Yes	Yes
Graphic Loop Editor	Yes	No	Yes
Voice Groups (predefined)	Yes	Yes	Yes
		Sound Shaping	
Modular Architecture	Yes	No	No
MIDI Learn	Yes	No	No
Individual MIDI Remote Controller per Group	Yes	No	No
Assignable External Controllers per Group	32	No	No
Internal Modulation	AHDSR Envelope, DBD Envelope 32-stage Envelope, Step-Modulator Envelope Follower, Glide, LFOs	Volume Envelope, Filter Envelope Free Envelope, Volume LFO Pan LFO, Tune LFO, Filter LFO	Pitch Envelope, Volume Envelope Free Envelope, 2 Free LFOs Envelope Follower
Max. Number of Internal Modulators	16 per Group	7 per Instrument	5 per Instrument
Tempo-Synchronizable Envelopes	Yes	No	No
Tempo-Synchronizable LFOs	Yes	Yes	Yes
Tempo-Synchronizable Step-Modulators	Yes	No	No
Multiple Targets for Internal Modulators (simultaneous)	Yes	No	No
Number of LFO Waveforms	6	3	3
Send Effects	Reverb, Delay, Chorus, Flanger, Phaser	Reverb, Delay, Chorus	Delay
Insert Effects	Distortion, Saturation, Lo-fi, Compressor, Stereo Enhancer, All 17 KONTAKT Filters, EQ	Filter	Lo-fi, Distortion, Filter
Number of Filter Types	17	6	6
Graphically Edited Master Filter	No	Yes	Yes
		Sample Content	
Included Sample Library	> 3 GB/5 CDs	> 2 GB/3 CDs	> 1 GB/2 CDs

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EQUIPMENT LEASING AVAILABLE

DJ LINE



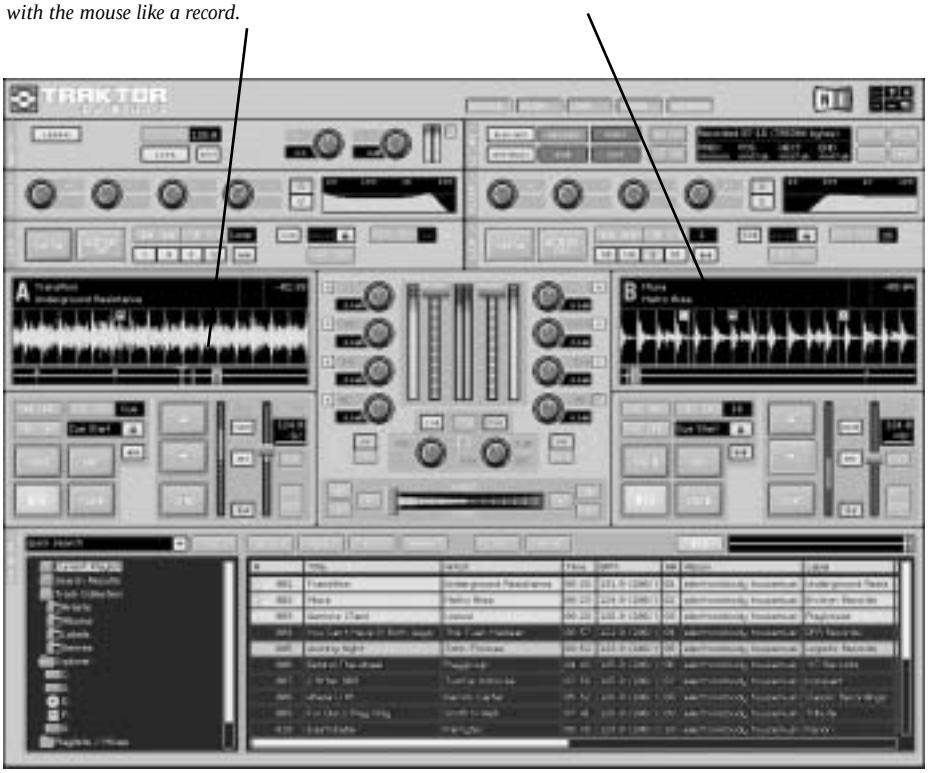
TRAKTOR DJ Studio 2.0

TRAKTOR DJ Studio 2.0 brings professional DJ mixing to the digital domain, unifying listening, interaction, mixing and composition. The modular concept of TRAKTOR DJ Studio 2.0 integrates two high-quality players with a convenient playlist section, DJ mixer, beat matcher, digital scratcher, cue section and filters. The innovative parametric control of the software makes it possible to record all control movements and export a mix in a dedicated file format.
Five cue points for setting loops or triggering the next track can be set and saved for each audio file. The unique interactive waveform display, full MIDI control, mix automation and audio file export make TRAKTOR DJ Studio 2.0 a superior music player.

FEATURES

- All user interactions during a TRAKTOR session can be recorded with sample accurate precision. You can generate a mix as a pure control file without replicating the audio material. Interrupt and resume mixing at any point or edit parts of a mix in a second pass for additional effects. TRAKTOR files contain all information needed to identify the tracks used in the mix and reduce hours of performance to a size perfect for sharing mixes over the internet.
- Realtime waveform displays on both decks allow inspection of the audio ahead of time, graphically showing cue points and detected beat events. The waveform can be manually dragged with the mouse for precise positioning of cue points. For turntablists, TRAKTOR DJ STUDIO 2.0 offers scratching in a most realistic way.
- Up to 10 cue points per track, assignable to trigger special mixing actions
- Up to 10 loops per track including a wide variety of loop setting
- Over 400 TRAKTOR actions and functions can be assigned to keyboard shortcuts or controlled via MIDI
- Powerful database with fast search engine processing ten thousands of tracks in fractions of a second
- DJ STUDIO 2.0 boasts a unique beat recognition and aligning feature. Tempo and phase of two tracks can be automatically matched, allowing the DJ to focus on the artistic aspects of mixing. Phase shift is displayed for easy manual beat matching. Plus, the beat detector makes it possible to set tempo accurate loops on the fly.

TRAKTOR DJ 2.0 is the first DJ player ever to display the running track's waveform showing two separate frequency bands in realtime, thus enabling the user to perceive beats, breaks and instrumentation ahead of time, regardless of whether an MP3-file, WAV-file or Audio-CD is being played. Saved loops and cue-points, as well as a beatgrids, are graphically displayed. The waveform can be moved and stopped to set cue points by dragging it with the mouse like a record.



- Non-destructive automation file recording (Mixfile) with multiple undo function, fast export to WAV or AIFF
- Has a crossfader with selectable sensitivity and a convenient auto-fade feature as well as punch-in, boost and mute buttons.
- A 3-band EQ with 100% kill switches per band for each deck. There is a powerful 3-band limiter for the master signal.
- On top of that an awesome 8-pole bandpass/notch filter allows you to totally impose your own sound on the mix.

REASON



Virtual Software Studio with Analog Synths, Samplers, Drum Machines, Sequencers and Effects

Finally a software synth that is harder than hardware. Reason is a complete studio package with synths, samplers, drum machines, loop players, a sequencer, mixing, effects and much much more. Reason is more than a program — it's a creative revolution. It's changing the way people approach modern computer-based music, blurring the boundaries between music genres. More than an infinitely expandable studio on a CD-ROM, more than analog synths, samplers, drum machines, loop players, effects, and pattern sequencers, Reason is the beginning of a new era of electronic music:

—A self-contained synth studio system, Reason provides an infinite number of great sounding devices with a fast and flexible sequencer.

—As a sub-system synced to your audio sequencer, you can process Reason's audio output with plug-in effects and mix it with your hard disk tracks.

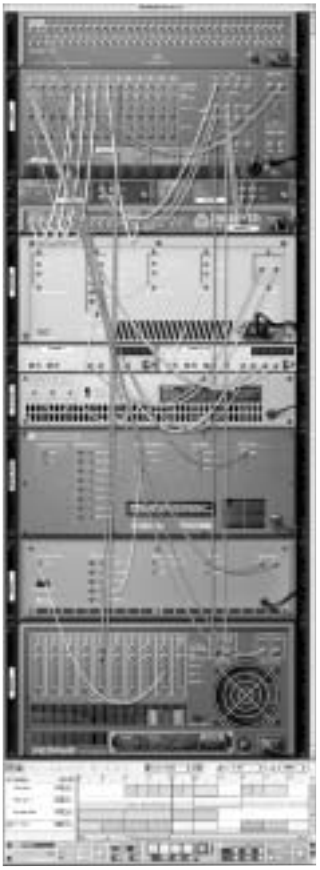
—As a virtual synth rack with your MIDI sequencer, Reason's devices function exactly as their hardware counterparts.

All of Reason's devices have the look and feel of the real thing, but more importantly they have the sound, performance and attitude to rival any hardware out there. Reason hooks up to your MIDI keyboard, giving you MIDI control of all devices, knobs, faders and parameters. So whether you're a producer or a bedroom DJ, a pro or a beginner, sounding this good has never been so easy. All the tools you need to create, edit and mix your music, all within the comfort of your computer.

FEATURES

- Reason is completely native. All sound is generated by your computer's CPU. No special hardware is needed, just an audio output. And ASIO, MME, DirectX and SoundManager support means that Reason is compatible with just about every sound card on the planet.
- AIFF and Wave export
- 32-bit floating point audio; 24-bit playback
- Sample rates up to 96kHz
- 64 audio outputs
- Each unit in Reason's virtual rack is edited from its own on-screen front panel. All the sliders, knobs, buttons and functions of the equivalent hardware are there.
- Choose what you need from the Create menu, and it appears in your rack, logically patched into the signal chain. If you ever wished you had eleven samplers, Reason is definitely for you. And if you run out of mixer channels, just create another mixer.
- When you save your music, your whole studio setup is stored with along with it. Include actual samples, loops and drum kits in the Reason file, for web publishing or e-mail distribution to other users.
- Comprehensive MIDI remote control
- Full synchronization with other MIDI equipment
- Full automation of all fader and control movements
- Zero latency when using Reason with a modern sound card with an effective ASIO driver
- 500MB of high quality samples, REX-files, kits and patches from e-Lab and Dunlab
- Included ReFill Packer utility lets you create your own ReFill sound banks including samples, kits and patches
- Includes a "database" for all types of files (samples, loops, drum kits, synth patches etc.), complete with its own Browser. Finding, auditioning, handling and organizing files has never been easier. A Reason Song file can contain all kinds of music files in a single chunk. Package sounds, loops and patches along with your music, publish the file on the web, and any Reason user in the world can enjoy your music exactly the way you intended it to sound.

A single keypress will turn Reason's rack around, and there you are, in patch cord heaven. Most audio connectors are made automatically. When a new device is created, it appears immediately below the currently selected device, and Reason patches it into the system in the most logical way. Repatch by dragging the patch cord plug to the desired connector, or just make a pop-up menu choice.



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REASON

.MIDI In Device/64-Channel Audio Interface

The hardware interface routes the audio output channels either to your sound card, or to ReWire channels in another program like Cubase and Nuendo, Logic and Digital Performer. The MIDI in device routes 64 channels of incoming MIDI to Reason's devices. With ReWire, you can process Reason's output with plug-in effects and filters and use it exactly like a hardware synth rack.

Mixer

Reason's mixer features 14 channels, each with four Aux sends. Insert effects are handled by patching the effect between the device's output and the mixer channel. If you're running out of channels, simply create another mixer. Chaining is more or less automatic, including sends. The equalizer is simple but effective. For more detail use the Parametric EQ effect.

Effects

Reverb, Delay, Phaser, Chorus/Flanger, Compressor, Parametric EQ, Envelope Filter and Distortion. All are high quality processors with personality, attitude and great sound, featuring parameters that can be modulated via Control Voltage (CV) or MIDI.

Subtractor Polyphonic Synthesizer

A two-oscillator analog synth in the classic mold, Subtractor matches the best hardware on the market. Phase modulation lets you replicate pulse width modulation effects as well as create all-new timbres. FM, ring modulation and two fat, resonant filters are included. Use the Gate and CV ins to connect it to the Matrix pattern sequencer for instant loop satisfaction, complex envelopes and cool LFO effects.

NN19 Digital Sampler

Powerful and easy-to-use, the NN19 does the job fast with top-notch audio quality. Map your own multi-samples with point-and-click ease. Or use one of the multi-samples from Dublab. Includes over 250 different sampler patches, and powerful synth functions provide all you need for creative sound design.

Dr. Rex Rex Player

Based on Propellerhead's groundbreaking ReCycle software, it allows you to do wondrous things to sampled loops. Change the tempo without transposing or timestretching. Pitch and pan individual samples or replace a single sound within a loop. Built-in filtering and modulation functions lets you turn any loop inside out and upside down.



Matrix Pattern Sequencer

A classic monophonic pattern sequencer, just the thing for hypnotic synth loops. Just click in notes and gate levels in the front-panel window, and hit Play. The Gate and twin CV outs can be hooked up to most Reason devices, often in several ways. This, combined with the fact that you can use as many as your computer can run, makes Matrix the ultimate power tool for loop sequencing.

Redrum Durm Machine

Redrum is a recreation of an old-school drum machine with 909-style step-time programming and classic features like shuffle, accent and flam. But there is more. Drum kits are included, but each sound can be replaced. Use Redrum as a passive sound module. Build basic patterns with Redrum's on-board programmer and fills with the Reason sequencer. Or convert Redrum patterns to sequencer data.

Rebirth Input Machine

This device lets you stream 18 channels of audio from ReBirth, right into Reason's mixer and effects.

The Reason Sequencer

Edit songs by drag and drop and copy and paste. You can even cut, copy and paste individual sequence events direct from the arrange view. The Group system lets you handle chunks of music graphically in a flexible, intuitive way. Select a track, and the rack will scroll to the relevant instrument. Move a knob on a device, and the movement will be recorded. Device-specific editors addresses every parameter by name. No need to keep track of anonymous controller numbers. All important features are there: quantize with swing and groove, REX-files as quantize templates, variable time signature, cycle recording, graphic editors, etc.
And because Reason knows what instrument is played for each track, the correct editor is automatically selected when entering the edit view. Key, drum, rex, velocity and controller editors are all included, providing full control over every detail of your MIDI recordings. Change the note length by dragging, draw in controller data or cut, copy and delete. There is even an editor for pattern changes in Redrum and the Matrix sequencer. And all sequencing is done in real-time.

RECYCLE



The Ultimate Tool for Sampled Grooves

Loopists, groovists, samplists! A new world is about to open up before your very ears! ReCycle solves all your groove related problems – and lets you get truly creative in the process.

Loops, grooves and breakbeats: Powerful sonic building blocks, and great inspirational triggers. No matter what style of music you're into, you can be sure there's a loop out there that can spice your track up a little, or lift it to completely new heights.

But handling loops and grooves used to mean hard work. Hours of pitching and stretching just to get a loop to fit your song's tempo and timing. And if you need to change the key, you're in for even more work. In the end, your loops are controlling you. ReCycle changes all that. The ultimate tool for sampled grooves, ReCycle gives you full creative control over your looped material.

FEATURES

What Does ReCycle Do?

ReCycle does more than just solve groove problems and clean up loops – it's a highly creative tool that helps you make the most of your grooves. Simply, ReCycle lets you do with sampled loops what you can do with beats programmed from individual drum sounds – like alter the tempo, or replace sounds and process them individually.

How is it Done?

Start out with a regular audio file or a sample in your sampler, preferably one of a groovy nature. Load the groove into ReCycle, and the program will "look" at the groove, analyze it, and break it up into its rhythmic components. Each part is called a "slice". The process itself is fully automated, but once the slices are there, they are yours to move, monitor or delete, using the on-screen tools and controls. Other tools allow you to set the length, attack and decay of the slices, and to change your grooves' overall tempo or pitch, without one affecting the other.

Then What?

Bring your improved groove into a song. Depending on your equipment and working method, you can either use ReCycle simply as a problem solver for loops (load a drum loop into ReCycle, set a new tempo or pitch, and save the results as a new file), or load up any groove, and use ReCycle's on-screen signal processors: Compressor, EQ and Transient Designer, to give it some punch and distinction. Anything you choose to do can be applied to your loop, and saved as a new file.

To use your loop directly in Propellerhead Reason, Cubase . Logic or other programs supporting REX 2 files, all you need to do is save your sliced-up loop as a REX 2 file and import it onto an Audio Track in your sequencer. The imported loop will play back like the original, but you can change the tempo freely, and you will have full control over the original slices! Silence, move or replace individual hits, change volume and panning – your loop has come to life!

If you're using a sampler (AKAI, SampleCell, EMU etc.), ReCycle creates a soundbank containing the samples/slices, and transmits it to your sampler. ReCycle then creates a MIDI file based on the timing of the original groove. Import the MIDI file into your sequencer, and it will trigger the slices in your sampler, playing back the groove you started out with. Only this time – you make the rules. Quantize it, change the tempo, retune or replace the sounds – total loop control!

ReCycle Up Close



REX 2: ReCycle Export Format

REX 2 is a file format that allows your audio sequencer to play and manipulate ReCycled audio files as if they were MIDI parts. All you need to do is save your sliced-up, imported groove as a REX 2 file and import it onto an audio track in your sequencer or the Dr. REX Player in Reason. Your groove will play back just like the original, but this time, you can change the tempo freely, rearrange the individual sounds, or apply realtime effects to any or all parts of the groove—couldn't be easier or more flexible.

B&H PHOTO - VIDEO - PRO AUDIO
ONE HOUR FREE PARKING
AT 349 W. 34th STREET (with purchase of $100 or more)

REBIRTH RB-338



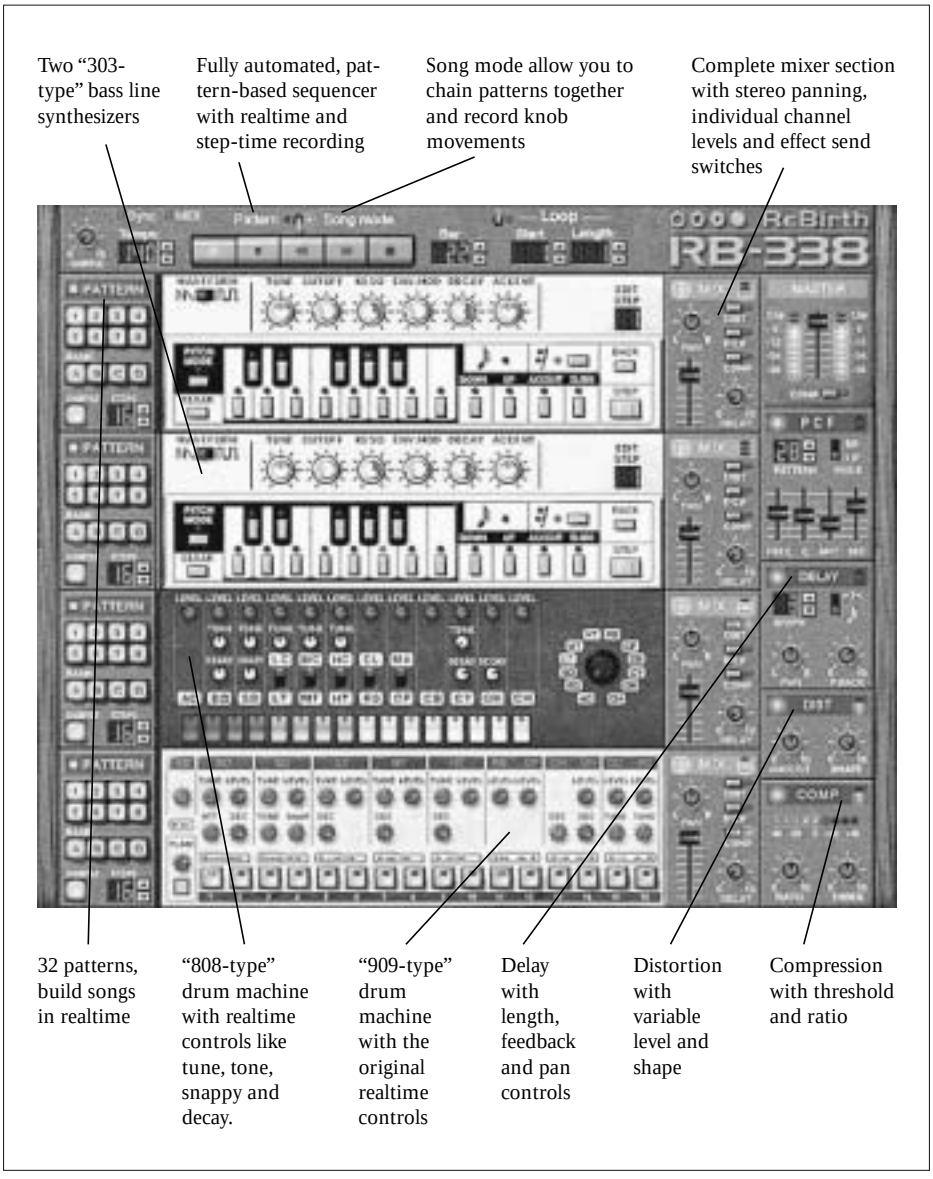
Bassline Synths and Drum Machines

ReBirth is a stunningly accurate reproduction of the bare essentials of techno music making. Silver boxes, blinking LED's, a myriad of knobs - and a sound like nothing else on this planet. A software reincarnation of two analog bassline synths (TB-303), and two classic drum machines (TR-808 and the TR-909), ReBirth also offers a digital delay, a quad-input distortion unit, a compressor and an analog filter emulation unit, the PCF (Pattern Controlled Filter). All the quirks and subtle qualities of analog, combined with the convenience of modern computers (a minimum of cables, integration with your sequencer software, complete front panel automation, real-time audio streaming and much more).

Here's how they did it: Instead of sampling the sound, they "sampled" the actual synth hardware. It was a question of analyzing the mysterious inner workings of the analog design, and then creating at mathematical model of it, incorporating practically every nuance. Then they converted this theoretical model into lightning fast computer code. So fast that it runs on the computer you already have. No customized cards, DSP chips or other expensive accessories are required.

Just as important as the sound, is the way the sequencer controls the sound. To get the unique character you need access to all programming functions of the analog original: Accent and Slide on the bass lines, Snappy, Decay and Tone on the drums. The subtle shift in character from beat to beat. ReBirth, integrates a vintage sequencer section and knob-loaded front panel into the machinery.

- All control work in realtime and in the same way as on the real machines
- Pattern and song editing functions like copy and paste, transpose, shift, random
- Export songs as AIFF or .WAV files
- MIDI clock sync to external hardware or sequencer software plus MIDI control of all front panel knobs
- User modifications for the program are your chance to get seriously creative: Dig right into the code and change the graphics, exchange the durm sounds— even create your personalized ReBirth.
- Total integration with Reason, Logic, Digital Performer and Cubase/Nuendo via ReWire. Full sync and realtime streaming of up to 18 individual audio channels for post-processing through effect plug-ins.



PRO SESSIONS

Sound Library



ProSessions is a fresh new collection of affordable top-quality samples created by some of today's top musicians, composers and producers. The ProSessions library delivers inspiration across a wide variety of genres and styles including hip-hop, dance, pop, Latin, R&B, rap, drum & bass, techno, world beat and more. Designed with songwriters, producers and remixers in mind, each ProSessions sample features a top-quality recording, meticulously edited by hand for rock-solid professional timing and easy integration into other grooves. Each CD delivers over 400MB of world-class samples in multiple formats for today's hottest Mac and PC software—AIFF, WAV, REX2 and Acidized WAV. The initial CD-ROM titles in the ProSessions Sound Library include:

Alex Tinsley Electro Crash: Mixed blend of big-hair 80s attitude remixed with today's hot sounds

Bobby Wilks Vector Field: Dangerous synths meet melodic outcries, twisted nature and psychoacoustics

Chris Falson Worldbeat Café: Full- and partial-mix grooves drawing from popular Latin percussion styles

Chris Falson These Drums Are Loud: Natural pop/rock drums recorded and edited in sections with songwriting in mind

Discrete Drums World Rock & Percussion: Solid rock song sections from starmaker producers and engineers

Discrete Drums R&B Drums & Percussion: Great collection of laid-back funky grooves presented in song sections

DJ Logickal Sounds Logickal: A deviant library of raw, experimental abused synths and samples

Diavola Mechanically Separated: Reality is stranger than fiction with this edgy, moody industrial aberration

James Galvez Latin Street: Deep funky vibe packed out with pure Latin hip-hop and R&B nastiness

James Galvez Latin Element: Sizzling Latin loops of authentic instruments and rhythms

Neldome St. Hydrosonix Disc 1: Incredibly diverse CD featuring a progressive mix of hip-hop, trip-hop and R&B

Neldome St. Hydrosonix Disc 2: Dives even deeper into the world of trippin', hippin' and skippin'

3RDedree Productions Hella Bumps Disc 1: Fresh booty from the best of the hip-hop and rap skools

3RDedree Productions Hella Bumps Disc 2: Who says you can get too much of good thing?

Strings Refills by Propellerhead



Strings is a high quality collection of live played strings turned into Dr Rex loops and NN19 sampler patches. This ReFill adds new flavors to the already powerful Reason program with both solo instruments and ensemble material. All string phrases have been edited in Propellerheads ReCycle program, which means that both tempo and pitch can be changed seamlessly for each loop. The ReCycle Rex file format also allows complete reorganization of the music phrases in Reasons Dr.Rex editor, all in total real-time. Materials include:

• 35 Orchestra Strings Dr.Rex Loops • 40 Full String Quartet Dr.Rex Loops • 80 Solo Violin Dr.Rex Loops • 40 Solo Viola Dr.Rex Loops • 40 Solo Cello Dr.Rex Loops • 48 Upright Bass Dr.Rex Loops • 5 Harp Dr.Rex Loops

Strings also contain patches for Reasons NN19 sampler featuring the following base samples

• Solo violin • Violin ensemble tremolo • Solo Violin Pizzicato • Violin Ensemble Pizzicato • Solo viola • Viola ensemble tremolo • Solo Viola Pizzicato • Viola Ensemble Pizzicato • Solo Cello • Cello ensemble tremolo • Solo Cello Pizzicato • Cello Ensemble Pizzicato • ContraBass ensemble tremolo • ContraBass Pizzicato • ContraBass Ensemble Pizzicato • Ensemble (Cello, Viola, Violin) • and finally some harp riffs

MAGENTA

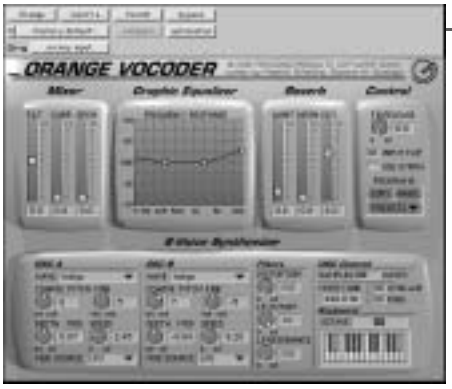
Realtime Resynthesis/Sound Manipulation Plug-in (VST Mac)



Magenta is an audio effects plug that uses any VST track or the live input as basic waveform that can be tweaked using the incredible power of realtime resynthesis—the most versatile and powerful synthesis technology available. Prosoniq's proprietary Multikernel Spectrum Resynthesis Engine (MSRE) reproduces any sound by resynthesizing it in real time, giving you utmost control over all aspects of the recorded instrument or music track. Design your own sounds without the limitations of traditional sound synthesis techniques.

Play and control multi-voiced musically monophonic and polyphonic instruments directly from any MIDI keyboard in real time while adding a wealth of spectral effects that are fully automatable and can be dynamically controlled. Produce vocal doubling, to create additional voices or melodic choruses from single vocal lines. Tweak sounds to create alien atmospheres, clusters or textures that have nothing in common with the original sound.

- Real time resynthesis for extensive sound processing
- Works on any input signal in real time
- Use as software synthesizer with live audio input
- Pitch shifting and versatile spectrum filtering capabilities
- LFO control over all important parameters
- 2D graphic control over filtering and spectral manipulations
- All key parameters can be controlled over definable MIDI controllers
- Formant preservation during pitch shift
- Over 10 spectral macro filters for all sound design situations
- Directly playable over MIDI keyboard
- Free pitch, pitch quantize and pitch lock capabilities to correct the input pitch or to create mouth tube or vocoder fx
- Built-in reverb effect
- Create vocal accompaniments by playing harmonies on your MIDI keyboard
- Change the timbre of any sound
- Create mouth tube and vocoder fx
- Pitch shift musically monophonic and polyphonic sounds
- Create yet unheard effects by tweaking the harmonic developments of sounds
- Use any VST track as oscillator for an incredibly versatile synthesizer!



ORANGE VOCODER

Realtime Vocoder (VST Mac and PC)

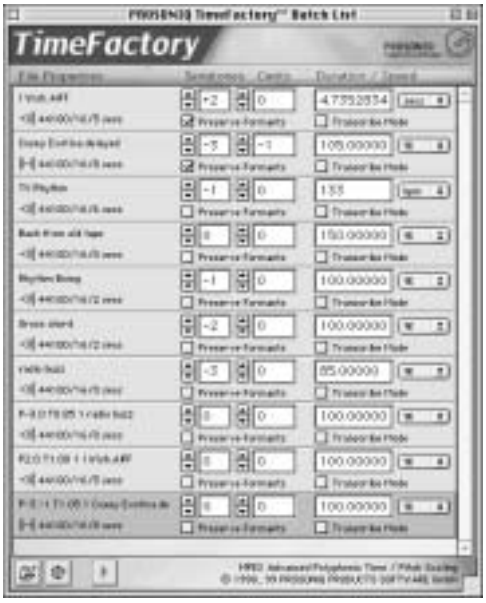
Perhaps the fastest, most flexible and most transparent sounding simulation of an analog vocoder effect as a VST plug-in. Following the long tradition of German Vocoder manufacturers, Prosoniq offers you an all-digital simulation of a realistic analog vocoder effect that is fully customizable and comes with an 8-voice virtual analog synthesizer unit, Freeform EQ and Filterbank Reverb, all in one plug in.

- Realtime Vocoder effect usable in either the Master or Track Effect Racks
- Uses either any input source/VST audio track or the built in oscillator as carrier signal, and any VST audio track as modulator
- Full MIDI controllable through the VST standard
- Post-Vocoder Filterbank Reverb Effect with Mix, Decay and Density
- Fully customizable Freeform EQ with max. 12 nodes, ranging from 20Hz to 20kHz and resolving from +30 to -30 dB FS
- Input Channel flip exchanges carrier and modulator signals without re-routing tracks
- Integrated 8-voice virtual analog synthesizer with 2 oscillators per voice featuring 10 basic wavefoms and 7 sampled sounds, voice detune, pitch LFO, 4-pole Lowpass filter with cutoff and resonance, oscillator hard sync and ring modulator.

TIMEFACTORY

Time Stretching/Pitch Shifting Software

TimeFactory is a powerful tool dedicated to performing high quality, near-lossless time and pitch shifting on your final mix or other audio material. Using the latest in DSP technology, TimeFactory achieves time scaling amounts of up to one third the original length, corresponding to a time stretching factor of 133% and more (or pitch scaling of up to 5 semitones) without audible loss in quality. This is as much as slowing down a complete music piece from 120 BPM to 90 BPM, or vice-versa. Conventional methods used in other software either suffer from unpleasant harmonic distortion, timing inconsistencies and echoes even at small time scaling ratios, or introduce severe smearing or reverberation that makes the results unacceptable for professional audio purposes when applied to complete musical pieces. With TimeFactory, you are free to choose the tempo of your recording at a result that is indistinguishable from the original within a large range of scaling ratios, without endlessly tweaking dozens of parameters. Whether you wish to time scale individual sound files of a recording session or the final mixdown, whether you are processing classical or 'drum & bass' style music, TimeFactory is a simple click-and-go application for Mac and PC with straightforward user interface and easy to use batch processing capabilities.



- Accepts sound files in AIFF, SDII (Mac only) and WAVE file formats of any word length and sample rate, even 24-bit/96kHz or split stereo files are no problem.
- Works with any material of any genre, be it classic, pop, rock, techno or other
- Proprietary timbre Wizard algorithm allows for natural pitch shifts
- No more worrying about the outcome of time consuming parameter tweaking sessions, just set the desired time scaling ratio and click to process your audio files within minutes.
- No timing inconsistencies
- Input speed in new length, BPM or %
- Double click to edit any soundfile in your favorite editing program
- Transcribe Mode for crystal clear 200% time scaling of complete songs
- Pitch Shifting including formant correction
- Batch Processing
- High quality resampling with anti-aliasing for pitch shifting
- No sound coloration, no phasing with stereo files
- MPEX Time Scaling technology for best results at the first mouse click
- Mac and PC version in a single box - no worrying about switching platforms. Also includes SonicWORX Essential audio editing software (for Mac).

PROSONIQ VST BUNDLE



The Prosoniq VST plug-in bundle combines many advanced sound design tools with professional plug in products for mastering and post production, all in one package. Whether you are processing individual tracks like vocals, guitar or synth or are looking for professional tools to improve your final mix, the Prosoniq VST plug-in bundle leaves nothing to desire. In fact, once you've used it, none of your tracks will ever go unprocessed!

- Ambisone places your tracks in virtual 3D sound space without requiring costly encoders or playback equipment.
- VoxCiter adds the warmth and clarity of an analog recording to your vocal or instrumental tracks.
- Includes Roomulator, a powerful reverb toolkit with 5 distinct reverb algorithms well suited for all studio application
- Dynasone with its powerful Maximizer/Multiband Compressor gives your mix that final touch and turns any music track into a CD-ready master.
- PiWarp is an indispensable tool if you're after special effects that are both unique and make your tracks easily recognizable.

PROSONIQ VST BUNDLE, *continued*

Ambisone: 3D Positioning



Mix your VST tracks in full 3D stereo. This includes placing sound sources outside or above the speakers to provide your mix with a previously unthinkable width and depth. And Prosoniq's Virtual Scenery Modeling approach allows for 3D placement when monitoring the mix through headphones as well as through speakers.

Includes modulation sources to automate 3D placement allowing for sound sources that rotate around your head on a single mouse click. The flexible and intuitive interface lets you adjust the different parameters while monitoring both position and elevation in real time.

Roomulator: Reverb Kit



Optimized for different size ambiences, the Roomulator toolkit provides highly flexible, natural sounding reverb which meets the highest requirements of today's studio standard. Dense Hall Reverb with canyon-like decay times as well as simulating narrow spaces or closeup situations in utmost quality and brilliance are provided to all your recordings with this all-in-one reverb effects kit. The intuitive 2D manipulation capabilities let you easily adjust the overall room timbre while listening to the results in real time. No matter whether you are recording guitar, acoustic instruments or voice, this powerful tool will place your instruments to pleasantly integrate in a clear and brilliant mixdown.

VoxCiter: Vocal/Single Instrument Track Toolkit

Add natural brilliance to vocal recordings, refreshing and even restoring the quality and fidelity of any kind of instrumental of vocal material. It is great for finalizing vocal recordings or any instrumental track and will bring old recordings back to life again.

Use it to add that 'pro' sound to your vocal recordings and give your audio tracks a touch of life that got lost during the process of sampling and intense post processing.



- Flexible Noise Gate with Attack, Hold and Release
- Dual Vocal parametric EQ (Low Pass, High Pass, Band Pass and Band Reject selectable)
- Realtime Neural Automatic de-esser circuitry
- Vocal Compressor/Expander with Attack/Release/Auto features
- Independently equalizable Sidechain feed for Compressor/Expander
- Vocal Refresh
- Vocal Saturation (simulates tube distortion to add warmth)
- On-board Bypass switch for instant comparison without having to switch windows

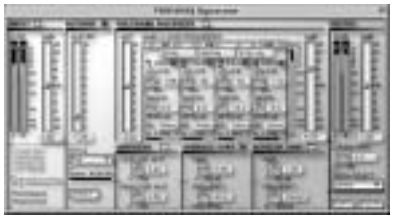
PiWarp: Frequency Warping



Create interesting special effects and atmospheres. Used with strings and sustained sounds it creates dense ambient atmospheres reminiscent of alien worlds or space ship ambiances that let you create sonic landscapes for film or TV in an instant. Used with dry vocal and drum sounds it creates interesting counter-melodies and alien voice effects that are unique and useful for a wide variety of experimental and contemporary music.

- Warp Factor parameter sets the frequency warping amount
- Tune Factor sets the basic pitch of the output
- Filter parameter filters the output with a 6dB/Oct. Low Pass Filter
- Reverb Amount and Room Size parameters included
- True stereo processing

Dynasone: Mastering/Post Production Tools



Powerful processing functions provide your final mixdown with that special 'pro' sound. Neural Automix function adjusts the overall sound of your mix to sound both transparent and have an optimal frequency response. Dynasone analyzes your work and proposes a setting for optimum results.

Different templates let you adjust your overall sound to match any desired musical style, from Classic to Dancefloor. What's more, you can even choose an optimization scheme for your final release medium. Select 'Vinyl', 'Tape', 'CD' or 'Linear'. Multiband maximizer and peakstop limiting functions optimize the loudness of your mix to match any desired dynamic range and for optimum punch and brilliance. Subsonic and Refresh functions add new spectral contents allowing for restoring and optimizing older recordings.

SERATO

PITCH 'N TIME2

COMPUTER SOFTWARE

1112

Time Stretching and Pitch Shifting

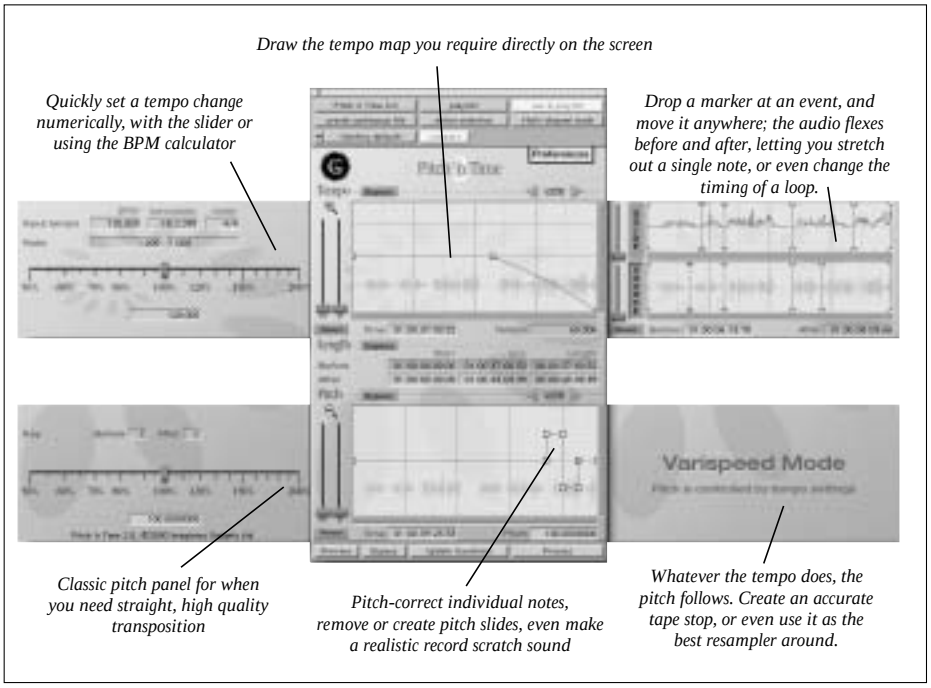


Pitch 'n Time 2 is a high quality time stretching and pitch-shifting AudioSuite software plug-in that will work with all versions of Pro Tools on the Mac and PC, from Pro Tools free to Pro Tools HD.

Traditionally pitch/time manipulating application used math to either remove or add samples, or change sample rates to achieve the desired result. Some program material would respond well, while others had artifacts introduced— rendering the processed files useless. Serato's decided that the output of the process must sound the same as the input, just faster or slower. In other words the problem isn't about waveforms or kilohertz - it's all about hearing. So Serato developed a sophisticated model of the human auditory system that "listens" to the music, performing a sophisticated auditory scene analysis and deciding how best to treat the source material.The ranges of transformation within Pitch 'n Time 2 are 50% to 200% for time compression / expansion, and ± 12 semitones for pitch shifting. Both functions each have three interchangeable panels, ranging from simple fixed ratios to complex tempo and pitch alterations that vary over time.

FEATURES

- ◆ Modify tempo from 50% to 200% of original and simultaneously pitch shift by ±12 semitones:
  - – Time stretch by tempo change, length change, target length, or target BPM
  - – Pitch shift by frequency change or semitone shift.
  - – Preview changes in real time
- ◆ Integrates perfectly with Pro Tools (Mac and Windows) as well as other DAWs and applications that support AudioSuite.
- ◆ Works with all bit depths, and sample rates and audio file formats available within an AudioSuite compatible application — can even be used to change the sample rate of your sample without losing quality.
- ◆ Preserves sample accurate timing while at the same time maintaining sonic integrity
- ◆ Works as well on solo instruments and drums as it does on a complete mix.
- ◆ Process stereo tracks without phasing
- ◆ Multi-Channel Mode - allows you to process up to 48 tracks together while maintaining their original phase coherency.
- ◆ Time-Morph allows you to drop a marker at an event and move it anywhere; the audio flexes before and after to allow you to stretch out a single note, or even change the timing of a loop.



- ◆ Variable Pitch Mapping lets you pitch-correct individual notes, remove or create pitch slides, or even make a realistic record scratch sound.
- ◆ Variable Tempo Mapping gives you the power to create variable tempo maps and changes over the length of your sample.
- ◆ Waveform Overview means you can actually see the effects of your tempo map as it is applied directly to the waveform.
- ◆ Supports all SMPTE time code formats: 30 fps, 29.97 fps (Drop/Non-Drop Frame), 25 fps (EBU) and 24 fp. Just select your desired end time position time code, and it calculates the required time stretch.
- ◆ Time Stretching from one Time Code frame rate to another.
- ◆ A quick and easy BPM calculator for loops determines the tempo of a selection against the number of bars and beats selected

SCRATCH – STUDIO EDITION

Serato Scratch - studio edition RTAS & HTDM Scratch any digital sample or sound file in Pro Tools using your existing turntables or mouse as the controller. Serato Scratch is a software plug-in from the company that brought you the multi award winning plug-in, Serato Pitch 'n Time. Serato Scratch - studio edition brings DJ scratching directly into your studio like never before. Scratch allows you to scratch any digital sample or sound file on your computer using your existing turntables or mouse as the controller. Special vinyl included with Scratch contains a control signal that allows your computer to track the motion of the record, simulating the same movement within the digital sample. The result is a feel and sound that is indistinguishable from scratching vinyl. No more record burn and no more dub plates!

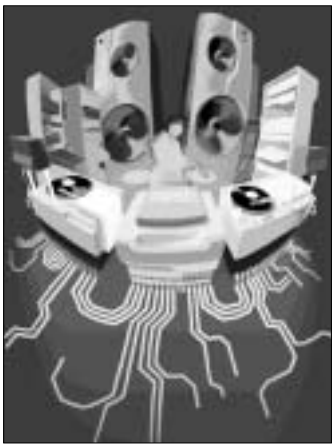


- Scratch any digital sample or sound file on your computer using your existing turntables or mouse as the controller
- Record mic input and scratch on the fly
- A full compliment of standard turntable controls are provided including reverse play and pitch control
- Available as a MAS plug-in on the Mac and as RTAS and HTDM plug-ins for Pro Tools TDM including HD, and Pro Tools LE - Mac OS & Windows

How It Works

- A special vinyl record included with Scratch contains a control signal that allows your computer to track the motion of the record, simulating the same movement within the digital sample
- The control signal output from the turntables are routed into the software by plugging them directly into your audio interface
- The selected audio sample is then routed to the DJ mixer for cross fading, cutting etc
- Audio from the DJ mixer then goes to your speakers, or back to the computer for recording
- Feels and sounds just as though you are scratching a real record – latency is limited primarily by your I/O hardware's buffer size, converter latency, and session sample rate. For example, a Pro Tools system configured with a 128 sample buffer results in a total latency of 9ms when operating at a 44.1 kHz sample rate
- The package includes two vinyl control records and the software CD.
- Replacement records are available for $10

SCRATCH – Live



Serato Scratch Live is a stand alone version of the ground-breaking software plug-in Scratch SE that allows you to scratch and mix your mp3's and digital sound files using your existing turntables and mixer as the controller.

- A stand alone version of Scratch SE for Mac OS 9 & X and Windows XP.
- Requires a Mac or PC computer, up to two turntables and a DJ mixer
- The package includes and two vinyl control records, software CD and a USB interface to connect your turntables to your computer.
- Scratch and mix mp3, aif, wav, ogg and CD sound files using your existing turntables and mixer as the controller.
- A 'feel' and sound that is virtually indistinguishable from scratching vinyl.
- The package includes two vinyl control records, a compact portable USB interface for 2 turntables and a microphone input.
- Record mic input and scratch live audio on the fly.
- Mix multiple tracks live from the same CD.
- Needle dropping - cue the way you are used to.

SONIC FOUNDRY

SOUND FORGE 6.0



Professional Digital Audio Recording, Editing and Mastering Software for Windows

The industry-standard for audio recording, editing, effects processing and streaming media creation, Sound Forge is used worldwide for production in leading recording studios, post-production houses, broadcast facilities, radio stations and home studios. It includes over 35 realtime audio effects and processes with more than 200 presets for manipulating audio, and supports a wide range of audio and video file formats, including Windows Media, RealMedia, QuickTime and MPEG 1&2. Version 6.0 expands on a decade of leadership by adding features that make it faster and more powerful than ever.

From non-destructive editing to new formats for import and rendering, Sound Forge 6.0 provides the most intuitive and efficient digital audio editing environment available for media professionals worldwide. Features include realtime non-destructive editing, modeless audio plug-in chainer, multitask background rendering, 32-bit/64-bit float/192kHz file support, enhanced time zoom, fully customizable tool bars, tabbed docking windows and DirectX plug-in manager.

FEATURES

Fast, Powerful Audio Editing

Sound Forge provides the power and flexibility you need to create a clean, professional final product—cut, paste, and delete audio with extreme speed and accuracy — down to the sample level. Sound Forge 6.0 combines fast and powerful non-destructive editing with simple drag and drop operation, allowing you to focus on your project and not time. And a fully customizable interface allows you to build projects the way you want. Also supports standard Windows keyboard commands and mouse shortcuts.

- Edit, record, process, and encode audio on your PC
- Cut, paste, move, mix, delete, mute, reverse, fade, and cross fade audio
- Drop markers during playback or while editing. Create custom marker names and use them for creating regions and loops.
- Edit files down to the sample level. Sound Forge has a 24:1 zoom ratio, allowing you to perform more precise, sample accurate editing. View, select, edit, and snap to sample selections. You have more control over your audio editing than ever before.
- Import and save 32-bit files (record them if your hardware supports 32-bit recording)
- Multitask background rendering lets you work on one file while Sound Forge processes another—saving you valuable production time. Open, play, preview, cut, copy, paste, and delete files while other project files render in the background.
- Customize your toolbars to create the interface you want for quick access to commonly used functions.
- Dock floating windows, including, regions list, playlist, keyboard, play meters, and time display, individually or on top of each other.
- Customize your toolbars to create the interface you want for quick access to commonly used functions.
- Use an unlimited number of undos/redos to restore a file to any previously edited state.
- Synchronize audio and video, frame by frame for streaming media productions and multimedia presentations.
- Create and edit ACID loops. Permanently assign root notes, number of beats, and tempo to ACIDize new loops.
- Trigger audio playback via MIDI commands from any MIDI software or hardware device.
- Use the Spectrum Analysis tool to analyze waveforms by frequency and help identify noise problems.
- Use the Preset Manager to back up, transfer, and organize your favorite presets (even ACID presets). Save time in accessing most commonly used settings.

Audio Effects and Processing

- Choose from 35 audio effects (including Amplitude Modulation, Chorus, Dynamics, Delay/Echo, Flange/Wah-Wah, Distortion, Envelopes, Gapper/Snipper, Noise Gate, Pitch Bend/Shift, Reverb, Vibrato and more) and processes with over 200 presets, including 18 DirectXAudio Plug-Ins from Sonic Foundry's XFX1, 2 and 3 series.
- Use the Audio Plug-In Chainer to chain multiple plug-ins together and hear the results immediately.
- Use the DirectX Plug-In Manager to customize folders and rename audio plug-ins.
- Apply acoustic responses of real environments such as concert halls, caves and stairways with the Acoustic Mirror plug-in.
- Wave Hammer mastering tool limits dynamic range, adds punch to recordings, and boosts signal levels without clipping
- Audition effects before applying them and make "before and after" comparisons
- Use Time Compress/Expand to easily turn 33 seconds of narration into a 30-second spot without altering the pitch.
- Use the Spectrum Analysis tool to perform precise FFT analysis, and display resulting data in two graphical formats. Use the Spectral Graph to monitor input or playback in real-time.

SOUND FORGE 6.0

Protected Work Environment

Sound Forge includes crash recovery tools, which allow you to recover lost work after a power failure or system crash. Also includes a powerful undo/redo history feature, which allows you to see your entire work history at a glance and undo edits.

Powerful Encoding Tools

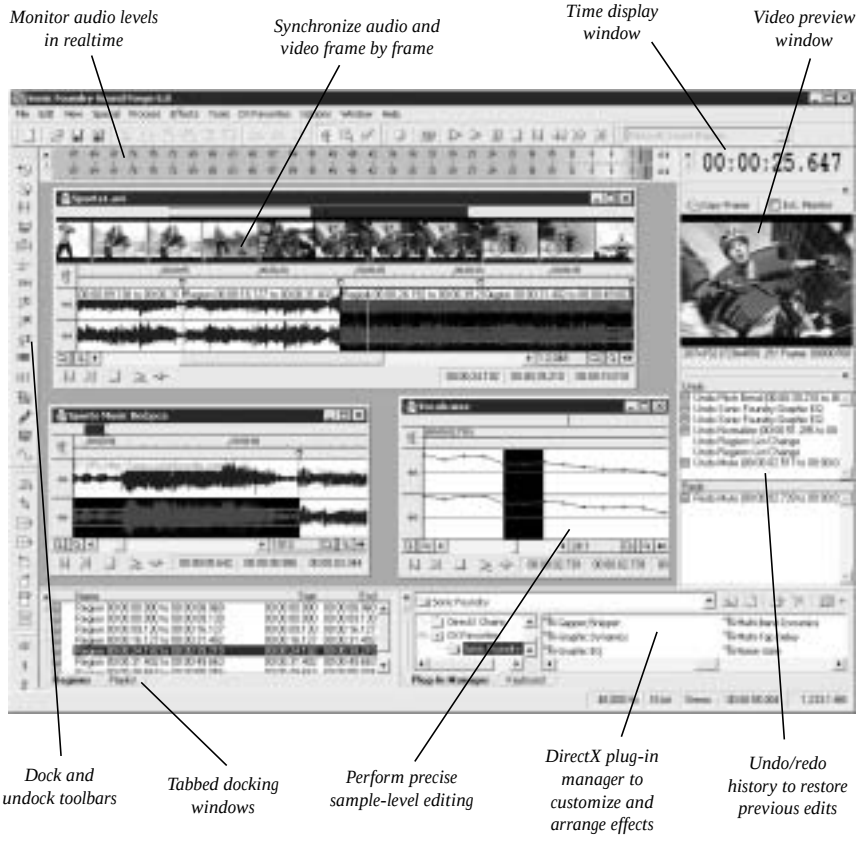
- Encode your audio files for the Internet in RealAudio, Windows Media, and MP3 formats. You can even encode video, insert metadata command markers, launch web sites, display captions, and embed URL flips into your audio and video streams.
- Render audio to just about any file format including WAV, ASF, RM, MP3, WMA, and AVI, eliminating the need for another program to perform file conversions.
- Sonic Foundry's Batch Converter 5.0 utility allows you to apply the same process across multiple files automatically

CD Features

- Rip audio from CDs, transferring music to your hard drive, and save files in MP3, WAV, and WMA formats.
- Create your own audio CDs. Sound Forge includes track-at-once CD burning, allowing you to take your audio projects with you and share them with the world.

Video Features

- Video render options include fast video resizing, source video resampling, and video stretching.
- Sync audio with video at sub-frame accuracy. With external monitor previewing through IEEE1394 devices and more render options than ever, Sound Forge is the perfect companion to your video editor.
- If you're new to the world of digital video, Sonic Foundry's bundled Vegas Video LE 3.x provides the perfect introduction to professional DV and audio production.
- A professional digital video and audio multitrack editor, Vegas Video LE 3.x lets you add up to eight tracks of video, apply effects and transitions, and time-stretch audio and video. Perform edits in real-time and hear the results immediately.



Sound Forge Studio

Incorporating much of the award-winning technology found in Sound Forge, this streamlined version makes it easy to record, edit, and process audio using your PC. Record from a CD, microphone, LP, cassette, or musical instrument, and then edit your recording using a wide range of tools and effects. Liven up presentations, recordings, and music using over 30 digital audio effects and processes, including 1001 Sound Effects. Create your own "Best Of" audio CDs using track-at-once CD burning.

- Edit files in real-time and hear the changes immediately. Sound Forge Studio includes dozens of editing tools including: cut, paste, move, delete, mute, reverse, crossfade, trim, normalize, fade, resample, pan, enhance, insert silence, and more.
- Studio supports many popular audio and web formats, including WAV, Windows Media Audio and Video, QuickTime, RealAudio, and MP3. It is perfect for creating streaming media for the web.
- Sound Forge Studio includes a 10-band EQ, reverb, delay, distortion, mute, reverse, fades, crossfade, pan, volume, normalize, and more.
- Free sound effects library lets you enhance your audio with over 1,000 professional-quality sound clips - from explosions and flying saucers to animals and spooky noises. This royalty-free library is perfect for presentations, PowerPoint productions, and web sites.

ACID PRO 4.0



Loop-Based Music Creation Tool for Windows

ACID PRO is the often-imitated original loop-based music creation tool for the PC. This powerful award-winning tool allows you to create and produce royalty-free music for audio production, multimedia projects, broadcast music beds, Web sites, Flash animations and DV scoring. Bundled with over 600 music loops, you'll be creating royalty-free music in minutes. Though incredibly easy to use, ACID PRO takes no shortcuts when it comes to providing the depth that advanced users require. From unlimited tracks of audio and MIDI events, DirectX plug-in support, CD ripping/burning, streaming media creation, and support for 10 popular audio formats, no program is better than ACID at matching the tempos and keys of ACIDized WAV files and songs for loop-based music construction.

FEATURES

Editing

Create original, royalty-free music in minutes. Easy to use, you just pick the loops you like, paint them on a track, play your results and enjoy the music. Work with hundreds of multi-genre loops. Use unlimited tracks of audio. Import audio samples and complete songs, even MP3s. Audition loops before you add them to your project, automatically matching up tempo and pitch in real-time. And ACID automatically saves your work so can recover your work after a crash.

- Record music and vocals. Import audio tracks from CDs and control the volume, panning, and effects for each track.
- Use as many loops and MIDI tracks as you like with unlimited tracks.
- Restore projects to any previous state using unlimited undo / redo history
- Edit multiple tracks at once with ripple editing
- Perform tempo and key mapping
- Make tempo and key changes over time
- Insert volume and pan envelopes
- Mix several tracks into a new one (track bouncing)
- Create custom remixes of your favorite songs using the Beatmapper. A remixer's dream, Beatmapper helps you add tempo information to song-length files so you create your own remixes and combine music loops with songs. It's simple. Just import a song, add some beats and give it a powerful new edge. You can even change the tempo of your song without affecting the pitch.
- Time stretch and pitch shift Beatmapped tracks more efficiently with minimal echo or artifacts. ACID PRO 4.0 improves the playback of your time-stretched tracks with less stuttering, and provides you with much more detail in your Beatmapped projects.
- Drop markers in real-time along the timeline
- Modify volume, effects, and panning at the sub-group level using the new bus tracks in ACID PRO 4.0. Use these new bus envelopes to set track effect parameters, volume, and panning for your entire ACID project.
- Produce amazing drum fills, stutters, and DJ-style effects with the Chopper. This amazing tool allows you to select a portion of an audio loop and place it on a track as an event. Simply create a "chopped" event, place it on the timeline, and let your imagination run wild. All "chopped" events play in sync with your music.
- Score original music alongside your video to create the ultimate video production. With ACID's seamless ability to match pitch and tempo, scoring video has never been easier.
- Synchronize audio and video files at the frame level
- Select time signatures anywhere from 2/4 to 9/8, or customize your own settings anywhere from 1/1 to 99/32. Set tempo, key, and time signature markers in your projects, and make adjustments to the beat ruler during playback. Use the included odd-timed loops or create your own. Explore and expand syncopated beats and odd times used in Jazz, Latin, and other music styles.

Effects Processing

Add finishing touches to your projects with 20 DirectX Audio plug-ins from Sonic Foundry's XFX 1, XFX 2 and XFX 3. Alter effects over time using volume and track envelopes. Apply fades and transitions. Even apply effects on tracks, busses or in assignable effects chains.

- DirectX Audio plug-ins include Amplitude Modulation, Chorus, Delay, Distortion, Flange, Noise Gate, Reverb and more.
- ACID PRO 4.0 also includes three EQs—Graphic, Parametric, and Paragraphic and supports third party DirectX plug-ins.
- Create resonant sweeps, dramatic fades, and EQ changes using automation parameters and envelope control.
- Audition effects in real-time and tweak the results during playback
- Apply effects on tracks, busses, or as assignable effects chains

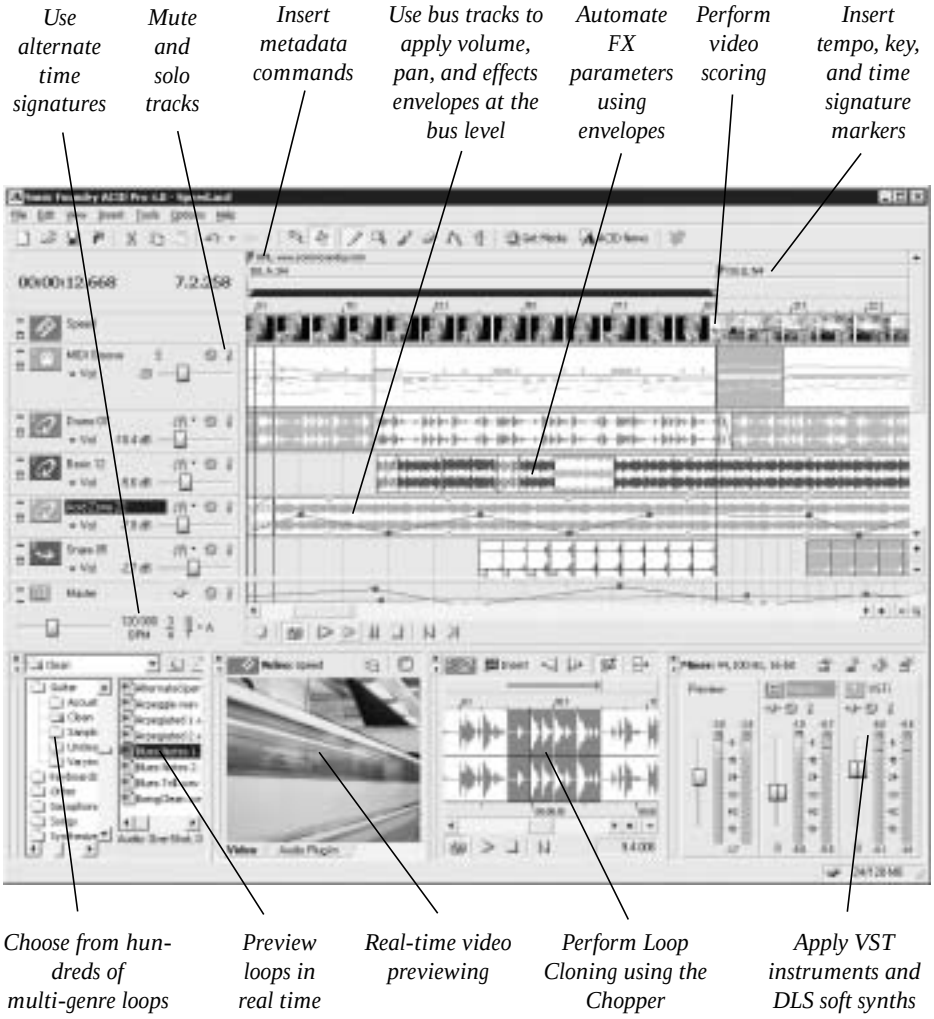
5.1 Surround Mixing and Panning Automation

- ACID PRO 4.0 makes it easy to set up and mix 5.1-channel surround project files for audio and DVD productions. Use the new Surround Panner dialog to easily set channel positions. Automate your surround mix using fully keyframable panning.
- After creating your mix, export your project files and render output using any supported third-party 5.1 encoding software to provide your audience with the ultimate surround experience.

ACID PRO 4.0

MIDI

- Perform MIDI recording, editing, and playback
- Use unlimited tracks of MIDI
- Record MIDI tracks directly into ACID using a MIDI controller
- Combine MIDI tracks with audio tracks
- Generate MIDI Time Code and synchronize with external devices
- Play MIDI files using external or internal MIDI sound sources
- Time stretch MIDI tracks
- Quickly view, filter, and edit MIDI events using ACID PRO's new MIDI list editor. Edit, add, or delete MIDI parameters including velocity, duration, and pitch.
- Align MIDI notes with beats using quantizing. Each MIDI command is listed in a sequence, allowing you to see exactly what notes and parameters are being changed at each point in your mix.
- Use step recording to add events to your project without having to use an external MIDI controller. Gain complete control all of the MIDI parameters you use including interval, duration, attacks, and release.
- Create and edit MIDI tracks in a piano roll format, just like a sequencer. The MIDI piano roll editor allows you to draw new notes, and makes it easier to graphically edit MIDI parameters.
- Use MIDI Quantizing to align your notes with beats based on the parameters you specify, including start times, and note durations. Select any quantize resolution between whole notes and 64th notes.
- Supports a wide variety VSTi plug-ins. VST instruments allow you play back your MIDI tracks through virtual synthesizers, filters, and effects. Software Instruments can be played live or in the studio in real-time via MIDI keyboards and other MIDI controllers.
- Use VST instruments and DLS soft synths. Now you can route any track to a new soft synth bus in the mixer window. Each bus controls its own DLS set, rather than being tied to individual tracks. The soft synth works as a Sound Module that allows you to choose which DLS set or VSTi you want to use with your MIDI tracks.



Video and CD Features

- Open and use WMA or WMV files in your projects along with loops, one-shots, and MIDI tracks. Its easy to render out your mix and create streaming media for the web.
- Two options for rendering projects containing video: Stretch to fill frame and Fast video resizing. The video preview window has an option to display square pixels.
- Rip audio tracks from CDs directly into ACID for immediate remixing. Burn audio projects to CD with track-at-once CD burning.
- Customizable track lets you specify default settings when adding new audio and video tracks. Set volume, pan type, height, and track effects to default values.

Bundled Software

ACID PRO 4.0 comes bundled with Sonic Foundry's XFX 1, XFX 2 and XFX 3, Sound Forge XP Studio and VegasAudio LE. Also includes a music content CD containing more than 600 music loops and dozens of projects from Sonic Foundry's Loops for ACID collection. This huge assortment will get you on your way to creating royalty-free music in minutes. Genres include: Dance, Hip-Hop, Techno, Industrial, Pop, Rock, Jazz, Ambient, Orchestral and more.

SONIC FOUNDRY

LOOPS FOR ACID



ACID DJ Expander Pack

A complete construction kit that delivers more of the same kinds of beats, basses, scratches and synths that make ACID DJ 3.0 the ultimate loops-and-software combination package for Breakbeat, Downtempo, Jungle, Hard House, and Tech House tracks. For the best in modern dance music from Europe to Detroit. 586 (450MB) original loops.

SOLADJEP **38.95**



ACID Techno Expander Pack

Companion kit to ACID Techno 3.0. It gives you more materials in the exact proportions needed to make stunning electronica with superb sonic balance and fidelity. Make your tracks today, play them at the club tonight, and then publish your slammin', royalty-free, license-free mix tomorrow. The best value in adrenalin-pumping dance loops. 575 (460MB) original loops.

SOLATEP **38.95**



ACID Rock

More than 800 loops from a variety of great players, including drummers Siggi Baldursson and Dave Abbruzzese, killer guitar loops from blues master Paul Black and industrial shredder William Tucker. The collection is rounded out with a fine assortment of even more acoustic and electric guitar loops, plus bass, piano, clavinet, synth, and sax loops. 877 (498MB) original loops.

SOLAR **38.95**

George Pendergast: alt.rockdrums



Forty years of rock has seen drummers come and go, and the backbeats still rule. This assortment of unique grooves comes with all the right breaks and fills you need to make complete tracks. Each groove comes in two variations: clean and dry, and also with full-on production. 662 (498MB) original, loops.

SOLAGPARD **38.95**

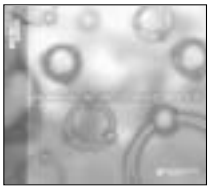
Ambient Realms



Create innovative audio atmospheres and explore new musical environments with this collection of high-quality sound loops. You'll work with a broad spectrum of highly unusual textures, effects, noises, and beds, organized in easy-to-reference categories. Take an out-of-the-box approach to your next multimedia project or original composition. 121 (234MB) original loops.

SOLAAR **38.95**

Aaron Mellinger, Aural E: Eclectic Electronica



An eclectic standalone music construction kit, Aural E is also ideal for multimedia developers who need positive, high-impact music and sounds of any duration. Aural E taps and unifies several music genres under rubrics of fidelity, immediacy, pop culture aesthetics, a little bit of kitsch, and the knowledge of what useful sample libraries provide.

SOLAAEEE **38.95**



The Bill Laswell Collection

The first four loop libraries in the Bill Laswell Collection. Laswell supplies fuel for ACID users who are redefining music just as Laswell does — with total disregard for genre boundary restrictions and obsolete conventions in making, listening to, even conceptualizing, modern music and sound. Includes three libraries, plus a bonus library only available in the box set — Covert Diaspora, a 2 CD collection with superior 24-Bit sound.

SOLABLC **149.95**

The Bill Laswell Collection

Volume I: False Encryptions

Bill Laswell reveals secret codes from the underground. Break them and reformulate it into new transmissions for broadcast within your own music communication networks. 664 (451MB) original loops.

SOLABL1FE **38.95**

The Bill Laswell Collection

Volume II: Undocument

Realizations achieved in Bill Laswell's domain are decontextualized and prepared for the world of chaos and chance. Source materials are freely co-opted by end-users who incorporate ACID smear tactics in their tireless campaigns against conventional music. 675 (450MB) original loops.

SOLABL2U **38.95**

The Bill Laswell Collection

Volume III: Letter of Law

The basic principals of Bill Laswell's production genius are delivered intact, but those with the most musical intelligence will now redefine their interpretation. 662 (450MB) original loops.

SOLABL3LL **38.95**

The Bill Laswell Collection

Volume IV: Covert Diaspora

Dissemination of five strains of music virus via CD-ROM. There is currently no vaccine available. Develop mutations and upload them — we believe in your ability to infect and eliminate generic music forms. Retain your anonymity if desired. Join the flow. 656 (683MB) original, license-free loops.

SOLABL4L **38.95**



Alex Spurkel: Electro-World Percussion

Traditional Middle Eastern rhythms in a series of performances that are energetic, technically accurate, and well-recorded. Addresses every major rhythm with traditional instruments such as doumbek, djembe, rik, frame drum, bendir, and wide assortment of hand percussion. Also includes programmed rhythmic counterparts for electronica—ideal for trance, jungle, and drum 'n' bass tracks.

SOLAASEWP **38.95**



American Piano

Doug Colosio's (performance credits with Willie Nelson, Merle Haggard, Charlie Pride, Jewel, and many others), American Piano offers solo piano phrases that breathe pure American spirit. From traditional country blues, swing, and stride riffing, to country rock and ballad figures that are edited to perfection, enjoy the sounds of smooth, natural, hand-played piano performances in ACID.

SOLAAP **38.95**

LOOPS FOR ACID

BASS-X



A collection of floor shakin' bass loops by Groove Addicts, the world-renowned L.A. music production and sound design house. A collection of intros, choruses, bridges, plucks, clicks, muffs, and song patterns. BASS-X gives up the big bottom you need to anchor your rock, pop, alternative, dance, hip-hop and rap tracks. 686 (419MB) original loops.

SOLABA **38.95**

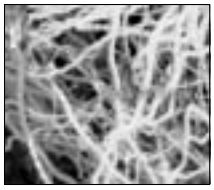
Ambient Grooves: Bill Laswell's Sample Material



This stunning array of loop-based tracks ranges from the world of techno to the beat of the street and features many exotic instruments from different continents. Blend or destroy musical boundaries with a new level of experimentation and expression, using some of the most compelling music on the planet, from a truly original artist.

SOLABLS **38.95**

Bradley Fish Unstrung



Outrageous blend of riffs and licks, produced with an assortment of instruments. Seven-string metal, blues-based rock, and sweet acoustic riffs mix it up with Chinese zithers, banjos, sitars, and more. Ever hear a mountain dulcimer shredded by a full Marshall stack, Banjo licks strained through a Morley wah? All this and more, with classic chops and a playful spirit.

SOLABFU **38.95**



Classic Drum Machines: Syntonic Generator

An eclectic array of high-quality sounds for the ultra-discriminating loop user. This disc includes hundreds of cutting-edge drum and synth loops for killer mixes–house, techno, dance, jungle, and industrial. 191 (267MB) original loops. Classic Drum Machines — the gear you need to mix.

SOLASG **38.95**



Brian Daly: Metarock Visions

Metarock Visions is the brainchild of Brian Daly, one of the world's most accomplished loop library producers. Here is a perfectly-proportioned modern rock construction kit with a collection of guitars, basses, drums, and synths. Brian's massive experience in every aspect of loop library creation culminates here. 765 (546MB) original, license-free loops.

SOLABDMV **38.95**



Classic Country

A grab bag that will add some twang to your next audio project. If you want to capture that Nashville sound, you'll find everything you need in this grab bag of musical loops: bluegrass, drum fills, guitar licks, and bass riffs. Add the whine of a pedal steel guitar or the jangle of a mandolin. Played with top-notch studio musicianship. 410 (261MB) original loops.

SOLACC **38.95**

DJ Puzzle: Scratch Tactics



The DJ rules on Scratch Tactics, the first Loops for ACID library to give it up completely to the decks. This collection is proof that the turntable is a 'legit' musical instrument in a class by itself, and can supply nearly all the beats, breaks, riffs, and FX you need to make grooving, fully-produced tracks. Includes folders full of basses and textures. 515 (457MB) original loops.

SOLADJPST **38.95**

Bunker 8: Nu Groove Pop



Made-to-order dance tracks are minutes away with this collection of beatbox rhythms, tight bass lines, simple melody loops, and compact instrumental motifs. Plenty of infectious song starters are included, ready for your pop vocal track or video drop. Instant access to indelible, royalty-free hooks for making quality tracks with minimum effort and maximum chart potential.

SOLAB8EAB **38.95**

Cinematix Volumes 1 and 2



The award-winning Groove Addicts team is all about music for media, and they understand the power of ACID. The Cinematix series contain high-energy, high-impact instrument loops, stingers, stagers, instant music beds in many styles, and great sound effects too.

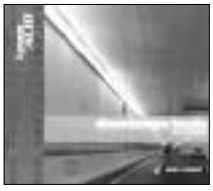
Vol. 1 300 (450MB) loops *SOLAC1* **38.95**
Vol. 2 285 (452MB) loops *SOLAC2* **38.95**



Crimson, Blue & Fabulous: Horncraft for R&B

Known as 'the greatest horn section in the world', Birch 'Crimson Slide' Johnson (trombone), Lou 'Blue Lou' Marini (tenor saxophone), and Alan 'Mr. Fabulous' Rubin (trumpet) offer blistering riffs, solos, and sections, presented in several keys for maximum flexibility and fidelity. 544 (532MB) original loops.

SOLACBFHRB **38.95**



Downtempo Beats

Italian sound designer Leo Cavallo checks in with a collection of beats and grooves on the slower tip. This collection contains a whole year's supply of fresh beats, trading tips on jazz, funk, hip-hop, electronica, and dance music. These slow groove exhibitions emit the inescapable vibe of the downtempo scene, converging styles. 427 (454MB) original loops.

LLDB1000 ..... **50.95**



Drum Components: RADS

Build customized beat patterns with ease,d unmatched flexibility and a user-friendly structure. You get a vast assortment of kick, snare, hi-hat, and ride cymbal patterns, along with dozens of drum fills and cymbal samples. Create your own intricate, infectious drum loops in an incredible range of styles. 974 (235MB) original loops.

SOLARDC **38.95**

LOOPS FOR ACID

Pandora's Toolbox



Add a degree of strange to your production with the dark side of sampling — Dark Ambient Soundscapes. This collection comes from the mind of David Torn and was created for the soul who requires unusual sounds—creepy, kooky, edgy, and weird loops. Make your next project a little twisted with this collection of textural mayhem. 54 (308MB) original loops.
SOLAPT.....**38.95**

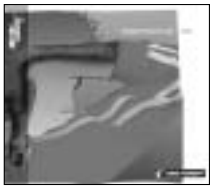
David Torn's SPLaTTeRCeLL:

Textures for Electronica and Film Music



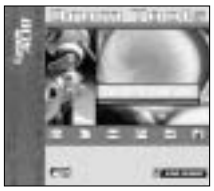
Master guitarist and sound sculptor, David Torn, delivers a brilliant library of abstract guitar performances and studio effects to generate massive soundscapes and twisted sonic implants. Depend on this rich collection to provide the sounds that have made Torn famous for his originality. 173 (545MB) original loops.
SOLADTS.....**38.95**

designer.dance.tools: ddt



Peace/Love Production's Jason Donnelly is as skilled as they come, and D.D.T. proves it with a perfect balance of beats, basses, synths, decks, and FX – all the must-haves for making techno, house, tech-house, progressive, trance, jungle, breakbeat, hip-hop, and downtempo tracks. 596 (501MB) original loops.
SOLADDT.....**38.95**

Drum Tools



Percussion-based loops that will make your next musical project a snap, a kick, and a ride you won't forget. All tracks were recorded live with the creative team from Roxette, the Swedish Top 40 group. Includes hundreds of infectious and irresistible percussion loops in a variety of styles. A must-have musical resource for pop and rock enthusiasts. 1215 (519MB) original loops.
SOLADT.....**38.95**

Discrete Drums Volume 2



Meticulously crafted and affordable beat crafting tools. Customizable rock/pop acoustic kit drum and percussion grooves. At the macro level, each volume contains three fully-mixed foundation beats, with all the fills and variations for creating complete, seamless drum tracks instantly. At the micro level, get everything, from multitrack session files to isolated one shots.
SOLADD2.....**38.95**

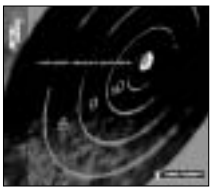
Electrocution

Big Beat / Darkside Construction Kit



The best of electro-hop, darkside, big beat and intelligent industrial music. Aaron Mellinger delivers a generous supply of all the sonic elements you need. Beats range from slamming to ethereal, and an excellent assortment of basses, synths and textures roll from grooving to disquieting. 462 (507MB) original loops.
SOLAE.....**38.95**

electronic point-blank



Presented by Mark Pistel, founder of Consolidated, this unparalleled construction kit is aimed squarely at the underground techno / dance music scene. A solid construction kit of synths, drums, and FX — to make your remix shine, or come on with fully-produced tracks — license-free and royalty-free in ACID. 494 (452MB) original loops.
SOLAMPEPB.....**38.95**

Eletronica Grooves: Cyclotronic Resonator



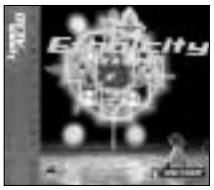
House, dance, techno, electronica, pop, acid, noise, and many other styles, all with the composer and producer in mind. DJs to radio producers can easily create thumpin' funkin' tracks. Every loop resonates with creativity and flexibility for the ease-of-use of the composer/producer. 258 (261MB) original loops.
SOLACR.....**38.95**

Essential Sounds II



Pick up where the ACID PRO 2.0 (Essential Sounds I) leave off. With Essential Sounds II, you receive even more "bread and butter" loops. Essential Sounds II features a huge variety of instrument types and styles—from superfunky electric bass to tasty sax licks and everything in between. Get every sound you need to make great music! 630 (459MB) original loops.
SOLAES2.....**Call**

Ethnicity



Imagine a tabla from India or an Indonesian anklung in your musical palette. With Ethnicity, imagination becomes reality. Add soulful, global sounds with hundreds of traditional rhythms. Explore new sounds and create audio experiences that have a unique cultural feel, using ready-to-use loops of percussion instruments from around the world. 324 (233MB) original, loops.
SOLAEN.....**38.95**

Euro Techno: Ground Loops



A collection of rave-ready beats, percussion, synths, pads, and arpeggios that reflect the energy and intensity of today's European dance scene. Create an unlimited supply of richly textured, distinctive tracks that will capture the club vibe—from bleepy experiments to raging, four-on-the-floor techno excursions. 600 (460MB) original loops.
SOLAET.....**38.95**

Electro Hip-Hop: Mac Money



Deliver the sounds of the hottest Top 40 hip-hop and R&B tracks on the charts. The complete package of beats, basses, keys, guitars, and more, including the sweetest vocal harmonies, speech, and scratchin' you've ever heard. Whether you're an experienced producer or just getting started with ACID, these are hit-making sounds. 421 (475MB) original loops.
SOLAMMHH.....**38.95**

B&H PHOTO - VIDEO - PRO AUDIO
EQUIPMENT LEASING AVAILABLE

LOOPS FOR ACID



Extremely Abrasive Beats

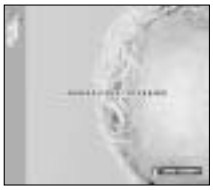
Extremely Abrasive Beats (EAB) is an unrelentingly harsh collection of fried beats and severe accents made for launching multiple rhythmic sorties on every industrial, postrock, crunchy downtempo, abstract hip-hop, and sheer noise frontier on the map. Pain-inducing, speaker-shredding, and highly toxic, EAB transforms ACID into a piece of sonic ordnance that will effortlessly incapacitate the enemy and render it harmless. Use it without mercy.
OLAEAB **38.95**

Fluid Dynamics: Computational Drum 'n' Bass



Complicated packets of rhythmic information can be painted across your screen to create smooth, high bpm rhythm beds which can be edited at the microscopic level to achieve all precarious twists and turns that make drum 'n' bass exciting. 607 (452MB) original loops.
SOLARMFD.....**38.95**

Headstrong Grooves



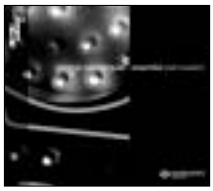
Contains most current beats and basslines. Paint your grooves into ACID and then adorn them with grungy synth leads, futuristic analog effects, filter sweeps, scratches, and more. Has everything you need to make dance music in the garage and 2-step styles. Add an MC or some diva samples, and achieve complete control. 634 (527MB) original loops.
SOLAHG.....**38.95**



Intelligent Dance Music: Twine Components

Intelligent Dance Music is an exclusive view of electronica from the stratosphere of sound design. Create harsh, twisted beats or lush ambience using an assortment of electronic rhythms, melodies, bass lines, electro-acoustic treatments, digital breakdowns, and evocative sonic environments. 350 (449MB) original loops.
SOLAIDM **38.95**

George Pendergast: Essential Percussion



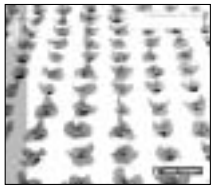
Packed with inspiring patterns played on custom kits, and a wide variety of hand percussion, including congas, bongos, djembe, doumbek, slit drums, timbales, bells, cymbals, shakers, tambourines, and more. In addition to patterns, an included extensive one shot collection allows you to customize your tracks and achieve unique results.
SOLAGPARD **38.95**



Futurist Drum 'n' Bass

Create one of the most versatile and exciting styles of music happening today—fast tempos and fractured breaks that resonate throughout the worlds of rock, pop, jazz, and hip-hop. Futurist Drum 'n' Bass delivers perfect beats and takes the tedium out of editing and matching difficult rhythms. 800 (400MB) original loops.
SOLAFDB **38.95**

Hydroponic Hip-Hop



There's some super phat hip-hop hydro budding in Brian Daly's digital hothouse, and it's all bumpin' to the beat. Brian's hydro is the best, and Hydroponic Hip-Hop is one killer harvest of mad blunted Loops for ACID. Get your mic under control, open your ACID, put the hydro on the tray, and click-slide on the perfect grooves. 428 (534MB) original loops.
SOLABDHHH **38.95**



Junkyard Rhythms

Superstar percussionist Joe Vitale and producer Joe Vitale Jr. spent countless hours pounding away at a junkyard, in search of metallic objects that would yield maximum klang. What they found will amaze you! So, if you need some percussion that'll make people say, "What was that!?" look no further than Junkyard Percussion. 868 (470MB) original loops.
SOLAJYR **38.95**

Essential Sounds III



All the sound-building components needed to complete a variety of projects. In the music department, you'll find acoustic and electronic drum grooves, wicked guitar and keyboard loops, funky bass grooves, and more. Elsewhere on the disc, you'll find game-sound elements and sound effects, foley effects, soundscapes, menu wipes, and more! An invaluable tool for composers, producers, and sound designers alike.
SOLAES3 .....**38.95**

Fast Breaks



Fast Breaks delivers speedy drum 'n' bass grooves on a budget. With all the beats and basses on this disc, you'll take a permanent vacation from long hours of recycling and editing loops. With Fast Breaks, you will enjoy the fun process and creative satisfaction of working with beats and basses. 328 (224MB) original loops .....**Call**

Industrial Toolkit: Methods of Mayhem



Create sonic disturbances with this organized catalog of strange and deranged loops. Space vibes, metal on metal hits, and short synth stabs are just the beginning. The disc also includes hundreds of aggressive samples, from twisted guitar riffs and distorted synth loops to feedback rhythms and mouthbox beats. 1090 (554MB) original loops.
SOLAMOM **38.95**

Infinite Soul Volumes 1 and 2



The best in R&B construction kits. Both Infinite Soul volumes are perfect for variety, balance, and fidelity and provide an unbeatable rhythm section for holding down your lyrics, raps, and jams. They also contain complementary ACID starter projects made by the MAC MONEY team.

Vol. 1 300 (369MB) loops. *SOLSKIS1*.....**18.95**
Vol. 2 296 (266MB) loops. *SOLSKIS2* **18.95**

LOOPS FOR ACID



Orchestral Series

The Orchestral series is a four volume collection of ***Classical***, ***Modern***, ***Cinematic*** and ***Rock & Pop*** loops and compositions. An international team of gifted composers and producers produces this astounding variety of authentic original music, presented as inspiring collections of royalty-free sounds. Precisely edited to provide you with the loops you need to conduct your own virtual symphony. 2334 (2169MB) original loops.
SOLAOLP **149.95**

Orchestral Series 1:
Classical
Composers and musicians steeped in the history of European music pay tribute to the most beloved Baroque and Romantic period composers. Create a genuine and complete orchestral experience. 589 (536MB) original loops.
SOLAO1C.....**38.95**

Orchestral Series 2:
Modern
An astounding collection of harmonically rich and impressionistic spaces, as well as the propulsive, kinetic energy of formal minimalism. 720 (544MB) original loops.
SOLAO2MO **38.95**

Orchestral Series 3:
Cinematic
Evoke a sense of time and place, or underscore mood and atmosphere. Complete orchestral pieces provide instant soundtracks. Includes individual instrument and solo section loops. 412 (529MB) original loops.
SOLAO3C.....**38.95**

Orchestral Series 4:
Rock & Pop
String, brass, woodwind, and percussion loops edited to provide intros and transitions, follow chord changes, and add more sparkle to your tracks. 635 (561MB) original loops.
SOLAO4RP.....**38.95**



Jazz Solos and Sections

Vince Andrews Horns & Woodwinds

Traditional and effected sounds including silky and sparkling brass and woodwind riffs as well as counterpart files created by Sonic Foundry's sound design crew. Make a solo statement on your jazz track, or add a blue note or a blaring distress call to your next composition with these 529 essential jazz loops. (409MB).
SOLAVASS.....**38.95**



On the Jazz Tip

Bring in the swing, the sting, and genuine jazz thing with this virtual band-in-a-box that contains an assortment of red hot-'n'-blue horn licks, beat crazy drum fills, hot electric piano licks, and much more. You'll get hundreds of drum sounds and instrumental tracks, including sax squeals and bass grooves. Juice up your musical palette with On the Jazz Tip. 1010 (557MB) original loops.
SOLAOTJT **38.95**

James Johnson: Slow Silhouette



A double disc collection of long, sparse music segments designed for sustained, non-looping musical passages. Used alone, each segment exists in a single, simple, highly refined sound space. Layering these sounds will allow you to create an infinite number of stunning buildups, transitions, or even complete long form pieces of flowing, seamlessly composed music.....**38.95**

Knutrix Drum n' Bass



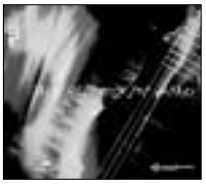
The beats and textures that define drum 'n' bass are sophisticated and positive. Integrating with virtually every genre, flexible drum 'n' bass soundbeds permeate clubs, TV commercials, and multimedia presentations. Drum 'n' Bass is an articulated, super-smooth construction kit that provides all the instrumentation needed for creating complete compositions.
SOLAKDB.....**38.95**

Ilona! Universal Female Vocals:



Take three full octaves of 'ooh' and 'aah' vocal samples, with long and short durations provided for every note. Then, add the production candy: separate, stereo Lexicon 480 reverb tracks, with your choice of decay times - for each and every note. Create a conventional mix, or experiment with unique combinations of sung notes and reverbs for startling vocal effects. Includes a an assortment of royalty-free diva samples.
SOLAIUFVTK **38.95**

Jade Hill: Rock/Pop Guitars



A beautifully crafted collection of guitar loops designed specifically as tools for chartbound singers and songwriters. Choose from a variety of jangly, bluesy, funky, and hard rock chord structures, and modify these in ACID to suit your arrangements. Then, personalize your creations with guitar solos that range from clean and melodic to chunky and shredded. Over 600 loops.
SOLAJHRPG **38.95**

Sound Genius Studios: Jazz Trap Kit



Get ready to enjoy a rainbow of styles: swing, smooth, 3/4, 5/4, sambas, waltzes, marches, and more! Customize your jazz rhythm beds with brushwork, isolated hi-hat and cymbal patterns, rolls and licks, and a comprehensive selection of one shots. Includes some bongo, conga, and doumbek patterns to add some extra cool to your mix .....**Call**

Latin Percussion



Latin Percussion is a collection of sizzling Latin rhythms from straightforward to exotic. Whether it's a simple shaker, hypnotic clave, driving conga, or intense ensemble, these meticulously tight and dynamic performances will lend movement and energy to any track. Turn up the heat on your next audio masterpiece. 600 (452MB) original loops.
SOLALP **38.95**

LOOPS FOR ACID



Leo Cavallo's Bass Taster

A superbly played and produced collection of bass loops that'll have you well covered for funky downtempo grooves, chilled out low end pulsations, and articulated post prog sonic tapestries. Inspired performances on fine instruments, smart processing, and Leo Cavallo's evolved loop library creation aesthetics are all present and accounted for here. Bonus folder full of drum loops, synth riffs, and special FX.

SOLALCBT **38.95**



Leo Cavallo: The Songwriter's Acoustic Guitar Companion

Browse through these sounds and you'll be endlessly entertained and amazed as you try to name all the smash hit singles born with just an acoustic guitar, a timeless lyric, and chord progressions just like these! Your amazement will turn into pure inspiration as new songs emerge from your desktop studio one after another.

SOLALCAGC **38.95**



Ma Ja Lé: Chromosphere

Intense percussion loops, disintegrating analog instruments, forbidden Theremin transmissions, solarized electric guitars, and more are presented with the awesome fidelity and advanced production techniques that define the current state of the Ma Ja Le' sound. An extremely effective one-stop scoring tool, as well as a great resource for sweetening tracks in almost any modern music genre.

SOLAC **38.95**

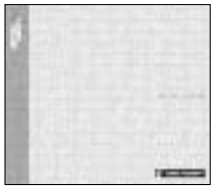
Mac Money R&B 101



A stunning crash course in building superb R&B tracks. This impeccable construction kit contains a wealth of loops conceived and executed in classic Mac Money style - killer beats, super phat basses, funky keyboards and guitars, and all the other ear candy it takes to round out a one-disc hit factory of infinite possibilities. 561 (498MB) original loops..

SOLAMMRB101 ..... **38.95**

Machine Language



This comprehensive hard techno construction kit by creator Jon Drukman delivers intense dance floor action — no polite, standard fare here, just five hundred maximum techno loops created by a well-established pro, all imbued with the special bits that'll make 'em fly in ACID — the ultimate dance music creation engine. 500 (487MB) original loops.

SOLAML **38.95**

Mick Fleetwood: Total Drumming



Fleetwood Mac's Mick Fleetwood is the living history of rock. Total Drumming reveals the shining core of the Fleetwood style, and showcases all the beats and rhythms that literally changed the course of modern music. Let the power of rock history propel you into the future.

SOLAMFTD **38.95**



Numina:

Numina synthesist / composer Jesse Sola delivers startlingly pure sounds that reference common emotions and cinematic events with the kind of flair that will make your listeners feel as if they're having a completely new experience. This construction kit contains a range of flexible, useful elements including beats, beds, melodies, and effects, that share an obvious, universal appeal.

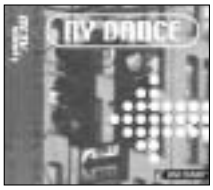
SOLANRRQ **38.95**

New Roots Reggae



New Roots Reggae boasts a brilliant abundance of roots, dub, and dancehall reggae grooves and instrumentation — presented with a modern electronic slant. It's an exotic island collection that will appeal to roots rockers and dub scientists alike. 425 (450MB) original loops.

SOLANRR **38.95**



New York Dance

New York Dance is a dance music construction kit produced by Doug Beck, one of NYC's hottest young producers and remix artists. A perfect blend of contemporary styles including big beat, house, deep house, electro, jungle, and more. Combine the sounds in New York Dance to create your own glittering opus. 604 (491MB) original loops.

SOLANYDP **38.95**

New York Dance I & II



New York Dance I is one of two original, exclusive construction sets created by Doug Beck, one of the Big Apple's best-known remix artists. It contains your most requested styles including big beat, house, electro, jungle, and more.

Vol. 1 301 (252MB) loops. *SOQLNYD1* **19.95**

Vol. 2 303 (239MB) loops. *SOQLNYD2* **19.95**

Phat Trax XXL



Put some real backbone under your rhymes with this all-in-one Hip-Hop workshop. Phat beats and basses, solid instrument riffs, turntable scratches, a folder full of MC announcements, and everything else you need to compete. 301 (182MB) original loops.

SOQLPTXXL **19.95**

Prototechno: Techno Foundations



A complete construction kit for making pure, fully developed techno tracks. Created using vintage modular synths and digital modeling instruments, this is a collection of very stern techno, with no intrusive signal processing added to interfere with the massive krunch and klang of the authentic instruments. 532 (490MB) original loops.

SOLAPTF **38.95**

SONIC FOUNDRY

LOOPS FOR ACID



Psychedelic Guitar

The Serpentine Magic of Harvey Mandel

Harvey Mandel is one of the pioneers of psychedelic music, with a style that changed the world of rock guitar. Psychedelic Guitar includes his signature soaring leads saturated with effects, supple rock rhythms, special effects, and mountains of tone and sustain—with the feel of an improvised live performance. 269 (405MB) loops.

SOLAPG.....**38.95**

Rapoon: Sci-Fi Tribal



The spaces between Robin Storey's mesmerizing loop points are filled with strange combinations of ethnic instruments, assorted noisemakers, found objects, obscure sound bites, and severe processing. Sci-Fi Tribal is a gateway to moods and atmospheres that will make lasting impressions.462 (510MB) original loops.

SOLARSFT.....**38.95**

Stylus Pressure



Apply the Pressure and enjoy the strange fruits of years spent diligently cataloging and preparing fine soundz, distilled by Stylus Pressure down to thirty-seven construction kits that effuse pure urban magik – trippin' grooves and freaky sonic collages that flow into ACID as quickly as you can act and react with the paintbrush tool.

SOLASP.....**38.95**

Sounds of Asia: Opium



Capture the spirit of the Orient in a collection of truly original audio loops. Opium contains unique tracks that encompass a variety of traditional Asian instruments. Weave mesmerizing musical spells from the Far East, add instruments like the dizu flute, the pip, and the yangchin. Give your next musical composition a touch of the exotic. 241 (229MB) original loops.

SOLAO.....**38.95**



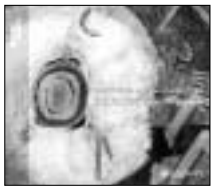
R&B Drums: Groove Spectrum

R&B Drums: Groove Spectrum is the funk drum groove encyclopedia. This is a virtual history of R&B, from its cold sweat beginnings to the machine-driven grooves of today's smoothest R&B hits. R&B Drums includes classic acoustic kit drums, congas, and one-shots as well as modern machine grooves. 588 (410MB) original loops.

SOLARBD.....**38.95**

Robin Storey's Rapoon

Textures & Soundscapes



Robin Storey is a founding member of the music ensemble :zoviet-france: and the genius behind many critically-acclaimed solo releases. Immerse yourself in Rapoon—a neo-primitive depiction of a post-industrial dreamtime. Create unsettling and serene sonic images. 590 (486MB) original loops.

SOLARSR.....**38.95**



Ma Ja Le':

Saturn & Elsewhere

Ethereal Textures and Psychotropic Rhythms

Space Music features intricate soundscapes, and lush, high fidelity sound. Enjoy the pulsating rhythms and swirling textures of Saturn and Elsewhere - a group of tribal ambient sounds. Utilized as a standalone kit, or a source of rhythms and atmospheres for electronica, experimental music, and film scores. 277 (479MB) original loops .....**Call**

Spektral Minimalism



Sonic Foundry together with ambient minimalist composer James Johnson present a Loops for ACID library filled with shifting, amorphous atmospheres, long drifting soundscapes, elegant piano passages, and textures that span the entire aural spectrum. Here is a collection of sounds made to dance in and out of the listener's consciousness 123 (526MB) original loops.

SOLAJJSM.....**38.95**



Ambient Atmospheres & Rhythms

Robert Rich's Liquid Planet

An organic and synthetic world filled with compelling sounds. Populate your songs with a generous array of rhythms, melodies, and atmospheres. Choose from flutes, drones, and textured soundscapes, and impart to your music hypnotic, nuanced minimalism. 482 (569MB) original loops.

SOLARRLP **38.95**

Rudy Sarzo: Workingman's Bass

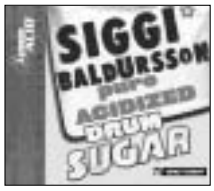


Rudy Sarzo's rock sold bass has worked like a charm for metal bands. For Sonic Foundry, the ACID-savvy bassman stamped out big pallets full of thick, round-wound bottom fasteners, trimmed for optimum treble and bite, perfectly looped, and ready for delivery to your local ACID assembly plant. 1582 (436MB) original loops.

SOLAWB.....**38.95**

Siggi Baldursson's Drumsugar

High-Carb Acoustic Drum Grooves



Loose, funky, rockin', and sophisticated, this generous array of beats, breaks and made-to order drum tracks will meet your creative demands. Recorded at Butch Vig's Smart Studios by veteran library producer Brian Daly, Siggi creates an atmosphere that is musical and inspiring. 780 (555MB) original loops.

SOLASBDS.....**38.95**

Street Beats



Offering incredibly funky drum loops, this practical disc delivers everything from hardcore hip-hop, jungle, and house to traditional R&B and soul beats. Authentic raw sounds, live, studio, and even true street sounds - recorded all over the world - are included. Street Beats delivers the real deal... beats that you want to use, sounds that you want to hear. 147 (34MB) original loops.

SOLASB.....**38.95**

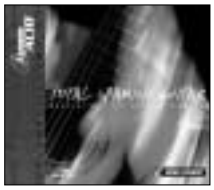
LOOPS FOR ACID



Synthetic Sound Effects: Twisted Reality

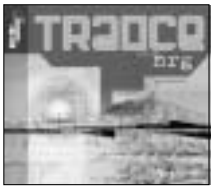
Take a trip through warped worlds, fantastic futurescapes, and unidentifiable ethereal realms. Synthetic Sound Effects is a collection of sounds that defy imagination and categorization, from vocal weirdness to vintage FX noises. Let your creativity explore strange, uncharted territory with Synthetic Sound Effects. 269 (221MB) original loops.
SOLATR **38.95**

Total Spanish Guitar



A collection of traditional and modern acoustic guitar performances rendered in a comprehensive range of flavors and feels. From lightening-fast picado leads to rasgueado rhythms and beautifully played soft ballads, all of the loops in this collection are imbued with stylistic authenticity, real musicality, and breathtaking performance and engineering skill. 928 (774MB) loops.
SOLAPBG **38.95**

Trance NRG



The veteran PLP sound designers are proud to present Trance NRG, a balanced set of do-it-yourself beats, crisp arpeggios, long sweeps, saturated leads, clean basses, and club style scratches. Pick loops with complex, full-bodied structures for making quick tracks on demand, or dive in and start from scratch with minimal beat component loops and their one shot counterparts.
SOLATNRG **38.95**

Vir Unis: Infusion
Fractalized Grooves and Atmospheres



Vir Unis has proven time and again his mastery of blending beats and atmospheres into compelling electronic biorhythms and psychedelic landscapes. Infusion delivers morphing and shapeshifting electronica loops, electro-tribal beats, industrial noisescapes, and deep space atmospherics. 257 (454MB) original loops.
SOLAVU **38.95**



Techno Club Grooves 1: Funky xtreams I

TOKYO-originated grooves that take the lead for the new club scene. All the percussive rhythms for the latest in techno, hip-hop, trip-hop, and more. Add hard-core techno beats, machine beat loops, cool vintage rhythm box patterns, and stylized grooves, or create your own grooves with kick bass, hi-hat, and tom loops. 211 (298MB) original loops.
SOLAFX1 ..... **38.95**

Twine: Build



Electronica duo Twine provides access to some of the most evolved sounds in electronica, in a self-contained music construction kit of beats and textures derived from Twine's world of deeply-processed audio. Experience an alchemical fusion of electronic and acoustic elements that constitute a truly unique perspective of music and sound. 341 (534MB) loops.
SOLATB **38.95**

Universal Groove Elements



A multitude of musical genres boiled down to the core essentials. These tracks are all the building blocks you need to construct fully-realized pieces of exquisite, cutting-edge music. From hip-hop to techno, basslines to beats, this disc has it all—the definitive collection of all that moves and grooves. 886 (536MB) original loops.
SOLAUGE **38.95**

Voices of Native America



Douglas Spotted Eagle, Grammy Award winner, takes you into the soul of North American indigenous music with soul-stirring sounds from exquisite solo vocals and flute passages to a variety of thrilling drum sounds. Sounds of the earth, the sky, the wind, and the rain—indigenous music at its best. 133 (173MB) original loops.
SOLAVNA **38.95**



Techno Club Grooves 2: Funky xtreams II

Get hip, vintage loops from the KORG Rhythm55 preset rhythm box. These rare sounds are essential for any serious vintage rhythm machine user. For a refreshing departure from other rhythm machines, add sounds like the "Disco" or "Bossa-Nova" presets, which sound great even to this day. 186 (115MB) original loops.
SOLAFX2 ..... **38.95**

Techno Synth Loops Volume 2



The essence you need to create music for dancing all night. Create wicked techno, drum 'n' bass, ambient, and house grooves. This collection of funky dance sounds was created on the greatest synthesizers - Korg MS20, Minimoog, Prophet-5, Roland Jupiter6 and Jupiter8, and Yamaha DX7. 193 (109MB) loops.
SOLAS2 ..... **38.95**



Vir Unis: Bug Report

A complete construction kit that extends the Vir Unis lexicon of bionic grooves and deep soundscapes to incorporate elements of contemporary minimal electronica. The tight integration of organic and synthetic structures allows even the most static forms and heavily quantized passages the ability to breathe with the dynamism of real life.
SOLABR **38.95**

Voices from the Blackbox



Hip-Hop/R&B Vocals: Mac Money Voices from the Blackbox is your source for male and female vocals built on rap, soul, hip-hop, and R&B. Filled with solos, harmonies, and scat phrases, the Mac Money team delivers the sounds you need to add professional vocals to your latest mix. 784 (474MB) original loops.
SOLAMMV **38.95**

SONIC FOUNDRY

LOOPS FOR ACID



Ugly Remnants Vol. 1 & 2

The original Ugly Remnants collection was celebrated as "...massive... scary... twisted... ghostly... weird... wonderful… an excellent body of work." (EM Magazine). Now updated over two volumes, each one a complete kit containing an enormous variety of music and sound.

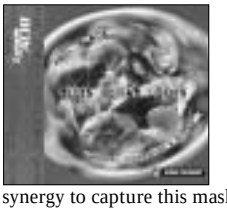
Volume 1: *SOLAUR1*.....**38.95**
Volume 2: *SOLAUR2*.....**38.95**

World Percussion: Marc Anderson's Dragon Dance



Marc Anderson, a student of African percussion, understands ancient sounds and modern technologies, and World Percussion marries them both. This library contains sounds that range from essential single solo instruments and performances to blistering walls of tribal thunder. 586 (465MB) original loops.
SOLAMADD **38.95**

Blues Guitar: Paul Black's Whiskey, Cigarettes & Gumbo



A gold mine of authentic, traditional blues: deep delta stylings and scorching urban riffs, rich with tone and character. Artists, engineers, and editors unite in a synergy to capture this mash of tasty licks. Add the essence of Paul Black's slide and fingerstyle work to your own musical creations. 660 (611MB) original loops.
SOLAPBG **38.95**

Steve Tibbetts: Friendly Fire



Steve Tibbetts' releases have received five star reviews in publications ranging from Rolling Stone to Downbeat to Guitar Player. This collection explores the full spectrum of Steve's unique approach to music and sound, from the trademark psychedelic rainbows of distorted guitar, to the impossibly delicate and meticulously produced acoustic work that never fails to leave you transfixed.
SOLASTFF **38.95**

Underground Soundlab: Hip-Hop Construction Kit



Allows you blend straight rhythm section instruments with skillful vinyl resourcing and advanced production techniques to create hip-hop with unmatched levels of sophistication and fidelity. Sound is open and spacious, lush, detailed, and extremely elastic and phat. Saturated with classic funk / R&B references.
SOLAUS **38.95**

World Pop



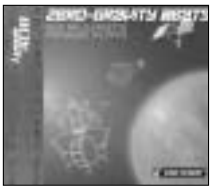
In World Pop, producer Richard Michos blends music genres with world influences. This kit offers many styles and textures, including rhythm sections, percussion loops, electric guitar washes, and Pakistani and East Indian vocal flourishes — melded to produce shimmering, exotic results. World Pop will enhance the musical landscape of your next composition. 389 (451MB) original loops.
SOLAWP **38.95**



Vintage Analog Synths: Vortexual Amplitude

The ultimate analog loop sampling disc. Hundreds of loops from great synthesizers such as the ARP 2500, Odyssey, EML Electrocomp 101, CAT SRM, Roland SH-1, Sequential Circuits Pro One, MultiMoog, and Memory Moog. Great for analog heads, DJs, radio producers, or anyone wanting to add classic synth sound. 119 (131MB) original loops.
SOLAVA **38.95**

Processed Drumkits: Zero-Gravity Beats



Futuristic, processed drum loops from Siggi Baldursson using a range of unusual and slightly outrageous recording techniques and effects processing. Intensely creative drumming adds flair and builds rhythmic foundations for trip-hop, jungle, dub, techno, space rock, trance, and other ultra-modern musical styles. 765 (543MB) original loops.
SOLAZGB **38.95**

CD Architect 5.0

Professional Red Book Audio CD Mastering



Produce professional audio CDs to Red Book specification with CD Architect 5.0— the most precise audio CD mastering and burning tools available. Arrange your CD Architect project using simple drag-and-drop operations. Produce complex crossfades and DJ-style megamixes. Generate multiple takes of a song to use in your project before burning. Create live-style CDs with audio in the time between tracks, apply volume envelopes and event ASR envelopes – even create hidden tracks. Perform full PQ code editing, trim events from full-length media and add as new tracks, import regions from Sound Forge as tracks, master audio by applying over 20 real-time DirectX effects to events and the master output, and perform disc-at-once CD burning.

- 32-bit, 192kHz source audio support
- High-quality resampling and dithering
- Trimmer window
- Undo/redo history list
- Multiple file format support
- Media file previewing
- Tabbed/docking windows
- Greater than 1:1 zoom •Ripple editing
- Audio layering to create complex crossfades
- Real-time pitch shift/time stretch
- Unlimited volume envelope points
- Expanded drive support including USB and FireWire devices

B&H PHOTO - VIDEO - PRO AUDIO

ACID LOOP BUNDLES

Drum Pack

Drummers fear this bundle. An infinite number of potential hit singles are hidden in this amazing collection of killer kit drum and percussion loops. 4142 (2159MB) original, license-free loops.

Kit contains:

- Drum Tools is the arbiter of clean, simple rock and pop beats, and the 'Laidback' section has the drum loops you need for your slow songs and ballads.
- R&B Drums: Groove Spectrum is a drum groove timeline that documents decades of the genre, while Drum Components: RADS allows you to go in and build great drum tracks from the kick drum on up.
- Latin Percussion is the work of megahit percussionist Joe Vitale, and Processed Drumkits: Zero-Gravity Beats is there when you need something a bit more on the startling side.

Dance Pack

Make your own club scene with this brilliant mix of Electronica and Techno loop libraries. 2499 (1863MB) original loops.

Kit contains:

- New York Dance supplies the basic materials needed to create music that spans House, Electro, Jungle and more.
- Futurist Drum 'n' Bass gives you even more fine beats and synth elements.
- Techno Club Grooves 1: Funky xtreams I and Vintage Analog Synths: Vortexual Amplitude add character by giving you access to rare and vintage gear.
- Processed Drumkits: Zero-Gravity Beats adds something completely unique and original - acoustic drum performances by former Sugarcube Siggi Baldursson, tweaked and processed into pulverizing grooves, made to order for intense dance music.

Ambient Pack

Explore new dimensions in sound with this carefully constructed pack. Each library brings something different to your ambient creations. A sonic adventure through uncharted musical frontiers. 1510 (2522MB) original, license-free loops.

Kit contains:

- Master guitarist and sound sculptor David Torn delivers a wide range of abstract sounds in SPLaTTeRCeLL: Textures for Electronica and Film Music.
- Robin Story's Rapoon: Sci-Fi Tribal offers found sounds and other objects de musique concrete.
- Journey through space music with Ma Ja Le: Saturn and Elsewhere.
- Manipulate the latest in electronica from Twine: Build.
- Vir Unis: Infusion provides shapeshifting loops and noisescapes that will ensure your music remains unique.

Guitar Pack

Two pure guitar libraries and three guitar-heavy construction kits comprise this genre-spanning bundle. 3059 (2290MB) original loops.

Kit contains:

- Psychedelic Guitar
- Blues Guitar:
  Paul Black's Whiskey, Cigarettes & Gumbo
- Industrial Toolkit: Methods of Mayhem
- Classic Country
- Essential Sounds II

Urban Pack

Your hip-hop and R&B dreams will become a reality when you add these five smooth libraries full of funky stuff to your toolbox. 3332 (2325MB) original loops.

Kit contains:

- On the Jazz Tip is a perennial favorite that features real players and a warm, old-school feel.
- Electro Hip-Hop: Mac Money gives up some serious machine grooves and big bottom in a construction kit that speaks hi-fi on every loop.
- The R&B Drums: Groove Spectrum is an exceptional R&B drum loop library that works anywhere in the R&B timeline.
- Last but certainly not least, Hip-Hop/R&B Vocals: Mac Money Voices from the Blackbox will give your tracks all the vocal candy they need to make your music truly unique.
- Urban Pack also includes contemporary, essential sounds from Jazz Solos and Sections: Vince Andrews Horns & Woodwinds.

Producer's Pack

This bundle supplies you with Sonic Foundry's finest assortment of textural elements for creating atmospheres and sound-tracks.2468 (2276MB) original loops.

Kit contains:

- Universal Groove Elements and World Pop provide a generous assortment of beats and rhythmic elements,
- Textures & Soundscapes: Robin Storey's Rapoon, Ambient Atmospheres & Rhythms: Robert Rich's Liquid Planet, and Ambient Realms serve up the atmospheres and textural details you need to create musical narratives that will leave a lasting impression.

World Pack

Travel the globe and beyond with a group of libraries that explores traditional world music and cross-cultural musical hybrids. 2386 (1937MB) original loops.

Kit contains:

- Sounds of Asia: Opium and Ethnicity are full of Asian treasures performed on ancient instruments.
- Latin Percussion features the driving rhythms of session star Joe Vitale.
- World Pop combines Indian and pop elements into something completely contemporary
- Ambient Grooves: Bill Laswell's Sample Material is a large sample of the genius that defines Laswell as the most significant catalyst of world music working today.

Scoring Pack

ACID is fast becoming filmmakers' scoring tool of choice, and this pack shows why. Creating and composing an original score for your production has never been easier. 1460 (2588MB) original, license-free loops.

Kit contains:

- Emotional Peak Sounds provides tension, stabs, and other sonic devices.
- Create soundscapes and moods with Spektral Minimalism.
- Robert Rich's Liquid Planet supplies atmospheric rhythms and melodies.
- Put a symphony to work on your score with Orchestral Series 3: Cinematic.
- Drop in stingers, stagers, and music beds from Cinematix 1.

Scoring Pack II: The Sequel

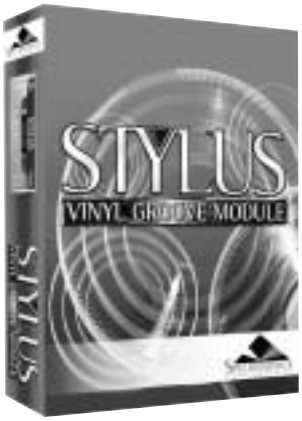
Scoring Pack was a hit, so Sonic Foundry followed Hollywood tradition by releasing a sequel!

Kit contains:

- Cinematix Volume II provides film-grade stingers, effects and more.
- Orchestral Series Volume II: Modern brings 20th Century composition to your mix.
- Ugly Remnants, Volume II adds unique, otherworldly sonic elements.
- Aural E: Eclectic Electronica injects processed sounds and manipulated media.
- Finally, ILONA!: Universal Female Vocal Toolkit offers three octaves of female vocal samples to bring a human touch to your score.

SPECTRASONICS

STYLUS -VINYL GROOVE MODULE



World's First Groove Control Instrument

Stylus is an innovative software plug-in instrument that integrates a massive core library of thousands of cutting-edge groove elements, loops and samples, with a powerful user interface for creating your own grooves. Stylus can be used as a native plug-in instrument in hosts like Logic, Digital Performer, Cubase, Nuendo and Pro Tools— no sampler is necessary!
The huge 3 GB core library of Stylus includes over 1000 brand-new loops, featuring insane new breakbeat remix loops with a cutting-edge vinyl attitude, from award-winning producer Eric Persing. The grooves are inspiration generators for application in a huge variety of remix genres including R&B, 2-step, Trip-Hop, Chemical, Epic House, Underground Hip-Hop, UK Garage, Downtempo Dub, Nu-skool, Acid Jazz, Trance, Funk, Alternative, Progressive, Rap, Abstract, Urban Neo-Soul, Big Beat, Slo-Jamz, Drum 'N Bass, and many others.



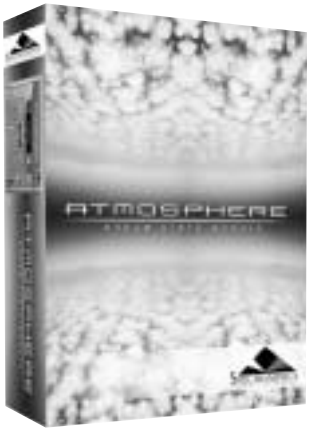
Each groove is presented in the widely-acclaimed Groove Control activated versions‹which allow you to change the pitch, tempo, feel and pattern independently ...without using any DSP! Since every groove can be mixed and matched at any tempo or feel, there are literally endless combination possibilities. You can also program your own grooves from scratch with thousands of edgy drum samples. In addition, the inspiring Groove Menus let you remix loops in real-time at every tempo from 50-180 BPM in both straight and swing feels. It's as simple as selecting a patch and playing your keyboard!

Includes over a thousand wild turntable FX and DJ tricks as well as a killer live percussion loop section including congas, bongos, djembes, shakers, triangles, agogos and tambourines, all of which can be mixed into any groove separately! But it is it's powerful interface and synthesis engine that make Stylus a monster creative tool. Perform techniques with Groove Control like randomizing the accents of a loop, turning a conga loop into a melodic part, radical filter sequencing, tweaking the tuning of just the snare, or filtering just the kicks, panning each slice in a different part of the stereo field, each all inside the loop! This is possible because the Stylus custom interface goes way beyond any of the current conventional samplers - each sample and slice has it's own adjustable synth parameters, and selecting the samples is as simple as playing the sample from your keyboard. The possibilities for creative manipulation are truly amazing... and yet it is so easy to use.

- Massive 3 GB core library
- Insane new remix grooves and samples by Eric Persing
- Over a thousand turntable tricks, skips, stops, beat-juggling and scratch FX
- Killer live percussion loops can be added to any groove
- Elastic Tempo, Pitch, Pattern and Feel with Groove Control
- Built-in Groove auditioning system
- Real-time groove jamming at any tempo with Groove Menus
- Lightning-fast load times
- 1000 Kicks, 1000 Snares, 500 Hi-hats for creating your own grooves
- Built-in patch management system
- Sample-accurate timing
- Fully programmable
- Total recall with your sequence
- Multimode resonant Filters for each sample, plus Master Filters
- Three envelopes for Pitch, Filter and Amplitude
- Matrix-style modulation routing, two LFOs
- Unique interface allows independent parameter control of each sample, and Groove control automation.
- Powered by custom 32-bit UVI Engine
- Cross platform, plug-in compatibility

B&H PHOTO - VIDEO - PRO AUDIO
ONE HOUR FREE PARKING
AT 349 W. 34th STREET (with purchase of $100 or more)

ATMOSPHERE - DREAM SYNTH MODULE



World's Most Massive Synth

Atmosphere is a software plug-in instrument that integrates a massive 3GB core library of more than one thousand unique sounds and layer elements, with a powerful and intuitive interface for shaping new textures. Atmosphere works as a native plug-in instrument in hosts like Logic, Digital Performer, Cubase, Nuendo and ProTools. Designed from the ground up by acclaimed synthesist Eric Persing (famous for his sound design of classic Roland synthesizers and award-winning sample libraries) and his team of sound designers, Atmosphere's core library was developed using over a hundred sound design devices and nearly every conceivable synthesis method including Granular, Additive, Wavescanning, Graintable, Neural Processing, Vintage Analog, Vector, Virtual Analog, FM, Plug-In manipulations, and unusual Acoustic sources as well. It is the custom core library that gives Atmosphere its unprecedented variety of tonal textures, especially when compared to standard soft-synths.



The range of sounds this instrument is capable of is astonishing, from lush ethereal pads, to powerful leads, crystalline glass swells to dark brooding atonal clouds, unusual synth basses, evocative ambiences, deep drones, complex textures, vintage analog sounds, disturbing noise FX, europhasers, gorgeous ambient string ensembles, thick trance/techno sounds, wavetable sweeps, vocoded choirs, dense vocal washes, transparent evolutions, and shimmering prisms of harmonic convergence. Puts even the best hardware synths to shame!

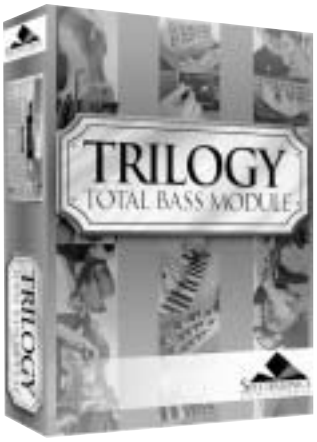
Stellar sound quality is this instrument's number one feature. The scope of sounds it can produce is very broad, making it extremely versatile in a wide variety of musical situations. This is made possible because there has never been a programmable synth released with such a massive, high resolution core wavetable to work with…completely dedicated to sound design elements (no stock pianos or oboes!)

The core library of a thousand powerful sounds is just the starting point for your own creations. The beauty of Atmosphere is in its "dual-layer" concept. You can immediately mix and match the layers of any of the 1,000 patches, and tweak each layer fully independently. That's over a million possible combinations! Each layer has its own multimode resonant filters, four LFOs, three envelopes and matrix modulation. There is also an additional master filter for quick tone shaping. The possibilities for creative manipulation are phenomenal... and yet it is so easy to use.

- 3.7GB core library offers vast array of textures to explore
- 1,000 patches by award-winning sound designer Eric Persing
- Dynamic layering of sounds for over 1,000,000 mix combinations
- Powerful interface and synthesis capabilities
- Built-in patch management system
- Multimode resonant Filters for each layer, plus Master Filters
- Three envelopes per layer for Pitch, Filter and Amplitude
- Fully programmable
- Total recall with your sequence
- Matrix-style modulation routing, four LFOs
- Powered by custom 32 bit UVI engine
- Cross platform plug-in compatibility

SPECTRASONICS

TRILOGY -TOTAL BASS MODULE



Electric, Acoustic and Synth Bass Instrument

Serious about Bass? This is the instrument! Trilogy is a programmable, sample-based sound module plug-in, that integrates a massive 3GB core library of hundreds of remarkable Bass sounds, with a powerful user interface for creating your own unique patches.

Trilogy's power lies within its layering concept. Beyond the core library of incredible Basses, every patch in Trilogy has two layers that you can tweak individually. You can also immediately mix and match any of the layers in the core library together, and edit each one independently. Imagine the power of combining the sound of a real Minimoog with a Fretless bass, or a Virus with a TB-303, or even adding a Juno sub-oscillator to an Upright Bass! There are thousands of earthshaking combinations, and you'll never run out of options.



The highly detailed Acoustic Upright Bass is one of the highlights of this instrument. Because of Trilogy's unique interface, you can get a wide variety of gorgeous tones from this Bass. This is made possible because the interface has separate control of the Neumann U-47 Tube Microphone signal and the Direct Pickup signal, which was sampled through a vintage Neve 1083 Console. This stunning sounding acoustic is chromatically sampled, with multi-dynamic velocity switching. The hardest velocity in every Bass patch in Trilogy brings in real bass glisses, similar in concept to the idea Spectrasonics pioneered on the Hans Zimmer Guitar libraries. An acoustic Martin Dreadnought Bass Guitar is also available in the Acoustic section -perfect for those "unplugged" sessions!

Electric Bass

The huge selection of electric basses feature classic 4, 5 and 6 string models, performed in Fingered, Picked, Muted, Rock & Roll, Slapping, Ballad, Fretless and R&B techniques through rare, custom-made tube preamps. Special variations are presented including Harmonics, Glisses, Fuzz, Trills, FX and thousands of slides! Both modern and vintage "Old School" electrics are available, providing an array of tones for different musical settings. All the patches are mapped the same way, so that one sequence will work with any bass you select.

There are thousands of samples used in these instruments, mapped with multiple dynamics for amazing expression. You don't have to "assemble" Bass parts with Trilogy, it's extremely playable as a highly expressive real-time instrument too!

Finally, every Acoustic and Electric Bass in Trilogy has a special "Finger Noise" release layer with release triggering, that gives these basses amazing realism -perfectly simulating the fretboard response of the real bass. When a note is released, the Finger Noise layer triggers an appropriate noise sample from among hundreds of tiny fret noises, subtle string scrapes, squeeks, x-notes, taps and mutes, which add a user controllable "human" imperfection element to your performance.

Synth Bass

Trilogy is one of the most awesome Bass Synthesizers ever made! Have you ever struggled with trying to fit your synth bass sounds to sit in a track properly? The sound of Trilogy is not only ultra-fat, but it sits great in a mix. High-resolution core samples were created from legendary analog bass synths like the Minimoog, Roland Juno 60, Roland TB-303 and SH-101, Oberheim SEM, Moog Taurus, OSCar, Virus, Yamaha CS-80, Arp Odyssey and 2600, Studio Electronics SE-1, Omega and ATC Tone Chameleon, Sequential Circuits Pro One, the mighty Moog Voyager and many others! Trilogy's multimode resonant filters allow you total control over the sound, while still retaining the original character and power of the classic analog instruments.

Each fully programmable layer has its own multimode resonant filters, four LFOs, three envelopes, adjustable sample start and matrix modulation. There's also an additional master filter for quick tone shaping.

Trilogy also has a smooth analog synth-style Legato triggering and Glide that software samplers lack. This allows for Synth Bass sounds that truely play like a vintage analog synth—and can also be used with any of the Electric or Acoustic Basses, for unique hybrid Bass sounds that you've never heard before.

COOL EDIT PRO 2.0

Complete Multitrack Recording Studio for Windows

More powerful than ever, Cool Edit Pro offers PC users powerful features such as real-time effects and track EQ, disk-at-once CD burning, MIDI and video playback support, MTC master generation, DSP effects, loop-based music composition, and a compressed loop file format—yet retains the ease of use that has won them thousands of users worldwide. Extremely affordable, Cool Edit Pro is ideal for everyone from the studio professional to the audio hobbyist. And with enough professional effects and tools to fill a room full of effects racks, Cool Edit Pro has everything needed for recording, editing, and mixing up to 128 high-quality stereo tracks with just about any sound card.



FEATURES

- 32-bit processing, 24-bit/192kHz and higher sampling
- Powerful wave editing with mutli-level instant undo and redo
- Multichannel encoder with WMA and WAV 5.1 file output
- More than 45 DSP effects to bring new life to your dry tracks with a depth and clarity previously obtainable in only the best studios around the world... now in your PC!
- Stretch loops using resample, time-scale, or beat splicing methods.
- Placement option allows the master fader to either control the levels before they go through the busses and the bus effects, or to serve as an output level control at the end of the signal chain.
- Move tracks in multitrack by right clicking and dragging on track controls
- Real-time effects:

– Track effects rack for each track

– Busses for track grouping with real-time effects on each bus

– Automation of some parameters over time (e.g. dynamic EQ and dynamic delay)

– Dedicated real-time EQ on each track with quick A/B comparisons

- Dedicated real-time EQ on each track with quick A/B comparisons
- Supports Red Rover as well as the Mackie Control, Tascam US-224, and Event EZbus
- Spectral Decay option makes possible much larger reductions in noise without adding unwanted artifacts.
- Loop-based song creation, including session tempo and key matching
- Customizable interface and look including dockable/sizeable windows
- Organizer window with tabs for open files, effects and favorites
- Dual monitor support
- Envelope control of Volume, pan, effects mix and some effects parameters
- Cue/Play lists, with auto-cue features
- Wave properties including labels, loop info, EBU extensions and sampler info
- Metronome
- Data Analysis, including Spectral View, Frequency Analysis, Phase Analysis, Statistic and Histogram
- Enhanced SMPTE/MTC master and slave accuracy
- MIDI trigger support for program control
- Supports over 20 file formats, Direct-X plug-in support
- High quality sample rate conversion; batch-processing and file conversion



For even greater functionality and ease of use, the optional Red Rover provides a fully featured dedicated remote control device (up to 10´ away) that operates seamlessly with the software.

Compatible with any sound card, Red Rover is a small USB device designed with ergonomic and space-saving functionality in mind. It features standard multitrack transport controls that seamlessly interact with and control Cool Edit Pro's on-screen displays. In addition, it provides unlimited access to all 128 tracks of possible audio on Cool Edit Pro.

Red Rover shows you all the key information you need to record and even do basic mixing right on its LCD panel. When you're done recording, take Red Rover back to the computer to use its transports while you mix and master. Once you start using Red Rover, you will wonder how you ever got along with out it.

GIGASTUDIO

Hard Disk Sampling Workstations



The incredible GigaStudio is the world's most powerful sampling tool — period. A complete Windows-based sample playback system for creating fully-realized audio productions with mixing and effects, GigaStudio is based on TASCAM's famous GigaSampler and its Endless Wave technology which allows samples to be streamed off a hard drive instead of being limited to RAM like other samplers. Instead of relatively small sample sizes that need to be looped, the Giga platform lets you work with HUGE sample sizes....up to 4.3GB! The resulting detail and realism must be heard to be appreciated. Plus, these enormous samples may be triggered with extremely low latency on all voices using approved GSIF interfaces, allowing for tight performances with truly musical feel. Features include dedicated MIDI input for real-time effects, high-quality 32-bit effects, and GigaStudio's DSP Station Mixing Console for professional quality mixing, eliminating the need for additional plug-ins or hardware.
Three versions of GigaStudio are available, from the economical GigaStudio 32 to the powerful GigaStudio 96 to the sophisticated GigaStudio 160, which is capable of streaming 160 simultaneous voices of audio. GigaStudio's advanced features make it ideal for professional scoring composers, engineers and producers, while the easy-to-use interface makes it a great choice for musicians and songwriters. GigaStudio is compatible with Windows 98/ME/XP/2000, and can be set up to play samples triggered from a sequencer like any hardware sampler.

FEATURES

MIDI Channels

They deliver 16 MIDI channels (GigaStudio 32) and 32 MIDI channels (GigaStudio 96) over two unique ports, as well as dedicated MIDI input for real-time effects and mixer automation, allowing the creation of massive performances with dynamically controllable effects. GigaStudio 160 delivers 64 MIDI channels over four ports.

Streaming Disk-based Voices

They provide up to 32, 96 or 160 disk-based, streaming voices of polyphony in real-time. In combination with your MIDI sequencer, you can create anything from a simple pop tune to an advanced, fully-orchestrated film score, with sound quality that's indistinguishable from recordings of performed instruments.

Sampling

GigaStudio's sample format conversion utility offers compatibility with WAVE format samples as well as CD-ROM libraries, including Akai S1000/S3000, SoundFont and DLS. Additional conversion tools are also available from third party developers.

Professional Effects Processors

High-quality, 32-bit effects are available on the GigaStudio 96 and 160 with the NFX real-time signal processing series (operates with true zero-latency performance with GigaStudio). The GigaStudio 96 and 160 includes the NFX1 Reverb/Multi-Effects, the GigaStudio 160 only includes the NFX2 Chorus/Flanger, the NFX3 Tap Delay/Auto Pan and the NFX4 EQ.

Mixing Console

GigaStudio's DSP Station Mixing Console includes Volume, Pan, Effect Sends and Insert points that enable professional quality mixing without the need for additional plug-ins or hardware. All parameters can be automated via MIDI and employ a 32-bit signal path to maintain pristine digital clarity.

Network Streaming Capability

Giga-compatible instruments don't have to be resident on the client hard drive. Users of 100-baseT or better networks can access mapped drives in real-time (not on GigaStudio 32), allowing instant access to Giga-compatible systems and instruments via LAN connectivity.

- Expandable sampling workstation
- DirectSound compatible
- Patented EndlessWave technology allows use of huge samples up to 4.3GB
- They support GSIF-compatible hardware with up to 32 audio channels (GSIF compatible soundcard required)
- Behavioral Sampling and Dimensions performance sampling technologies
- Multi-layering, dynamic, release-triggered sampling for soundboard resonance, staccato modeling
- Ultra-low latency (3-10 millisecond) performance via GSIF interface
- Enhanced support for GigaSampler file formats: intelligent auditioning, up-to-500% faster load times, extended instrument information
- Bundled sounds include:
  GigaPiano 1GB sampled grand piano,
  RetroDrums analog/digital percussion Set,
  Soundware Demo #1

GIGASTUDIO

Professional Features

With additional professional features like capture-to-wave (records entire performances, including effects, faders, pan, and MIDI automation directly to disk), support for 24bit/96kHz hardware, embedded help tools, an advanced instrument editor featuring synthesis tools like multi-mode resonant filters (low-pass,band-pass, high-pass, notch), multiple envelopes and LFOs, drag and drop sample assignment, and up to 32 samples per key—GigaStudio is the must-have tool for any serious composer or sound designer.

QuickSound Realtime Interactive Sound Database

For busy professional musicians, sound designers and recording facilities, GigaStudio's unique QuickSound technology enables instant location and previewing of samples, waves, and instruments in real-time via an intuitive, database-assisted technology for cataloging instruments and sound samples using plain language.

Just type in the type of sound you need — piano, drums, french horn and so on — and QuickSound will pull up all the samples that fit the description. GigaStudio literally makes terabytes of sound content instantly accessible and playable.

By instantly mapping the dozens of sound objects matching any search criteria to a MIDI control surface, automatically coupling these sound objects to an advanced audio streaming and DSP processing engine, and making each event audible in less than 10 one-thousandths of a second, QuickSound affords unrivaled power to scoring professionals.

GIGA-PCI

PCI-822 Interface Card with GigaStudio 96 Bundle



Combining Tascam's PCI-822 computer digital audio/MIDI interface card with the GigaStudio 96 software results in an complete package with software and computer interfacing that's ready straight out of the box. The PCI-822 offers compatibility with GigaStudio via its GSIF drivers, and is an ideal computer interface for those who use digital mixers and recorders with TDIF or S/PDIF digital interfaces. With the Giga-PCI package, you get the power of a full version of GigaStudio 96 along with computer interfacing for about the same cost of the card alone.

The PCI-822 is a PCI-based interface card with everything you need to send digital audio and MIDI signals to your computer for recording and sequencing. The 8-channel DTRS compatible TDIF interface and stereo S/PDIF interface supports 16-, 20- and 24-bit audio as well as 44.1 and 48kHz sampling rates. The two 16-channel MIDI ports provide 32 channels of MIDI I/O for keyboards, synth modules and mixer automation. The PCI-822 ships with GigaSampler LE — a powerful software-based digital sampler. The PCI-822 also includes ASIO and MME drivers which provide compatibility with most digital audio and sequencing software applications.

- 8-channel TDIF digital input & output
- S/PDIF digital input & output (RCA coax)
- 2x2 MIDI Interface (32 MIDI channel I/O
- 16, 20, 24 bit resolution
- Supports 44.1 and 48 kHz sample rates (internal) as well as input tracking from 39-51 kHz
- 75-ohm BNC Word Clock output for sample rate synchronization
- The PCI-822 has a software control panel for digital audio input status, sample rate selection, and clock source control.
- ASIO driver provides low-latency performance when adjusting parameters and monitoring inputs with applications such as Cubase. The PCI-822 also has a direct GigaSampler Interface (GSIF) driver.
- Includes a breakout cable for S/PDIF, MIDI and word clock connectors

Includes GigaSampler LE

- Multi-Gigabyte, 44.1k Sampler Playback
- GigaSampler Instrument Editor
- 32-bit Audio Signal Processing
- Resonant HP/LP/BP/BR filters with Dynamic Resonance
- Advanced Release Triggered Sampling

GIGASTUDIO SOUND LIBRARIES– *STRINGS/ORCHESTRAL*

Sample and Loop Libraries Optimized for GigaStudio

Taking advantage of GigaStudio's powerful architecture, Giga-compatible sound libraries offer more than most sample sets. Since only a small portion of the sample is saved in RAM, GigaStudio sound libraries don't need to be looped and compressed to work in playback. In fact, may Giga libraries, from acoustic pianos to strings, guitars and many more have no looping at all.....just masterfully-captured full recordings of instruments the way they were meant to sound. You'll find GigaStudio libraries for virtually any genre of music. For the most up-to-date complete list of Giga libraries, please visit the B&H website.



Garritan Orchestral Strings

One of the most detailed sound libraries ever developed for any sampler. Featuring masterful 24-bit recordings of the very finest string instruments (including Stradivarius, Guarneri, Montagnana, Testore, Gagliano and more) performed by distinguished virtuosos at Lincoln Center, this 16 CD-ROM set features over 8,500 captivating sampled string sounds with over 500 different instrument patches & combinations. This extensive library offers a myriad of playing techniques, complete coverage of the entire range of the sections, multiple dynamics and much more. It also adds extra touches like MaestroTools Auto-Alternator (an exclusive feature that alternates automatically between up & down bows or alternative samples), MaestroTools Legato Mode to help achieve more realistic legato phrasing, expression control, vibrato control and warmth control using multi-velocity cross-fading between samples.....**Call**

Garritan Orchestral Strings "Lite"



Garritan's world-renowned orchestral string library is also available in a more economic package. This 3 CD-ROM library includes two dynamic levels, with basic articulations and bowing including sustain vibrato, sordino, trills, detache and more. GOS Lite offers the same excellent quality level from the advanced GOS version, and comes with a free edition of Chicken Systems' Translator software.

TAGGSTRINGLT.....**224.95**



Denny Jaeger Master Violin Library

The acknowledged standard in sampled violins. Sections of 14 violins available in every dynamic level. Includes looped sustains, pizzicato, tremolo, half and whole step trills, slides and calegno. Patching allows control over tuning and bow strokes. Original comprehensive 3 disc and one disc CD-ROM sets.

TAQUPVIOLIN.....**224.95**

Sonic Implants Symphonic String Collection



Symphonic String Collection is a 20 CD-ROM (3 DVD) box set that enables musicians and composers to evoke the true essence of an ensemble string orchestra with unsurpassed clarity and ease. Recorded within the exquisite acoustic confines of Sonic Temple Studio in Roslindale, MA by Emmy and RIAA award-winning engineers Antonio Oliart and John Bono, this stunning collection showcases world-class musicians from both the Boston Symphony and Boston Ballet Orchestras. Rich, multi-dynamic instruments with down and up-bow samples, gbs of articulations, convenient switching and cross-fade parameters, and release samples which preserve each note's native resonant string decay make this an eminently playable and expressive tool. Great care was taken to preserve the aural integrity of each sample with a minimum of applied processing. The result is a collection which is truly an instrument unto itself and an essential vehicle for any composer who values their creative medium as much as their craft.....**Call**



Dan Dean Solo Strings

Dan Dean Solo Strings is a culmination of many months of recording and editing. Solo Violin, Solo Viola and Solo Cello were recorded in stereo in various articulations (arco/vibrato, pizzicato, half-step and whole-step trills, spiccato and tremelo). Each note of each instrument was recorded in each articulation. There is no pitch-shifting or unnatural transposition artifacts commonly found in other libraries. Also, since each and every note of the instruments appears unaltered by pitch shifting, the bow and instrument body resonances as well as the overtones characteristically generated, are true throughout the entire range of all of the instruments. This creates a smoother, more "organic" sounding bank with even note-to-note transitions. Instruments and banks on this collection include violin, viola and cello with arco/vibrato, staccato, pizzicato, tremelo, and half-step and whole-step trills.

TADDSTRINGS.....**149.95**

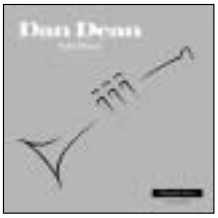
Symphonic Fields Forever



Beautifully evocative solo and small section orchestral instruments. For use in pop and classical orchestration as well as acoustic textures. Features superbly recorded, multi-samples of Celli, Violins, Choir, Silver Flute, Saxophone, Double Reed, Bassoon, Trombone, Tuba, Double Basses, Clarinet & Bass Clarinet, plus ensembles of these instruments.

TAQUPFIELDS.....**112.50**

GIGASTUDIO SOUND LIBRARIES — *BRASS/WOODWIND*



Dan Dean Solo Brass

- 10 CD set with up to 8 layers per note (ppp, pp, p, mp, mf, f, ff and fff).
- All notes chromatically sampled.
- Articulations include: NV, Portato, Staccato, Straight Mute, fp, fp long/medium and short crescendo.
- Vibrato on Trumpet and Piccolo Trumpet.
- Straight mutes on the Bass Trombone, Trombone and Trumpet.
- Advanced programming features include "ensemble" presets, full keyswitching, mod wheel switching implementation.
- Select between 2 light versions: LT1 retains all layers with less notes. LT2 retains all notes with less layers.
- Source captured in 24-bit stereo, UV-22 bit-reduced to 16-bit.
- Instruments exhibit extreme dynamic/timbral range

TADDBRASS **489.95**



Dan Dean Solo Woodwinds

10 CD set with up to 6 layers per note (pp, p, mp, mf, f, ff). All notes chromatically sampled. Vibrato and NV on most instruments. Extremely natural sounding.

TADDWOODWIND **409.95**



Pyschic Horns by Jason Miles

Beautifully evocative solo and small section orchestral instruments. Features superbly recorded, multi-samples of Celli, Violins, Choir, Silver Flute, Saxophone, Double Reed, Bassoon, Trombone, Tuba, Double Basses, Clarinet & Bass Clarinet. Includes long and short sustains, loops, riffs, swells, falls, & stabs. Mixed mode CD-Audio/CD-ROM and CD-ROMs.

TAQUPHORNS **74.95**

Bigga Orchestral Brass: French Horn



302 MB of French Horn. All wavs Full envelope/no loops. Two separate versions of every Instrument Preset allow unique, true Doubling Presets to be included. You get two separate Multi-Velocity Fr Horns and their Doubler Presets, plus separate Presets of each velocity layer (Piano, Mezzoforte, Forte) & Doubled Presets of each velocity type. In addition to multi-velocity main instruments, there are single velocity (Mezzoforte) versions of special attacks. Includes 2-Staccato, 2-Sordini, 2-Stopped, 2-Tongued variants & Doubled Presets of these variants. Also includes short & long crescendo presets + some "gliss" attacks.

TABGFRENCHHO **44.95**

Bigga Orchestral Brass: Trombone



440 MB of Trombone. All wavs Full envelope/no loops. Two separate versions of every Instrument Preset allow unique, true Doubling Presets to be included. You get two separate Multi-Velocity Trombones and their Doubler Presets, plus separate Presets of each velocity layer (Piano, Mezzoforte, Forte) & Doubled Presets of each velocity type. In addition to multi-velocity main instruments, there are single velocity (Mezzoforte) versions of special attacks. Includes 2-Staccato, 2-Sordini & 2-Tongued variants & Doubled Presets of these variants. Also includes short & long crescendo presets and some special "gliss" attacks.

TABGTROMBONE **44.95**

Bigga Orchestral Brass Complete



341 MB of Tuba, 441 MB of Trombone, 302 MB of French Horn, 348 MB of Trumpet. All wavs Full envelope/no loops. Two separate versions of every instrument preset allow unique, true doubling presets to be included. Get two separate Multi-Velocity lead Trumpets, Fr Horns, Trombones, Tubas and their Doubler Presets, plus separate Presets of each velocity layer (Piano, Mezzoforte, Forte) & Doubled Presets of each velocity type.

In addition to multi-velocity main instruments, there are single velocity (Mezzoforte) versions of special attacks. Includes Staccato, Sordini, Tongued variants of each instrument & Doubled Presets of these variants. *no sordini Tuba. *Fr Horn also includes "Stopped" version. A complete orchestral Brass solution in one package.

TABGORCHESTR **139.95**

Bigga Orchestral Brass: Tuba



341 MB of Tuba. All wavs Full envelope/no loops. Two separate versions of every Instrument Preset allow unique, true Doubling Presets to be included. You get two separate Multi-Velocity Tubas and their Doubler Presets, plus separate Presets of each velocity layer (Piano, Mezzoforte, Forte) & Doubled Presets of each velocity type. In addition to multi-velocity main instruments, there are single velocity (Mezzoforte) versions of special attacks. Includes Staccato, Tongued variants & Doubled Presets of these variants. Also includes short & long crescendo presets

TABGTUBA **44.95**

Bigga Orchestral Brass: Trumpet



348 MB of Trumpet. All wavs Full envelope/no loops. Two separate versions of every Instrument Preset allow unique, true Doubling Presets to be included. 2 separate Multi-Velocity Trumpets and their Doubler Presets, plus separate Presets of each velocity layer (Piano, Mezzoforte, Forte) & Doubled Presets of each velocity type. Additionally, there are single velocity (Mezzoforte) versions of special attacks. Includes 2-Staccato, 2-Sordini & 2-Tongued variants & Doubled Presets of these variants. Includes short & long crescendo presets and some special "gliss" attacks.

TABGTRUMPET **44.95**

GIGASTUDIO SOUND LIBRARIES — *PIANO*

Post Piano Suite V1



The main instrument is a Steinway D Concert Grand (the largest size Steinway makes) sampled with multiple velocity layers and sustain pedal sampled individually. Patented disk acceleration technique allow this 1.2 GB piano to fit on a single CD and there are 12 preset choices for special voicings in addition to the Full preset. (A Light preset allows a smaller version of the piano to load on less powerful systems.)

Not stopping at this, Michiel Post added a unique section of Orchestral perc instruments including Glockenspiel, Hackbrett, Marimba and Vibes. The samples are from the orchestra pit of the renowned Concertbegouw in Amsterdam, whose natural hall ambience adds a charming realism. Also includes a set of Electric Piano instruments, and a "Simple Piano" section with multiple preset choices such as Pianissimo Only, FF Only etc. A must for the accomplished pianist or sequencing musician.

TABGPIANOSUI **99.95**

The Holy Grail Piano



Finally, a sample piano that you can actually use in many different situations. Using a proprietary method of sampling called Virtual Model Sampling, this new approach offers you more realism than you've ever heard using samples. Every aspect of the piano including note let-offs, pedals and sympathetic strings are all controllable. The notes decay more slowly, and sustain longer. The samples are more resonant because more of the sypathetic resonances are supported. They've captured the actual change in harmonic content of each note over time.

- ◆ Set up for use in surround environments
- ◆ A real time V.M.S. Modeled Sustain component, identical to a real piano
- ◆ Controllable pedal/hammer action, resonant harp, and note on and off
- ◆ Multiple layers and multiple versions to fit any need
- ◆ Two different miking perspectives offered

TAQUPGRAIL **224.95**

Grandioso Steinway D



Recorded on the best grand found— a Steinway model D3 built in 1965 and fully refurbished by Steinway Hamburg in 1999. This piano served the Rotterdam 'DOELEN' concert hall for several decades, where hundreds of famous musicians, from Claudio Arrau to the Rolling Stones, performed for live audiences and broadcast concerts. Captured with up to 6 articulations (PPP, PP, P, MF, F, FF and FFF) for sustain pedal up, sustain pedal down and 4 articulations for the release triggered samples. Samples were recorded using Prism Sound, a 24-bit ProTools TDM Mix+ system and Waves processing. Mapped up to 16 levels of velocity, true multiple velocity release layers, ultimate staccato, and sustained pedal-down samples with a carefully chosen amount of resonance. The library has 5 GB of samples. The end result was tested by several concert pianists, who helped develop a sampled instrument to meet their highest expectations.

TASTEINWAYD **183.95**

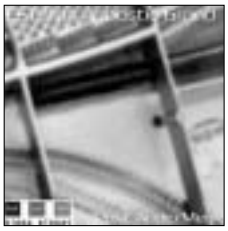
FortePiano



Play the same type of piano as Haydn and the other fathers of classical music. This Gigasampler instrument has multiple velocity layers/pedal down/pedal up AND separate Moderator layer, as well as True Key-Release samples. Listen to the MP3 demos and hear the actual key mechanisms just like the real thing. The largest, most ambitious sample version of FortePiano ever.

TABGFORTEPIA **69.95**

Estonia Acoustic Grand Piano



Estonia Grand pianos have been praised by the top pianists in the world. Now a Giga sample version with 4 velocity layers Pedal Down & Pedal Up plus 4 velocity levels of Release samples for extra expressiveness. Recorded by Michiel Post it is designed to take advantage of Gigasampler's best features. 15 instrument presets within the gig include the full version plus variations for Jazz, Rock Ambient etc.

TABGESTONIA **69.95**

GigaPiano



Over a GB of rich, stereo samples result in the most realistic sampled grand piano available anywhere. GigaPiano encompasses multiple velocities, rich soundboard resonance for damped staccato, as well as undamped pedal down samples for unparalleled playability. Recorded by Grammy-winning engineer, Larry Seyer, this is the piano that everyone is raving about, and is bundled with the 'full' version of GigaSampler.

TAGP **79.95**



Rain Piano

Sometimes a Grand Piano is simply too much or lacks personality. Rain Piano is professionally detuned to add character while staying in tune with your other instruments. A true 8 velocity X2 (pedal Up / pedal Down) The extra velocities provide smooth response from softest to loudest timbres. Recorded at Studio Kuling, Sweden using only the finest gear and mastered in 24-bit stereo.

TABGRP **69.95**

Studio Grand 88



A huge sample instrument with ultra realism from true 8 Velocity X2 (Pedal Up / Pedal Down) which provides extra responsive velocity layering, smooth transition from softest to loudest timbres. Malmsjo Concert Grand Piano. Recorded at Studio Kuling, Sweden. All samples 44.1kHz. Stereo originally mastered in 24-bit depth for ultra-low noise. 6 User Preset Voicings included.

TABGSG88 ..... **69.95**

GIGASTUDIO SOUND LIBRARIES — *PIANO*

Ultimate Yamaha C7 Grand Piano Library



The result of almost a year of hard work and dedication from Vintaudio, the Ultimate Yamaha C7 Grand Piano Library is the world's largest grand piano Library ever created. Vintaudio sampled the Best Yamaha C7 Conservatory Grand Piano they could get their hands on. They were so dedicated to getting just the perfect sound that the whole process was redone 3 times until we found the Ultimate Yamaha C7 Grand Piano. The concept was simple, often times composers and producers need a certain type of Grand Piano sound and the Piano libraries they have at their disposal are either too "wet" or too "dry". We decided to offer our costumers the works. With our library you get three 2.5 Gig sized Instruments: Close Miked, Ambient and Player Perspective. All 3 instruments were sampled at 6 Velocities Pedal Up and 6 Velocities Pedal Down with a Release Layer and modwheel controlled filter modulation for either a very bright sound to a really dark tone. All samples were recorded in 24 bits using only the very best audio equipment available today (Avalon, Millenia preamps and B&K Microphones). The results are simply incredible, all the range of expression from this beautiful instrument has been captured in stunning detail to offer what we believe to be the most complete Grand Piano Library ever assembled. Whatever style of Piano your project requires, you can achieve with this library. Pop, Jazz, classical, this library gives it all to you in one comprehensive library. At Vintaudio, we strive for excellence, innovation and most important of all, a reasonable price point that all musicians can afford.



Giga Upright Piano Collection

Vintaudio's Giga Upright Piano Collection is the largest Upright Piano library in existence today. Two wonderful uprights were captured, first a Brand new Steinway Boston Upright was sampled and then a 1908 Clinton Upright for that old school blues and ragtime sound. Each instrument is 2 gigs in size and was sampled at 4 velocities pedal up and down. A release layer was also created to enhance th realism of this wonderful library. The Steinway sounded so good that no processing or Eq was needed to make this piano sound as beautiful as it does.

GIGASTUDIO SOUND LIBRARIES — *GENERAL*

Conexant GM150 & GM500 General MIDI Kits





A general MIDI collection that gives you 128 baseline GM instruments, plus a drum kit for one low price. Pianos, Fender Rhodes, string ensembles, basses, French horn, pads, and ethnic instruments - everything you need to complete your arrangements, fully realized with the expressive potential of EndlessWave. The libraries come in two formats: the GM150 (150MB) for general use and the GM500 (500MB) for professionals.

GM150: *TAGM150* **69.95**
GM500: *TAGM500* **124.95**



Gary Garritan's GigaHarp

The instrument of heaven and angels, every string of a Salvi Pedal Harp has been sampled in stereo with four attacks/velocities per string, two harmonics per string and hand-dampening and muffling. Each pluck was recorded in 24-bit resolution using Neumann mics and Neve preamps, capturing the full duration of each string's natural decay. This library features 300 stereo samples totaling over a GB of harp sounds.
TAGIGAHARP **134.95**

giga_module vrsound

Offers a multitude of elaborate 3D instruments using hundreds of samples to achieve mixes of unprecedented sonic quality. The vrsound_giga_module is the beginning of a line of virtual reality instruments for the GigaStudio platform. It offers a variety of patches for immediate music composition and production. Instruments include: Rock drums, Funk drums, 4 pianos, 3 Fender Rhodes, various 3D synthpads, dirty electric bass, 88 key conga, timpani, small string ensemble, and more.
TAVRSMODULE **109.95**



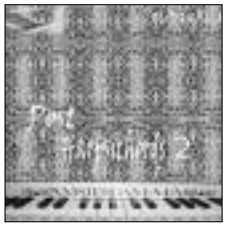
GIGASTUDIO SOUND LIBRARIES — *KEYBOARD/ORGAN*

Sune's L100 Hammond Organ



This is the new standard for sampled organs. Every note of every Drawbar setting recorded in extremely long looped samples averaging over 15 seconds BEFORE the loop begins... Fully controllable Virtual Drawbars screen in GigaStudio. This instrument is exclusively in GigaStudio Type 2 Format and extremely demanding on system resources. Best with a fast computer and at least 128 MB of RAM.
TABGHAMMOND **59.95**

Post Harpsicords Vol. 1 and 2



Two antique harpsichords captured in detail using world-class mics and mastered in 24-bit audio directly into Pro-Tools. Perfect for keyboard purists seeking to reproduce the great early keyboard compositions on the instruments for which they were written. One Flemish and one French harpsichord are covered in all their voicings including true "release" samples and even special voicings such as the "Lute" settings. All samples are full-envelope and the total library runs to over 700 MB compressed with easy install instructions.
Vol. 2 of Post Harpsichords contains 517 MB of Gigasampler format all dedicated to capturing a "Virginals" keyboard in every aspect. Like Vol. 1, this library uses true "release" samples and is the most comprehensive coverage available for this historically important instrument
Post Harpsicords Vol. 1: *TABGHARPSICO*..**69.95**
Post Harpsicords Vol. 2: *TABGHARPSICQ*..**56.95**

Purgatory Creek
Rhodes for GigaSampler / GigaStudio



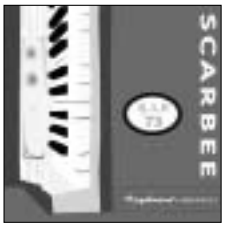
554 MB native Gigasampler format instrument with 4 velocity layers and separate release triggers! Classic 70's Fender Rhodes 73 Suitcase Model A real Rhodes that won't break your back or your wallet .....**Call**



Hammond B3 Organ

Since the first introduction in 1935, the Hammond B3 Organ has been favored by rock, blues, gospel and jazz artists alike. And for good reason, it's an incredible sounding instrument! Sonic Implants Hammond B3 Organ CD is a massive collection. 45 different Hammond B3 instruments all run through Leslie Speaker and programmed by one of Boston's best B3 players. Long loops, stereo samples, complete keymaps, slow, fast and stopped rotor variations give you a ton of sounds to choose from. Giga Crossfades move smoothly between stopped, slow and fast rotors for ultimate playability. This B3 isn't the pristine, clean, and pretty B3 sound you often find in a sample collection or soft synth. Instead, we chose to record a B3 with some character, some grit; what a B3 really sounds like after its been played a lot and gigged a lot. Documentation includes complete drawbar settings.
TASIHAMMONDB **114.95**

Scarbee R.S.P. '73



Master sample developer Scarbee brings you one of the most comprehensively sampled Rhodes electric pianos ever made with the Scarbee R.S.P. '73. Its depth and sophistication captures elusive soul of the original Rhodes!
The R.S.P. '73 consists of a collection of truly authentic 'deeply sampled' renderings of the classic Fender Rhodes Stage Piano MK I - '73, captured in high definition 24 bit quality. The piano sound was captured directly from the harp, bypassing the passive tone control on the front panel, in order to capture the full frequency range of the instrument and minimize unwanted noise. All sample editing - including noise reduction and normalizing - was also done at 24 bit resolution.
All samples on R.S.P. '73 are full length and unlooped. Each of the 73 keys of the Rhodes was sampled at 12 different velocities, ranging from 0 to -26 dB...but that's only half the story. In order to capture that magical feel of really 'playing' a Rhodes, Scarbee also recorded 'release' samples for each of the 12 velocities, at the exact same level as the corresponding 12 sustain samples. The release tone - a sound created when a damper bounces against the tine as a key is released - is an intrinsic element of what musicians recognize as the original 'Rhodes Sound'.
For that duller 'old school' sound, a special Hi-cut version of each program is also included. Plus, all programs are available in both Equal and Stretch tunings. Each Giga version utilizes 1752 samples and weighs in at 1.5 GB. Each key controls 24 unique samples: 12 different velocities of samples for the sustained notes, and another 12 velocities of samples for the release samples. If you're looking for the ultimate Rhodes sound for Giga, you need to check out Scarbee's R.S.P. '73!
TARHODES **149.95**

GigaStudio Mastery Tutorial



Dave Govett, one of the developers of GigaSampler, has created a 4-CD multimedia tutorial that details GigaStudio. Each aspect, from setup through use of the Instrument Editor through making the most of the Quick Sound feature is covered.

CDxtract—GigaEdition

CDxtract was designed for musicians who want to manage and convert their Roland, Emu, SampleCell, SoundFont, and AKAI sample libraries to Giga. For example, an AKAI CD can store hundreds of programs and samples. To find the right sound you usually have to load them one by one in your sampler, which takes a lot of time. CDxtract reduces the time needed. Spend more time playing music!
A sampler library is an expensive investment and usually if you buy a new sampler that uses a different format then you probably would have to buy the same sampler library again. CDxtract allows you to convert your sampler libraries between popular sampler formats thus protecting your investment. Moreover, CDxtract cost the same as a single sampler CD.

GIGASTUDIO SOUND LIBRARIES — *PIPE ORGAN*

Post Positif Organ



The classic "Positif" organ captured with and without Release samples. It was recorded in full 24-bit digital audio quality using the worlds best mics, converters and recorders. Incredible attention was paid to every detail during editing. The recording was done using DPA 4000 cardioids & omnidirectional microphones going straight into ProTools, keeping both close and ambient microphones on separate tracks. The historic organ sound benefits in terms of realism from the note for note sampling. All typical characteristics of this instrument are captured. The release sample layer was added to achieve true realism!

**Please note, Gigasample libraries are 16 bit, though capture may be achieved at 24-bit resolution.*

TABGPOSITORG **49.95**

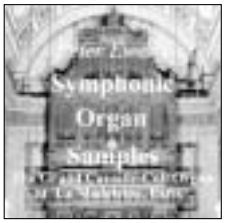
VRSound 3D Pipes



The instrument represented on this CD is The McLane Organ at the Baylor University, Texas. 92 ranks 4 manual designed by Petty Madden Organ builders in 1993. The presets are a reflection of organ stops used by the great performer Dr. Joice Jones. This organ is so realistic that not even she could tell the difference between live and "memorex".

TAVRS3DPIPES **299.95**

Peter Ewers: Symphonic Organ



Bring the entire, historic, grand La Madeleine, Paris cathedral and the Cavaillé-Coll organ into your studio! For the first time ever — in any sampled pipe organ — the original cathedral ambiance is included via release triggered samples. Combinations and pedals, all windchest activity, all interferences, and all bellow noises are captured with breathtaking accuracy. Well just hear the difference.

TAPEORGAN **134.95**

Post Organ Tool Kit



The POST ORGAN TOOLKIT is one of the largest and most versatile collections of high quality organ samples ever assembled! There are 111 Gigasampler instrument files (291.4 MB); all perfect sounding, looped, and most in full stereo, divided into 6 folders: Solo Instruments, Tutti & Pleni, Max Organ, Simulations, Italian, and the Concertgebouw Organ By combining several programs simultaneously you can create every possible registration any organ can produce...!

TABGORGANTOO **69.95**

The Denny Jaeger Private Collection



Audio quality and patching that belong in the 21st century. Proven in high-end record, TV and film production. A comprehensive collection of high power, high utility sounds featuring keyboard pads, powerful orchestral string stabs, lyrical English horn and bassoon, vocal sounds, SFX & percussion.

TAQUPPRIVATE **149.95**

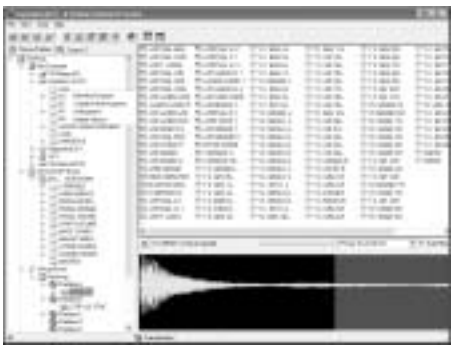
Post Theater Organ: Volume 1



In 1935, Dutch organ builder Strunk built an immense organ in the Amsterdam City movie theatre. It consists of three rooms. The main room counts 11 ranks of pipes; the first Solo room has 7 ranks and the 2nd Solo room had 8 additional ranks, but was lost 10 years ago. Samples are 10 to 15 seconds long with perfect loops to enable playing of unlimited length of notes. The true release sound of the organ pipes was also recorded and is reproduced upon key-release. All Preset programs have 50 multisamples recorded in stereo, with the true release reverberation tail recorded in the theatre. The programs are around 100 MB each, making the total size of these individual stops approximately 1.4 GBs

TABGTHEARORG **69.95**

Chicken Systems Translator for Giga



Roland, Emu, Ensoniq, SoundFont, and AKAI file conversion to Giga products. (Kurzweil, SampleCell, and others coming soon.) Use your sampler's hard drives and CD-ROM's with Giga products. Translations are precisely optimized for the best possible sound - keymaps, looping, all possible program parameters, even effects.

Browsing any sample disk or file system is familiar, quick and easy. See your disk and PC files in a tree-view fashion, exactly how it is laid out on the disk. Translator allows you to audition any sample directly from disk or file, within the PC's hard drive or proprietary hard drive, allowing you to find a specific sound. Single file or bulk translations (entire disks, folders, volumes, banks, directories, etc.) are available. For example, you can convert a Roland Performance to a .gig file with a single Giga instrument, or multiple Giga Instruments representing each Roland Patch contained in the Performance. Convert a single Emu E4 Preset, or a whole Bank. **69.95**

Many of the special utility functions of the full version (see Chicken Systems Translator on page 1065 for further details) are also included.

GIGASTUDIO SOUND LIBRARIES — *SYNTHESIZERS*

ARP Solina Strings



Created by Worra of Bigga Gigga, this is the classic ARP String Synthesizer of the late 70's faithfully captured in all its analog warmth. Every note of every Factory Preset is sampled! It is like buying the actual synth in perfect working order for an unbelievably low price. Now you can add the authentic sounds of this much sought after synth and take advantage of the awesome increase in polyphony made possible by Gigasampler. There are multiple variations of each Preset available through convenient "Key-Switch" mapping, making it possible to go from the stock, factory sound to a huge pad at the touch of key and without missing a beat. Stacking these warm analog sounds with modern strings and pads leads to endless hours of creative fun, searching for that perfect mix of old and new.

TABGARP **45.95**

Vintage String Boxes



Stunning strings, vocals, flute and brass samples from the famous Mellotron and Arp String Ensemble. Today we rely on digital samples and modern synthesizer technology for the culmination of realistic instrument sounds. But before there were microprocessors, there were string boxes. The Mellotron is one of the most well known and easily recognized early legends. It's signature sound has shown up on hundreds of albums from the Beatles to Beck and has spanned decades of musical changes without ever going out of style. The Arp Solina String Ensemble is a product of the late 70's. This keyboard has been made famous by Elton John, The Eagles, New Order, and The Cure to name few.

The CD contains a choice selection of sounds from the Mellotron tapes. All 8 seconds of every note is included! (Small memory versions also included.) Three different string recordings from the Mark2, the M200 and the M300 plus the Mixed Choir and of course the infamous Mellotron Flute. The Solina Arp String Ensemble was recorded and programmed to give you every sound and knob variation the instrument is capable of. Fantastic pads, filter sweeps, FX and leads.

TASIVINTAGE.....**54.95**

Dex Rex Collection



With 4 oscillators, 8 waveforms assignable to each, 8 algorithms possible per voice plus 3 assignable LFO waveforms, the Yamaha DX11 could do more than any FM synthesizer. And it was the quietest by far. It can duplicate any sound ever made by any previous FM synth and almost all the original analog sounds such as Moogs and ARPs. It's kiss of death was that it had no onboard FX processor. So what happens when you apply OUTBOARD FX to the Yamaha DX11 and give it 64 note polyphony as a Gigasampler instrument? You get some of the best sounding FM ever.

Fat pianos and clavs, killer organs, great lead synth sounds, and even some big analog strings that rival modern synths and have unique qualities .....**Call**

Roland JX-3P Synthesizer



Over 950 MBs of classic synth samples contained in a single 76 preset multi-instrument Gigasample file. All 32 original factory preset sounds are covered plus many custom created pads, filtered sounds and more. The JX-3P was Roland's first MIDI capable synth in 1983. Now you can own every sound from this timeless classic and enjoy the advantages of modern polyphony and the endless possibilities by "stacking" these sounds. All samples are long envelope and retain the internal volume scaling of the original hardware synth, so you will get the exact "feel" and easy playability of the original and all the subtle nuances that only really long samples can provide.

TABGJX3P **44.95**

ATTACK SYNTH: MemoryMoog



Another huge native Gigasample format library from Bigga Giggas' Vintage Series of classic synths. The MemoryMoog library contains 101 Presets in 33 Gig files. Total size is 1.08 GB. Grab some vintage "feel" at a great price!

TABGATTACKSY **44.95**

Worra's Prophet: Sequential Circuits P-5



It set the standard in the 80's, yet the Prophet 5 is still loved by many. 5 voice polyphony - 2 oscillators per voice and a white noise generator. The analog filters, envelope and LFO all sound great and are extremely flexible. There are 36 gigs on the CD, ranging from ultra-punchy analog basses to spacey pads.

All the samples were recorded digitally directly into the PC. The sounds that have no infinite sustain are full envelope recordings. The sample ends only when the synth note faded on its own. The voices that would sustain indefinitely ARE looped, but the loop point doesn't occur until 10-12 seconds into the sample so in most cases you'll play the unlooped sound. Each Prophet voice has been sampled in thirds, C, D#, F# and A. The Prophet had no velocity sensitivity, but Gigasampler adds velocity sensitivity by default, which helps make the voice more expressive.

TABGPROPHET.....**23.50**

Monster Matrix



2 full CDs of native Gigasample sounds. 27 Gig files, 101 Preset synth voices. The definitive library of classic Oberheim Matrix-12 voices. Leads, Pads, Basses; all are extremely long source samples to take full advantage of the Gigasampler's power. Sounds designed and recorded by SonicEmulations; Gig files tweaked and post-edited by Bigga Giggas.

TABGMONSTERM.....**44.95**

MiniMoog



A native Gigasampler format collection. Over 500 megabytes of pure synth sounds. Includes many of the most sought after Moog sounds; FX voices, basses, fat pads & great leads. Nothing compares to the real thing unless it is a Gigasample version that allows 160-note polyphony!

TABGMINIMOOG **44.95**

B&H PHOTO - VIDEO - PRO AUDIO
ONE HOUR FREE PARKING
AT 349 W. 34th STREET (with purchase of $100 or more)

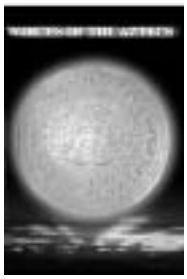
GIGASTUDIO SOUND LIBRARIES — *WORLD ETHNIC*

Silk Road



The Silk Road is the most well-known trading route of ancient Chinese civilization. Spanning 7000 miles through China, Central Asia, Northern India, and the Parthian and Roman Empires, the trade route blended culture, music, art and religion for centuries. Performed by Middle Eastern musicians, the Silk Road CD captures 50 authentic melodic and percussion instruments from this region. The CD includes single note instruments and percussion loops and melodic improvisations. Use the loops & improve in your pieces or as guides to the style. Melodic Instruments include: 3 Kawals, 3 Nays, May, Zurna, Arabic Violin, Bouzuk, Kanuun, Oud, Persian Violin, Persian Sitar, Santur, Tar. Percussion Instruments include: Egyptian Dumbek, Clay Dumbek, Bass Dumbek, Low & Hi Tombek, Low & Hi Udu Drums, Small-Med-Large Negato Drums, Copper Darbuka, 14″ Frame Drum, 22″ Frame Drum, Persian Daff, Bandir Frame Drum, Small & Large Bell Shakers, Egyptian Riqq 1 & 2, Asian Zill, Indian Zill, Turkish Zill, Sufi Triangle, Ankle Bells, 3 Copper Bowls, Small & Large Tibetan Bowls, Morocan Gargaba, Vessel Shakers.

TASISILKROAD **189.95**



Voices of the Aztecs

Add ancient culture to your music... Voices of the Aztecs is a full compilation of Aztecs instruments, loops and vocals. These samples were recorded in Mexico from archeological artifacts. From the big Huehuetl drum to the smallest of Ocarinas, all of the sounds are unique. Water drums and wooded percussion, clay flutes and incredible sound of the sea shell horn all remind us of a distant culture disappeared 400 years ago. Voices of the Aztecs also includes Nahuatl poems recited by Santos de la Cruz, a poet of Aztec descent. The booklet included offers a cultural history of the Aztecs.

TAQUPAZTECS **149.50**







NFX2 Chorus/Flanger

Complete range of modulation algorithms with spectral qualities ranging from smooth, rich chorusing to stiletto-sharp flange effects. Full control and MIDI automation of wet/dry levels, delay time (stereo), feedback, LFO rate, LFO depth and phase adjust. Screenshot, Expert Opinions.

NFX3 Tap Delay/Auto Pan

Quad multi-tap delay algorithm with individual delay time, level, pan, feedback and damping per tap. BPM mode allows entry of delay times in musical terms (beats per minute, note values). Screenshot, Expert Opinions.

NFX4 EQ

NFX4 is a semi-parametric stereo contour Equalizer. This effect gives you control over low-frequency shelving, mid-range, crossover point (low-to-mid) and high-frequency shelving.

Spirit Voices: Native American Flutes



The unforgettable sound of handmade Native American flute captured by Werner John of Early Light Music. Werner's flute music CDs have sold over 150,000 copies. Now his handmade Native Flutes are available as Gigasample instruments. Multiple attacks such as Legato, Legato with vibrato, Tongued, Tongued with vibrato plus one super Gig that contains all these attacks at three velocity levels. Five separate flutes in all.

TABGFLUTES **59.95**

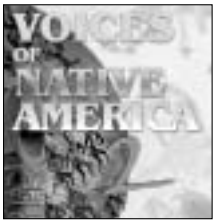
Voices of Native America Vol.1



An authentic collection of Native American & indigenous instruments. Includes percussion, Native vocables, wooden & clay flute samples & performances, hand drums, rawhide drums & shakers. Vocal phrases are performed by members of the Navajo (Dine'), Crow, Shoshone-Bannock, Hopi, Apache & Nez Perce Nations. Flute and drum performances by Douglas Spotted Eagle. Produced by Douglas Morton.

TAQUPNATIVE **149.95**

Voices of Native America Vol.2



Sample producers Douglas Spotted Eagle and Doug Morton have teamed up again. This volume focuses more on component-based loops and rhythms of Native America. Instruments include deer toe shakers, Taos Drums, Social Drums, and shakers. Flute phrases, single hit big drums and vocals are also included.

TAQUPNATIVE2 ..... **187.50**

Voices of Istanbul



Welcome to the world of Turkish sounds and rhythms. Recorded in Istanbul, this collection features single hit and multi-instrument loops of Turkish music.

TAQUPISTANBU.. **149.95**

GIGASTUDIO SOUND LIBRARIES — *GUITAR/BASS*

Jim Corrigan's Nashville High-Strung Guitars



From pioneering synth visionary, Jim Corrigan, comes the coolest, most playable acoustic guitar collection for GigaSampler. Recorded using proprietary studio techniquest, this totally authentic collection of up and down strums represents the sound of Nashville at its finest. This incredible library also includes dynamically playable single strings for solo parts.

TANHSG1.....**124.95**

Jacaranda Guitars: Vol. 1



The standard of excellence for acoustic 6 string guitar samples. Fingerpick and Flatpick solo gigs and 12 different Chord Types recorded with down and upstrokes mapped so that you can strum them yourself. Popular chord types include up to three inversions, so you can jump octave registers like real guitarists. Includes Midi Templates and demo songs, GSP Performance Presets for GigaStudio users and extensive Help File. Over 1400 MBs of 44.1kHz Stereo Gig files.

TABGJACARAND.....**109.95**

Whole Lotta Country



Get ready for a Hoe Down! The definitive collection of country instruments, plus the most outstanding array of country/ acoustic loops phrases and riffs ever assembled—and all performed by country guru Larry Campbell. Pedal Steel, Banjo, Fiddle, Dobro, Acoustic Guitar, High String Guitar, Mandolin, Electric Guitar, 6-String Bass, Tub Bass 1,133 events - More than 2 HOURS of loops and notes. Pre-looped, ready to sample & 100% copyright free. 2 CD-ROM set.

TASHCOUNTRY.....**189.95**

Matt Ragan: Max Strength Acoustic Guitar



With over 1200 discreet, unlooped samples, the musical character of the beautiful Martin 000-16 is expressed in every possible hammer-on, pull-off, palm mute, release-damp, or slide combination with any open string or fretted sustain in 6 velocities - double-tracked effects, and other variations are also included for production versatility.

TAMRSTEELGTR.....**109.95**

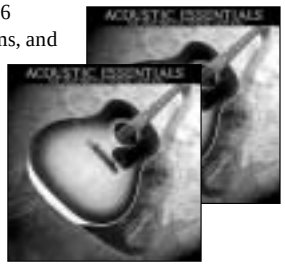
Pocket Syndrome Vol.1 Funk Guitar



Groove Master Bernd Schoenhart brings his unique "In The Pocket" guitar feel to this copyright free sample disc. Vol. 1 contains 74 minutes - more than 700 "in the pocket" guitar loops at multiple BPMs and keys per loop. Bernd played loops in more than 20 unique styles, ready to give your tracks the "Pocket Syndrome Edge". Your tracks will be come alive with that "funky guitar feel" - that you've only dreamed of achieving with GigaSampler - until now!

TASHFUNK.....**114.95**

Accoustic Essentials: Standard & Premium Bundle CD



Vol. #1 consists of 6 string guitar, Drums, and Acoustic Washburn Bass. Vol. #2 consists of the entire 6 & 12 string acoustic guitar collection. Guitars include pristine, single-note instruments as well as unique chord strum .gigs. Drums include Snare .gigs of Gretsch,Tama, and more....Cymbal .gigs of Zildjian rides, crashes etc. There is a full .gig of Timbales. The KIT .gig has that hard to find, true acoustic feel. Bass is the classic acoustic Washburn guitar. A must- have essential for authentic acoustic recordings.

Standard: *TABGSTANDARD*.....**44.95**

Premium: *TABGPREMIUM*.....**59.95**

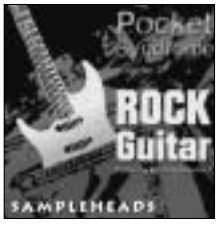
Heavy Guitar Library



It ain't pretty! This is a grungy, harsh, ruthless collection of guitar samples.... leads, mutes, scrapes, scratches, power chords, slides, feedback, harmonics and more. Insanely produced by Reek Havok and Sounds Amazing. Bonus 60Hz hum sample included on CD-ROMs.

TAQUPGUITAR.....**109.95**

Pocket Syndrome Vol.2 Rock Guitar



Bernd Schoenhart (John Secada, George Michael, Kenny G, Slash, Marc Anthony, Dave Valentin, C+C Music Factory) does it again, this time focusing on rock guitar feels. All new unique "In The Pocket" rock guitar loops and riffs at multiple BPMs and keys per event. All loops are calculated to 1 or 2 bars, ready to load into your sampler or computer. Once again, Bernd performed in more than 20 unique styles, grouped by BPM, ready to give your tracks the "Pocket Syndrome Edge".

TASHROCK.....**114.95**

Amps and Pick-ups - Complete Guitar & Bass Collection



Sonic Implants guitars and basses are unlike any you've heard. Awesome sonic quality, sampled tones, chords, tones, upstrokes, downstrokes, harmonics, etc. for incredibly authentic guitar tracks. And basses with so much variety youll find anything you need here.

TASIAMPS.....**114.95**

B&H PHOTO - VIDEO - PRO AUDIO

GIGASTUDIO SOUND LIBRARIES — *GUITAR/BASS*

Dan Dean Giga Bass



This the Gigasampler version of the highly regarded Dan Dean Bass Collection #1 and #2 combined with additional bonus banks not found on either of the first two disks. What's different about our samples? First, each and every note of all of the instruments has been sampled. This means that there is no pitch shifting, stretching or any other alteration of the original sample data. The notes that were played are the notes that you play. The body resonance is constant. The string overtones are constant. The harmonic content of each of the notes is consistent, and so on. This all translates into digital instruments that are more natural sounding.
Basses include Alembic Spoiler Bass, Toucan Fretless Bass, Guild Pilot 5-String Bass, Fender Jazz Bass (60's vintage), Fender Precision Bass (60's vintage), Rickenbacker 4001 Bass, Hofner Violin Bass (60's vintage), Gibson Thunderbird Bass, Alembic Spoiler 5-string Bass, Washburn AB-20 Fretless Acoustic Bass Guitar, Guild Ashbory (Rubberband) Bass, Hohner Fretless Acoustic Bass Guitar and an Acoustic Bass (german 100+ years old).

TADDGIGABASS.....**149.95**

J-Slap & J-Fingered Electric Bass "Light"



Now get bass samples for Giga at a more affordable price. Based on the full versions of Scarbee J-Slap and J-Fingered the handmade Celinder J Update 4 has been recorded without the use of compression, tubes or advanced EQ to ensure you maximum flexibility in the sound production.



With the J-Fingered Light you can play sustained notes, do hammer-ons, pull-offs, fast grace-notes, staccato-releases or slides, using only the mod wheel and the foot controller. You can also add all kinds of fretnoise and pick-up clap to you bass line to make it sound even more real. 423 samples are dedicated to each of the 3 pickup settings: bridge, both and neck, giving you the total of 1269 samples (about 496 MB on this CD).

J-Slap Light gives you the chance to thumb and pluck sustained notes, do hammer-ons, pull-offs, fast grace-notes, staccato-releases or slides using only the mod wheel and the foot controller, like J-Fingered Light. Add all kinds of fretnoise and string clap to you bass line to make it come alive. 524 samples are dedicated to each of the 3 pickup settings: Bridge, Both and Neck, giving you the total of 1572 samples - about 464 MB on this CD.

The programming of J-Slap Light is almost identical to J-Fingered Light as these two libraries are meant to be used as one bass sound.

J-Slap Light *TAJSL*...**Call** **J-Fingered Light** *TAJFL*....**Call**

Larry Seyer: Upright Acoustic Bass



This library is over 500 MB in size, dedicated to this one fine classic Italian instrument. Every note of every string was sampled in stereo at 4 velocities with no loops. Features 'finger damped' staccato release resonance samples that will play on the note-up (release) and body resonance volume controlled by the Mod Wheel. There are fast and slow slides up and down all four strings, up and down from each note. Also included are some riffs, special effects, harmonics and noises.

TALSUPRIGHTB.....**109.95**

Will Lee's Ultra Freakin' Bass Rock



A collection of licks, riffs & patterns inspired by many bass influences, as well as "conditions of the moment" stumbled upon at the sampling sessions. Sampled with an array of some of my favorite cool effects pedals handy including the Boss Octaver (the "Brown Pedal") and Digitech's DOD Envelope Filter (the green pedal) and Whammy (the purple one). Also used was the spring reverb from Will's Fender Deluxe Reverb amp. This library also contains samples of 5 of Will's favorite instruments, sampled all kinda ways to give you tools with which to make music at a high level of realism.

TASHFREAK.....**114.95**

Scarbee J-Slap & J-Fingered Electric Bass



Novel sampling techniques and pristine audio signal quality combine to make the Scarbee J-series the standard in sampled electric bass libraries. Designed to maximize the enormous sample capacity, high polyphony, and pristine 32-bit signal processing exclusive to the GigaStudio system, the Scarbee J-series bass libraries provide levels of interactivity and musicality not found in other sampled electric bass libraries.

For the Scarbee J-Slap bass, more than 900 samples are dedicated to each of the 3 pick-up settings: bridge, both, and neck, providing a total of 2700 samples - or about 800 MB on the CD. The musicality of this handmade Celinder J Update 4 is expressed in every possible hammer-on, pull-off, grace-note, staccato-release or slide, with any sustain, mute, thumbed or plucked note combination reproducible in every authentic context.



For the Scarbee J-Fingered bass, 1046 samples are dedicated to each of the 3 pick-up settings: bridge, both, and neck, providing a total of 3138 samples, or 1.15 GB on the CD. As in J-Slap, the musicality of this handmade Celinder J Update 4 is expressed in every possible hammer-on, pull-off, grace-note, staccato-release or slide; with any sustained note combination reproducible in every authentic context.

For the first time, the sustained bass notes have been sampled with both right hand index and middle finger (up to 8 sustained note per note), providing very realistic sounding bass lines. This library also contains 2 different sets of staccato-release samples: loose and tight which really brings life into your bass playing. J-Fingered has also both soft and hard legato-tails, even more slides than J-Slap and a complete new set of fret-noises. The programming is almost identical to J-Slap as these two libraries are meant to be used as one bass sound.

J-Slap *TAJSLAPBASS*.....**104.95** **J-Fingered** *TAJFINGEREDB*.....**104.95**

GIGASTUDIO SOUND LIBRARIES —*PERCUSSION/LOOPS*

Latin Groove Factory

The standard for Latin Music and percussion samples. This is the most comprehensive 3 volume collection of authentic traditional and contemporary drum and percussion grooves from Cuba and Brazil. Grooves from the cultures of Puerto Rico, Trinidad, and the Dominican Republic are also included in this smoking set of drum and percussion loops and samples. Produced by experts in the Latin music field, these CDs offers the maximum amount of feel and groove allowed in samplers today. The project is laid out in such a way that even the lay person will be able to understand and use "clave".

Vol. 1 Afro-Cuban



Styles represented include: Mambo, Cha-Cha-Cha, Songo, Guaguanco, Bomba, Merengue, Mozambique, Bolero, Salsa, Sambas - Cruzado, Funk, Partito Alto, & Batucada, Bossa Nova, Bairo, Calypso.

TAQUPAFROCUB.....**149.95**

Vol. 2 Brazil



Styles include Baion 96, 116, 132. Batucada 96, 116 ,138, Bossa Nova 120,142, Brazilian Pop 96, 116, Marcha 152, Partido Alto 96, 108, Samba Cancao 96, Samba Cruzado 76, 96, 116, 142, Samba Funk 96, 116. Plus multi-sample single hits of percussion instruments.

TAQUPBRAZIL.....**149.95**

Vol. 3 - Caribbean



Styles include Caribbean: Calypso 96, 108,128 Bomba 90, 108 Merengue 124, 148 Afro-Cuban Traditional 6/8 traditional: Bembe 108, 128 Guiro 108, 128 Yongo 108, 128 Mozambique 108, 124 Guaguanco 76, 100, 116, 138 Guaguanco Matanzas 76, 100, 116, 138 Bolero 76 Cha-Cha 120, Danzon 120 Son 120 Afro Cuban Contemporary Bata Songo 116 6/8 Popular 108, 128. 4/4 Bata: ChaChaElekefun 116, 132 Rumba Iyesa 116, 132.

TAQUPCARRIBN.....**149.95**

Dave Samuels - Marimba & Vibes



Dave Samuels is a multiple Grammy award winner who has established himself as the top mallet player of his generation. He is recognized for his fresh new sound and creative approach to both the vibraphone and marimba. This disc contains the highest quality multi-samples of marimba and vibes ever recorded! Multiple mallet variations (timbre) and dynamics. Chromatic samples of the entire range of the instruments: Yamaha YV3710 _ 3-1/2 octave vibe and the YM5000 _ 5 octave rosewood marimba. Sustained Notes, Muted Notes, Rolls, Riffs, and more!

TASHMARIMBA.....**114.95**



Christian/Lane: Ultimate Marimba & Vibes

This two disc library features a Marimba One five octave rosewood marimba, and a Musser Gold Century set of vibes. With four sets of mallets to choose from for each instrument your composing possibilities are endless. The marimba patches include four velocities strikes, rolls, muted strikes, and rattan handle strikes. The "marimba extra's" patch includes resonator strikes, glisses, various runs and more. The Vibraphone patches include three velocities strikes, pedal up strikes, bowed vibes, and many ethereal sounds. Three selectable motor speeds are programmed via the mod with each mallet of the vibes.

TACLMARIMBA.....**149.95**

Christian/Lane: The Ultimate Timpani Library



Using a set of Hinger Touch Tone timpani with calf heads, this six-disc library contains every performance nuance to allow the most realistic timpani imaginable. These incredible drums produce tones not possible with any standard drums using conventional heads.
Five mallets from a warm legato cartwheel to wood have been recorded at eight velocity levels for amazing flexibility and playability. Both right and left hand samples are included to make fast articulate passages flow seamlessly and realistically. Files for each mallet include: hits (right and left hands) at eight velocities, rolls at four velocities with release triggers programmed for realistic roll endings, crescendo and decrescendo rolls at three lengths, and true hand muffled strikes at eight velocities. Also includes a fingers and hand patch at four velocities (right and left hands) ranging from a light tap with one finger to the entire hand at "ff". And don't forget the standard "special effect" timpani sounds including; glissando rolls, bowl hits, booming detuned earthy tones, and eerie rolling glissando's with a cymbal on the head.

TACLTIMPANI.....**259.95**

Christian/Lane: Ultimate Percussion Library



A true pallet of orchestral percussion sounds with capabilities unmatched by anything else available. The samples were recorded with the natural acoustics of a professional hall built in. This allows sounds to instantly meld into your work with little or no additional special placement, and instruments sound like they are coming from their correct position in the ensemble, not "in your face". Each instrument is provided with a vast array of performance possibilities including: right and left hand snare hits for realistic rapid passages; consonant and dissonant triangle strikes to create the perfect tonal shading; tambourines with right and left hand samples, thumb rolls, shake rolls, crescendos; multiple pairs of crash cymbals with sustained crashes, choked crashes and scrapes suspended cymbal rolls, crescendos, mallet hits, stick hits, chokes, bells and scrapes; mallet instruments and timpani with multiple mallet choices and all velocity switched for incredibly realistic dynamic changes. Also included is a great collection of exotic percussion: wind chimes, thunder sheet, almglocken, ratchet, finger cymbals, clay chimes, bamboo chimes, porch chimes, Chinese bells, Chinese bowl, Chinese nipple gong, and anvil.

TALCLPERCUSS.....**227.95**

GIGASTUDIO SOUND LIBRARIES—*PERCUSSION/LOOPS*

Percussion Wall



This sampling CD offers Afro Techno Ritual Rhythms with drums and percussion as its foundation. Percussion Wall offers a wide range sounds, and arrangements with rhythmic pulse divisions (2, 3, 4, 6, 8 and quarter note triplets - 3 against 2). All the 559 loops on Percussion Wall are organized into 8 tempos 160, 140, 130, 120, 118, 110, 100 and 80 beats per minute. Also includes a listing of the original stereo loop points. Important data is listed for each loop which includes sound and music properties. The original loop points can be easily obtained with the information listed in this manual.

- Percussion Wall is an extremely versatile sampling CD because it contains compete larger than life rhythm loops that are production ready, also background or rhythms textures that can mix in with other existing music, as well as sonic construction material for creating entirely new loops.
- Electronic Drummer Dick Gail performed and created a series of drum loops based on African/Jazz rhythm concepts but used contemporary percussion samples for his sound source. His inspiration for these rhythms are the songs from the tribes of the Yoruba, the Bambala, the Zingili the Mahafaly, the Baya, the Watutsi, and the Badouma, and the work of the great BeBop drummers, such as Elvin Jones, Max Roach, Art Blakey and Philly Joe Jones.
- Ernest Cholakis took these loops and processed and transformed some of these loops into "Electro-Acoustic transformations" —a class of sound processes Cholakis developed to dramatically alter and transform percussion loops in unique ways that are commercially unavailable. The transformed loops on this CD were designed to expand the range of sonic and musical creative potential for the user. The results are rhythm building blocks which can be manipulated by any musician or sound designer to create entirely new and different sounding loops from those heard on this CD.
- And lastly, in the process of separating the musical elements much work was done to enhance the clarity of many of the very complex rhythms as well as having expanded and extended the timbre of the original performances.

TAPW **149.95**

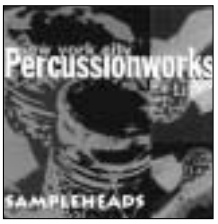
vrPercussion



A collection of hits and techniques of percussion instruments from around the world. High quality percussion samples for any kind of music.
Single hits, riffs and technique samples. Create extremely realistic grooves without being locked into ethnic patterns.

TAVRSPERCUSN **109.95**

New York City Percussionworks



Amazing percussion played by 7 of New York's top percussionists. Congas, Bongos, Clave, Shakers, Triangles, CowBells, Tamborines, GoGo Bells, Whistles, Tabla, Shakere, FrameDrum, Udo, Dumbec, Talking Drum, Pandero, Cabasa, Guiro, Flexitone, Wood Blocks, Timbale, Gongs and Bells. 1,482 events. Over a GB of hits and loops - pre-looped, ready to sample & 100% copyright free! 2 CD-ROM set!

TASHPERCUSSN **189.95**

Afro-Cuban (Latin) Percussion



This is the Latin Percussion collection you've been waiting for. Ambient stereo and close miked mono versions included to give you complete control over your mix. TONS of articulations (hits) per instrument and up to 4 multi-velocity samples from soft to loud to make you sound like a pro. Vibraslap Instruments and playing provided by Joe Galeota of JAG Drums. Recorded at Blue Jay studios in Carlisle, Mass. with engineering by Mark Tanzer (NRBQ, Carly Simon, Patty Larkin, Jerry Marotta, Livingston Taylor, Lyle Mays and others).

TASIPERCUSS **74.95**

GIGASTUDIO SOUND LIBRARIES— *DRUMS/LOOPS*

New Orleans: The Delta Groove



Inspired by the exception acoustic performances of New Orleans' musicians Robin D. Rorie and Dennis Shaikewitz, Numerical Sound has produced a sampling package that offers several exceptional features available to users for the first time. The Delta Groove contains tons of loops and samples of drums and bass to give you that distinctive New Orleans sound.
On this collection, all the kick, snare, and tom toms samples have 32 velocity levels and the hi hat and ride cymbal have 8 velocity levels. Each track has a partial set of the appropriate drum samples in the upper register of the each keyboard patch. The user can play along in real time with either the acoustic or electo-acoustic loops which are triggered from the lower register the same keyboard patch.
New Orleans; The Delta Grooves contains over 1.3 Gbs of sampling material. This package gives the user 250 acoustic loops all between 4-8 bars in length at tempo's ranging from 72-120 BPM of drums, bass & drums, bass, guitar & percussion loops. Also included are 485 E.A.M. loops, 1336 drum samples in either 8 or 32 velocity levels and 170 bass riffs, slides, harmonics as well as 210 perfectly looped individual electro-acoustic Fender bass tones organized in 6 velocity levels. Recorded in New Orleans, the overall sound is an acoustic sampling CD that is clean, warm and transparent. Into the Delta sound? Get Delta Grooves today!

TANSNEWORLNS **149.95**

GIGASTUDIO SOUND LIBRARIES — *DRUMS/LOOPS*

Heavy Hitters Library



Three stellar drummers; Tommy Lee (Motley Crue), Alan White (Yes), and Jim Keltner (studio legend). Multiple dry/ambient hard, medium and soft hits. CD-ROM versions offer modular patching allowing easy kit construction. Recorded in world-class drum rooms like A & M Studios & Ocean Way. CD-Audio & CD-ROMs.
TAQUPHITTERS **109.95**

Steve Gadd Drum Scores



Loops & multi-sampled drum hits played by Steve Gadd. Steve's credits include Paul Simon's "Rhythm of the Saints" & "Graceland", Steely Dan "Aja", and many more world class hits. DrumScores include Brush Samba, Afri-Latin & Reggae. Sticks & brush samples. Loopable sections and complete pieces of drum music.
TAQUPDRUMSCR **109.95**

Sly Dunbar - Reggae Drumsplash Remastered



Sly Dunbar, one of the world's greatest drummers, has set new standards for dance music using his unique blend of Reggae, Funk and Hip Hop with his long-time associate Robbie Shakespeare. Dubbed the Riddim Twins, Sly and Robbie pioneered the fusion of Hip Hop and Reggae that is today taking the charts by storm - Dancehall Reggae has left a trail of music rich in rhythm and bass. Sly plays the numerous styles of Reggae and Ska, then opens the vault to his Dancehall grooves, which can only be described as a sonic hurricane. This CD is a chronology of the past , present and future of Reggae music. Included are 150 otherwise impossible to get Dancehall loops by Sly for hits including Shabba Ranks, Maxi Priest, Chaka Demus & Pliers, Apache Indian, Snow, Cutty Ranks, Black Uhuru, Yellowman, Gregory Isaacs.
TANSSLYDUNBA.....**149.95**

Peter Erskine - Living Drums!



What do Steely Dan, Weather Report, Rickie Lee Jones, Chick Corea, Joni Mitchell, Joe Henderson, Jaco Pastorius, Bob James, McCoy Tyner and Michael Brecker have in common? Peter Erskine! Thousands of carefully recorded drum performances, played and personally selected by Peter from his collection of world class drum sets, are compiled on this definitive collection of drum sounds guaranteed to make your tracks come alive.
TASHLIVINGDR **114.95**

Bun E in a Box



Multiple loops from every Cheap Trick hit, plus a comprehensive collection of single hits from his extensive drum collection. Produced and recorded by legendary engineer Steve Albini (Nirvana). Available in 24-bit/44k. Drum tracks from Cheap Tricks arsenal of hits highlight this stunning collection, such as: I Want You to Want Me, California Man and many others are offered in looped sections for easy customization into your existing productions.
TAQUPBUNE **149.50**

Drum Series 1



Get ready for the drums of your dreams. From the real to the surreal, these drums sound amazing. Recorded at the famous Blue Jay Recording Studios and Wellspring Studios near Boston, all drums and all cymbals are stereo, no loops, and heavily multi-velocity (multiple samples from soft to loud hits). Even the snares are sampled at multiple places on the drum. This attention to detail is guaranteed to give you the most realistic, most playable drum sounds ever. Includes 250 Gigasampler programmed drum kits and instruments.
TASIDRUMS **149.95**

New York City Drumworks



1.2 GBs of awesome rhythm played by 6 of New York City's top session drummers! Funk, Rock, House, Hip-Hop, Latin, Cajun, R&B, Brazillian, Rock-a-billy, African, Reggae, Swing, Gospel, Country, Thrash beats and lots more! Played with sticks, blastix, brushes and mallets at multiple tempos per groove or style... The definitive collection of drum loops and hits from 6 of New York's finest. 2 CD-ROM set!
TASHDRUMWRKS **189.95**

Mark Walker - Latin Drums



Mark Walker (Paquito D'Rivera, Oregon, Caribbean Jazz Project, Michel Camilo) lays down slamming Cuban, Brazilian, Puerto Rican and other Latin American rhythms on drum set with percussion & complete assortment of drum hits and cymbals meticulously recorded at 24-bit resolution. Recorded specifically for the sampling musician with an extremely wide variation of tempo and feel. The drum hits are superb - some say the best ever recorded.
TASHLATINDRM **114.95**

TOPAZ STUDIO KITS: Volume One



The ultimate in live acoustic drum sampling. Features 8 complete virtual drum kits created from Yamaha and Gretsch premium acoustic drum sets! From large rock kits, to studio and pop kits, to acoustic jazz and jungle, you are sure to find the ideal drumming solution here. All kits were played by New York session drummer Zach Danziger, who is well known in musicians' circles, having recorded with such artists as David Holmes, Manic Street Preachers, U2, Jim Beard, Bill Evans, and Randy Brecker. The library fully utilizes the expressive power of GigaSampler and GigaStudio. Lightning fast performance, even on modest systems.
TABGTOPAZ.....**74.95**

GIGASTUDIO SOUND LIBRARIES– *DANCE/ELECTRONIC LOOPS*

Dance Partition



1500 fresh, modern synthetic drum and percussion single shot samples. A huge variety of high quality sounds for many different styles; classic dance-floor, techno/hardocore, hip hop/rap, pop, trance & jungle. Produced by Andreas Luhymeyer and Christian Hoppner/TIB Studios/Germany
TAQUPDANCE.....**112.50**

HipHop



This is the phattest multi drum & loop cd rom period. It contains a 1000+ Kicks, Snares, Hats, 3D Live Loops and Kits, virtual speaker Guitar licks, chords and funky wahwah stuff, real bass, bass noises, and distorted bass.
TAVRSHIPHOP.....**109.95**

Streetbeats by Poogie Bell



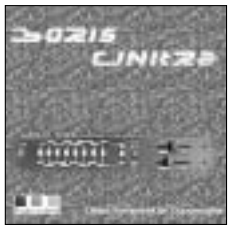
Drummer/producer Poogie Bell (Chaka Khan, Teddy Riley, Marcus Miller, Hugh Masakela) offers you sounds from his recording and performing travels. Hip Hop, Rap, R&B, soul, jungle, acid jazz, reggae dance hall are genres represented on this disc. Includes loops, plus hits of drums, doors and much more.
TAQUPSTREET.....**109.95**

David Torn Pandora's Tool Box



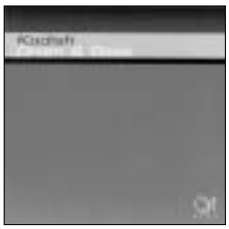
Features 70 tracks (73:01 mins.) of Textural Mayhem. Created for the producer/musician who requires highly unusual sounds. Organized into 3 categories: Textural, Riffic, and Rhythmic, these loops are suitable for many applications including sound for picture, music production, multimedia, or adding a desired edge and degree of unusualness to existing music.
TAQUPPANDORA.....**109.95**

Boris' Unitra Eltra Rytm16



Collectors' item analog rhythm box circa 1989 from Poland. 32 rhythm patterns sampled at multiple tempo ranges and perfectly looped. Mapped to Gigasampler files clearly marked by tempo for ease of use. Check out the mp3 demo to see if this funky beat box appeals to you!
TABGBORIS.....**23.50**

Kodish Drum 'n' Bass



Pure adrenaline by Kodish, who offers the machine-like energy of Drum 'n' Bass with the emotional impact of real drums. Subsonic synth grooves and hits played by Cliff Hewitt (Apollo 440) round out this disc. You've heard Kodish live and on recordings by Apollo 440, Chemical Brothers, Maximum Roach, Jean Michel Jarre remix concert. Sampler layouts offer complete control over each instrument/loop in the set. This enables access to only the hat loop or kick loop. Overhead and room mics were recorded on discreet tracks thus enabling user control over tight and dry or big and wet. Mixes of the entire groove are also included. Some loops are heavily processed and trashed out. A complete set of multi-sample single hits are also included.
TAQUPKODISH.....**149.95**

Dream Experience



Dream house, the new rage in hi-tech dance music sweeping Europe. Innovative bass, drums, FX, choir, textures, ambient film tools and complex sound scapes. Perfect for electronica, score to picture and remixes. Wake up and smell the dream. Produced in Germany by TIB/Andreas Luhmeyer and Chris Hoppner.
TAQUPDREAM.....**112.50**

Increased Velocity— Surround Tools for Post Production



Surround and astound your listeners with radically futuristic loops, beats, ambiences, drones, hits, stinger FX & atmospheres. All sounds are processed in Dolby Stereo Surround (4-2-4). Fully endorsed by Dolby Laboratories, this library is the Swiss army knife for use in film, music, re-mixing, multimedia development, or any project requiring cutting edge audio production. The sounds were created Canadian-based sound designers Sean Charles and Gordon Durity, who have created signature compositions & sound design work for the film, music, multimedia and video game industries.
TAQUPVELOCIT.....**152.95**

Drone Archaeology



Constantly evolving drones are based on original sounds and acoustic palettes and have completely unique characteristics. These tones are the result of computer resynthesis with a series of proprietary computer programs. Each drone is rich yet is completely absent of noise. Contemporary and original in nature and scope, Drone Archaeology is a powerful tool in the hands of the contemporary and avant garde composer/musician, as well as being an invaluable resource for film composers creating soundtracks, soundbeds and sound design.

- Used as foundation material, they can be processed and transformed in any number of ways to assist you in developing your own particular sound.
- Every drone is between 32 and 64 seconds of original material with no built in loops.
- Film Composers: In order to aid multimedia and film production, the drones have been classified into seven categories:Nature, Science, Drama, Industry, Voice, Bell and Drones. They were designed to have "focused spectrums". In other words: to occupy a measured amount of the sonic space in the stereo field. Even though they are extremely rich and evocative tones, they do not dominate or overpower the overall sound.

TANSDRONE.....**149.95**

TC POWERCORE

Effects Processor for VST and MAS



TC PowerCore is an open DSP hardware platform for your favorite VST/MAS recording application—Mac or Windows. A PCI card that acts as a DSP Turbo, TC PowerCore enables native users to run extremely DSP-intensive high-end effects that previously taxed the host computer to the utmost. Using state-of-the-art Motorola 56K DSPs designed for audio, TC PowerCore provides significant performance over sole native processing, providing the equivalent power of four Mac G4 processors and a PowerPC chip— on a single card. With TC PowerCore you can run 8 studio-quality TC MegaReverbs, TC Master X or up to 24 other high-end PowerCore plug-ins simultaneously without reducing your track count or using up all available CPU horsepower.

TC PowerCore adds incredible power to your existing work environment. It seamlessly integrates into any VST-compatible sequencer or audio application such as Cubase, Nuendo, Digital Performer, Spark, Peak or Logic—no proprietary add-on software is required to access the board's audio processing power. PowerCore works in conjunction with any audio hardware supported by the hosting audio application, from ASIO to Direct I/O and even SoundManager. 24/96 is supported, of course requiring twice the amount of processing resources per plug-in. Multiple cards are also supported by the design, allowing for expansion as the need grows. Hardware I/O is supplied by the user's existing I/O solution.
Finally, PowerCore includes a stack of extremely powerful high-end DSP tools, right out of the box. All in all, 22 different tools from Reverb to Synthesis and Metering to Dynamics processing are included—all in TC-quality.

FEATURES

Flexibility

Each TC PowerCore DSP can run any type of effect thanks to plenty of power and the dedicated external S-RAM. Depending on processing requirements, a single DSP can normally run multiple effects. Super-algorithms may even take up to 2 DSPs, as required. Open 3rd party policy ensures that there will always be plenty of options available to expand the scope of the system!

PowerPC

The unique design includes a Motorola PowerPC with 8MB of D-RAM for perfect integration with floating point processing. This chip is TC POWERCORE's brain and bridge to the host-based native world - so the DSPs can do what they are best at: professional audio processing!

Low-Host Load

TC PowerCore doesn't drain your computer's performance with its sheer existence in the system. Even the PowerPC chip has its own RAM. So no effect has to access the host computer's RAM, saving precious bandwidth on the PCI bus.

Quality

Four Motorola 56362 DSPs ensure professional signal processing quality. Each chip allows up to 24-bit/96kHz audio and 48-bit double-precision processing, which many consider superior to 32-bit floating point processing. The 56362 is the most widely used DSP chip around, found in many high-end products from digital audio workstations to effect processors and synthesizers!

TC POWERCORE

Multiple PowerCore Cards

Up to 4 cards may be used simultaneously in one system, though each card provides you with so much power that you probably won't need that. In fact, with 4 cards you can go as far has having a MASTER X3 virtual Finalizer on every track in a 32-track production.

The VST PowerCore Plug-in Format

With TC PowerCore you don't give up any of your VST features – instead, PowerCore adds the reliability and power of a DSP-based system to your native environment. The VST PowerCore plug-ins run as an extension to VST-plug-ins. The user interface code runs on the host and acts like a VST Plug-In, while all the CPU intensive DSP code runs on the card. This frees up precious performance resources on the host computer for other native processes like VST instruments. Of course, you can use VST and TC PowerCore plug-ins at the same time – even in the same signal path. PowerCore plug-ins are even stored in the same folder as your VST plug-ins.

I/O

TC PowerCore functions with all I/O systems supported by your application, making it the perfect addition to almost any existing system.

Compatibility

TC PowerCore supports both, MAC and PC computer platforms. The card is compatible with MacOS 9.x and Windows 98, 2000, ME, XP. It also compatible with any host application that supports VST or MAS-plug-ins, as well as any sound system supported by the application software (ASIO, Digidesign Direct I/O, SoundManager etc).

Options

An integral part of the TC PowerCore is it's open processing platform, allowing many more plug-ins by well-known 3rd party developers. Software like SONY's Oxford EQs, D-sounds Simulator and Waldorf's D-Coder take advantage of PowerCores's processing power to provide you with seamlessly integrated FX in your VST/MAS sequencer.

Bundled Plug-ins



TC Megareverb

Megareverb's unique algorithms are based on years of research and the core technology of the flagship TC Electronic M5000. As performance requirements aren't issue with TC PowerCore, the software provides studio quality reverb inside the native production environment. Designed for professional use, you will find all the parameters required for proper reverb design— from separate initial reflections and tail controls to a reverb tail with 3 independent bands. Run up to 8 Megareverbs simultaneously on a single TC PowerCore card .

- With better diffusion, denser tails and more realistic room impressions than ever before, TC Megareverb is the ultimate reverb solution. Adds Initial Reflection and Tail Balance Faders for even more creative control!
- Intuitive user interface with an overview of all relevant parameters, combined with easy control of the parameters takes the guesswork out of the reverb creation process. You get the sound you want - fast.



TC VoiceStrip

This highly integrated plug-in is a complete channel strip for voice sound processing, with Compression, DeEssing, a dedicated Voice EQ, Lo-cut Filter and Gate! The acclaimed TC-saturation technology and the vintage compressor model ensure that VoiceStrip can add the highly coveted TC emulation of tube gear warmth and presence to your recordings whenever required.

TC Vintage CL (Compressor/Limiter)



The Vintage CL employs a different compression algorithm from VoiceStrip - emulating the behavior of industry-standard analog compression, aiming at general purpose rather than specific applications. Run up to 6 of these versatile Compressor/Limiter plug-ins on only one of the four available DSPs.

TC POWERCORE

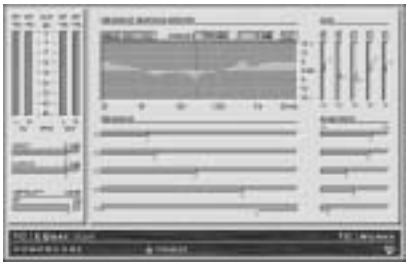
MASTER X3

Based on the technology of TC's industry-standard Finalizer, the MASTER X3 provides knock-out punch for mixes and masters. MASTER X3 expedites the mastering process by integrating several phases of mastering into a single, easy to use interface. It features multiband expansion, compression and limiting, all optimized for mastering applications. High quality uncorrelated dithering is included as well.



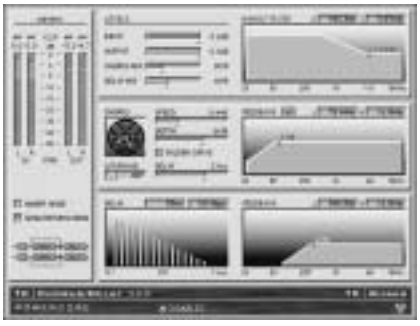
- Intuitive user interface makes this critical process faster and easier. Overall processing characteristics are controlled via "Target Curves" that simplify handling by providing a global style for the bands in all modules.
- The interaction between the bands can be fine tuned by applying "Target Factors" which determine the frequency focus of each processing module, significantly reducing the number of required user parameters.
- During mastering, you need to know exactly what's going on, so MASTER X3 includes accurate level metering with clipping counters and peak hold indicator.
- Digital ceiling parameter lets you precisely limit maximum volume whenever brick-wall limiting is required. "Soft Clipping" even provides "analog" sound when desired.
- Up to 8 instances of MASTER X3 can run simultaneously on one PowerCore board.

TC EQsat



TC EQsat provides the same high-quality EQ that you find in the TC Finalizer hardware. This 5 band EQ has 3 parametric bands plus one Low shelf and one High shelf band (±18 dB boost/cut per band). Naturally, bandwidth, frequency and gain are selectable per band. Plus, the unique SoftSat technology creates that warm, analog sound whenever desired. Clean-sounding and very easy to-use, it is ideal for any situation from recording to mastering. Up to 12 instances of EQSat can run simultaneously on one PowerCord card. A combination of TC Megareverb, EQSat and Chorus/Delay will only require one DSP to run (at 48kHz).

TC Chorus/Delay



Chorus/Delay was modeled after the TC Electronic 1210 Spatial Expander, the legendary classic device for modulation and delay FX with very flexible routing and filtering facilities. This plug-in provides Flanging, Chorus, Delay, Slap Delay and combined modulation and delay multi FX, complete with BPM-based delay time input! Chorus/Delay creates unique images which are rich, yet never sound "overdone". Hear it to believe it!

Spark LE

Spark LE is the bundle version of Spark, TC's effects processing and mastering application for Macintosh. However, if you only expected a "Lite" version, think again: Spark LE supports 24/96, VST plug-ins, includes a native mastering limiter, delay line and EQ, realtime crossfades, etc., etc.! And if you want even more power – an upgrade path to Spark is included as well. Compatible with Mac OS X and Mac OS Classic, Spark LE comes with a mastering quality Audio Maximizer, a flexible Tempo Delay and a One-Band EQ with selectable Notch, Low Pass, High Pass, Parametric types of equalization and a Master window with high resolution metering and 4 slots for real-time plug-in processing.



B&H PHOTO - VIDEO - PRO AUDIO

TC POWERCORE

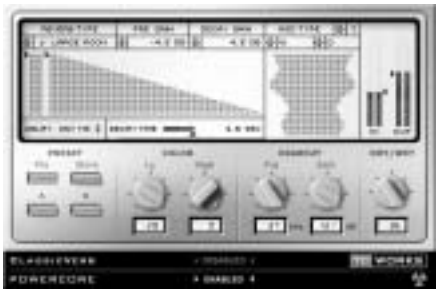
TC PowerCore 01 Synthesizer



PowerCore 01 is a synthesizer plug-in for PowerCore. Modeled after a classic monophonic analog synth, it sounds fantastic.

The TC PowerCore instrument format allows virtual synthesizers to use the available DSP-power for signal processing while running seamlessly integrated within any VST-instrument compatible sequencer, such as Logic, Nuendo or Cubase. As PowerCore is an open platform, other manufacturers such as Waldorf and Access are building synths based on PowerCore's processing power.

Classic Verb



Modeled on a world renowned studio reverb to achieve its distinctive rich and textured classic reverb sound, Classic Verb's algorithm follows a completely different design philosophy, offering a completely different sound from Megareverb. Apart from the elegant user interface, Classic Verb features a choice of modulation types for sonically rich timbres. Fifteen room types provide a wealth of different unique spaces, featuring Halls, Plates, Cathedral, Spring as well as three rooms tuned especially for vocal processing. 2 instances of the plug-in require only one single DSP to run, allowing a total of up to 8 Classic Verbs on a single PowerCore– leaving room for more plug-ins.

24/7-C Limiting Amplifier



24/7-C is meticulously modeled after a legendary dynamics processor, allowing it to match its distinctively warm tone, while adding sophisticated extras only possible with today's flexible software technology. 24/7-C extends PowerCore's dynamics processing to include a total of 4 plug-ins dedicated to a variety of compression needs. 7 mono instances of the plug-in require only one single DSP to run, allowing a total of up to 28 mono 24/7-C's on a single PowerCore. Supports 96 kHz.

PowerCore FireWire



TC Electronic introduced PowerCore Firewire, providing audio professionals on the move with audio processing on laptop or desktop systems. This addition to the PowerCore platform offers almost twice the power of the PCI-card version. Now even complex mixes become a reality using just a laptop computer and one PowerCore Firewire. Of course the PCI-card and the Firewire version of PowerCore can be used at the same time.

PowerCore Firewire offers the same unique architecture as PowerCore PCI, featuring a Motorola PowerPC and 4 Motorola DSPs. All signal processing is performed on the PowerCore unit, freeing up precious host performance resources for virtual instruments or native Plug-Ins. The hardware specifications go beyond those of the original card, featuring faster DSPs, a faster PowerPC and 3-times the amount of RAM per DSP.

OPTIONAL PLUG-INS

Assimilator

EQ-Curve Assimilation and Morphing



Learn the sound of your favorite mix and apply it to your own. This in short is what Assimilator does. Equalization is one of the more difficult tasks in recording, mixing and mastering – especially when you are trying to match a certain sound. Assimilator makes the process fast and instantaneous—3 steps and you're done.

The user interface provides two complete sets of curves and a morph page. On the morph page, you can morph between two complete curve sets by simply moving one fader. The applications of this processor range from film sound to mastering and of course creative use.

- FFT-resolutions of 2048, 4096, 8192
- Freely adjustable Apply strength
- Draw tools for EQ-curve editing
- Curve presets (Loudness etc.)
- Limiter
- 1 per DSP in high resolution

TC WORKS

Optional PowerCore Plug-Ins

Waldorf D-Coder
Synthesizer Vocoder



Waldorf's D-Coder for PowerCore goes beyond classic Vocoder concepts by providing an integrated synthesizer as tone generator. This way no external carrier signal is required to make the vocoding effect audible. D-Coder is based on DSP technology used in the popular Waldorf synthesizers, made accessible to PowerCore users from inside any VST or MAS compatible sequencer.

D-coder has everything you need to create sophisticated vocoding effects, in one easy to use interface! The included polyphonic synthesizer, featuring that classic Waldorf sound, eliminates the need for external carriers. Integrated effects include 3-band EQ, Chorus and Drive

D-coder can be played from a MIDI track or in real-time from a keyboard. D-coder is so flexible, the synthesizer section can be played independently of the vocoder section! If you are using a VST 2.0 compatible sequencer, all LFOs can follow MIDI sync, allowing you to independently modulate the synthesizer and the vocoder in musically relevant values such as "8 bars".

DSound VL2
Multichannel Valve Interface



The VL2 is an 8-channel virtual valve interface for mixing and mastering, adding the warmth and life of tube processing to any digital recordings. This Plug-In was developed by DSound as a direct result of the phenomenal growth in popularity of affordable hard disk recording systems, to fix the sometimes cold and clinical sound of digital recordings with its unobtrusive processing. The VL2 is an elegant solution with a simple interface and a superior audio quality. The VL2 goes beyond the functionality of common valve interfaces by allowing you to control the amount of saturation with a single, finely tuned control. Each channel of the VL2 has unity gain – so levels are unaffected – but as each channel incorporates a valve stage, the audio signal benefits from the warmth of the finely modeled valve characteristics.

Steinberg Surround Verb (For Nuendo)



TC-quality 5.1 reverb has been available to users of Steinberg's Nuendo for quite a while thanks to this plug-in's native version. Now this version adds Powercore support. Surround Verb features a graphic display for visualizing the reverb expansion in the virtual space, and a powerful algorithm core delivering dense and convincing state-of-the-art reverbs. The tail parameter offers the same flexibility as MEGA REVERB with 3-band control and adjustable crossovers. The additional EQ stage provides additional sound flexibility with its low and high shelving features.

- 4 Room Types: Hall, Horseshoe, Club, Room
- Power-panning for position in a room
- Discrete tail for each channel in a 5.1 setup

Optional PowerCore Plug-Ins

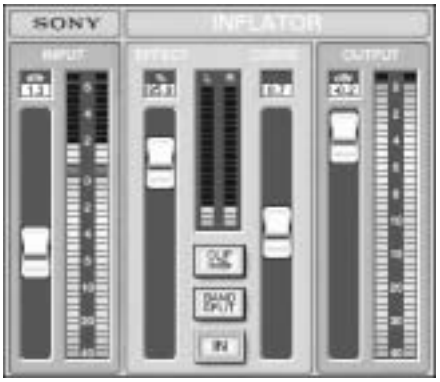
Sony Oxford EQs *EQ Suite*



Sony Oxford EQs deliver an unparalleled quality and flexibility with very musical results and extremely low noise and distortion, below -130 dB. The EQ and Filters plug-in comprises out of 5 different sections including HF and LF selectable to shelf. Two additional HF and LF filters add to the power with variable slopes from 6 dB to 36 dB per octave. The models and technology of these plug-ins are based on the same as in the popular Sony Oxford consoles and Pro Tools plug-ins. The package includes different plug-in versions for greater DSP-efficiency. If you are looking for the ultimate in EQ, this is your solution!

- Very musical results
- 24-bit/96kHz processing
- 4 EQ-style models for a maximum of artistic flexibility – from British to transparent
- Low noise and distortion (below -130 dB)
- HF sections are fully decramped
- A/B and selection between two complete sets of settings

Sony Oxford Inflator



Tube Emulated Dynamics Processor

The Inflator is a unique process that can provide an increase in the apparent loudness of almost any program, without obvious loss of quality or audible reduction of dynamic range, yet avoiding damaging increases in the peak level of the signal. The Inflator process can also bring power, presence and warmth to program material and even provide headroom overload margin above digital maximum with a subtlety and musical character reminiscent of tube systems.

- Increases the loudness of almost any program material
- Creates warmth, character and dynamic excitement, similar to that of analog systems
- Provides virtual headroom above digital maximum
- Creates artistic effects ranging from subtle tube-like harmonic characteristics for warmth, presence and 'in your face' fatness, to outright saturation distortion modeling
- 2 modes of operation for maximum flexibility, artistic creativity and optimal DSP usage

Sony Oxford Dynamics

Full Dynamics and Compression from the OXF-R3



Resulting from many years of research into professional dynamics applications, the channel dynamics offers separate Compress, Limit, Expand, Gate and side chain EQ functions, with full independent control of all parameters. Also separately included is a bus Compressor/Limiter which is fully surround multi-format compatible.

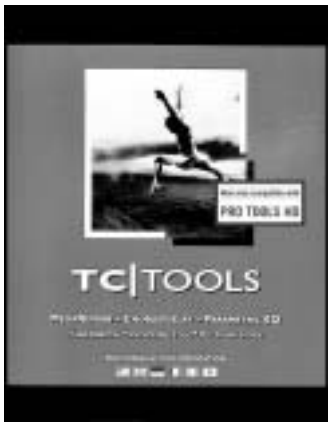
Features such as selectable time constant curves and variable soft compress functions allow you to confidently tackle all common uses of compression, from subtle level control and mastering functions to the production of great artistic effects. The use of a feed-forward architecture with logarithmic side chain processing ensures exemplary sonic characteristics and dynamic accuracy. This highly sophisticated and professional product has the power and flexibility to provide an artistic capability unavailable from other units, analogue or digital.

- Channel Dynamics with separately controlled sections for Compressor, Limiter, Gate and Expander.
- Channel Dynamics includes fully featured 2 Band side chain EQ which can also be used in the signal path.
- All functions are fully automated.
- Separate Bus Compressor/Limiter with surround support and selectable Sub channel filtering and gain contribution control.
- Variable harmonic enhancement for extra loudness, presence and 'punch''.
- Fully variable soft ratio function for extreme program tolerance and highly musical compression.
- Selectable re-dithering function for word length reduction in mastering situations.
- Low noise and distortion (below -130 dB)

TC WORKS

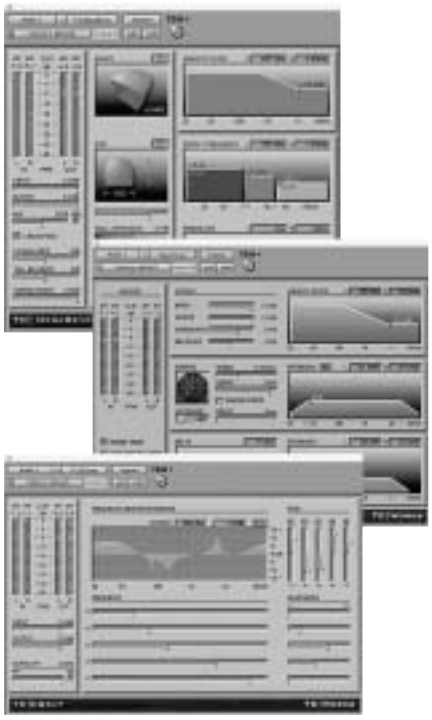
TC|TOOLS 3.6

TDM Plug-ins



TC/TOOLS 3.6 includes the essential processing tools for your Pro Tools 24 Mix and HD systems including systems running on OS X and Windows XP. These include the powerful TC MegaReverb, TC EQSat and TC Chorus/Delay. Support is provided for sampling rates up to 96 kHz using HD systems.

TC MEGA REVERB's unique algorithms are based on years of ongoing research and the core flagship technology of the M5000. With even better diffusion, denser tails and more realistic room impressions than ever before, TC MEGA REVERB provides the ultimate reverb solution.



TC EQSat is a 5 band EQ with 3 parametric bands plus one Low shelve and one High shelve band. It is capable of boosting or cutting up to 18 dB. Naturally, bandwidth, frequency and gain are selectable per band. And what's more, the unique SoftSat circuit creates that warm, analogue sound whenever desired.

TC CHORUS/DELAY recreates the sound known from the classic TC 1210 Spatial Expander, creating unique images which are effective without sounding "overdone". This Plug-In allows you to create all sorts of different modulation effects like Chorus, Flanging, and Slap Delay with very flexible routing and filtering facilities.

TC VOICE TOOLS *Plug-in Bundle for Pro Tools 24 MIX Systems*

TC VoiceTools is a dedicated all-in-one voice processing solution plug-in bundle featuring Intonator for pitch correction and Voicestrip for sound and dynamics adjustment.

TC INTONATOR TDM

TC INTONATOR TDM fixes out of tune notes on the fly - by adjusting a few intuitive parameters, an otherwise great vocal performance can be brought into shape in realtime. A definable pitch "window" allows you to narrow or widen the re-intonation range, set scale and amount - done!

- Realtime pitch correction (maximum range ±400 cent)
- Choose from major, minor, chromatic and manual and user definable custom scales
- Adaptive Lo Cut Filter: Automatically adjusts its frequency according to the material's shape
- Tune Parameter for adjusting the tuning reference
- Tune Reference Mode (keyboard acts as synthesizer)
- Extremely intuitive interface: very few parameters required to set up the process
- Up to instances per Mix DSP



TC VOICE STRIP

TC VOICE STRIP includes all you need for voice sound processing: Compression, De-essing, a dedicated Voice EQ, Lo Cut Filter and Gate - that's total recall of your vocal processing sound! Thanks to our acclaimed SoftSat technology TC VOICE STRIP can even emulate the sound of analog gear, adding the highly coveted TC emulation of tube gear warmth and presence to your audio whenever required.

- All-in-one voice sound processing
- Single Band Vintage Compressor
- Level independent De-esser
- Dedicated 3-Band Voice EQ
- Switchable SoftSat saturation for analog emulation
- Gate and Lo Cut EQ
- All processes can be used simultaneously

TC NATIVE BUNDLE 3.0

The TC Native Bundle 3.0 includes seven high-end processing tools for VST and MAS users that cover all needs for equalization, dynamic processing (including de-essing), reverberation, limiting and maximizing. Tools are heavily performance optimized, so they really work in your multi-track environment - without bringing the computer to its knees. Their intuitive user interfaces allow you to use them instantly. All of Native Bundle 3.0's plug-ins have been completely rewritten in order to achieve even better sound and system performance. They process you audio with 32-bit floating point accuracy, and support 24-bit/96kHz files. Compatible with Mac OS X, OS Classic and Windows.



Filtrator

Synth style filtering-LFO Modulation, Envelope follower, SoftSat for vintage sound, and more...Filtrator underlines the creative possibilities with Native Bundle as its innovative controls support intuitive and fast effects design. Its Envelope Follower allows for classic Filter effects with the filter being modulated by the amplitude of the incoming audio signal. The LFO modulation is syncable to MIDI clock, providing easy synchronization for modulated filter sweeping. Furthermore, Filtrator includes 12, 18 and 24dB Slope, Lo and Hi Pass Types, a Separate Drive Section for Filter Distortion and a Saturation algorithm for a warm, analog sound.

Limiter

A TC-quality single-band limiter, the Limiter is the power tool for limiting and maximizing your audio recordings. Adjustable Hold Time, Auto Make-Up Gain, a huge reduction meter and a level histogram, more punch and perfect levels turn this plug-in into the perfect tool for mastering. The histogram is a true eye opener: it displays the level distribution in your file before and after processing. You can even monitor the difference between input and output, so you get the feedback you need to ensure you don't apply too much processing! The histogram's scale is switchable between full dynamic range (-96 dB to 0 dB) and hi resolution for the more significant range (-24 dB to 0 dB).

Compressor

In addition to standard controls like Attack, Release, Threshold and Ratio, the Compressor section features a Hold Time, Automatic Make-Up Gain and Soft Knee controls. These parameters allow you to "fine-tune" the sound characteristics of the compressor from "analog" to digital - and to adjust the overall behavior for processing a single track or mastering a complete playback. Make-Up Gain automatically brings peak levels up to 0dB.

De-Esser

Most de-esser's operate with an "absolute" threshold, requiring you to send the signal in at an optimal level close to 0 dB. This makes working with heavily dynamic material and fades a nightmare, because all the sibilant, unwanted frequencies would suddenly reappear. With the TC plug-in, de-essing is level independent so you get optimal results — even at very low input levels, with highly dynamic material or in fade outs.

Reverb PLUS

TC reverbs are the cornerstone of many professional recording studios. And the Native Reverb PLUS is no different. A successor to the original Native Reverb, the easy-to-use Reverb PLUS features an enhanced algorithm, new user interface with additional parameters to significantly extend the creative possibilities for creating convincing spaces, and over 100 new presets for a wealth of added flexibility.

Graphic EQ & Parametric EQ

All the equalization tools necessary for recording, editing or mastering - and more. Innovative controls make using these two plug-ins an intuitive and creative experience. Both EQs also have the SoftSat advantage: a proprietary algorithm which generates that warm sound often associated with analog equipment. The key to a better sounding EQ.

Parametric EQ—

The Parametric EQ is a powerful 10-band EQ (7 fully configurable bands and 3 additional bands) controlled by a unique virtual joystick for simultaneous and fast treble and loudness control. The 7 bands can be assigned to work as Parametric, Hi/Lo Shelf or even Notch filters – and any combination of these. Split gain controls for left and right channels give you instant access for working on channels discreetly, or moving the frequency spectrum around in the stereo image.

Graphic EQ—

The Graphic EQ is a 28 band graphic equalizer with a unique graphic interface (touch screen) for intuitive access to the band gain controls – simply draw the desired frequency response onto the display with the mouse, and the EQ is set. Operate with full 28 or alternatively 14 or 7 bands. The bands can be grouped for even easier handling. The Master Fader is an additional tool for quickly adjusting the overall gain relations of complex EQ settings.

TC WORKS

SPARK XL



Realtime Digital Audio Editor for Mac

More than a classic two-track editor, Spark XL is a complete mastering and editing environment with a stack of powerful tools to get the job done. Spark represents a new approach to stereo-based audio editing, realtime processing, and mastering, featuring a unique Browser window, integrating Wave editing, project file management, and play list, all at once. SPARK XL also includes the FX MACHINE and all its plug-ins. The Browser View is used to edit a file, or build a play list, the Master View gives you Master Faders with highly accurate PPM level meters, phase correlation meter, built-in Dithering, and a realtime "FXmachine" with extensive routing capabilities. Up to 4 stereo streams with 5 plug-ins each can be routed and processed simultaneously. A histogram display visualizes the "level distribution" in your recording, for valuable feedback about the signal processing being applied, thus simplifying finding the right settings.

FEATURES

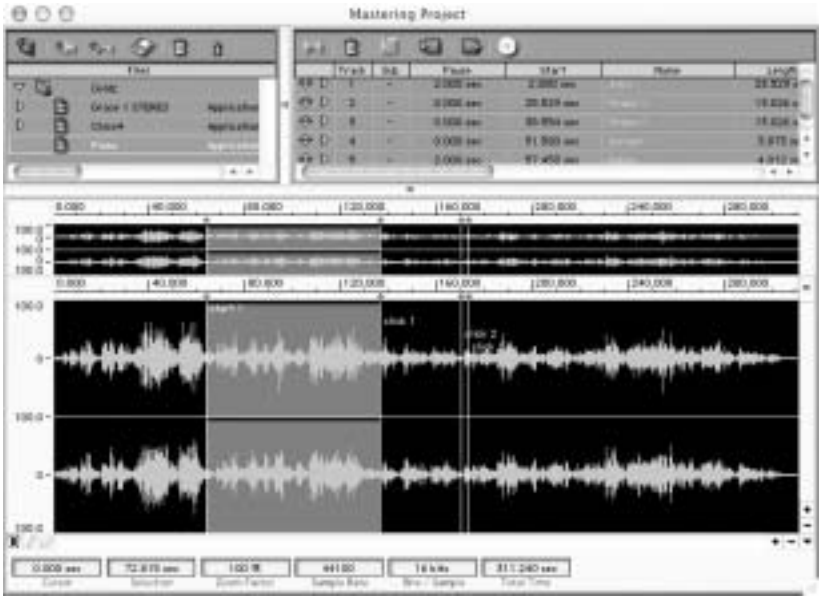
BROWSE

Thanks to SPARK XL's unique one window approach, you always keep the overview: The BROWSER VIEW integrates all aspects of editing into one window: Audio Database, Wave Editor and Play List.

The Database allows you to create a virtual view of all used audio material, and to store all project relevant information in one global Project file. Accessing files is instantaneous; the waveform overview is always displayed immediately in the editor. Import and export are a breeze – full drag and drop support allows importing files directly from the desktop and vice versa.

The Edit View features the waveform display with two views for details and overview. Separate Play and Edit cursors simplify the location in the file and the generation of Segments and Regions. Markers can be created on-the-fly, even during record, and a padlock function ensures they can not be moved by accident.

The Play List View provides a numerical play list for CD-preparation, complete with realtime Cut and Crossfade Editor. Multiple entries can be assigned to one index and the CD-Export saves the master into Roxio Toast and Jam compatible formats. The FX Machine's processing can even be applied to the Play List on export.



The realtime CUT & CROSSFADE EDITOR is accessible directly from the PLAY LIST and displays two transitions at the same time, with crossfades of unlimited lengths, different curves and even user presets. A temporary A/B memory allows to compare different settings for the same transition. Extensive preview and nudging features complete the picture.

Time Stretch processing can be performed in realtime, without affecting the pitch, and even accessible from the Play List, so the same take can be played at different speeds easily.

SPARK XL, our award-winning audio processing and mastering application, now features direct CD-Burning. Even complex Play Lists can be turned into a CD instantly, as crossfades and pauses are taken straight from Spark's Play List. Track indexes may be activated or deactivated per region - so several regions can be used to create one track without the need for any additional work or rendering.

SPARK XL

General

- Record/playback up to 32-bit/192 kHz files
- AIFF, SoundDesigner (SD2), WAV Files and all file formats supported by QuickTime
- MP3 encoding
- VST-plug-in support
- Supports ASIO and DigiDesign Direct I/O
- Keyboard commands for all important functions like transport controls etc.

Wave Editor

- Virtual File Engine with instant cut, copy and paste
- Single channel editing
- Multiple undo/redo
- Direct editing of split stereo files
- Markers (with "on-the-fly" creation, even during record)
- Off-line Processing: Normalize, change gain, DC Removal, reverse, sample-rate and bit-depth conversion and fades
- Off-line mode for VST plug-ins allowing for easy selection-based processing
- Info line with all important wave parameters at one glance
- Up to 40 realtime plug-ins during recording of files
- Drawing tool and eraser

Play List

- Linked to file database
- Plays up to 2 stereo tracks simultaneously
- Realtime crossfades
- Cut/Crossfade Edit Dialog: Extensive editing and nudging controls

Batch Converter

- Process loads of files in one go
- Off-line Processes: Format, bit-depth, sample rate, channels, DC removal, normalize and dither

Sampler Support

- AKAI S1000/1100 and 3000 series, EMU ESI and E4 series, Kurzweil K2000 and 2500 series, Roland S760, Yamaha A3000, EX5, EX7, EX5R, SY85, SY99, and any samplers supporting the SMDI protocol
- Reads AKAI S1000 format CD-ROMs

Master View & FX Machine

The FX Machine is part of the Master View, Spark XL's second main window with all the master controls and FX-routing matrix. Everything you need for processing at a single glance! This extremely flexible FX-routing matrix is a creative power tool with endless possibilities for creative realtime sound manipulation, sound design and of course mastering processing. The Matrix size is freely adjustable, so you can create whatever size you need... your CPU power is the only limitation.

Of course, SPARK XL utilizes the FX Machine's realtime effects. But for detailed section edits, the offline processes come in handy. They cover all the basics like Normalize and Remove DC but there's more: You can apply any VST plug-in offline as well. Additionally, Pitch Shifting and Time Stretching allow for speed and pitch adjustments. Time Stretch is also available in realtime.

Bundled with over 20 Plug-in (VST Compatible) Including:

- Metergraph and Sonograph Analyzer plug-ins let you easily check your material. The Sonograph provides detailed information on the material's frequency distribution. Issues like 15 kHz monitor hum become visible instantly.
- DeClick removes vinyl/digital clicks and crackle – the process is fully automatic and requires only 2 parameters for fine-tuning. An Audition function provides optimal monitoring possibilities. The Click/Crackle parameter can be automated so the algorithm checks the material for the characteristics of the artifact and adjusts the setting without user interaction.
- Denoiser is optimized for broadband noise and utilizes a fingerprint method, so it is easy to control the behavior. Three parameters are required to fine tune the denoising process. Audition function ensures that only noise is removed, not your material.



- TC Native Bundle cover all your needs for pro-equalization, dynamics processing and reverberation:
  - DeX compressor and de-esser
  - EQ-P 10-band parametric EQ
  - EQ-G 28-band stereo graphic EQ
  - Native L and Reverb: mastering limiter and award-winning TC-reverb

QuickTime Movies

Spark XL loads QuickTime movies and splits the audio off for editing. This way, mastering jobs for QuickTime movies become a breeze. Spark XL supports all file formats compatible with QuickTime including MPEG-4.

TDM Master

Spark XL comes with an additional MasterView for TDM plug-ins. You can use native and TDM plug-ins in the same audio stream! And with the Direct Connect, you can run Virtual Instruments in Spark XL, then stream the output directly into your simultaneously running Pro Tools.

MegabitMAX

MegabitMAX is a novel Word Length Reduction technology developed for high-end audio mastering. MegabitMAX is created for applications that demand the highest level of detail resolution from audio master. MegabitMAX is an excellent solution for recordings with high dynamic range, such as classical, jazz, film music, etc.

Unlike the noise of other word length reduction systems, the dither noise of MegabitMAX is not disturbing to the human ear. It has natural spectrum across the audible frequency range, so users can describe the sound as "vinyl-like" and "analog". While preserving the finest detail of your high-resolution master, the dither noise of MegabitMAX has very small amplitude, so it won't present any problems for post- processing/duplication equipment. MegabitMAX is completely optimized for 44.1 kHz and higher (96 kHz, 192 kHz...) sampling rates.

PLUGSOUND

Virtual Sample Library and Instrument Plug-ins

Plugsound unites two previously unrelated tools, and simplifies your music day after day. It is the result of two worlds merging into one: a professional sound library meets a virtual instrument. As a result each Plugsound consists of two elements on your hard drive: a plug-in and a sound library data file.United by the revolutionary UVI-Engine, Plugsound is a first-of-its-kind sample based series of virtual instruments designed to provide easy, flexible access to massive sampled sound banks.



The Plug-in:

Identical in every volume of the series, the Plug-in is a sample-playback virtual instrument powered by the enormously powerful UVI-Engine. The UVI-Engine is used in many great products like Spectrasonics Stylus, Atmosphere and Trilogy. Plugsound provides you with essential and useful synthesis tools:

- MIDI control of every knob and slider on the interface via standard MIDI continuous controllers.
- Two resonant filters, one Multimode, and one switchable between HPF and LPF called the UVI filter.
- ADSR-type Amplitude and Filter envelopes. The Multimode Filter has an envelope amount knob.
- Reverb' module with 3 knobs to set the time, the high damping and the reverb' mix
- Mono Mode with a glide time knob. This provides the most realistic mono-legato you've heard on a virtual synth.
- One LFO that can be used for tremolo, auto pan etc. This is in addition to the integrated LFO that provides the modulation you expect when using a Mod wheel.

The Sound Library:

This is what differentiates each volume, otherwise identical. Each volume comes with a "data file" full of multisampled presets that are easily accessed through the interface of the plug-in. No mess on your hard drive, just the two files. Plugsound is supplied on a CD-ROM and the data files are between 540 and 660 MB in size. This is roughly ten times more waveforms than even the most recent synthesizers. Enough to offer the most in-depth exploration of a given instrument that a tone module has ever done.

Each Plugsound volume is a virtual instrument based on a professional sample library, available as a VST2 plug-in (Mac-PC), as well as a MAS and RTAS plug-in on the Mac. If you're on PC you'll need Cubase, Logic or another VST2 compatible sequencer. Owners of Sonar should use DirectiXer to use Plugsound. On Mac, any VST2 program like Cubase or Logic, but also Pro Tools and Digital Performer can enjoy Plugsound.

Plugsound Volume 1: Keyboards Collection

The Keyboards Collection features acoustic pianos in classical, jazz and pop styles, and alternative sounds like honky-tonk and detuned. It also features a best-of selection of famous "suitcase-style" electric pianos from the most respected manufacturers of these bygone instruments. A wealth of clavinets, electric organs and synth-based keyboard sounds complete the modern part of the collection. Featured traditional keyboard instruments include church organs, accordion, celesta, vibraphone, music box and xylophone. Finally, an extensive Harpsichord section benefits from the "Release Trigger" feature that allows a realistic recreation of this difficult instrument. (Synth sounds can be found on Plugsound Volume 5.)

Keyboard instruments may seem simple to reproduce in a library, but that's only true when you let each preset consume hundreds of MBs, and even that doesn't guarantee a better sound. Plugsound's sound design team went out of their way to ensure that all instruments use the amount of RAM they need, and no more. Why create a 200 MB Piano if you can achieve a better sound with 96 MB? In return, this extra work allowed the team to include more sounds in the 650 MB sample library, and make sure that this collection is as complete as possible.

Plugsound Volume 2: Fretted Instruments

Fretted Instruments kicks off with an acoustic guitar section. It includes a Nylon guitar custom-made by acclaimed maker Gerome, and a wonderful "Spanish-style" nylon guitar. Moving on, numerous steel string guitars like a 12-string and several 6-string acoustics (from makers like Guild, Martin, Gerome and Maruha) are featured. Many Dobro and Bottleneck presets add their unique color to the acoustic section.

The electric guitar section packs several clean Strat sounds through D.I. or amps like Jazz Chorus or Twin reverb. The D.I. presets are especially handy if you own an amp modeling effect plug-in. Distorted Strat is included too, captured with the ubiquitous SM57. The Gibson Les Paul is next, sampled through several different amps. The electric section provides the Gibson 335, Jazz guitars, muted playing style guitars and more. The bass section starts with an awesome selection of Double Basses. A wide range of electric basses are covered, both fretted and fretless, including all the classic instruments like Jazz Bass, and many others, covering all current styles. 'Ethnic' section includes a great selection of rare multisamples like Banjo, Mandolin, Celtic guitar, Lute, Sitar (and electric Sitar), Baglamas, Bozouki, Jubus, Oud, a Lute from Morocco, Saz and Tzouras.

PLUGSOUND

Plugsound Volume 3: Drums & Percussion Elements

A superb collection of 5000 drum sounds noted for their versatile and professional characteristics. Acoustic Drum kits are sorted by style: Jazz, Natural (Brushes and Hot Rods), Raw (for pop and rock) and Treatment (distortion, gate and spring reverbs). Next comes the electronic drum section. The first folder offers a selection of sounds sorted by the machines from which they originate—from the 70s to the 90s there aren't many missing boxes. All the classics are there, and many forgotten models as well! The Stylistic kits offer another wide selection of electronic drums which are not so instantly recognizable, and are sorted by style rather than by machine: drums for Dance, Electro, Groove, House, Techno, Trash, Jungle, Disco, Lo-Fi, Vinyl, more.

With so many presets available, emphasis was put on the ease of use. This is why each drum sound is accessible in the three following ways:

- As part of a GM-compatible drum kit.
- In a drum-specific program, e.g. 61 snare samples mapped on the keyboard.
- In a unique "Element" folder allowing quick auditioning of every drum sound in the library simply by browsing presets one by one while your existing sequence plays.

To complete this rhythmic arsenal, an extraordinary percussion selection is provided. It features more than 60 different instruments such as Congas, Djembe, Bongos, Shakers, Triangle, Tambourine, Castanets, Windchimes, Clave, Darbuka, Woodblock, Timbales, Rainstick and many more. Most have velocity switched presets for added realism. The percussion is also used in GM compliant "drum and perc" presets.

Plugsound Volume 4: Hip Hop & R'n B Toolkit

This virtual instrument offers the most complete sonic arsenal for Hip Hop, R 'n B and New Pop/Rock productions. It's made of more than 600 presets using 1,500 samples. Though created by Hip Hop and R 'n B artists, this library offers sounds that will be useful to composers working in many modern musical genres.

The first section features "construction kits" of the latest styles: Timba-style, Funk Mood, Talk-boxed, M&M, Pure R 'n B, Hip Hop. The next section offers drum loops sorted by tempo, instrument phrase combinations, single phrases mapped to the entire keyboard for easy transposition (Bass, FX, Guitar, Hits, Keys, Synth), Vinyl Sounds, Basses (phat, R 'n B), Keyboards, Guitars, Synths, Bass/Synth splits, Vinyl FX, Drum Kits, and sorted Drum presets. This last section features the most powerful R 'n B style drum selection on the planet—the real, heavy stuff!

Plugsound Volume 5: World of Synthesizers

512 patches, over 600 MB of sounds, a huge collection of retro and modern synth sounds. The focus of this collection is to offer presets based on a very wide variety of analog, digital, virtual analog and plug-in synthesizers. World Of Synthesizers focuses on real synth sounds taken from a range of essential machines, from expensive workstations to specialized sound modules. It gives this collection a variety of textures and patches unrivalled by most hardware modules. When compared with recent sample-based sound modules, this Plugsound is the equivalent of having a synth with over thirty expansion cards! But this is only the beginning.

Synth categories offered in the library include Composites (stacks), Flutes, Voices, Tines & Bells, Basses (Acid, Analog, Disco/House, Garage, Electropop, Hip Hop/ R 'n B, Sub, Techno), Pads, Soft Pads, Filter Sweeps, Analog Brass, Leads, Organs, Piano-like, Textures/FX and Short-Reso.

Plugsound Volume 6: Global Collection

The latest and greatest General-MIDI tone module is here! In fact, Plugsound is more than that. It's the next generation of sound module, leaving its hardware counterparts in the dust. Most musicians will find this Plugsound invaluable. It provides a realistic and contemporary library designed to replace yesterday's cheesy GM players, whether software or hardware, with a high-end, professional sound. Fire up Plugsound Volume 6 and suddenly the music you thought dated sounds wonderful! It is also a dream come true for facilities that want to be compatible with musicians without investing in bulky and dated hardware, just for the sake of playing back a few sounds on a session. Finally it will allow musicians who work via the Internet to have an inspiring reference library instead of having to settle for the lowest common denominator.

Global collection faithfully follows the GM standard for sound classification. The Acoustic and Electric Piano category is first, followed by Pitched Percussions, Organs, Guitars, Basses, Strings, Ensemble Sounds, Brass, Reeds, Pipes and so on. The collection then moves on to Synth Leads, Pads and Composites, followed by Ethnic, Percussive and Sound Effect categories. Volume 6 also offers another 128 presets consisting of light versions of the original 128 GM presets. Finally, 7 GM drum kits are supplied, followed by 7 light versions. Most categories feature exceptional samples, including those difficult to get orchestral sounds that are particularly good in this global collection. The keyboards, drums and guitars are also outstanding



PLUGSOUND BOX

Complete Bundle of Plugsound Instruments

Plugsound Box regroups the six individual plug-ins, allowing musicians to access the entire sound library at once, a whooping 3.56 GB of sounds in total. Plugsound Box is a simple to use and CPU-efficient plug-in collection.

UNIVERSAL AUDIO

PRO TOOLS PLUG-INS

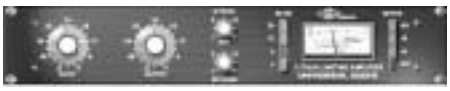
Universal Audio's line of plug-ins, the spot-on digital recreations of the LA-2A, 1176LN, Cambridge, DreamVerb Reverb and Pultec EQ, were previously only available for the UAD-1 card, but are now available for Pro Tools|HD and Pro Tools 24|MIX systems.

Cambridge EQ



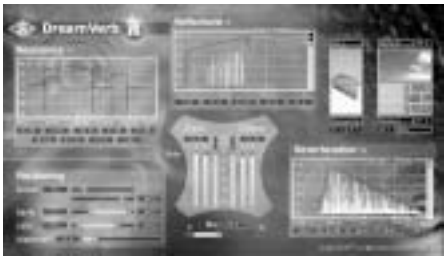
When the engineers at UA set out to create the ultimate EQ, they didn't compromise. The algorithm was designed from the ground up to emulate the response of a high-end analog EQ. The result is Cambridge, which boasts five bands of switchable parametric or shelving EQ and two comprehensive high and lowpass filters, including brickwall elliptical filters. Advanced editing features include graphical frequency response curve with zoom controls for fine adjustment, as well as adjustment of the gain, Q, and frequency of each band on the graphic display. A/B switching is also included for easy comparison of two settings. All these features allow the Cambridge EQ to handle every EQ'ing chore, from the subtleties of mastering to radical tone shaping.....**00.00**

1176LN Stereo Compressor



An analog classic that is well known throughout the audio industry. Designed and built during the 1960's, the 1176LN was the first to use a field effect transistor (FET) as a voltage controlled variable resistor, and that innovation was the key to the product's unique character. An in-depth analysis of the 1176LN's "personality" allowed Universal Audio to methodically reproduce those results within a plug-in architecture, and now offer that same character to Pro Tools TDM HD and Mix Systems.....**00.00**

DreamVerb Reverb



DreamVerb, Universal Audio's flagship stereo reverb plug-in, draws on the unparalleled flexibility of RealVerb Pro. It's intuitive and powerful interface lets you create a room from a huge list of different materials and room shapes. These acoustic spaces can be customized further by blending or "morphing" the different room shapes and surfaces with one another, while the density of the air can be changed to simulate different ambient situations.

DreamVerb also features a flexible 5-band active EQ and unique level ramping for the early and late reflections for ultra-realistic dynamic room simulation. And with Universal Audio's proprietary smoothing algorithm, all parameters can be adjusted in real-time with no "zipper-noise" or audible artifacts. From a vibrantly dynamic room to a rich, deep cathedral, DreamVerb is the reverb of your dreams.....**00.00**

Pultec EQ-P1A



A faithful electronic reproduction of the classic hardware equalizer. The Pultec EQ has long been a choice of recording and mastering engineers for its ability to bring out individual frequency ranges without significantly altering other frequencies. In addition, the Pultec is one of those magical pieces of gear that makes audio sound better - just by passing through it. The sophisticated modeling technology used in the Pultec EQP-1A plug-in captures both of these key characteristics.....**00.00**

LA-2A Leveling Amplifier



The Teletronix LA-2A leveling amplifier is practically the definition of vintage audio gear. The original opto-electrical attenuator circuit is responsible for the incredibly rich and magic sound embracing the tracks of so many classic recordings in circulation today. In an effort to bring this product and its character into the world of DAWs, Universal Audio studied not just the sound, but how each and every component interacted and behaved under a full range of studio conditions. The word, "emulation", hardly does this plug-in justice. It is in fact a digital copy of the LA-2A, and upholds all the classic, analog characteristics of the original with unprecedented clarity and sound quality.....**00.00**

BUNDLES:

Since you'll discover that owning only one of the finest plug-ins for TDM is not enough, Universal Audio offers plug-in bundles for a substantial savings.

Compressor Bundle (LA-2A and 1176LN)

EQ Bundle (Cambridge EQ and Pultec EQP-1A)

Studio Bundle (Comressor and EQ Bundles)

Super Studio Bundle (Studio Bundle plus Dreamverb)

SMARTCODE PRO — DOLBY DIGITAL OR DTS

Software Surround Encoders for Pro Tools (AudioSuite)

No surround production is complete without encoding. And in a busy production facility, efficiency is critical. SmartCode Pro streamlines the surround sound production process. By encoding in 5.1 within Pro Tools, your facility avoids the bottleneck of dedicated hardware encoders which saves you both time and money.

SmartCode Pro comes in three versions to accommodate the two most widely used surround formats: Dolby Digital and DTS. All three versions are AudioSuite software programs that allow Pro Tools users to preview their 5.1 surround mixes in real time 5.1, then encode and decode the mix to create a 6-channel surround master.





FEATURES

One Step Encoding

All three SmartCode products allow you to create encoded files ready for DVD production or CD playback directly from Pro Tools. To create an encoded file in SmartCode Pro/DTS or SmartCode Pro/Dolby, select the six Pro Tools tracks that comprise your surround mix—then press the "Encode" button to create your new .dts or .AC3 file.

Preview Your Mix

There is simply no easy way to preview a complete surround mix outside the studio. With SmartCode Pro/DTS-CD (included with the full DTS encoder), you can make a preview CD that will play on any CD/DVD system equipped with a DTS decoder. Even if the project is destined for Dolby Digital, you'll love this valuable tool for providing a quick and cost-effective way to test your surround mix.

Smartcode Pro Dolby

- Encodes Dolby Digital (AC3) surround format
- Includes right to use Dolby logo on encoded material
- Complete integration with Pro Tools
- One button to launch encode process
- Supports all Dolby Digital bit rates
- Supports all surround modes including 5.1, mono, stereo, LCR, LCRS, 5.0, etc.
- Flexible channel/track assignment
- Pre-processing options include Low-pass, LFE low-pass, DC filter and 90° phase shift
- Selectable 3dB attenuation option for surround channels

Smartcode Pro DTS-CD
Smartcode Pro DTS-DVD

- Right to use DTS logo on encoded material
- Complete integration with Pro Tools
- One button to launch encode process
- Supports 44.1 (DTS-CD) and 48 kHz/24 bit audio files
- Flexible channel/track assignment
- LFE encoding option
- Rear channel attenuation option
- Supports HDCD (High Definition CD) flag

VIRSYN TERA

The Next Generation of Software Synthesis



Tera is a modular software synthesizer based on five different synthesis technologies including the revolutionary "Spectrum Synthesis". This synthesis allows the creation and manipulation of resonant bodies with up to 128 formant frequencies. The innovative user interface of TERA sets a new standard in design and functionality. This simplifies the access to the huge sound space of its predecessor. The classical modulation matrix together with a simple method for patching modules with virtual cables makes operating the modular system of VirSyn TERA as easy as other synths with a fixed structure. The integrated step sequencer is pattern/song oriented and can even be used in a VST host environment.

WAVE MECHANICS

PITCHDOCTOR



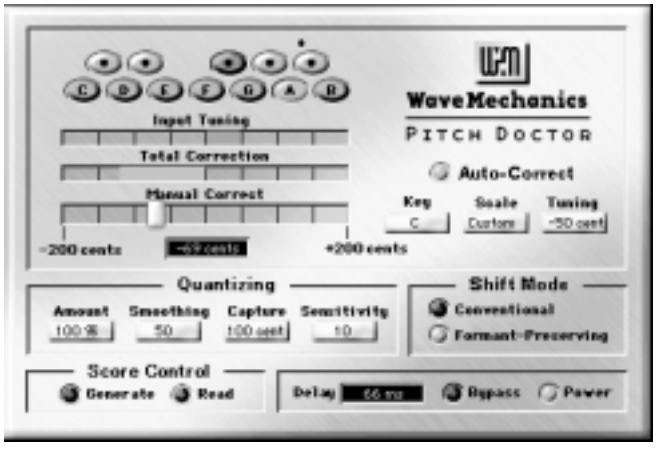
Formant-Preserving Automatic Pitch Corrector

PitchDoctor is a realtime, fully automatable TDM-based plug-in that combines ultra-smooth pitch-shifting with an advanced intelligent pitch analyzer for pitch correcting out-of-tune vocals and solo instrument performances. When an out-of-tune performance is a problem, PitchDoctor is nothing less than a session-saving miracle. Simply enter the desired key and scale, and Pitch Doctor automatically adjusts the intonation of any out-of-tune notes. Because Pitch Doctor is the only intonation correction plug-in with total automation and PurePitch's formant-preserving quality (no chipmunks!), it's the only choice for truly challenging correction work. And it's only available for Digidesign's TDM hardware system.

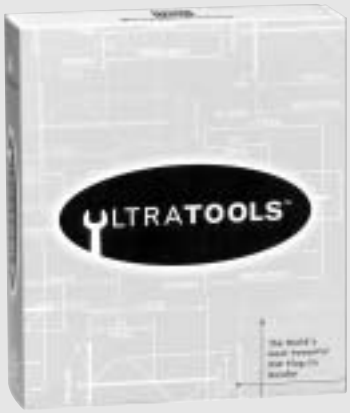
FEATURES

- Automatically adjusts the intonation of any out-of-tune notes to a preset or user-defined key and scale in realtime.
- Full automation capabilities (Pro Tools 4.0 or later) allows key, scale, and scale customizations to be changed mid-track.
- Score Control allows you to specify the correct melody to which Pitch Doctor will correct pitch. The score editing feature works seamlessly with ProTools automation editor, allowing easy, and intuitive editing of the desired musical score.
- On-screen keyboard displays an editable musical scale for correcting pitch as well as a visual readout of the detected note.
- Guide Track works in with score generation to allow any instrument to be used to create a score to guide PitchDoctor through the pitch correction process.
- Pitch Quantizing controls including pitch correction amount, smoothing, capture range and sensitivity can be used to fine tune the pitch correction process.
- Shift Mode uses PurePitch's formant-preserving quality to eliminate 'chipmunk' effect and give PitchDoctor a wider usable pitch shift range.
- In addition to automated pitch correction, Pitch Doctor provides a large, high-resolution pitch correction fader. The manual correction fader can either be used alone, or in conjunction with the automatic pitch correction mode. Data is easily editable in Pro Tools' automation window.

Pitch Doctor's on-screen keyboard displays the musical scale used for pitch correction, allowing easy scale customization while also providing a visual readout of the detected note. Since Pitch Doctor is fully automatible, the key, scale, and even scale customizations may be changed in mid-track.



UltraTools — The Complete Bundle



Now get all four Wave Mechanics plug-ins in one convenient bundle. Three of the processors, PitchDoctor, PurePitch and SoundBlender are some of the most powerful and DSP-intensive plug-ins available for TDM and feature full realtime controllability as well as complete automation capabilities within ProTools 4.0 or later. The fourth processor, Speed uses the AudioSuite environment and is therefore compatible with all levels of Pro Tools including Pro Tools LE and even Pro Tools Free.

PUREPITCH

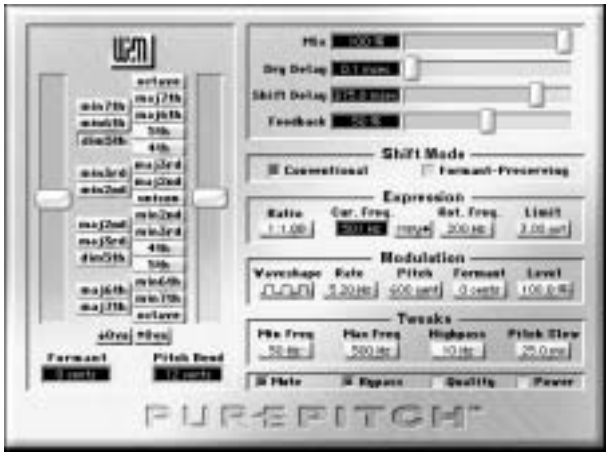


Formant-preserving Pitch Effects Processor

A high quality TDM-based plug-in designed to accomplish natural-sounding pitch-shifting in real-time, PurePitch has become the standard for its high-quality pitch alteration and unique vocal-processing effects. Using PurePitch, you can easily create realistic harmony parts from a lead vocal or instrument, deepen a vocal part without changing it's pitch, or add vibrato to a flat performance. For voice-over work, PurePitch can be used to alter the formants and pitch inflections of a speaker, producing subtle character alterations, or more extreme gender and species morphing effects. PurePitch includes over 50 preset patches for music production, sound design, and post-production. And since it is a realtime plug-in — as soon as a knob is tweaked you'll hear its effect. PurePitch is even automatable under ProTools 4.0 or higher so you can edit harmonies to create a virtual background vocal section or morph parameters through extreme sound transformations. And it's only available for Digidesign's TDM hardware system.

FEATURES

- Pitch Control Panel uses a keyboard display with interval indicators for selecting vocal harmonies. This allows you to create musically useful harmony vocals without another singer. All harmony intervals can be easily recorded and edited using Pro Tools automation.
- The Pitch Bend slider alters the amount of pitch shift in cents (100 cents = 1 semitone) and can be used for double tracking or detuning a track.
- Vocals can be pitch shifted without affecting their formant structure — the natural resonances that make voices unique.
- The Formant slider can be adjusted in small amounts to create subtle double-tracking effects, in larger amounts to deepen voices or for gender bending effects or at the most extreme warp voices for animated 'chipmunk' and 'alien' effects.
- The formant amount can even be automated to morph vocal characteristics in realtime.
- Mixer Control Panel — Provides control over Wet/Dry mix, Dry delay time and pitch shifted delay time with feedback control.
- The Modulation Control Panel allows you to apply an LFO (Low Frequency Oscillator), with a choice of 6 waveforms and a variable rate control, to modulate pitch shift amount, formant amount and amplitude.
- The Shift Mode Control Panel allows you to select between conventional or formant preserving pitch shifting.
- PurePitch's exclusive Expression Control Panel allows you to dynamically compress or expand the pitch envelope of vocal tracks. This enables the emotional content of dialogue tracks to be modulated, in subtle, or not-so-subtle amounts. The more the track is compressed, the more monotone it becomes. (Adding expansion increases expressiveness).
- Tweaks Control Panel — helps optimize the quality of pitch processing by offering control over minimum and maximum frequency ranges, a Highpass filter and a Pitch Slew (portamento) control adjustable in ms.
- Includes over 50 preset patches for music production, sound design, and post-production — Vocal Harmonizing, Chorus and Doubling, Tremelo & Vibrato, Dialogue Enhancement, Vocal Transforms for game designers, animated characters, and sound design, Spacey Delay Effects, Wacky sound design effects.

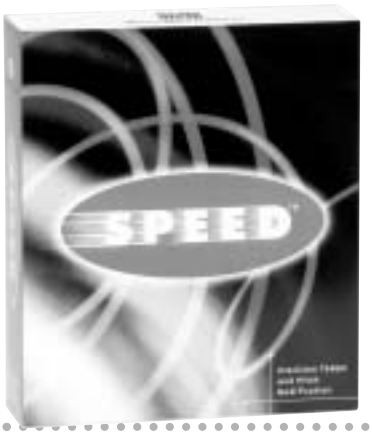


Create Bizarre, Unheard-of Effects

- With its built-in digital delays, multi-waveform LFO, and pitch, formant and amplitude modulation, PurePitch is a veritable cornucopia of new and fantastic designer sounds. But you don't need to be a sound designer to use PurePitch - they've included dozens of hand-crafted effects.
- Perfect for game designers, animated characters, and sound design.
- One-hundred percent real-time.
- Uses one DSP chip for each channel of pitch processing.
- Automation of all parameters

WAVE MECHANICS

SPEED



Time Compression/Expansion for AudioSuite

Speed is an AudioSuite plug-in that allows you to change the tempo of audio tracks without affecting pitch, transpose pitch without affecting tempo or even alter tempo and pitch simultaneously. Speed works equally well on drum tracks, single instruments and even entire mixes — processed audio tracks are free from unwanted artifacts while transients and rhythmic consistency are preserved. Speed is perfect for time conforming commercial spots and is the ideal tempo and pitch matching utility for today's loop-based music productions. Available by itself or as part of the UltraTools bundle for Mac and Windows XP.

- Alter tempo from half-speed to double-speed, transpose pitch over a two-octave range (±1200 cents) without distortion
- Maintains phase synchronization between left and right channels when processing stereo tracks
- Compatible with Digidesign's AudioSuite native plug-in format — no TDM hardware required
- Preview tempo and pitch changes in real-time before committing the changes to disk (CPU dependent)

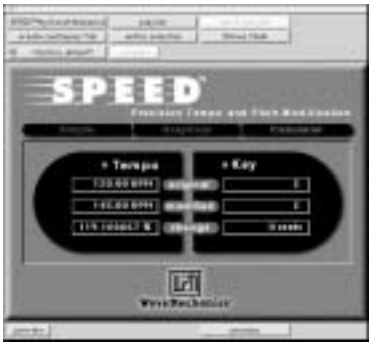
Simple Control Panel

- The Simple control panel provides straight forward tempo and pitch knobs adjustable in different units — percentage of tempo, length, pitch in semitones, cents, or frequency ratio

Graphical Control Panel

- The Graphical control panel works just like ProTools automation, letting you draw time-varying pitch and tempo changes using a waveform/timeline graph
- Effects created in the graph mode can be saved as a preset





Calculator Control Panel

- Makes it easy to transform any tempo, length, or musical key to any other. Values for the original, modified and change for Tempo/Length and Key are clearly displayed.
- Tempo and Length calculators control speed without affecting pitch and the Key calculator transposes pitch without affecting Tempo/Length.

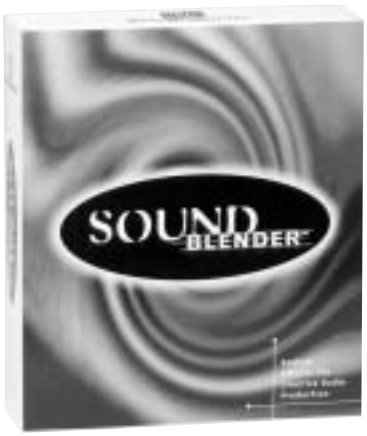
Sound Toys – FilterFreak

FilterFreak is an analog-modeling filter plug-in that takes a truly musical approach to creative sound design. Though it was inspired by studio filtering boxes like the Mutronics Mutator and the Sherman FilterBank, FilterFreak goes far beyond what any of those boxes are capable of.



- The super-steep 48dB per octave analog modeling filter features variable filter slope, resonance and filter shape, allowing you to create nearly any filter character - from smooth and gentle to aggressively resonant.
- Overdriving the input or output stage allows you to achieve an even dirtier sound.
- The modulation section offers five different modulation sources for creating radical filtering effects. These include: Tempo-locked LFO, Envelope Follower, Random LFO, ADSR, and Triggered Random Stepper.
- User programmable waveshapes, programmable rhythmic patterns are also provided.
- All of this flexibility is available with a user interface that is as familiar and intuitive as an analog compressor or EQ.
- Support is currently provided for RTAS, HTDM and AudioSuite plug-in formats.
- Keep an eye out for seven more plug-ins over the next several months.

SOUNDBLENDER



The Ultimate FX Processor for Pro Tools

SoundBlender is a TDM-based plug-in that combines classic hardware effects such as pitch-shifters and digital delays with the resonant filtering, arpeggiation and modulation capabilities of analog synths to produce thousands of never-heard-before signal-processing effects. SoundBlender is broken down into two distinct realtime plug-ins — PitchBlender and TimeBlender each with their own effects-processing modules with a unique Tri-Modulation Matrix and Tempo-Based control. Hundreds of pre-programmed patches are included ranging from subtle choruses to radical over-the-top sonic experiences. Available by itself for the Mac or as part of the UltraTools 2.0 bundle for Mac and Windows.

FEATURES

- BPM Control synchronizes delay and modulation effects to the tempo of your song.
- Three modulation sources can be mixed and routed to any of dozens of effect parameters.
- The available modulation types include — LFOs, Random, Triggered, Toggled, Envelope Detector or Gate
- Variable Soft-Clipping algorithm adds the warmth of analog saturation to any signals boosted beyond the digital signal range.
- Sidechain input lets you trigger the input source with any of the modulators and can be used to synchronize triggered effects to a rhythm track, or to create ducked effects.

Pitch Shifter/Pitch Mapper

- Pitch-shift range of ±2400 cents (2 octaves) can be used to create anywhere from traditional chorusing, double-track effects and harmonies to abstract arpeggiations and wild pitch modulation. Each pitch-shifter has its own pan and level control and can be modulated by the modulation matrix.
- Pitch Mapping is an intelligent harmony and arpeggiation feature that detects the pitch of the input signal, and dynamically adjusts the pitch shift interval depending on the selected scale, key and pitch shift interval.
- Pitch Map parameters let you use a chromatic instrument (piano), and map it through one of 21 western and ethnic scales.
- Any modulation source may be used to control the pitch-shift interval to create interesting in-key arpeggiation and randomized harmony effects.

TimeBlender

- TimeBlender is a two-channel effects processor that combines two reverse pitch-shifters with two delays, two filters, and a feedback mixer.

PitchBlender

- PitchBlender is a two-channel effects processor that combines two pitch-shifters, two digital delays, two filters, a mixer and a signal flow/feedback matrix.



Delays

- The two Digital Delays can be controlled in either time (up to 1000ms) or beats.
- Each delay has its own pan and level control and can be controlled by the modulation matrix.

Filters

- Each of the two Filters has an adjustable center frequency and resonance (Q) control and can be configured as a lowpass, highpass, bandpass or notch filter. The center frequency can be modulated by the modulation matrix.

Signal Flow Parameters

- There are four selectable algorithms that allow you to re-wire and modulate the effects-processing modules in a variety of useful configurations.
- The relative feedback level for channel 1 and 2 can be adjusted as well as blended and crossfaded for creating dense chorus and reverb-like textures
- Filter effects can be assigned to any location in the signal chain.

Reverse Pitch Shifting

- Reverse pitch shifting with a range of ±2400 cents (2 octaves) samples sections of the input audio (up to 1000ms) and plays it backwards mimicking backwards tape effects
- The two Digital Delays can be controlled in either time (up to 1000ms) or beats
- Each Reverse Pitch Shifter has its own pan and level control and can be modulated by the modulation matrix.
- The relative feedback level for channel 1 and 2 can be adjusted as well as blended and crossfaded for creating dense chorus and reverb-like textures.

Presets

- Bundled with hundreds of presets hand-crafted by audio professionals. Includes stereo detuning, intelligent pitch-shifting, diatonic arpeggiation, ducked delay effects, envelope filtering, auto-panning, tremolo, flanging, chorus, reverse pitch-shifting, and much more.

WAVES

NATIVE POWER PACK / TDM BUNDLE



Plug-in Software Bundles

You need pro-audio equalizers, compressors, reverbs, dynamics, multi-tap delay, mastering processors and more to make your productions the best they can be. They're all here. Waves' Native Power Pack and TDM Bundle are complete software bundles of audio processing essentials.

Designed for musicians in mastering, remix and restoration applications, they are also ideal for those creating multimedia titles, authoring audio for the web, or designing sound for games. Everything you need to process audio is provided, so you can finish an entire project without requiring any other plug-ins.

The high-quality plug-ins bundled in the Native Power Pack and TDM Bundle include the famous TrueVerb virtual-space reverb, the "Swiss-army knife" Q10 equalizer, C1-compressor and C1-gate, the incomparable S1-Stereo Imager, and of course, the essential L1-UltraMaximizer mastering peak limiter. Together, they let you shape your audio with precision and creativeness.

The Native version supports RTAS, AudioSuite, VST, MAS and DirectX formats on Mac and Windows with sample rates up to 88.2/96kHz. The TDM version adds support for Digidesign's ProTools TDM format on the Mac, including 192kHz compatibility for ProTools HD. And for those that will later need more processing, Waves offers an upgrade path to the Gold or Platinum Native and TDM Bundles.

Surgical toolkits for tracking and mastering. Superb tools for everyday needs including limiter, EQ, compressor delay, reverb, and dynamics.

Bundles Include	❧ L1 Ultramaximizer ❧ Q10 Paragraphic EQ ❧ C1 Compressor/Gate ❧ TrueVerb Reverb/Distance Processor ❧ S1 Stereo Imager ❧ De-Esser ❧ SuperTap 2-Tap Delay ❧ PAZ Psychoacoustic Analyzer

L1 Ultramaximizer: The L1 is a mastering level brickwall limiter that combines state-of-the-art peak-limiting with advanced requantization, delivering the maximum level and highest resolution with little or no loss of dynamics.

SuperTap: Provides two taps with up to 6 seconds of delay with complete precision control of gain, panning, and filtering on every tap, plus feedback with filtering. Use the tap pad to set rhythms or tempo.

S1 Stereo Imager: The S1 offers unmatched stereo image tools, with widening, rebalancing, recentering, and MS functions. Adjust left and right independent of center channel. Includes phase-compensated Blumlein shuffler.

Q10 Paragraphic Equalizer: The Q10 is the most powerful transparent EQ available. It offers over 200 presets including notch, super notch (-46dB), hum removal, comb, Baxandall, brickwalls and more.

De-esser: Excellent dynamics processor gives you precision editing of annoying "hiss" that can mar your recordings.

Trueverb Room Emulator: TrueVerb offers virtual real-room acoustic emulation. Implementing the acoustically correct approach, you define the room size, frequency response and distance the listener is from the sound source. Includes a comprehensive setup library.

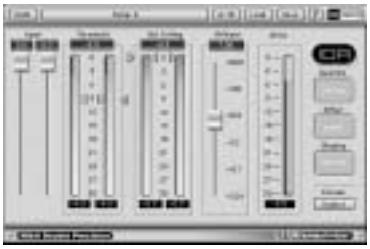
C1 Compressor/Gate: The C1 is a powerful and transparent compressor capable of frequency specific compression and gate using the sidechain. Its presets include de-essers, enhancers, expanders, dehissers, dereverb, ducking, keyed effects and more.

PAZ Psychoacoustic Analyzer (TDM Bundle only): A realtime analyzer based on human hearing, PAZ shows 52 or 68 bands with RMS, peak, weighting, variable integration time, and more. Perfect for mastering, trouble-shooting, environmental analysis.

Waves has built a worldwide reputation for quality and technical leadership in the professional audio signal processing market. They are the market leader for software Plug-Ins used in the audio content creation process for music, movies and computer games with over 150,000 users worldwide. Waves offers both the broadest selection and the highest quality set of software solutions available. Its software solutions support over 20 different audio editing environments, both for native CPU operation (Windows and Mac operating systems) and DSP accelerator solutions.

NATIVE POWER PACK / TDM BUNDLE

L1 Ultramaximizer



The L1 is a mastering level brickwall limiter that combines state-of-the-art peak-limiting with proprietary word length reduction that maximizes audio levels and resolution while minimizing quantization noise — create 16-bit files with 19-bit perceived resolution. Use L1 to maximize audio levels for CD's and multimedia and as as the final audio processor in mastering. The L1 installs two separate plug-ins into your host application: L1 limiter, a mono/stereo wideband limiter without IDR, best used as a track insert for voice, instruments or effects. L1-Ultramaximizer, the "full" mastering plug-in with limiter and all IDR options.

- Peak limiting interface consists of input attenuation, threshold, output ceiling, release time controls and a gain reduction meter.
- As the Threshold is lowered below the level indicated by the peak threshold meters, the output level is increased but only to the maximum set by the output ceiling control
- Release time controls how fast the L1 recovers to the constant gain after a peak is encountered
- Lookahead peak limiter for maximum level
- 48-bit internal resolution for TDM and 64-bit on Native systems allows dithering back to 24-bit output for DVD authoring and high-resolution mastering
- Although your audio may not exceed the digital maximum set by the out ceiling control, it may clip when converted to an analog, especially if an inferior A-D converter is used. To combat this, choose Analog Domain for your final mastering ceiling out reference and this will prevent sample from going over when converted to analog.

IDR (Increased Digital Resolution)

- Waves' proprietary word length reduction system, IDR implements Quantize, Dither and Noise-Shaping to ensure maximum digital resolution when reducing the number of bits in your audio
- Quantize allows you to chose the final bit depth output (8-, 16-, 20-, 24-bit). The L1 works at 48-bit resolution, and is dithered back to 24-bit when this option is chosen
- Dither controls the type of dithering process to be used:
  - Type 1 yields lower distortion but adds about 5 dB to the noise floor
  - Type 2 yields lower dither level without increasing noise floor
- Noise-shaping options include Moderate, Normal, Ultra or non and can be applied to suit your preference for each application

Q10 ParaGraphic EQ



The Swiss-army knife of equalizers, the Q10 provides up to 10-bands of precision mono or stereo EQ. An extensive Setup Library includes 200+ presets, from gentle sweetening EQs for mastering to extreme filters for hum removal and creative effects, suitable for every aspect of audio production.

- ➤ *88.2/96 kHz ready for many native platforms*
- ➤ *Separate Left and Right, Strapped and Mono processing control*
- ➤ *Frequency Curve Graph and Value Windows provide interactive control over Frequency, Gain and Q*
- ➤ *Supports both mouse and keyboard control*

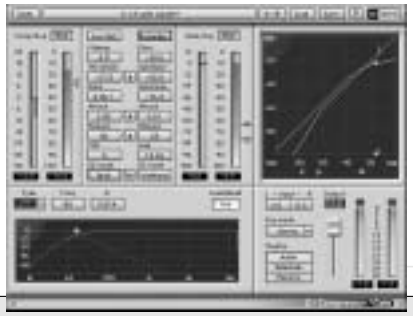
Easy To Use Graphic Interface

- Left and Right Input trim controls with phase reversal switches for each channel
- Left and Right Output trim controls plus an Auto Trim button
- Left/right peak meters with clipping indicators
- Frequency Curve Window lets you adjust the gain and frequency of active bands and offers a graphic representation of the adjustments in relation to the filter type being used and the interaction between any of the active bands.
- Matrix style interface with separate controls for each of the 10 EQ bands
- Selectable filter types- Bandpass (or bell curve), Low shelf, Hi shelf, Low and Hi Pass
- Gain Control ±18.0 dB in 0.1 dB increments
- Frequency control with 16 Hz - 21.357 kHz range in 1/16-tone steps (96 steps per octave)
- Q parameter offers exceptionally wide to narrow bandwidth control; constant bandwidth filter boost or cut gain as needed.

Extensive Setup Library

- Instant access to over 200 complex filters divided into manageable categories including Multimedia Tools, Pseudo Stereo, Band-limiting, Pre/De-emphasis, Plateau and Tilt filters, Crossovers, Harmonic combs, etc.
- Setups are adjustable and cover applications for mastering, hum and hiss reduction, AM radio and telephone simulation effects, pseudo-stereo, comb filtering, brickwall crossover filters and many more.

NATIVE POWER PACK/TDM BUNDLE, *continued*



C1 Parametric Compander

The C1 consists of three independent stereo processing modules: Compressor/expander, Gate/expander, and Filter/equalization. The EQ mode of the Compressor/expander may be chosen independently of each other. For example, you can combine a wideband compressor with a high frequency gate. The Threshold/GateOpen, Attack, Release, EQ Mode controls of the two dynamics modules can be operated simultaneously.

C1 consists of six component plug-ins based on the three available processor modules allowing you to select the specific audio processing you need without wasting DSP power. Each component plug-in has its own factory and user presets.

C1comp -Wideband compressor/expander with no sidechain, for use as a basic track insert

C1gate - Wideband downward expander/gate with no sidechain

C1c/sc - Compressor with sidechain or split-band compression can be used as a high or low frequency limiter, de-esser, de-popper, sidechain drum gate, standard limiter/compressor/expander or no-pump sidechain mastering compressor

C1comp/gate -Wideband compressor and gate combo with no sidechain for general use outboard track insert

C1c/g+- The "full" plug-in, with compressor, gate, sidechain EQ, and IDR for use as a precision multi-function, multi-band mastering processor; as a wideband compressor with high frequency gate; simultaneous compression and upward presence enhancer

C1c/g — Same as +C1c/g without IDR

Compander/Expander Module

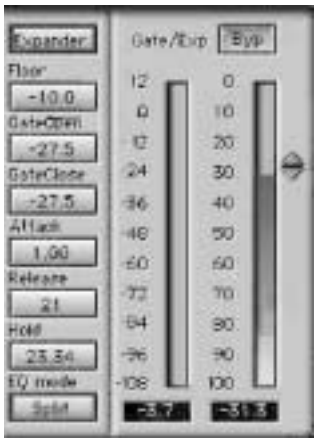
Provides basic soft knee variable-ratio compression, expansion and limiting using the following controls:



- ◆ Toggle between two reference modes to control how the Makeup Gain stage will react to gain reduction:
- **LowRef** (low-level) is the standard mode for compressors
- **PeakRef** (Peak-level) the output gain automatically increases as the threshold is lowered, maintaining a consistent output level

- ◆ **Makeup Gain** controls the output level with a range of -40 dB to +40 dB
- ◆ **Threshold** sets the input level, in dB, above which the compression/expansion takes place
- ◆ **Ratio** control is variable from 0.5:1 (Expansion) to 50:1 (Limiting)
- ◆ **Attack** time from 0.01 msec to 1 second
- ◆ **Release** time from 1 msec to 10 seconds
- ◆ **Program Dependent Release** can apply a short release time to transient signals while at the same time maintains a more relaxed release time for normal dynamic material.

Gate/Expander Module



- ◆ Full range of adjustments independent of those in the Comp/Exp module. However, some of the controls of the two modules may be linked together for ease and speed of adjustment.
- ◆ **Floor** sets the lowest level, up to 12 dB, that a sound will be taken to when gated or expanded. When set to infinity, the level can be taken down to digital silence.
- ◆ **GateOpen** sets the input threshold level, in dB, to let signal through. In Expander mode, GateOpen is the level at which soft knee expansion starts acting to a significant degree
- ◆ **GateClose** controls the level below which gate is closed to stop signals from getting through.
- ◆ **Attack** time from 0.01 msec to 1 second
- ◆ **Release** time from 1 msec to 10 seconds
- ◆ **Hold** adjusts the length of time in milliseconds that the gate is to be held open — useful for creating gated effects such as gated reverb
- ◆ **Control level** meter shows the level of the wideband or sidechain EQ signal controlling the processor. The meter time constants match the Gate/Exp attack and release times.

NATIVE POWER PACK/TDM BUNDLE *continued*

◆ The **EQ Mode** switch has three modes:

— **Wideband mode** operates as a conventional expander/gate or compressor

— **Sidechain mode** passes the desired sidechain audio signal used to control the dynamics through the filters in the EQ module so that the dynamics respond only to a defined frequency band

— **Split mode** not only filters the sidechain audio signal, it also separates the main signal into two bands: an active band which will be processed and a passive band which is not processed

◆ The **Gain Reduction Meter** shows instantaneous gain reduction and increase changes below and above 0 dB. A value window above each meter shows the largest absolute peak since the last 'reset' of the meter.

Filter Module

The Filter module allows you to designate frequency bands for use with the Sidechain EQ and Band split modes:



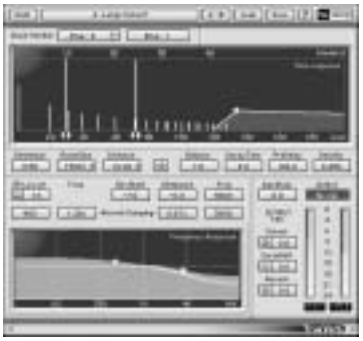
◆ Type selects one of the four filter (bandpass, band reject, high pass, or low pass) to be applied to the active band

◆ Frequency range is selectable from 16 Hz to 21 kHz. When the band pass or reject filter is selected, the chosen frequency is at the center of the pass or reject band. When a low or high pass filter is used, the chosen frequency is at the crossover point between the two bands

◆ Q range is selectable from 0.1 to 0.6. When using a band pass or reject filter, Q controls the width of the band. For low and high-pass filters, Q determines the steepness of filter cutoffs

◆ Graphic Display shows the filter type and the interaction of the active/passive bands

TrueVerb Reverb/Room Simulator



TrueVerb is a room emulator/reverb processor that uses separate early reflection and reverb algorithms to generate compelling virtual spaces. Control the Room size, its decay time and frequency characteristics as well as the Distance between the sound source and listener.

➤ *Create natural sounding rooms without distorting the original stereo image*

➤ *True Stereo compatibility*

➤ *Two user modes: Send mode or Thru Mode where TrueVerb is inserted on a stereo submix or master outputs to create a virtual room for groups of instruments or an entire mix*

➤ *Constant Perceived Level ensures a consistent overall mix as you tweak TrueVerb's parameters*

➤ *Extensive setup library with dozens of presets, many tailored specifically for either Send or Thru mode usage*

➤ *Separate editable graphic Time Response and Frequency Response displays*

Input/Output Section

◆ Input Gain A full-range control, from 0.0 to silence (-infinity), adjusting the input level

◆ Direct On/Off toggle switch for the dry input signal with a Direct level gain control

◆ EarlyRef On/Off toggle switch for the early reflection signal with an Early Reflection level gain control

◆ The Reverb On/Off toggle switch with a Reverb level gain control

◆ VU meters show the stereo output level. A clip indicator lights red if a single sample reaches 0dBFS (Full Digital Scale). A numeric sample-accurate peak hold shows the highest output level value

Frequency Response Controls

◆ ER Absorb– output HF filter for the Early Reflections

◆ Freq control– sets the frequencies for the RevShelf and the ER Absorb

◆ RevShelf:– HF shelving filter for the Reverb

◆ Control the high and low frequency character of the Reverb using adjustable frequency and Damp Ratio controls

Time Response Controls

◆ ***Dimension*** controls the character of the Early Reflections to simulate a 1-4 dimensional space

◆ ***RoomSize*** controls the size of the room in cubic meters

◆ ***Distance*** controls the apparent distance, in meters, from the listener to the input source.

◆ ***Link*** matches the level of the Reverb and PreDelay to the last reflection in the Room

◆ ***Balance*** controls the balance between the Reverb and the Direct/EarlyReflections levels

◆ ***DecayTime*** determines the rate at which the early reflections and reverb fade away

◆ ***PreDelay*** controls the delay time, in milliseconds before the reverb begins

◆ ***Density*** controls the initial buildup of reflections in the Reverb

NATIVE POWER PACK/TDM BUNDLE *continued*



S1 Stereo Imager

S1 combines a number of stereo processing techniques, based on recognized engineering principles, with an intuitive user interface. Four component plug-ins, S1 provides powerful stereo enhancing and re-balancing tools for use in mixing, mastering and digital editing without adding unwanted tonal, phase coloration and with a high level of mono compatibility with minimal side effects. Supports 88.2/96kHz native host applications as well as 48-bit double precision resolution for TDM.

Easy To Use Graphic Interface

- **Width** control alters the width of the input stereo signal. Can be narrowed to mono or widened to move beyond the loudspeakers
- **Asymmetry** is a post-width control that adjusts the relative level of left and right sounds without moving the mix off-center or favoring either side of the stereo image
- **Rotation** control is used for centering a stereo mix biased to one side without altering the mix level balance, or as a true stereo 'pan' control for mixing stereo-miked sources
- **Shuffling** and **Frequency** work in tandem to increase the stereo width and spaciousness of bass frequencies which are usually perceived as omnidirectional sound energy
- **Shuffling** controls the bass width while the **Frequency** controls the frequency (between 350 and 1400 Hz) below which the Shuffling effect is increased
- **Bass Trim** adjusts the level of bass which may shift when Shuffler is in use
- **LR/MS** button changes the input mode to accept either standard Left-Right inputs or **Mid/Side (MS)** inputs for decoding tracks recorded with a MID/Side stereo mic configuration. Left and right channels can be swapped and polarities can be switched
- **Gain** controls overall gain level

Four S1 component plug-ins let you select the amount of audio processing you need.

- **S1 Shuffler** is the full plug-in, with all controls, for mastering, fixing off-center or unbalanced mixes, precision image adjustment, changing wideband width or expanding the low frequency image as well as Rotating true stereo sources without narrowing the stereo stage
- **S1 Shuffler (48)** is the full mastering plug-in for TDM users with 48-bit double precision resolution dithered to a 24-bit output
- **S1 Imager** is less processor intensive then S1 Shuffler with Width, Rotation, and Asymmetry controls for enhancing stereo mixes or Rotating stereo-mic and submixes into place)
- **S1 MS** converts left-right input into Mid/Side stereo (MS) output or vice versa.

De-Esser

High Frequency Dynamics Processor



Equipped with only a few simple controls, De-Esser provides fast and easy de-essing and High Frequency limiting that can be applied to a wide variety of sources especially singing and voice-overs to remove sibilant 'esses' and 'sshs'. A choice of Wideband and SideChain mode allows you to use De-Esser as a high frequency dynamics processor or as a straightforward compressor/limiter. Supports 88.2/96kHz native host applications.

De-Esser Controls

- **Threshold** slider adjusts the level above which attenuation takes place in the SideChain. To adjust the threshold, click and drag the slider down until the 'esses' are sufficiently reduced
- **Audio** control toggles between **Wideband** and **Split** compression modes. In **Wideband** mode, attenuation is applied to the entire audio band. In **Split** mode the audio is split into low and high frequency signals and attenuation is applied only to the **HighPass** signals
- **Frequency** window allows you to set the center frequency used in the SideChain filter
- **SideChain** button toggles between a **HighPass** filter which looks at all frequencies above the set frequency and a **BandPass** filter which is a narrow band around the set frequency
- **Monitor** button toggles between **Wideband** and **SideChain** only monitoring

De-Esser Displays

- **Attenuation** meter displays instantaneous gain reduction applied to the audio (in dBs)
- **Output** meter provides an infinite peak hold feature that shows the highest output level
- **Energy Detector** located next to the threshold control, displays the energy of the **SideChain** in dB below 0 dBFS). The energy bar-meter and threshold-slider are aligned for easy adjustments and visual feedback

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NATIVE POWER PACK/TDM BUNDLE *continued*

SuperTap 2-tap



SuperTap 2-tap is a two-tap delay with up to six seconds of mono or true stereo delay. There's a dedicated Eq/Filter section for each tap derived from the acclaimed Q10 as well as Global LFO and a feedback section capable of overmodulation and positive gain feedback loops capable of some serious distortion. SuperTap 2-tap can produce a wide range of creative rhythm effects and loops and includes 19 factory presets covering a broad range of multi-tap delay effects, including analog and tape delays, slapback echo, chorusing and more.

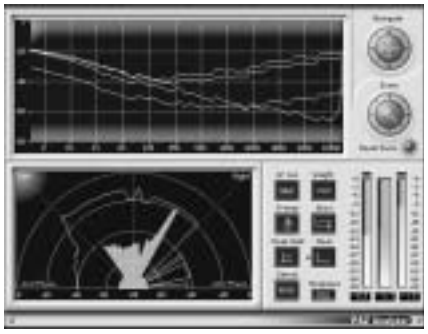
Each Tap Features

- Controls for each tap are arranged in a row from left to right — On/Off, Gain, Rotate, Tap Marker, and Eq/Filter section
- Gain is adjustable in 0.1 dB steps
- Rotate controls the tap's stereo position ±45°
- Each tap has its own slider located in the Tap Delay Grid that is adjusted by dragging the slider from left to right. Each slider shows a value either in BPM or milliseconds (ms)
- Each tap has a EQ/filter section. Each filter has a selectable frequency (100Hz to 20kHz) and six filter curves — Bell (wide) Bell (narrow), Hi and Low Shelf, High and Low Pass. Gain control lets you boost or cut filtered bands without affecting the pass bands
- Pan Graph shows the stereo location of each tap with a marker. Tap markers can be dragged from left to right to adjust Rotation (±45°) and up and down to adjust Gain (from 0.0 to -24dB).

Global Controls

- Grid Mode button changes between BPM (adjustable from 40 to 1200bpm) or ms
- Snap-to-grid feature lets you quantize each tap to a beat selected with the Grid button. Note divisions include quarter, quarter triplet, eighth, eighth triplet, and sixteenth. A Free mode can be used to accommodate complex rhythms and odd time signatures
- Modulation section allows you to control the depth (from 0 to 500ms) and the LFO rate (from 0.0 to 20Hz) of the taps
- Tempo Controls allow you to raise or lower the BPM and corresponding quarter note value in milliseconds
- Two distinct Feedback Modes:
  Normal mode where the feedback sends each tap back into the input again and Tap Feedback mode, where the feedback has its own delay time that sets the repeat time for a rhythmic pattern.
- Feedback Gain controls how much signal is fed back into the input. In Normal mode, this effect can range from gently decaying repeats to positive gain feedback that develops into distortion. In Tap Feedback mode, a gain setting of 100% will continuously cycle the delay loop while a lower percentage will cause the loop to gradually fade out.
- Tempo Mode button changes the function of the Tap Pad between Tempo and Pattern mode which allows you to tap a rhythmic sequence to use for the delay values
- Feedback Rotation control allows you to shift the entire stereo signal ±45° without collapsing the stereo image
- Feedback EQ/Filter section offers 3 filter types: Low Pass, Low Shelf, and a unique Low Decay that gently attenuates treble frequencies not unlike the behavior of tape or analog delays, or vintage digital delays with low sampling rates.

PAZ Realtime Psychoacoustic Analyzer



PAZ (TDM bundle only) provides accurate visualization of psychoacoustic experiences and contains three modules: Frequency Analysis, Stereo Position Display, Loudness/Peak Meter. Useful for mastering and analysis, troubleshooting sound systems, room tuning and more.

Four Plug-in Modes

Each 'mode' can be used as a separate plug-in, or all three can be used in a single window

- **PAZ-Analyzer** — This is the full plug-in with all 3 measurement functions (Frequency, Stereo Position, and Level);
- **PAZ-Position** — Shows only the Stereo Position Display with the controls related to it
- **PAZ-Frequency**— This shows only the real-time frequency analysis with related controls
- **PAZ-Meters** — Shows only the meters for Left, Right, and Sum. The Sum meter can be selected for Peak or RMS display

Frequency Analysis Display

- The Frequency Analysis display can be used for Peak or RMS display of dual channel, mono analysis and total stereo energy from DC to Nyquist (the highest frequency allowed by the current sample rate)
- It employs 52 frequency bands nearly identical to the constant Q critical frequency bands of human ears. This provides the most accurate visualization of sound as we experience it
- The resolution below 250Hz can be increased to 10 Hz steps for a total of 68 bands
- Energy spectrum can be "A" or "C" weighted

Stereo Position Display

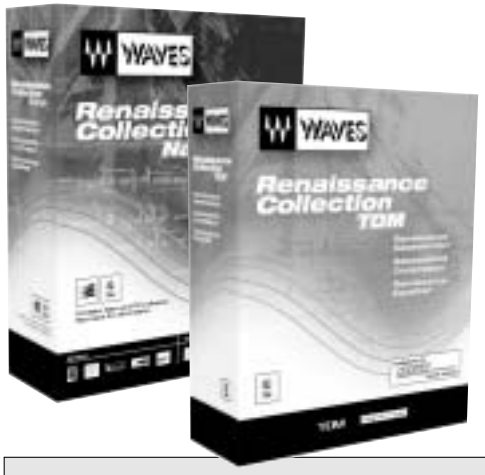
- Stereo Position Display shows the loudness energy across the Stereo stage as well as anti-phase information.

Loudness/Peak Meters

- Loudness/Peak Meters are shown against each other. Loudness is computed according to the selected weighting curve

WAVES

RENAISSANCE COLLECTION — *NATIVE or TDM*



Software Emulation of Vintage Analog Hardware with Simple Intuitive Controls

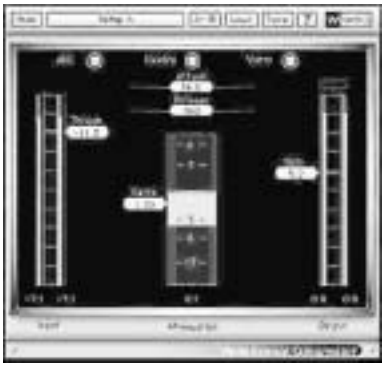
Unbeatable sound and power at an incredible price. The Renaissance Collection puts the three essential processors at your fingertips: equalization, dynamics, and reverb—all with classic interfaces. Each sports an elegant and simplified interface, and respond immediately to any parameter changes. The Native version supports RTAS, AudioSuite, VST, MAS and DirectX formats on Mac and Windows with sample rates up to 88.2/96kHz. The TDM version adds support for Digidesign's ProTools TDM format on the Mac, including 192kHz compatibility for ProTools HD.

Renaissance EQ—
Perhaps the most musical EQ plug-in ever developed, the award-winning Renaissance EQ offers two to six bands of audiophile EQ. Selectable filter types respond like classic analog/tube hardware adding air and smoothness while maintaining clarity and space.

Renaissance Compressor—
The Renaissance Compressor recreates the warm distinctive sound and simple operation of classic analog hardware. Features ARC and Opto/Electro modes with brickwall (L1 style) limiter on the output. Has an intuitive interface for comprehensive, detailed control.

Renaissance Reverberator—
Easy to use, with incredible detail and flexibility, the Renaissance Reverberator offers the classic sound of professional reverb with intuitive controls. It offers Hall, Plates, Chambers, Gates plus Reverse, Resoverb and Echoverb.

Renaissance Compressor



The Renaissance Compressor is a vintage style compressor/expander with a barebones, no nonsense interface. The five standard compressor controls — threshold, ratio, attack, release and gain are enhanced with Mode, Character and Behavior controls that let you choose between manual or auto-release functionality, analog warmth or an uncolored sound, and Electro or Opto circuit emulation. A diverse range of factory presets for use in mastering, voiceovers, vocals, drums, guitar and bass, transient restoring expanders as well as aggressive pumping effect, provide a starting off point to get you going.

- 64-bit floating point resolution dithered to 32-bit for Native applications
- 56-bit resolution dithered to 24-bit (TDM)
- Auto Release (ARC) or Manual release control. ARC dynamically chooses the optimal release value for a wide-ranging input, allowing greater compression with less artifacts.
- Compression behavior can be set to emulate either an Electro or Opto circuit for applications ranging from tracking to mastering
- Electro mode increases release time as gain reduction approaches zero. This ideal for increasing the level of sound sources for loud applications such as voiceovers and pop music
- Opto mode slows down release time as gain reduction returns to 0 and increases as gain reduction exceeds 3 dB, adding sustain to percussive tracks while taming transient peaks
- Choose between Smooth (leaves the sound uncolored) and Warm (adds low frequency harmonics as more gain reduction is applied).
- Threshold fader, adjustable in 0.1 dB steps, controls the input level above which the soft knee compression or expansion begins
- Adjust the compression and expansion ratios (1.01:1 to 50.0:1 and 0.99:1 to 0.50:1 respectively), for signal above Threshold
- Adjustable attack fader (0.5 to 5000 ms in 1/100 ms steps) to control the response time of the onset of compression or expansion
- Adjustable release fader (5 to 5000 ms in 1/100 ms steps.
- Gain fader adjusts the output gain of the compressor in 0.1dB steps, ±30 dB. Gain meter displays realtime gain reduction or increase
- Input/output Meters are logarithmic displays from 0 to -60 dBFS , with resettable peak hold values at the bottom of each meter
- L1-style limiter at the output stage is activated if the output level exceeds 0dBFS.

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ONE HOUR FREE PARKING
AT 349 W. 34th STREET (with purchase of $100 or more)

RENAISSANCE COLLECTION — *NATIVE or TDM*



Renaissance EQ

Renaissance EQ is a 2, 4 and 6-band audiophile quality EQ. The Q10 style functionality of Renaissance EQ includes separate Gain, Frequency Q and Filter Type controls for each band that can be controlled via dedicated buttons or graphically using the intuitive Frequency Response Graph. The various filter-curves, based on the theoretical designs of renowned engineers emulate the warm, sweet sounds of vintage-style EQ filters including prized Pultec EQs.

- Michael Gerzon and Baxandall Filter-curves, based on vintage analog equalizers, include Resonant shelves that emulate the simultaneous boost and cut characteristics of vintage Pultec EQs with a unique Q control that adjusts the slope of the shelf
- Stereo, dual-mono and mono operation
- Simultaneous multi-parameter control lets you make EQ adjustments while maintaining the integrity of a desired curve
- 48-bit double precision internal processing dithered to a 24-bit output (TDM), 64-bit floating point resolution dithered to a 32-bit floating point output (Native)

Each of the six EQ bands has its own Gain, Frequency, Q, In-Out and Filter Type controls

- Gain for each EQ band is adjustable ±18 dB, in 0.1 dB increments, via the Gain Value Window or by dragging the EQ curve in the frequency response graph
- Frequency control displays the center frequency for each band when a bell filter is selected or the corner frequency of the filter when in cut or shelf modes. Adjustable in 1/16th tone increments (96 steps per octave)
- Q control works differently for each filter type. When using a Bell filter, Q adjusts the width of the filter in relation to the center frequency.
- When using a Shelf filter - Q controls the slope of the "side"of the shelf and the resonant dips and peaks
- When the Cut filter is used, available for bands 1 or 6 only, Q controls the slope of the cut filter (from about 10dB/oct to 18dB/oct)
- Individual EQ bands can be bypassed
- There are different sets of filters available depending on the band — bands 1 and 6 have Cut, Bell, and shelf filters, bands 2 and 3 have Bell and Low Shelf filters, bands 4 and 5 have Bell and High shelf filters

Renaissance Reverberator

Renaissance Reverberator is a classic sounding reverb processor that excels in any level of music production. Designed to be straightforward and musical, it provides twelve selectable reverb types, controllable with an uncluttered yet highly functional graphic interface.



- Nine sliders control the properties and levels of of the Early Reflection and Reverb
- Installs two component plug-ins in your host application — the full Reverb, and a Reverb Tail only plug-in without Early Reflections
- Three intuitive graphic displays — Damping Graph with moveable High and Low Freq. markers, Early Reflection and Reverb Tail Response Time Graph and Reverb EQ Graph with moveable High and Low Freq. markers

Reverb Controls and Faders

- Reverb types control the sound characteristics of the reverb and early reflection: Hall 1/2, Room, Chamber, Church, Plate 1/2, Reverse, Gated, Non-Linear, EchoVerb, and ResoVerb
- Six variations of the amount of correlation between the left and right channels within the early reflections
- Diffusion fader serves as a balance control between the direct and early reflection signals that feed the reverb tail.
- ±160.0 milliseconds predelay range. Positive delay times offset the signal sent to the reverb tail generator (negative delay the direct signal
- Time fader sets the decay time of the reverb tail from 0.1 to 20.0 seconds
- Adjust size related parameters corresponding to the current Reverb Type such as early reflection spacing or reverb tail dimension
- Control the decay envelope of the reverb tail from non-linear gating to a more natural linear slopes.
- Early Reflections and reverb tail's output level have a range of 0.0 to Off
- The Wet/Dry fader controls the balance, from 0 to 100%, between the dry and wet signals

Pre Early Reflection and Reverb EQ Controls

- Low Frequency Shelving EQ has a frequency range of 16 to 1600 Hz with a variable gain of -24.0 dB (low cut filter) to +12 dB
- High Frequency Shelving EQ has a frequency range of 1k to 21 kHz with a variable gain of -24.0 dB (high cut filter) to +12 dB

Damping Controls

- Control the "knee" of where the LF damping begins (16 to 1600 Hz) and where the HF damping begins (1000 to 2100 Hz
- Control the progressive frequency response of the reverb's low or high range with a multiplication factor of 0.10 to 2.00

WAVES

PRO-FX PLUS — *NATIVE or TDM*



The Ultimate Sound Design Tool Kit

Retaining the sonic excellence that Waves plug-ins are known for, Pro-FX are potent and innovative implementations of classic (and new) effects including emulation of vintage analog devices and effects possible only in the digital domain.

The Pro-FX Bundle includes the UltraPitch, a 6 voice formant-corrected pitch shifter; the SuperTap, a 6 voice multi-tap delay for effortless rhythmic repeats and doubling; the MetaFlanger for vintage tape-flanging and phase emulation; and MondoMod, a unique combination of AM, FM and Rotation (panning) modulations. Each plug-in uses Waves' proprietary digital signal processing algorithms, with 56-bit audio math on TDM systems.

UltraPitch -
Formant-corrected pitch shifter with 6-voices that creates huge and thick stereo chorusing, doubling, parallel harmonies and vocal slap/spread effects. It also creates excellent gender-bending effects. Independent stereo panning and delay for each voice. An animator delay-randomizer sets the pitchshift by musical intervals (with 5 cent resolution). Includes manual formant mapping as well as presets.

Doppler -
Software processor developed for post, film sound and game designers.

MetaFlanger-
Vintage tape-flanging, phaser-emulation, and special effects with true dual-delay flanging sounds. Wet signal include filters so you can flange or phase just part of the signal. Presets of vintage emulations (Mutron, MXR, Itchycoo Park) and more.

MondoMod-
Combines AM (level), FM (frequency), and stereo rotation (panning) into a single modulator, with built-in LFO and the ability to modulate from a sidechain source. It is able to create bizarre and unexpected effects, both destructive and subtle.

Enigma -
Combination of a new-technology "notchbox", a feedback circuit with filtering, and a modulator, but as the name implies, has plenty of mysterious and creative corners for designers to explore.

SuperTap-
Six taps of mono or true stereo delay (up to 6 seconds) with global LFO modulation, 2 feedback modes, Q10-style filtering for each tap as well as rotation (stereo panning). Delays are adjustable in milliseconds and note values and you can tap out delay times or patterns using the Tap Pad.

UltraPitch

UltraPitch provides up to 6 voices of formant corrected pitch shifting each with independent level, pan and delay control for each voice. It allows you to raise or lower the pitch, of each voice, up or down one octave by semitone or with 5 cent resolution. Variable formant-corrected pitch shifting prevents chipmunk effects and is excellent for gender bending. With fast pitch detection and completely variable stereo panning and rotation for each output, UltraPitch can create huge and thick stereo chorusing, doubling, parallel harmonies, slapback effects and more. Includes three separate component plug-ins; single voice, 3-voice harmony maker and 6-voice harmony maker, that can be used in either mono or stereo.



PRO-FX PLUS — *NATIVE or TDM*

Pitch Detection Display & Controls

- Pitch detector graph displays the current pitch of the direct monophonic signal in real time. A keyboard display on the right hand side of the pitch detection graph displays the notes of the pitch being detected
- Adjust your monitor's scrolling speed to clearly see the pitch-detection and adjust the Min and Max-pitch thresholds accordingly (these limiters adjust the upper and lower frequency limits of pitch detection).
- Set the optimum pitch detection type for the input signal — Default, Voice Steady, Voice Jumpy, Voice Opera, Music Steady, Clear tones or Bass

Pitch Character Display & Controls

- Voice character graph displays a color-coded Marker for each active voice indicating its current pitch and formant settings. Markers can be dragged up and down to raise or lower its pitch, and from side-to-side to adjust its formant
- Three different ways to raise/lower the pitch for each component up to one octave
- Eight Link presets (Free, Voice, Song, Brass, Woods, Strings, Linear and Inverted), designed so that as you change pitch, formant-correction will follow the ideal curve
- Formant-correction control adjusts the resonant frequencies for each voice ±12

Mix Display & Controls

- Direct Level can be set (0.0 to -24.0 dB)
- Mix graph displays each active voice in the stereo field via color coded markers.
- Adjust Gain range (0.0 to -24.0 dB) and Delay time (0 to 100ms) for each voice
- The Pan control allows adjustment of each voice ±45º within the stereo sound field
- Live mode (TDM version only) is delay setting that syncs the direct voice with the input sound, permitting UltraPitch to work in real time situations such as a live show
- Animate feature adds a random delay (0 to 100ms) between voices, adding more realism to the overall blend.

MetaFlanger



MetaFlanger is a plug-in for creating unlimited flanging, phaser and chorusing effects including classic tape flanging effects and other mutations of vintage modulation effects. It can emulate traditional analog flangers, fill out a simple sound, create intricate harmonic textures and even generate small rough reverbs and effects. The Filter section allows you to affect only the upper or lower frequency ranges, while phase inversion allows even greater, more dynamic effects control. Has an extensive library of factory presets.

Mix, Feedback & Filter Controls

- Mix control sets the Wet/Dry balance with a range from 0% (Dry) to 100% (Wet)
- Invert the phase of the wet signal creating an increased boost or cancellation of frequencies depending on the flanger's corresponding depth and delay settings
- The Feedback control, adjustable from 0 to 100% sets the amount of wet signal is fedback to the input. A higher feedback percentage increases the intensity of the effect
- Inv Feedback button adds even more tension and uniqueness to an effect
- Apply flanging effects to selected high or low frequencies
- Toggle between a low or high pass filter
- Sets the corner frequency for the selected filter within a range of 0.1 to 20.0 kHz.

Delay, Rate and Depth Controls

- Delay control is settable from 0 to 50.0ms in 0.1ms increments. The amount of delay dictates the type of modulation effect achieved:
  — 0.1 -0.5ms is ideal for phaser type effects
  — 0.5 -3ms for flanging
  — 3 -50ms for chorusing effects.
- The Delay control combines with Rate and Depth to alter the sweep of the modulator.
- Adjust the LFO waveform frequency from 0 to 20Hz in 0.1Hz steps. The LFO rate controls the speed at which the delay time will vary.
- Vary the delay time of the wet signal, from 0 to 100%, creating a slight pitch shift
- Control Rate and Depth simultaneously, maintaining a constant ratio between them
- Tape button provides true tape-flanging emulation by causing the dry audio to be delayed equal to the wet delay.

Waveform, Modulation Display and Stereo Controls

- Waveform button selects either a Sine wave with a continuous variation from the highest to lowest pitch or Triangle LFO wave modulation which only has two pitches
- Modulation display shows you graphically how the LFO parameters dynamically change when you alter the Rate and Depth settings
- The Stop button "freezes "the modulator wherever it is when you click it
- Stereo control sets the phase between left and right LFO's between 0 to 180º. When set to 180º, the Left side is flanging down while the Right side is flanging up
- Gain fader (±12.0dB) in 0.1 dB increments
- A Numerical Peak-level display shows the exact level to one decimal place

PRO-FX PLUS — *NATIVE or TDM continued*



Doppler

Doppler is the effect heard when a sound source, such as a car or train, passes by, and the pitch changes. Pitch sounds higher as the sound approaches, and sounds lower as it goes away. Easy to operate, Waves' Doppler processor lets you create realistic and unusual Doppler effects with full control over pitch, path curve and high frequency damping, gain, reverb and start/stop points. Waves' Doppler is based "on reality"—some controls behave according to the laws of physics (air damping, gain hanging with distance). However you can set the values of Doppler to work well outside those laws; it is said there is reality, and there is "Hollywood", when it comes to effects and the appropriate perception of them. Sometimes what an audience would expect is not based on the laws of physics at all!

SuperTap Multi-Tap Delay

SuperTap is a six-tap delay with up to six seconds of mono or true stereo delay. There's a dedicated Eq/Filter section for each tap derived from the acclaimed Q10 as well as Global LFO and a feedback section capable of overmodulation and positive gain feedback loops capable of some serious distortion. SuperTap can produce a wide range of creative rhythm effects and loops and includes 19 factory presets covering a broad range of multi-tap delay effects, including analog and tape delays, slapback echo, chorusing and more. To conserve processing power, SuperTap allows you to choose between 2 or 6 tap plug-ins, in mono or stereo, with or without modulation.



Each Tap Features

- Controls for each tap are arranged in a row from left to right — On/Off, Gain, Rotate, Tap Marker, and Eq/Filter section
- Gain is adjustable in 0.1 dB steps
- Rotate controls the tap's stereo position ±45º
- Each tap has its own slider located in the Tap Delay Grid that is adjusted by dragging the slider from left to right. Each slider shows a value either in BPM or milliseconds (ms)
- Each tap has its own EQ/filter section derived from the Q10 plug-in. Each filter has a selectable frequency range of 100 Hz to 20kHz and six filter curves — Bell (wide) Bell (narrow), Hi and Low Shelf, High and Low Pass. Gain control lets you boost or cut the filtered bands without affecting the pass bands
- Pan Graph shows the stereo location of each tap with a marker. Tap markers can be dragged from left to right to adjust Rotation (±45°) and up and down to adjust Gain (from 0.0 to -24dB).

Global Controls

- Grid Mode button changes between BPM (adjustable from 40 to 1200bpm) or ms
- Snap-to-grid feature lets you quantize each tap to a beat selected with the Grid button. Note divisions include quarter, quarter triplet, eighth, eighth triplet, and sixteenth. A Free mode can be used to accommodate complex rhythms and odd time signatures
- Tempo Controls allow you to raise or lower the BPM and corresponding quarter note value in milliseconds
- Tempo Mode button changes the function of the Tap Pad between Tempo and Pattern mode which allows you to tap a rhythmic sequence to use for the delay values
- Two Feedback Modes: Normal mode where the feedback sends each tap back into the input again and Tap Feedback mode, where the feedback has its own delay time that sets the repeat time for a rhythmic pattern
- Feedback Gain controls how much signal is fed back into the input. In Normal mode, this effect can range from gently decaying repeats to positive gain feedback that develops into distortion. In Tap Feedback mode, a gain setting of 100% will continuously cycle the delay loop while a lower percentage will cause the loop to gradually fade out.
- Feedback Rotation control allows you to shift the entire stereo signal ±45° without collapsing the stereo image
- Feedback EQ/Filter section offers 3 filter types — Low Pass, Low Shelf, and a unique Low Decay that gently attenuates treble frequencies not unlike the behavior of tape or analog delays, or vintage digital delays with low sampling rates
- Modulation section allows you to control the depth (from 0 to 500ms) and the LFO rate (from 0.0 to 20Hz) of the taps

PRO-FX PLUS — *NATIVE or TDM continued*

MondoMod

MondoMod is software plug-in for pro-audio processing applications. It includes an AM (Amplitude Modulation) section for creating tremolo effects, an FM (Frequency Modulation) section for creating vibrato effects, and a Rotation Section for creating panning effects. The rate of change for all three effects are controlled by an LFO (Low Frequency Oscillator) with 5 different waveforms and a variable tempo adjustable by Hz (frequency) and BPM (Beats per Minute). Simple On/Off buttons allow you to use the three effects together or individually.



Tempo and Waveform Section

- **Tempo** adjusts the rate of an LFO (Low Frequency Oscillator) in cycles per second from 0.01 to 60.0Hz and beats per minute from 0.6 to 3600 BPM.
- The **x-factor control** multiplies the tempo by 0.5, 1, 2, 4, 10 or 100.
- Waveforms available for the LFO include — Square, Saw, Saw up, Triangle and Sine.

AM (Amplitude Modulation) Section

- Control the amount of AM from 0 to 100%. A setting of 50% means that the amplitude will drop by half and then return to 0 dB at a rate set by the LFO in the Tempo section.

FM (Frequency Modulation) Section

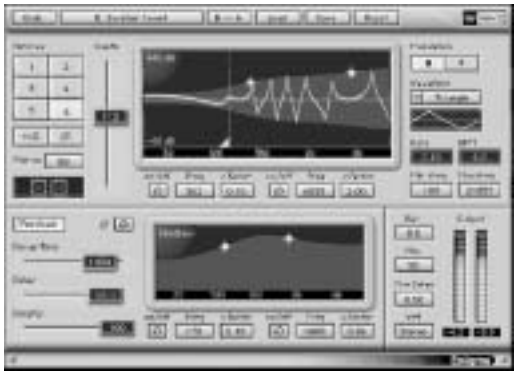
- Control the amount of FM from 0 to 100%.
- Adjust the phase relationship of the FM to Amplitude and Rotation Modulation (selectable from 0, 90, 180 or 270° out-of-phase.

Rotation Section

- Set the amount of rotation to the left and right channels. Negative settings place more audio to the right, positive settings place more to the left channel. Set the modulator's left and right sweep range, or width, between 0 and 360º.

Output Section

- Adjustable output Gain (±12.0 dB in 0.1dB increments). Dry/Wet balance can be adjusted from 0% (Dry) to 100% (Wet).



Enigma

Four processing blocks combine to create totally unique effects for vocal and instrumental leads, re-mixing/re-processing, sound design, subtle ghosting of effects, treatment of reverb sends, and more,

- *Notch Section is the "enigma processor" with filtering option.*
- *Modulator Section modulates the notch processor*
- *Feedback with filtering option*
- *Global Section with Gain, Mix,Wet and Predelay controls*

Notch Section -the Enigma processor

- Choose from 2 to 12 (1 to 6 pairs) of notches/frequency sweeps
- Plus Half ("+1/2 ") button) adds "half a pair" of notches at the high end.
- Depth controls the depth of the notches in the Enigma processor. When set to zero, the Feedback section is the main area of control.
- Phase button reverses the polarity (up/down orientation) of the notches.
- Control both stereo and tonal aspects of the processor.

Modulation Section

- **Modulation On/Off button** can be used to "freeze" the notches at a certain color and to automate the start and stop of the modulator.
- Waveforms include sawtooth up down, triangle, sine, square (50% pulse).
- Adjust modulation in seconds or beats per minute.
- Set the boundaries of the frequency range to be swept.

Feedback Section

- The **Decay Time slider** responds like a feedback control with longer delay times and a reverb decay control with shorter delay times.
- The **Feedback Phase button** reverses the phase of the feedback.
- Control the density and stereo spread of the individual delays.

Global Section

- **Gain** controls the output level indicated by the peak reading meters.
- Control the Wet/Dry balance between the processed and input signals.
- **Pre-delay** delays the wet signal before mixing it with the dry.
- The **Wet control** offers 3 modes that control the output signal — mono, stereo, and Mono+Phase Reverse which takes reverses the phase of the left channel and then sums it with the right channel, producing a stronger notching effect while phase cancelling the original input signal.

WAVES

GOLD BUNDLE — *NATIVE or TDM*



Professional Tools for Production, Mastering, Tracking and Multimedia Audio

Whether you've exhausted your old plug-ins or have simply been waiting for the right all-in-one audio toolbox to come along, the Waves Gold Bundle is simply everything you need, with several daily tools, sweetening and mastering processors, and sound design mindbenders. A near-flawless, must-have plug-in suite, Waves Gold includes a total of 19 plug-ins — enough tools to round out any plug-ins menu — or replace it all together. From the original Q10 and L1 to the Renaissance series, over 300 setups and processors provide exacting control of the highest-resolution tools, for musicians, mastering, remix, restoration, multimedia, film, web, games, everything audio. In fact, the Gold Bundle can handle hundreds of jobs—yet the whole collection falls into tight formation, thanks to its consistent interface design and expansive compatibility.

Gold Bundle (Native or TDM) includes
C4 Multiband Parametric Processor, MaxxBass and AudioTrack

Native Power Pack or TDM Bundle—L1 Ultramaximizer, Q10 Paragraphic EQ, SuperTap 2-tap, DeEsser, TrueVerb, S1 Imager, C1 Compander

Pro-FX Plus—Enigma, Doppler, Ultrapitch, MetaFlanger, SuperTap, MondoMod

Renaissance Collection — Renaissance EQ, Renaissance Compressor, Renaissance Reverberator

C4 Multiband Processor

The C4 Multiband Parametric Processor harnesses the compression design from the award-winning Renaissance Compressor including the ARC Electro and Opto bands.
Key features of the C4 include transparent cross-over design with global adjustable Q; true parametric control including independent threshold, range, gain, attack, release and bandwidth per band; flexible design allows compression; expansion or EQ independent of the functions of the other bands; adjustable global knee control; bit precision dithered to a 24 bit output.



- Pure innovation with 4-band up and down expansion, limiting, compression, plus dynamic and standard EQ.
- The C4 is simply the most flexible, musical device ever created. Compress the bass, expand the mid, De-ess the highs, EQ the air, all with complete understanding and control. No tiny LCD screens or confusing menus. Pure multiband power. Instant gratification. Each component is crafted to provide exquisite clarity and control.
- Each C4 band includes threshold, range, attack, release, and gain. The crossover has independent bandwidth control and global Q adjustment. This unmatched flexibility allows highly different functions in each band simultaneously. For example, you can compress the lows, expand the low-mids, add fixed gain to the upper mids (with no dynamic change), and limit the highs, all at once. The combinations are staggering and give you the ultimate in tonal control.
- The C4 brings a fusion between gain reduction metering and EQ. You can think of the C4 as a dynamic EQ, and therefore, Waves' displays the output as an EQ curve that moves, called the DynamicLine. Of course, as with all Waves processors, you can select-and drag any graphical representation, or type in numbers, tweak with the mouse, etc. And as always you can select multiple controls at the same time for great ease in adjustments.

GOLD BUNDLE — *NATIVE or TDM*



MaxxBass

MaxxBass provides a virtual subwoofer by extending the perceived bass response on any speakers. Whether for mastering enhancements or site-specific mixing, your ears will perceive an amazing low end.

Patented technology, MaxxBass takes your bass to the maximum by adding a series of harmonics to the signal. These harmonics stimulate a psychoacoustic bass-enhancing effect – a phenomenon that small speakers rely on to play audible bass. MaxxBass simply maximizes this known acoustical phenomenon. Thus bass frequencies are greatly enhanced without traditional EQ or bass compression. Taken to the extreme, you can actually remove the original bass signal completely and the ear will still "capture" the fundamental from the added harmonics. Employing dynamic graphics and a simple user interface, MaxxBass allows you to creatively add specific harmonics to bass signals.

The MaxxBass Graph

- **Frequency slider** controls the crossover point between the original bass and the added harmonics with a range of 32 to 256 Hz. Frequencies below this point have harmonics created for them; above it, are passed to the output.

Output Section

- **Harmonics fader** controls the level of the generated harmonics in the output mix from +12dB to -infinity.
- **Original Bass fader** controls the level of the low-passed bass signal in the output mix from +12dB to -infinity.
- **Input level fader** can be used to attenuate hot signals indicated by the peak reading output meters
- **Monitor buttons** toggle between Audio (mixed output), Harmonics (MaxxBass Signal), and Original Bass.

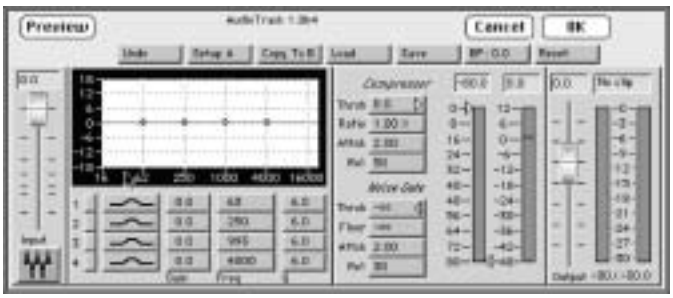
Harmonics Section

- **Highpass filter** toggles between a 24dB/oct filter at 16Hz for removing DC and very low frequency components), A 12dB/oct filter at the selected Frequency on the graph, and a 24dB/oct filter at the selected Frequency on the graph.
- **Decay control** adjusts the progressive rate of decay in level for each succeeding harmonic in the series.

Dynamics Section

- **Ratio control** applies upward compression to the generated harmonics between 1.00:1 to 4.00:1 in 0.01 increments. Higher ratios limit the dynamic range of the harmonics.
- **Response controls** adjust attack and release of the harmonic generation between 10 and 30 ms in 0.1ms increments.

AudioTrack Channel Insert

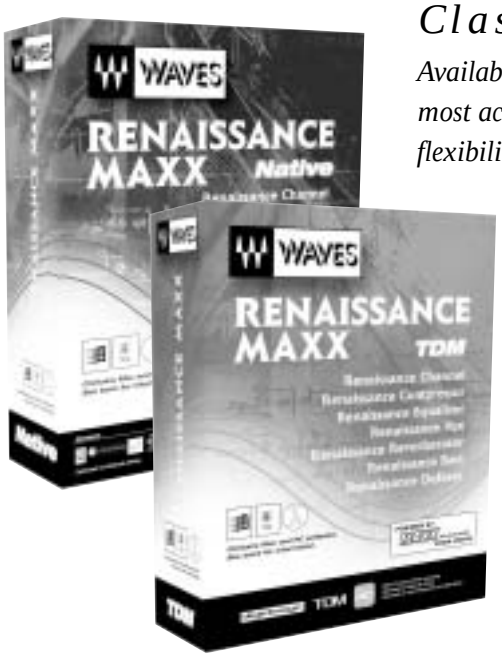


AudioTrack is a channel insert with 4-band, fully parametric EQ, compressor, and gate, all in one space-saving window, 88.2/96kHz-ready for many native platforms. Unbelievably efficient on any system, with undeniable quality, it is ideal for multimedia producers, voiceover, project studios, and more. The EQ is a true fully parametric 4-band EQ, with bell, shelf, and highpass/lowpass filters, all taken from Wave's Q10 design. The compressor includes expander ratios, plus a "one-hand" Autogain feature. The noise-gate removes background noise from voiceovers and more.

- Single window interface
- Volume and Gain Reduction meters
- Setup library with: Effect, Multimedia, Musical Style, RealAudio, and ShockWave audio setups. (Mac version only)
- Sample-Accurate peak hold and clip meters
- Single pass: Equalization, Compression / Expansion and Gate
- Instantaneous A/B comparisons between on-line settings
- Full compatibility between native and DSP hardware processing
- ShockWave Audio Preprocessing presets (PC version)

WAVES

RENAISSANCE MAXX — *NATIVE or TDM*



Classic Sound — Complete Control

Available for Native and TDM platforms, Renaissance Maxx offers seven of Waves' most acclaimed audio processors in one bundle to provide the MAXXimum in flexibility, efficiency and intuitive control.

Renaissance Maxx features vintage-modeled equalization, dynamics, and reverb plus a powerful vocal processor, a new de-esser, Renaissance Bass, and the new plug-in that tips the scale —Renaissance Channel— the ultimate channel insert processor.

Renaissance VOX combines compressor, limiter and gate functions to bring vocals and solo instruments to the front of the mix while pushing out unwanted noise. Renaissance Bass refines Waves' popular MaxxBass technology to allow bass frequency extension through speakers with limited low frequency response. Renaissance DeEsser is the next generation DeEsser, and Renaissance Channel is a true channel insert with EQ, comp/limiter, gating, and more incorporated from other award-winning Renaissance plug-ins.

Renaissance Maxx (Native of TDM) includes
Renaissance Bass, Renaissance VOX, Renaissance DeEsser, Renaissance Channel

Renaissance EQ—

Perhaps the most musical EQ plug-in ever developed, the award-winning Renaissance EQ offers two to six bands of audiophile EQ. Selectable filter types respond like classic analog/tube hardware adding air and smoothness while maintaining clarity and space.

Renaissance Compressor—

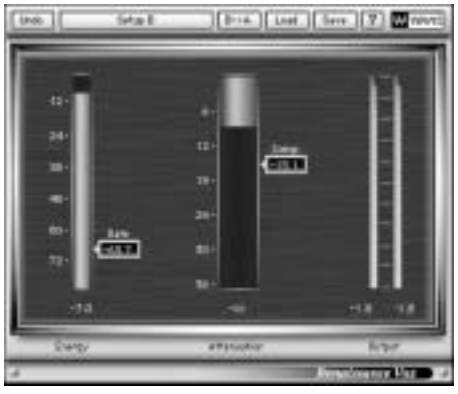
The Renaissance Compressor recreates the warm distinctive sound and simple operation of classic analog hardware. Features ARC and Opto/Electro modes with brickwall (L1 style) limiter on the output. Has an intuitive interface for comprehensive, detailed control.

Renaissance Reverberator—

Easy to use, with incredible detail and flexibility, the Renaissance Reverberator offers the classic sound of professional reverb with intuitive controls. It offers Hall, Plates, Chambers, Gates plus Reverse, Resoverb and Echoverb.

Renaissance VOX

This may be one of the ultimate "one-knob" compressor limiters in pro audio history. Actually it has two controls because it also has a gate. Renaissance VOX is optimized for vocal tracks, featuring a simple interface with only two controls and two meters. The elegant display controls a compressor, gate, and precision limiter with automatic gain staging for clip-free maximization of vocals. The Energy control adjusts the threshold of the multi-stage downward expander, and the Compression control adjusts the amount of compression and limiting while automatically providing output gain boost. This greatly simplifies voiceover or vocal track production by giving the user a single-control adjustment for the most-needed compression and maximization of such tracks.



RENAISSANCE MAXX — *NATIVE or TDM*

Renaissance DeEsser

Renaissance DeEsser relies on technologies used in Waves C4, DeEsser and Renaissance VOX plus it has several proprietary features that make it better then other DeEssers including Waves own DeEsser. Here are a few main advantages:

- Threshold is not the normal "absolute value" threshold that you usually have. Dynamically adapting to the input signal in order to provide a more unified DeEssing effect yielding more natural sounding results.
- Specify the maximum desired gain reduction for better control of the DeEssing effect.
- Phase compensated crossover – this is very important in order to avoid undesirable coloration to the sound and to eliminate phase modulation that would other wise occur as an artifact of the amplitude modulation applied by the compressor. Same type of crossover used in the Waves C1, and C4.



- As the DeEsser works only on part of the spectral range of the input, the graphic user interface incorporates a graph that lets you see what you hear.



Renaissance Bass

Waves developed the MaxxBass technology to allow audio engineers to selectively add harmonics which significantly enhance the bass perception by the listener, and most importantly, beyond the range of the speaker's conventional frequency response. Renaissance Bass takes the MaxxBass algorithm to greater effectiveness and simpler operation. The process has been refined and is even more effective than the original processor. In addition, a simplified user interface helps achieve top-quality sound with quick adjustment, whether in post, mixing, mastering, or multimedia.

Renaissance Channel

Renaissance Channel is a complete channel processor with true double precision operation for maximum fidelity and artifact-free audio. It includes vintage modeled EQ and compression inherited from the critically acclaimed Renaissance EQ and Compressor, selectable EQ/Dynamics ordering, and independent sidechain configuration of routing and filtering for compression and gating. In addition, it includes overload protection and stereo rotation, taking the concept of channel insert to the limit.

The EQ features consistent filter response over the entire audio range, with two user-selectable compression curves – one taken from the Renaissance Compressor, and the other from the Renaissance VOX processor. With true double precision processing, the internal processing bus always has the

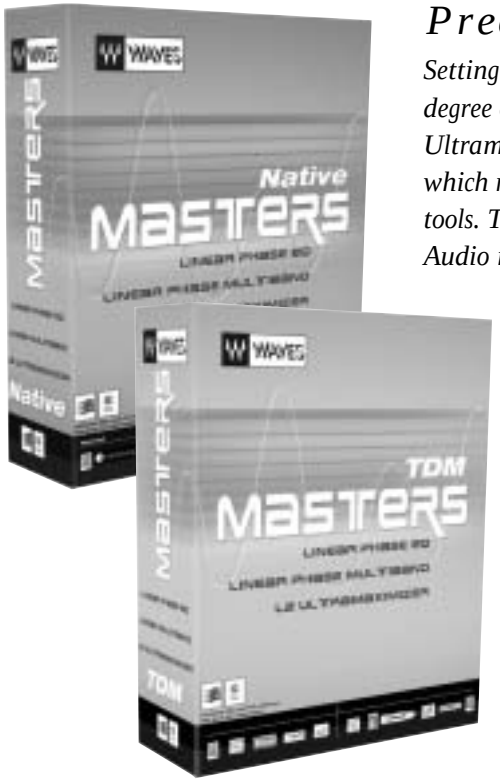
definition required to handle any input signal without artifacts or internal distortion, while overload protection ensures there is no output clipping. The entire stereo stage can be rotated, controlling the level balance without affecting center channel sound. In addition, sidechain routing and filtering can be configured independently for the gating and compression components of the processor.



- Selectable EQ/Dynamics signal flow
- Overload protection - no output clipping
- EQ with consistent filter response over the whole frequency range
- Setting of sidechain routing and filtering for gating and compression

WAVES

MASTERS— *NATIVE or TDM*



Precision Mastering Tools

Setting the standard for mastering applications with higher sonic capability plus a greater degree of functionality, the Masters bundle includes the award-winning L2 Ultramaximizer as well as innovative linear phase equalization and multiband processing which result in absolutely no phase distortion created by previous software and hardware tools. The sound is more transparent with better musical balance.

Audio mastering professionals are continually looking for methods to deliver more pristine audio quality in their recordings. In direct response to their demands to eliminate phase distortions from their work, Waves has developed the Linear Phase Equalizer and Linear Phase Multiband.

The Linear Phase Equalizer utilizes an innovative FIR (Finite Impulse Response) filter implementation that introduces no phase distortions. This innovative design also overcomes the variable latency of traditional IIR (Infinite Impulse Response) filter implementation that add phase distortions, while offering a user-friendly tool with a user interface similar to Waves popular Renaissance Equalizer.

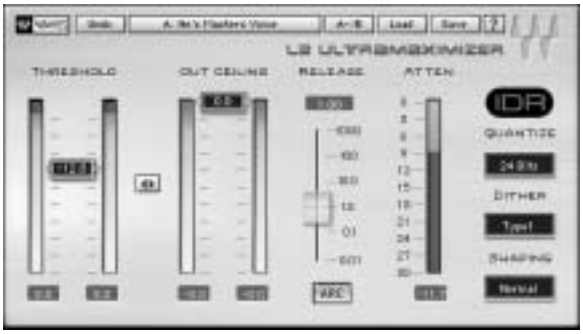
The Linear Phase Multiband supports independent gain and dynamics (compression, EQ, and limiting) on five user defined frequency ranges, while adding linear phase crossovers. The Linear Phase Multiband extends Waves award-winning C4 Multiband Parametric Processor to an even higher quality standard for purist mastering on full range mixes.

In addition, Waves award winning limiter, the L2 Ultramaximizer is included in the Masters bundle. The L2 adds ARC (Automatic Release Control) to provide extra gain during mixing.

Linear Phase Equalizer

Innovative phase linear filters eliminate your phase distortions to provide you more transparent sound that better preserves the musical balance. Linear Phase EQ includes a broadband component with 5 general bands and one LF band and a more surgical LF component. Uses the familiar Q10 and Renaissance EQ type interface. Hear what you have been missing.





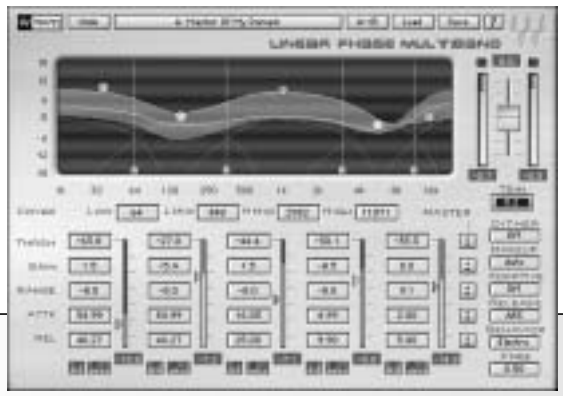
L2 Ultramaximizer

The L2 is the next step in peak limiting with ARC (Automatic Gain Control) to add the extra gain you need to master your projects. IDR and ninth order filters maintain the highest quality. Designed exclusively for mastering.

MASTERS— *NATIVE or TDM*

Linear Phase Multiband

Linear Phase Multiband supports independent gain and dynamics (compression, EQ and limiting) on five bands with linear phase crossovers. It includes Adaptive Thresholds, Automatic Makeup and Gain Trimm It uses the familiar C4 type interface and is ideal for purist mastering on full range mixes.



PLATINUM BUNDLE — *NATIVE or TDM*



The most complete set of audio processors ever offered

The Platinum bundle offers the highest-quality, most comprehensive set of audio processors to provide professionals with everything they need in a single cost-effective bundle. For those searching for a larger palette of creative options, the Platinum bundle is the perfect choice.

Containing 26 premium audio processing tools, the Platinum bundle includes all 19 processors in the award-winning Gold bundle, plus the three processors from the Masters bundle, and four additional processors from the Renaissance Maxx bundle.

The Native version supports RTAS, AudioSuite, VST, MAS and DirectX formats (Macintosh and Windows). The TDM version adds support for Digidesign's Pro Tools TDM format on the Macintosh.

Platinum Bundle (Native or TDM) includes:

Native Power Pack or TDM Bundle—
L1 Ultramaximizer, Q10 Paragraphic EQ, SuperTap 2-tap, DeEsser, TrueVerb, S1 Imager, C1 Compander

Pro-FX Plus—
Enigma, Doppler, Ultrapitch, MetaFlanger, SuperTap, MondoMod

From the Gold Collection —
C4 Multiband Processor, AudioTrack, MaxxBass

Masters Bundle—
L2 Ultramaximizer, Linear Phase Equalizer, Linear Phase Multiband

Renaissance Maxx —
Renaissance EQ, Renaissance Compressor, Renaissance Reverberato, Renaissance VOX, Rennaisance Bass, Renaissance DeEsser, Renaissance Channel

WAVES

RESTORATION — *NATIVE or TDM*

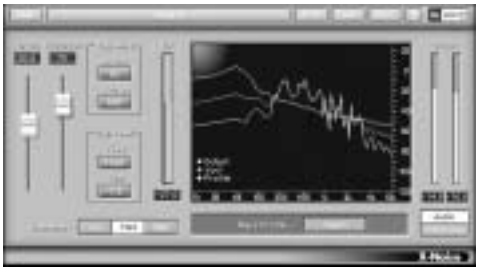


Audio Restoration and Noise Reduction

Removal of unwanted background noise and disturbing signal transients without affecting the original sound has always been a major challenge in audio postproduction, remastering, broadcasting, video, film and forensics. Waves' Restoration bundle has addressed the increasing demand for real-time noise and transient elimination tools. It establishes a new standard for quality and simplicity in software based noise reduction and sound restoration tools for post-production, remastering, and forensics applications. Fast and incredibly simple to use, it provides feedback through audio output, audio difference and intuitive graphics. Quality is comparable to reference hardware costing up to 10 times more.

The Restoration bundle includes four plug-ins: X-Noise, X-Click, X-Crackle, and X-Hum. X-Noise intelligently learns from a section of noise, and then applies a broadband noise reduction to eliminate background noise from any source. X-Click effectively removes clicks from 78's or vinyl records, as well as spikes arising from digital switching or crosstalk. X-Crackle is the second stage in the restoration of old records eliminating crackles and surface noise left after the X-Click process. X-Hum attenuates steady pitched low frequency disturbances, such as ground loop hum and its harmonics. It is also intended to reduce rumbles, microphone pops, and DC-offset.

X-Noise



A powerhouse for cleaning practically all types of noise, X-Noise intelligently learns from a section of noise, and then applies a broadband noise reduction to eliminate background noise from any source. Primary controls are similar to classic dynamic processors. Eliminates learning curve. Even stores noise profiles for future use.

X-Click



Offers high quality click removal. X-Click eliminates analog or digital clicks with the provided presets. Effectively remove clicks from 78's or vinyl records, as well as spikes arising from digital switching or crosstalk. Operates fully automated or manual click removal.

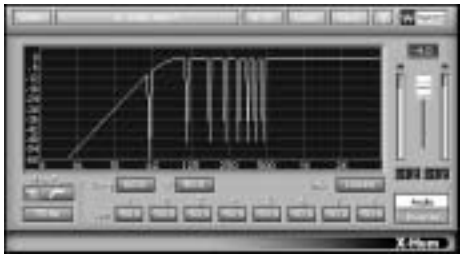
X-Crackle

X-Crackle is the second stage in record restoration eliminating crackles and surface noise from old vinyl or magnetic tapes. Simple and effective.



X-Hum

X-Hum is perfect for high quality hum removal. Eight harmonically linked notch filters attenuate ground loop hum by up to -60dB with very narrow Q. Removes 60 Hz power in a snap. High pass filter options remove rumble, mic pop and DC-offset.



360° SURROUND TOOL KIT



The 360° Surround Toolkit offers a complete set of tools for surround audio production to industry standards. Available for Pro Tools|HD and MIX systems on the Mac, the 360° Surround Toolkit consists of seven surround tools for the most impressive surround localization, spatialization, and envelopment, with enhanced panning, reverb, and dynamics:

*The **Surround Manager** allows the calibration of a studio setup to all industry standard surround release formats,and includes flexible bass management, while the **Surround Reverb** includes six channels of completely de-correlated reverberation with special front and rear surround control.*

*The **Surround Imager** adds distance panning, generating early reflections and shuffling (frequency-dependent localization) for greater LF spatialization.*

*The Surround Limiter is a 6-channel implementation of the acclaimed L2, and the **Surround Compressor** provides dynamic compression with flexible channel coupling and grouping for surround applications.*

*Finally, **Surround Mixdown** allows the derivation of Quad, LCR, Stereo and Mono mixes directly from the surround mix, greatly adding to studio productivity and compatibility between different format mixes of the same material; and the Surround Panner provides rotation and width for any input.*

BROADCAST & PRODUCTION BUNDLE — *NATIVE or TDM*

Complete production solution covering all your audio signal processing needs

Featuring Waves critically acclaimed Renaissance Maxx, Masters, and Restoration processors, this bundle includes limiters, compressors, EQ's, filters, noise removal, reverb, bass enhancement, de-essing & voice processing. You will be able to clean up audio, get killer voice overs, easily create both common and signature audio effects, fatten the sound, have the best level control available, and create superior feeds & mixes. These are the world-class standard processors used in Hollywood and in audio and music production the world over.



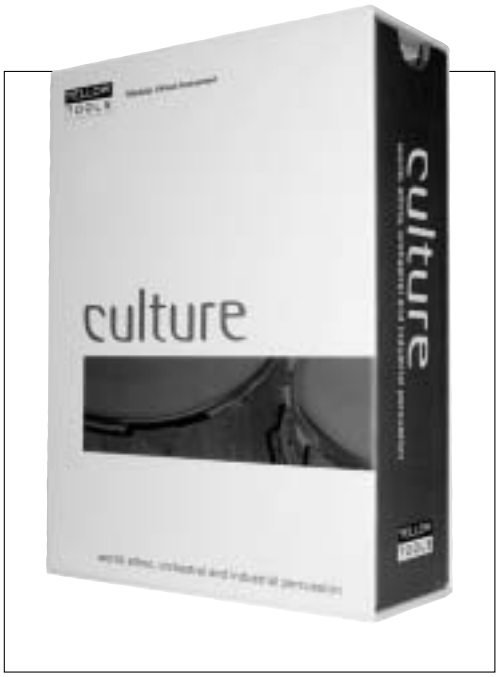
Broadcast & Production Bundle (Native or TDM) includes:

Renaissance Maxx—
Renaissance EQ, Renaissance Compressor, Renaissance Reverberator, Renaissance Bass, Renaissance VOX, Renaissance De-Esser, and Renaissance Channel

Restoration —
X-Noise, X-Click, X-Hum, and X-Crackle,

Masters Bundle—
L2 Ultramaximizer, Linear Phase EQ, Linear Phase Multiband

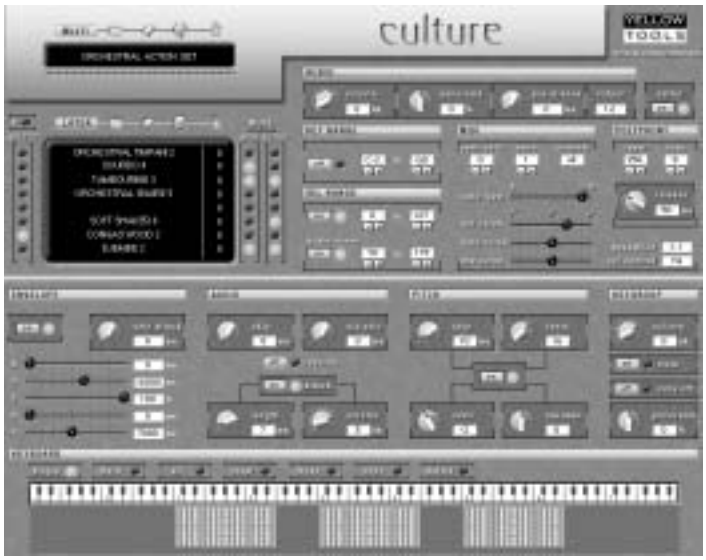
CULTURE



Modular Virtual Percussion Instrument

Culture delivers an unprecedented level of quality and control to satisfy the demands of today's top professional musicians.
With over 9GB (2 DVDs) of authentic and highly optimized sounds, Culture is the world's largest collection of world, ethno, orchestral and industrial percussion instruments. The essential sonic details of each instrument have been captured perfectly, making Culture sound incredibly real. Each instrument was sampled using every playing technique imaginable… 16 unique velocities per note for left AND right hand! The natural touch of each instrument makes Culture a percussion player's delight. For example, play crisp bass tones at the center of a djembe drum by playing the middle keys, then work your way out to the outer keys for some serious rimshots…brilliant!! Culture includes a full range of percussion instruments such as: djembes, tamdrums, udus, darabukas, shakers, barrels and cans, framedrums, orchestral timpani, orchestral snares, cymbals, tonal gongs, chimes, bongos, congas, claps, tambourines, dunun, sangban, kenkeni, balafon and many more standard and exotic percussion instruments.

FEATURES

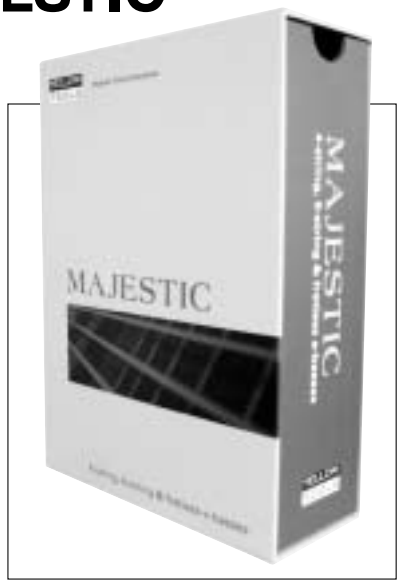


Culture's audio engine is a completely independent development of yellow tools and offers powerful features, an intelligent user interface and is optimized for the simultaneous use of several-thousand sound samples. The amazing results are made possible by the fact that the engine was developed for the sounds AND the sounds for the engine.

- Highly optimized engine for low CPU load
- Powerful MIDI processing including keyswitch option
- Flexible and intuitive handling of multis, layers and single keygroups
- Up to 16 audio outputs
- Simultaneous use of up to 8 different layers
- Hundreds of stunning single and multi patches using over 20,000 samples
- Intelligent polyphony management: protection of the natural release of the instruments and also the adjustment of voice usage per layers and notes to save CPU power
- The Pre Silence parameter applies a slight initial delay to each note that adds a humanizing element to your performance. For example, when a group of percussionists playing a note at the same time, they never hit the note right on the spot.
- Easy and fast switching between different instruments with previously assigned keys or the modwheel. So the hands can stay where the should be: on the keyboard...
- Basic and pro editor modes
- Cross platform plug-in compatibility: VST 2.0 (PC, Mac OS 9 and OS X); RTAS (MAC OS 9); MAS, RTAS OS X and AU for free download when available
- Compatible with Digital Performer, ProTools Logic, Cubase, Nuendo, SONAR, etc. up to 32bit/192kHz.

MAJESTIC

Modular Virtual Bass Instrument



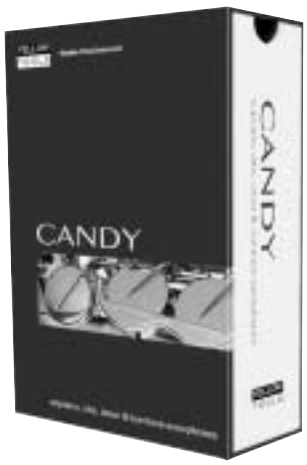
Majestic covers with 9GB finest sound samples the whole range of electric basses, useful for all styles of music. You can play Majestic's e-basses with the different playing styles fingered, fingered muted, slapped, thumbed, picked and picked muted and in addition with the playing techniques long notes, vibrato notes, short notes, ghosts, tappings, hammer ons, 8th notes, slow slides, fast slides, back slides, fret noises, FX... everything with up to 8 velocity splits per note. To achieve even more authentic e-bass performances, the instruments were recorded in two different scales. Next to the international well-known and specifically for the MVI optimized instruments of the yellow tools pure e-basses libraries, Majestic was enhanced with a vast number of e-basses.
Majestic delivers the whole range of e-bass instruments:
Music Man Sting Ray, Höfner Beatles Bass, Höfner shortscale, Ibanez BTB 5-string, Warwick Streamer Stage 1 fretless, Danelectro, Fender Jazz Bass, Fender Jazz Bass fretless, Fender Precision Bass, Rickenbacker, Epiphone Jack Cassidy Signature Model, Gibson EB-0, Steinberger headless and Warwick Thumb 5-string.



- Highly optimized engine for low CPU load
- Powerful MIDI processing including. keyswitch option
- Flexible and intuitive handling of multis, layers and single keygroups
- Up to 16 audio outputs
- Simultaneous use of up to 8 different layers
- Hundreds of stunning single and multi patches using over 20,000 samples
- Intelligent polyphony management: protection of the natural release of the instruments and also the adjustment of voice usage per layers and notes to save CPU power!
- Basic and pro editor mode
- Cross platform plug-in compatibility
- Compatible with Digital Performer, ProTools and any VST 2.0 compatible Host like Logic, Cubase, Nuendo etc. up to 32bit/192kHz

CANDY

Modular Virtual Saxophone Instrument



- The same 2 DVDs, 1 CD, MVI package as Culture
- Includes 11,000 Samples of the highest quality saxophones:
  - soprano, alto, tenor, bariton, bass saxophones
  - Classic, Jazz and Metal-mouth pieces
  - playing styles: long notes, short notes, vibrato, staccato, long and short sforzando, long & short falls, overdrives, FX - playing techniques: piano, mezzo-forte, forte, fortissimo, with up to 4 velocity splits per note.

Special features:

- after touch volume: press the already played key even more to increase the volume after the sample is already play. You can adjust the time and the volume
- alternate: if a sample is played twice, another sample of the same note is played with the second hit, so you will not have the same sample two times after another.
- SP-switch: you can switch between two layers with the sustain pedal. Amazing for long notes on the one layer and vibrato notes on the second. It is like keyswitch but you don't have even to move your fingers from the keys.


AKAI







B&H
PHOTO - VIDEO - PRO AUDIO

The ProfessionalAudio SourceBook

Section12

MIDI Equipment

AKAI.....**1190-1196**

Alesis.....**1197-1201**

E-MU.....**1202-1211**

Korg.....**1212-1231**

Kurzweil.....**1232-1239**

Roland.....**1240-1257**

Yamaha.....**1258-1277**

AKAI

Z4 • Z8

Z•Series Performance Samplers



Z8 shown with front panel removed and optional computer running ak.Sys Control software.

The Z4 and Z8 are the industry's first 24bit/96kHz hardware-based samplers. Z-Series samplers were designed for professional audio production with 64 voices of polyphony (32 voices at 24bit/96kHz), 272MB of RAM (expandable to 512MB) and a 60GB internal IDE hard drive for storing a serious sample library. Both units include Akai's award-winning ak.Sys PC/Mac Control and Networking System Software allowing you to control the Z4 and Z8 using your computer while taking advantage of the powerful DSP processing capabilities provided by Akai's custom Z-96 LSI sampling engine and a high-performance Intel CPU. Both units also feature balanced stereo analog I/O with 24-bit/96kHz converters. An option is available for adding eight additional analog outputs. What sets the Z4 and Z8 apart? Both units employ Akai's realtime controllers called Q-Link knobs that can be assigned to a variety of parameters. The Z4 has four Q-Link knobs while the Z8 has eight. The Z8 also features a removable front panel for remote editing as well as a S/PDIF digital I/O with a Word Clock input and a powerful 56-bit, 4-channel effects processor, both of which are optional on the Z4.

FEATURES

- Supports 24- and 16-Bit linear samples at sample rates of 44.1, 48, and 96kHz
- 272MB of internal SDRAM memory expandable to 512MB using off-the-shelf 168-pin DIMMs (PC133/PC100, CL2, 256MB max.
- 64-voice polyphony, 128-part multitimbral (32-voice polyphonic with the system clock set to 96kHz)
- 60GB internal IDE HD
- 248x60 graphic LCD display
- Built-in sample rate conversion is available on the digital input for capturing audio from peripherals such as CD players.

File Compatibility

- AKAI S1000, S3000, S5/6000 and MPC2000XL - Sample and Program files
- EIII, EIV, Roland samples and programs
- SMF (Standard MIDI File) playback
- FAT32/DOS disk format and .Wav implemented as native format
- .AIFF also supported using ak.Sys Networking Software on the Mac.

Q-Link Knobs For Realtime Control

- The Z4's four and the Z8's eight realtime Q-Link knobs can be individually assigned to control a wide variety of parameters including level, pan, filter cut-off and resonance, start offset, effect send, tuning and more. Control of an entire program or a sound assigned to a single keygroup is possible.
- Each assignable Q-Link knob can generate MIDI controller data so that parameters movements can be recorded and played back from a sequencer.

Digital Dynamic Filters

- 36 types of 6-pole digital dynamic resonant filters plus a special 'Triple' mode that supports up to three independent 2-pole filter blocks per voice.
- Realtime control of the cut-off frequency and resonance from the Q-Link knobs or via MIDI control change messages.
- Dual independent LFOs with internal, external modulation control and MIDI clock sync.
- Three Digital Envelope generators including two multi-stage envelopes.

Inputs and Outputs

- Two 1/4″ TRS balanced record inputs
- Two 1/4″ TRS balanced outputs
- 24bit/96kHz A-to-D and D-to-A converters with Enhanced Dual Bit 64-times oversampling A-to-D and Advanced Multi-bit 128-times oversampling D-to-A conversion.
- Two MIDI In and Two MIDI Out/Thru connectors
- A USB master port allows you to interface USB compatible floppy disk drives, hard disks, removable media drives, or a USB keyboard for quick name entry.
- An additional USB slave port is provided for connecting a Mac/PC running the AKAI Professional ak.Sys software.
- 50 pin high density SCSI port

Z8 Adds

- 2-channel S/PDIF digital I/O
- Word Clock input
- 8 x Q-Link assignable controllers for realtime modulation control
- 56-bit, 4-channel effects processor (EB4JS)
- Removable front control panel for remote operation.

Z4 • Z8

IntelliSample

- IntelliSample provides an automated sampling function that allows you to easily create custom sounds while eliminating complete steps in the sound creation process. For example, using IntelliSample is a convenient way to continually record from an audio CD, normalize, name, and assign the samples to a program automatically.
- IntelliSample uses an audio triggering to begin and end the sample recording process: Recording is triggered when the input level exceeds an adjustable threshold level, and stops when the level drops below the threshold level. The recorded samples are automatically named and assigned to a new program, or any specified program.

56-bit Multi-Effects

- The 24bit/96kHz 4-channel effects processor (Optional on the Z4, standard with the Z8) offers over 40 effect types including reverbs, chorus, flanger, phaser, auto and touch wah, rotary speaker, pitch shift, delay as well as dynamics processors such as compressors, noise gates.
- A Pitch Corrector effects is also provided for intelligently correcting out of tune vocals and other monophonic instruments.

Quick FX

Quick FX allows sounds to be easily resampled and simultaneously processed with the internal 24bit/96kHz effects processor. A variety of presets are available that describe the final result and further simplify the effects assign and programming process. Presets such as Dirty, Swell, Squash, Stretch, Distant, and Fat will provide immediate results for most effects applications while retaining the original sample.

Waveform Editing

Resample, Insert Sample, Delete Section, Silence Zone, Reverse Zone, Pitch Shift, Fade Up/Down, Normalize, Rescale (user defined normalize), TimeStretch, BPM Matching, Cross-Fade, Sound Slicing, and many more.

ak.Sys Control and Networking System software

- The included ak.Sys software provides brings intuitive, computer- based program editing, multi editing and file management functionality to your Mac or PC.
- The on-board USB Host/Slave port connects to your computer allowing you to take advantage of computer software control while relying on the processing DSP power of a dedicated hardware sampler.

Also Available

Z4 Studio A — All Z4 features plus IB-48P - 8-channel analog output board

Z4 Studio D — All Z4 features plus IB-4D, 2-channel S/PDIF digital I/O board with WordClock input IB4ADT - 2 in/8 out ADAT optical I/O

Z8 Studio A — All Z8 features plus IB-48P - 8-channel analog output board

Z8 Studio D — All Z8 features plus IB-4ADT - 2 IN/8 Out ADAT optical I/O

Options

EB4JS: 56Bit, 4-channel effects processor (Standard in the Z8)

IB-4D: S/PDIF 2-channel digital I/O with BNC Word Clock input (Standard in the Z8)

IB-4ADT: 2 in/8 out ADAT optical I/O

IB-48P: 8-channel analog output board

Z48-256MB: 256MB DIMMs

VZ8 – Software Sampler

The VZ8 is a 24-bit/96kHz compatible software sampler that offers the same specifications and performance capabilities as its hardware counterpart, the Z8 Performance Sampler. The VZ8 is available as a VSTi under Windows and as an Audio Unit plug-in under Mac OS X.

- 24-bi t/ 96kHz performance
- Up to 64 voices and 16 audio outputs per instance
- Over 30 filter types including 2, 4, and 6-pole lowpass plus various highpass, bandpass, notch reject, peak, phase shifting filters, a vowelizer and an assortment of combination filters.
- 'Triple' filter mode lets you to create outrageous and unique filter combinations.
- Three envelope generators (two multi-stage, one ADSR) per voice.
- Two multi-wave, MIDI synchable LFOs with variable waveform phase and shift.
- The comprehensive modulation matrix allows up to 35 control sources to be routed in completely variable amounts to over 50 targets.
- Eight virtual Q-LINK controls allow real-time performance.
- Easy to use – just insert the VZ8 into a track and start dragging and dropping sounds onto it.Layer up to 128 sounds across 16 MIDI channels.
- True compatibility with S1/1100, S2/S3000, S3000XL, S5/6000, Z4/8 sound libraries. Sounds can be loaded from your Mac/PC hard drive, CD-ROM drive or, using the ak.Sys Disk Browser.



- The Q-EDIT function provides quick and easy access to filter cutoff, resonance and envelope times.
- Seamless integration with with Akai's ak.Sys software, which provides access to all of the sampler's parameters as well as the ability to transfer data between your VZ8 and Akai's hardware samplers using simple drag-and-drop procedures.

AKAI

MPC2000XL

MIDI Production Center



The MPC2000XL is a complete MIDI production center that combines a digital sampler, 64-track sequencer and a tactile work surface based around 16 velocity sensitive drum pads. The sampling section features 16-bit stereo sampling, 32-voice polyphony and is expandable to 32MB with widely available standard SIMMs (2MB of RAM is included standard). Each voice has access to a dynamic resonant filter and envelope and samples can be precisely edited using the MPC2000XL's intelligent time stretch function, graphic cut-and-paste sample editing and resampling capabilities. The sequencing section has a 300,000 note capacity and allows you to record on up to 32 MIDI channels simultaneously. Two MIDI Out ports allow you to integrate external synths and synchronize with other workstations. The MPC2000XL can be pre-configured with a host of options including 8 individual outputs with S/PDIF I/O, SMPTE I/O and a 4-bus effects processor. These options, including memory upgrades are user-installable and can be added at a later date.

FEATURES

Sampler/Sound Generator

- Sampler, drum machine and MIDI sequencer in one instrument
- 44.1 kHz, 16-bit stereo sampling with 32-voice polyphony
- 2MB of memory is included as standard providing 21.9 sec. mono or 10.9 sec. of stereo sampling.
- Two memory expansion slots allow you to expand the available memory to 32MB using 16MB 72-pin SIMMs (70ns).

Control Surface

- 16 large, velocity- and pressure-sensitive drum pads and 4 pad banks provide a total of 64 pad/bank combinations.
- Up to three samples can be assigned to one MIDI note. The samples can be played simultaneously or can be switched via velocity, or with the Note Variation slider.
- Up to 256 sounds and 24 programs can be loaded into memory at one time.
- Note Variation slider for programming a wide variety of dynamics
- Large 248 x 64 dot flip-up LCD display for graphic waveform editing. Six soft keys under the LCD display provide direct access to functions on each page.
- Dedicated buttons for most-used functions

64 Track Sequencer

- The powerful and easy-to-use 64 track MIDI sequencer offers a great alternative to complex computer-based sequencers.
- A maximum of 50,000 notes (events) is available per sequence. The sequencer's total memory supports a maximum of 300,000 notes (events) and 99 sequences.
- Support is provided for both linear and pattern-style (loop-based) recording. The loop recording function allows you to quickly and easily loop short phrases.
- The Step Edit function allows you to view and edit the pitch, duration, position and velocity of individual events.
- A dedicated Undo Seq key allows you to undo sequence recordings or edits.
- Two (mergeable) MIDI inputs and two independent MIDI outputs allow you to integrate synths and other MIDI gear into the sequencing environment. Up to 32 channels can be recorded simultaneously.
- The sequencer's timing resolution is 96 parts per 1/4-note (ppqn).
- The Swing feature lets you add a swing-feel to the rhythm.
- The Auto Punch function enables you to execute automatic punch in or punch at designated points within a sequence.
- The 2nd sequence function allows you to have a separate sequence play at the same time that another sequence (or song) is being played/recorded.

Storage

- MPC2000XL models are available with either a 3.5˝ 2HD/2DD compatible 1.44MB disk drive or a 100MB Iomega Zip drive for storing and loading samples.
- 50-pin high-density SCSI port allows you to back up large projects to a variety of removable media such as Zip or Jaz disks, fixed hard drive or you can connect a CD Rom drive allowing you to access a vast array of commercially available sample libraries.

File Compatibility

- You can load samples from the vast Akai S-Series Sound Library as well as the Akai MPC60, MPC60-II, MPC3000 libraries, E-Mu EIII and Roland S750/S760 CD-ROM libraries, as well as .WAV files from PC disks.

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MPC2000XL

Inputs and Outputs

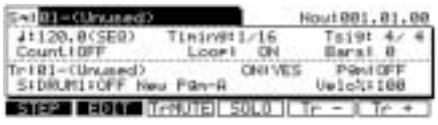
- Left and right balanced 1/4″ TRS record inputs with a Max. input level of +10dBu.
- Left and right unbalanced 1/4″ outputs. The optional **IB-M208P** I/O board provides an additional eight individual unbalanced 1/4″ outputs plus a coaxial (RCA) S/PDIF digital input and master output.
- 1/4″ stereo headphone output with dedicated volume control.
- Two independent MIDI outputs permit 32 simultaneous MIDI output channels.

Synchronization

- Synchronization and control capabilities include MIDI clock, MTC, MMC (MIDI machine control)
- The optional IC-M2TC board supports SMPTE time code synchronization. The supported SMPTE frame rates are 24, 25, 29.97 drop and 30.

Realtime Control

- Track mute can be turned on or off and sequences can be selected using the drum pads.
- The Note Variation slider allows you to control the decay or filter value of the sound in real time.
- The Next Seq key allows you to select another sequence to play directly following the one already selected.
- The Erase key works in conjunction with the drum pads to allow you to delete sequence data on the fly, without stopping the sequencer.
- The note repeat function and the after touch function pads enable you to easily enter drum rolls and hi-hat beats.
- The Tap Tempo function allows you to set the playback tempo by tapping a key in the time of 1/4-notes.
- The ability to convert MIDI sustain pedal data to note duration data allows you to insert sustain effects independently from the note data within a track.



Sample Editing



- Extensive cut-and-paste sample editing with waveform display.
- Waveform display with zoom function allows precision editing.
- Timestretch allows you to intelligently lengthen or shorten a sample using 18 different preset algorithms tailored to suit a variety of source material.
- BPM Match enables you to change the tempo of the sample so that it will fit into the track that you are working with.
- The Re-sampling function allows you to change the sampling frequency of the sample which is useful for conserving memory as well as for creating LoFi samples.

Synthesis



- A 12dB/octave dynamic resonant lowpass filter is available per voice.
- An Envelope with three parameters (Attack, Decay and Decay MD) is also provided.

Fully Editable Multi-Effects

- The optional EB20 effects processor offers two independent channels of multi effects plus two additional channels of reverb-only effects – all of which are fully editable.
- The two multi channels offer up to six simultaneous effects each, including reverb, echo/delay, distortion, EQ, ring modulation, and a choice of modulation effects such as chorus, phasing, flanging, pitch shifting and rotary speaker emulation.
- In Multi Mode, the EB20 is configured as four separate processors with effects sends available on each multi-timbral channel.
- The effects can also be configured to serve as a stand alone dual channel FX processor for other sound sources attached to the sampler's inputs.

Utilities

- The Utilities page provides lets you customize settings to suit your particular work style such as whether MIDI notes are shown as numbers or keys, loop lengths as a time value or samples and whether you want the Ent/Play button to play back a key as one note, a chord or an arpeggio.
- All customized settings can be saved to Flash ROM so that your sampler boots up exactly as you want it every time. Settings can also be saved to floppy disk or hard disk.
- Data can be exported to or imported from standard MIDI files.
- Tempo changes can be programmed mid-sequence or mid-song are supported.

Also Available

MPC2000XL Studio —
All MPC2000XL features PLUS IB-M208P, 8 output expander with S/PDIF digital I/O and IB-M20T, SMPTE Read + Generate

MPC2000XL Studio Plus —
All MPC2000XL Studio features PLUS EB16, 4 bus digital multi-effects processor, capable of 6 simultaneous effects

MPC2000XL-ZP —
MPC2000XL with built-in 100MB ZIP drive, 100MB ZIP disk loaded with 50MB of free MPC programs and sounds

MPC2000XL Studio-ZP —
MPC2000XL Studio with built-in 100MB ZIP drive, 100MB ZIP disk loaded with 50MB of free MPC programs and sounds

MPC2000XL Studio Plus-ZP —
MPC2000XL Studio Plus with built-in 100MB ZIP drive, 100MB ZIP disk loaded with 50MB of free MPC programs and sounds

Optional Accessories

IB-M208P: 8 individual output and digital I/O board

EB16: Multi-effects processor board

FMX008M: 8MB Flash ROM board

IC-M2TC: SMPTE IC

16MB SIMM memory board

AKAI

MPC4000

Desktop 24-bit/96kHz Sampler/Sequencer

The MPC4000 is Akai Professional's most advanced music production center ever. Built around the same Z-96 LSI sampling engine found in Akai's Z4 and Z8 Performance Samplers, combined with a high-performance Intel CPU, the MPC4000 provides ultra-fast processing with 24-bit audio resolution and support for sample rates up to 96kHz. This is the first MPC to integrate a full feature sampler that not only supports standard Drum programs, but also gives you the ability to use keygrouped programs such as piano, bass, strings, synth, etc. The MPC4000 uses .Wav as its native sample format and is compatible with Z-Series and S-Series multi, program, and sample files offering you instant access to a huge library of material. The MPC4000's 128 track sequencer boasts a 300,000 note capacity and 960 PPQN resolution. The large 6-inch backlit, hinged graphic display offers high resolution graphical editing of waveforms as well as sequence event editing, including piano roll style editing. Other features include: 272MB of RAM, expandable to 512MB; a built-in four channel, 56-bit effects processor; a 60GB internal hard drive and CD burner for archiving and loading data as well as burning audio CDs; plus a whole lot more.



FEATURES

- Includes all sampler functions of Z4/Z8 sampler – Z-96 sampling engine, combined with a high-performance Intel CPU provides ultra-fast processing and 24-bit audio resolution with sample rates up to 96kHz
- 64-note polyphony (up to 48kHz), or 32-note polyphony at 96 kHz
- Includes 272MB of internal memory. Expandable to 512MB maximum expandable using standard PC100 or PC133 SDRAM.
- Samples can be recorded to RAM or directly to the internal hard drive for streaming playback.

Data Storage

- A built-in 60 GB EIDE hard drive is included for sample storage.
- An internal CD Burner is also provided for master archiving, back-up, audio CD creation as well as loading commercially available samples CD-ROMs.
- FAT32 disk format is fully compatible with S5/6000, Z4, Z8, DPS24, and as well as the .WAV sample format allows data sharing with anything supporting .WAV files.

User Interface

- Sixteen velocity and pressure sensitive drum pads plus six pad banks provide quick access to up to 96 different sounds.
- Large backlit 320 x 240 dot hinged graphic LCD display.
- Up to six real-time controllers can be assigned to control a wide variety of parameters.
- Virtually all parameters are assignable to two Note Variation sliders, or four Q-Link knobs for real-time modulation control.
- All Q-Link knobs generate MIDI controller data that allows performances to be captured with the built-in sequencer.
- Memory meters display remaining free sampler memory, sequence memory and CPU memory
- IntelliSample automates sampling, naming, program creation and sample assignment

Inputs and Outputs

- Balanced stereo XLR-1/4-inch TRS combo inputs with reference-grade preamps, as well as (RCA) phono inputs that allows you to sample form a turntable without the need for an external mixer or amp.
- Balanced stereo XLR and 1/4″ outputs.
- All inputs and outputs utilize high quality 24bit/96kHz converters.
- Two MIDI In and four MIDI Out ports control up to 64 independent MIDI channels.
- SMPTE I/O is included standard
- A 50-pin Ultra SCSI port is provided for accessing additional storage peripherals.

Two USB Ports

- A USB host port supports an ASCII keyboard for naming, or USB floppy, Zip, CD-R, HD or MO drives for storage.
- A USB slave port is included for computer interfacing ak.Sys utility software.

MPC4000

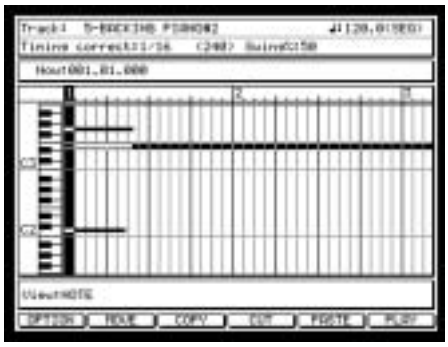
Sequencing

- You can record up to 128 tracks per sequence and up to 128 sequences can be stored in internal memory.
- The sequencer allows you to capture MIDI data at 960 ppq resolution and has a 300,000 note capacity per sequence..
- The length (1–999 measures), tempo, and time signature can be specified separately for each sequence.
- You can freely select the output destination for each track — the MIDI events recorded in the 128 tracks can be sent to the sampler section or you can access external synths by channelizing tracks to any of the four MIDI Outputs.
- Multiple sequences can be easily arranged in any any order to create a “song.” A song can be converted to a sequence.
- The large LCD also allows an unprecedented view of sequence data showing data as a piano roll or as a drum machine 'grid'. Continuous MIDI data can also be shown (and edited) graphically.

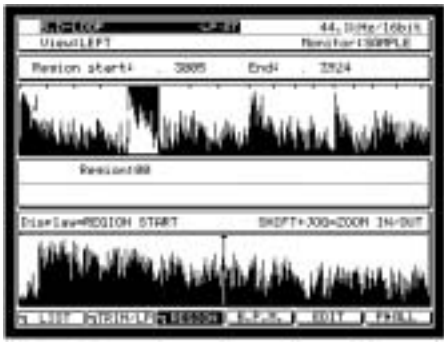
EB4JS 4-Channel 24-bit/96kHz Effects Processor

- Four-channels of 24-bit/96kHz compatible effects, processed at 56-bit. The four processors can be configured as mono in/mono out; mono in/stereo out or you can strap two processors together for true stereo I/O.
- Each of the four effect units has access to one of 52 effect types and each effect is fully editable allowing a wide range of effects to be obtained.
- The effects are accessed via four effects sends.
- Quick FX resampling using powerful presets from the four-channel 24-bit/96kHz effects processor. Samples can be edited by simple descriptive commands like Dirty, Swell, Squash, Distant, Fat, etc.

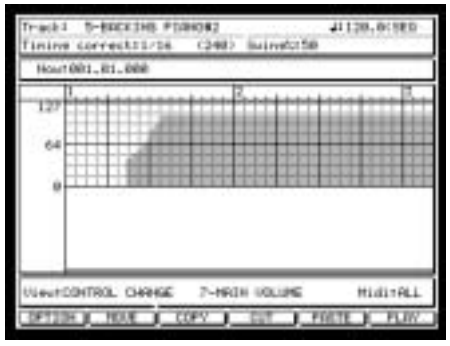
Large Backlit 320 x 240 Dot Hinged Graphic LCD Display



Graphical sequence event editing



Graphic editing of sample waveforms



Graphic editing of controller data

Sample Editing

- Besides standard cut copy and paste editing, the MPC4000 offers a wide range of tools for extracting the perfect groove for your productions
- The Slice Sound Function (also available on the MPC2000XL) allows you to divide a a sample into 32 samples and assigns them to a new program automatically.
- The Beat Loop Function calculates the tempo from the phrase length for Pitch Shift and Time Stretch functions to change the tempo.

Realtime Synthesis Control

- There are 26 types of 6-pole digital dynamic filters available with up to three simultaneous filters per voice (192 total). Filters can be independently controlled from a variety of assignable modulation controls.
- Dual multi-wave LFOs with internal/ external modulation control
- ADSR envelope generator, two multi-stage envelope generators,
- Virtually all parameters are assignable to two Note Variation sliders, or four Q-Link’ knobs for real-time modulation control.
- All Q-Link knobs generate MIDI controller data that allows performances to be captured with the built-in sequencer.

Computer Interfacing

- The free (downloadable) ak.Sys Network and Control Software allows complete Program editing, Multi editing, and file management via Mac or PC computers.
- The MPC4000’s intuitive interface combined with the generous ak.Sys software GUI makes the MPC4000 extremely easy to manage, and since all DSP processing takes place within the MPC4000 instead of the computer, it won’t overtax your CPU.



OPTIONS

- IB-4D stereo S/PDIF digital I/O with a built-in sample rate converter to eliminate rate-matching problems and a WordClock input
- IB-48P analog output board with eight assignable balanced outputs.
- IB-4ADT Two In/Eight Out ADAT optical digital I/O board.

MPD16

USB/MIDI Pad Control Surface

The MPD16 is a self-contained control surface based on the drum pad design found on Akai's legendary MPC series of music production systems.The MPD16 connects via USB and/or MIDI to your computer, synth, drum module or sampler — anywhere you want to take advantage of a velocity drum machine-style tactile interface.



- Designed with simplicity in mind – allows you to program your drum sequences using a drum machine style interface instead of your keyboard.
- 16 MPC-spec pressure-and-velocity sensitive drum pads
- 16-Level mode allows you to generate the note velocity in 16 steps with 16 pads. Full-Level mode produces data with velocity value of 127, regardless of the velocity of the pad tapped.
- Bank key allows you to switch between two banks of sounds.
- The Active key enables the fader which outputs MIDI control changes.
- Parameter settings such as MIDI Note Assign, MIDI Channel, MIDI Control Change, etc. can be assigned from the computer via USB.
- Set-up is facilitated with the MPD16 Utility and PC/MAC driver software CDROM.
- Powered via your computers USB port or the optional AC adapter.



MFC42 MIDI Filter Controller

The MFC is a digitally controlled analog filter module optimized for use with Akai's MPC-Series of Music Production Systems. It features four selectable filter types, real-time control of filter, LFO, distortion, phasing and envelope parameters. The MFC42 can be rackmounted or mounted directly on an MPC2000/MPC2000XL using the optional KIT-MFC Mounting bracket kit.

- Two independent channels: a stereo channel, for processing MPC drum sounds, with a choice of 2-pole and 4-pole filter curves; and a mono channel, for bass with a choice of a 2-pole, 4-pole and an awesome 8-pole filter.
- Four filter types are available including: low-pass, high-pass, band-pass and notch filters. Toggle switches allow you to immediately switch to any filter type, with having to scroll through unwanted choices.
- Dedicated controls are provided cutoff frequency and resonance allowing real-time sound sculpting – from sub-sonic to extreme weirdness.
- An assortment of standard inputs, plus direct turntable inputs, allows the MFC42 to be used in just about any application.
- In addition to the powerful filtering capabilities, the MFC42 provides one LFO, one Envelope Generator
- Almost all of the controls on generate MIDI controller data, allowing performances to be recorded in the MPC's sequencer.
- You can assign the LFO and EG (envelope generator) to either channel to provide real-time changes. The on-board LFO can sync to external MIDI clock to easily match the tempo of any beat.
- The LFO rate can also be controlled using TAP TEMPO as well as BEAT SYNC which allows the duration of the LFO/Envelope when using the tap tempo feature.
- When using the stereo channel, you can shift the LFO phase between the left/right outs.
- While each of the channels is independent, LINK and INVERT functions allow interaction between the channels. By linking the channels together, any changes made on one channel will actually affect both channels. The INVERT function allows the control of one channel to have the opposite effect on the other channel.
- A Trigger button allows instant filter triggering -no keyboard required.
- Large, dedicated control knobs

Built-in effects

- Even greater sound shaping possibilities are provided via the distortion, a stereo analog phaser, and EQ.

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DM5 • DM PRO

Drum/Percussion Sound Module



The DM5 is an affordable, single rack space drum module that offers drummers and MIDI composers a professional, full featured tool for high quality drum, percussion and special effect samples. The DM5 puts over 540 sounds, many recorded in true stereo and with ambient effects. You'll find a range of classic Alesis studio drum sounds as well as cutting-edge sounds representing many of the latest musical styles. The DM5 also features four assignable 1/4″ outputs as well as a 12 ultra-fast trigger inputs. Whether you're needs quality drums and percussion on a budget or are looking to add new sounds to your arsenal, the DM 5 provides a low cost solution.

- Four 1/4″ analog outputs (2 stereo pairs)
- 12 trigger input jacks – each with 5 user adjustable trigger parameters.
- 18-bit digital to analog converters and 48kHz samples
- 21 programmable drumsets.
- Over 500 sounds including stereo samples with reverb, ambience
- User programmable panning for each sample (7 positions)
- 16-note Polyphony
- 127 velocity levels are available via MIDI or trigger inputs.
- 21 drum kit memory locations w/defaults permanently stored in ROM.
- MIDI In, Out/Thru connectors
- 1/4″ TRS headphone output with variable gain control
- 1/4″ switchable footswitch input can be assigned to control hi-hat or program advance mode.
- Coarse (chromatic) and Fine (cents) tuning is provided for each sound.
- Data knob, front panel keypad sound auditioning via velocity sensitive Preview button on front panel.
- Expanded Dynamic Articulation uses velocity to modulate the tone and pitch of samples
- Random Sample feature brings new life to static tracks by realistically varying the sound of the drum as you play.

DM PRO — 64 Voice 20-Bit Expandable Drum Module



The DM Pro is Alesis' most advanced drum module. It boasts 64 voice polyphony, and a 16MB drum and percussion library with a diverse range of over 1,000 sounds designed to accommodate nearly any musical style. A PCMCIA card slot allows you to add an additional 8MB of customized samples for a total of 24MB. The DM Pro also provides 16 drum trigger inputs using Alesis' most advanced triggering system. The DM Pro Expandable 20 Bit Drum Module is the most advanced device available today. For composers who need a dedicated MIDI module for drums, or drummers looking for the professional solution for triggered electronic drum performances, the DM Pro represents the most powerful tool for creative inspiration.

- 1,024 different kick, snare, tom, cymbal, hi-hat, percussion and special effect samples derived from 16MB of PCM ROM.
- Create and store up to 32 virtual drumkits, and each kit can contain up to 64 different drum sounds.
- Up to four samples can be layered on each drum sound. The DM Pro also provides complete envelope control
- Customizable onboard multi-effects use the same processing engine as the Q20 multi-effects processor.

Expansion

- An expansion slot allows access additional sounds via 8MB PCMCIA-format cards.
- Alesis' Sound Bridge sample management software allows you to burn your own sounds and samples from your computer (PC or Mac) onto a PCMCIA-format card.

Inputs and Outputs

- Six 1/4″ audio outputs provide optimum mixing and processing flexibility.
- Stereo RCA audio inputs accommodate external audio sources, like CD players or tape decks, to be routed through an audio system so you can practice with recorded music.
- 16 fast trigger inputs for drummers using electronic pads or triggered acoustic drums.

ALESIS

QS6.2 • QS8.2

64 Voice Expandable Synthesizer





Alesis introduced the QS8 in 1996, and shattered the price barrier for professional 88-note keyboards. Normal working musicians, and those just aspiring to work more, could suddenly afford a no-compromise instrument for a very compromising price, and this tradition continues with the QS6.2 and QS8.2. Featuring serious sounds for serious players, the QS synths are loaded with stunning pianos and keyboards, a soup-to-nuts collection of orchestral instruments, killer drum and percussion kits, and enough amazing synth textures, fat basses, and sizzling leads to keep dance musicians and remixers grooving into the next time zone. A deep programmable synth architecture and four-bus effects processor add the finishing touches. And a wide variety of fresh samples are available from our extensive QCard library.Whether you need an 88-note workhorse or a do-it-all synth, the QS8.2 and QS6.2 keep your fingers flying for a down-to-earth price.

features

- QS8.2 features an 88-key, fully-weighted hammer-action keyboard with velocity and release velocity.
- QS6.2 has a 61-key synth-action keyboard with velocity, release velocity, and aftertouch
- Pitch and modulation wheels as well as four user definable edit/control sliders are provided for controlling synth and effects parameters in realtime.

Sounds

- 16MB of internal wave memory, expandable to 24MB via PCMCIA Q cards.
- 16-bit linear 48kHz samples
- 640 Programs (512 preset, 128 user) including Alesis' acclaimed Stereo Grand Piano, and 500 multimbral mixes (400 preset, 100 user).
- To make it easier to find a desired program, they are organized into 11 categories based on instrument type: Pianos, Chromatic, Organ, Guitar, Bass, Strings, Brass, Winds, Synth, Rhythm/FX, Drums/Percussion

QS Composite Synthesis

- True 64-voice polyphony and 16 multitimbral parts.
- Sweepable lowpass filter, 3 envelope generators, 3 LFOs and programmable effects send per voice along with user defined modulation routings.

QS Parallel Matrix Effects

- The powerful built-in 4-bus digital multi-effects processor is capable of generating a variety of effects for adding spaciousness, presence or even mutate sounds including: reverb, chorus, distortion, EQ, delay, rotary speaker simulation, and more.



Mix Mode

- Mix Mode allows you to access up to 16 different Programs at once from the keyboard or an external sequencer. Each of the 16 parts can have its own level, panning, transpositions and effect send settings.
- You can use Mix Mode to create complex layers and/or splits. Example you can create a lush piano, string layer or create a split with an acoustic bass on the left side of the keyboard with an organ on the right side.
- You can also use Mix Mode to receive data on 16 MIDI channels simultaneously from an external sequencer, with each channel representing a different instrument.

Inputs and Outputs

- Stereo L/R 1/4″ outputs using 24-bit 128x oversampling delta-sigma Digital to Analog converters.
- 1/4″ Stereo headphone output
- Assignable aux pedal, sustain pedal inputs.
- MIDI in, out, and thru connectors

QS6.2 • QS8.2

Expansion

- A QCard expansion slot, compatible with Type I PCMCIA SRAM and flash cards, allows you to add 8MB of additional samples and programs from Alesis' comprehensive QCard library.
- The programs in each QCard employs 16-bit, 48kHz uncompressed digital samples for exceptional clarity, dynamics and detail. Each 8 megabyte QCard also offers 100% plug-and-play convenience and can be used with any expandable Alesis synthesizer.

Sound Bridge

- Sound Bridge is a free (downloadable) software application for the Mac and PC that allows you to download your own sequences, Program Banks, and samples (.AIF, .AIFF, .SD, and .WAV) from your computer into a Flash card plugged into the QS. All you need is a Flash card and a computer with a MIDI interface.

Available QCards

Stereo Jazz Piano – featuring multisamples taken from a renowned concert grand favored by top jazz players everywhere.

Stereo Classical Piano – captures the rich, authentic tone of a world-class grand piano. It's ideal for any musical style that demands superb piano sound.

Latin –vibrant sounds and rhythms, ranging from traditional to contemporary including Tex-Mex, Tejano, Salsa, Merengue, Samba, and muc more

Classical – a collection of authentic classical instruments including solo and ensemble strings, brass, woodwinds, percussion and much more.

Sanctuary – A wide variety of instrument, choral and special effect sounds for the creation and performance of spiritual music for churches, musicians and composers.

EuroDance – Create powerful Techno, Trance, Ambient, Industrial or House music with this awesome collection of drums, bass, noise bursts, filter sweeps and synths



Hip Hop – a cool collection of sounds for Hip-Hop, R&B, Swing, Jungle, Acid Jazz, House and other urban music styles

Vintage Synthesizers and Classic Bea Boxes – Groovy synth sounds and lo-fi drums from the dawn of the age of electronic music.

Vintage Keyboards – a collection of super-realistic sounds from the greatest keyboard instruments from the '60s, '70s and '80s including electric pianos, clavs, organs and much more.

SR-16 *Drum Machine*



The SR-16 is one of the most popular drum machines ever made. It features a great selection of 233 realistic, natural drum sounds, offered both in dry form and sampled with Alesis' renowned digital reverbs. 50 preset patterns, performed by top studio players, are provided with enough rhythmic variations to create a complete arrangement making the SR-16 an ideal companion for songwriters. Live musicians and studios will appreciate the SR-16's straight forward drum programming, 12 velocity sensitive drum pads and four outputs as well as its ability to sync to external sequencers via MIDI clock and Song Position Pointer.

Sounds

- 233 internal 16-bit sampled sounds many of which are stereo samples with reverb and ambience.
- 50 preset drum kits are provide plus you can create up to 50 of your own kits. Drum kits program changes can be made via MIDI.
- You can access up to 12 sounds at a time via the drum pads, and up to 120 sounds via MIDI.
- 16 voice polyphony
- Adjustable volume and pan per sound

Drum Pads

- 12 velocity sensitive drum pads with a Dynamic Articulation function that enables a drum sound to change its tonal content as it's played harder, adding a humanizing element to performances.
- There 8 loudness (velocity) levels via pads, and 127 levels via MIDI.
- You can sync the SR-16 to external devices via MIDI Clock and Song Position Pointer.
- A footswitch input allows you to access a number of functions including: start/stop, count/A/B/fill.

Pattern-Based Sequencing

- 96 PPQN timing resolution
- You can create patterns with any time signature (1-128 beats per pattern)
- The tempo can be set between 20-255 bpm
- 12,000 note capacity with 400 pattern memory locations (200 user, 200 preset) and 100 memory locations for songs.

Outputs

- Four 1/4″ outputs (2 stereo pairs) with 18-bit D-A converters.

ALESIS

ION

Analog Modeling Synthesizer



The ION is a 49-key analog modeling synthesizer with a super-charged sound engine that packs eight voices of smooth, high-bandwidth analog-style punch, with three oscillators and two multi-mode filters per voice. This four-part multitimbral sound engine serves up everything from warm, round bass tones to lush pads and gritty lead sounds, making ION perfect for music styles ranging from rock to funk and the latest house, techno and trance dance styles. The ION's intuitive user interface begs to be tweaked with 30 knobs, over 70 direct-access buttons, a pitch bend wheel, and two assignable modulation wheels that provide realtime control over synth parameters. From radical filter sweeps to subtle tonal changes, everything is right at your finger tips, just like the classic analog synths of the past. The ION also has an array of onboard effects including chorus, flange, phaser, echo, distortion, fuzz, compressor and more — all developed using Alesis' legendary effects algorithms. The ION even offers a full-featured Vocoder, which can perform up to 40 bands of vocoding. Connections include four balanced analog outputs and stereo balanced analog inputs, as well as MIDI In and Out/Thru ports. A built-in switching power supply is also included.

FEATURES

- 49 velocity sensitive synth action keyboard
- Eight note polyphony and four multitimbral parts with a proprietary 500 MIPS sound engine that is capable of delivering everything from warm, round bass tones to lush pads and gritty lead sounds, perfect for a wide range of musical styles from retro rock and funk to futuristic house, techno and trance dance styles.
- 512 Preset Programs, all of which are user re-writeable

Synthesis

- 3 oscillators per voice, with continuously variable wave shapes, Sync and FM.
- 16 types of filters are available – up to two multimode filters can be used simultaneously in series, parallel, or stereo configurations.
- 2 LFOs, Sample and Hold and Arpeggiator - all Sync to MIDI Clock
- External signals can be routed via the balanced, 24-bit analog inputs where they can access the onboard filters, effects as well as a 40-band Vocoder.

Performance Friendly Interface

- Cleanly laid-out and intuitive user interface begs to be grabbed and tweaked.
- 30 infinitely adjustable, realtime control knobs - on high-resolution pots, not encoders, over 70 direct-access, backlit buttons
- Large 160x160 high resolution graphic LCD provides visual feedback for a variety of operations.
- Sustain and expression pedal inputs are also provided offering even greater control capabilities.
- Pitch bend wheel, and two assignable mod wheels.



Alesis Effects

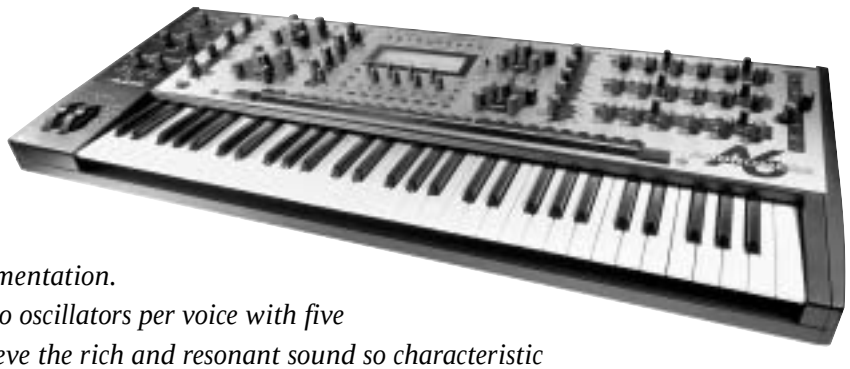
- 4 Individual Mono/Stereo Insert Effects and Stereo Master Multi-FX Processor.
- Includes many of Alesis' renowned effects including: chorus, flange, phaser, echo, distortion, fuzz, compressor and more.
- A full-featured Vocoder provides up to 40 bands of vocoding without affecting polyphony.

Connections

- Four 1/4″ balanced analog outputs and stereo1/4″ balanced analog inputs all with 24-bit converters.
- 1/4″ stereo headphone output
- MIDI In and Out/Thru ports.
- Built-in switching power supply

A6 ANDROMEDA

16-Voice Real Analog Synth



The Alesis A6 Andromeda is a real polyphonic analog synthesizer designed to uphold the legacy of classic American synthesizers. Andromeda offers a 61-note synth-weighted keyboard features velocity and aftertouch sensitivity, 16 true analog voices and 16-channel multitimbral functionality with full MIDI implementation. The keyboard's powerful voice engine includes two oscillators per voice with five waveforms each, and two sub-oscillators. To achieve the rich and resonant sound so characteristic of analog synths, Andromeda features 2-pole multimode and 4-pole lowpass classically derived resonance filters. To unleash the potential of Andromeda's true analog sound, there are 72 knobs and 144 buttons that allow real-time control of virtually every parameter. Other powerful features include MIDI-synchable LFOs, an arpeggiator and an analog-style step-sequencer. To manipulate sounds even further, Andromeda offers a high quality digital effects processor along with a dedicated analog distortion circuit. In addition to the standard pitch and mod wheels, Andromeda also has a large, multi-function ribbon controller that offers an expressive alternative for controlling real-time parameters.

FEATURES

- 61-note synth-weighted velocity sensitive keyboard with aftertouch.
- 256 internal preset programs and 128 user programs.
- 16 true analog voices and 16-channel multitimbral functionality with full MIDI implementation.

True Analog Subtractive Synthesis

- Two oscillators per voice with five waveforms each (sine, triangle, square, up saw, down saw), and two sub-oscillators.
- 2-pole multimode resonant filter and a 4-pole lowpass resonant filter per voice.
- Three MIDI-synchable LFOs each with six waveforms and three 7-stage envelopes.
- Portamento with nine slopes, and legato functions.
- Analog-Style Sequencer with 16 steps and four levels: Note, the note's Velocity, the note's Gate time (duration), and Type which selects whether a note will play or the step will be a rest.
- Up to 16 user-defined arpeggiators can be created and stored per program (one per voice).



Hands-On Interface

- 640 x 240 backlit graphic LCD display provides real world values (time, frequency, BPM, etc.) with high-resolution graphics.
- A total of 72 knobs and 144 buttons allows for real-time control during performances.
- Pitch and mod wheels are provided as well as a large ribbon controller, which offers multiple, assignable functions, and allows for greater tactile control.

Fully Progammable Effects

- Two complete effects systems are provided: the first produces a variety of analog-generated distortions (overdrive, classic fuzz box and more) and the other for producing an array of studio-quality digital effects (reverbs, delays, chorus, flanging, pitch-shifting and rotating speaker).
- The digital effects contain a number of single effects plus some very useful multi-effects where two or more single effect generators are linked.

Inputs and Outputs

- An extensive audio input/output array provides maximum routing flexibility.
- 16 mono outputs via eight 1/4″ TRS jacks — one for each of the 16 voices.
- Stereo 1/4″ main outputs
- Two 1/4″ mono aux outputs (jacks)
- 1/4″ stereo headphone output
- External audio sources can be routed to the 1/4″ analog inputs and processed via the A6's Filters and modulation sources.
- Dedicated 1/4″ Pedal/CV, Switch and Sustain jacks
- MIDI In, Out and Thru connectors

Conveniences

- An Auto-Tune function automatically tunes the 16 oscillators that may become detuned due to temperature changes.
- A card slot allows you to save and recall custom programs and mixes to a standard PCMCIA-format memory card.

E-MU

PROTEUS 2000 SERIES



Expandable Sound Modules

There are a total of six single rack space sound modules available in the Proteus family covering a wide spectrum of musical styles from Pop to HipHop and Techno to Classic Synths and Keyboards to Orchestral instruments. Each module features a powerful synth engine with 32MB of sound ROM packed with over 1000 presets, dual stereo 24-bit effects processors, an ultra-fast 32-bit processor and real-time control over an extensive range of synth parameters. In addition, each module can be further expanded by adding any of the large number of Proteus 2000 expansion -ROMS. Modules are available with either 64-note polyphony and 16 multitimbral parts or 128-note polyphony and 32 multitimbral parts allowing you to play back complex sequences with timing and expressivity that will meet the expectations of the most demanding composer and producer.

THEY ALL FEATURE

Powerful Synthesis

- Up to four synthesizer voices can be layered per preset making it possible to program just about any kind of sound. Layers can be switched or crossfaded using key position, velocity, real-time controllers or any modulation source.
- 50 different types of 2nd to 6th order deep resonant and modeling filters allow you to shape and modify the raw waveforms.
- 64 modulation sources are also provided including three multistage envelopes and two LFOs per layer, as well as full MIDI control over virtually every parameter.
- The LFOs and Envelopes can be synced to MIDI clock or to an internal tempo.

Control

- The digital patch bay, with 24 cords per layer, (and 12 more cords per preset) allow you connect modulation sources to up to 64 destinations in any imaginable way.
- Four front panel real-time controllers can be switched in three layers giving you control over 12 user-selectable parameters. The real-time knobs can also control multiple parameters at once.
- Virtually every synth parameter is controllable using the real-time knobs or by any internal or external control source.

Dual 24-Bit Stereo Effects

- 24-bit stereo effects allow you to add richness, depth and color to your sound. You can choose from over 60 different algorithms and effects can be linked to each preset or used globally.
- The effects section is actually two separate effects processors with control over each wet/dry mix level on four effects sends.
- Effects Processor “A” contains primarily ambiance algorithms like reverb and delays, while effects processor “B” contains primarily spectral algorithms such as chorus, flange, phase, distortion, and delay.

Expandability

- Each module has between one and three user installable expandable sound ROM sockets for adding any of fourteen available 16MB or 32MB ROM sets.
- You can also author your own -ROMS using the E4 Ultra sampler.

Additional Features

- The powerful Sound Navigator function makes it easy to find the exact sound you want from the thousands available. It's powerful, yet simple to use.
- 12 user-definable alternate tunings, and, of course, an extensive MIDI implementation.

Selected Models Include–

- Each module features either 64-note or 128-note polyphony.
- Modules may be outfitted with dual 1/4″ outputs or upto six outputs and a S/PDIF digital output.
- Modules may also be outfitted with either 16 or 32 MIDI channels.

SuperBEATS Mode (Available in Selected Models)–

- **SuperBEATs Mode** is a 16-track play-only sequencer that is optimized for live performance and groove creation. It allows you to trigger, latch and unlatch synced loops and grooves from your MIDI controller.
- SuperBEATS compatible sound modules include dozens of special 16-part BEATS Riffs created by some of the best producers in the business.
- Rhythmic Pattern Generator/ Arpeggiators can play up to 16 synchronized arpeggiator patterns at once.
- Patterns can be edited using pattern flow commands such as: delay for 2 bars, play for 4 bars, hold for 2 beats and repeat. There are 200 factory patterns included in addition, you can program or download up to 100 user patterns.

PROTEUS 2000 SERIES

Proteus 2000



This powerful 128-voice expandable sound module is E-MU's first sound module to employ the ultra-fast 32-bit processor, extensive real-time controls and powerful synthesis that allows you to playback complex sequences with remarkable timing and expressivity. The versatile 32 MB Composer soundset offers you 1536 presets (1024 ROM, 512 user) with a wide and highly useful assortment of instruments covering many different styles of music. Three additional 32MB ROM slots let you add up to 96 MB of additional sounds as well as the ability to author your own custom sound -ROMS using E-MU's E4 Ultra samplers.

Proteus 1000



The Proteus 1000 is a 64 voice expandable sound module that offers the same 32 MB Composer soundset with 1536 presets (1024 ROM, 512 user) and lightning-fast processor as the Proteus 2000. Proteus 1000 also provides you with an additional 32 MB ROM slot, so you can expand your sounds using any of the available Proteus -ROMS including custom sound -ROMS using E-MU's E4 Ultra samplers.

Virtuoso 2000



The Virtuoso 2000 is a 128-voice expandable orchestral sound module with a comprehensive 64MB orchestral sound set powered by the Proteus 2000 synth engine. E-MU's sound designers spent months sampling dozens of solo and section instructments, all from the same professional symphony orchestra, to develop an orchestral ROM sound set that is second to none in realism, versatility and consistency. Meticulously crafted into 512 presets that include: violins, violas, celli, double basses, trumpets, French horns, trombones, bass trombones, tubas, Piccolos, flutes, oboes, English horn, clarinets, bass clarinets, bassoons, contra bassoons, harp, orchestral bells, tubular bells, bell trees, celeste, xylophone, marimba and percussion battery with over 20 instruments.

Orbit 3



The Orbit 3 Super Synth is a 128-voice electronica/dance sound module with a electrifying 64 MB soundset created by famed sound designer Rob Papen – well known for his work on both Access and the Waldorf synths. From fat analog synth patches to "in your face" acoustic dance kits and percussion, Orbit 3 offers fresh, diverse and eclectic dance sounds. The Orbit 3 also employs SuperBEATS mode which consists of multi-layered sequences that you can interact with to create unique performances. Using the front panel, real-time controller knobs, you can manipulate different aspects of a groove or rhythm pattern.

Mo'Phatt



Mo'Phatt is a 64-voice Urban Dance Synth that is loaded with the hottest Hip-Hop and R&B sounds and beats. With its 32 MB expandable soundset, 12-pole digital filters, synchable/programmable arpeggiators, and interactive SuperBEATs Mode, Mo'Phatt has become a production tool staple, found in the studios of many composers, producers and musicians. A Turbo upgrade option increases polyphony to 128 voices, and provides 4 additional outputs (6 total), a S/PDIF digital output, two additional internal ROM expansion slots and 16 additional MIDI channels. Also available with the Turbo upgrade pre-installed.

Vintage Pro



The Vintage Pro is a 128-voice retro-styled sound module with a 32 MB soundset that packs an extensive collection of sampled vintage keyboards from the 60's, 70's and 80's, as well as an assortment of samples from E-MU's B-3 module. Each of the keyboards that were sampled had its own identity, both sonically and from a design standpoint. Using samples derived from the finest vintage instruments available, the Vintage Pro is an indispensable tool for musicians from classic rock and blues purists to the latest in dance music productions.

E-MU

PROTEUS FAMILY KEYBOARDS



PK-6 – Proteus Keys

The PK-6 offers cutting-edge Proteus sounds and features in a high quality keyboard package. The 32 MB composer soundset can be expanded to 128 MB using any of the Proteus expansion ROMS available. The 16 real-time controllers allow you to tweak the PK-6's synthesis architecture and filters, giving you total performance control over your sounds. And the PK-6's 16 synchable/programmable arpeggiators and interactive Super BEATs Mode let you perform entire pieces in real-time, all in perfect unison. Together with pristine 24-bit DACs and a lightning-fast processor, the PK-6 offers professional results and playability at an incredible price.

XK-6 – Xtreme Keys

The XK-6 is a 61-key Electronica performance/production keyboard that delivers the advanced synthesis architecture and 12th order filters found in the Xtreme Lead-1 Sound Module. A host of interactive SuperBEATs, arpeggiators and effects, all synced to BPM, deliver unprecedented rhythmic control. 16 mute/enable buttons bring beats in and out of your mix on the fly, while the 12 controllers allow you to tweak filters and synth parameters in real-time. The diverse 32MB Electronica/Techno soundset features everything from cutting lead sounds to drumkits to ethereal pads, and can be expanded using any of the many Proteus expansion -ROMS available. You can even create your own custom -ROMS using E4 Ultra samplers. Whether it's your main axe or an addition to your arsenal, the XK-6 delivers total sonic satisfaction in at an amazing price.

MK-6 – Mo'Phatt Keys

The MK-6 puts Hip-Hop sounds and beats at your fingertips. Based on the sound set of Mo'Phatt, it features a fresh new collection of presets and SuperBeats, giving you hands-on control of your sounds with a host of real-time performance controllers and buttons. And keeping your grooves tight is no problem, as your synth parameters, arpeggiators and interactive SuperBEATs all lock to your master BPM. The 61-key MK-6 ships with a 32 MB soundset that has been carefully crafted by top sound designers to create the first "state-of-the-art" keyboard dedicated to Hip-Hop and R&B. Three additional internal slots accommodate all of the Proteus expansion -ROMS available (up to 128 MB total).



Vintage Keys

The Vintage Keys keyboard delivers over 30 classic keyboards in a high quality, 61-note package. The 32 MB soundset includes the legendary synth and keyboard sounds that defined music in the 60's, 70's and 80's. All sounds are optimized for the Proteus synthesis and filter architecture, and many presets were voiced side-by-side with the the original instruments, for unmatched realism. Expand your sound palette further using any Proteus expansion ROM to configure your soundset to match your individual sound requirements. Together with its 16 simultaneous, synchable arps, real-time controllers and other performance features, Vintage Keys gives you an incredible selection of classic keyboard sounds.

They All Feature

- High-performance 61-note keyboards with velocity and aftertouch.
- 64-voice polyphony and 16-part multimbral with a super-fast sound engine for tight layered sounds, arpeggiators and grooves.
- 32 MB soundset, expandable to 128 MB using any Proteus expansion -ROMS (three additional ROM slots available).
- 64 multi setups, 640 Factory and 512 User presets with up to 4 layers per preset
- Over 100 synthesis parameters per preset
- Dual 24-bit effects processor with over 70 effects parameter including a number of BPM-synced effects.
- Over 50 types of E-MU's legendary 6 and 12th order filters.
- >100 BPM/ MIDI synchronized parameters per Preset
- Use up to 16 simultaneous arpeggiators that are perfectly synced to each other
- 300 Factory and 100 User arp patterns
- Control of the number of steps (up to 32 steps) with Key, Velocity, Duration and Repeat parameters for each arpeggio.
- 2 line by 24 character LCD display
- 4 programmable real-time control knobs give you instant control over 16 synth parameters including filters.
- Interactive groove control with 16 dedicated trigger buttons
- Other controllers include: Footswitch, Foot pedal, Modulation Wheel and Pitch Wheel
- Four 1/4″ analog outputs, with 24-bit D-to-A converters, configured as 2 stereo pairs.
- MIDI In, Out and Thru connectors
- Universal power supply: 90VAC – 260VAC at 50Hz – 60 Hz
- 40.75 width, 13.30″ depth, 4.32″ height (21 lbs.)

PROTEUS 2500

128-Voice Expandable Command Module



The four rack space Proteus 2500 Command Module is E-MU's 128-voice flagship sound module with three times the processing power of the Proteus 2000, a host of real-time editing controls, a full-featured MIDI sequencer, and USB connectivity to a host computer. This latest processor, combined with high quality 24-bit analog and digital audio outputs, delivers outstanding audio quality. The Proteus 2500 ships with a 32 MB soundset that can be expanded to 128 MB using any of the fourteen Proteus Expansion -ROMS currently available. The Proteus 2500 also ships with over 100 MIDI sequences, allowing you to play and modify existing sequences as well as your own.

FEATURES

- 128 voice polyphony and 32 simultaneous internal MIDI channels.
- 32 MB soundset, 512 presets and 512 user presets, with up to four layers per preset, provides you with a versatile sound palette that includes everything from lush strings and pristine pianos to synth stabs and hip-hop scratches.
- Three internal ROM slots allows expansion to up to 128 MB of sounds using any of the 14 available Proteus expansion -ROMS as well as Flash -ROMS authored using E4 Ultra samplers.
- Over 100 synchable synthesis parameters per preset, as well as 50 of E-MU's legendary 6/12th order filters.
- The internal processor is three times more powerful than the Proteus 2000, delivering better audio performance and flawless MIDI timing of even your most massive sequences.
- Six 1/4″ analog outputs with high quality 24-bit A-to-D converters and 24-bit coaxial S/PDIF digital output.
- One MIDI input and two MIDI outputs supports up to 32 external MIDI channels.

16 Track MIDI Sequencer

- Sixteen track sequencer with up to 16 MIDI channels per track can be used in either linear or pattern based modes.
- The sequencer's intuitive and flexible operating system allows you to select sounds, toggle between real-time, grid, and step record modes, edit parameters, and mix without ever stopping the recording process.
- 300,000 note sequencer memory with room for 512 songs and 1024 patterns
- 384 PPQN resolution
- Standard MIDI File import/export

Multi-channel Pattern Arpeggiators

- 32 simultaneous rhythm patterns or arpeggiators.
- 300 ROM and 100 User Arp Patterns
- Each arpeggio can have up to 32 steps with variable Key, Velocity, Duration, Repeat
- Arpeggiators can be saved with your sequence and can be modified in real-time using the control knobs.

Control

- Over 40 knobs and buttons deliver total hands-on control of all synth parameters (cutoff, Q, attack, delay, etc.) as well as sequencer functions (track mute, pan, volume, BPM, etc). Up to 24 modulation routing patch cords can be assigned per preset for customizing controller setups.
- Knobs and buttons can be programmed to control all of your external MIDI gear, allowing you to use the Proteus 2500 as the central control system for your studio setup and live performance rig.

24-bit Dual Effects Processor

- Two internal effects processors provide you with over 60 effects algorithms — from tempo-based delays to reverbs, distortion, chorus, flanger, phasers and more.

USB Port

- The USB port offers easy connectivity to a host computer allowing you to import and export Standard MIDI Files to and from the computer and accommodate easy software updates from the internet using E-MUs E-Loader application.

E-MU

XL-7 • MP-7

Command Stations

The XL-7 and MP-7 Command Stations are tabletop groove-oriented instruments each containing powerful multi-track sequencers and 128-voice synthesizers. Each Command Station features 128-note polyphony, a 16 track pattern based sequencer with real time, grid and step recording capabilities, 13 velocity sensitive pads, 32MB of ROM with over 1000 waveforms, three additional ROM expansion slots, digital output, and over forty performance realtime knobs, buttons and switches. The XL-7 Combines the Proteus 2000 and Xtreme Lead — ideal for High Energy Dance and Electronic music production. MP-7 Combines the Proteus 2000 and Mo-Phatt soundsets — ideal for Hip-Hop, Rap and R&B music production. The Command Stations can also function as a dynamic MIDI controller that will work with your external MIDI devices via its Sequencer, Arpeggiators, Controller Knobs, Trigger button section and Key Pad.



FEATURES

Control Surface

- The control surface is designed to provide quick and intuitive access to sound and sequence editing.
- One octave keyboard — thirteen velocity sensitive key pads (one octave) with after-touch, transposable over a 7 oct range.
- 16 programmable controller knobs for accessing synth parameters (LFOs, envelopes and filters) in realtime.
- Center detented touch strip
- 2 footswitches
- 4 button switches
- Controllers can be assigned to access external MIDI gear.
- Controller setups can be stored into multiple multi-setups.

Sounds

- 32 MB sound ROM — 512 ROM and 512 User presets with up to 4 layers per preset
- 128-note polyphony, 16 part multitimbral
- 50 types of 6 & 12th order filters
- Expandable to 128 MB using three additional ROM slots — choose from a large selection of Proteus 2000 Sound -ROMS to enhance and increase the types of available sounds.

Interactive Sequencing

- 16 track per pattern sequencer with up to 16 MIDI channels per track.
- Realtime, Step and Grid recording modes that can be switched in realtime without interruption to your creative process.
- 384 PPQN timing resolution
- 300,000 note internal sequencer memory with up to 512 Songs and 1024 Patterns
- Tap Tempo control
- 16 internal MIDI channels, 32 external
- 128 factory sequences
- Sequence editing tools include cut, copy, paste, quantize, swing, and more.
- 16 dedicated Mute/Select buttons allow you to add, monitor and modify parts on the fly — perfect for remixing.
- Song mode allows you to chain patterns in a cue list.
- E-MU's E-Loader program allows you to import and export MIDI files to and from your Mac and PC with ease.

24-bit Dual Effects Processor

- Two internal effects processors provide you with over 60 effects algorithms — from tempo-based delays to reverbs, distortion, chorus, flanger, phasers and more.

Multi-channel Pattern Arpeggiators

- 32 simultaneous rhythm patterns or arpeggiators.
- 300 ROM and 100 User Arp Patterns
- Each arpeggio can have up to 32 steps with variable Key, Velocity, Duration, Repeat
- Arpeggiators can be saved with your sequence and can be modified in real-time using the control knobs.

Inputs and Outputs

- Six 1/4″ analog outputs (configured as 3 stereo pairs) with 24-bit D-to-A converters. Outputs 3 - 6 can be configured as a stereo effects loop.
- Stereo S/PDIF digital output
- One MIDI Inputs and two MIDI Outputs

Other Features

- 2 line by 24 character backlit LCD display plus a 4 Digit 7 Segment LED.
- Optional rack ears for mounting in a standard 19-inch rack (6 rack spaces).
- 2 line by 24 character backlit LCD display plus a 4 Digit 7 Segment LED.
- 12 volt goose-neck lamp adapter

B&H PHOTO - VIDEO - PRO AUDIO
EQUIPMENT LEASING AVAILABLE

EXPANSION -ROMS

Definitive B-3

John Novello and E-MU present the Definitive B-3, a diverse collection of tone wheel organ drawbar settings sampled direct and with rotary speakers (chorale and tremolo) to give you the most versatile and playable B-3 available!

X-Lead

X-Lead consists primarily of analog, noise and modern synthesizer waveforms, which are rich in harmonic texture and are a perfect match with the Z-plane filters in your Proteus family sound module. With the included software upgrade, you now have access to dynamic and rhythmic features not found on any other instrument.

Orchestral Sessions Vol. 1

A 32MB collection of the most realistic and comprehensive section strings — the same sounds featured in E-MU's Virtuoso2000 sound module, complete with legato, spiccato, pizzicato and tremolande samples, and includes special presets for 4 speaker applications.

Orchestral Sessions Vol. 2

A realistic and comprehensive collection of woodwinds, brass, percussion, and solo strings – the same sounds featured in E-MU's Virtuoso 2000 sound module, complete with velocity switching between p/mf/ff samples, special presets for 4 speaker applications and a percussion battery with over 20 instruments.

World Expedition

An amazing collection of traditional instruments from the four corners of the globe including exotic stringed instruments, winds, percussion, and chromatic percussion. World Expedition delivers the most realistic ethnic soundset in the world.

Siedlaczek Advanced Orchestra-Special Edition

Some of the finest orchestral samples ever from Peter Siedlaczek, one of Europe's most respected sound designers. It brings together the finest string, brass, wind and percussion ensembles of Siedlaczek's legendary Advanced Orchestra library.

Beat Garden

Dutch sound designer Rob Papen and E-MU introduce the 32 MB Beat Garden expansion ROM, a diverse palette of all-new Electronica/Dance beats, synths, basses, drumkits, acoustic and electronic percussion.



Sounds of the ZR

Sounds of the ZR faithfully reproduces the diverse sounds of ENSONIQ's popular ZR-76 keyboard, including William Coakley's Perfect Piano. You'll find everything from synths and orchestral instruments in this one soundset.

Pure Phatt

E-MU's latest Hip-Hop soundset produces the freshest and newest sounds and grooves from top notch producers from the East, Southeast and West coasts. Pumping leads and basses, silky pads and those unique hits, scratches and drones that your mix needs to bring it to the next level are ready to provide the love.

Techno Synth

Rob Papen and E-MU deliver a compilation of synth, techno, and ambient sounds from one of Europe's finest programmers. From atmospheric pads to punchy sub-basses and crunch drum loops, these sounds will help you tear up the dance floor!

Protozoa

E-MU presents the ultimate sound compilation containing presets from the groundbreaking Proteus sound modules, all optimized for your Proteus 2000's advanced synthesis and filter architecture.

Proteus Pop Collection

The bread and butter sounds of E-MU's Proteus 2500 Module and PK-6 keyboard. Provides 640 diverse presets that cover the sonic spectrum with everything from traditional keyboards and orchestral instruments to electric guitars, world percussion and drumkits.

Ensoniq Project

The sounds of Ensoniq's acclaimed Urban Dance Project and Real World collections from their legendary MR and ASR sound libraries. It features the Hip-Hop and World sounds heard on countless hits, all optimized for E-MU's powerful synthesis and filter architecture.

Vintage Collection

The 32 MB Vintage Collection delivers the keyboard sounds that defined popular music in the 60's, 70's and 80's. From tonewheel organs and suitcase pianos to analog and digital synthesizers, Vintage Collection gives you an all-in-one, no-compromise source for all your classic keyboard sounds. Meticulously sampled from the finest vintage instruments available and optimized for the powerful Proteus architecture.

E-MU

E4 Ultra Series Sampling Synthesizers

E5000 ULTRA • E5000 ULTRA Turbo



The E5000 Ultra is E-MU's most affordable E4 Ultra Series sampling Synthesizer. It offers you all access to all of the state-of-the-art features available to the powerful EOS (Emulator Operating System), lightning-fast 32-bit Risc processor and proprietary tools like Beat Munging and Z-Plane filters. In fact, with the exception of its four standard outputs, 64 voice polyphony, 2 fewer ROM slots and new LED backlit display, the E5000 Ultra features the identical hardware and software of its more expensive models of the E4 Ultra line. And with the revolutionary RFX Card option, your E5000 Ultra will push the envelope even further. Never before has Emulator technology been so affordable. The E5000 Ultra Turbo adds a 3.2GB hard drive as well as the 16MB Orbit/Phatt Sessions ROM card with 500 presets derived from E-MU's Orbit and Planet Phatt sound modules.

E6400 ULTRA



The E6400 Ultra is E-MU's 64 voice expandable sampling synthesizer that delivers professional results at an affordable price. The E6400 is perfect for the budget-minded professional, providing superior sampling and synthesis flexibility with all of the latest advances in the powerful Emulator Operating System (EOS). The E6400 Ultra is fully upgradeable to E-MU's top-of-the-line E4XT Ultra, as well as supporting the various available sound -ROMS and flash -ROMS, allowing your E6400 Ultra to grow with your needs. The E6400 Ultra features Digital Modular Synthesis (DMS), 16MB of RAM (expandable to 128MB), a 48 track sequencer and more. The E6400 Ultra offers you all the features you need to deliver unique, complex and expressive sounds no matter what style of music you create, now and in the future.

E4XT ULTRA



The E4XT Ultra is E-MU's workhorse digital sampling synthesizer, created for the world of composition, performance and sound design. Equipped with a new ultra-fast 32-bit RISC processor, 128 voice polyphony, a 4 GB hard drive and 64 MB of RAM (expandable to 128 MB), the E4XT Ultra is a professional design tool to realize any sound imaginable. The E4XT Ultra also comes with a comprehensive palette of digital audio sculpting tools like Beat-Munging and Bit Converter, EOS Link software to control the E4 Ultra from your computer desktop, 24-bit dual stereo-effects processor and the capability to author flash -ROMS for E-MU's Proteus 2000 sound module. A legend in its own time, the E4XT puts the professional tools you need at your fingertips.

E4 Ultra Series Sampling Synthesizers

Sound Storage

- Arrange up to 1000 samples per bank in up to 1000 presets.
- The SoundSprint function automatically loads presets from the hard disk as you browse through them. You can continue to play while the new preset is loading in the background.
- The Sound ROM option gives you 16 MB of great samples with hundreds of user programmable presets which are always on-line and ready to go.
- A 50-pin SCSI connector is compatible with a variety of storage devices, including external hard disks (up to 18 GB), CD--ROMS, DVD-RAM, Jaz, Zip and more.

File Compatibility

- Roland S-700 & AKAI 3000/1100/1000 sound library compatibility

Built-in 24-bit Digital Effects

- The dual 24-bit stereo effects processor allows you to add reverb and chorus effects without having to leave the digital domain.
- There are 76 different effects available, each with a number of controllable parameters
- Sounds with effects applied can easily be resampled and then saved as an entirely new sound.

Digital Modular Synthesis

- Access up to 128 analog-sounding filters from the 21 different 6th order Z-Plane Filters with variable, cutoff frequency and resonance controls. These Filters include:
  - ❑ 12, 24, or 36 dB/octave Low-pass filters with resonance
  - ❑ 2nd & 4th order High-pass filters with resonance
  - ❑ 2nd & 4th order Band-pass filters with resonance
  - ❑ Contrary Band-pass filter
  - ❑ Three types of swept EQ filters
  - ❑ Three Phasers and one Flanger with resonance
  - ❑ Six powerful Morphing filters
- Two LFOs with variable Rate, 17 waveforms, variable Delay, Variation and Sync (key and free run)
- Amplifier Envelope (6 rates, 6 levels)
- Envelope (6 rates, 6 levels)
- Auxiliary Envelope (6 rates, 6 levels)
- The virtual Patch Cord architecture allows you to easily target the Z-Plane filters, LFOs, tuning functions, and three fully independent six segment envelope generators per sample to create any sound imaginable.
- You can create up to 24 Patch Cords derived from 60 sources and 60 destinations.

Advanced DSP

- EOS's advanced digital signal processing (DSP) features make sampling a breeze.
- Undo and Digital Re-Sampling functions
- Samples can be truncated, normalized and placed keymapped automatically. Samples can be controlled dynamically from the keyboard using velocity and positional crossfading and switching functions.
- Auto Correlation, Loop Compression and Crossfade Looping allow even the most difficult sounds to be looped easily.
- Digitally splice and mix samples with other samples.
- Advanced digital processing features such as Compression, Parametric EQ and Digital Tuning let you shape raw samples quickly and with greater precision than computer based systems.

48-track Linear Sequencer

- The 48-track sequencer contains a host of powerful features such as high resolution cut and paste editing, quantization, data filters, and external sync capability.
- Tweak sequences to perfection with intuitive Copy and Paste style editing and an array of tools which allow you to: transpose, quantize, scale velocity or rechannelize tracks.
- Program changes, pan, initial volume, submix, tempo, controller changes and SysEx data can all be stored with the sequence.
- Record and save up to 50 sequences along with each EOS bank.
- Note capacity is dependent upon the amount of RAM installed and how much you have allocated to sequences. As an example, over 400,000 notes can be stored if you have 4 MB of preset RAM.
- The virtual "Jukebox" lets you link up to 7 sequences and play them back in any order.
- Standard MIDI files (format 0 & 1) can be read and played back direct from a DOS floppy disk and saved to a hard drive.

Arpeggiator

- The arpeggiator is a powerful and easy to use performance-oriented function that moves a pattern of notes sequentially over a range of the keyboard.
- Control is provided for tap tempo, octave extension, note value divisor as well as the note pattern (up, down, or up and down or random).
- Arpeggiated notes can be recorded into the sequencer and are transmitted on the MIDI Out port.

Additional DSP Functions Include:

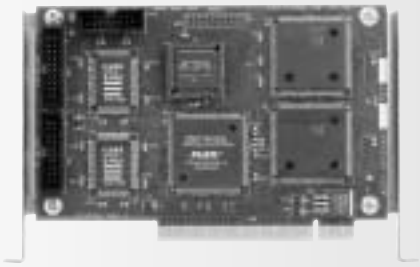
- **Time Compression and Expansion:** Shortens or lengthens the sample time without changing the pitch.
- **Pitch Change:** Changes the pitch of a sample without altering the time.
- **Transform Multiplication:** Creates weird and wonderful new timbres.
- **Aphex Aural Exciter:** Adds brilliance and sheen to your samples.
- **Dopplerization:** Moves samples forward and backwards in space as well as from side to side.
- **FIR Filters:** Non-realtime digital filters with "Ultra" high Q values and phase-linear filter response.
- **Bit Converter:** Removes unwanted bit resolution from your samples.
- **Beat Munging:** Analyzes drum loops and phrases and allows you to change tempo, time signature, swing and individual beats in real-time.

E-MU

E4 Ultra Series Sampling Synthesizers

RFX-32

Digital Effects and Mixer Card



Take a giant step into the future of sampling and synthesis with the RFX-32 digital FX/mixer card for E-MU's E4 Ultra Series Sampling Synthesizers, and blur the line between MIDI and Audio forever. Transform your sounds at the sample level with 32-bit effects, run external signals through E-MU's patented morphing filters and synthesis architecture, and then blend and modulate them in real-time using your MIDI controller. Take total control of your sound with the RFX-32.

- 16 studio-quality 32-bit stereo effects-large effects polyphony means you can have all those specialty effects as well as plenty of fat reverbs spread across RFX's many mixing busses
- Multiple FX setups and routing configurations-program each sample with its own exact blend of effects (up to 4 stereo sends with amount per voice)
- External signal processing through FX, filter and synthesis architecture-your E4 Ultra becomes a world-class digital effects/filter/mixing system with up to 16 inputs
- Multi-bus resampling-blend external and internal audio with powerful effects to create new samples
- Perfect Live System with instant recall of multiple RFX and MIDI setups-your E4 Ultra becomes the perfect centerpiece of your live and studio rig for mixing, effects and samples
- MIDI synchronization of FX parameters
- Support for new 24-bit analog and ADAT I/O cards

6860
128 Voice Polyphony Upgrade

This kit allows you to upgrade your 64 voice E4 Ultra to 128 voices of polyphony (standard on the E4XT Ultra).

6861
16 Out/ 8 In A-DAT

This kit allows you to connect 16 channels of ADAT outputs and 8 channels of ADAT inputs.

6862
D-WAM (AES/EBU, Word Clock I/O, ASCII, additional 16 MIDI)

This daughter card upgrade for the E6400 allows you to add digital I/O, Word Clock I/O, ASCII and an additional 16 channels of MIDI to your E6400 Ultra (standard on E4XT Ultra and E-Synth Ultra).

6863
EDI (E-MU Digital Interface)

This kit allows you to add a multi-channel EDI interface to connect your E4 Ultra to EDI equipped E-MU products.

6864
R-Chip Multi-channel Studio Effects Processor

This kit allows you to add the revolutionary, multi-channel R-Chip Studio Effects Processor to your E4 Ultra.

6870
E-Synth 16 MB Sound ROM

This kit provides your E4 Ultra with 16 MB of instantly accessible samples and 500 presets that are optimized for the E4 Ultra's synthesis capabilities. The E-Synth Sound ROM features a diverse blend of both standard instruments and digital waveforms to provide you with an array of useful and cutting-edge sounds (standard on E-Synth Ultra).

6871
Orbit/ Phatt Sessions 16 MB Sound ROM

This kit is similar to the E-Synth Sound ROM with 500 funky/electronica presets, and features all of the samples from E-MU's famous Orbit and Planet Phatt sound modules.

6876
16 MB Flash ROM

This kit allows you to create your own custom 16 MB sound -ROMS that never require loading and provide you with the perfect way to take your sounds on the road. You can edit your flash -ROMS whenever needed, putting your most-used sounds at your fingertips.

6877
32 MB Flash ROM

This kit is similar to the 16 MB Flash ROM above, but offers you 32 MB of Flash memory for even more flexibility in storing your sounds.

6301
Hard Drive Mounting Kit

This kit allows you to mount an internal hard drive into your E4 Ultra (standard on the E4XT Ultra).

6313 (8 Analog Output Expander)

This kit enables you to expand your E4 Ultra's number of balanced 1/4″ outputs from 8 to 16.

E4XT Ultra

	E4XT Ultra	E6400 Ultra	E5000 Ultra	E5000 Ultra Turbo
Polyphony	128 voices	64 voices (expandable to 128)	64 voices (not expandable)	64 voices (not expandable)
RAM	64 MB (expandable to 128)	16 MB (expandable to 128)	4 MB (expandable to 128)	
Hard Drive	4.0 GB	Optional	Optional	3.2GB
MIDI Channels	32 channels	16 channels (expandable to 32)	16 channels (expandable to 32)	16 channels (expandable to 32)
RFX-32 Effects Mixing System	Optional	Optional	Optional	Optional
Stereo-Effects Processor	24-bit	24-bit	24-bit	24-bit
Balanced Analog Outputs	8 (expandable to 16)	8 (expandable to 16)	4 (expandable to 12)	4 (expandable to 12)
Word Clock & AES/EBU I/O	Standard	Optional (with Option 6862)	Optional (with Option 6862)	Optional (with Option 6862)
ADAT	Optional	Optional	Optional	Optional
RISC Processor	Ultra-fast 32-bit	Ultra-fast 32-bit	Ultra-fast 32-bit	Ultra-fast 32-bit
EOS 4.61 Software	Standard	Standard	Standard	Standard
Included CD--ROMS	9	9	2	2
16 MB Internal Sound ROM	Optional	Optional	Optional	Orbit/Phatt Sessions
6860 128 Voice Polyphony	Standard	Optional	Not Available	Not Available
6861 16 Out/8 In ADAT	Optional	Optional	Optional	Optional
6862 D-WAM	Standard	Optional	Optional	Optional
6865 2nd SCSI Port	Optional	Optional	Optional	Optional
6870 E-Synth 16MB Sound ROM	Optional	Optional	Optional	Optional
6871 Orbit/Phatt 16MB Sound ROM	Optional	Optional	Optional	Standard
6876 16MB Flash ROM	Optional	Optional	Optional	Optional
6301 (for SCSI drives)	Not Available	Optional	Optional	Optional
6303 (for IDE drives)	Standard	Optional	Optional	Optional
6313 8 Analog Output Expander	Optional	Optional	Optional	Optional
6864 RFX-32	Optional	Optional	Optional	Optional
6866 16 out/8 in 24-bit ADAT	Optional (requires RFX)	Optional (requires RFX)	Optional (requires RFX)	Optional (requires RFX)
6867 8 24-bit Output Expander	Optional (requires RFX)	Optional (requires RFX)	Optional (requires RFX)	Optional (requires RFX)
6868 4 24-bit Input Expander	Optional (requires RFX)	Optional (requires RFX)	Optional (requires RFX)	Optional (requires RFX)

Technical Specifications

Presets:	2000 (1000 Ram, 1000 Internal/Flash)
Layers:	32
Zones:	255
Crossfade/Switch zones:	128
Sound Memory:	expandable to 128 MB; 72-pin simms (RAM, Flash or ROM)
Data Encoding:	20-bits
Signal to Noise:	105db

Frequency Response:	20Hz-20kHz ±1db
THD + Noise:	.044%
Power Consumption:	<50 watts
Voltage Input:	100-250 VAC, 50/60Hz. auto select
Maximum Output Level:	+10 dBv
Output Sample Rate:	44.1Khz, 48Khz
Analog Inputs/Outputs	Balanced Stereo 1/4″
Digital Inputs/Outputs	AES/EBU (optional for E6400, E5000), Optional 16 out/ 8 in ADAT card

KORG

TRITON LE 61 • TRITON LE 76 • TRITON LE 88

Music Workstations





The TRITON Le inherits the main functionality of the "classic" TRITON music workstations within a streamlined and cost-effective package. There are three TRITON Le models available: the 61-key TRITON Le 61 and the 76-key TRITON Le 76 both featuring synth action keyboards; and the 88-key TRITON Le 88 which employs a piano weighted, graded hammer action. Featuring Korg's HI (Hyper Integrated) synthesis technology, TRITON Le provides 62-voice polyphony. The 61-key and 76-key TRITON Le models offer a robust 32MB of PCM ROM that provide numerous classic TRITON Programs and Combinations, along with a host of cutting-edge sounds geared toward electronic music. The TRITON Le 88 adds an additional 16MB piano-based ROM samples. The user-installable EXB-SMPL Sampling option adds two mic/line audio inputs, SCSI, and supports 16-bit 48kHz linear mono/stereo sampling along with the sophisticated sample waveform editing functions you'd expect to find in a dedicated sampler.

FEATURES

TRITON Sound Data

- 512 Programs (448 factory and 64 user) and 24 drum kits are available in internal memory. 128 programs and 9 drum kits are also included for GM compatibility.
- Programs and drum kits are derived from 32 MegaBytes of internal PCM ROM using Korg's HI (Hyper Integrated) synthesis system and high-quality 48 kHz sampling. These samples are divided among 425 multisamples and 413 drum samples and cover a broad range of instruments from piano, guitar, brass, strings and tuned percussion to cutting modern synth sounds. The Triton Le 88 uses 48 Mbyte of PCM ROM and 429 multisounds.
- There are also 384 Combis available each comprised of up to eight Programs that can be layered, velocity layered and split. Combinations are a great way of creating rich textures and for configuring complex live performance setups.
- 62 note polyphony in single oscillator mode and 32 note polyphony in double oscillator mode.

Studio Quality Effects

- There are a total of four independent studio quality effects processors available, based on 89 effects types, including one stereo insert effect, two master effects and a master 3-band EQ.
- Effects types include: filter and dynamics effects, such as EQ and compressor; Pitch and phase modulation effects, such as chorus, phaser, rotary speaker and pitch shifter; early reflection, delay effects and reverbs; as well as a number of mono and mono-chain effects in which two mono effects are connected in series.
- Each effect is fully editable and many of the effect's parameters can be controlled in realtime.
- The BPM/MIDI SYNC function allows you to synchronize the modulation/delay time of effects to the internal or external tempo. You can specify each delay time as a note value, easily setting up rhythms that will follow any changes that occur in the tempo of the arpeggiator, sequencer or external MIDI clock.

Full-featured 16-track Sequencer

- The 16 track / 16 part multitimbral sequencer features a 200,000 note/event, 200 song capacity and offers the flexibility to record in real or step time.
- The Cue List allows you to assemble your song from separate elements (Intro, Verse, Chorus, Bridge, etc.), and even convert them back into a single song for further recording or editing.
- 150 preset drum patterns and 100 user patterns are provided.
- Track Looping lets you play back specified measures of a track as a loop.
- The Template Song function provides 16 presets and 16 user template memories containing program and effect setups appropriate for various styles of music.
- Sequence data can be saved as SMF format 0 or 1, making it simple for you to exchange data with other sequencers.

TRITON LE

Dual Arpeggiator

- Two powerful polyphonic arpeggiators can be used to create intricate and expressive pitched patterns or drum grooves. One arpeggiator can be assigned within a Program, and two arpeggiators can be assigned in Combi and Sequencer modes. The arpeggiators can be synchronized to MIDI clock, internal clock and can be recorded into the internal sequencer.
- There are five Preset patterns (UP, DOWN, ALT1 ALT2, RANDOM) available as well as 216 User patterns.

RPPR (Realtime Pattern Play/Recording) Function

- The RPPR function allows you to trigger up to 74 different sequences directly from individual keys on the keyboard.

Four Outputs

- The sound from each oscillator, drum, timbre/track, or insertion effect can be freely assigned to the 1/4˝ L/MONO and R main stereo audio outputs or two additional individual audio outputs.
- MIDI In, Out and Thru connectors are provided for interfacing with external synths, samplers, sequencers, etc..

User Interface

- The AUDITION key allows you to preview Programs using a selection of musically-appropriate riffs while the CATEGORY key allows you to find Programs and Combis quickly and easily as well as selecting oscillator waveforms and effect algorithms.
- Large, backlit 240 x 64 pixel LCD display



Control

- A joystick allows expressive pitchbend and modulation control.
- Two assignable switches and four realtime control knobs allows you to apply modulation to the LFO and filters etc., control effect parameters or to control the arpeggiators.
- Three pedal inputs include: a dedicated damper pedal; a foot pedal input that can be assigned to continuous controller functions such as volume; and a foot switch input that can be assigned to a number of functions such as soft pedal and arpeggiator on/off.

HI Synthesis

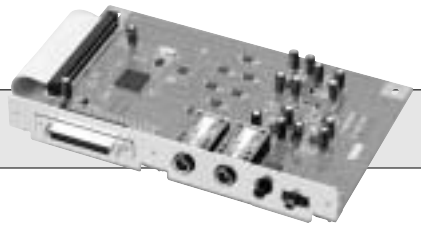
- HI (Hyper Integrated) synthesis is Korg's system for manipulating PCM sound data. By filtering, shaping and modulating the Triton's high-quality PCM data and then processing the result in the powered-up effect section, Triton can create richly expressive and nuanced sounds.
- Internal PCM samples or your own custom samples can be assigned to an oscillator, and processed via the filter, EG's and LFO sections.



- Each oscillator has access to multiple filter types, including resonant, hi and low-pass, and two LFO's with a choice of 21 different LFO waveform. LFO Frequency can be linked to the sequencer or arpeggiator using MIDI or TempoSync, so that the LFO speed will follow any changes in tempo. There are also independent Pitch, Filter and Amplifier envelopes available.
- The HI synthesis system also lets you create sounds that have unique effects by using 42 AMS (alternate modulation) sources to apply modulation to any of 55 destinations

Memory Back-Up

- Internal memory (Program, Combi, Sequence and sample data) can be backed up onto SmartMedia cards (3.3V, 4MB - 128MB) or external fixed and removable drives via the optional, user-installable EXB-SMPL Sampling/SCSI upgrade board.
- SmartMedia cards can also be used to exchange data with your computer via the readily available adapters commonly used with digital cameras.



USER-INSTALLABLE EXB-SMPL SAMPLING UPGRADE

- The sampling upgrade supports 48-kHz 16-bit linear mono/stereo sampling and adds two mic/line switchable inputs with a variable gain knob as well as a SCSI interface.
- The EXB-SMPL Board comes standard with 16MB of SIMM memory, which allows about 2 min / 54 secs of monaural sampling, and can be expanded to a total of 64-MB using two 32MB Fast Page or EDO-compatible 72-pin SIMM modules with an access speed of 60 nsec or faster.
- Sophisticated waveform editing functions include Crossfade Loop, Truncate, Normalize, Time Slice, Time Stretch and Sample Rate Convert.
- The SCSI interface is compatible with a wide range of storage media and will allow you to access CD-ROM sound libraries including AIFF, WAVE, AKAI S1000/3000 (samples and multisamples) as well as Korg format data.



KORG

TRITON • TRITON PRO

Music Workstation / Sampler





The classic 61-key TRITON and 76-key TRITON Pro represent where the legendary TRITON series began. Employing Korg's HI (Hyper Integrated) synthesis system along with powerful sequencing, onboard sampling, built-in effects and a wide selection of user installable options, the classic TRITON series is still on the cutting of music workstation technology. Both TRITON models contain 32MB of internal PCM waveform ROM, 425 multisamples and 413 drum samples, that range from edgy dance/techno synths to real-world instruments. Both TRITONs come with the EXB-SMPL sampling board already installed and includes 16MB of RAM that can be easily upgraded to 64MB using standard 72-pin SIMMs modules. Both models also feature Korg's intuitive TouchView Graphical User Interface that will have you zipping around the TRITON's extensive programming, sequencing and performance capabilities with style and grace. The TRITONs also add an assignable ribbon controller allowing even greater expressiveness to these already inspiring synths.

All the features of the TRITON LE Plus—

- The internal memory is re-writable and contains 640 Programs, 512 Combinations and 64 drum kits.
- 256 programs and 9 drum kits are also provided for complete compatibility with the latest GM Level 2 specification.
- Floppy disks or an external fixed or removable SCSI drive (optional EXB-SCSI interface board required) can be used to back up this data, or to create your own data library.

Outputs

- Sounds can be freely assigned to the 1/4″ stereo outputs or to any of the four 1/4″ individual outputs.
- External audio sources can be routed through the sampling input and processed using insertion and master effects.

Effects Section

- The Effects section consists of five stereo insert effects, two master effects, a stereo three-band master EQ and a mixer that controls the routing of these effects
- Insertion effects can use any of the 102 available types, and the master effects can select from 89 types.
- In addition to standard effects like reverbs, chorus and delays, Triton has unusual effects such as stereo random filter and ring modulator.
- A vocoder effect is also available that can control internal sounds via the sampling input.



TouchView GUI

- All functions and parameters are displayed in a large and highly-visible 320 x 240 pixel screen using Korg's original TouchView Graphical User Interface that lets you perform operations smoothly just by touching the screen.
- A ribbon controller and an assignable slider and two assignable pedal inputs.

TRITON RACK

Expandable Sound Module / Sampler



The TRITON-Rack is a two rack space module that features the same high quality sound, effects and sampling functionality of the TRITON Workstation/Samplers with a few enhancements including an increased number of program and combination locations along with significantly more expansion and digital I/O possibilities. In Multi mode the TRITON-Rack functions as a 16-part multi-timbral tone generator with 60 voices of polyphony and offers room for up to 2,057 programs and 1,664 combinations of up to eight timbres each. To navigate this vast array of sounds, the TRITON-Rack features an Audition Riff function that can playback 382 different riffs tailored to various types of sounds. The TRITON-Rack provides room for up to eight PCM expansion boards, allowing you to expand the PCM wave memory from a very respectable 32MB to a whopping 160MB. In addition, up to 96MB of sample memory may be added via three SIMM slots. The TRITON-Rack also supports the SCSI and MOSS options that are currently available for the TRITON keyboards. The TRITON-Rack provides 60-voices of polyphony and functions as a 16-part multi-timbral tone generator that can play back tracks from an external sequencer or from the internal playback-only sequencer.

FEATURES

- The same powerful 48 kHz 16-bit linear mono/stereo sampler section as the TRITON keyboards, plus the ability to export samples as .WAV and AIFF files for editing samples on computer systems or use on other products, such as Korg's ELECTRIBE-S.
- Room for up to eight PCM expansion boards, each with their own dedicated bank for programs and combinations.

Sampling

- 48 kHz 16-bit linear mono/stereo sampling with visual waveform editing, Time Slice, Time Stretch, Crossfade Loop, and more.
- 16 MB of sample memory is included, expandable to up to 96MB using three 32 MB 72-pin SIMMs.
- Export samples as .WAV and AIFF files for editing samples on computer systems or for use in other products, such as Korg's ELECTRIBE-S.
- The optional EXB-SCSI SCSI Interface Board will allow you to load and save data on an external SCSI device, and load AKAI sample data from CD-ROM.

Multi Mode

- You can edit and store up to 200 Multis consisting of up to 16 programs with transposition settings along with five insert and two master effects.
- You can use Multis for playing back tracks from an external sequencer or the internal playback-only sequencer which supports TRITON and Standard MIDI File (Format 0 and 1) sequence data.
- Time-sliced pattern data, from the sampler, can be saved as MIDI data (SMF file) and loaded into an external sequencer.

Arpeggiator

- The dual polyphonic arpeggiator provides access to 5 preset and 328 User patterns – 200 patterns are preloaded.

User Interface

- 240 x 64 dot backlit LCD display
- Four realtime controller knobs (with 3-step selector), Arpeggiator ON/OFF key.
- 3.5 inch, 2DD/2HD compatible Floppy Disk Drive

Expanded TRITON Stereo Digital Multi-effects System

- Access up to two master effects (mono in, stereo out), 5 insert effects (stereo in / out), 1 master EQ (3-band stereo) simultaneously per multi.
- Choose from 102 insert and master effects types

Inputs and Outputs

- Six 1/4″ outputs (two main and four individual), plus S/PDIF optical digital I/O included standard.
- 1/4″ stereo headphones outputs with level control for the Main and Headphones.
- Two 1/4″ mic / line switchable inputs with volume control for sampling.
- MIDI In, Out and Thru connectors

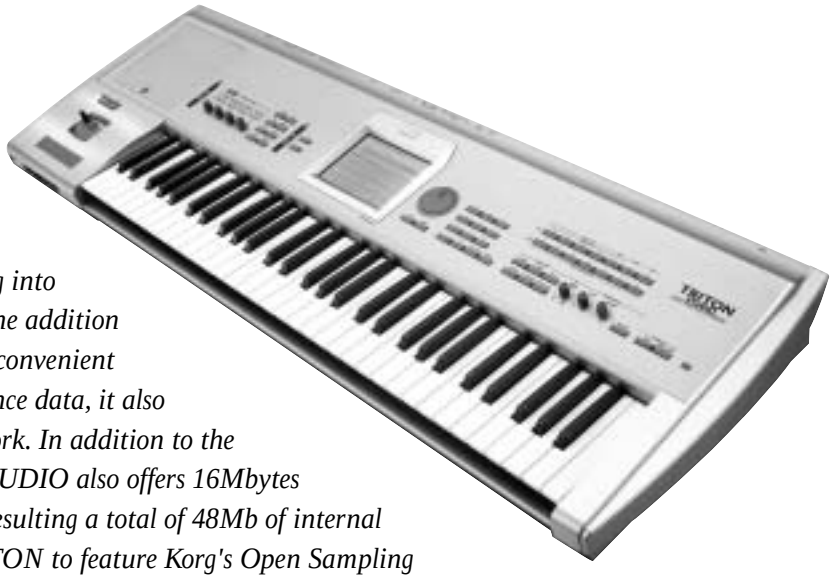
I/O Options

- EXB-DI Digital Interface Board – 6-channel ADAT optical output
- EXB-mLAN mLAN FireWire audio/MIDI interface

KORG

TRITON STUDIO

Workstation / Sampler



TRITON STUDIO is Korg's most advanced and expandable music workstation ever. Available in 61 and 76-key synth action versions, as well as an 88-key weighted action model, each TRITON STUDIO combines Korg's HI (Hyper Integrated) synthesis technology, effects, sampling and sequencing into what can only be described as a sonic powerhouse. The addition of an internal CD-R/W burner not only facilitates a convenient means of backing up and loading samples and sequence data, it also allows you to create an audio CD of your finished work. In addition to the original "classic" TRITON PCM ROM, TRITON STUDIO also offers 16Mbytes of ROM devoted to a stereo acoustic grand piano – resulting a total of 48Mb of internal PCM ROM. TRITON STUDIO is also the first TRITON to feature Korg's Open Sampling System, which allows direct access to sampling and resampling functionality from within the Program, Combination or Sequencing modes, as well as from the Sampling Mode.

All the features of the TRITON Plus—

- Three models are available: 61/76 note synth action and 88 note weighted action keyboards using Korg's RH2 graded hammer action.
- Ultra-powerful processor with 6 times the power of the Triton and 2 times the Karma.
- Expandable to a total 256 MB of waveform data:
  - ➢ 48 MB of PCM ROM onboard, with 429 multisamples and 417 drum samples
  - ➢ Includes a 16 MB stereo acoustic piano with two levels of velocity switching.
  - ➢ Room for up to seven PCM expansion boards
- There are 1,536 Programs (512 preloaded) and 1,536 Combinations (512 preloaded) available in user writable memory locations as well as 144 user drum Program locations (20 preloaded) — Open memory locations can be used for sounds that you edit/create, load from expansion board packages or get from other sources
- An additional 256 sounds and 9 drum kits are provided for GM2 compatibility.
- 120 note polyphony
- Expandable to up to 96 MB of sample memory (16 MB included standard)

Re-Sampling

- Resampling is possible in Program, Combination and Sequencer modes using the full functionality of each mode (filters, effects, arpeggiator, sequencer, etc.): Sample keyboard phrases in realtime; or bounce an entire mix from the 16 track sequencer to a two track .Wav file.
- The In-Track sampling function allows you sample external audio sources directly into the sequencer while your track is playing.
- Samples recorded to hard disk can be loaded into sample memory for editing and use in Programs and drumkits.



Create Audio CDs

- An optional internal CDW/R drive is available for burning audio or data CDs as well as ripping samples from audio CDs.
- A 5 GB internal hard drive is provided for backing up samples
- The internal hard drive is also available as a two track HD recorder that allows you to record live audio, or sequenced tracks onto two dedicated tracks. Audio can be recorded through onboard effects, and a number of editing functions are available.

Enhanced TouchView Graphic Interface

- The enhanced, easy-to-read 320 x 240 pixel TouchView graphical user interface is faster than ever and provides visual waveform editing for samples, a clean mixing layout for sequencing and a logical, category-based system for finding sounds, effects and even waveform data.
- 24-bit optical (S/PDIF) I/O is provided with support for both 48 kHz and 96 kHz audio sources.

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KARMA

Music Workstation with Variable Performance Modeling



The KARMA is a totally unique Music Workstation that combines the sounds, effects and sequencing capabilities of Korg's award-winning TRITON with a powerful realtime performance modeling technology called KARMA. The result is a workstation that provides an unprecedented amount of user-interaction that will allow you to reach, what was once considered, unobtainable expressive control of your musical ideas. KARMA stands for Kay Algorithmic Realtime Music Architecture and is the invention of musician/programmer Stephen Kay, who has worked closely with Korg for the past seven years to integrate and further enhance KARMA's functionality. Based on the notes and chords that are played, KARMA will automatically generate and manipulate phrases and patterns that are impossible to produce with conventional arpeggiators or static pattern playback functions. KARMA can produce anything from unhuman electronic rhythms and melodic textures to acoustic emulations of harp glissandos and finger-picking and provides control over almost 400 musical elements that allow you to interact with the phrases intuitively, and to change them in ways that can range from subtle to extreme.



FEATURES

KARMA Function (Variable Performance Modeler)

- Karma (Kay Algorithmic Realtime Music Architecture) is a patented algorithmic MIDI data generating technology that takes input notes, chords and controller movements to generate complex and sophisticated musical phrases and effects.
- Phrases and patterns are produced by a GE (Generated Effect). Each GE consist of nearly 400 parameters, such as note data, rhythm development, chord structure and velocity. Of these, 16 parameters are available for realtime control.
- There are over 1000 types of phrase or pattern available covering a wide range of instruments, playing styles and musical genres: Create swirling cascades of complex interweaving notes, tricked-out techno arpeggios, dense rhythmic and melodic textures, natural sounding glissandos for acoustic instrument programs, guitar strumming and finger-picking simulations, to name a few. And because phrases and patterns can evolve in response to realtime controller movements, your performance will never sound "canned" or static.

Realtime Control

- 61 note, synth action keyboard, Karma Realtime control knobs 1-8, plus Joystick & latching controller switches.
- Alternate Modulation and Effect Dynamic Modulation functions allow you to freely assign modulation to parameters that affect the pitch, filter, amp, EG, LFO, and effects etc.

HI Synthesis System

- Includes the same HI synthesis system and 32MB of PCM wave ROM with 425 multisamples and 413 drum samples as the classic TRITON.
- 640 user Programs and full General Midi level 2 with 256 Programs and 9 Drumkits. 768 user programmable Combi sounds and 64 User Drumkits.
- 5 Stereo Insert and 2 Stereo Master Effects, with 102 different effect types available and a Master EQ section.
- The 16 Track Sequencer has the same capabilities as the classic TRITON with the added ability to capture the performance of KARMA phrases and patterns.

KARMA MW Software

- KARMA MW software allows PC and Mac users to access all internal parameters of existing GE's as well as make new ones via their computer. Kamra MW also offers the ability to import SMF phrases and grooves (from any SMF source) as well as TRITON Arpeggiator data and convert them into GE's.

Connectors

- Left and right 1/4″ main outputs plus two 1/4″ individual outs.

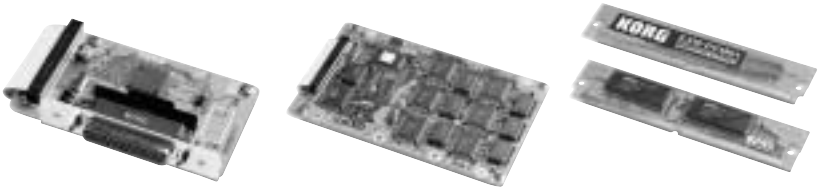
Options

- Up to two TRITON series PCM expansion boards can be installed, for a total of 64 Mbytes of waveform data. The six-voice MOSS DSP Synthesizer Board can also be installed (EXB-MOSS), allowing you to use both DSP and PCM synthesis technologies without connecting external equipment.

KORG

TRITON ACCESSORIES

Korg offers a variety of user-installable ROM and I/O options for enhancing your TRITON Music Workstation including a number of 16 MB PCM data boards, each consisting of high-quality PCM data and 128 programs and 128 combinations.





EXB-PCM01
Pianos / Classic Keyboards
- 128 Programs and 128 Combis, plus 16MB of sample data focusing on piano, electric piano, organs, harpsichord and other keyboard classics.



EXB-PCM02
Studio Essentials
16 MB of sample data delivering strings, brass, solo instruments, guitar and even gospel choir. 128 Programs and 128 Combis.



EXB-PCM03
Future Loop Construction
16MB of time sliced sampled grooves can be played back at any pitch or tempo. 728 RPPR grooves arranged into 18 song templates.



EXB-PCM04 Dance Extreme
Modern contemporary sounds enhance many dance genres. 16 MB highlighted by tons o' drums; Synthesizer leads, pads and basses; plus guitar and choir hits. 128 Programs & Combis



EXB-PCM05 Vintage Archives
16 MB from the glory days of analog synth provide rare classics treasures – including beat boxes, vocoders, tape-trons and more. 128 Programs & Combis



EXB-PCM06 / 07
Orchestral Collection
A 32 MB, two-board set covering every instrument and element needed to re-create a symphony orchestra.



EXB-PCM08
Concert Grand Piano
16 MB devoted to faithfully capturing the Concert Grand. 32 Programs & 64 Combis provide a variety of tonal shadings.



EXB-PCM08
Trance Attack
16 MB devoted to faithfully capturing the Concert Grand. 32 Programs & 64 Combis provide a variety of tonal shadings.



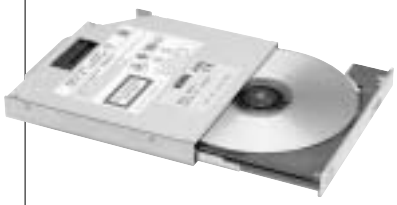
DSP Modeling Synthesizer (EXB-MOSS)
Add Korg's Multi-Oscillator Synthesis System technology to the TRITON STUDIO for more synthesis capabilities



ADAT Digital Output (EXB-DI)
Triton Studio's six channels of audio output can be sent in 24-bit 48kHz optical format. Also provides a Word Clock In for Synchronization.



mLAN Interface (EXB-mLAN)
Using a FireWire connector, the Triton Studio can be connected to other mLAN equipped devices for transmission of MIDI and digital audio data.



CDRW-1x8
CD-R/RW Rewritable CD Drive for TRITON Studio
For burning audio or data CDs and audio ripping.

TRITON SERIES COMPARISON

	TRITON Le	Karma	TRITON	TRITON Rack	TRITON STUDIO
Polyphony	62	62	62	62	120
Processor speed	1x	3x	1x	1x	6x
Waveform memory	32	32	32	32	48
# of multisamples	425	425	425	425	429
# of drumsamples	413	413	413	413	417
PCM expansion	-	2 slots	2 slots	8 slots	7 slots
Total Program Locations	512	640	512	1664	1536
Preload Programs	448	384	128	512	512
User drumkits / Preload drumkits	24 / 16 preload	64 / 16 preload	64 / 16 preload	144 / 16 preload	144 /20 preload
GM2 Programs / GM2 Drum Kits	128 / 9	256 / 9	256 / 9	256 / 9	256 / 9
Additional Programs w/ EXB-MOSS	-	128	128	128	128
Total Combination Locations	384	768	512	1664	1536
Preload Combinations	384	384	512	512	512
Insert Effect Processors	1	5	5	5	5
# of Insert Effects	89	102	102	102	102
Master Effect Processors	2	2	2	2	2
# of Master Effects	89	89	89	89	89
Master EQ	3-band	3-band	3-band	3-band	3-band
Sequencer	16 track 200,000 note	16 track 200,000 note	16 track 200,000 note	Pattern sequencer 140,000	16 track 200,000 note
Sampling	Optional EXB-SMPL	-	standard	standard	standard
Resampling	-	-	-	-	yes
In-Track Sampling	-	-	-	-	yes
Maximum Sample RAM / SIMM slots	64 MB / 2	-	64 MB / 2	96 MB / 3	96 MB / 3
Sample to HD	-	-	-	-	Yes
Burn audio CD	-	-	-	-	Yes
Arpeggiator	Dual Polyphonic	KARMA GE System	Dual Polyphonic	Dual Polyphonic	Dual Polyphonic
Arp patterns: Preset/User/Preload	5 / 216User (200 preload)	1190 GE (V2 +300 User)	5 / 232 User (180 preload)	5 / 328 User (200 preload)	5 / 507 User (367 preload)
Display	240 x 64 pixel	240 x 64 pixel	TouchView 320 x 240	240 x 64 pixel	TouchView 320 x 240
Ribbon	-	-	Yes	-	Yes
Assignable knobs	4	4	4	4	4
Assignable switches	2	2	2	-	2
# of Audio Outputs	4	4	6	6	6
# of inputs	2 with EXB-SMPL option	-	2	2	2
SCSI	part of EXB-SMPL option	-	EXB-SCSI option	EXB-SCSI option	standard
S/PDIF digital I/O	-	-	-	standard	standard
ADAT optical output	-	-	-	optional	optional
mLan digital I/O	-	-	-	optional	optional
EXB-MOSS	-	optional	optional	optional	optional
Floppy drive	-	Yes	Yes	Yes	Yes
Internal Hard Drive	-	-	-	-	5 GB
Smartmedia slot	Yes (128 MB)	-	-	-	-
CDR/W bay	-	-	-	-	Yes, optional CDRW-1x8
Keyboard type	Fatar	Fatar	Yamaha	-	Yamaha (Korg/Fatar f/88)
Keyboard sizes	61, 76	61	61, 76, 88	-	61, 76, 88
Power supply	external	external	internal	external	internal

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KORG

MS2000 & MS2000R

Analog Modeling Synthesizers





The MS2000 44-note velocity keyboard and the MS2000R 4U rack-mount are analog modeling synthesizers that use the latest DSP technology to recreate the coveted analog sounds and hands-on functionality of Korg's classic MS20 analog synth, VC-10 vocoder and SQ10 analog sequencer. All major sound creation parameters are available on the front panel, with 35 dedicated knobs providing the ability to create and manipulate sounds in real-time. MS voices are built around two multi-waveform oscillators and a noise generator and have access to four types of analog-style filters, dual ADSR envelopes and two MIDI-synchable LFOs along with dedicated volume, pan and distortion controls. The Korg MS2000 is housed in a futuristic metallic blue-green body, with the keyboard version echoing traditional analog instruments with rosewood side panels. The rack-mount MS2000R can also be used on a tabletop. Both have an angled front panel for excellent visibility and easy operation.

FEATURES

Hierarchy

- There are a total of 128 programs divided into 8 banks of 16 sounds.
- Each program can contain one or two timbres (4 voices total), effects and an arpeggitaor— If two timbres are used, they can be split, layered or accessed independently from different MIDI channels.
- Each timbre consists of Osc1, Osc2, and Noise followed by a Mixer that allows you to adjust the levels of each.
- The tone generator uses a DSP-powered analog modeling system based on the KORG Prophecy, Z1, and Electribe series and has been further refined to produce a wide range of sounds. From sharp, cutting leads to powerful and thick pads.
- After the mixer, signals are sent to the Filter, Amp, EGs (envelope), LFO, Virtual Patch, and a Mod Sequence.
- All major sound creation parameters are available on the front panel, with 35 dedicated knobs providing the ability to create and manipulate sounds in real-time.

OSC1/OSC2/NOISE

- OSC1 allows you to select from eight different oscillator algorithms, including basic analog synth waveforms such as SAW and PWM, Cross Modulation, and 64 DWGS (Digital Waveform Generator System) used on the Korg DW-8000 synthesizer.
- OSC2 allows you to select from three types of oscillator algorithms: SAW, SQU, and TRI. It can also be used as a modulator oscillator for the Sync or Ring modulation characteristic of analog synthesizers.
- You can also mix NOISE into the signal.
- There are a total of 70 possible waveforms which can be "wave-sequenced" using the Mod Sequencer.
- All of the analog-modeled waveforms include modulation/shaping to help recreate the sound of pulse width modulation as well as generating frequency modulation from the Cross Wave.
- Oscillator 1 can be linked to Oscillator 2 for Ring Mod and Sync timbres
- External audio signals can even be used as a waveform, and processed and modified with filter, effects, and parameters

Mixer

- The mixer section allows you adjust the levels of OSC1, OSC2 and NOISE, and send the combined signal to the Filter section

Filter Section

- Four filter types are available: 12 and 24 dB low-pass as well as band pass and high pass, each with variable resonance that can be adjusted into self-oscillation

AMP Section

- In addition to volume and pan controls for each timbre, the amp section provides a Distortion on/off setting so that even soft sounds can be transformed into aggressive solo parts.

Envelope / LFO Section

- Dual ADSR envelope generators and two four-waveform MIDI-synchable LFOs provide flexible sound shaping control

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MS2000 & MS2000R

Mod Sequence

- The Mod Sequence function allows you to apply time-variant changes to sound parameters in a way similar to varying the control voltage on an analog synthesizer.
- Each timbre can have up to three sequences with up to 16 steps each.
- Sixteen knobs on the front panel allow you to set the value for each of the 16 available steps. Knobs can be operated in realtime, and you can record the movements of the parameter values in each step using the Motion Rec function.
- In addition to the 28 parameters on the front panel, you can apply time-variant change to the pitch or step length.

Inputs and Outputs

- Left and Right 1/4″ line level outputs and a 1/4″ stereo headphone output.
- Two 1/4″ inputs are provided allowing you to use an external source as an waveform for the oscillator 1. Input 2 can be switched to accommodate mic or line level signals.
- MIDI In, Out and Thru connectors and two footswitch inputs (one switch and one pedal) are also provided.

16-band Vocoder

- A powerful 16-band, four voice vocoder modeled on the classic Korg VC-10 is also provided. You can use vocal, using a microphone connected to Audio In 2, or rhythm sounds as an input source.
- The vocoder uses two 16-bank filter sets to simulate the vocoder sounds. Original vocoder sounds can be created by shifting the filter frequency, or adjusting the level and pan of individual frequency bands.

Arpeggiator

- A MIDI-synchable arpeggiator is also provided with six different patterns (UP, DOWN, ALT1/2, Random and Trigger), along with adjustable gate and swing parameters.

User-Programmable Effect Section

- The programmable effects section provides each program with access to a modulation effect (phaser, chorus/flange and ensemble), three types of delay, and 2-band EQ.

Virtual Patch

- The Virtual Patch system, based on the patching capabilities of the legendary Korg PS series and MS series synthesizers, delivers an array of rich creative possibilities. It allows you to use the EGs, LFO, velocity or keyboard tracking as modulation sources to affect parameters such as pitch, noise level, and pan.
- You can select up to eight sources and destinations directly from the front panel.



Classic Korg Instruments to Return in Virtual Form

Korg has a long history in the world of synthesizers and they will now be bringing many of the classic, discontinued instruments back to life for digital studios in the form of virtual instruments - the ***Korg Legacy Collection****. Korg uses Component Modeling Technology (CMT), to model the components that make up the signal chains of the instruments and how they interact in the original hardware instruments.*

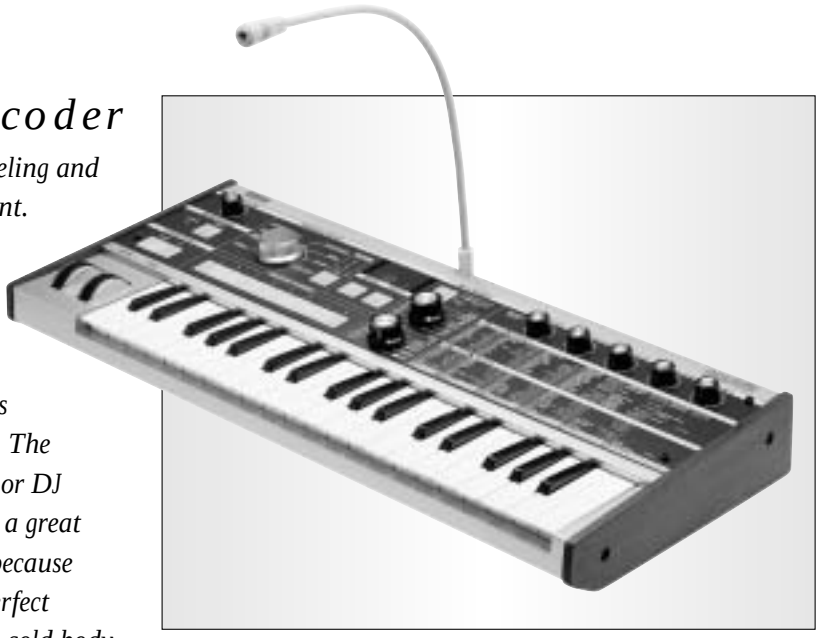
Currently, Korg has only confirmed that they will be producing virtual instrument versions for three of their synths: the ***MS-20****,* ***Polysix****, and* ***Wavestation****.*

Others Korg models are expected to be recreated as well, however.

The plug-ins will be available for both Windows and Mac OS (VST and AudioUnit formats). Release is expected in the third quarter of this year (2003). More details will likely be released at the Summer NAMM show in July.

Korg has created a special preview section on their web site, currently offering a video summary with an audio track produced with preliminary versions of the Legacy Collection instruments.

microKORG



Analog Modeled Synth/Vocoder

The microKORG delivers state-of-the-art analog modeling and multi-band vocoding in a compact, portable instrument. With 37 keys and 128 user-rewritable programs, the microKORG Synthesizer/Vocoder is perfect for any performer, producer, computer musician or beginner looking for an affordable synthesizer. The microKORG offers you the quality sounds and features you expect from Korg at a price that will astound you. The microKORG's compact size is perfect for the musician or DJ looking to add a portable synth to their setup. It's also a great choice for use in a computer-based music setup. And because the microKORG will run on battery power, it is the perfect choice for using with your laptop computer. The beige-gold body, wooden side panels and retro-influenced design, the microKORG has a look that's both vintage and funky.

FEATURES

Powerful Synthesis

- 128 great-sounding presets, including 16 vocoder programs are available covering a wide range of contemporary and vintage musical styles.
- The same dual-oscillator DSP synthesis engine, arpeggiator, and effects found in Korg's critically acclaimed MS2000 and offers a wider selection of waveforms than any other modeled synth.
- Multi-Mode resonant filter, two ADSR envelope generators and two MIDI-synchable LFOs.
- The Virtual Patch matrix allows you to create advanced modulation settings, providing an unprecedented level of control over synth parameters at this price range: route the Mod Wheel to control filter cutoff or pulse width; use an LFO to control panning or amp level, etc.

Classic Vocoding

- An 8-band vocoder including its own microphone.
- The voice formant capture function allows you to "freeze" the formants of your voice, and then play it across the keyboard, or shift the formant frequency to make your voice sound male, female, or childlike, etc.

An Ideal Compact MIDI Controller

- 37-note velocity-sensitive mini-keyboard with Pitch Bend and Modulation wheels.
- All editing parameters are located on the panel for easy access to any function.
- A large dial and LED illuminated buttons provide easy program selection even on a dark stage.
- Five realtime knobs on the front panel that not only provide flexible realtime control over filter cutoff, attack, release and other important parameters. The knobs can be used to transmit MIDI CC messages for controlling both hardware and software synthesizers.



Audio and MIDI I/O

- Two external audio input jacks permit you to run other instruments, CDs and more through the microKORG's filter and effects. This allows you to process sounds in creative ways.
- 1/4″ left and right outputs as well as a 1/4″ stereo headphone output.
- MIDI In, Out and Thru connectors



KP-2 KAOSS PAD

Dynamic Effect/Controller



The KP-2 is Korg's second generation KAOSS PAD, an innovative sound manipulation tool hat has become a must-have item for the cutting-edge DJ and remix artist. Using the concept of "the effect processor as musical instrument", the KP-2 allow real-time control over digital effects processing for any audio source (records, CDs, samplers, keyboards, voice). The KP-2 puts 100 killer effects, along with synthesizer and sampling functions, quite literally, at your finger tips – an unique X-Y Touch Pad allows you to control effects parameters with the touch (or sweep) of a finger. A wide variety of functions are provided to augment the X-Y pads realtime capabilities including; input mute (Transform), Pad Notion record/playback, effect Hold and more. Plus, MIDI I/O allows the KP-2 to control or be controlled from external sources. The KP-2's compact size allows it to be easily integrated in to your studio or DJ rig.

FEATURES

Realtime Control

- The Touch Pad controller provides unique and intuitive ways of manipulating effects using your fingertip.
- The Pad Motion function lets you record and playback complex finger movements on the touch pad.
- The Hold switch lets you freeze the effect setting at the current location of the X-Y pad. The effects depth knob allows you to adjust the exact mix of effects desired.
- A Mute function, aka "Transform" in DJ speak, allows you to cut off the incoming signal leaving the effect's "tail" or gate the input to create interesting rhythms.
- MIDI IN/OUT is provided, letting you control the KP2's effects via MIDI IN from an external sequencer.
- The color of the pad changes dramatically according to your finger position and the type of effect (delay or reverb, etc.), adding an even greater impact and visual feedback during performances.
- 100 different effect programs can be easily selected by turning a knob, and can be assigned to any of eight programmable buttons for immediate recall.

Effects

- 13 effects types including: Filter, Isolator, Distortion, Wah, Talking Modulator, Flanger, Vibrato, Tremolo, Slicer, Auto Pan, Phaser, Pitch Shift, Delay, Ping-Pong Delay, Tape Echo, Multi Tap Delay, Reverb, Gate Reverb, Reverse Delay, Ring Modulator, Broken Modulator.

Sampling

- You can sample any incoming audio (up to 6 seconds each at 44.1 kHz) onto two sample playback keys.
- You can sample through the effects, play a sample through a new effect, or even resample a sound through effects.
- Samples can be modified using a variety of effects (such as time stretch, reverse playback, loop time modulation and pitch change). This is ideal for integrating sampled effects into a live performance situation or create sound material for further sampling.

Synthesis

- A wide variety of cool analog modeling synth and drum sounds, drum patterns, slicer, vocoder and other essential DJ effects are also provided.

BPM Control

- The BPM function allows effects to be synchronized to the tempo of the music.
- The Auto BPM function can detect the tempo from an incoming audio source or MIDI Clock.
- BPM can also be set using the Tap tempo button.

Inputs and Outputs

- Independent stereo RCA Line and Phono inputs and a 1/4″ Mic input
- Stereo RCA line outputs
- 1/4″ TRS stereo headphone output with volume control
- 20 bit linear A-to-D and D-to-A converters
- MIDI In and Out connectors allow you to use the KP-2 as both a MIDI controller for synths, samplers, effects or other external devices and as a unique, MIDI controlled effects processor/sampler for sequencing and remixing applications.
- Includes DC12V AC adapter and Touch panel protection sheet.
- The compact size (8.03″ W x 8.15″ D x 1.89″ H)

KORG

ELECTRIBE–A

Analog Modeling Synth



The ELECTRIBE·A (EA-1) is a compact tabletop analog modeling synth with an onboard sequencer and a host of realtime controllers designed to allow you to create and record phat analog synthesizer, bass, and sound effect sounds. Operation is easy and intuitive because the results of tweaking any of the realtime knobs is reflected immediately in the sound. The sounds you create on this compact synthesizer can be input to the internal sequencer to create phrases. The ELECTRIBE·A's high quality sound is descendant from the DSP synthesis technologies used on the Z1 and Trinity V3 MOSS synthesizer engines, as well as the legendary Prophecy synthesizer platforms. Having produced classic analog synthesizers such as the "Mini Korg 700," the "PS series," and the "MS series," Korg has a long history of analog sound creation experience. And with that heritage, Korg's ELECTRIBE·A provides the ultimate in analog modeling to give you the richest, most versatile analog-style sounds that are sure to inspire today's electronic dance music artists- from DJs to remixers to producers.

FEATURES

- Two tone generators (or parts) based on the analog modeling technology made famous on Korg's "Prophecy" and "Z1" synthesizers generate the warm, fat tone and rich distortion that make analog sounds stand out.
- Each of the two parts has access to two VCOs (Voltage Controlled Oscillator) with a choice of Saw, Triangle and Square waverforms, and a VCF (Voltage Controlled Filter) with variable cutoff, resonance and envelope controls.
- Oscillators can be further modulated using Ring Modulation, oscillator Sync, or the Decimator effect.
- Realtime controllable distortion, delay, and chorus/flanger effects are provided for each part.

Motion Sequence Function

- You can control volume dynamics and filter cutoff of sounds or patterns in realtime using dedicated control knobs. These movements can be recorded into the sequencer allowing you to create continuously evolving variations to the sounds.

Step Sequencing

- The onboard sequencer uses a 16 step-grid interface, with 16 step keys, that allows you to create two-part sequenced patterns using the two onboard analog modeling tone generators.
- The pad keys can be played as a pitched keyboard in either step or real time. Pad keys light up so you can easily see what rhythm a part is programmed to play.
- Phrases can be stored as a pattern that can be from one to four measures long and include a chosen knob movement as well as an accent function.
- You can create up to 16 songs consisting of multiple patterns.
- You can also assign different patterns to each of the sixteen step keys allowing you to quickly switch between different patterns during live performance.
- Each step of the pattern can be set to pass or gate the audio input, so rhythm patterns can be programmed to synchronize the gating of the audio in time with the rest of the pattern.
- Tap Tempo allows you to synchronize with a turntable.
- There's room for creating 256 patterns, each with its own tones programmed as well as the sequences recorded.

Inputs and Outputs

- Left and right 1/4″ outputs and a 1/4″ stereo headphone output.
- A 1/4″ audio input allows you to process an external audio source through the resonant filter or effects knobs, and then sync them to your sequence. using the step-grid interface.
- MIDI In Out and Thru connectors allow you to use the Electribe units as tone generators. They also output note and controller information as well as MIDI clock for synchronization with other MIDI gear including other Electribe units.

ELECTRIBE–R

Analog Modeling Rhythm Synthesizer



*Korg has harnessed its 35 plus years of rhythm programming expertise to create the ultimate analog beat machine. The ELECTRIBE*R (ER-1) combines a four-part DSP-based synthesizer with four additional PCM sound sources, providing classic hi-hat and cymbal tones. Its synth voices includes dual oscillators, waveform modulation, amp and panning parameters. Effects include low boost for booming low-end, normal or tempo delay settings and the same Motion Sequencer as its sibling, the ELECTRIBE·A. The onboard sequencer allows you to program your own custom analog-style beats using the 16 trigger pads. The combination of DSP and sampled sound sources is what gives the ER-1 its unique character - the voice is much warmer and more modulatable than other sampled-only drum boxes. And with unlimited real-time control of every sound parameter, you know your sound will be uniquely your own.*

FEATURES

- Input two external audio sources and use the ELECTRIBE·R's synth parameters to process the sound.
- Input sequences into the ELECTRIBE·R and hear your results in realtime as you play the steps.
- Record Oscillator, amp or effect parameter knob movements in realtime using the "Motion sequence" function.
- Sync the ELECTRIBE·R to other MIDI devices using tap tempo or MIDI clock, or tap in the beat and sync to a turntable or other non-MIDI sound sources.
- Built-in low boost and delay effects
- Simply by tweaking the knobs you have unlimited control over all the elements that make up the sounds - like pitch, modulation depth, waveform shape and much more.
- Drop these sounds into the built-in sequencer and you've got unique rhythms with your own creative signature. And with the ELECTRIBE·R, you can hear every change you make as it happens, allowing you to mold the groove on the fly.
- You can store a total of 256 (64 x 4 banks) phrases or other patterns plus 16 songs consisting of multiple patterns, so all your beats are with you.

"Motion Sequence"

- Record the realtime movement of any of oscillator, amp or effect parameters as a Motion Sequence. Change sounds as your groove progresses in unique and expressive ways.
- Every sound or rhythm you create can be stored as a pattern that includes your knob movements. And you can save up to 256 1 to 4-bar patterns in internal memory.

External Control

- The ELECTRIBE·R can also be controlled from an external sequencer, keyboard, or sampler to function as an analog sound rhythm tone generator, or set up to control or be controlled by
- You can also control the ELECTRIBE·R from an external sequencer or keyboard and it becomes a MIDI expansion tone generator.

Four Modeling Synths In One

- Four analog modeling tone generators, based on Korg's renowned Prophecy and Z1 synths, allow the creation of four completely different kinds of rhythm sounds that can be used simultaneously.
- Four PCM (samples are also provided that cover essential crash, open hi-hat, closed hi-hat and clap sounds.

Step And Real-Time Input

- The built-in step sequencer's 16 pad keys make it easy for you to produce rhythms from the sounds you create.
- Each pad keys lights up so you can easily see what rhythm a part is programmed to play. You can also play in beats in real-time using the 16 trigger pads.
- Four of the pads are assignable to any analog modeling sound you come up with, two are reserved for controlling sounds accessed through the audio input, and the other four permanently control the indispensable PCM sounds. So you can tailor your own custom "kits" to fit each musical situation.

KORG

ELECTRIBE–S

Rhythm Production Sampler



Korg's ELECTRIBE•S (ES-1) combines the ELECTRIBE Series easy and intuitive realtime tweaking and sequencing functions with extensive sampling, resampling, signal processing capabilities. The ELECTRIBE•S was designed to make sampling straightforward: play it, sample it and then combine samples to create rhythm patterns and grooves. The ELECTRIBE•S has 95 seconds of sampling time (mono), and can store up to 100 samples in memory at a time. In addition, the ELECTRIBE-S features a broad range of editing capabilities, allowing you to create many unusual sounds and grooves. Sounds may be sampled through any of 11 different effects. With the realtime resample function you can apply pitch, filter effects to previously recorded samples and then resample the result. Sampled sounds can be assigned to as many as ten parts, and then sequenced in real or step time. Samples can be stored and exported to your computer via readily available SmartMedia cards.

FEATURES

- 95 seconds of mono sampling time with room to store up to 100 samples in memory at a time.
- Sounds can be sampled through any of the 11 different effects, including standard reverb, chorus/flange, and distortion as well as unique effects such as the Decimator, Isolator, and Resonant Filter.
- Effect parameters can be edited in realtime using dedicated controller knobs and can be switched on/off for each of the ten available parts
- A Master tempo-synchable Delay offers a wide range of realtime rhythmic possibilities.
- Samples can be played in reverse with the touch of a button, or you can trigger rolls to create complex rhythmic variations.

Time Slice

- The Time Slice function detects the attacks of each part within a phrase, creating individual elements that can be played back independently. This allows you the flexibility of rearranging rhythm patterns or modify the the pitch or filter values of individual slices without affecting the overall loop.

Motion Sequence Function

- The Motion Sequence function allows you to capture realtime knob movements of pitch, filter, level and pan parameters as well as effect on/off assignments and record them as part of the rhythm pattern.
- An Effect Motion Sequence is also provided so you can record changes in effect parameters. A Delay Motion Sequence can be used on the overall pattern.

Sequencing

- Rhythms can be created by assigning sampled sounds to as many as ten parts, and then triggering them in real or step time.
- The step sequencer can be used to edit the pattern via 16 step-keys, which control when each sound is to play within the measure.
- Patterns can be from 1-4 measures long, and there are 128 pattern locations available, including 64 preloaded patterns and 54 preloaded samples.
- Up to 256 patterns/steps can be combined to create a song, and then stored in one of 16 internal song memories.

Real-time ReSampling

- The real-time resample function allows you to apply effects to previously recorded samples and then resample the result without and loss in audio quality.
- Pitch, filter and effects knobs can be manipulated in realtime while resampling.
- Incoming audio can be gated in time with a pattern for rhythmic effects and the gated audio can then be run though the Master Delay for further processing.

Smart Media Sample Storage

- Samples, songs and patterns can be stored on 3.3 V Smart Media ranging from 4 MB to 64 MB. You can also use Smart Media to load AIFF and WAV files of sampled data stored on computer.

The Ins and Outs

- Mic/Line switchable 1/4″ stereo input with variable Gain adjustment knob.
- Left and right 1/4″ outputs, and a 1/4″ stereo headphone output.
- MIDI In, Out and Thru are provided for easy connection to other MIDI equipment, to create finished tracks. Power is supplied via 9V AC adapter.

ELECTRIBE–M

Music Production Section



The ELECTRIBE•M, the next step in the evolution of Korg's ELECTRIBE series. Now you can focus on creating scorching dance tracks with one easy-to-use production tool. With powerful sounds and the same motion sequence functionality featured on other ELECTRIBE models, the ELECTRIBE•M enables you to create an entire song with a single piece of gear. Plus, its easy operation and killer sounds help you to produce sophisticated music quickly and easily. With ELECTRIBE-M you have access to a huge array of sounds and phrases including 144 drum waves and 50 synth waveforms, covering the latest dance music styles such as Techno, Trance, House, 2step, Drum'n'Bass, R&B, and Big Beat, so that you can produce tracks quickly and easily. The Electribe•M features the classic 16-keypad, illuminated interface found on all Electribe dance tools. Also on-hand is Korg's unique "Motion Sequence function" that can memorize the movements of knobs used to vary the volume, panning or filter cutoff of a phrase you create. You can sync the ELECTRIBE-M with other models of the ELECTRIBE series simply by connecting a MIDI cable. You can even use the Electribe-M as a step sequencer for external tone generator modules, or connect it to a computer for running alongside another sequencing program or as a tone generator.

FEATURES

Dance Oriented Sounds and Phrases

- 144 drum waves include vintage kicks, snares, classic electronic tones, cymbals, hi-hats percussion and more.
- 50 synth waveforms cover everything from deep bass tones and vintage analog synth waves, to moving modulation timbres like sync and PWM, plus chord tones, various hits and sound effects.
- A variety of cutting edge patterns covering the latest dance music styles such as Techno, Trance, House, 2step, Drum'n'Bass, R&B, and Big Beat are also included giving you the leg up when producing new tracks.

Realtime Pattern Performance

- Realtime controller knobs and functions such as Part solo/mute let you produce a range of variations from a single pattern.
- Using the Pattern Set function, you can arrange up to 64 Patterns across the step keys, and then switch patterns in realtime by simply touching the keys.

Effects

- A variety of effects are provided for applying subtle to dramatic changes to the sound. Effects can be edited via dedicated knobs, and can be controlled by separate Motion Sequences.
- You can access up to two different effects simultaneously per Pattern to create a highly polished mix. Eleven different Insert effect choices are available, and you can choose to send parts individually to your selected effect or not.
- The eleven types of Insert effect include standards such as Reverb and Flanger, Chorus, as well as radical effects such as Decimator, Resonator, and Ring Modulator.
- A Master tempo-synchable Delay effect is also available.
- The Drive effect allows you to add a subtle presence to synth sounds to extreme and intense timbres when opened up and used in conjunction with filter resonance.

Step Sequencer

- The 16-keypad step sequencer, with its illuminated interface and intuitive visual feedback, allows you to create rhythm patterns and phrases with ease and efficiency.
- Each pattern can consist of up to eight drum parts and two synth parts allowing for extensive arrangement possibilities.
- Each pattern can be from 1-4 bars long, and each Song can hold a maximum of 256 patterns, and up to 35,700 events!

Motion Sequence Function

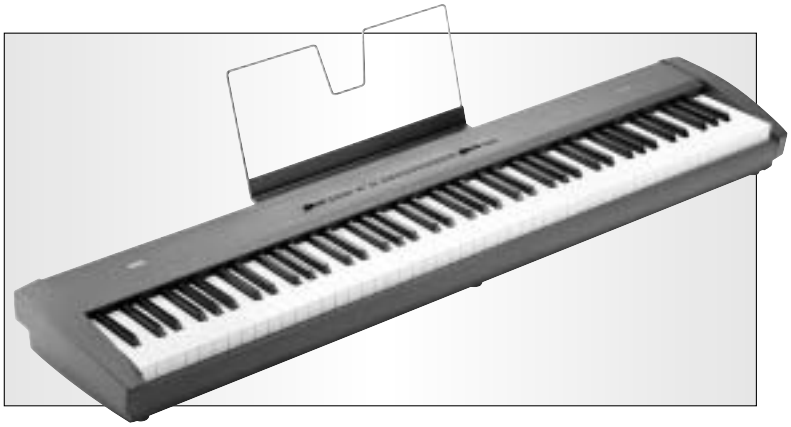
- The Motion Sequence function memorizes the movements of knobs used to manipulate the volume, panning or filter cutoff of a phrase you create. You can record up to two Motion Sequences per Drum Part, and up to three per Synth Part.

Inputs and Outputs

- Left and right 1/4″ outputs, and a 1/4″ stereo headphone output.
- MIDI In, Out and Thru connectors

KORG

SP-200/SP-300



88-Key Portable Digital Pianos

The SP-200 and SP-300 are slab-style stage pianos that are an answer for any pianist — from student to professional — looking for rich piano sounds in a convenient and portable package. They come equipped with an authentic-feeling 88-note hammer-action keyboard that faithfully reproduce the touch of an acoustic piano. Complementing this piano action are four full stereo and mono grand piano sounds. Three carefully tuned velocity-response curves are available so players can tailor the action of the SP-200 and SP-300 to their own playing styles. The SP-200 and SP-300 feature 30 sounds in total, including a number of Vintage 60's Electric Pianos, an Electric Grand, plus Harpsichord, Clavinet, Wah Clav, Vibraphone, Marimba, Acoustic Guitar, Jazz and Church Organs, Strings, and Choirs. All of the sounds can be played individually or in two-timbre layers. Reverb and chorus effects add spaciousness and depth to these sounds, recreating everything from a subtle chorus to a concert hall. Effects settings have been optimized for each individual sound, and the performer can choose from three levels of effect — Light, Normal, or Deep.

The Headphone output and built-in Metronome with adjustable tempo and time-signature are ideal for student use. For pro players, they provide mono and stereo outputs, and for those interested in period tunings — the addition of Kirnberger and Werckmeister Temperaments. They also offer a Transposition Function and up to 60-note polyphony. The SP-200 and SP-300 include a music stand and a sustain pedal, while the optional DS-1H pedal provides a Half-Pedal Sustain function that will meet the needs of even the most demanding pianists.



Otherwise identical, the SP-300 features a built-in amplification system consisting of a main amp and two 15-watt speakers. The SP-300 also features mono/stereo audio inputs to play external MIDI modules and other instruments through its internal speaker system. The SP-300 is housed in a sleek silver cabinet with dark cherry end pieces that fits into any décor. The optional ST-300 stand—specifically designed for use with the SP-300, has a dark cherry finish that matches the end pieces of the SP-300.

Keyboard:
88 notes, Velocity-sensitive, Hammer Action keyboard

Touch Control:
Light, Normal, Heavy

Sound Generation:
Stereo PCM System

Programs:
30 sounds in ROM

Polyphony:
60 notes maximum (the use of 2-oscillator sounds, layer mode and/or the effects reduces the maximum number of simultaneous notes)

Effects:
Reverb, Chorus (3 levels each)

Metronome:
Tempo, Time Signature, Accent and Volume controls

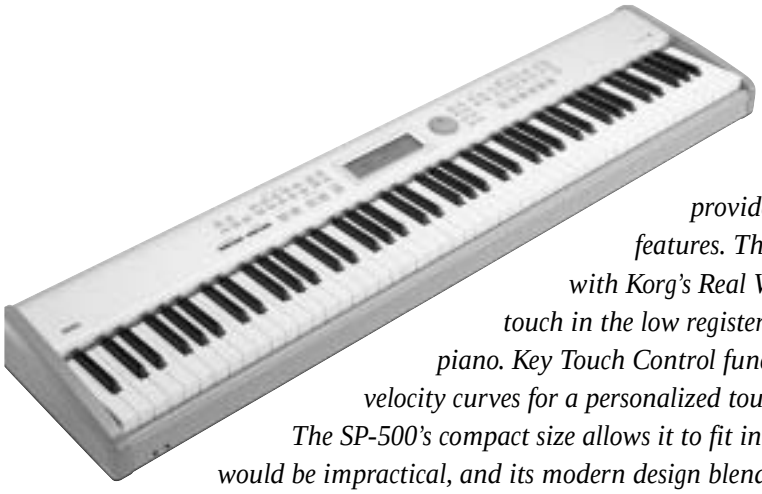
Tuning:
Transpose, Fine tuning, Temperament (Equal, Kirnberger, Werckmeister)

Pedal:
Damper (included). Half-pedaling supported with DS-1H (optional)

Connections:
Audio output
Headphones
MIDI
Damper

Controls:
Power switch
Master Volume
Reverb
Chorus
Touch
Sound selectors
Transpose Function
Tempo
Metronome
Demo

SP-500



88-Key Portable Digital Piano with 396 Sounds

Housed in a stylish wood and brushed metal cabinet, the SP-500 Digital Piano uses Korg's TouchView display to provide fingertip access to numerous piano and accompaniment features. The 88-key SP-500 is a "slab" style digital piano equipped with Korg's Real Weighted Hammer-Action 2 keyboard, providing a heavier touch in the low register and a lighter touch in the high register, just as on a real piano. Key Touch Control function allows the performer to select from six different velocity curves for a personalized touch.

The SP-500's compact size allows it to fit in places where a traditional piano or larger digital piano would be impractical, and its modern design blends well with any décor — from home, to church, to school.

FEATURES

- Highlighted by a stereo grand piano, the SP-500 is loaded with 396 sounds derived from Korg's professional synthesizers, and an ample 64 voices of polyphony. In addition to 19 different piano sounds, the SP-500 boasts an assortment of electric piano, organ, flute, sax, brass, strings, guitar, drum, and synthesizer sounds. Any two sounds can be layered together or split across the SP-500's keyboard.
- Two independent stereo effect processors offer up 37 different effect types, including reverb, delay, chorus, flanger, exciter, distortion, overdrive, phaser, rotary speaker, auto-pan, tremolo, and more. The amount of the effect can be easily adjusted to suit the user's taste.
- The SP-500 contains an advanced auto-accompaniment system with professionally produced styles that make it easy to create great-sounding ensemble performances. A wide range of Styles covering rock, pop, jazz, and other musical genres enable the SP-500 to deliver the full-sounding performance of a complete pro backup band with up to six musical parts.
- The Pianist Styles provide an accompaniment featuring piano-type left-hand passages that enable even novice players to sound like accomplished pianists.
- The sequencer features a full set of event editing and quantization tools. Realtime, step-entry and punch in/punch out record modes are also available.
- Comes loaded with 128 Rhythm Styles, 26 Pianist Styles, and room for 64 User Styles.
- GM compatible, the SP-500 can be connected to a computer or other MIDI-equipped devices for playing back song files, taking interactive music lessons, and much more.
- Advanced TouchView Graphical User Interface is a touch-panel LCD that lets you select or modify sounds and accompaniment styles with the touch of a finger.
- Eight-track internal sequencer can capture and record performances created using the accompaniment system. This unique arrangement saves all the accompaniment data down to three tracks, leaving five tracks available for overdubbing additional parts.
- Additional features include dual headphone outputs, audio and MIDI I/O, Key Transpose, a built-in metronome, and ten tuning temperaments.

PA60 *Professional Arranger*



Whether using the Pa60 as a songwriting station or as a back-up band, you can always be sure that its performance will be truly musical. The Pa60 combines enhanced sequencing functions, powerful performance features, comprehensive arranging tools and impeccable sound quality. Its huge, custom, backlit LCD screen displays the status of multiple features in an organized and easy-to-read manner.

- Relies on the same HI synth engine found in Korg's TRITON workstations. Included with its 660 sound programs is a set of very expressive solo instruments, sampled with a natural vibrato. All sounds are fully editable, and simple sound edits can be made inside the sequencer itself. Four studio-quality effect processors are available and feature 89 different effect types.
- Provides a huge collection of music Styles created by top musicians from around the world. Each Style provides up to eight accompaniment parts – drum, bass, percussion and more – that respond interactively with your playing. With 256 editable Styles onboard (plus 48 user Styles) the Pa60 can cover any situation.

KORG

CX-3 / BX-3

Single Manual and Dual-Manual Combo Organs





For years, if you wanted the sound of a real tone-wheel organ, there was only one instrument that could deliver – until now. Korg's Combo Organs accurately and faithfully re-create that classic sound in two portable models – the single manual CX-3 and dual-manual BX-3. The original Korg CX-3 first went on sale in 1979 followed by its dual-manual sibling, the BX-3. Both went on to become favorites in their own right, and are still widely used. Now, using advanced DSP technology and painstaking attention to sound, touch and style, they have been reborn and vastly improved upon.

FEATURES

- DSP technology makes it possible to precisely re-create the unique sound that, until now, could only be produced by the real thing. Two different tone-wheel sounds are provided: "VINTAGE" and "CLEAN". The CX/BX-3's amazing realism even extends to such details as allowing you to add the "leakage" noise generated by the tone-wheels, and the ability to adjust their overtone levels.
- The CX/BX-3 feature two sets of nine drawbars that duplicate the sonic characteristics of the originals – including the proper foldback settings and harmonic distortion. Select between the presets and these two sets of "live" drawbars while performing for real organ-playing experience. An EX mode in which all 18 drawbars can be utilized as one extended palette is provided. The resulting harmonics and percussion timbres can be used to create organ sounds never before heard.
- One of the unique characteristics of a genuine tone-wheel organ is the percussive effect that adds a readily identifiable sound to a note's attack. The CX/BX3 re-creates that beautifully. The volume, decay rate and harmonic pitch (2nd, 3rd) of the percussion can be adjusted – and the percussion has the proper "multi-triggered-yet-monophonic" characteristic for authentic organ performance. You can even control the amount of key-click heard when you press or release a key, simulating the dirty contacts often found on older instruments.
- With 128 programs, the CX/BX-3 are ready to cover a wide range of musical styles including pop, hard rock, jazz, gospel, and rhythm and blues. Detailed drawbar and parameter settings are displayed, and can be easily edited and stored.
- Proprietary "REMS" modeling technology is applied to create effects that add further authenticity to the organ sound. The rotary speaker effect, so important to the organ experience, is recreated with total realism. You can even adjust the time required for the rotor and horn to speed up or slow down when switching speeds, or stop the rotation altogether.
- Three types of amp simulation are provided: a traditional rotary speaker sound, a blend of rotary and tube amp sounds, or a pre-amp-only signal. The amp simulations are dynamic, so your sound naturally evolves from a warm and fat sound to a natural overdrive as your volume increases. In addition, the classic vibrato and chorus settings are provided along with three types of reverb, featuring adjustable time and depth.
- Attention has been paid to the long-neglected aspect of keyboard response speed. The key scanning method used on the CX/BX-3 starts the sound at the first key contact, ensuring a much faster response to your playing. Notes speak more crisply, allowing glissandi, trills and "smears" to be played with total authenticity.
- Not limited to a vintage recreation, the CX/BX-3 is equipped with many advanced MIDI features. Every drawbar, switch and control can be transmitted as MIDI data so your performance can be perfectly captured by a sequencer. The keyboard can be set to transmit key dynamics via MIDI to an external device that can even be layered with the onboard organ sound.
- The BX-3 provides all these features in a dual manual model. Pitch Bend and Modulation Wheels for improved MIDI control, additional voice parameters for even more authentic sound reproduction, and a newly-designed "waterfall" keybed for absolutely authentic look and feel have all been added.
- Different sets of drawbars, and Chorus/Vibrato On/Off can be assigned independently to the BX-3's UPPER and LOWER manuals, just like using a real tone-wheel organ. The single manual CX-3 gives you this functionality via a Split Mode, enabling you to play the single keyboard in two sections. Or you can plug in a second MIDI keyboard to recreate the dual manual playing experience.
- Drawbars, vibrato/chorus, and percussion switches are placed in the same location as on the classic tone-wheel organs, demonstrating a total commitment to tradition and playability. The sturdy wood cabinet gives them the proper vintage look, as does theBX-3's wooden stand.

B&H PHOTO - VIDEO - PRO AUDIO

EQUIPMENT LEASING AVAILABLE

C-3200/C-4500/C-6500

Concert Pianos



Korg's Concert Series piano's are housed in a stylish cabinet and come with all the essentials, including two types of piano sounds and a variety of valuable features only available from a digital piano. And with their stunningly beautiful sound, they are sure to bring years of musical enjoyment. The C-3200 and C-6500 feature a real weighted hammer action keyboard that responds to every movement of your fingers. Just like a grand piano, the keyboard has a heavier action in the lower register and lighter action as you play upward, giving you the truest piano feel possible. All three models offer a three-position key touch control that lets you select light, medium, or heavy response, so you can choose how the sound will respond to your playing dynamics. A two-track, 14,000 note sequencer lets you record two parts separately (like a backing track and a melody). This can be a convenient function to use when you are practicing or working on a new lesson.

FEATURES

- Three separate pedals are provided for damper, soft, and sostenuto, allowing professional-level piano performance.
- They also support half-pedaling, a technique where you only slightly depress the damper or soft pedals for a wide range of tonal possibilities.
- In addition to two grand piano sounds, they give you a total of nine sounds (six on the C-4500) including electric piano, harpsichord, organ, string, and bass, all with the same quality as on Korg's professional synthesizers. You can choose to split these across the keyboard with different sounds in the high and low ranges, or layer them for rich new sounds.
- They come with chorus and reverb effects that enable you to further enhance its piano and other sounds. (On the C-3200 the effects can be adjusted in three stages.). Brilliance is also adjustable in three steps to control the brightness of the sound.
- MIDI I/O allows connection to a synthesizer or other MIDI-equipped device. (IN addition, the C-4500 has a PC interface).
- Key transpose feature lets you transpose the pitch of the instrument so you can play in any key with ease.
- A convenient metronome function is built-in for practicing.
- In addition to the standard temperament, they provide two historical temperaments, Kirnberger and Werckmeister.
- With a maximum 64-note polyphony, you'll have plenty of notes for full piano performance even while using the sustain pedal.
- Utilizing Korg's world-acclaimed technology, the C-4500 and C-6500 boast beautiful, professional piano sound and feel. Their tone generator is equipped with the same HI synthesis system as best-selling Korg synthesizers, like the award-winning TRITON and KARMA workstations. With a new memory-rich, dynamic stereo-sampled piano sound, the C-4500 and C-6500 sound like a true grand piano and respond faithfully to your keyboard playing dynamics.
- On the C-6500, "Dynamic Stereo Sampled" piano sound actually switches between two different piano samples, one played softly and the other hard so your expressive playing is perfectly represented.
- Audio output jacks allow their sounds to be heard through external speakers. Or, via the audio inputs (not on the C-3200) you can plug an outside audio source and listen through their internal speakers.
- Their sophisticated and classic design features a brown walnut finish that looks good in any room. On the C-6500, the key cover that protects the keyboard is designed to close slowly, like a fine acoustic piano, giving an extra measure of safety in homes with small children.

New! EC-120 Concert Piano

The EC-120 is the most affordable 88-key Concert Series Piano. Using the latest sounds from Korg's professional voicing technology, the 32-voice EC-120 delivers a high-quality, spacious stereo piano sound, plus electric piano, harpsichord, vibraphone, organ and string sounds. The EC-120's full-scale keyboard provides the same feel as a traditional hammer-action acoustic piano and includes three levels of touch sensitivity.

Two pedals provide true piano performance, with the ability to switch the Sustain pedal to Sostenuto function when needed. Reverb and Chorus effects add spaciousness and color to the sounds. A built-in, 4,000 note MIDI recorder can capture and playback the user's performance for tracking progress or playing along to.Also features MIDI I/O, two headphone jacks, line inputs, and key transposition. Housed in a rich black walnut finish cabinet.

KURZWEIL

KME61

Micro Ensemble Keyboard



The KME61 offers the power and versatility of the Micro Ensemble in a 61-note keyboard. This synth action keyboard features 256 sounds from the PC2 series, including Triple Strike Stereo Piano, electric pianos, award-winning Take 6 vocals and much more. The KME61 has thirty effect combinations, with a choice of reverbs, choruses, delays and flangers. Each effect has a wet/dry mix which can be controlled per channel. The synth action of the KME61 also includes aftertouch, for a realistic playing experience. Pitch and Mod wheels give expressive control for any musician. The KME61 allows for the creation and storage of 16 multi-timbral setups, as well as customization of effects for each of the 256 sounds. KME61.

- 256 Preset programs using samples from the Kurzweil PC2 including: Stereo Triple Strike Piano, Electric Pianos, Strings, Basses, Guitars (Acoustic and Electric), Take 6 Vocals, Brass, Drums, and Percussion.
- 30 effects combinations, with a choice of reverbs, choruses, delays, and flangers. Each effects has a wet/dry mix, which can be controlled per channel. Change and store effects for each of the 256 presets.
- 32 Multi-Timbral Setups with up to 3 Zones. You can create and store up to 16 of your own Setups.
- MIDI In, Thru, Out responds on all 16 Channels
- 32 Voice Multi-Timbral Polyphony
- 20 Character x 2 line backlit LCD display with contrast knob
- Two 1/4″ Analog Outputs
- 1/4″ TRS Stereo Headphone output
- External Power Supply
- Pitch and Modulation wheels
- Data slider
- 1/4″ hold pedal input (Pedal is included)
- 1/4″ continuous pedal input
- Dimensions: 3.25″ x 37.25″ x 12.0″
- Weighs 22 lbs.

Micro Essemble ME-1



In the tradition of Kurzweil's highly popular and successful Micro Piano, comes our newest half rack module, the ME-1 Micro Ensemble. This low cost module features a wide variety of piano, keyboard and synth sounds, along with strings, guitars, basses, voices, and drums - a total of 256 programs in all. The samples and synth programming is taken directly from our PC2 series instruments, including our newest stereo triple strike acoustic piano sample, and our new multi-strike electric pianos (both Rhodes and Wurlitzer). It should be noted that the KB3 Organs from the PC2 are not found in the Micro Ensemble, but the regular organ programs ARE included. The Micro Ensemble has 32 voice polyphony and is multi-timbral on all 16 MIDI channels. A Global effects processor features the same effects found in our Stage Piano series. You can choose from one of 9 different effects, along with the wet dry settings for reverb and chorus, directly from the front panel, or via MIDI (the effects are applied the same for the programs on all MIDI channels). The Micro Ensemble can also be transposed up or down up to an octave.

- 1/2 Rack Module: will fit into a regular rack with a standard 1/2 rack adapter
- MIDI: Receives on all 16 channels, In and Thru jacks
- Internal Voices: 256 programs include Grand Piano, Tine E. Piano, FM E. Piano, Ensemble Strings, Voices, Organ, Harpsichord, Clav, Vibes, Marimba, Electric Bass & Acoustic Bass (with and without layered Ride Cymbal), Acoustic Guitar, Drums, Percussion, Synth Pads.
- Polyphony: 32 voices
- Effects: 9 preset effects, featuring 8 different reverbs, plus chorus, with separate controllable wet/dry levels for reverb and chorus components.
- Outputs: 2 Analog Outputs (1/4 in. jack Left/Mono and Right), Stereo headphone out.
- Power: external power supply.

SP76 • SP88 • SP88X

Lightweight Stage Pianos for the Next Century

The SP Series Stage Pianos are the performance keyboards you've been waiting for. Well-constructed and affordable, these stage pianos have everything you need. The SP Series comes in lightweight 76 and 88 key semi-weighted action models, plus an 88 note fully weighted action model. The Stage Pianos have the award-winning Kurzweil sounds and transmit on two, user-selectable, independent MIDI channels. Put your music on stage with this new model of performance piano from Kurzweil, the leader in sound technology.



SP88X



SP76

SP88



FEATURES

- Available with 88- / 76-note Semi-weighted action keyboard, or 88-note fully weighted action keyboard
- Light weight design for easy transportation
- 32-note polyphony

MIDI Controllers

- Two Ribbons (used instead of traditional wheels) provide flexible control.
- The Pitch ribbon has a user programmable software "spring", allowing the pitch to jump back to center or remain at the value where the finger lifts off the surface.
- The Mod Ribbon also has the programmable "spring", and in addition, allows for different functions (software destinations) to be programmed for the 'up' position separately from the 'down' position.
- A front panel control slider
- Pedal inputs include a continuous control pedal input plus a 1 dual footswitch input for two pedals.

32 High Quality Programs

- The award-winning Grand Piano offers stretched and non-stretched tuned versions, and programs with different EQ.
- Dual-Hardness Tine Electric Piano offers soft and hard strike samples arranged in many program variations, FM Electric Piano is also provided in a variety of programs.
- Programs based on the Ensemble Strings include selections with different EQ's and envelope response.
- Tone Wheel Organ programs include organ w/percussion, ballad organ, jazzy organ and more.

Setups

- 32 user programmable MIDI setups, each with 2 independent zones that can be used to configure a variety of splits and layered combinations with useful controller assignments.
- 2 independent external MIDI zones + 1 internal MIDI zone

Effects

- 9 preset effects, featuring 8 different reverbs, plus chorus, with separate controllable wet/dry levels for reverb and chorus components.

Inputs and Outputs

- Two (left/right) 1/4″ outputs
- 1/4″ stereo headphone out
- MIDI In and Out connectors: Transmits on 2 channels, Receives on 1 channel.
- Accepts remote setup change on any MIDI channel
- 12V DC external power supply

Dimensions:

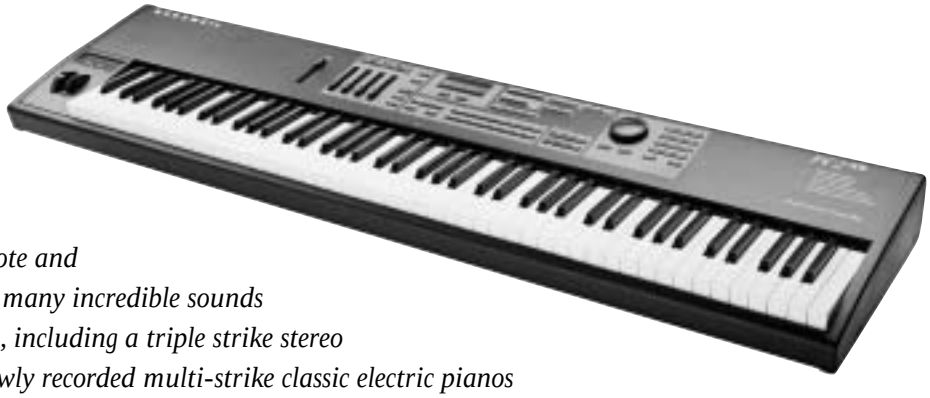
- **SP76**: 47.2 x 11.5 x 3.54″ (27.5 lbs.)
- **SP88**: 53.3 x 11.5 x 3.54″ (30 lbs.)
- **SP88X**: 54.8 x 14 x 4.8″ (52 lbs.)



KURZWEIL

PC2 • PC2X

Performance Controller



Following on the design of their award winning PC88, the PC2 features outstanding sound and impressive master controller capabilities. The Easy Split and Layer feature makes it fast and easy to create quick layers and splits. The PC2 series comes in both the PC2 76 note and PC2X 88 note keyboards. The PC2 features many incredible sounds not found on any previous Kurzweil models, including a triple strike stereo grand piano, the new lush stereo strings, newly recorded multi-strike classic electric pianos (both Fender Rhodes and Wurlitzer), brass sections, drums, percussion, voices (featuring Kurzweil's outstanding Take 6 vocal samples), and more. It also features Kurzweil's KB-3 modeled tone wheel organ mode, giving you ultra-realistic tone wheel organ programs. Orchestral Sounds include String Sections, Solo Strings, Woodwinds, Orchestral Brass, Jazz Brass, Orchestral Ensembles, Choir, Organ, Harp, Nylon String Guitar, Orchestral Percussion, and Ethnic Percussion. Additional sound include mallet percussion, guitar, bass, Clavinet, harpsichord, synth sounds, and more.

FEATURES

- 4 zone MIDI controller operations with easy Split/Layer functions
- 76 note lightly weighted keyboard with mono pressure (PC2) or 88-note, two-piece weighted keyboard with mono pressure (PC2X).
- Pitch wheel, modulation wheel, 4 front panel sliders, 5 front panel switches (momentary or toggle, software selectable), 2 continuous control pedal inputs, 3 footswitch inputs, mono pressure, breath controller input, Kurzweil Ribbon input.
- 64-note polyphony dynamically allocated, 128 with the Polyphony Expansion option
- 64 factory and 128 user programmable MIDI setups, each with 4 independent zones, effects and arpeggiator settings.

Effects (Dual Processors)

- Processor A presets provides a choice of: 58 Reverbs, 6 Delays, 10 Choruses, 6 Flangers, 3 Phasors, 4 Shapers, 2 Enhancers, 8 Filtered Effects, 4 Distortions, 1 Mono>Stereo, 3 Wide Stereo, 4 Compressors, 2 Panners, 7 Rotarys, Stereo Tremolo, and 44 combination effects chains utilizing Kurzweil's unique Laserverb.
- Processor B effects offers a choice of 30 Reverbs including booths, rooms, chambers, halls, and plates.
- All 16 parts can send to A and to B in any amount. A global send allows serial effects changing from A to B.

400 Internal Voice Programs

- 256 incredible sounds like the triple strike stereo grand piano, or the lush stereo strings, recorded multi-strike classic e. pianos and keyboards, tone wheel organs, brass sections, drums, percussion, mallet percussion, basses, guitars, voices, synths and more. 16 KB3 Programs use Kurzweil's modeled tone wheel organ mode. General MIDI Bank also available.
- An Orchestral ROM block comes pre-installed with 2 banks of 128 programs.
- 256 locations are provided for User created programs (128 for regular programs and 128 for KB3 programs). Expandable with an optional ROM black which adds 128 preset programs and 32 preset Setups.

Voice Control

- User Adjustable Voice Parameters (Timbre, Envelope, LFO)

Arpeggiator

- Full-featured MIDI note arpeggiator with multiple latch modes, selectable play order, velocity and duration as well as tempo and sync functions.

Inputs and Outputs

- Two (L/R) 1/4″ analog outputs
- 24-bit stereo, AES/EBU and S/PDIF compatible coaxial digital output
- 1/4″ stereo headphone out.
- MIDI In, Out, Thru (hardware switchable to copy of MIDI Out): Transmits on 4 channels, Receives on 16 channels.
- 9V AC external power supply.
- KFP-1 Single Piano-Style Footswitch

Dimensions and Weight

- **PC2**: 47⅞ x 14 x 4⅝″ (39 lbs.)
- **PC2X**: 54⅝ x 14 x 4⅝″ (55 lbs.)

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EQUIPMENT LEASING AVAILABLE

PC2 • PC2X



Conveniences

- Flash ROM allows software updates via system exclusive dumps.
- Lightweight and compact design

Optional Accessories

- KFP-2M Double Piano-Style Footswitch
- CC-1 Continuous Control Pedal
- DPC2-MDS Music Rack
- XP-1 Polyphony Expansion - increases polyphony to 128 voices
- PCR-1 16MB Orchestral ROM expansion (standard in current units, available as an option for older units)
- PCR-2 - 16 meg ROM Block adds additional sound (not yet released)

PC2R *1U Performance Rack Module*



The PC2R is a 1U rack module, which incorporates the feature set from Kurzweil's new PC series. 256 programs featuring triple strike stereo grand piano, new stereo strings, great multi-strike classic keyboards and organs and more. Kurzweil's award winning KB-3 tone organ models are also included. Dual stereo FX processors offer flexible routing and extensive effects algorithms. 4 real-time performance knobs are provided and allow users the same real-time control over functions as the sliders on their PC2 series. 64 voices expandable to 128 voices, easy split and layering capabilities, powerful setups (allowing for up to 4 split /overlapping zones with extensive MIDI control), expandable sound ROM, and 24 bit digital output with better than 117 dB S/N place the PC2R in a class of its own.

- Designed for anyone in search of great sound quality with great sounding effects in a compact 1U rack module.
- 4 multi-function controller knobs
- Four programmable zones
- Onboard arpeggiator with multiple note order modes, latch modes, selectable play order, velocity, duration, tempo, and beat values; synchable to MIDI clock
- 64 voice polyphony, user-expandable to 128
- Same dual sounds and effect processors as the keyboard models.
- 20-character x 2-line backlit LCD display with front panel contrast knob.

Inputs and Outputs

- Two 1/4″ balanced TRS analog outputs
- Coaxial digital with selectable format (AES/EBU, S/PDIF), bit width (16, 18, 20, 24), and dithering
- Front panel 1/4″ stereo headphone output
- MIDI jacks: In, Out, Thru

Options

- PCX-1 Polyphony Expansion
- PCR-1 16MB Orchestral ROM expansion (standard in current units, available as an option for older units)
- PCR-2 16MB ROM expansion (not yet released)



KURZWEIL

K2600 SERIES

Keyboard Workstations



Professionals around the world use Kurzweil instruments for all their production needs. From Film Studios and composers to the U.S. Government, Kurzweil delivers the quality and award-winning audio that is required by the best. Continuing this tradition, Kurzweil is proud to offer the K2600. Fully compatible with all models that Kurzweil offers, professionals can continue their work without interruption, always confident that archived work will be compatible. Building on Kurzweil's world-renowned operating system, the K2600 can do more than ever before. The new operating system expands the algorithms from 32 to 127 with Kurzweil's exclusive Triple Modular Processing. The K2600 is a complete workstation, allowing the user to work from concept to completion without changing from station to station. Full MIDI controller capabilities including eight sliders, two ribbon controllers, footswitches, pedals and breath controllers.

FEATURES

- 76 note semi-weighted and 88 note fully weighted action keyboard versions available.
- True 48 polyphony with 192 oscillators
- 16 MIDI channels, each can be a 32 layer program
- 64MB sample memory upgradable to 128MB
- Sampling option with analog and digital I/O and sample-while-play capability

Sound ROM

- 12MB of onboard ROM, including an acclaimed Stereo Grand Piano.
- Includes over 450 programs (with 100s more available) for a wide and useful selection. A search function allows you to locate programs, setups or objects quickly.
- KB-3 Organ Mode provides powerful tone wheel organ synth models
- Four expansion slots allow you to expand your K2600 to a total of 44MB of sample ROM.
- Optional ROM boards include the well known Orchestral and Contemporary - ROMS which already exist for the K2000 and K2500 series, as well as the newer Stereo Dynamic Piano, and Vintage Electric Pianos ROM options.

V.A.S.T.

- Extensive synth and sample editing is available via Kurzweil's V.A.S.T. (Variable Architecture Synthesis Technology) which features different 60 DSP functions.
- New Triple Modular Processing offers 126 sound shaping algorithms for per-voice processing.
- Even without the sampling option installed, you can take advantage of the many sample processing DSP capabilities. Some of the Sample Processing DSP functions include: time compression/ expansion, pitch shifting, re-sampling, pasting samples on beats-per-minute timeline (Mix Beat), volume ramps, mix sample, and many other functions.

File Compatibility

- K2000 and K2500 compatible, enabling it to read Kurzweil or third-party libraries for each instrument from disk or CD-ROM.
- Reads/writes AIFF and WAVE files.
- Reads libraries in Roland 700 series, Akai S900, S950, S1000, S1100, and S3000, and Ensoniq EPS, and ASR file formats.
- Import and export either Type 1 or Type 0 Standard MIDI files.

Powerful KDFX 5 stereo bus, 24-bit digital Effects

- Kurzweil KDFX utilizes Kurzweil's VLSI DSP chip that delivers a versatile 5-stereo-bus design, 24-bit signal processing, cutting edge DSP functionality, and unparalleled real-time control, all with an extremely flexible routing scheme.
- KDFX offers over 100 effects algorithms arranged into 260 effects presets.
- All I/O can be routed through KDFX including 8 outputs and 8 inputs from Kurzweil Digital Stream (KDS) digital I/O, or sampling I/O.

Sequencing

- Full-function 32-track sequencer with an internal resolution of 768 ppq.
- Sequencer functions include: Linear, Pattern and Step Record Modes
- Editing functions include: Event Edit List with view filters and audible scroll; Cut, Copy, Paste facilities; as well as Reference Quantize which allows you to quantize to a groove track, and to quantize tracks not previously recorded with a click..
- Advanced Auto Arranger functions (allows sequences to be triggered from keys with velocity sensitivity and transposition).

K2600 SERIES

Assignable Controllers

- The front panel has eight independently assignable sliders, useful as MIDI sliders, for sequencer mixdown, or as organ drawbars in the KB-3 Mode.
- A 600mm ribbon controller, running across the top of the keyboard, is programmable in three distinct zones or one long zone. It can be assigned to respond to finger positions as follows: relative to absolute center; relative to "virtual center" (initial finger-press defines the virtual starting point); return to center when finger is removed; or maintain current value when finger is removed.
- A shorter, pressure and direction sensitive ribbon controller below the pitch and mod wheels allows natural vibrato and subtle nuances of expression.
- Two buttons above the pitch and mod wheels are completely programmable.
- Additional controllers include sequencer "transport control" buttons
- 4 footswitch inputs, and 2 continuous control pedal inputs (1 CC2 and 1 CC2/breath controller). The breath controller input accepts Yamaha BC-1, BC-2, and BC-3 breath controllers. The multitude of flexible control surfaces make the K2600 the ideal center piece for any MIDI production.

CONNECTIVITY



- Outputs are 1/4″ TRS balanced and include a pair of mixed audio outs, plus eight individual audio outs.
- A 1/4″ stereo headphone output is also provided.
- Two 25-pin SCSI connectors
- MIDI In, Out and Thru (Thru switchable to a copy of Out).
- Controller inputs include 4 footswitches, one breath-controller and two continuous control pedal jacks 1 CC2 pedal and 1 CC2/breath input).

Models

- **K2600** – 76-note semi-weighted piano-style keyboard.
- **K2600S** – 76-note semi-weighted piano-style keyboard with the SMP2X Sampling Option.
- **K2600X** – 88-note fully-weighted piano-style keyboard.
- **K2600XS** – 88-note fully-weighted piano-style keyboard with the SMP2X Sampling Option.

Also Available



- The K2600R – All of the same features as the K2600, minus the keyboard and MIDI controllers, in a 3U rack space
- K2600RS – Same as above with the Sampling Option installed.

Live Mode

- Process any mono or stereo signal via the analog or digital sample inputs (requires the sampling option) with powerful V.A.S.T. synthesis functions and built in KDFX capabilities.
- Plug your guitar, microphone, mixer or tape track directly into your K2600 and customize the sound using any of the V.A.S.T. parameters, all in real-time. Then add effects using the K2600's on board effects or the state-of-the-art KDFX.
- With the digital I/O option you get 8 in and 8 out KDS I/O which can bi-directionally communicate with the DMTi and any other devices wired to the DMTi, such as MDMs (Adat, TDIF), Computer I/O (AES, S/PDIF) and more.

Options

- SMP-2X Advanced Sampling Option with Digital I/O, Optical I/O, and Sample-while-play. DSP functions include time compression/expansion, pitch shifting, re-sampling, pasting samples on a beats per minute time line (Mix Beat), volume ramps, mix sample and many other functions.
- DIO-26 Digital Input/Output Option adds the ability to input and output eight channels of digital audio to/from the K2600. All incoming audio can then be routed through V.A.S.T. and KDFX processing.
- DMTi Digital MultiTrack interface performs real-time digital data format conversion between K2600, AES/EBU, S/PDIF, ADAT and DA-88. It simultaneously performs real-time sample-rate conversion to 44.1 kHz, 48 kHz or a variety of external clock rates.
- Contemporary ROM, Orchestral ROM, Stereo Dynamic Piano ROM, and Vintage Electric Pianos ROM Soundblocks add additional ROM samples, along with new preset programs and setups
- PRAM-26 Program RAM Memory Expansion increases the amount of battery backed RAM from 486k to 1503k.

KURZWEIL

KSP-8

MIDI EQUIPMENT

1238

Multi-Bus Signal Processor



The KSP-8 is an eight channel, 2-space rack mount signal processor designed for the pro and semi-pro musician/engineer/producer requiring high quality signal processing and demanding unparalleled real-time control. It is the world's only 8-channel multi-effects processor with twice the processing power of Kurzweil's KDFX system. The Kurzweil KSP8 includes all the high-quality effects algorithms from KDFX, and adds surround reverbs and other new effects. The KSP8 breaks through traditional barriers and delivers with a wide range of flexibility. The base unit comes with 4 analog inputs and outputs, and stereo AES/EBU in and out, for a total of 6 possible channels. An I/O option slot provides a choice of expanding with 4 additional channels of analog I/O or 8 channels of Alesis Lightpipe I/O, Tascam TDIF, or AES/EBU I/O. Inputs and outputs may be flexibly routed. 24 bit precision is maintained throughout. The unit offers a selectable master clock, dedicated EQ, and can process 8 mono channels of audio simultaneously, making the KSP 8 an ideal tool for today's MDM based studios.



FEATURES

Factory ROM

- 249 effect algorithms (expandable)
- 636 effect presets (up to 999 user presets)
- 138 effect chains (up to 999 user chains)
- 14 studios (up to 999 user studios)

Effect Routing

- Any 8 of up to 14 possible inputs can be processed and routed to any 8 of the 14 possible outputs. This allows the KSP8 to be setup with 8 mono in, 8 mono out; 4 stereo in, 4 stereo out; 5.1 plus stereo I/O configurations and anything in between.
- Each input signal has its own multi-band EQ and stereo or multi-channel panner.
- Input and output routings can be saved as part of the Studio.
- Effect buses can be routed directly to the KSP8's outputs, or to internal mixdown buses for submixing.
- The unique multi-stage metering section allows for metering at 4 points in the signal path, for all 8 channels.

Architecture

- At the lowest level is the effect algorithm which are designed for use in mono, stereo, and surround applications.These high quality, and highly tweakable algorithms have been crafted exquisite effect presets ranging from spacious and gorgeous, to shocking and in your face rude.
- Effect presets are easily linked together to form effect chains on up to 8 buses. You can fully customize effect presets and chains and have all the control they want.
- Parameters can be controlled in realtime and automated via MIDI. A host of local modulation sources including up to 36 LFOs and 36 envelope generators, with 72 functions is also available for combining control sources.

The Studio

- The highest level of the KSP8 architecture is called the studio — a kind of "snapshot" of the total state of the KSP8 at one time.
- Studios contain all the physical connection settings, signal routing (analog and digital I/O selections), analog and digital signal levels, as well as the assigned effect presets, effect chains, effect bus assignments (including all MIDI channel and controller settings) and all multi-band EQ settings.
- A studio allows you to save and recall your entire project, quickly and easily.
- The KSP8 offers up to 999 locations for user-created objects. SmartMedia cards are used to store your work.

Effects Algorithms

Reverbs (booths, rooms, chambers, halls, plates, stages and more), Delays, Choruses, Flangers, Phasers, Shapers, Enhancers, triggered Filters, Distortions, Tube Simulators, Compressors, Limiters, Expanders, Gates, Bit Resolution Degenerators, Aliasing, Ring Modulators, Cabinet Simulators, Spatializers, Rotary Speaker Emulations, Tremolo, Resonant Lowpass, Highpass Filters, Bandpass, Notch, Parametric, Graphic and Multi-band equalizers, Kurzweil1s unique Laserverb, Pitcher' and Chaos, plus others.

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KSP-8

- A SmartMedia slot can be used to store data on cards up to 64Mb.
- 240 x 64 CCFL backlit LCD display with front panel contrast
- MIDI In, Out and Thru conectors with 16 channel operation. Data can be stored and retrieved via MIDI Sys Ex.
- ROM Upgrades: OS and objects upgradeable via SmartMedia card, or via SysEx dump from a MIDI sequencer.

Intelligent Interface Design

- The easy-to-use interface includes a 240 x 64 CCFL backlit display, scroll knob, data entry keypad, and dedicated buttons for navigation and actions.
- The intelligent and intuitive design offers features like Quick Pages, allowing instant access to the most commonly needed effect parameters.

Analog Inputs and Outputs

- Four 1/4″ TRS balanced inputs and outputs, expandable to eight with the optional KANA4 analog I/O board.
- 24-bit, 128x oversampling A-to-D and D-to-A conversion

Analog I/O Specs

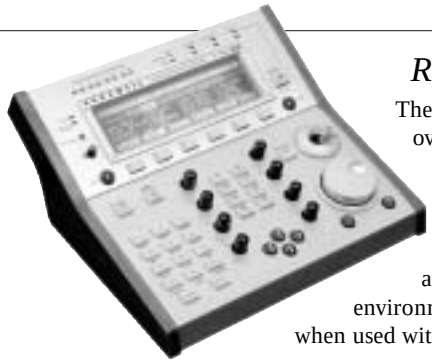
- 20Hz - 20kHz frequency response ±0.15dB
- Dynamic Range: > 110dB unweighted @ 1kHz
- Crosstalk: < -100dB @ 1kHz

Digital Audio I/O

- 24 bits Digital I/O with XLR connector – 110 ohms (AES/EBU) or 75 ohms (S/PDIF)
- 16, 20, or 24 bits output with a choice of None, Flat, and 2 types of Shaped dither.

Optional I/O

- KANA4 - 4 channel of 1/4″ analog I/O
- KAES8 - 8 channel AES/EBU or S/PDIF I/O (multi-pin connector, serial connectors, BNC word clock)
- KADT8 - 8 channel Alesis Lightpipe I/O plus 8 channel TDIF I/O with BNC word clock connector
- KMLN8 - mLAN option



RSP8 – Remote Control

The optional RSP8 offers totally remote operation over all KSP8 functions. The remote offers the same display, navigation and data entry controls as the KSP8. In addition, the RSP8 adds a joystick and 8 knobs to control the KSP8 more efficiently. The RSP8's compact, attractive design will be at home in any studio environment. One RSP8 can control up to 7 KSP8s when used with the optional HUB7 multi-port hub.

RUMOUR / MANGLER



Effects Processors

Say hello to the Rumour and Mangler, Kurzweil's two low cost effects processors. Based on the award winning KDFX technology found in the KSP8 and K2600, These single space rack units pack a powerful punch for a very low cost. The Rumour features a comprehensive set of Kurzweil's finest reverbs and reverb variations, including gated, reverse, and reverb/compression combinations, along with Kurzweil's unique LaserVerb(tm), and chorus/flange+delay+reverb combinations. The Mangler focuses on effects other than, but not excluding, reverb, featuring chorus, flanger, phaser, tremolo, rotary speaker, panner, delay, filters, LaserVerb, Pitcher(tm), synth trigger, ring mod, distortion, and compression effects, and many combinations.



- Stereo Balanced Analog I/O
- Stereo Digital S/PDIF I/O
- Stereo and Mono Input Channels
- Stereo Effects Buses
- 44.1 kHz and 48 kHz Sampling Frequencies
- Selectable Bit Depths of 16, 20, and 24 bits
- 3 band parametric EQ
- 192 Effects Presets
- 64 User Locations
- Extensive MIDI Control
- Up to 16 Effects Parameters
- 3-LED Input Signal Level Metering
- Output Clip LED
- Tap Tempo and BPM Sync
- EQ and/or FX Bypass
- Tap/Bypass Pedal Input

ROLAND

XV-5050

128-Voice Synth/Sample Player



The Roland XV-5080 128-Voice Synthesizer Module is the ultimate sound module based on the best-sounding, most expressive and expandable synthesizer/sample player architecture in Roland history. This flagship module offers 128 voices, true stereo waveforms per tone, expressive Matrix Control, expandability via new 64MB SRX- and popular SR-JV80-Series wave expansion boards, Roland/Akai sample library compatibility and a host of new studio-quality effects. Put simply, cutting-edge synthesis doesn't get any better than this.

FEATURES

- 32-part multitimbral and 128-note polyphony allows you to create lush arrangements without note robbing.
- The massive 64MB internal wave memory (16-bit linear equivalent) offers four times the capacity of Roland's JV Series synths.
- Patches are comprised of up to four tones, each capable of using stereo waveforms. Each of the four tones can be velocity-switched and assigned its own key range, providing a high degree of realism and expression at the basic Patch level.
- 1,152 Preset patches (128 x 9 bank with GM2 banks) and 128 User patches
- Performance Mode allows you to create complex setups, such as layers and/or splits, by combining up to 32 different patches. You can also assign each of the 32 patches to a different MIDI channel for use with MIDI sequencers.
- 64 (32 x 2 bank) Preset performances and 64 User performances.
- 23 (14 + 9 GM2) Preset rhythm sets and 4 User rhythm sets
- User memory can be greatly expanded using SmartMedia storage.
- Upward compatible with Patches from the JV-2080/1080/1010 and XP-80/60/50/30.

Sample Playback

- The XV-5080's sample playback capabilities will allow you to load Roland S-700 and Akai S1000 sample libraries via its 25-pin SCSI port.
- You can add up to 128 MB of sample playback memory via two 72-pin FP/EDO SIMM slots.

Expansion

- Four expansion slots are available for adding Roland's popular SR-JV80 Series expansion boards.
- Another set of four expansion slots is provided for Roland's latest SRX Series of 64MB Wave Expansion boards.

Intuitive

- 320 x 80-dot graphic backlit LCD
- The PatchFinder button allows you to quickly locate sounds by category. The Favorite List function calls up your favorite Patches in succession for live performance.
- The Matrix Control offers an extensive array of modulation routing capabilities, providing a wide variety of expressive control and sound shaping possibilities.

Studio Quality Effects

- Independent reverb, chorus and a multi-effects processors are provided.
- Many of the onboard reverbs are based on algorithms from Roland's top-of-the-line SRV-3030 24-bit Digital Reverb Processor.
- The multi-effects algorithms include unique Roland effects like RSS, Isolator, Shuffle Delay and more.
- In Performance mode you can assign up to 3 sets of Roland's acclaimed COSM algorithms guitar amp models and other unique and powerful effects.

Connectors

- 1/4″ stereo headphone output
- 1/4″ L/R main outputs plus three additional 1/4″ stereo pairs with 24-bit D-to-As.
- Coaxial and Optical S/P DIF digital outputs (24bit, 44.1kHz)
- A proprietary 24-bit R-BUS digital output port is available for sending eight channels of audio to the latest VS recorders, or ADAT and Tascam T-DIF compatible gear using the optional DIF-AT Interface Box.
- Two MIDI Ins (for accessing all 32 parts, plus a, MIDI Out and Thru connector.

XV5050

64-Voice Synthesizer Module



The XV-5050 is a single rack space, 64-Voice Synthesizer Module that gives musicians access to Roland's flagship XV sound engine without breaking the bank. This space-saving 1U rack module offers four stereo Tones per patch, expressive Matrix Control and over a thousand inspiring Presets—expandable via high-quality SRX-Series Wave Expansion Boards. There are also three independent effects processors, digital outputs, plus a new USB port for easy connection to PCs. Don't compromise your sounds. Get the best for less!

FEATURES

- Affordable, 64-voice 16 part multitimbral synth module based on acclaimed XV-5080 synth engine
- Stereo 4-Tone architecture and Matrix Control for ultra-expressive sounds
- 64MB of internal wave memory (16-bit linear equivalent) – 1,083 (XV-5080 equivalent) waveforms.
- Includes all presets from the XV-5080, select Fantom patches and all-new sounds
- 1,024 (128 x 8 Banks) Preset patches and 256 (GM Level 2) Preset patches
- 16 (2 x 8 Banks) Rhythm Sets plus 9 (GM Level 2) Rhythm Sets
- 64 (32 x 2 Banks) Performances
- 128 User patches, 4 User Rhythm Sets and 64 User Performances

Expansion

- Two SRX-Series Wave Expansion Board slots allow you to expand and customize your XV-5050 with up to 192MB of high-quality sounds designed to suit your needs.

Studio Quality Effects

- Three independent effects processors with everything from 24-bit reverb to chorus, plus 90 multi-effects including COSM Guitar Amp Modeling.

User Interface

- Display 20 letters 2 lines (backlit)
- Convenient 1U rackmount design

Inputs and Outputs

- 1/4″ L/R main outputs plus two individual 1/4″ outputs using superb sounding 24-bit D-to-A converters
- Coaxial and Optical S/P DIF digital outputs (24bit, 44.1kHz)
- 1/4″ stereo headphones output on the front panel
- A USB port on the front panel can be used as a MIDI interface for your USB-equipped computer.
- MIDI In, Out and Thru connectors
- Included accessories:
  Owner's Manual, CD-ROM (USB Driver), Power Cable

XV-2020



64-Voice Expandable Synthesizer Module

The XV-2020 64-Voice Expandable Synthesizer Module puts Roland's flagship XV sounds in a convenient half-rack design complete with USB. Expandable via two SRX-Series Wave Expansion Boards, it's like a powerful "plug-in" synth with up to 192MB of sounds, 3 independent effects processors and intuitive editing software.

- Half-rack synth module with 64MB of XV sounds and USB port
- 64-voice polyphony and 16-part multi-timbral operation
- Presets include 768 Patches, 17 Rhythm Sets and 64 Performances
- USB port for easy MIDI connection and control via software
- 3 independent effects processors: reverb, chorus and 40 multi-effects
- Stereo 1/4″ and RCA outputs
- Includes editing software for Mac or PC
- Expandable via two SRX-Series Wave Expansion Boards

ROLAND

FANTOM-S • FANTOM•S88

Advanced Music Workstations





Roland's Fantom-S workstation keyboards are a complete solution for professional music production. The 61-note Fantom-S and 88-note Fantom-S88 both feature seamless integration of audio and MIDI—plus advanced sampling with realtime timestretch, sample chop and new "Skip Back Sampling." A Dynamic Pad Bank makes it easy to arrange and trigger audio, while all-new factory sounds, an onboard mastering processor and USB file exchange give the Fantom-S the power to take your music from idea to completion.

FEATURES

Advanced music production studios with 61-note keyboard (Fantom-S) or 88-note Progressive Hammer-Action Keyboard (Fantom-S88).

Internal Sounds

- 64 voice polyphony and 16 part multitimbral (shared with Sampling Section).
- All-new Patches and Rhythm Sets built from 64MB-equivalent wave ROM. These stereo-sampled waves were painstakingly sampled in world-class studios around the world, and are the finest building blocks ever offered in a Roland keyboard.
- 640 Preset Patches and 256 General MIDI 2 Patches; 32 Preset Rhythm Sets plus 9 General MIDI 2 Rhythm Sets; 64 (16 part multitimbral) Preset Performances
- 256 User patches, 32 User Rhythm Sets and 64 (16 part multitimbral) User Performances
- A range of tweakable modulation routings are available via the Matrix Control
- The user programmable LFO allows you to program cyclic modulation (such as filter cutoff, pitch, etc.) using 16 steps.
- Expandable to 192MB of internal sounds via four optional SRX-Series Wave Expansion Boards.

Flexible User Sampling

- Includes 32MB of sample RAM (expandable to 160MB) — enough to resample an entire song.
- Audio (samples, loops, etc..) can be used as a waveform in the expressive stereo 4-tone synth engine, or sequenced seamlessly alongside MIDI data - complete with realtime timestretching.
- Your favorite samples can be saved to the internal 16MB flash memory or to an optional SmartMedia card.
- The unique "Skip Back Sampling" feature continuously records performances, enabling you to retrieve an idea that might otherwise be lost.
- Sampled waveforms are displayed graphically for quick and precise editing
- Waveforms can be automatically divided into 16 "slices" using the Chop function.
- The onboard USB port allows you to grab .WAV and/or AIFF files from your Mac or Windows computer.

16-Track High Resolution Sequencer

- The internal pattern-based 16-track sequencer features a 480 PPQN (pulse per quarter note) resolution, and a 400,000 note capacity with support for both realtime and step recording.
- Because the sequencer is always active, you can start recording as soon as inspiration hits — whether it be while playing live, tweaking the realtime controls or playing Arpeggiator patterns.
- The oversized LCD display makes it easy to view and edit track data with unparalleled precision when using Microscope editing.
- Compatible with MRC Pro Songs and Standard MIDI Files (format 0, 1)

Mixing

- The Mixer View screen takes advantage of the large 320-240 LCD display to offer the look and feel of a digital mixer. This screen offers intuitive graphic editing of a multitimbral Performance allowing you to change track levels, panning and effects settings using the four front-panel knobs.
- A Layer View screen offers quick graphic editing for layering and key zones.

B&H PHOTO - VIDEO - PRO AUDIO EQUIPMENT LEASING AVAILABLE

FANTOM-S • FANTOM•S88

Arpeggiator and Rhythm Generator

- The onboard Arpeggiator provides 128 Preset and 128 User patterns that allow you to create anything from wild synth patterns to bass lines and chord comping.
- Graphically build your own arpeggiator patterns by inputting notes in the 32-step grid.
- Parameters such as pitch, velocity and note duration can be specified for each step.
- The Rhythm Generator is designed to allow you to quickly and easily add realistic sounding drums using a choice 256 Preset and 256 User rhythms.

Studio Quality Effects

- 3 MFX processors, plus dedicated reverb, chorus and new mastering processor.
- An onboard 24-bit reverb on par with rackmount processors and professional plug-ins
- A dedicated chorus processor that serves up warm, rich chorus and delay.
- Three multi-effects processors handle everything from tempo-synced delay and Lo-Fi processing to COSM Guitar Amp Modeling, Slicer, Isolator and more.
- A dedicated mastering processor with multi-band compression lets you add that finishing touch to your mixes.
- You can resample phrases or entire songs with effects to create entirely new sounds or bring a mix into your computer for CD burning.

Expressive Interface

- 16 (velocity-sensitive) Dynamic Pads make triggering internal sounds (such as drums), RPS Phrases, samples and audio loops easy.
- 320 x 240 dots LCD Display with helpful graphic icons including full waveform editing with zoom capabilities.
- Pitch Bend / Modulation joystick-style lever as well as 4 realtime assignable control knobs.
- The D Beam Controller uses an infrared beam of light that allows you to controller a variety of synth and effect parameters.

Rear Panel



- Two pairs of stereo 1/4″ line outputs
- Two 1/4″ inputs: input one is mic/line switchable; while input two is line only.
- Coaxial and optical S/PDIF digital outputs
- 1/4″ stereo headphone headphone output.
- Hold and control pedal inputs.
- MIDI In, Out and Thru connectors
- The onboard USB port makes it easy to load .WAV/AIFF files, or export a finished song to a computer for CD burning or further editing.
- A SmartMedia storage slot accepts 8 to 128MB SmartMedia cards (3.3V) for backing up samples, sequences, presets and performances.

Sound collection optimized for XV-5080's sample-playback capabilities*; Includes samples taken from popular L-CDP-Series for S-Series samplers; Perfect for composer/performers requiring very high quality sounds.

**Samples are in S-700-Series format and must be convert loaded for use with XV-5080.*



L-CDX-01
Rhythm Section Instruments

Two collections on one CD-ROM: Drums & Cymbals Vol. 1 and Guitar & Bass Vol. 1.



L-CDX-02
Keyboard Instruments

Keyboards of the 60's & 70's Vol. 1 and Vol. 2 on one CD-ROM optimized for the XV-5080.



L-CDX-03
Brass & Woodwinds

Two CD--ROMS containing Orchestral Winds, Brass Sections, Super Sax and Solo Brass.



L-CDX-04
Strings/Orchestral Percussion

Orchestral Percussion, Solo Strings, Symphony Orchestra and String Sections on two CD--ROMS.

ROLAND

SRX Series Expansion Boards

Roland's Premier Sound Library

The SRX-Series is Roland's next-generation sound library for "SRXpandable" instruments like the Fantom-S/-S88, XV-Series sound modules and RD-700 expandable synthesizer. With a strong focus on quality, each 64MB-equivalent expansion is loaded with sounds for all types of applications and genres.

- User-installable SRX wave expansion boards with all-new stereo sounds with astonishing quality
- Fully compatible with XV-Series synthesizers, Fantom Workstations, and RD-700 Expandable Keyboard
- Equivalent to 64MB of high-quality waveforms with zero loading time (when converted into 16-bit linear format)

SRX-01 Dynamic Drum Kits



- 64MB of world-class drum kits
- Drum sounds developed in conjunction with Spectrasonics
- Stereo samples with multiple velocity switches for unsurpassed authenticity

SRX-02 Concert Piano



- User-installable wave expansion board offering the best concert grand piano sound in Roland history
- Entire 64MB memory dedicated to stereo samples of a single, very rare European piano

SRX-03 "Studio SRX" Wave Expansion Board



- An extensive 64MB collection of all-new pop music sounds
- Focus is on superb sound quality rather than quantity

SRX-04 Symphonique Strings



- Includes violin, viola, cello, bass and full orchestra selections
- Multiple performance articulations of virtually every type

SRX-05 Supreme Dance



- 64MB of all-new waveforms and patches for dance music producers
- Focus is on individual sounds: 818 waveforms, 312 patches, 34 rhythm sets including 113 kicks and 133 snares alone!

SRX-6 Complete Orchestra

Includes 64MB of waves from SR-JV80-02/07/13/16



SRX-7 Ultimate Keys



Includes 64MB of waves from SR-JV80-03/04/08/09/10

SRX-8 Platinum Trax



Includes 64MB of waves from SR-JV80-11/12/15/19

SRX-9 World Collection



Includes 64MB of waves from SR-JV80-05/14/17/18

SRX-6/7/8/9 Wave Expansion Boards feature waves from 4-5 classic SR-JV80-Series boards with all-new patches programmed for XV-Series engine. They include stereo waveforms, Matrix Control and multi-effects, plus most original Patches and Rhythm Sets on CD-ROM.

B&H PHOTO - VIDEO - PRO AUDIO
EQUIPMENT LEASING AVAILABLE

KC SERIES



Keyboard Amplifiers

The Roland KC-Series has earned a reputation for both powerful sound and flexible mixing features. Now with the introduction of the KC-550/350/150, that legacy is set to continue. All three models feature a flexible 4-channel design with a stereo Auxiliary input and XLR Microphone input—plus compatibility with the new KCW-1 Powered Subwoofer and more.

- 4-channel stereo mixing keyboard amplifiers with clean, powerful sound
- Stereo Auxiliary input (RCA type) for connecting CD/MD players
- XLR Microphone input for adding vocals or acoustic guitar
- New Subwoofer output creates powerful low end with optional powered subwoofer
- KC-550/350 include Output Select and Shape switch, plus XLR or 1/4″ Line output

KC-550
Stereo Mixing Keyboard Amplifier

The KC-550 boasts 180 watts of powerful sound via a 15″ speaker and horn tweeter. In addition to its four input channels, this flagship amp includes an Output Select switch for selecting the monitor sound destination, a Shape switch for quick sound adjustment and an XLR Line output for direct connection to a mixing console.

- Powerful 180-watt/15″ speaker and horn tweeter
- Output Select feature and convenient Shape switch for tonal adjustment
- 3-band EQ and Subwoofer output compatible with new KCW-1
- XLR Line output for connection to mixing consoles
- Stereo Link In/Out for chaining to KC-550s for stereo applications

KC-350
Stereo Mixing Keyboard Amplifier

The perfect combination of power and value, the KC-350 is ideal for medium-sized gigs. It features a 120-watt amplifier with convenient 3-band EQ, Output Select and Shape switch features, plus a Subwoofer output for extra low end. The KC-350 can also be connected to mixing consoles via its XLR Line output.

- 120-watt/12″ speaker and horn tweeter
- Output Select feature and convenient Shape switch for tonal adjustment
- 3-band EQ and Subwoofer output compatible with new KCW-1
- 1/4″ phone output for connection to recording devices
- Stereo Link In/Out for chaining to KC-350s for stereo applications

KC-150
Stereo Mixing Keyboard Amplifier

A compact 60-watt amp with 4-channel capability including Aux and Mic inputs. The KC-150 can also be expanded with an optional subwoofer for a powerful low-end boost.

- Affordable 4-channel keyboard amplifier with 60-watt/12″ speaker and piezo tweeter
- Includes stereo Aux input (RCA) and XLR Mic input
- 2-band EQ and Subwoofer output compatible with KCW-1
- Headphones output; can be mounted on speaker stands

KCW-1 Powered Subwoofer



A convenient way to add 200 watts of punchy low-end to sub-out equipped KC amps, or standard amps via the stereo Thru output.

- 200-watt powered subwoofer for use with KC-Series amps
- Stereo Thru Out jack (XLR and 1/4″) for amps without a Subwoofer output
- Intuitive controls including Crossover knob, Phase switch and Shape switch

ROLAND

RS-50 • RS-70

Synthesizers



The RS-70 and RS-50 are streamlined and affordable synths that boast hundreds of CD-quality sounds that can be easily called up via simple Direct Access Buttons. Both units offer a number of performance features including a D Beam Controller, arpeggiator and chord memory. The RS-70 goes even further by adding an easy instrument-based Loop Sequencer, USB-MIDI interface and floppy drive for playing back Standard MIDI Files.

- Full of new sounds based on an entirely new set of waveforms sampled at 44.1kHz.
- Direct Access buttons let users call up sounds by category, while bundled editor software makes it easy to dive in and program new sounds.
- The RS-70's Loop Sequencer demystifies MIDI sequencing by allowing you to simply choose a sound and record. As the recording loops, additional sounds can be selected and overdubbed until all parts are entered.
- Realtime and step recording are both supported, along with Microscope editing.
- The Multi Chord Memory function allows you to play a variety of complex chords using a single note.
- You can also trigger phrases at the touch of a key using the Arpeggio/Phrase Generator. The RS-70 adds the ability to program your own Arps and phrases as well as the ability to record these to the sequencer).
- The RS-50's Rhythm Guide metronome offers a number of preset rhythm patterns in a variety of styles.
- Both models employ Roland's D Beam Controller which offers realtime control over a variety of functions like pitch and filter cutoff.

Inputs and Outputs

- Stereo 1/4″ outputs and a 1/4″ stereo headphone output
- Hold and Control Pedal jacks
- MIDI In and Out connectors

The RS-70 Also Adds

- The built-in floppy drive allows you to load GM/GM-2-compatible Standard MIDI Files.
- The onboard USB port can be used as a MIDI interface for direct connection to your computer.

XV-88 128 Voice Synthesizer



The XV-88 128-Voice Expandable Synthesizer is Roland's finest performance synthesizer to date, featuring powerful XV-3080 synthesis, an 88-note precision hammer-action keyboard, and other expressive features like the infrared light-sensing D-Beam controller. This synth offers 128 voices, true stereo waveforms per tone, expressive Matrix Control, expandability via new 64MB-equivalent SRX- and popular SR-JV80-Series wave expansion boards, and a host of new studio-quality effects-marking the XV-88 as performance synth of the highest order.

- 88-note precision hammer-action synthesizer with D-Beam controller and aftertouch.
- 128-voice synth engine for extensive layering and multitimbral applications
- Massive 64MB-equivalent internal wave memory includes incredible new sounds, plus all sounds from JV-2080 and a collection of waveforms from the JD-990 Super JD
- Stereo playback capability per tone for amazing velocity switches and keyboard splits
- Powerful Matrix Control gives sounds an unprecedented level of nuance
- Studio-quality reverb and effects derived from Roland SRV-3030, V-Studios, and RSS technology
- "SRXpandable" via SRX-Series 64MB-equivalent
- Compatible with SR-JV80-Series wave expansion boards and XP/JV-Series Patches (XP-80/60/30, JV-2080/1080/1010)
- Advanced arpeggiator functions

RD-170

Stage Piano



The RD-170 is an entry-level stage piano, for students and live performers, with an 88-key weighted hammer action that feels and sounds like it should cost hundreds more. And with an upgraded sound engine that's equally realistic, the RD-150 sets a new standard for affordable digital pianos.

- 88-note weighted hammer-action keyboard with improved touch response
- Full 64-voice polyphony with 64 sounds including Roland's best stereo grand piano plus electric pianos, clavinet, vibraphone, strings, pad, voices, jazz and church organ, guitar, acoustic bass and more.
- Master Keyboard functions for controlling external MIDI modules
- 16 Programmable Setups allow instant recall of custom settings
- 8 levels of Sympathetic Resonance add realistic acoustic piano resonance when using the damper pedal
- Keyboard split and layer functions with separate volume controls for upper and lower parts.
- Stretch Tuning function provides instant alternate tunings
- Built-in 2-track Recorder for practice or accompaniment
- Onboard reverb and chorus effects with eight levels
- MIDI In and Out connectors
- Half-pedal recognition for sustained playing just like an acoustic piano (requires optional DP-8 Damper Pedal).

RD-700

Expandable Keyboard



The RD-700 Expandable Keyboard is loaded with 64MB of stereo-sampled piano and instrument sounds — expandable using quality-intensive SRX-Series Wave Expansion Boards—plus an 88-note Progressive Hammer-Action Keyboard for the ultimate musical expression. And with its new graphic LCD, intuitive knob- and slider-based editing and powerful arpeggiator, the RD-700 is everything an expandable keyboard should be.

- Flagship expandable keyboard with 128-voice/16-part multitimbral sound engine and 88-note Progressive Hammer-Action Keyboard
- Massive 64MB-equivalent internal wave memory includes stereo-sampled piano sounds and a collection of Roland's best instrument sounds
- Large graphic LCD for easy operation; dedicated knobs and sliders for quick sound shaping and direct volume control over parts
- Expandable via two SRX-Series 64MB-equivalent Wave Expansion Boards
- Onboard effects include Sympathetic Resonance and 63 algorithms identical to the XV-3080
- One-touch Piano button instantly switches RD-700 for piano performance
- Realistic tonewheel organ sounds with graphical drawbar editing
- Powerful arpeggiator with 45 styles including guitar strumming; 50 onboard rhythm patterns for practice and composing
- MIDI Tx button for easy control over external MIDI devices and sound modules

ROLAND

SR-JV80 EXPANSION BOARDS

Roland's SR-JV80 Sound Library

Genre-specific sound expansions compatible with a wide range of Roland synths and modules including the famed JV Series sound modules and XP Series synths and workstations as well as the current XV-5080 128 voice sound module.

- User-installable SR-JV wave expansion boards with all-new patches and waveforms
- Fully compatible with XP-80/60/30, JV-2080/1080/1010, XV-5080 and RD-700 Expandable Keyboard
- Equivalent to 16MB of high-quality waveforms with zero load time when converted into 16-bit linear format

SR-JV80-01 Pop

A collection of basic sounds ranging from rhythm sections, keyboards, bass and guitars to strings, brass and synth patches. 224 waveforms and 145 Patches.

SR-JV80-02 Orchestral

Realistic sound for classical orchestration, covering a variety of string, woodwind, brass and percussion sounds. 174 waveforms and 255 Patches.

SR-JV80-03 Piano

Features an unprecedented variety of piano sounds, including world-class grands and electric pianos. 73 waveforms and 111 Patches.

SR-JV80-04 Vintage Synth

The definitive collection of legendary "vintage" synth sounds, including most of the classic Roland synths. 255 waveforms and 255 Patches.

SR-JV80-05 World

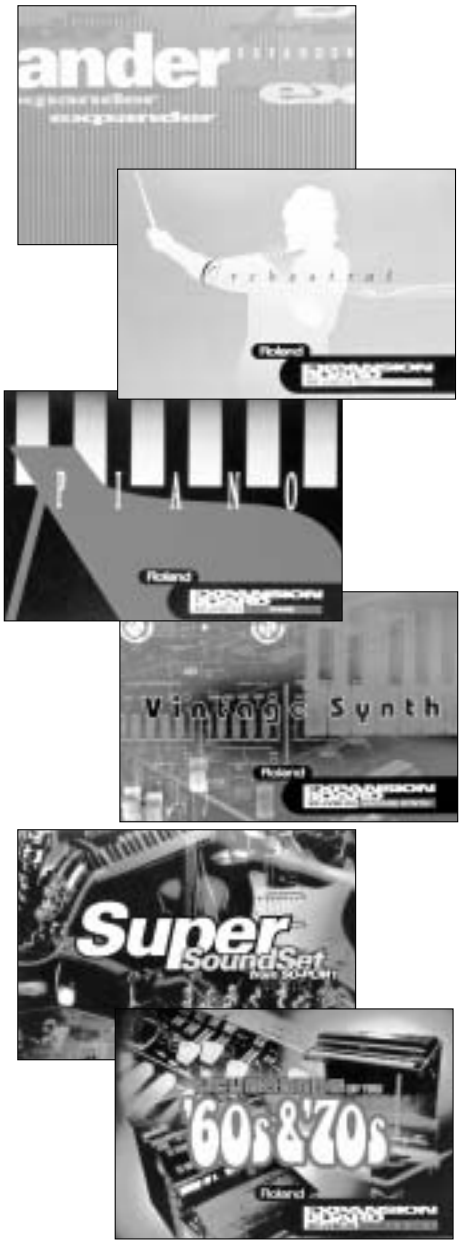
Extensive collection of ethnic instruments from around the globe, including bagpipes, quena, kayakcom and tablabaya. 255 waveforms, 255 Patches and 1 rhythm set.

SR-JV80-07 Super Sound Set

Contains sounds ranging from rock and pop to Baroque and traditional music, such as harpsichord, accordion, fiddle and banjo. 255 waveforms, 255 Patches and 8 rhythm sets.

SR-JV80-08 60s & 70s Keyboards

Offers various classic organ and electric piano sounds, including a vast number of Hammond B-3 settings. 255 waveforms and 255 Patches.



SR-JV80-09 Session

Includes lush stereo grand pianos, high-quality strings, choirs, nylon and electric guitars, brass and synth textures. 206 waveforms, 255 Patches and 8 rhythm sets.

SR-JV80-10 Bass & Drums

Contains cutting-edge bass and drum sounds meticulously sampled from renowned studio musicians including Marcus Miller, Abraham Laboriel, John Patitucci, Abraham Laboriel Jr. and Bob Wilson, adding a high-powered rhythm section to your music. 241 waveforms and 204 Patches.

- Includes various playing techniques such as slapping, fingering and picking, slides, harmonics and assorted playing noises
- Dynamic drum waveforms with expressive velocity control

SR-JV80-11 Techno Collection

High-quality Dance/Techno sound expansion board featuring 255 waveforms, over 200 Patches and 8 full rhythm sets.

- Provides a wealth of classic Roland drum and synth sounds (TB-303, TR-808, -909, etc.) as well as effected/distorted bass, synth and rhythm sounds
- Sound effects include noise, hits and industrial sounds

SR-JV80-12 Hip Hop Collection

Features 256 Patches and 8 full rhythm sets of the latest, fattest hip-hop sounds.

- Over 40 cutting-edge hip-hop phrase loops for quick song creation
- A wealth of hip-hop instrument sounds including drums, bass and keyboards
- Includes a variety of "Lo-fi" sounds, human beat box sounds, vocal sounds and sound effects

SR-JV80 EXPANSION BOARDS

SR-JV80-13 Vocal Collection

High-quality vocal sounds ranging from choir groups and jazz scatting to Gregorian chants. Extremely expressive and strikingly realistic, they can add life to music ranging from soundtracks to classical compositions.

- Stereo sampled Jazz scatting includes five-stage velocity switching with very responsive vocal sounds (in Performance Mode only)
- Contains 94 different waveforms focusing on unrivaled sound quality

SR-JV80-14 "Asia" Collection
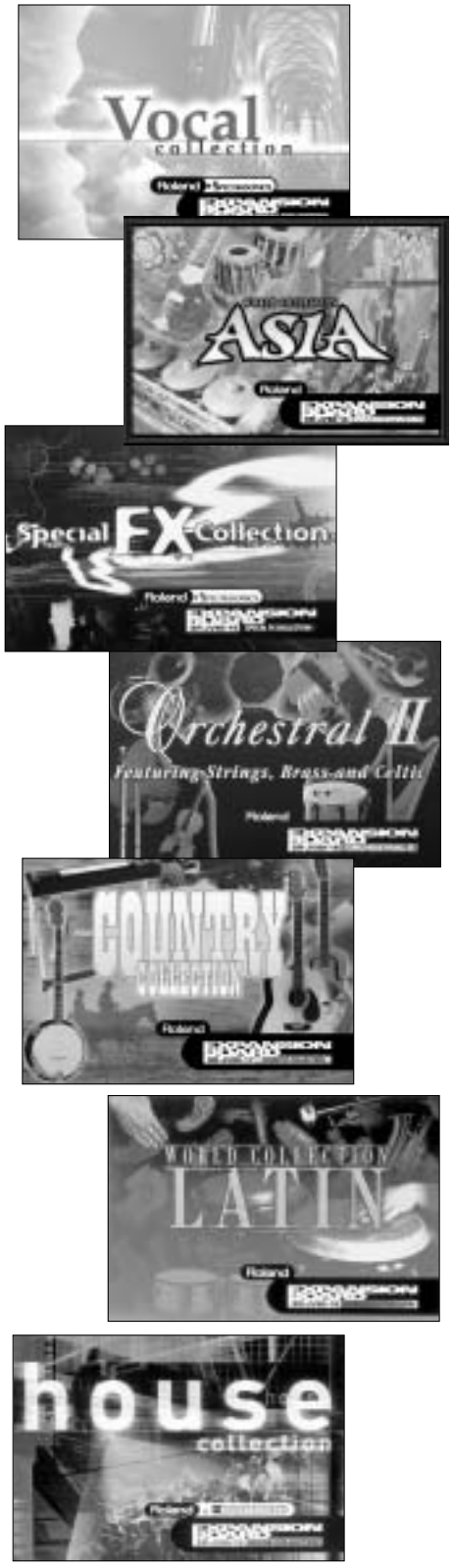
Authentic Asian instruments, including distinctive sounds from China, Indonesia, India, Japan, Korea and more.

- Features Asian string instruments with a variety of distinct "weeping" sounds, as well as rhythm instruments
- Phrase loops include both Standard (even-metered) and Asian-style timing that can be tempo sync'd to MIDI
- Chinese instruments include Er Hu, Yang Qin, Suona and many others
- Gamelan (Indonesian orchestral music) instruments include Jegogan, Jublag and Pemade — other Korean and Japanese sounds also included
- Indian instruments include Tabla, Dholak, Dhol, Sitar, Santoor and more
- Expressive possibilities are easily enhanced with bender lever and Arpeggiator functions

SR-JV80-15 FX Collection

A range of sounds including ambient sounds, noise, rhythmic hits, environmental sounds, nature sounds and more – many derived from the popular Distorted Reality CD-ROM.

- High-quality wave expansion board with 100 waveforms and 200 Patches
- Includes a variety of synth-type effects, Ambient sounds, Noise and Rhythmic Hits
- Environmental sounds include rivers, ocean, birds, insects, weather sounds, industrial sounds and machines, city and construction sounds, and much more



SR-JV80-16 Orchestral II

Includes stunning new string, brass, wind and other orchestral sounds, as well as Celtic instrument samples – great for film scores and television commercials. 100 waveforms and 200 Patches.

- Contains a wide variety of useful sounds, including stereo string sounds, brass, wind instruments, percussion, ensemble, breath noise and more
- Celtic instrument sounds including Bosouki, Uillean Pipe, Bodhran, and Clarsah Harp

SR-JV80-17 Country Collection

Easy access to hundreds of authentic country music sounds.

- Includes acoustic guitars, pedal steel, resonator guitar (metal hollow-body), clean electric guitars, fiddle, banjo, wash-tub bass, and a variety of natural drum sounds
- Perfect for studio use and live performance

SR-JV80-18 "Latin" Collection

Designed to meet the unique needs of Latin music, this is an authentic collection of Latin drum and percussion sounds, pianos, horns, and other genre-specific instruments.

- Authentic Latin sounds including pop, mariachi, cumbia, and more
- Genre-specific instruments include percussion, pianos, horns, etc.
- Onboard rhythm/phrase loops for easy groove creation and song composition

SR-JV80-19 House Collection

Loaded with all the fresh loops, phrases and keyboard sounds needed to lay down your next house track. Developed in collaboration with Ueberschall, many of the sounds and loops were created by legendary house music producer Mousse T.

- Based on Ueberschall's acclaimed "HOUSEWORX!" and "jam box" CD--ROMS (from producer Mousse T.)
- Includes 255 waveforms: drum and percussion loops; piano and guitar phrases; piano, bass, brass and string sounds; hits and vocal samples
- Contains 256 patches including key split settings of loops and keyboard sounds for instant house music creation

ROLAND

V-SYNTH



61-Note Synthesizer

The Roland V-Synth is a distinctive and inspiring new instrument that opens up a world of new sounds full of life and motion. The V-Synth's sound engine is designed to be configured in several different ways by choosing from preset Structure types--just like a semi-modular synth. Imagine a synth that combines multiple oscillator technologies with realtime control of a waveform's pitch, time and formant. Now add user sampling to create entirely new waveforms, powerful COSM filtering, a killer arpeggiator, touchscreen and a host of realtime controllers including the revolutionary TimeTrip Pad.

FEATURES

- 61-note synthesizer with velocity and channel aftertouch
- 16 part multitimbral and 24 voices of polyphony (depending on available DSP).

Sound Generator

- Two variable oscillators, each with a choice of three sound generation techniques.
  - The PCM oscillator uses Roland's VariPhrase technology for independently manipulating the pitch, time and formant of the over 300 preset or user-sampled waveforms.
  - The second oscillator type is Analog Modeling which offers nine traditional waveforms, utilizing Roland's latest modeling technology to deliver sounds that are fat and warm.
  - External Audio Processing is the third oscillator type, which allows any external sound to be used as a synth waveform.
- All oscillator types can be mixed and layered in several ways, or modulated using FM, ring mod and hard sync to create sounds never heard before.
- Powerful COSM processing including filter modeling, guitar amp modeling, resonator and Side Band Filter.

Effects

Global reverb, chorus and multi-effects enhance the sonic palette even further.

TimeTrip Pad

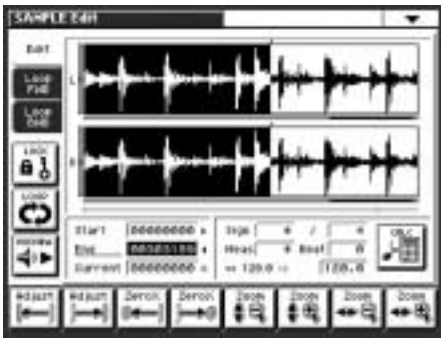
- The TimeTrip Pad is an innovative controller that lets you adjust the time parameter of a waveform in real time—without affecting pitch or formant—simply by dragging your finger along the pad. You can speed up a waveform, slow it down, or freeze it at any point.

Realtime Control

- Over dedicated 20 knobs and sliders allow you to grab essential sound parameters quickly and easily.
- The large and intuitive touchscreen interface responds to dragging motions, allowing you to access more detailed parameters – Imagine setting an envelope curve just by sliding your finger on the screen.
- The Twin D Beam controllers provide an expressive interface for controlling parameters in realtime

Programmable Arpeggiator

- Programmable arpeggiator provides additional rhythmic and timbral controls. It allows you to create complex riffs from a single note or chord, or to create timbral changes that add rhythmic motion to a sound.



Inputs and Outputs

- 1/4″ stereo Mic/Line analog as well as coaxial and optical digital sampling inputs
- Two 1/4″ Main and and two 1/4″ Direct outputs each configured as stereo paris
- Coaxial and optical digital I/O
- 1/4″ stereo headphone output
- Hold pedal and two Control Pedal jacks
- MIDI In, Out and Thru connectors
- A USB port allows .WAV/AIFF file import, .WAV export and MIDI interfacing with your computer.



VARIOS



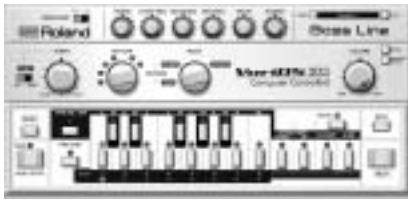
Open System Module

VariOS (short for Variable Operating System), is an open-ended hardware/software system that can be customized for different needs. In its first configuration, the VariOS module comes standard with the software program V-Producer. This software lets musicians manipulate a sample's pitch, time and formant—giving musicians the power to build complete audio-based arrangements with seamless software/hardware control and without draining the host computer's CPU. The VariOS module can also be updated via PC cards so the system can take on entirely new functions, making it the most open-ended and flexible hardware-based product ever.

FEATURES

- Open-ended realtime audio construction system consisting of VariOS module and new V-Producer for VariOS software.
- Play back up to 14 phrases simultaneously with 6-part multitimbral operation
- Award-winning VariPhrase technology for intuitive realtime manipulation of pitch, time and formant—control audio phrases like MIDI!
- Batch load/auto-encode .WAV/AIFF audio files from computer via USB and browser
- Internal memory holds up to 300 seconds of mono samples (150 seconds in stereo)
- Easy drag-and-drop sample arrangement—build a complete song in minutes!
- PhraseScope window provides free-form editing of a phrase's melody content
- GrooveScope window allows non-destructive reconstruction of rhythm loops
- Graphic editing of VariOS module's 40 multi-effects, 8 chorus types and 9 reverbs
- V-Producer for VariOS supports MIDI clock/MTC sync and audio/SMF export
- Flash memory and PC Card sample storage for using VariOS module without computer
- Open-ended design allows for future software updates





Firmware Updates

Two firmware sets with control software allow you to turn the VariOS module into a polyphonic Analog Modeling Synth (which can re-create Roland's vintage Juno and Jupiter synths) and monophonic Analog Modeling Bass Synth (to re-create a Roland TB-303 Bassline synth) respectively, but with modern conveniences like onboard effects and more.

V-PRODUCER SOFTWARE





Connections



- Stereo 1/4″ inputs (mic/line switchable) with variable gain
- Two pairs of stereo 1/4″ line outputs
- Optical and coaxial S/PDIF digital outputs
- 1/4″ stereo Headphones jack
- MIDI In and Out and a USB port.

ROLAND

SH-32

Desktop Synthesizer

The SH-32 Synthesizer is a nod to Roland's past with a look toward the future. This 4-part multitimbral synth uses Wave Acceleration Synthesis to deliver up to 32 voices of fat synth waves and entirely new waveforms. There's also a full complement of knobs and sliders—plus analog-style drums, DSP effects and a killer arpeggiator—giving musicians desktop access to a range of stunning synth sounds at an unbelievable low price.





FEATURES

Wave Acceleration Synthesis

- Wave Acceleration Synthesis provides a total of 67 waveforms including standard synth waves as well as new waveforms like Spectrum and Noise.
- There are also 63 rhythm waves available for rhythm sets
- 32-note polyphony and 4-part multitimbral operation.
- The internal preset memory features 128 Patches and two rhythm sets. User memory provides 128 Patches, Two Rhythm sets as well as 64 Performances
- Each Patch consists of up to two main oscillators and 2 sub oscillators with ultra-thick Subsonic mode. You can also apply pulse width modulation plus a choice of oscillator "hard" sync or ring modulation. Patches using oscillator sync will be monophonic.
- The Filter section features variable cutoff and resonance with a selectable Low Pass, Band Pass, High Pass and Peaking filters and a choice of -12dB or -24dB slopes.
- There are two ADSR envelopes. The first is a filter envelope which also features a depth control a key follow parameter. The second envelope controls amplitude.
- Two MIDI synchable LFOs, with a choice of 7 waveforms and variable depth, can be assigned to Osc 1, Osc 2, Filter, Amp (envelope) or pan.

Arpeggiator

- The onboard fully programmable arpeggiator can be used to create incredible user synth lines or drum patterns. The Arpeggiator can be programmed using step input or in realtime.
- There are a total of 64 Arpeggio styles and 64 Rhythm styles available.
- The Chord Memory function, with 64 Chord Forms, lets you produce complex arpeggios well suited for modern techno styles including trance. And with so many front-panel controls, programming is still ultra-intuitive.

Inputs and Outputs

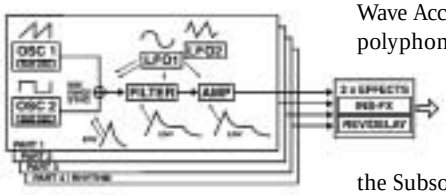
- 1/4″ Left/Mono and Right output connectors.
- 1/4″ stereo headphones output
- MIDI in and out connectors
- 1/4″ TRS footswitch input

Additional Features

- Tempo variable from 20.0 to 250.0 BPM.
- Three character LED Display
- Dimensions: 11 15/16 x 9 x 3 5/8″ (WDH)
- Accessories: AC Adapter (ACI Series/ PSB-1U)

Two High-Quality Effects Processors

- The first processor is a Loop (Send/Return) effect processor that gives you 10 varieties of reverb/delay effects.
- The second is an Insert effect processor that offers 35 multi-effects algorithms. These include everything from distortion, compressor and pitch shifter to proprietary Groove effects like Slicer, Lo-Fi and Isolator.



Wave Acceleration Synthesis overcomes the limited polyphony and sounds of modeling-based synths. You get up to 32 voices, plus 67 waveform variations from traditional analog waves to entirely new waveforms like Spectrum and variable Noise. And when combined with the Subsonic mode, you can create everything from thick bass sounds to entirely new sounds not possible on traditional synths. There's even drum maps with samples taken from the TR-808/909 Rhythm Composers.

B&H PHOTO - VIDEO - PRO AUDIO
MC
DISCOVER NOVUS
AMERICAN EXPRESS Card

A-37

MIDI Keyboard Controller



The A-37 MIDI Keyboard Controller is designed with today's MIDI musicians in mind. Whether controlling a rack of synths or a sophisticated computer-based setup, the A-37 puts you in control with 76 velocity- and aftertouch-sensitive keys, GM2 compatibility and an easy-to-use interface. And all of these practical features come in a cool-looking design that will appeal to your sense of style as well.

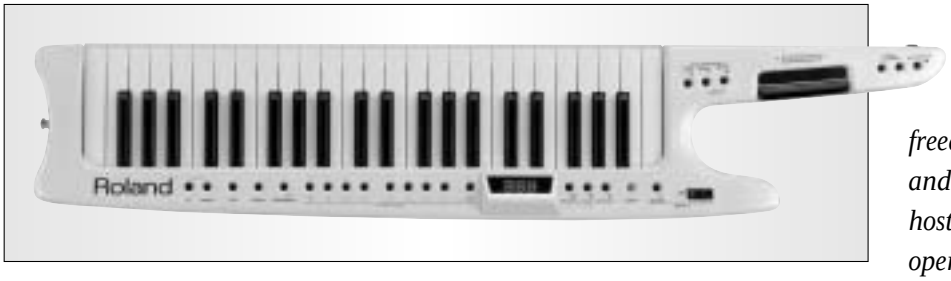
- Affordable 76-note controller keyboard for MIDI studios and performance
- Synth-weighted keys respond to velocity and channel aftertouch
- GM2-compatible; 128 User patches and 2 independent keyboard zones
- Easy to use with improved 7-segment, 3-digit LED and assignable Data Entry slider



- Convenient sequencer Start/Stop buttons, Clock TX and Transpose buttons
- Includes 2 independent MIDI outputs, MIDI In and Thru, plus Hold and Foot Pedal inputs
- Battery or AC powered

AX-7

MIDI Controller Keyboard



Building on the legendary design of the AX-1, the new AX-7 MIDI Keyboard Controller offers stage performers more freedom and expressiveness. Just strap it on and you're front-of-stage with the band. A host of controllers—including a D-Beam—opens up a new dynamic for live players, while the AX-7's attractive pearl white design looks great on stage. It's also very easy to use, thanks to a new LED display, expanded patch memory and GM2/GS compatibility.



- 5 realtime controllers: Data Entry knob, Touch Controller, Expression Bar, Hold button and light-sensing D-Beam
- 128 patches with data backup via MIDI
- Stage-oriented shoulder keyboard in attractive pearl white body
- 45-note velocity-sensitive keyboard—perfect for solos and bass lines
- Intuitive operation via new 7-segment LED display
- 2 MIDI zones for independent control of Upper/Lower parts
- GM2/GS Compatible
- Battery or AC powered

ROLAND

MC-909



Sampling Groovebox

The MC-909 Sampling Groovebox is Roland's ultimate tool for professional music production. Combining an expandable synthesizer based on a custom wave ROM with a professional sampler and sequencer, this revolutionary studio offers power, flexibility and unrivaled ease-of-use. A USB port makes it possible to exchange samples and Standard MIDI Files with a PC, while onboard effects and mastering processors give producers the ability to create "finished" tracks. Audio and MIDI can be integrated, synthesized and sequenced together within a single, easy-to-use environment.



FEATURES

An "SRXpandable" Sound Set

- 16-part multitimbral with 64-voice polyphony.
- 32MB of cutting edge dance-oriented ROM containing 800 Patches, 64 Rhythm Sets with 693 all-new waveforms exclusively sampled and created for this next-generation workstation. Each patch contains up to 4 tones.
- Sounds include everything from house pianos and organs to edgy techno hoovers and stabs—plus favorites like the JP-8000 "SuperSaw" waveform and distorted TB-303 sounds. There are also plenty of basses, pads and sound effects perfect for any track, along with the most complete assortment of dance drum kits ever assembled.
- As easy to program as an analog synth thanks to a generous supply of knobs, buttons and sliders.
- An expansion slot is provided for adding an optional SRX-Series Wave Expansion Board. The SRX-05 "Supreme Dance" expansion, in particular, features special Patches that can only be accessed with the MC-909.

Flexible Sampling

- 16MB of sample memory, expandable to 272MB using optional DIMMs.
- Audio can be recorded via the stereo analog inputs or digitally using the optical or coaxial S/PDIF inputs.
- Audio loops and phrases can be sequenced right alongside MIDI tracks as the MC-909 synchronizes your loops automatically.
- Samples can also be used as raw waveforms in the synthesizer section.
- Samples can be triggered chromatically via the velocity-sensitive pads or via an external MIDI keyboard.
- .WAV and.AIFF files can be imported from a computer via the onboard USB port.
- Upon completing your final mix, you can resample the output and export it back to your PC for CD burning.

Pro Sequencing

- 16 track linear and pattern ased sequencer with Tempo and Mute Control Track.
- 16-note drum machine-style keyboard with velocity-sensitive pads.
- 480 pulse per quarter note resolution and an approx. 1,300,000 note capacity
- Tempo is variable from 5 - 300 bpm
- Built-in "Piano Roll" style editing) allows you to to view multiple notes at once using the pads and LCD.
- Store up to 50 songs in internal memory
- 128 Preset and 128 User Arpeggio Styles
- Import and export Standard MIDI Files (Format 0/1) via USB.
- Patterns can be up to 998 measures

Chop, Stretch and Loop

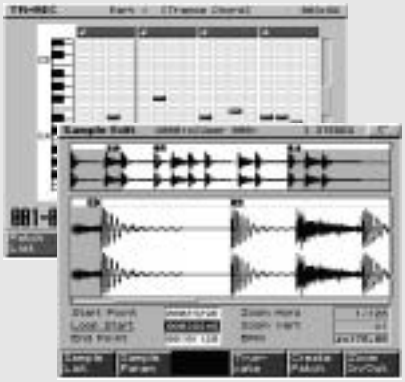
- Numerous graphic Wave Editing functions are provided including truncating and looping samples
- The Timestretch function allows you to change a loop's tempo, to match your song, without affecting pitch.
- The Chop functions provides the means of slicing up a loop into individual beats that can be re-arrange and re-quantized to create customized grooves

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ONE HOUR FREE PARKING
AT 349 W. 34th STREET (with purchase of $100 or more)

MC-909

Club-ready Presets with all of the Latest Dance Styles

- 215 Preset patterns, programmed by professional producers, fully exploit the onboard synth and drum sounds. These hard-hitting grooves include many popular styles like UK hard-house, trance and the latest"broken-beat" hip-hop and R&B. Use the presets as a starting point for your own grooves or play them "as-is."
- You can create or customize an additional 200 of your own patterns plus you can store another 999 to using an optional SmartMedia card.



Large Backlit LCD display with "piano-roll" and waveform editing screens

Studio Quality Effects

- A 24-bit reverb is available for applying to each sequencer part.
- A dedicated compressor allows you to get tight bass and drums without sacrificing a multi-effect.
- Two independent multi-effects (MFX) provide the means of taking the internal sounds and your own samples into new dimensions. These effects include a steep 36dB/oct filter and an amazing Step Filter as well as up to four seconds of tempo-synced delay, phasing, distortion and a host of other proprietary effects like Lo-Fi, Slicer and COSM Amp Modeling.

Dedicated Mastering Effects

- The dedicated Mastering processor includes a 3-band compressor, with dedicated knobs for each frequency band, for creating "final" mixes that have incredible clarity, definition and punch.

PRESET PATTERN LIST

Techno	1-22	Psy. Trance	1-7	G-Funk	1-11
Minimal	1-13	US HardHouse	1-7	Abstract	1-6
Hardcore	1-5	Progressive	1-7	Electro	1-6
Ambient	1-6	Garage	1-10	Reggae	1-5 (Dance Hall)
Drum'n'Bass	1-13	Disco	1-3	Reggae	6-9 (Lovers)
Break Beats	1-14	House	1-6	Reggae	10-11 (Roots)
Euro Trance	1-9	R&B	1-18	Reggae	12 (Dub)
Hard Trance	1-11	Hip Hop	1-13	...total 215 patterns	

SmartMedia Storage

- A SmartMedia slot is provided for storing User data such as samples, Patches, Rhythm Sets and patterns—all on ultra-thin, 3.3V SmartMedia cards from 8 to 128MB.

Easy Operation and Performance Control

- The large LCD screen makes it easy to locate sounds, sample audio, edit waveforms and sequence MIDI arrangements.
- Almost all major parameters are accessible via dedicated knobs, sliders and buttons.
- The dedicated Part Mixer provides real hands-on mixing.
- The Turntable Emulation function uses real-time timestretching to allow you to adjust pitch and time independently for MIDI or audio
- The twin D Beams use an infrared beam of light for controlling patches, tweaking filters and more.
- Tempo can be dialed in directly or tapped with the dedicated Tap Tempo button



Rear Panel Connections



- Six 1/4″ line outputs configured as 3 stereo pairs
- Two 1/4″ inputs: input one is mic/line switchable; while input two is line only.
- 1/4″ stereo headphone output
- MIDI In and Out connectors
- Coaxial and Optical S/PDIF digital inputs and outputs
- The onboard USB interface makes it possible to import or export .WAV/AIFF files with a computer, or even exchange Standard MIDI Files with software sequencers like Cubase, Logic and Performer.

ROLAND / BOSS

SP-303

Dr. Sample



The BOSS SP-303 Dr. Sample offers DJs and musicians a compact, easy-to-use portable phrase sampler with CD-quality sound, 8-note polyphony, 26 onboard effects and a real-time pattern sequencer–all at a price unmatched by the competition. For sampling and effects processing anytime, any place, the SP-303 can't be beat.

- Compact phrase sampler with 44.1kHz sound quality and 8-note polyphony
- Onboard sampling time expandable via optional SmartMedia cards
- DJ-friendly operation via large, luminous sample pads and three knobs
- 26 groove effects including new Vinyl simulation; internal resampling
- Realtime pattern sequencer with quantize function for easy song creation
- Selectable Gate, Loop and Reverse triggering of samples
- Accepts up to 64MB SmartMedia (optional) for sample/sequence storage
- Imports .WAV/AIFF files stored to SmartMedia cards. (Requires optional SmartMedia adapter or PCI adapter card.)
- AC adapter included



SP-505

Groove Sampling Workstation



The BOSS SP-505 Groove Sampling Workstation is the ultimate tool for working with samples, loops and laying down beats. This compact box gives you eight voices of CD-quality sampling, sophisticated sample editing tools including Chop, Pitch and BPM Sync, plus a 4-part sequencer for creating grooves using samples or onboard Tones. And with 26 killer DSP effects—plus an easy graphic interface with realtime control knobs—the SP-505 has everything groove musicians need to build cutting-edge tracks.

- Compact sampling workstation with onboard sequencing and effects
- 8 voices of CD-quality sampling, expandable up to 128MB via SmartMedia*
- Chop function divides loops and maps individual samples to pads
- Pitch function for playing back samples at new pitches as on a keyboard
- BPM Sync function instantly matches up to 16 phrases to the same tempo



- 64 onboard Tones with drums, bass, keyboard, synth sounds and more
- 4-part Pattern Sequencer with Quantize for easy song creation
- Intuitive graphic LCD (with waveform display) and three knobs
- 26 effects like Tape Echo, Isolator and Vinyl Simulator with resampling and realtime control
- Imports .WAV/AIFF files via SmartMedia*; Coaxial/Optical Digital inputs

**SmartMedia cards sold separately; SmartMedia adapter or PCI adapter card required for .WAV/AIFF conversion.*

DRUM MACHINES



DR-3 *Dr. Rhythm*

With the DR-3 Dr. Rhythm, guitarists and songwriters can add incredible-sounding drums and bass to their music. This ultra-affordable drum machine includes some of BOSS' best acoustic and electronic drum kits and a selection of bass sounds. Great for jamming, the DR-3's Rhythm Progressions and multiple variations make it easy to create a realistic-sounding drum track for any style of music. And with Total Sound Control, you can tweak your kits to perfection.

- BOSS' most affordable drum machine for jamming or practice
- Realistic drum, percussion and bass sounds with velocity-sensitive pads
- New Rhythm Progressions with 3 variations each for Intro, Verse, Fill and Ending
- New TSC with Sound Shape function and Ambience for professional drum sounds
- 100 Preset Styles and 100 User styles programmable via realtime or step recording
- Footswitch input for easy live control; runs on AC or battery power

DR-5 *Dr. Rhythm Section*

The DR-5 Dr. Rhythm Section is an innovative new concept in rhythm composers. Designed for guitarists, songwriters, live performers or anyone, it features a programmable backing band including bass, drums and keyboards.



- 256 16-bit percussion/instrument sounds, 64 drum kits (48 Preset /16 User)
- 400 patterns (200 Preset + 200 User)
- 20 songs
- External pitch function for triggering the internal sounds from a guitar
- 4-track composer, MIDI In/Out for interfacing to external devices
- Guitar In with built-in tuner and amp simulator



DR-670

Dr. Rhythm

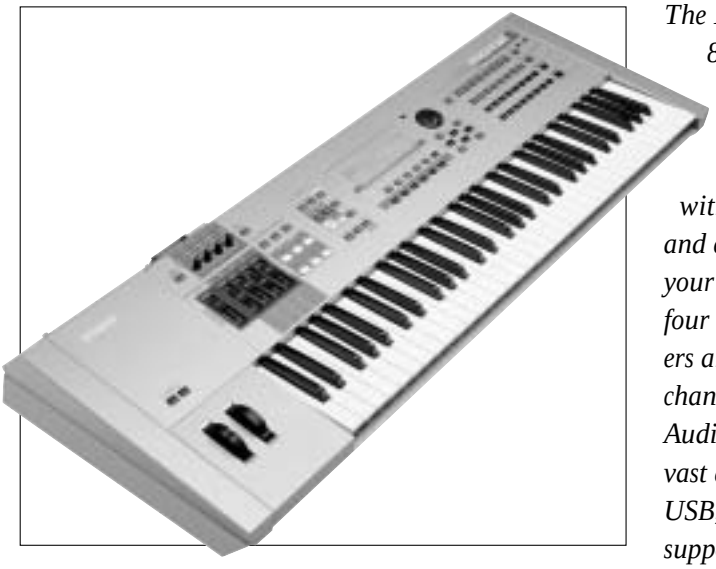
As the follow-up to BOSS' hugely successful DR-550mkII, the DR-670 Dr. Rhythm stands as one of the most feature-packed and affordable drum machines in BOSS history. This compact rhythm-creation tool offers all the acclaimed sounds from BOSS' flagship DR-770 model, while adding enhancements like velocity-sensitive pads, footswitch control and much more.

- Affordable BOSS drum machine with 255 superb-quality drum and percussion sounds
- Velocity-sensitive pads permit very expressive rhythm pattern programming
- New Bass part allows for programming bass lines using 16 different bass sounds
- 200 versatile preset rhythm patterns and 200 user-programmable patterns
- Large LCD for confirming pattern tempo at all times
- Foot control jack permits connection to optional footswitch for hands-free start/stop and fills
- Unique pad interface designed like a bass guitar neck for easy programming of rhythm/bass lines
- Direct Pattern Play function lets users assign internal patterns to pads for triggering "on the fly"
- Selectable battery or AC operation

YAMAHA

MOTIF 6 / MOTIF 7 / MOTIF 8

Music Production Synthesizers



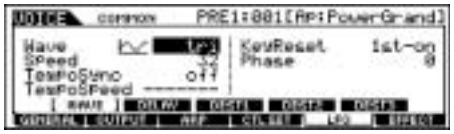
The MOTIF 6, 7, and 8 are 61- and 76-note synth action, and 88-note weighted action keyboards respectively. All three models are potent music workstations that feature the equivalent of 85MB of wave ROM, onboard sampling with 4MB of RAM (expandable to 64MB), a high resolution 16 track sequencer with a 200,000 MIDI note capacity, studio-quality effects processing and extensive expansion options for customizing the MOTIF to suit your style and taste. The MOTIF's real-time control modes allow the four sliders and real-time control knobs to function as 16 virtual sliders and 16 virtual knobs. These controls can even be used to as a 16 channel control surface for computer-based DAWs including Logic Audio, Pro Tools, Cubase , and Cakewalk. The MOTIF also features a vast arsenal of connectivity including dual 1/4″ inputs and outputs, USB, high-speed SCSI, and an expansion slot that can hold a card supporting Yamaha's mLAN interface. Or you can use that last expansion slot for the AIEB board which adds six assignable analog outs and S/PDIF. A Smart Media slot is also included to transfer data, using standard 3.3V media.

FEATURES

- The 61-key MOTIF 6 and the 76-key MOTIF 7 each feature synth action, FS keyboards with initial touch and aftertouch.
- The 88-key MOTIF 8 is a fully-weighted, balanced hammer effect keyboard with initial touch and aftertouch.

AWM2 Modular Synthesis

- 16-part multitimbral and 62-note polyphony (expandable with Modular Synthesis Plug-in System boards)
- 85MB of internal wave ROM (when converted to 16-bit linear format) containing many of the best waveforms from Yamaha's highly acclaimed S80, 9000 Pro, EX Series and RS7000 synthesizers as well as a wealth of new and improved waves.
- 384 Preset voices, 128 GM voices and 128 User programmable voices ranging from expressive stereo acoustic instruments like pianos, drums, strings and brass to never-before- heard dance sounds and lush ambient textures.
- Voices are comprised of up to 4 elements (waveforms) and have access to a myriad of editing functions including a resonant filter (21 types), multi-waveform LFO, amplitude and filter envelopes and more.
- 48 Preset drum kits; 1 GM kit and 16 User programmable drum kits
- You can also create up to 128 User Performances each containing up to four voices. Each of the four voices can be layered and/or split to create even rich and more expressive sounds.
- The Category Search Function allows you to quickly search and browse sounds via a selected voice category (such as pianos or strings).
- A Favorites category provides easy access to the voices you need the most.



Expandability

- The MOTIF's Modular Synthesis Plug-in System allows you to add up to three expansion boards, each of which adds a completely new, full-fledged synthesizer or sound processing "engine" to your arsenal.
- These expansion boards not only add new sounds, but you gain more polyphony and additional multi parts — without taking any processing power away from the keyboard itself.
- A diverse range of synthesizer expansion boards is available, including analog physical modeling, FM synthesis, vocal harmony processing and more.

Sampling

- 4MB of sampling memory expandable to 64MB using affordable PC SIMMs for a maximum total sampling time of 6 minutes and 20 seconds (stereo, 44.1kHz).
- Reads sample maps and voice parameters for Akai S1000/S3000 and Yamaha A3000/A4000/A5000 samplers. you can also import AIFF and WAVE files.

MOTIF 6 / MOTIF 7 / MOTIF 8

Integrated Sampling Sequencer

- The 200,000 MIDI note sequencer supports linear based recording as well as pattern-based sequencing which offers the flexibility of assembling your song as individual elements (Verse, Chorus, Bridge, Breaks, etc.).
- The Song Mode recording feature allows you to combine MIDI and audio data in a linear fashion from beginning to end.
- The Slice with Sequence function takes a recorded audio phrases and slices it into pieces (based on amplitude peaks and divisions of the beat) and then generate an extremely accurate MIDI sequence to perfectly re-trigger each of the pieces. This will allows you to rearrange or change the tempo of grooves without effecting the audio phrase's pitch.

Groove-Oriented, Real-time Pattern Chain Record

- The various MIDI and audio parts of your songs can be arranged in real-time using dedicated Section, Mute and Song Scene buttons. These buttons allow you to instantly audition different arrangement possibilities, and then record these performances as Pattern Chains. The Pattern Chains can then be edited using the detailed event editor, and upon completion, you can convert them back to a linear song formats like SMF.

Interactive Arpeggiator

- The Arpeggio features a an extensive range of rhythmic sequences, as well as "human" patterns such as the strumming of a guitar or the trilling of a flute. These performances were create by using real MIDI data recorded by real musicians.
- 256 Preset and 128 User arpeggitaor types are available.
- A velocity window which opens up unprecedented possibilities; for example, while playing a lead guitar voice, you could trigger a sampled MIDI guitar phrase on the fly by hitting a key hard.
- Arpeggio performances can be recorded into your sequences and you can import new arpeggios from Standard MIDI files.
- The Arpeggiator can be synced to MIDI

Effects

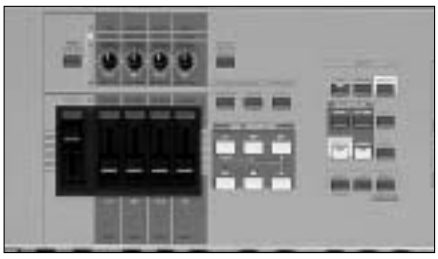
- There are 2 categories of effects processors available: System Effects; and Insert Effects.
- The System Effects section provides access to an independent Reverb (12 types) and Chorus (25 types) processor. A Send control allows you to dial in the desired wet dry mix for each effect. In Performance and Song modes, you also have access to 25 different Variation effects including amp simulators, wahs, sonic enhancers, LoFi effects and more.
- There are two Insert Effects available that can be used on a single voice or assigned to individually to different part, when used in Performance or Song modes. Insert 1 offers a choice of 25 effects types, while Inert 2 offers 104 types
- A Master 5-band Equalizer is also provided for fine tuning the overall sound before the final output stage.

SmartMedia Card Slot

- Easily store your samples on a SmartMedia Card Slot card and transfer them to your personal computer for editing. This storage method also makes it easy to save and load Standard MIDI Files.

Total Hands On Control

- 240 x 64 dot backlit LCD display
- Four physical sliders and four physical knobs are available on the front panel, with up to four functions available to each, can be assigned to edit the Reverb, Chorus, Pan, Cutoff, Resonance, Attack, Release and EQ as well as 4 user-assignable parameters.



- In Mix mode, these sliders and knobs work in conjunction with the Track Select buttons to give you all the capabilities of a 16 channel digital mixer.
- Mix mode can even be used as a real-time external control surface for your favorite sequencing software, including Emagic's Logic Audio, Steinberg's Cubase, Digidesign's Pro Tools and Cakewalk's SONAR.

Inputs and Outputs



- Two stereo pairs of 1/4″ analog outputs plus an optical S/PDIF digital output.
- Dual 1/4″ analog inputs with gain control are provided for sampling
- 1/4″ stereo headphone output
- MIDI In, Out and Thru connectors
- The Built-in USB MIDI interface is ready to connect directly to your iMac, G4 or newer G3 Macintosh computer as well as any USB-equipped Windows PC (Windows 98 or later).
- The 25-pin SCSI Port allows you to connect an external ZIP, MO, or hard drive to MOTIF for easy storage of large amounts of sample data. The SCSI interface provides fast loading times expected by performing musicians.

Optional Multi-Out Expansion Board

- The AIEB2 expansion board gives you 6 assignable 1/4″ analog audio outputs as well as both optical and S/PDIF digital outputs.

Connect to the Future of MIDI and Audio — mLAN

- Yamaha's mLAN protocol allows you to link together all your digital audio and MIDI data through a single high-bandwidth IEEE-1394 FireWire cable.
- MOTIF is ready for the latest generation of mLan expansions and future mLan products, so you'll be able to stay current with future possibilities.

YAMAHA

MOTIF RACK

128-Voice Sound Module



Perhaps the most requested Yamaha product of '02, the Motif Rack is more than just a single rack version of the award winning Motif synthesizer — its newly designed sound chip that delivers 128 note polyphony and enough DSP to accommodate 8 simultaneous insert effects, in 16 Part Multi Timbral Mode. The core Voices of the rack are identical to the keyboards with the addition of two new Preset Voice Banks and 256 User Editable Voices (twice the size of Motif). Two Yamaha PLG cards slots allow for expansion of polyphony, effects and synthesis capabilities. To insure complete connectivity, the Motif Rack offers six 1/4″ outputs (Stereo Main plus four individual assignable outs) as well as coaxial and optical digital outputs. Plus the Motif Rack comes with both standard MIDI and USB interfacing along with Voice Editing, File Utility, Multi Part Editing software, allowing you to take full advantage of your computer and the Motif Rack's MIDI capabilities The Motif Rack is perfect for both Motif owners who want to expand the power of their Motif system or for those of you who already have plenty other keyboards and just want the Motif sounds in a convenient rack mount size.

FEATURES

Single Rack Space Tone Generator

- A newly designed sound chip provides 128 notes of polyphony as well as up to 4 insert effects when used in the 16-part multitimbral Mix mode.

768 Preset Voices and 59 Preset Drum Kits

- 48MB of onboard wave ROM (84MB when converted to 16-bit linear format) with 896 Preset voices (including the General MIDI bank) and 65 preset drum kits – the core Voices of the Motif keyboard are all there plus there's an additional two new Preset Voice Banks.
- 256 User voices and 32 User drum kits

Effects

- System Reverb (20 types including 3 New "R3" type Enhanced Reverbs) and Chorus (44 types)
- Up to 8 Insert Effects (107 types) -2 for voice mode, 4 x 2 for Multi mode.
- Each of the 16 Parts has access to it's own 3-band EQ in Multi Timbral Mode.
- A 5-band Master EQ is also provided

Multi Library

- A Multi Library of 124 different Multis is provided, each with its own specially programmed effect, equalizer and other settings, and each designed particularly for a specific style of music or application.
- Since they are categorized according to music type, you can quickly and easily call up the settings you need.
- The Multi Libraries also serve as convenient templates to create new, original Multis – just select a Multi, then tweak it and customize it as needed.

User Interface

- An intuitive user interface with a large data wheel and dedicated function buttons provide quick access to all functions.
- The 160 x 64 dot backlit LCD provides clear and easily viewable display of all the Motif Rack's parameters.

Expansion

- Two expansion slots are provided for adding any of Yamaha's PLG Modular Synthesis Plug-in Expansion boards, resulting in new sounds, effects, additional notes of polyphony and new synthesis engines.
- Bundled with a Windows and Mac compatible CD-ROM that contains a dedicated Voice Editor and Multi-part editor.

Arpeggiator

- Onboard arpeggitor with 256 presets patterns.

Inputs and Outputs

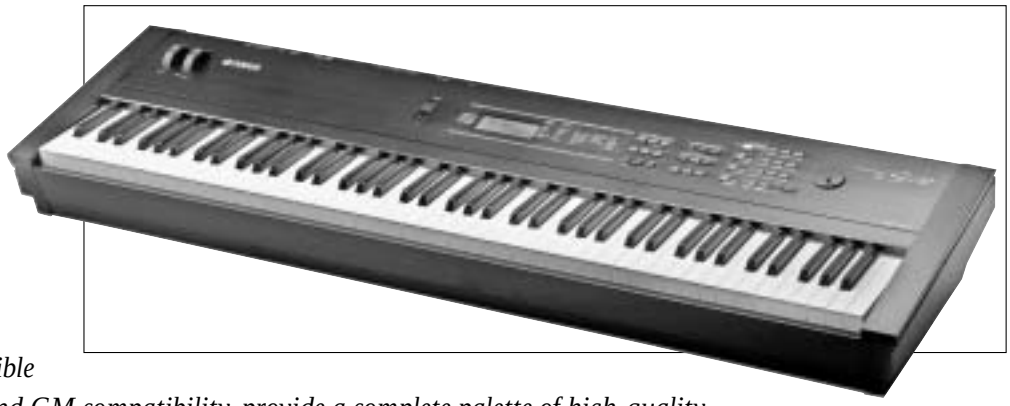
- Stereo 1/4″ main outputs plus four individual 1/4″ outputs.
- Optical and coaxial digital outputs
- 1/4″ stereo headphone output on the front panel
- Built-in USB MIDI interface plus MIDI In, Out, and Thru connectors



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EQUIPMENT LEASING AVAILABLE

S03 • S08

Performance Synthesizers



The 61-key S03 and 88-key S08 are versatile, yet highly cost effective, performance synthesizers with full 64-note polyphony and 25 MB of wave ROM (at 16-bit linear conversion). Yamaha's AWM2 tone generation and over 700 incredible sounding voices, including full XG and GM compatibility, provide a complete palette of high-quality sounds: stereo grand pianos, funky basses, analog pads and pounding drum kits. A selection of onboard effects, including chorus and reverb and variations such as distortion and "wah," may also be applied. You can create your own sounds (including analog-style synth sounds with the 64 resonant low pass filters) and store them in any of 128 user-voice memory locations. Panel switches, a large custom-designed LCD display and controllers that include pitch bend and mod wheels provide full control and access to all functions.

The portable S03 features 61 keys with Initial Touch, while the larger S08 features an 88-key balanced hammer design with Initial Touch. Both units feature left and right 1/4″ outputs, a stereo headphone out, MIDI In, Out and Thru connectors, as well as foot switch and foot controller inputs. Both units come bundled with Yamaha XGWorks and Voice editing software for both the PC and Mac. The S03 also offers a TO HOST serial port for direct connection to a PC or Mac. The S08 offers a USB port for computer interfacing and has a smart media slot for MIDI file playback and patch storage.



FEATURES

- 64 note polyphony and 16-part multitimbral
- 25.2 MB wave ROM (in 16 bit linear format) with over 700 sounds including: 480 XG/GM compatible presets, 128 additional Presets and 20 Drum Kits
- 128 User presets and 2 Drum Kits are also provided
- 3 Effects Blocks with 64 types
- 32 user-definable Multis

High Quality Digital Effects

- Independent Reverb (11 types) and Chorus (11 types) processors as well as a wide selection of Variation effects (42 types) such as delays and distortions

User Interface

- Large custom backlit LCD display makes it easy to see the names of voices and other parameters at a glance.
- Quick Access and Category Search
- Pitch bend and modulation wheels

The S08 adds

- 88 key weighted (Balanced Hammer) action Synthesizer
- SmartMedia slot provides an cost effective
- Playback Sequencer for playing Standard MIDI Files direct from SmartMedia
- USB interface

XGworks lite V3.0A

- The included XGworks lite software is a comprehensive yet easy-to-use MIDI and audio sequencing package that gives you everything you need to produce fully arranged songs with your S03, including 100 MIDI/audio tracks for recording, a built-in database of music styles, full editing and mixing capabilities and more.
- The Voice Editor features an intuitive graphical user interface, making it a snap to edit the keyboard's onboard voices.

Dimensions

- S03: 38¼ x 11¼ x 3¼″ (Weighs: 12.56 lbs.)
- S08: 52¾ x 15¼ x 6″ (Weighs: 44 lbs.)

YAMAHA

S30 • S90



Music Synthesizer

The S30 is an expandable synthesizer with a 61-note keyboard, full 64-note polyphony, 64 multi-mode resonant filters, effects, an arpeggiator and 24MB of wave ROM. On-board voices include 256 presets with stereo-sampled pianos, strings, brass, as well as a complete sonic palette of other high-quality sounds. You can create your own sounds and store them in any of 128 user-voice memory locations. There are also 128 user-storable performances for layers, splits and multitimbral sequencer setups. A plug-in expansion slot allows the S30 to accept Yamaha's PLG Modular Synthesis Plug-in Expansion boards, which essentially adds new synthesizers and effects processors to the system.



The S90 is Yamaha's most feature-packed S series synth to date. It features an 88-note weighted Balanced Hammer Effect keyboard and a synth engine that contains the best sounds of Yamaha's flagship MOTIF synth with an exquisite three-layer grand piano sample. The S90 is also equipped with three expansion slots for Modular Synthesis Plug-in System boards, so you can add new and exciting synthesis capabilities as your musical needs dictate.

FEATURES

- 61 key synthesizer with initial and after-touch
- 64 note polyphonic and 16 multitimbral parts (17 parts with PLG installed)
- 256 Preset voices and 8 drum kits
- 128 internal and 128 external User voices plus 2 internal and 2 external drum kits.
- 128 internal and 64 User Performances.
- A SMF playback sequencer is provided with direct play from Smart Media and chain playback for up to 100 songs.

Effects

- Dedicated Reverb (12 types) and Chorus (23 types) processors as well as an Insert effects (92 types)

Controllers

- 4 zone MIDI master controller functions with 128 memory locations.
- 5 assignable and 2 data entry knobs plus 4 assignable sound control sliders.

Expansion

- 1 slot for PLG series Modular Synthesis Expansion Boards
- 3.3 volt SmartMedia memory card slot for expansion and storage (cards up to 128 MB are supported)

Inputs and Outputs

- Four 1/4″ outputs: Stereo Main plus two assignable outputs.
- 1/4″ A/D inputs
- To Host serial Port for interfacing with a PC or Mac.
- MIDI In, Out and Thru connectors
- 2 foot pedals inputs, 2 foot switch inputs plus Breath Controller input

Arpeggiator

- The onboard arpeggiator provides full MIDI capabilities and offers 128 patterns, (from standard Up and Down patterns to catchy melodies and drum patterns)

Bundled Software

- CDROM with XGworks Sequencer, Voice editor and card filer included

The S90 Adds

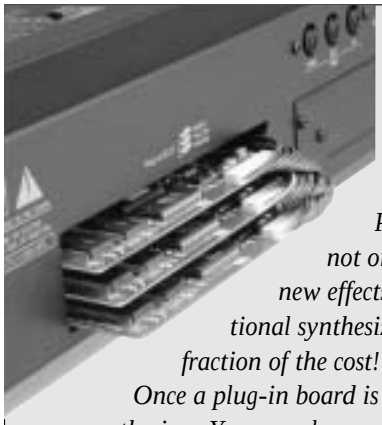
- 88 key weighted action keyboard with initial and aftertouch
- 110 MB wave ROM featuring a 25MB 3 layer S700 piano (in 16 bit linear format)
- 384 Preset voices and 48 drum kits plus 128 GM voices and 1 GM drum kit.
- 128 User voices and 16 drum kits
- 128 User performances
- 256 preset and 128 user Arp patterns
- 3 PLG slots for Modular Synthesis Expansion System Boards
- USB port for easy computer interfacing

Dimensions

- S30: 39 7/16 x 13 3/16 x 3 7/8″ (Weighs: 17.6 lbs)
- S90: 53 3/8 x 15 1/8 x 6 3/8″ (Weighs: 51 lbs.)

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EQUIPMENT LEASING AVAILABLE

PLG Plug-in Expansion Boards





Yamaha's Modular Synthesis Plug-in System is a revolutionary expansion system that adds a COMPLETE SYNTHESIZER or SOUND PROCESSING ENGINE to any Modular-Synthesis-Plug-in-compatible synthesizer. This amazing plug-in board lineup, which includes the PLG150 Professional Series and the PLG100 Standard Series, not only gives you new sounds, but also new synthesis architecture, new effects processors, extra polyphony and more. It's like placing an additional synthesizer or digital effects processor inside your existing keyboard—but at a fraction of the cost!

Once a plug-in board is installed, you can directly access any of its voices and edit them from the front panel of the host synthesizer. You can also use a PC-based software editor (bundled with each plug-in board) to edit any of the voices in detail. Many voice parameters are MIDI controllable, which, when used in conjunction with MIDI sequencing software, allows you to record realtime edits, such as filter cutoff frequency and effect levels, for automated voice transformation during playback.

PLG100 Standard Series

PLG100-VH

MIDI Controllable Vocal Harmonizing

• 4 types • Vocoder harmony
• Chordal harmony • Detune and Chromatic harmony

PLG100-XG *XG Wave Set*

• 32 note polyphonic • 3 effects blocks • 16 Part Multi-Timbral



PLG150 Professional Series

PLG150-PF

64-note Polyphonic Piano Tone Generator

• 16 MB Wave ROM • 132 voices

PLG150-DX

Vintage FM Synthesizer Plug-in Expansion Board

• 900+ preset voices • 16 note polyphonic • 64 custom voices
• Part EQ

PLG150-AN

Analog Physical Modeling Tone Generation

• 5 note Polyphony • AN1x compatible

PLG150-VL

Virtual Acoustic Plug-in Expansion Board Monophonic

• 256 preset, 64 internal, 6 custom voices • Compatible with VL70M

PLG150-DR *Live Room Drums*

• 80 preset kits (New Ambient Drums + Motif core kits)
• 8 User kits • Built in editable effects
• CDROM includes patterns, phrases and PLG editing software

PLG150-PC *Latin Groove Factory*

• 80 kits (Afro-Cuban + Brazilian Percussion)
• 8 User kits • Built in editable effects
• CDROM includes patterns, phrases and PLG editing software

YAMAHA

RM1X

Sequence Remixer



The Yamaha RM1x is a complete Dance Music Workstation that combines a powerful AWM2 tone generator with realtime performance controls and a professional suite of sequencing tools. An intuitive "hands-on" interface provides realtime control over synth parameters, pattern playback functions as well as the three, fully editable effects processors. The 32 note polyphony, 16 part multitimbral synth engine offers access to over 650 voices and 46 drum kits covering all major music styles. The RM1X's 16 track sequencer section delivers a professional-level timing resolution of 480 ppqn and a sizable memory of 110,000 note events. To help get you right into the groove, the RM1X provides a library of over 7,000 drum, bass, guitar, percussion and keyboard phrases organized into various styles including: Trance, Trip Hop, Hip Hop, Acid Jazz, House and Techno.

FEATURES

- MIDI synthesizer tone generator, professional sequencing, and a real-time MIDI controller surface.

700 Dance Oriented Sounds

- 16-part multitimbral, 32-note polyphonic, synthesizer with 654 sounds and 46 drum kits designed specifically for today's dance oriented music genres.
- In-depth voice editing parameters allow you to customize voices using a comprehensive range of envelope generator, LFO, filter and other parameters.

Styles, Patterns and Phrases

- 60 Preset and 50 User Pattern Styles each containing up to 16 Patterns for a total of 960 preset and 800 User patterns.
- Each pattern can playback up to 16 Phrases at a time, such as a bass line or kick drum pattern. You can create up to 256 phrases per pattern.
- You can easily mix and match a variety of rhythm, bass, and chordal/melodic phrases to create a virtually unlimited range of new patterns quickly and easily.
- The Patch function make it easy to select from the huge selection of 7,726 preset phrases.

16 Track Sequencer

- Powerful 16 track, 64-note polyphony sequencer allows you to record using a variety of record modes: real-time replace, real-time overdub, punch-in, step and grid and has a large 110,000 note memory along with a 480 pulse per quarter note resolution.
- You can easily record MIDI data via the two octave keyboard or an external MIDI keyboard using features
- Detailed sequence editing functions allow you to refine and shape your musical ideas with extraordinary precision.
- Because sequence data can be transmit via MIDI you can integrate outboard synths and samplers into your compositions.
- The Pattern Chain mode, allows patterns to be easily "chained" together for automatic sequential playback.
- Up to 20 Songs and 20 Pattern Chains can be stored in internal memory.
- The floppy disk drive allows you to load both type 0 and type 1 Standard Midi Files generated by other sequencing applications.
- Track data can be freely moved back and forth from Song mode (linear tracks) to the looped-based Pattern tracks.

Real Time Controller Knobs

- The two octave keyboard not only allows you to trigger MIDI events: it is also a real-time interface for accessing a range of functions including track selection, transposition, numeric data entry, track mute/solo, and section selection.
- Eight controller knobs provide realtime access to virtually every parameter including: synth parameters such as filter cutoff and resonance; envelope decay: pitch bend; effects and more
- Controller knob movements can be recorded in real-time, or applied to the sequence data and edited.
- The knobs can also be used transmit MIDI control change data for real-time control of external MIDI instruments and devices.
- Editing detailed sequence parameters, adjusting track pan or volume, or tweaking effect send levels is made easy thanks to the large 64 x 240-dot backlit graphic LCD and control interface including 4 multifunction knobs located directly beneath the display.
- Tap BPM entry lets you define tempos on the fly without entering specific numbers.
- The mute memory function lets you specify groups of tracks, which can then be instantly muted or un-muted, by pressing a single-key.

SU700

Sample Factory



The SU700 is a powerful and professional multi-functional music production center which is a hybrid combination of a phrase sampler, 42 track sequencer, tone generator and drum machine. It is an ideal workstation for the DJ and hip-hop artist as well as a valuable compositional, live performance and recording tool for the professional musician.

Three Fully Editable Effects

- A fully editable multi-effect system provides independent Reverb, Chorus and Variation effects processors that can be used to add anything from studio-quality reverb and subtle ambience, to delay, modulation, distortion, and an extensive range of other effects.
- There are 11 Reverb types, 11 Chorus types and 42 Variations to choose from.
- The dedicated Digital Low Boost function delivers earthshaking bass allowing you to boost and cut the main output signal ±24dB over a 50Hz to 2.0kHz range.

Play Effects

- An extensive Play FX section includes a number of real-time MIDI processes that can be applied dynamically to song data as it plays. These include: the harmonizing of parts with octave and unison parameters, clock-shifting to move instrument tracks forward or backwards in time, note-duration quantizing and the ability to modify note velocity offset.

Arpeggiator

- You can create your own arpeggios via an easy-to use grid interface.
- Arpeggio data can be transmitted via MIDI, and recorded to the internal sequencer.
- There are five different Arp types to choose from: (Up, Down, Alternate 1, Alternate 2, Random)
- You also have control over octave range and hold.

Inputs and Outputs

- Dual 1/4″ (L/Mono, R) outputs
- 1/4″ stereo headphone output
- MIDI In and Out connectors
- 1/4″ footswitch input

Quick and Easy Sampling

- Sampling times range from 44.1 kHz 16-bit stereo down to 11.025 kHz 8-bit mono
- Eight velocity-sensitive composed loop pads
- Includes 4MB of sampling wave memory expandable to 64MB.
- Sample from a CD-ROM or audio CD, transfer the audio information into the SU700, then edit it, chop it up and slice it any way you want.
- Adjustable data format (16-, 12-, or 8-bit).
- The Groove Function is a time-slicing function divides loops into discrete sections that can be individually manipulated to create new and interesting rhythms.
- Instantaneous Time compression and expansion without pitch change — simply assign a loop or phrase to one of the Auto Loop tracks and you'll instantly have a looping rhythm that automatically matches any selected tempo.
- 64-note polyphony
- Four types of resonant filters: Low Pass, High Pass, Band Pass, and Band Eliminate.
- An LFO with variable rate and depth can be assigned to a sample's pitch, filter or volume for adding tremolo, wah and vibrato effects.

Sequencing

- 42 track sequencer with Real Time and Step recording modes.
- Tempo is adjustable from 40.0-299.9 BPM, and has a resolution of 480 ppqn.
- Built-in 3.5″ floppy disk drive for saving songs.
- Up to 8 Audio Loops can be synced to MIDI clock.

Built-in Multi-Effects

- Access up to 3 simultaneous effects from a total of 43 high-quality types including: rich-sounding reverbs, warm choruses and MIDI sync'd delays as well as a wide selection of bizarre effects for creating unique textures.
- You can even process any external audio source, such a vocal mic, a keyboard, or a turntable, using the SU700's audio inputs.

Realtime Control

- 12 assignable knobs for realtime control of sound, groove, LFO, Filters, effects and more
- Use the Ribbon Controller to scratch a sample over your mix.
- The unit's sampling, looping, ribbon controller and multi-effects processing capabilities are perfect for the lo-fi, vinyl, scratch approach used by hip-hop music.
- You can store up to 8 scenes containing the machine's current knob settings, mute settings, and effect settings, and recall them all at the touch of a button.
- Bright Florescent Display

Inputs and Outputs

- Left and right 1/4″ line outputs
- Left and right 1/4″ mic/line inputs
- 1/4″ stereo headphone output with high output level
- MIDI In and Out connectors
- Optional SCSI interface

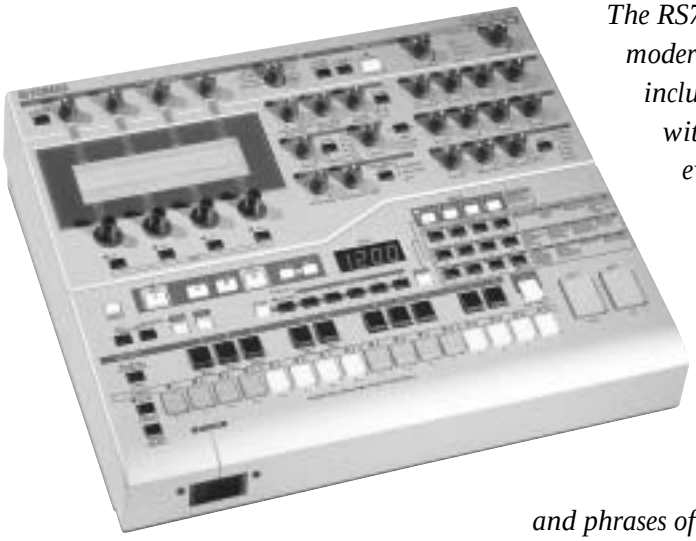
YAMAHA

RS7000

MIDI EQUIPMENT

1266

Music Production Studio - Integrated Sampling Sequencer



The RS7000 is a complete desktop studio designed to meet the needs of modern music production. It offers a unique and powerful feature set including sampling, a AWM2 tone generator, sequencing and effects with an intuitive user interface and real-time control of virtually every parameter. Perhaps the most unique function is the ability to seamlessly integrate audio and MIDI data. You can import samples from a variety of formats, sample actual sounds from external sources—or even created unique loops and phrases by re-sampling internally sequenced sounds within the RS7000 itself. The Integrated Sampling Sequencer (ISS) automatically breaks the samples into discrete parts (based on amplitude peaks and divisions of time) and generates a corresponding MIDI sequence. This allows loops and phrases of differing tempos to be synched together and processed using MIDI controls to change the Tempo, Swing or any other musical parameter controllable by MIDI. Add to this a plethora of studio and DJ-style effects, SCSI and SmartMedia storage and you realize that the RS7000 is the ideal nerve-center for your next remix, electronica or hiphop project.

FEATURES

On-Board Synthesis

- Internal 62-voice AWM2 tone generator with over 1,000 onboard ROM-based synth voices and 63 drum kit sounds.
- Six types of filters, including 24dB / 18dB / 12dB Low Pass, Hi Pass, Band Pass and Band Eliminate all controllable with a multi-stage envelope.
- The LFO section includes a number of BPM-sync'd waveforms plus a unique user programmable wave for creating unique BPM-sync'd effects.
- Sounds edits are stored as part of the song and pattern data, so a virtually unlimited range of user sounds can be created.

Sampling

- 4-MB of onboard sampling memory, expandable to 64 MB.
- Import samples in RS7000, Wav, Aiff, Yamaha A3/4/5000 and SU700, and AKAI S1/3000 format. Export RS7000 format, WAV
- Re-sample internally sequenced sounds to create new audio loops and phrases.

Advanced MIDI Sequencing

- The MIDI sequencer offers a choice of pattern-based recording with 16 sections of 16 tracks each, and linear sequencing with a 200,000 note-per-song capacity and a 480 ppqn resolution.
- Real-time recording with input quantize, step recording and grid style recording methods are all supported. Advanced editing macros help refine sequences.
- Different arrangements can be performed and auditioned in real time using the dedicated section buttons. All pattern and section data can be easily converted to linear sequence song format for final tweaking.

Effects Processing

- Three effects processors: Reverb (11 types), Delay/Chorus (22 types), Variation (92 types), plus a separate Master effect section (8 types) with a multi-band compressor, slicer, isolater, and other DJ-style effects.
- 4-band parametric Master EQ, plus 5 types of Track EQ (Lo and Hi, 1-band parametric EQ, Boost6, Boost12, Boost18)

Arpeggiator

- Type (Up, Down, Alternate1/2, Random), Sort, Hold, Octave range

Hands On Control

- 18 assignable knobs and two pads for subtle or radical real time control over sounds, samples and MIDI sequences.
- Knobs can be assigned to control the synth and sound parameters, onboard MIDI processors like MIDI echo, harmonize and beat stretch, or to standard MIDI CC messages for controlling external devices.

Inputs and Outputs

- 1/4″ stereo headphone output
- Dual 1/4″ inputs and outputs with 20-bit D-to-A and D-to-A conversion
- One MIDI IN and two MIDI Outs
- 1/4″ foot switch input

Storage

- SmartMedia Card Slot
- 50-pin half pitch SCSI port for accessing HD, removable and CD ROM drives

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MRT-3 Micro Rhythm Trak



Zoom's MRT-3 Micro RhythmTrak is an incredible stand-alone drum machine, and is also the perfect add on to any digital audio recorder that lacks a built in rhythm programming. It has a consistent appearance and functions with the MRS-4 4 track digital recorder. The MRT-3 features 7 velocity sensitive drum pads, 50 preset and 20 user drum kits (user can make an original drum kit), 200 preset and 99 user patterns, 99 song memory (maximum notes/events: 10,000), 21 note polyphony, quantizes to 96 PPQN, and tempos of 40-250 BPM.

Connections include a Mono out (Line) 1/4˝ phone jack, Stereo out (Line and Headphone) 1/4˝ stereo phone jack, a MIDI Input and a Control foot switch jack. The unit is powered by 4 "AA" batteries or an optional AC adapter. The MRT-3 links to the MRS-4 4 track, sending its output to the MRS-4's AUX input and is synchronized through the MIDI clock, acting like a built in drum machine.

- Works seamlessly with MRS-4
- 7 Velocity sensitive Drum pads
- 50 Preset, 20 User Drum Kits (User can make original drum kit)
- 200 Preset, 99 User Patterns and 99 Songs
- 10,000 Maximum notes/events
- 21 note Polyphony
- Quantize: 96 PPQN
- Tempo: 40.0-250.0 BPM
- Stereo out (Line and Headphone): 1 x 1/4˝ stereo phone jack
- MIDI In
- Mono out (Line): 1 x 1/4˝ phone jack
- Control In: 1 x Foot switch
- Optional AC Adapter: AD-0006D
- 4 AA-size battery

RhythmTrak RT-123 - Drum Machine

The functionality of RhythmTrak 234 in a compact body. Play cool backing patterns made up of two drum tracks and one bass track. 297 preset patterns provide a wide variety of popular rhythms. 99 additional patterns can be programmed and stored by the user.

- Create a backing sequence with up to 99 patterns. As many as 99 songs can be stored.
- Self-lighting pads indicate when sound is produced.
- Choose from a variety of 105 built-in super-realistic drum and bass sounds.
- LINE IN jack allows connection of external instruments.
- Simple and intuitive controls make it a snap to operate even if you have never used a drum machine or sequencer before.
- Groove play mode allows changing patterns during play simply by tapping pads on the unit.
- Optional FS01 foot switch allows pattern start/stop control or tempo switching. You can also operate an assigned sound such as bass drum or open/closed hi-hat.
- MIDI IN connector allows synchronization with external MIDI sequencer or control of the RT-123 from other MIDI equipment.



RhythmTrak RT-323 - Drum Machine



- 500 Rhythm Patterns (400 Preset + 100 User)
- 55 Bass Programs
- 128 Drum Kits (64 Preset and 64 User)
- Store up to 100 songs
- Assign 12-note Scale for Melodic Bass Patterns
- 13 Velocity Sensitive Backlit Pads x 3 Banks
- 32 Voice Polyphony
- Smart Media Storage Capability
- Edit PS-02 Drum / Bass Patterns
- Groove Play and Sound Jammer Slide
- MIDI In and Out
- (6 x AA) Battery or AC Adapter Operation
- Large Custom LCD easy-to-read Display
- Line Input / 2 Control Inputs / 2 Sub Outputs

YAMAHA

P60 • P60S

Digital Pianos



The P60 or P60S are the perfect pianos from some one who is just starting out and wants to learn to play piano or for the accomplished player who wants a very light weight, portable instrument.
It comes in two striking designs – the P60 in Black and the P60S in Silver.
It features the same the GH (Graded Hammer Effect) keyboard found on more expensive Yamaha digital pianos. The GH action of the P60 has the right keyboard feel to insure correct technique and inspire you to play more.
The hi quality AWM (Advanced Wave Memory) sampled sounds feature Grand Piano with 30 different sample points for an accurate digital emulation of the nuances of a Grand Piano. The built-in speaker system is specifically designed to reproduce all the tonal quality of the P60's voices including the built in digital reverb to add the ambience of a concert hall to your living room. The MIDI in/out/thru allow you to expand the P60's musical and educational capabilities by connecting it to a home computer.

- Graded Hammer 88 Note weighted action keyboard for realistic piano feel
- 32 Notes of polyphony
- 10 Voices including Grand Piano, EP, Pipe Organ and Strings
- Dual voice mode to combine two sounds
- Digital reverb
- 20W x 2 amplifier
- MIDI IN/OUT/THRU

P120 • P120S *Stage Pianos*

Whether you're practicing at home, playing with the band, or doing a live solo performance, the Yamaha P series offers a range of models with just the features you need. All boast superb acoustic and electric piano voices along with a number of other essential instruments such as organs, bass, and even strings. And thanks to their GH (Graded Hammer effect) keyboard, the action on these digital pianos is as close to the real thing as you can get. Of course, what else would you expect from the world's leading manufacturer of acoustic pianos. The voices onboard the P-120/P-120S are among the very best available on the market today. In addition to their rich and expressive stereo-sampled piano voices, they also feature a range of other essential instruments including electric pianos, harpsichord, organs and strings.



- 88 key weighted Graded Hammer action
- 64 note polyphony
- 2 Stereo acoustic and 2 electric piano voices, 2 organs, strings , bass, harpsichord, Clavichord and more each with Variation
- Split and layer modes
- Sustain pedal with half-pedaling capabilities
- 3 song 2 Track 10,000 note sequencer
- 2 effect processors
- PC interface
- Stereo Speakers with On/Off/SW (sub-woofer) switch
- 120 Black/Mahogany
- 120S Silver/Cherry
- Optional Stand LP120 & LP120S

P90

Portable Professional Piano



As the successor of the famous P-80 the P-90 recommends itself as a professional portable Digital Piano for the stage. Due to the triple Dynamic Stereo Sampling (DSS) and special sustain and key-off samples its piano sound is incredible natural and accurate. With its 64 notes polyphony the P-90 stands even long sustained passages. 24 Voices include not only piano sound, but organs, strings, choirs and basses, too. Using the Split or Layer functions two Voices can be played simultaneously - in two separated keyboard ranges (Split) or unison or layered (Layer). Eight digital effects do the polish on the sound. The Graded Hammer keyboard is responsible for the realistic touch - like on a acoustic piano the weight of the keys increases to lower notes. Its 2 headphones jacks and many internal demo songs make the P-90 ideal for music lessons. MIDI In/Out and the TO HOST interface recommend it as a master keyboard for large setups or computer aided music production. Power adapter, music rest and a sustain pedal are included.

- 88 weighted keys (Graded Hammer)
- 24 Voices
- 64 notes polyphony
- 8 effects: Room, Stage, Hall 1, Hall 2, Chorus, Phaser, Tremolo, Delay
- Dual/Split mode
- 2 track sequencer
- Build-in stereo-amplification
- 2 Headphones jacks
- MIDI In/Out, integrated TO HOST interface

P250 *Professional Stage Piano*



The P250 features three levels of Yamaha CFIII Grand piano stereo samples for dynamic selection according to your touch on the keyboard. The result is a delicacy of tone and expression that emulates the experience of playing a real piano. And to give the P250 an even more richly detailed sound, note off samples, string resonance samples and sound board samples simulate the acoustic interaction between different notes. The sound is unmatched, and so is the touch, thanks to the GH (Graded Hammer Effect) keyboard. The GH keyboard uses weights and mechanical linkages to achieve an entirely new level of realism. And to go even further, the key weight is graded from heavier in the lower registers to lighter in the higher, exactly the same as a grand piano's. To deliver the sampled instrument voices and 128 notes of polyphony with all the power and authority they deserve, the P250 features a new speaker system and high power stereo amplifier. The P250 features 16cm polypropylene woofers and a 30 watts per side stereo amplifier. The result is astonishingly lifelike power.

- Graded Hammer 88 Note weighted action keyboard for realistic piano feel
- 128 Notes of polyphony
- Acoustic Piano Voices with Three Velocity DSS (Dynamic Stereo Sampling), Note Off samples, String Resonance samples for realism and expressiveness
- 38 Panel Voices plus an extra 480 XG voices for a complete palette of sounds
- 16 Track Song Recorder with 223 Songs and 770 KB of Internal Flash Memory
- Digital reverb and digital effects, Dual mode and split mode, MIDI/TO HOST PC interface
- Built-in Speaker system with 16cm polypropylene woofers and 30 watts stereo amplifier

YAMAHA

PF500



Contemporary Keyboard

The PF500 brings the Yamaha's years of experience in creating great sounding digital pianos to a new market with the fresh contemporary design of the new PF series of products.
The Yamaha engineers built the ultimate piano sounds into the PF500: AWM with Dynamic Stereo Sampling (DSS). The PF500 features three levels of Yamaha CFIII Grand piano stereo samples for dynamic selection according to your touch on the keyboard. The result is a delicacy of tone and expression that emulates the experience of playing a real piano. And to give the PF500 an even more richly detailed sound, note off samples, string resonance samples and sound board samples simulate the acoustic interaction between different notes.
The sound is unmatched, and so is the touch, thanks to the GH (Graded Hammer Effect) keyboard. Unlike conventional digital piano keyboards that use springs or oil dampers to simulate a piano feel, the GH keyboard uses weights and mechanical linkages to achieve an entirely new level of realism. And to go even further, the key weight is graded from heavier in the lower registers to lighter in the higher, exactly the same as a grand piano's.
To deliver the new sampled instrument voices with all the power and authority they deserve, the PF500 features a new speaker system and high power stereo amplifier. The PF500 features 16cm polypropylene woofers and a 30 watts per side stereo amplifier. The result is astonishingly lifelike power.

FEATURES

The PF500 includes all the features you would expect from a Yamaha Digital Piano including:

- Graded Hammer 88 Note weighted action keyboard for realistic piano feel
- 128 Notes of polyphony
- 16 Track Song Recorder with 223 Songs and 770 KB of Internal Flash Memory
- 16 voice demo songs, 50 preset songs
- Custom 3-foot pedal (Damper, Sostenuto and Soft) with half pedal effect
- Selectable dual mode and split mode

AWM Dynamic Stereo Sampling

- Acoustic Piano Voices with Three velocity DSS (Dynamic Stereo Sampling), Note Off samples, String Resonance samples for realism and expressiveness
- 38 Panel Voices plus an extra 480 XG voices for a complete palette of sounds including strings, brass, woodwinds, synthesizers, drums and more
- Digital reverb, chorus and brilliance
- Bundled CD-ROM with USB driver, File Utility and demo software
- Built-in Speaker system with 16cm polypropylene woofers and 30 watts stereo amplifier
- MIDI In, Out and Thru, To Host and USB ports
- Two 1/4″ stereo headphone outputs
- Dual Aux Inputs and Outputs
- Pedal and Aux Pedal
- Keyboard stand, music rest and Three Pedals included
- Dimensions: Main unit with keyboard stand 56 5/16″ x 19 5/16″ x 30″ (92 lbs., 9oz)

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EQUIPMENT LEASING AVAILABLE

PF1000



The PF1000 features a weighted action 88 note GH keyboard , 96 notes of polyphony and three velocity layer Dynamic Stereo Sampling to provide the highest quality piano sounds and performance available on the market. It also features unique home entertainment features including auto accompaniment music styles and lyric and notation display.
Dynamic Stereo Sampling features three levels of Yamaha CFIII Grand piano stereo samples for dynamic selection according to your touch on the keyboard. The result is a delicacy of tone and expression that emulates the experience of playing a real piano. And to give the PF1000 an even more richly detailed sound, string resonance samples and sound board samples simulate the acoustic interaction between different notes.
The addition of styles adds instant musical enjoyment no matter the level of your musical skills. A new 240 X 340 LCD makes operation incredibly easy and offers a score display for more performance versatility and learning support than ever before. So it's easy for anyone to grow, learn and enjoy music on the New Yamaha PF1000. For a new level of performance versatility, Yamaha offers a huge selection of over 800 voices. Besides incredibly realistic Piano voices and XG voices, these include new Sweet, Live, Cool and Natural voices.
You get a choice of built-in Normal, Session, Pianist and Piano Combo accompaniment styles covering virtually any musical genre. What's more, each style has three intros and endings, four variations, four fill-ins and one break. Using these accompaniment styles lets everybody, even beginners, give accomplished solo performances. Music Finder offers an extensive range of music styles with pre-programmed voices and effects setups.

FEATURES

- Graded Hammer 88 Note weighted action keyboard for realistic piano feel
- 96 Note Polyphony
- Large 320x240 LCD for ease of use and for displaying musical scores and lyrics
- 194 Music Styles each with 3 Intros & Endings, 4 Variations and 4 Fills
- Tunings include: Flat/Stretch (Natural Piano Voice only) Equal Temperament, Pure Major/Pure Minor, Pythagorean, Mean Tone, Werckmeister/Kirnberger, Arabic 1/2

AWM Dynamic Stereo Sampling

- Acoustic Piano Voices with Three velocity DSS (Dynamic Stereo Sampling), Note Off samples, String Resonance samples
- Huge selection of high-quality voices (308 panel + 480 XG + 16 drum kits)
- 6 blocks of digital effects (Reverb, Chorus, DSP)
- Custom 3-foot pedal(Damper, Sostenuto and Soft) with half pedal effect
- Score/lyrics display function and video out jack
- Built-in Speaker system with 16cm polypropylene woofers and 30 watts stereo amplifier
- Two 1/4″ stereo headphone outputs
- Keyboard stand, bench, music rest and Three Pedals included
- Microphone Input
- MIDI In, Out, and Thru A & B
- To Host Connector for PC
- Dimensions: Main unit with keyboard stand 56 5/16″ x 19 5/16″ x 30 7/8″
- Weighs 94 lbs., 13 oz.

YAMAHA

PSR172



Portable Digital Keyboard

The PSR172 represents a remarkable value in a 61key, MIDI capable keyboard, complete with 100 voices and styles and educational components in the form of the Yamaha Education Suite. Finally, a portable keyboard in this price range designed specifically for the more demanding needs of Music Retailers. The PSR172 is perfect for the beginning music enthusiast on a budget. Yamaha quality at an affordable price.

- 61 Full-size keys
- 100 AWM voices
- Yamaha Education Suite 2
- Portable Grand
- DJ button
- MIDI In/Out
- 100 Preset rhythm styles
- 4 Multi-pads for drum sounds or musical phrases
- Multi-fingering auto accompaniment
- 100 “One Touch” settings
- 16 Note polyphony
- Large LCD screen
- 100 Demo songs with Music Minus One and Song Book
- Headphone jack
- Stereo speaker system with bass ports
- Music rest
- Sustain Pedal jack
- Power: 6 ‘D’ batteries or PA3C power adapter
- Bundled with Portable Keyboard Survival Kit

PSR273 *Portable Digital Keyboard*



The all new PSR273 is a breakthrough in sound quality for a keyboard at such a low price. For the first time in this price position, Yamaha has included its famous XG-Lite voice compatibility resulting in the very best sound while still being GM compatible. XG-Lite also results in more voices. A lot more voices!
The PSR273 is perfect for computer/internet use with a dedicated “PC” button that makes connectivity fast and easy. With GM compatibility, a touch sensitive keyboard, backlit screen, the Yamaha Education Suite, and more, the PSR273 is more value and performance than has ever been offered at such an affordable price point.

- First model in line that has a backlit LCD
- General MIDI compatible: Internet ready
- Touch Response: expressive touch like a piano
- XG Lite offers 100s of high quality Voices & DSP effects for better Internet compatibility
- PC Button: Memorizes your favorite computer setup
- “Portable Grand” button with stereo-sampled piano
- Yamaha Education Suite: helps you learn to play
- 100 Internal songs with Song Book
- 100 “One Touch” Settings
- 100 Styles
- 32 Note polyphony
- Dual & Split mode
- Transpose
- Headphone jack and Sustain pedal jack
- Music rest (included)
- Power: 6 ‘D’ batteries or PA3C power adapter

EZ150



Lighted Key Keyboard

The new EZ150 represent the second generation of lighted key technology. The EZ150 brings Yamaha sound quality, the learning support of lighted keys, and the Yamaha Education Suite to a new, lower, price point. With 100 built in songs, you'll be playing in minutes. The LCD display actually shows the notes on a staff so that you can see what you're playing. This second generation lighted keyboard takes all this form and function to a lower more affordable price point. It's all good.

- 61 Full-size lighted keys
- 100 AWM Voices
- 16 Note polyphony
- Yamaha Education Suite 3 with 100 Built-in Songs and Music Book
- Portable Grand
- MIDI In/Out
- 100 Preset rhythm styles with two variations each
- 4 Multi-pads
- Multi-fingering auto accompaniment
- 100 "One Touch" settings
- LED screen
- Headphone and Foot Switch Jack
- Stereo speaker system with bass ports
- Music rest
- Power: 6 'D' batteries or optional PA3C power adapter

EZ250i *Lighted Key Keyboard*



The EZ250i marks a new direction in Yamaha Portable Keyboards. Portables with MIDI connectors and General MIDI compatibility have always been a natural fit for computers and the Internet. The EZ250i brings that relationship even closer with the addition of a USB connector and associated software. EZ250i, with it's lighted keys and educational functions and features, is the first Yamaha portable to be bundled with Konami's Keyboard Mania software. Making playing the keyboard fast, fun, and easy, is what this combination is all about. Kids relate to the gaming and parents relate to the learning.

- 61 Full-size lighted keys with Touch Response
- "Keyboard Mania" learning game and Song Filer bundled with keyboard
- USB Connection for easy Computer Connectivity
- Flash ROM: Download new songs into the keyboard for learning
- Portable Grand
- 480 Voices
- GM/XG Lite Compatible
- Dual & Split mode
- PC Button: Memorizes your favorite computer setup
- 32 Note polyphony
- 38 Digital Effects
- Yamaha Education Suite 3
- MIDI In/Out
- 100 Styles with 2 variations each
- 6-Track Sequencer
- Multi-fingering auto accompaniment
- 100 "One Touch" settings
- 2 Registration pads (x 8 banks)
- Large Backlit LCD screen
- 100 Songs with Song Book
- Foot Switch Jack
- Headphone jack
- Two-way stereo speaker system with bass ports
- Music rest
- Power: 6 'D' batteries or PA3C power adapter

YAMAHA

PSR350



The PSR-350 comes complete with a remarkable new Music Database. Simply choose from over 200 song titles and the entire keyboard prepares itself for that song; it couldn't be easier. It also helps that the PSR-350 includes a beefed up version of the popular Yamaha Education Suite 2.
The sonic quality has also been ramped up with an incredible selection of 128 General MIDI voices, 480 Extended voices and significant increases in voice memory and new bass ports. Plus, the PSR-350 gives you the ability to write Standard MIDI files to disk so you can send music over the Internet for others to enjoy.

- 61 Full-size keys with Touch Response
- 2 Note Polyphony
- 4MB Wave ROM (PSR-340 X 2)
- Yamaha Music Database (208)
- 128 Panel voices + 480 XG Lite Voices + 10 DJ voices
- 38 Digital effects
- 26 Harmony types
- 3.5″ Disk Drive Writes Standard MIDI file to disk
- GM Compatible
- Yamaha Education Suite 2 with 100 built-in Songs
- Portable Grand
- MIDI In/Out
- 106 Preset rhythm styles with 2 variations
- 6-Track sequencer
- Multi-fingering auto accompaniment
- "One Touch" settings: One per style
- Large backlit LCD screen with icons and characters
- Headphone jack
- Footswitch jack
- 2-way stereo speaker system with Bass Ports
- Music rest
- Demo disk
- Power: 6 'D' batteries or PA5D or PA6 power adapter

PSR550



Yamaha PSR-500-series keyboards have always been powerful, now they're also colorful. The PSR-550 is one of the first portable keyboards in its price range to include a backlit, multi-color LCD display. The color changes by function to simplify operation. The EZ Navigator — Yamaha's great interactive help — has also been improved to make the PSR-550 still easier to use.
Yamaha's remarkable new Music Database makes playing songs a simple proposition as well. Choose from over 200 song titles in the database and the PSR-550 will set itself up for that song. Yamaha improved the PSR-550's sound over its predecessor by more than doubling the voice memory and by adding new voices and styles.

- 61 Full-size keys with Touch Response
- Multi color display with icons and characters
- 8MB Wave ROM Memory (PSR540 X 133%)
- Music Database (220)
- AWM Voices: 219 panel + 14 drum kits + 480 XG (includes GM)
- 74 Digital effects
- 22 Harmony types
- 3.5″ "Disk Direct" disk drive
- Pitch bend wheel
- 4 Multi-pads
- MIDI In/Out
- "To Host" connector
- 112 Preset rhythm styles
- 16-Track sequencer
- Multi-fingering auto accompaniment
- 448 "One Touch" settings
- 128 Registration memories
- 3 User styles
- "Direct Access" button
- 32 Note polyphony
- Improved "Easy Navigator" on-screen guide
- 9 Demo songs
- Footswitch jack
- Headphone jack
- 2-way stereo speaker system with Base Ports
- Music rest
- Demo disk
- Power: 6 'D' batteries or PA5D or PA6 power adapter

B&H PHOTO - VIDEO - PRO AUDIO
EQUIPMENT LEASING AVAILABLE

PSR1100 • PSR2100



The first thing you'll notice about the PSR1100 and 2200 are the displays. Large, back lit, bit mapped graphic displays are able to show lyrics, scores and tons of information available in seven different languages. Just put in a disk, push play and the song starts, the sheet music is displayed and the words to the song appear in time to the music. If you care to sing along, the PSR2200 includes a microphone input with vocal harmony technology. As the song plays through the PSR2200, you read the words in the display as you sing through the microphone, and the PSR2200 harmonizes with you in your voice in real time. Amazing. Improvements over previous models include double the voice wave ROM memory and new and improved style and voice content for spectacular sound, a USB connector for fast and easy computer connectivity, and improved operating functions. Added to the PSR1100 over its previous model is the ability to display scores like the PSR2200.

PSR1100

- Contemporary new look
- 61 Full-size keys with Touch Response
- Huge Bitmap Screen displays both Lyrics and Score
- USB Connection makes computer hookup a breeze
- Download & Play Scorch XF Files (Piano Score with Lyrics & Chords) from the Internet using built-in styles
- Double the Wave ROM and Flash ROM of the PSR-1000 translates to better sound and more User Memory for storing Songs, Registrations & other data
- First in the line with 4 Auto Accompaniment Variations & includes many new Styles
- Over 700 Realistic instrument voices including stunningly realistic "Sweet" and "Cool" Voices and 169 Preset Rhythm Styles
- 16-Track Sequencer
- Music Finder provides hundreds of keyboard setups by Song Name
- 32 Note polyphony
- 74 Digital effects
- 22 Harmony types
- 3.5″ "Disk Direct" disk drive
- Pitch bend wheel
- 4 Multi-pads
- MIDI In/Out
- Multi-fingering auto accompaniment
- 676 "One Touch" settings
- "Direct Access" button
- Help wizard in 6 Languages
- Footswitch and foot volume jacks
- Headphone jack
- 2-way stereo speaker system
- Includes Music rest, demo disk and PA300 Power Adapter

PSR2100

- More Wave ROM means Better Sound
  - 50% More Wave ROM than the PSR9000
  - 100% More Wave ROM than PSR2000
- Double the PSR2000 Flash ROM
- Organ Flutes with Virtual Drawbars
- Microphone input with Vocal Harmony
- 64 Note Polyphony
- New Styles
- Mod Wheel
- Large LCD with Notation & Lyrics Views
- Ultra Realistic Sweet, Live & Cool Voices
- 16-Track On-Board Sequencer - 32 Tracks via USB
- 61 Piano-Sized Touch Sensitive Keys for Expressive Playing
- 793 Instrument Voices and 181 Accompaniment Styles Play Any kind of Music
- Custom User style capability – create your own styles
- 3.5" Disk Drive to load/store songs, styles & setups
- Interactive help in 6 languages
- Stereo Two-Way Speakers with Bass Ports Create a Dynamic Sound System
- Music Rest Included
- 2 Footswitch Jacks

YAMAHA

PSR K-1



Portable Digital Keyboard with Karaoke

The Yamaha PSR K-1 is new from the ground up with innovations never before available in this price point and some never before available at any price. Start with the new, bitmap graphic display that, in addition to the standard information, displays both the score and lyrics during song playback. USB replaces the standard MIDI jacks and make connecting to a computer a breeze. The all-new "Performance Assistant" function allows you to play along with the song. You can't play a wrong note. Save your music to the smart media card, another first at this price point. And, for the singers among you, sing through the included, handheld microphone. There's something for everyone and every member of the family in the PSR K-1.

- First Portable Keyboard from Yamaha with a Karaoke Focus
- 8MB Smart Media card for song storage (included)
- USB equipped for ultra fast computer connectivity
- MusicSoft Downloader included on CD-ROM: Download new songs from the Internet
- Microphone with Digital Effects included
- Large FullDot LCD displays Score, Lyrics & Chords
- 61 Full-size keys with Touch Response
- Magic Performer: play chords along with XF songs - you can't make a mistake
- 480 AWM voices: General MIDI & XG Lite Compatible
- 46 Digital effects
- 26 Harmony types
- Portable Grand
- 100 Demo songs with Music Minus One and song book
- 6-Track sequencer
- 100 One Touch Settings
- Registration Memory
- 32 Note polyphony
- PC button
- 100 Preset rhythm styles with Multi-fingering auto accompaniment
- Dual/Split Voice
- Footswitch jack
- Two-way stereo speaker system with bass ports
- Headphone jack
- Music rest
- Power: 6 'D' batteries or PA3C power adapter

SURVIVAL KIT 2



Portable Keyboards are just that — portable. So if you're looking for the perfect accessory for your Portable Keyboard, this is a must-see. The Survival Kit 2 consists of a Yamaha bag that will protect your keyboard when you want to take it with you and a lightweight, but sturdy, X-Style stand for putting your keyboard on when you get where you're going. There's even a pouch for your music!

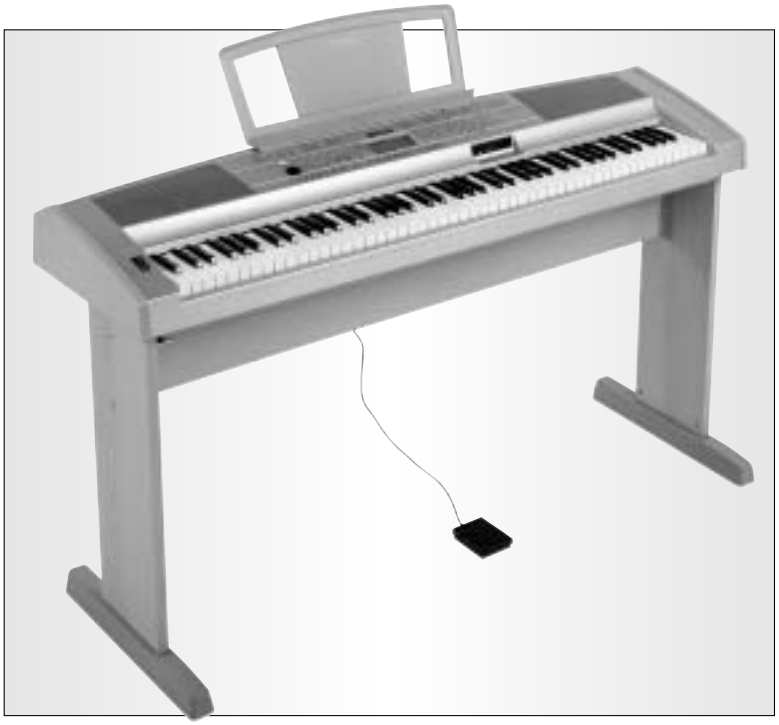
FEATURES

- Custom Yamaha keyboard bag
- Sturdy X-Style stand
- Padded shoulder strap
- Limited lifetime Warranty
- Bag dimensions: 46 x 20″

STAND DIMENSIONS

Height range: 25.63 x 38.75″
Length range: 31.50 x 11.0″ Width: 13.75″

DGX500



Portable Grand

The DGX500 is the first Portable Keyboard with piano keys and is the flagship in a new series of instruments from Yamaha called Portable Grand. It is packed with functions and features designed specifically for the new piano user. Included with the DGX500 is a furniture-type stand that completes its piano image. The Yamaha Portable Grand feature in the DGX500 is immediately apparent when you see the large, piano-shaped button in the center of the front panel. When you push it, the best piano is automatically chosen, the auto-accompaniment is turned off, putting the keyboard in full keyboard mode — just like a piano. A piano accompaniment style is automatically selected and is waiting for you if you choose to use it. Next to the piano button is a metronome-shaped button, giving you a built in metronome for practice.

- 88 Piano type keys with Touch Response
- Large Wave ROM features a New AWM (Advanced Wave Memory) Stereo Sample of the Yamaha Portable Grand
- Blue Backlit LCD
- Floppy Disk Drive for storing/learning new songs & loading new styles
- 619 AWM Voices: GM & XG Lite Compatible
- 46 Digital effects
- 26 Harmony types
- Yamaha Education Suite 2
- 208 Music Database Selections
- Pitch bend wheel
- MIDI In/Out
- 135 Preset rhythm styles with Multi-fingering auto accompaniment
- 6-Track sequencer
- 32 Note polyphony
- 100 Demo Songs with Music Minus One and Song Book
- Dual/Split Voice
- Headphone jack
- Two-way stereo speaker system with bass ports
- Stand and Sustain pedal Included
- Music rest
- Power: 6 'D' Batteries or PA5D

DGX300 *Portable Grand*

This is the keyboard that Yamaha customers have been asking for. The DGX300 has all of the incredible features of the DGX500 in a more compact 76-key body.* **Keyboard stand and sustain pedal optional.*



- 76 note touch sensitive keyboard
- 3.5″ disk drive records your settings and musical creations
- 135 upgraded styles
- 610 high quality sounds including a incredible sample of a real Yamaha Grand Piano
- 6 track recorder to create your own performances
- Music Database gives you instant access to a keyboard set up of a well known song
- Yamaha Education system will help teach you to play.
- PC ready button makes it easy to use with a home PC.
- Large Back lit Blue LCD screen
- New speaker system


DANCE MEGA
HOUSE/DANCE
Audio CD
1300 DRUML
DANCE MEGA
JUNGLE/RAVE
Audio CD
+450 MB
CD-ROM
1200 DRUMLOOPS
+250 JUNGLE & RAVE SOUNDS


Wild World
Of Animals
Wild World
Of Animals
01
02



B&H

PHOTO - VIDEO - PRO AUDIO

The Professional Audio SourceBook

Section 13

Sample Libraries

Big Fish Audio **1280-1295**

EastWest **1296-1321**

Ilio **1322-1343**

Sound Ideas **1344-1375**

BIG FISH AUDIO

R'n'B & HIP HOP

Abstract Hip Hop



Prepare yourself for the deepest Hip Hop CD on the market. Caspar returns with some deep, vibey turntable phunk, packing out this disc with pure Hip Hop and R&B nastiness. From the producers of Strictly Hip Hop, Strictly RnB & XTcFiles of Hip Hop, this joint is jammed with deep, groovy, dubby, smooth & jazzy construction kits, music loops, drum-loops, beats & breaks, sounds, chords, vocals, drums, guitars, bass loops, FX & tons of the phunkiest kicks, toms & percussion hits.

Format: Acid, AKAI S5000, AKAI Z4/Z8, Cakewalk, Cubase, Digital Performer, EXS24, GigaSampler, HALion, Kurzweil, Live, Logic Audio, ProTools, Reason, Sonar, Triton, Unity, WAV

Audio/WAV **89.95**

XTc Files of Hip Hop



Packed with phat breaks, sloppy fills, ghetto bass, funky licks, crackling Rhodes and wild-style FX. Plenty of loops and samples from the masters of ol' dirty Wu Boy beats* construction kits, master cuts, drumloops from 77-121 bpm, horror FX, guitars, strings, brass, bass, fills, vinyl FX, keys, pads, ambience, drum hits and more!

Format: Acid, AKAI MPC2000, AKAI S1000, AKAI S2000/3000, AKAI S5000, AKAI Z4/Z8, Audio CD Player, Cakewalk, Cubase, E-mu E4, Ensoniq, EXS24, GigaSampler, HALion, Kontakt, Kurzweil, Live, Logic Audio, Mac, PC, ProTools, Reason, Roland 700 Series, Sonar, Triton, Unity, WAV, Yamaha

Audio/WAV **89.95**

Wall of Vinyl 3



This 2-CD set SLAMS down the baddest Hip Hop, Rap and R&B - sure to hang your music on the wall in gold! Crackly, poppy, clean construction kits, hits, horns, piano, scratches, guitar, bass, synths, pads, strings, vocals, percussion, and even some xylophone!

"... this CD is worth the price of admission for its drum loops and breakbeats alone. . suitable for almost any style of music that relies on sampled loops of live drumming." 5 out of 5 - Remix

Format: Analog sampling devices; Software that can extract files from an Audio CD

Audio.....**89.95**

Vinylistics 3



Another massive amount of stretchin' & bendin' of ol' Ruffstyle breaks, keeping all the dust and dirt! Ruff Hip Hop and Funk breaks with that distinguishing Vinylistics sound... over 1,000 measures of new Hip Hop and R&B beats and breaks filled to the brim with loops and variations, plus the component drum samples that made the loops.

Format: Acid, AKAI MPC2000, AKAI S1000, AKAI S2000/3000, AKAI Z4/Z8, Audio CD Player, Cakewalk, Cubase, E-mu E4, Ensoniq, EXS24, GigaSampler, HALion, Kontakt, Kurzweil, Live, Logic Audio, Mac, PC, ProTools, Reason, Roland 700 Series, Sonar, Triton, Unity, WAV, Yamaha

Audio/WAV.....**89.95**

Strictly R 'n' B



Are you a remixer? A DJ? A producer of Soul? Do you need some smooth, dope loops? From the makers of XTcFiles of Hip Hop & Vinyl-istics comes Strictly R'n'B. 800 exclusive swing and soul, loops and samples with variations and all the drum samples that made each loop. A great mixture of slo & groovy loops from 60-100 bpm.

Format: Acid, AKAI MPC2000, AKAI S1000, AKAI S2000/3000, AKAI S5000, AKAI Z4/Z8, Audio CD Player, Cakewalk, Cubase, E-mu E4, Ensoniq, GigaSampler, HALion, Kontakt, Kurzweil, Live, Logic Audio, Mac, PC, ProTools, Reason, Roland 700 Series, Sonar, Triton, Unity, WAV, Yamaha

Audio/WAV **89.95**

Strictly Hip Hop



From the makers of Strictly R'n'B and Vinylistics, comes this new collection of the most groovy, freaky beats and breaks! Jam-packed with 'em from 59 to 112 bpm, plus variations and the drum samples that made each loop. Lush, slo grooves mixed with mad, hectic double-time beats, in Street, Hip Hop, R&B and Soul- get busy and make ya own wildstyle traxx!

Format: Acid, AKAI MPC2000, AKAI S1000, AKAI S2000/3000, AKAI S5000, AKAI Z4/Z8, Audio CD Player, Cakewalk, Cubase, GigaSampler, HALion, Kontakt, Kurzweil, Live, Logic Audio, Mac, PC, ProTools, Reason, Sonar, Triton, Unity, WAV, Yamaha

Audio/WAV **89.95**

B&H PHOTO - VIDEO - PRO AUDIO
EQUIPMENT LEASING AVAILABLE

BEATS

Big Beat



In the battle for the biggest beats, Big Fish has gone behind enemy lines to bring you a Mega-ton of them! We've plundered and pillaged our way across the war-torn landscape of modern Breakbeat dance music to bring you the fourth in the Megaton Bomb series. Big Beat is a 2-disc arsenal of the freshest, meanest, toughest sounds around- packed with insane construction kits, drumloops, hits, percussion, ruff-n-tuff background noises, vocals, ambient & atmospheric sounds from 85 to 150 BPM.

Format: Analog sampling devices; Software that can extract files from an Audio CD

Audio.....**89.95**

Breakbeat



Those absolutely authentic beats with the crispy vinyl sound used by top producers. The all time sweetest breaks along with every hit from every loop! PACKED with nothing but 4/4 beats- punchy kicks, crunchy snares & sizzling hi-hats. These extended beats can be dropped into anything! Breakbeats.. Extended Loops.. Scratchy Poppy Sections.. Shorter Loops..

Format: Acid, AIFF, AKAI MPC2000, AKAI S5000, AKAI Z4/Z8, Cakewalk, Cubase, Digital Performer, EXS24, GigaSampler, HALion, Kurzweil, Live, Logic Audio, ProTools, Reason, SampleCell, Sonar, Triton, Unity, WAV

Audio or AKAI MPC2000 or WAV/Acid.....**89.955**

Multi-Format: AIFF/SampleCell/Audio.....**129.95**

Breakbeat 2



We've raised the bar on Hip Hop drumloops with this one. All new, all fresh, all never-before-heard breakz for you beathedz and digital funk-masters. The disc features extended breakz with variations - no filler! If you're looking for the sound of a drum machine, look elsewhere; but if you want only hot, greazy, honey-suckin' beatz with the hits from every loop... you know who to call.

"If the urban thing is where you and your music live, you'll find tons of stuff ripe for picking... strictly for producers who want to add some fresh beats to their library...." - Keyboard

Audio.....**29.95**

Breakbeat 3



More nasty, dirty breaks, with all updated sounds. Live splice-n-dice drums, with subtle electronic sounds. Perfect for hip hop, R&B and dance. Contributions come from major label recording artists Jamie Myerson (Columbia Records), as well as up-n-coming producers Alan Moore and Sean Dvorak. Tempos from 70-126, emphasis in 80-100 range. Long loops of 8-16 bars, variations every two bars, hits at the end of each, plus a section of 2-bar loops.

Format: Acid, Cakewalk, Cubase, EXS24, GigaSampler, HALion, Live, Logic Audio, ProTools, Reason, Sonar, Unity, WAV

Multi-Format: Audio/WAV/Acid.....**89.95**

Hit Zone



From deep inside the war-riddled Pop Chart bunker come the beats that'll bump the charts for a thousand years. Timbaland to Max Martin, Neptunes to Jimmy Jam & Terry Lewis, that clean, defined R&B/POP sound – a true Top 40/R&B/Hip Hop assault. Basses, beats, keys, licks, guitars, fx, and more. Ready for use in your sampler or software.

Format: Acid, AKAI MPC2000, AKAI S1000, AKAI S2000/3000, AKAI S5000, AKAI Z4/Z8, Audio CD Player, Cakewalk, Cubase, E-mu E4, Ensoniq, EXS24, GigaSampler, HALion, Kontakt, Kurzweil, Live, Logic Audio, Mac, PC, ProTools, Reason, Roland 700, Sonar, Triton, Unity, Yamaha

Multi-Format: Audio/WAV/Acid.....**99.95**

Hardcore Breakbeat Action



Construction kits with big bonus section of drumloops from HUGE producers. Aimed at the breakbeat/big beat producer, with a focus on nu-school breaks, as well as old-school hardcore (i.e. fast-tempo breakbeat). Modern style elements of hip-hop, breakbeat and drum-n-bass, elements of happy and dark hardcore, and the occasional guitar-based riffs of big beat. Incorporates both the west coast funk-breaks sound, as well as the hi tech sound of the UK.

Format: Acid Cakewalk, Cubase, EXS24, GigaSampler, HALion, Kontakt, Live, Logic Audio, ProTools, Reason, Software that can extract files from an Audio CD Sonar, Unity, WAV

Multi-Format: Audio/WAV/Acid.....**89.95**

HARD CORE/NU METAL

Black Butta 3



If you thought the first two were great... you better sit yourself down. Madjef is back! He's been listening to all your feedback from the first two volumes and has raised the stakes for volume 3! Black Butta 3 BUSTS out with two CDs of brand new Hip Hop and R&B loops. Beats, separate kicks, snares, hats, hits, wet, dry, scratches, basslines, guitars, percussion, keys, synths, Rhodes, FX and more, all in construction kit format. This hot, brand new CD has it all! Don't waste anymore time. Pick it up!

Format: Analog sampling devices; Software that can extract files from an Audio CD

Audio **89.95**

R&B Backline



Chicago has a rich African-American music tradition and has been the birthplace of many soul and R&B groups over the last fifty years. Features loops from Chicago's best rhythm section. Chuck Webb on bass (David Sanborn, Grover Washington Jr., Ramsey Lewis, Joe Sample & many others) and Khari Parker who currently has the hottest drumming gig in the business - Destiny's Child. Funky, laid-back bass & drumloops in construction kit format, with section variations.

Format: Acid, AKAI S5000, AKAI Z4/Z8, Audio CD Player, Cakewalk, Cubase, GigaSampler, HALion, Kontakt, Kurzweil, Live, Logic Audio, Mac, PC, Reason, Sonar, Triton, Unity, WAV, Yamaha

Multi-Format: Audio/WAV/Acid **89.95**

Electric Ghetto



Electric Ghetto is the toughest Rock/Rap/Industrial disc ever produced and only followers of the likes of Limp Bizkit, Korn, Ministry, NIN and Zombie will be pleased by the groove-grinding madness of these sounds. Two audio CDs and 2 Acidized .wav CD-ROMs of extended construction kits that develop and change for ultimate variation, self-contained songs, guitar, synth, bass, guitar riffs, Hip Hop beats, creepy atmospheres and more crazed mayhem sure to set fire to your mix! Insert this nastiness only if you dare.

Format: Acid Cakewalk, Cubase, EXS24, GigaSampler, HALion, Live, Logic Audio, ProTools, Reason, Sonar, Unity, WAV

Multi-Format: Audio/WAV/Acid **89.95**

King Cortez



Drum legend, Jody Cortez (Boz Scaggs, Crosby Stills & Nash, Joe Cocker, Elton John, Jackson Brown, Christopher Cross, Billy Idol, Mike McDonald, Steve Lukather, Cinderella, Jody Watly, Chicago, Woodstock 94, the Porcaro Brothers...) has laid down the most infectious disc of grooves. Assorted sexy R&B, Rock, Hip Hop, Rock Funk, Latin, Jazz, BeBop, Afro-Latin, Blues, Reggae, Dance, Alternative Hip Hop and many more grooves. Live drums, loops, fills, kicks, snares, toms, hats..

Format: Acid, Audio CD Player, Cakewalk, Cubase, EXS24, GigaSampler, HALion, Kontakt, Live, Logic Audio, ProTools, Reason, Software that can extract files from an Audio CD, Sonar, Unity

Audio/WAV/Acid.....**44.95**

L.A. Riot 4



The LA Riot team is back in it! This time they're breakin' out with a 4- CD set of the stuff to make you rich and famous. Producers of Hip Hop, R&B, Drum&Bass and Big Beat will find it ALL in here for the mixin' of some smokin' traxx! 50 construction kits, over 300 exclusive beats and breaks (from Chronic Music's secret stash), drum fills, bass riffs, vocal parts, accapella raps, freestyle raps and hooks, hooks, hooks! Keyboards, guitar and vocoder parts, horn riffs, tons of drum kits, cymbals, kicks, snares, bass tones and some messed up scratches!

Format: Analog sampling devices; Software that can extract files from an Audio CD

Audio.....**89.95**

L.A. Riot 5



Over 80 Nu Skool and classic construction kits. Nu Skool, live drum breaks, stop-start drum breaks, scratches and found sounds, drumloops, new bomb banks, Ghetto vocals / vocal hooks, Diva vocals, Ghetto talk vocals, raps, Gospel organ master riffs, guitar licks, funky/bizarre guitar, scratches, basslines, FX.

Format: Acid, AKAI MPC2000, AKAI S1000, AKAI S2000/3000, AKAI S5000, AKAI Z4/Z8, Audio CD Player, Cakewalk, Cubase, E-mu E4, Ensoniq, EXS24, GigaSampler, HALion, Kontakt, Kurzweil, Live, Logic audio, Mac, PC, ProTools, reason, Roland 700 Series, Sonar, Triton, Unity, WAV, Yamaha

Audio **89.95**

HARD CORE / NU METAL

Ballistix



Phat, phunky, uptempo nu school breaks and chemically treated music loops & phrases. 750 loops & samples on the revolutionary Tc WAV CD-ROM! 500+ uptempo phunk driven drum loops and loop elements, 300 music, bass & fx loops. Over 630 Mb of sounds . . . a massive 1700+ loops and samples, grouped into 3 categories: Ballistic Breaks: a massive folder of extended super cool breaks each with 6 variations and fills; Chemical Treats & Music Loops: hardcore music & bass loops, fx & turntable loops; Phatboy Drumtools: 3 folders of the best drum samples.

Format: Acid Cakewalk, Cubase, EXS24, GigaSampler, HALion, Live, Logic Audio, ProTools, Reason, Sonar, Unity, WAV

Audio/WAV **89.95**

Cut'n It Up 2



Packed to the gills with more brand new, all license-free sounds- construction kits, drum hits like no other, breaks, bass, guitars, keys, organs and more of John Tejada's unmatched scratching madness! A 4 disc set (2 audio, 2 WAV/ACID) of raw Hip Hop and R&B sounds. Insert these sounds into you mix and see for yourself what all the hype is about.

"Loads of material... Excellent programming... Lots of variation in extended play loops" - Keyboard

Format: Acid, Cakewalk, Cubase, EXS24, GigaSampler, HALion, Live, Logic, Audio, ProTools, Reason, Sonar, Unity, WAV

Multi-Format: Audio/WAV/Acid **89.95**

Def Jef Kontrol Freaks



After years of producing the hottest artists like Krayzie Bone, Snoop Dogg, The Eastsidaz and Shaquille O'Neal, multi-platinum producer Def Jef has produced a breakbeat CD. His assembled multi-talented production team dubbed "Kontrol Freaks," whose members' production credits range from Public Enemy's front man Chuck D to the multi-platinum sensation TLC, has composed some of the hottest and most authentic hip hop tracks, drum sounds and loops from their extensive library for you to use.

Format: Acid, Cakewalk, Cubase, EXS24, GigaSampler, HALion, Live, Logic Audio, ProTools, Reason, Sonar, Unity, WAV

Multi-Format: Audio/WAV/Acid **89.95**

Dubb Trax



Loops, back spins, kicks, snares, claps, hi-hats, cymbals, keyboard lines... Just trax, plus a huge section of KICKIN' hits* this brand new collection was designed by world-famous producers Taj Mahal and C-DUBB for the serious gangsta rap, hip hop and R&B artist who need a beat on-the-spot. Need a cut for a demo track? Need that snare or kick you just can't get anywhere? Snatch this! No breakouts so there's plenty of room to jam 96 separate trax into this essential CD for the serious producer. Don't miss the this phat, delicious collection of Hip Hop ecstasy!

Format: Audio CD Player; Software that extracts files from Audio CD

Audio **89.95**

Off the Hook



The smoothest, the sexiest, the absolute sweetest R&B grooves ever. The hottest tracks with today's vibe- just like you like it and made by professionals who know what's what! These loops are ready for the pickin'... just spread your sweet lovin' on the mix and watch it flow.

"The drums, synths, keys and guitars are well played and programmed throughout... Very smooth." - 8 out of 10 Computer Music.

"The one word that kept coming to mind... was smooth." - Electronic Musician.

Format: Analog sampling devices; Software that can extract files from an Audio CD

Audio **89.95**

Off the Hook 2



When Off the Hook was released, no one could stop the tidal wave of producers from using every inch of the set. After thousands of requests for a sequel, producer Anthony Myershas produced a package that rivals even Volume 1! Nasty R&B construction kits, beats, hits, sounds and more await your skills and your music. It also includes a demo version of Abelton's Live, so there's no stopping. You better snatch this one up quick, cause those who do, are gonna be smellin' the platinum!

Format: Acid, AKAI S5000, AKAI Z4/Z8, Audio CD Player, Cakewalk, Cubase, E-mu E4, EXS24, GigaSampler, HALion, Kontakt, Kurzweil, Live, Logic Audio, Mac, PC, Sonar, Triton, Unity, WAV, Yamaha

Multi-Format: Audio/WAV/Acid **89.95**

BIG FISH AUDIO

NOIZY NIGHTS

Atomizer



Finally, Atom releases his wild world of digital percussion on an unsuspecting public. Looking for typical drumloops & breakbeats? Keep looking. But if you want something so bizarre your friends will look at you funny, this is the disc. atom divided it into 9 categories, based on his decade of recording styles & aliases- Slo Motion- Downtempo & Atmospheric Beats; supertropical- a fusion of Latin American grooves & digital abstractness; scientific freestyle- odd time signatures & quantizations. You get the idea.

Format: Analog sampling devices; Software that can extract files from an Audio CD

Audio.....**89.95**

Methods of Mayhem II



The sequel to the legendary Methods of Mayhem CD. Prepare your ears for another onslaught of grinding guitar riffs, spooky soundscapes, darkish drum loops, booming basslines and nasty noises in general. Hundreds of drum and noise loops, 300 drum sounds, more than 270 savage guitar and bass guitar riffs, eerie soundscapes, background noises, etc.

Format: Acid, AKAI MPC2000, AKAI S1000, AKAI S2000/3000, AKAI S5000, AKAI Z4/Z8, Audio CD Player, GigaSampler, HALion, Kontakt, Kurzweil, Live, Logic Audio, Mac, PC, ProTools, Reason, Rex, Roland 700, Sonar, Triton, Unity, WAV, Yamaha

Multi-Format: Audio/REX/WAV.....**89.95**

Noize Loops



NOISE (noiz) n.: Sound or a sound that is loud, unpleasant, unexpected, or undesired. ... - gut wrenching brutal drum loops! Contains drumloops which have been manipulated and tweaked to the extreme. If you like bands like NIN, Coil, Aphex Twin or Atari Teenage Riot, you will love this CD. Crazy futuristic sounds, Mad loops, ambient drones & "Orch Drones".

Format: Acid, AKAI MPC2000, AKAI S1000, AKAI S2000/3000, AKAI S5000, AKAI Z4/Z8, Audio CD Player, Ensoniq, EXS24, GigaSampler, HALion, Kurzweil, Live, Logic Audio, Mac, PC, ProTools, Reason, Roland 700 Series, Sonar, Triton, Unity, Yamaha

Audio/WAV.....**89.95**

pHATfactory 001



A brand new loop-oriented virtual instrument, built on the award-winning pHATmatik PRO loop resynthesis engine. Unlike a standard loop library, pHATfactory loops can play back at any tempo, perfectly synced to any VST 2.0 compliant host software. Twist loops beyond recognition with included DSP & synthesis tools.

Format: Acid, AKAI S5000, AKAI Z4/Z8, Cakewalk, Cubase, EXS24, GigaSampler, HALion, Kontakt, Kurzweil, Logic Audio, Mac, PC, ProTools, Reason, Sonar, Triton, Unity, VST, WAV, Yamaha

Multi-Format: pHATmatik/Plug-in/WAV..... **129.95**

Things That Go Bump In The Night



Ten of the top sound designers and electronic music artists of our time, including Atom Heart, Robert Rich, Sliver, Thomas Schobel and control-X. Top-notch sound design with a rich, dark undercurrent, comprised of pulsar data, proprietary synth engines, incredible processed effects, pitched and percussive loops, crescendos, ambient beds and more!

Format: Acid, Aiff AKAI MPC2000, AKAI S5000, AKAI Z4/Z8, Cakewalk, Cubase, Digital performer EXS24, GigaSampler, HALion, Kurzweil, Live, Logic Audio, ProTools, Reason, SampleCell, Sonar, Triton, Unity, WAV

AIFF/SampleCell/Audio or AKAI S1000/Audio.....**169.95**

Audio.....**89.95**

Things That Go Bump In The Night 2



Building on the trend-setting chaos that was Things That Go Bump in the Night, "Bump 2" brings even more producers & more weird sounds, for a sequel that outshines even the original! Atom Heart, Plug Research Records, The Rip-Off Artist and more top-notch producers have assembled heaps of off-kilter percussive loops, subsonic drones, poisonous atmospheres & more madness. If your music or score is crying for more, then come out from under the covers.

Format: Acid, AKAI MPC2000, AKAI S5000, AKAI Z4/Z8, Cakewalk, Cubase, Digital Performer, EXS24, GigaSampler, HALion, Kurzweil, Live, Logic Audio, ProTools, Reason, Sonar, Triton, Unity, WAV

WAV/Acid or Audio.....**89.95**

AKAI S1000/Audio.....**169.95**

B&H PHOTO - VIDEO - PRO AUDIO

GROOVY

X-Static Goldmine 5



1500 pure, artificial loops and samples! Electronico! Jampacked with synthetic loops, FX & drum samples, electro phunk loops, lo-fi Bronx breaks, klang grooves, tribesmen beats, warped pads, synthetic synth sweeps, old modular arps, deep fx kicks & synthetic snares, electronic drum FX samples clix & rims, twangy bleeps, zapping blips, sy-fi birds and tons more! There's nothing real in here!

Format: Acid, AKAI MPC2000, AKAI S1000, AKAI S2000/3000, AKAI S5000, AKAI Z4/Z8, Audio CD Player, Ensoniq, EXS24, GigaSampler, HALion, Kontakt, Kurzweil, Live, Logic Audio, Mac, PC, ProTools, Reason, Roland 700 Series, Sonar, Triton, Unity, WAV, Yamaha

Audio/WAV **89.95**

Xperts of Techno



Just when you have it in your gunsights, Techno shifts, changes and wriggles out of your grasp. If you make Techno, you know how hard it is to stay current. The answer to your problems: Xperts of Techno. Designed by some of the top Techno producers in Europe and so fresh and current that your tracks will leap out of the DJs mix set. Construction kits, drumloops, music loops, pads, chords, hits and more.

Format: Acid, AKAI MPC2000, AKAI S1000, AKAI S2000/3000, AKAI S5000, AKAI Z4/Z8, Audio CD Player, GigaSampler, HALion, Kontakt, Kurzweil, Live, Logic Audio, Mac, PC, ProTools, Reason, Roland 700 Series, Sonar, Triton, Unity, WAV, Yamaha

Audio or WAV/Acid.....**89.95**

Phat Beats from the Box



That's human beat box sucka! Features world-renown Kenny Muhammad, "The Human Orchestra". There's nothing out there like this. Will add an organic vibe to your tracks that you can't get from nowhere else. All the loops are full mixes then laid out with all the elements and breakdowns. 75-147 BPM plus all the bass drums, snares, cymbals, effects and everything that made the loops.

Format: Acid, Audio CD Player, Cakewalk, Cubase, EXS24, GigaSampler, HALion, Kontakt, Live, Logic Audio, ProTools, Reason, Software that can extract files from an Audio CD, Sonar, Unity

Multi-Format: Audio/WAV/Acid **89.95**

Uncivilised G'rooves



Get into these brand new programmed beats with plenty of variation to give your music a tonal edge like never before!

"... the vibe here is futuristic with very tonal type beats....sound quality and production is excellent..... for all types of futuristic dance music ... special effects CD, as well as beats ... well worth checking." - Knowledge

Format: Acid, AKAI MPC2000, AKAI S1000, AKAI S2000/3000, AKAI S5000, AKAI Z4/Z8, Audio CD Player, GigaSampler, HALion, Kontakt, Kurzweil, Live, Logic Audio, Mac, PC, ProTools, Reason, Roland 700 Series, Sonar, triton, Unity, WAV, Yamaha

Audio/WAV **89.95**

Nu Directions



e-LAB sets a NU DIRECTION for house, so anyone looking for pure soul, heart and "true to the cause" house will not be disappointed. We are sure that this disc will seriously challenge your vision of house music's past, present and future with screaming construction kits, loops, beats, samples, percussion and more! Compatible with all PC, MAC and WAV reading samplers and software.

Format: Acid, AKAI MPC2000, AKAI S1000, AKAI S2000/3000, AKAI S5000, AKAI Z4/Z8, Audio CD Player, Ensoniq, EXS24, GigaSampler, HALion, Kontakt, Kurzweil, Live, Logic Audio, Mac, PC, ProTools, Reason, Roland 700 Series, Sonar, Triton, Unity, WAV, Yamaha

Audio/WAV **89.95**

Nu Groove RnB



Here's the follow-up to the highly acclaimed Strictly RnB. Nu Groove RnB finds its inspiration from Timbaland- the genius future funk guru. Check the awesome, phat compressed loops, & if you're into the sound of the current R&B charts, you can NOT afford to miss this groove euphoria. With an audio CD and Tc WAV CD-ROM, there's no stoppin'!

Format: Acid, AKAI MPC2000, AKAI S1000, AKAI S2000/3000, AKAI S5000, AKAI Z4/Z8, Audio CD Player, Cakewalk, Cubase, E-mu E4, Ensoniq, EXS24, GigaSampler, HALion, Kurzweil, Live, Logic Audio, Mac, PC, ProTools, Reason, Roland 700 Series, Sonar, Triton, Unity, WAV, Yamaha

Audio/WAV.....**89.95**

BIG FISH AUDIO

FUNK/FUNKY/FUNKIER

Freaky Jazzy Funky



Get your freak on with Rich Mendelson, the critically acclaimed producer of Phatter Phunkier! Rich assembled the East Coast's premier session talent, and contributed his own amazing production & programming skills to make this set pop like no other! Construction kits, drumloops, fills & sounds, bass loops, riffs & sounds, percussion loops & sounds, guitar, trumpet, trombone, flute, synths & more- plus midi files! With all this, it's a breeze to customize the loops and swap sample elements.

Format: Analog sampling devices., MIDI Files, Software that can extract files from an Audio CD

Format: Audio/MIDI Files.....**89.95**

Phatter Phunkier



The long-awaited follow-up to Rich Mendelson's Phat & Phunky is finally here! This critically acclaimed producer of Freaky Jazzy Funky has done it again, and done it right! This set breaks out with 3 discs, including midi files- PACKED with construction kits, synth bass, drumloops, guitar, bass loops, drums, horns, sax loops, scratches, some funky strings, FX & way more! Whatever your production calls for, it's in here. There's 215 minutes of Hip Hop, Trip Hop, Phunk & Live production to add to your music.

Format: AKAI MPC2000, AKAI S1000, AKAI S2000/3000, AKAI S5000, AKAI Z4/Z8, E-mu E4, Ensoniq, EXS24, GigaSampler, HALion, Kontakt, Kurzweil, MIDI Files, Roland 700 Series, Sample Tank, Triton, Unity, Yamaha

Audio/MIDI Files.....**89.95**

AKAI S1000/Audio/MIDI Files.....**169.95**

Old School Funk Bass



The funk stops here! World renown session bass player and producer Josquin des Pres captures the essence of Old School Funk Bass with over 500 of the funkiest bass loops, each performed along with other live musicians, then isolated, ensuring to capture the true feel of these nasty, stylistic grooves. Look no further!

Format: Acid, AKAI MPC2000, AKAI S1000, AKAI S2000/3000, AKAI S5000, AKAI Z4/Z8, Audio CD Player, GigaSampler, HALion, Kontakt, Kurzweil, Live, Logic Audio, Mac, PC, ProTools, Reason, Roland 700 Series, Sonar, Triton, Unity, WAV, Yamaha

Multi-Format: Audio/WAV/Acid.....**89.95**

SloFunk



Having trouble adding the funk to your R&B creation? Need some funk with a little R&B flavor? Now you can funk up your mix with a little nastiness from the veterans of live funk: the funkmasterz of Freaky Jazzy Funky, Phatter Phunkier and Big Fish Audio. SloFunk takes you on a trip to the slower side of funk with construction kits, loops, sounds, guitars, bass, keys and piles more - all from 84 to 120 bpm.

Format: Acid, AKAI S5000, AKAI Z4/Z8, Audio CD Player, Cakewalk, Cubase, GigaSampler, HALion, Kontakt, Kurzweil, Live, Logic Audio, Mac, PC, Reason, Sonar, Triton, Unity, WAV, Yamaha

Multi-Format: Audio/WAV/Acid.....**89.95**

Tower of Funk



David Garibaldi is a true drum legend. His unmistakable style is the foundation of the 1970's icon funk/R&B band Tower of Power. Without David Garibaldi, Funk, R&B and Hip Hop would just not be the same.

The most comprehensive loop and sample library by one of the greatest innovators, most imitated funk drummer in history.

Format: Acid, AKAI MPC2000, AKAI S1000, AKAI S2000/3000, AKAI S5000, AKAI Z4/Z8, Audio CD Player, GigaSampler, HALion, Kontakt, Kurzweil, Live, Logic Audio, Mac, PC, ProTools, Reason, Roland 700 Series, Sonar, Triton, Unity, WAV, Yamaha

Multi-Format: Audio/WAV/Acid.....**89.95**

Funk Stew 2



There's no way to fake the funk, and every jam comes out funkier with a little flavoring courtesy of Funk Stew 2. It serves up the funky stuff in a two-disc portion. It's all here- three way harmony wah wah guitars (performed by Dre's and Time's studio talent), down & dirty live bass, hundreds of phat, live East coast and smooth West coast beats- plus Hammond, Rhodes, Moog, amazing scratches, sax squeals and too much more to list.

"Funk Stew 2.....a real winner!" 5 out of 5 - Sound on Sound

Format: Analog sampling devices; Software that can extract files from an Audio CD

Audio.....**89.95**

DRUMS

Mike Cartellone - Power House Drums



Michael Cartellone's work as a band member and touring/studio drummer with Lynyrd Skynyrd, Damn Yankees, John Fogarty, Freddie Mercury, Adrian Belew, Peter Frampton and others makes him one of the most sought-after musicians on the scene. Michael's ability to capture the essence, feel and groove of many genres, coupled with his technical precision, is the guarantee for success to the modern composer/producer.

"Michael Cartellone is my favorite Damn Yankee ...one of the all-time greatest drummers in the world today....rocks with spirit and attitude." - Ted Nugent

Audio/WAV.....**89.95**

G-Force Drums



Rock-solid session drummer, Andrea Ge, lays down some serious rock drumloops. All the drum grooves were played by this highly sought-after Italian rock drumming legend. He used three different drum kits: an all maple drum kit, a "Roswell" aluminum set and "Bonsai Kit" made from birch. Six different snare drums: brass, maple, steel, aluminum and bronze plus tons of different cymbals. Recorded at Pickaboo Studio and Umbi Studio in Milan, Italy, these drum grooves are a serious necessity for the rock producer.

Format: Acid, Cakewalk, Cubase, EXS24, GigaSampler, HALion, Live, Logic Audio, ProTools, Reason, Sonar, Unity, WAV

Multi-Format: Audio/WAV/Acid.....**89.95**

Ross Garfield Drums 1



The industry standard for quality, quantity and bang for the buck. Over 1,200 samples, multiple ambience and up to 6 velocities per instrument. From the first kick drum to the last conga, these are the best drums on earth.

"Top-notch ... great-sounding instruments recorded in great-sounding rooms with excellent mics. Kicks are excellent... snares are beautiful... hi-hats are exquisitely EQed... .." 5 out of 5 - Keyboard

Format: Aiff/SampleCell, AKAI S1000, Audio, E-mu E4, Kurzweil, Roland 700 Series, SampleCell/WAV, Synclavier

AIFF/SampleCell or AKAI S1000 or E-mu E4 or Kurzweil or Roland 700 Series or SampleCell/WAVT.....**169.95**

Audio.....**71.95**

MIXtended Drums



A new kind of drum library offering enhanced mixing options.Features five huge-sounding, ultra-professional drum kits offering unbeatable possibilities. Per kit you get close-miked dry kit with signals routed to individual outs, overhead kit and room kit all featuring up to 12 velocity levels per kit. Mix and process these three components to create the perfect drum sounds for any application, because combining drum sounds with their natural room while placing and mixing instruments individually is easy thanks to HALion's multitimbral functionality.

Format: EXS24, Kontakt, LM4

EXS24/Battery or HALion/LM4.....**89.95**

Ross Garfield Drums 2



Over 1500 samples from the Doctor's library. Ross Garfield has dramatically raised the standard for the best drums on Earth. More than 350 kick samples & 800 snares, hundreds of toms & cymbals plus an extensive percussion selection. With most having five velocity levels per instrument, you just can't get better sounding drums!

Format: Aiff, AKAI S1000, AKAI S2000/3000 & S5000, AKAI Z4/Z8, Audio CD Player, GigaSampler, HALion, Kontakt, Kurzweil, Roland 700 Series, SampleCell, SampleTank, Sonar, Triton, Unity, Yamaha

GigaSampler or AIFF/SampleCell or AKAI S1000 or Kurzweil.....**169.95**

Audio.....**71.95**

Scott Rockenfield–Queensryche Drums



From driving force behind the multi-platinum and grammy nominated band, Queensryche, comes the ultimate rock drum loops collection. Scott Rockenfield recreates the drum tracks for many of his favorite Queensryche songs. The drum tracks for each performance have been split into many different loops, variations, fills, and individual hits that you can string together to create a complete drum track that fits your song!

Format: Acid, AKAI MPC2000, AKAI S1000, AKAI S2000/3000, AKAI S5000, AKAI Z4/Z8, Audio CD Player, GigaSampler, HALion, Kontakt, Kurzweil, Live, Logic Audio, Mac, PC, ProTools, Reason, Roland 700 Series, Sonar, Triton, Unity, WAV, Yamaha

WAV/Acid.....**89.95**

INTERNATIONAL

Hadeeth — Arabic Rhythms



There's nothing like old world rhythms to add flavor and essence to tracks that would seem otherwise a bit ordinary. These Arabic rhythms are just that- traditional percussion rhythms that will add the authenticity that makes any traditional track or film score come alive, but also modern rhythms that add an edge off to the traditional, and add a new flair to the ancient rhythms that originally inspired dance. Hadeeth was recorded by seasoned Arabic professionals and contains loops and hits from tabla, taar, duff, tamborine with skin, congas, sakat (hand cymbals) . . .

Format: Analog sampling devices; Software that can extract files from an Audio CD

Audio.....**89.95**

Roots of the Middle East & North Africa



Take a journey to countries like Saudi Arabia, Algeria, Egypt, Syria, Iraq, Libya, Tunisia, Morocco and more. The most authentic percussion hits and rhythms, straight from these regions, are sure to add the truest sound to any traditional or classic production, as well as give dance tracks a flare that is otherwise unattainable. Add authenticity to a movie score or create genuine music – you go back to your Roots.

Format: AKAI MPC2000, AKAI S1000, AKAI S2000/3000, AKAI S5000, AKAI Z4/Z8, GigaSampler, HALion, Kontakt, Kurzweil, Roland 700 Series, SampleTank, Triton, Unity, Yamaha

AKAI S1000/Audio **169.95**

Audio.....**89.95**

Roots of South America



Enjoy the rich sounds and fascinating grooves in styles such as candombe, murga, milonga, chacarera, huayno, festejo, joropo, cumbia, samba, afoxe, cueca, vals peruano, carnavalito and more. Over 70 distinct traditional percussion instruments create more than 300 grooves. More than 250 single hits from the Caribbean coast of Venezuela to the south of Argentina. Multiple dynamic levels and tempos.

Format: AKAI MPC2000, AKAI S1000, AKAI S2000/3000, AKAI S5000, AKAI Z4/Z8, GigaSampler, HALion, Kontakt, Kurzweil, Roland 700 Series, SampleTank, triton, Unity, Yamaha

Audio **89.95**

Audio/Yamaha or AKAI S1000/Audio.....**169.95**

Roots of the Pacific



Ensemble, solo drum performances; hits from the Hawaiian Islands, Fiji, Tahiti. . Pahu (bass drum), to'ere (slit drum or "log" drum), fa'atete (tom tom). . .a large tin can played (snare drum style). Pahu heiau (temple drum), pahu hula (dance drum), ka'eke (bamboo pipe drum), ipu (bottle gourd), Ili'ili (small stone castanets), pu'pu (shells), pu'ili (split bamboo sticks), uli'uli (feather covered gourd rattles), kala'au (wooden sticks), chants and samples of 'olelo (various Hawaiian words), location recordings of gentle Maui waves, a bamboo forest running stream on a rainy night, the sound of paradise . .

AKAI S1000/Audio **169.95**

Audio.....**89.95**

Roots India



Delve deep into the roots of India and explore the vast complexity of the rhythms from its many regions. With loops & hits from instruments such as Pakhwaj, Taal, Tabla, Ghungroo, Dhol, Chanda, Chimta, Mridangam, Ghattam, Tavul, Khanjari, Madol, Khol, Tuntuna and many more; creation of this emotionally driven fusion of earth and music is only limited to your inspiration. These traditional rhythm patterns of devotional, folk and classical styles are perfomed by seasoned Indian musicians.

Format: AKAI MPC2000, AKAI S1000, AKAI S2000/3000, AKAI S5000, AKAI Z4/Z8, GigaSampler, HALion, Kontakt, Kurzweil, Roland 700 Series, Sample Tank, Triton, Unity, Yamaha

AKAI S1000/Audio **169.95**

Audio.....**89.95**

Megatron Trance Bomb
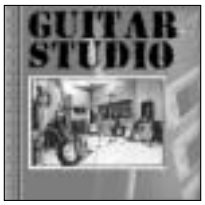


From the soaring melodies of Goa, to the pounding machine rhythms of Berlin, Megaton Trance Bomb delivers everything you need to create trance. You make trance, so you know how hard it is to keep current. This ever-changing music requires up-to-the-minute sounds, & we've got them right here! Megaton Trance Bomb was created by two top producers in the world of trance, and the quality is evident in these massive beats. This 2 disc set gives you mostly construction kits with extra drumloops and synth arpeggios for that custom sound. In short, everything you need to bliss out right now!

Format: Analog sampling devices; Software that can extract files from an Audio CD

Audio.....**89.95**

INSTRUMENTAL

Guitar Studio



The "swiss army knife" of guitar loops and performances, Guitar Studio gives you everything you need to make authentic guitar-based tracks. Whether you're looking for cutting-edge alternative rock grooves, or retro jazz riffs and comps, you'll find them here. Blues, metal, funk, acoustic- it's all here, and it all grooves! And not just static loops; you'll get intros and endings too, so you can make the most realistic guitar performance possible.

Format: Acid, Audio CD Player, Cakewalk, Cubase, EXS24, HALion, Live, Logic Audio, ProTools, Reason, Sonar, Unity, WAV

Multi-Format: Audio/WAV/Acid.....**89.95**

Technoid Guitars



Brand new from Wizoo and producer Craig Anderton. Pushes the guitar's sonic envelope, creating a hybrid organic/electronic sound that's perfect for the 21st century world of dance music and Hip Hop. Electronic music guru Craig Anderton combines guitar, synths, processing and cutting-edge music into a double-CD set that is truly groundbreaking. Over 1.1 GB of guitar-like sounds and synthetic, highly processed material. Retains guitar's organic 'vibe.' Power chords and expressive leads.. Effects, such as slides, squeaks, reverb oceans or vocoder chords. Rhythmic qualities perfectly suited to modern dance and hip-hop styles.

Multi-Format: HALion/REX/WAV.....**89.95**

Celtic Instruments



Add the mesmerizing essence of the Celtic tradition to your music or score. Visit the misty hills of Scotland and Ireland with Celtic Instruments. High-quality, large and detailed multisampled instruments, such as great highland pipes, uillian pipes, Scottish small pipes and other bagpipe-styles. Pennywhistle, bodhran (Irish drum), fiddle and more. Astounding quality, many instruments sampled in stereo. Many cases feature multiple samples for every pitch.

Format: Acid Cakewalk, Cubase, EXS24, GigaSampler, HALion, Live Logic Audio, ProTools, Reason, Sonar, Unity, WAV

GigaSampler or AKAI S1000.....**254.95**

Chronic Horns Vol. 2



Horn riffs and patches for phunkin' and "live"ning up your tracks. Huge 5-piece section of the dopest players in L.A. Hundreds of solo / mute trumpet, trombone, baritone sax, tenor sax riffs. 5-piece sections in chromatic unison, swells, stabs, sustains & falls. Triads and seventh chords in diatonic progressions of swells, stabs, sustains and falls.Reggae, 70's TV-Funk , Ska, Swing, Blues, Acid Jazz and Groovy Phunk.

Format: AKAI MPC2000, AKAI S1000, AKAI S2000/3000, AKAI S5000, AKAI Z4/Z8, GigaSampler, HALion, Kurzweil, Roland 700 Series, SampleTank, Triton, Unity, Yamaha

AKAI S1000.....**169.95**
Audio.....**89.95**

Brush Artistry



A collection of classic and contemporary drum loops and samples – the wonderful sound of brushes! Beautifully recorded. Full variety of organic brush patterns. Traditional swing, be-bop swing, ballad swing, jazz waltz, bossa nova, samba and country. Modern six-eight, folk, funk and latin styles. Each brush style recorded at multiple tempos. Includes full kit loops, intros, fills, ending and "snare only" tracks. Over 71 minutes of loops and samples... nearly one thousand audio files!

Format: Acid, AKAI MPC2000, AKAI S1000, AKAI S2000/3000, AKAI S5000, AKAI Z4/Z8, Audio CD Player, GigaSampler, HALion, Kontakt, Kurzweil, Live, Logic Audio, Mac, PC, ProTools,Reason, Roland 700 Series, Sonar, Triton, Unity, WAV, Yamaha

Multi-Format: Audio/WAV/Acid.....**89.95**

Dread: The Reggae Collection



Reggae is a hugely popular art-form, and yet rarely has it been represented properly in a sample CD, so Big Fish is extremely pleased to present Dread. Quit planting that stuff in your back yard, get back inside and make some sweet rhythms. These construction kits are gorgeous...the drums are authentic, the bass is low and throbbing, and in the 'pocket' like it should be. Also included: extra drum hits, loops, keyboard bits, guitar and bass loops, and lots more. This fat one will take you from dirty ol' raggamuffin dub to joyous island sounds, you can't find a better reggae disc.

"If you want to 'Marleyfy' your music-making then this'll be just the ticket." - Computer Music

Format: Acid, Cakewalk, Cubase, EXS24, GigaSampler, HALion, Live, Logic Audio, ProTools, Reason, Sonar, Unity, WAV

Multi-Format: Audio/WAV/Acid.....**89.95**

PIANO/STRINGS

First Call Vintage Keyboards



A complete library of vintage synthesizers, electric pianos and organs for your sampler. All the classic sounds, including: Arp, Farfisa, Hammond B3, Clav, Mellotron, Moog, Oberheim, Wurlitzer, Korg, Crumar, Emu, Sequential, Yamaha CS-80, Fender Rhodes, Vox Continental & way more.

"First I loaded the Hammond B3...shocked at the authenticity. I quickly moved to Fender Rhodes and Mellotron ...the original intent of each machine is captured by the sample or brilliantly simulated by the patch." - Keyboards, France

Format: AIFF, AKAI MPC2000, AKAI S2000/3000, AKAI S5000, AKAI Z4/Z8, GigaSampler, HALion, Kurzweil, Roland 700 Series, SampleCell, SampleTank, Triton, Unity, Yamaha

AIFF/SampleCell or AKAI S1000 or Yamaha **254.95**

John Cage Prepared Piano



Created for use in his magnum opus composition, Sonatas & Interludes (Feb. 1946-March 1948). Forty-five of the piano's 88 keys were prepared, eleven with double preparations; 3 dynamic levels in 5 articulations comprise a total of 1,320 samples! Whether you're a classical pianist looking for a portable prepared piano, an electronic musician looking to experiment with unusual organic sounds, or an admirer of Cage's unprecedented genius, you will be left in reverent awe.

Format: AKAI, MPC 2000, AKAI S2000/3000/S5000/Z4/Z8, E-mu E4, Ensoniq, EXS24, GigaSampler, HALion, Kurzweil, Roland 700 Series, SampleTank, Triton, Unity, Yamaha

GigaSampler or AKAI S1000.....**169.95**

Prosonus Grand Piano



From the acclaimed Prosonus library, a sampled Steinway grand piano without compromise. For the first time, the sampler equipped studio can reproduce every nuance of this spectacular instrument in the digital realm, allowing a depth of expression and level of authenticity previously available only to Steinway owners. With beautiful multi-samples at 16, 32, 64 and 128 meg patches and samples of all 88 keys, this is the most "playable" piano CD-ROM available.

Format: AKAI S1000, AKAI S2000/3000, AKAI S5000, AKAI Z4/Z8, E-mu E4, Roland 700 Series

Roland 700 Series or AKAI S1000, S5000 .....**169.95**

Kurzweil or E-mu E4 .....**169.95**

The Orchestral Collection



From percussion to strings, brass to woodwinds. Marcato, pizz and FX; section and solo violin, viola, cello and bass; trumpet, piccolo trumpet, flugelhorn, french horn, trombone, bass trombone, tuba/tuba FX; brass ensemble and brass orchestra; flute, alto flute, piccolo, clarinet, oboe, oboe d'amore, English horn, bassoon, and contrabassoon; harp single notes and glissandi; timpani, tubular bells, triangle, cymbals, cymbal FX, gong, snare, snare rolls...

Format: AKAI MPC2000, AKAI S1000, AKAI S2000/3000, AKAI S5000, AKAI Z4/Z8, Audio CD Player, Cubase, E-mu E4, Ensoniq, EXS24, GigaSampler, HALion, Kontakt, Kurzweil, Logic audio, MAS, ProTools, reason, Roland 700 Series, Triton, VST, Yamaha

Kurzweil, AKAI S1000, Yamaha or UVI Plug-in.....**254.95**

London Orchestral Percussion



Four discs jam-packed with over 2.5 GB of the finest orchestral percussion ever produced. A huge selection of instruments: 8 crash cymbals, 7 snare drums, etc. all at up to 16 stereo velocities. Multiple articulations per instrument, such as rolls, glissandos, flams, scrapes and more for the most realistic simulations. Recorded in the classical tradition - no close mic'ing - to allow the sound to develop properly. Melodic instruments, such as glockenspiel, feature multiple dynamics and samples for every available pitch.

Format: AKAI S1000, AKAI S2000/3000, AKAI S5000, AKAI Z4/Z8, E-mu 4, Ensoniq, EXS24, GigaSampler, HALion, Kontakt, Kurzweil, Roland 700 Series, Triton, Unity, Yamaha

AKAI S1000 or GigaSampler.....**399.95**

London Solo Strings



Picks up where every other string library leaves off The most playable string library out of the box – load it up and start playing. Violin, Viola, Cello and the oft-omitted Double-Bass like you've never heard them before. Top of the line instruments, like the beautiful violin from 1740. Articulations: arco, marcato, martele, spiccato, sul tasto, ponticelli, ponticelli tremolo, half step / whole step trills, tremolo, pizzicato, ricochet, short, medium and long crescendos, harmonics, bartok (snap) pizzicato. A huge assortment of FX per instrument.

Format: AKAI S5000, AKAI Z4/Z8, EXS24, GigaSampler, HALion, Kontakt, Kurzweil, Triton, Unity, Yamaha

Multi-Format: Audio/WAV/Acid .....**Call**

PLATINUM 24

Platinum 24 Grand Piano



Adds a top-flight concert grand to your sampler's arsenal. This is all the concert grand you will ever need. Enabling you to adapt this versatile instrument to every situation and song, you are free to choose from a rich selection of options at any time: One to 10 velocity layers, with or without pedal string resonance, 22 or 42 samples per layer, from 56 MB to 1.1 GB in size or four timbres (Natural, Soft, Bright, Hard). Excellent sound quality. The original 8-track, 24-bit recordings were mastered in 16-bit stereo format. As a result, you enjoy the fidelity of 24-bit samples at significantly reduced computer and hard disk loads.

Format: Kontakt

EXS24.....**124.95**

Platinum 24 Electronic Drums



Takes the legendary sounds of the Roland TR-808 and TR909 to your sampler. Enriched with contemporary analog and digital options, you get the highly diverse variations of every individual instrument and up to ten dynamic levels, both clean and overdriven. Sampled from the originals at 24-bit, mind blowing dynamics through analog compression and, in part, true tube distortion. Digital 32-bit high-end processing and maximized 15-bit mastering. More than 40 kits and menus. The most diverse musical styles, plus reverb, reverse and synthesizer FX. All instruments may be swapped among kits. Up to ten dynamics levels per instrument, variable via velocity or pitch.

AKAI S1000 .....**Call**

Platinum 24 Acoustic Drums



The perfect drum library for professional demands in the studio, at home or on stage. Mix 'n match instruments at will. Features complete drum sets and loads of variations (e.g., rolls, flams, brushes). 24-bit recording (multi-track micing), 32-bit high-end processing, 16-bit stereo mastering, up to 32 MB per kit. Perfectly tuned drum sets; instruments can be swapped individually. Chromatic kits with up to 20 dynamic levels per instrument. Exclusively the best that top-flight professional drummers, engineers and sound engineers have to offer.

Format: AKAI S1000, AKAI S2000/3000, AKAI S5000, AKAI Z4/Z8, E-mu E4, Ensoniq, EXS24, GigaSampler, HALion, Kontakt, Kurzweil, Roland 700 Series, Triton, Unity, Yamaha

AKAI S1000..... **Call**

Platinum 24 Latin Percussion



All the percussion you could ever wish for, and more. Featuring everything from congas, bongos and timbales, to shakers, maracas, cabasa, tambourine, triangle, cymbals and exotics such as caxixi, pandeiro or surdo. Played by pro percussionists and recorded in premium Platinum quality. 24-bit recording (multi-track micing), 32-bit hi-end processing up to 100 MB per kit, perfectly tuned sets, chromatic kits with up to 20 dynamic levels per instrument.

Format: AKAI S1000, AKAI S2000/3000, AKAI S5000, AKAI Z4/Z8, Analog Sampling Devices, E-mu E4, Ensoniq, EXS24, GigaSampler, HALion, Kurzweil, Kurzweil, Roland 700 Series, Software that can extract files from an Audio CD, Triton, Unity, Yamaha

Multi-Format **89.95**

Performance Loops – Drums Vol. 2



The Performance Loops series represents a unique approach to sample libraries, taking several loops, variations, fills and hits from the tracks of real song performances. By using many different loops, variations, and fills from a performance, you can construct a complete track that sounds like it was played by a real drummer instead of sounding like the same loop over and over (unless that's the sound you're going for). Or, if you are searching for that perfect loop, you have a much wider range of variations to choose from in each style with the Performance Loops series. Contains over 1000 of the highest quality stereo drum loops, variations, fills and hits from top session drummers recorded at world class recording studios in Seattle and LA.

Multi-Format: Audio/WAV/Acid **89.95**

Performance Loops – Acoustic Guitars



Finally, a usable acoustic guitar library! What better way to have it than in the Performance Loops series? Similar to the Performance Loops Drums series, you're able to use several guitar loops and variations from a single performance (or session) and string them together to create a complete track. Each guitar performance is organized by chords or root key and laid out for you to construct a whole song instead of just relying on one ambiguous loop. 6-string steel, 6-string nylon and 12-string guitars from Taylor, Guild, Gibson, Ovation and others. Over 1000 loops in folk/rock, rock/pop, acoustic singer/songwriter, with some jazz, funk and a touch of country thrown in. The Performance Loops series has revolutionized production and this is one not to miss!

Multi-Format: Audio/WAV/Acid.....**89.95**

BIG FISH AUDIO

LOOPZILLA

Loopzilla Underground



The monkey is back! He's been lurking in the underground, gathering the newest and freshest sounds, waiting for the perfect moment to emerge. That time is now. Loopzilla busts up through the streets of America's urban landscape to *yet again* dictate the course of Hip Hop! From the producers of the acclaimed Loopzilla and Wall of Vinyl, comes the deepest construction kits, hits, scratches, guitar stabs, breaks, Rhodes, bass, horns and more!

Format: Acid, AKAI MPC2000, AKAI S1000, AKAI S2000/3000, AKAI S5000, AKAI Z4/Z8, Cakewalk, Cubase, E-mu E4, Ensoniq, EXS24, GigaSampler, HALion, Kurzweil, Live, Logic Audio, Mac, PC, ProTools, Reason, Roland 700 Series, Software that can extract files from an Audio CD, Sonar, Triton, Unity, WAV, Yamaha

WAV/Acid or Audio **89.95**

Loopzilla Funk, Guitar & Bass



Can you smell the funk?! All you producers of Hip Hop, R&B, Rap and Trip Hop know that nothing funks up programmed rhythm tracks like live guitar and bass. Loopzilla Funk Guitar & Bass is the answer, and this set is packed with top quality funk! Featuring players from the Gap Band, Boo-Ya Tribe and Dr. Dre's posse, this set covers everything from clean & jazzy to low-res greasy- played by pros who understand what Hip Hop feels like. Featuring wah-wah, multi-guitar harmonies, mutron, clean and distorted funk, plus a ton of variety guaranteed to make a straight track swing and a good track great!

Audio.....**89.95**

Loopzilla 1



This Hip Hop monster is still tearing the roof off from coast to coast! Still the leader in authenticity and usability, and the standard for Rap and Hip Hop construction kits! Loopzilla is two discs jam-packed with some of the phattest construction kits ever made! Way over 1,000 Breakbeats, bass loops (greasy old electric, stand-up and synth), funky wah wah guitar loops, scratches and FX. There's West Coast Gangsta, Acid Jazz, Old School and more top quality Hip Hop Funk for your trunk than anybody else is selling anywhere.

Audio.....**89.95**

Loopzilla 2



The monkey is back with two discs jammed full of the phattest grooves and construction kits in funky, soulful music for R&B and Hip Hop! Loopzilla 2 combines the best elements of programmed grooves and live playing with new melodic & hook elements destined to be the hallmark of the new sound. Breakbeats, guitar, bass, analog synth loops, horn hits, organ, silky vocal harmonies & pop hooks, combined in slow jams, R&B & Hip Hop from Atlanta, Miami, New York, Oakland and So Cal schools of Hip Hop, Funk and R&B.

Audio.....**89.95**

Loopzilla 3



Loopzilla 3 busts out the cage with 145 minutes of brand new drums, guitar, bass, rhodes, vocals, horns, turntable wizardry and much, much more! The monkey will take you from mad, hard Hip Hop Funk to silky Hip Hop Soul* Quality makes the difference! This 2-disc set features the veterans of Loopzilla 1 & 2, plus new talent from Boo Ya Tribe and The Dazz Band. There's nothing about this set that ain't great - so c'mon and get your groove on with Loopzilla 3!

Audio.....**89.95**

Rizing Drum and Bass



TJ Rizing finally lets you in on a few of his drum & bass secrets. Rizing Drum & Bass is a collection of some of his best- construction kits from 165-180 bpm, bass sounds, keys and FX, sub bass, 808 bass, Rhodes, kicks, snares, hatz, percussion and more! Produced by legend TJ Rizing and mastered by the same team who brought Underground Garage, this audio CD is brand new and ready to be used. The loops are tight, the sounds are superb and the production is right in the pocket!

Format: Analog sampling devices; Software that can extract files from an Audio CD

Audio **99.95**

Drum 'n Bass: Journey to the Light



When it comes to Drum 'n Bass, Jamie Myerson needs no introduction. America's premiere Drum 'n Bass producer has finally made a sample CD, and it slams! This set features a variety of construction kits and loops, ranging from hard 2-step to dreamy Lookin' Good-style beats. This is a must-have for all you Jungle/Drum 'n Bass fans.

"A drum'n'bass offering...comes packed to the hilt with 764 samples ... on two CDs in both audio and wav format." - Computer Music.

Format: Acid, Cakewalk, Cubase, EXS24, GigaSampler, HALion, Live, Logic Audio, ProTools, Reason, Sonar, Unity, WAV

Multi-Format: Audio/WAV/Acid.....**89.95**

HIP HOP/DRUMS

Platinum Essentials



Keith Clark, a.k.a Clizark, is the best kept secret on the West Coast! Shhhh, don't tell no one, but he's lettin' you in on his multi-platinum bag of essentials! His wicked versatility as a producer can be heard on albums like the debut of Def Jam Recording Artists "The Dove Shack", Snoop's first album with No Limit Records "Da Game Is To Be Sold Not To Be Told", along with his follow-ups – including Suited N Booted on Snoop's brand new "Paid Tha Cost to be Da Bo$$" and multi-platinum hits from B-Legit, Mack 10, Kurupt, Eastsidaz and TRU. These hits propelled Keith to multi-platinum status in a matter of minutes! Now he is among an elite group of producers putting it down on the West Coast and lapping up the worldwide recognition. "Platinum Essentials" is JAMMED with construction kits, beats, sounds and more from his bag of platinum sounds! This joint is so hot, a few of the tracks even got sold before it was done. Now you're hearing them on the radio. You want the platinum? Pick up the essentials.

Format: Acid, AKAI MPC2000, AKAI S5000, AKAI Z4/Z8, Audio CD Player, Cakewalk, Cubase, EXS24, GigaSampler, HALion, Kontakt, Kurzweil, Live, Logic Audio, Mac, PC, ProTools, Reason, Sonar, Triton, Unity, WAV, Yamaha

Multi-Format: Audio/WAV/Acid.....**44.95**

Metriam



Metriam is your gateway to parallel dimensions; the ancient ambiences, ritual rhythms and shaman sound effects on this CD will elevate your music to another level... or just make it a little more interesting.

Metriam is stacked with tribal drum beats, rhythmical mood loops, shimmering backgrounds, walls of noise, ethnic-electric percussion sounds and more. If you ever find a sample collection as unique as this one, you're dreaming.

The sounds and loops on Metriam are ideal for soundtrack composers, perfect for technoheads and dance music producers, and indispensable for makers of ambient and chill-out music.

Drumwise: Single Drum Sounds

Rhythmbeds: Harmonical, rhythmical pads and beds.

Soundwise: Atmospheric add-ons.

Pads/Plains: Two-dimensional pads and backgrounds. Shallow

Pads/Pools: Three-dimensional pads and backgrounds. Deep.

Audio/REX/WAV **89.95**

LA Drum Sessions



Prepare yourself for the ultimate collection of drum performances to date. Phat, thick, luscious beats, in about every style you can think of, with so much variation, you won't need anything else for a long while! Three discs of acidized WAV, divided up into folders containing a number of related loops in a given style and tempo. Allows you to construct a song in your style by picking from similar loops. A variety of recording setups were used in the studio, to fit the needs of each style being recorded. There are three versions of each performance: dry recording, room-mic recording only and mixed (dry and room-mics together). The mixed versions have been mixed for each style separately to create an appropriate mix... but if you don't like it, you can mix your own version! Over 80 categories of styles; over 6000 loops; Jazz, Rock, Disco, 60's Fun, Funk, Punk, Country, Blues, Texas Shuffle and more. Call up your own session drummer today, and add the quality only real session pros can offer.

Styles include: Jazz, Big Band, Rap/Rock, Hard Rock/Metal, Slow Rock, Fast Rock, Pop-Rock, 60's Fun, Slow and Fast Surf-Rock, Disco, Slow Groovin', Funk, Gypsy/Bossa, Punk, Reggae, Blues (slow, fast, chicago, etc.), Country Waltz, Texas Shuffle...and more.

24-bit/WAV/Acid (3 CD Set) **89.95**

Dirt Keeps the Funk



The mothership has landed! Inspired by Parliament, Funkadelic and other P-funk pioneers, the live played loops and grooves on this unique CD are sure to shake any booty. Dirt Keeps The Funk is jam-packed with bubbling, Bootsyrific bass guitar lines, laid-back drum beats, wah-wah rhythm guitars, go-go style percussion and psychedelic synth riffs, all waiting to tear the roof off your sampler. P!Funk!

•Bass guitar •Rhythm guitar •Drums •Rhodes •Clavinet •Organ •Percussion •Analog synths

Audio/REX/WAV **89.95**

BIG FISH AUDIO

SAMPLEHEADS

Peter Erskine – Living Drums



One of the most amazing drummers in music today! Killer recordings of drum hits and loops played and selected by Peter from his collection of world class kits, compiled on this definitive collection, guaranteed to make your tracks come alive. Actual Swinging 4/4 time! Odd-meter loops, Brazilian and Caribbean beats, Jazz, Pop & R&B Loops, transitions & endings, fills, loops, ballad and brush performances, multi-sampled drum kits w/textural, dynamic and timbral variations, blastix/rute, mallets, head/rim hits, multiple dynamics, right & left stickings.

AIFF/SampleCell, AKAI S1000, E-mu E4, Giga or Roland 700...**129.95**
WAV **114.95**
Audio **89.95**

Mark Walker – Latin Drums



Mark Walker lays down slamming Cuban, Brazilian, Puerto Rican and other Latin American rhythms* but this is not your typical Latin Percussion disc of traditional timbales and congas. What sets this one apart from all the others is that the loops are played on a full drum kit plus a complete assortment of hits and cymbals. The loops cover a HUGE variety of tempos and feel plus a section of superb drum hits that get up to nine levels of velocity!* some say the best ever recorded. If you liked Peter Erskine - Living Drums!, you're going to LOVE this!

AKAI S1000 , E-mu E4 or GigaSampler.....**149.95**
WAV **129.95**
Audio **99.95**

NYC Drumworks



One of the best selling sample discs ever! 146 minutes of awesome rhythms played by 6 of New York City's top session drummers on 2 audio CDs or 2 CD-ROMs. Funk, rock, hip-hop, latin, R&B, brazilian, rock-a-billy, reggae, gospel, country, thrash, funk and more! Kenwood Dennard, Shawn Pelton, Steven Wolf, Thommy Price, Dave Johnson and Frank Duncan laid it down with sticks, blastix, brushes and mallets at multiple tempos per groove and style... The definitive collection of drumloops and matching hits from 6 of New York's finest. Maximum bang for the buck!

AIFF/SampleCell, AKAI S1000, E-mu E4, Giga or Roland 700 ..**219.95**
WAV **134.95**
Audio **89.95**

NYC Percussionworks



The follow-up to the blockbuster NYC Drumworks, Percussionworks is a 2 audio or 2 CD-ROM set featuring 7 of NYC's hottest percussionists. Hundreds of percussion samples, hits and loops, all meticulously recorded. Whether you're looking for that perfect conga part or a Hip-Hop tambourine groove, it's all here. The best part is that this library is totally interactive with NYC Drumworks (in tempo and style). Sammy Figueroa, George Jinda, Steve Thornton, Emedin Rivera, Mark Johnson, Todd Isler and Mark Johnson give it to you straight!

AIFF/SampleCell, AKAI S1000, E-mu E4, Giga or Roland 700 ...**214.95**
WAV **134.95**
Audio **89.95**

Pocket Syndrome v.1 – Funk Guitar



Groove Master, Bernd Schoenhart (John Secada, George Michael, Kenny G, Slash, Marc Anthony, Dave Valentin, C+C Music Factory) brings you his unique "In The Pocket" guitar feel. More than 700 "In The Pocket" guitar loops at multiple BPMs and keys per loop. All loops are calculated to 1 or 2 bars in more than 20 unique styles, grouped by BPM, ready to give your tracks the Pocket Syndrome Edge. Your tracks will come alive with the "funky guitar feel" you've only dreamed of achieving with your sampler or computer- until NOW!

AIFF/SampleCell, AKAI S1000, E-mu E4, Giga or Roland 700...**129.95**
WAV.....**114.95**
Audio.....**89.95**

Pocket Syndrome v.2 – Rock Guitar



Groove Master, Bernd Schoenhart (John Secada, George Michael, Kenny G, Slash, Marc Anthony, Dave Valentin, C+C Music Factory) does it again* this time focusing on rock guitar. All new unique "In The Pocket" rock guitar loops and riffs at multiple BPMs and keys per event. All loops are calculated to 1 or 2 bars, ready to load into your sampler or computer. Once again, Bernd performed in more than 20 unique styles, including the slamming "Rock-It-Pocket" grouped by bpm, ready to give your tracks the "Pocket Syndrome Edge."

AIFF/SampleCell, AKAI S1000, E-mu E4, Giga or Roland 700 ..**149.95**
WAV.....**129.95**
Audio.....**99.95**

SAMPLEHEADS

Dave Samuels – Marimba & Vibes



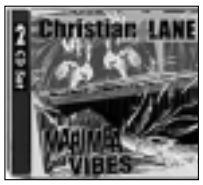
The highest quality multi-samples of marimba and vibes ever recorded from the top mallet player of his generation* 2 time Grammy award winner, Dave Samuels. Dave's credits include Gerry Mulligan, Oscar Peterson, Stan Getz, The Yellowjackets, Pat Metheny, Bruce Hornsby, Frank Zappa and Spyro Gyra. Multiple mallet variations (timbre) and dynamics. Chromatic samples of the entire range of the instruments: Yamaha 3-1/2 octave vibe and the rare Yamaha 5 octave rosewood marimba. Sustained & muted notes, rolls, riffs, and more!

AKAI S1000 , E-mu E4 or GigaSampler.....**129.95**

WAV **114.95**

Audio **89.95**

Christian Lane - Ultimate Marimba & Vibes



This gorgeous library features a Marimba One five octave rosewood marimba, and a Musser Gold Century set of vibes. 4 sets of mallets to choose from with each instrument which should give you an idea of possibilities that await your compositions. The marimba patches include 4 velocity strikes, rolls, muted strikes and rattan handle strikes. The "marimba extra's" patch includes resonator strikes, glisses, various runs & more. The Vibraphone patches include 3 velocity strikes, pedal up strikes, bowed vibe & many ethereal sounds. 3 selectable motor speeds are programmed via the mod with each mallet of the vibes. The days of dealing with one velocity and one mallet choice are finally over!

GigaSampler .....**Call**

Whole Lotta Country



Another first from Sampleheads! There is nothing like this in the industry! The definitive collection of country instruments, performed by Country Guru, Larry Campbell. Fiddle, mandolin, pedal steel, banjo, dobro, hi-string guitar, electric telecaster, 6-string bass, and gut bucket, all chromatically sampled to the highest of standards. Plus, the most outstanding collection of country/acoustic loops, phrases and riffs ever assembled, including endings and fills. This double audio CD or 2 CD-ROM set will put a smile on anyone looking for

AIFF/SampleCell, AKAI S1000, E-mu E4, Giga or Roland 700 ...214.95

WAV **134.95**

Audio **89.95**

Jazzistic



At long last, a sample CD totally devoted to Jazz! Pure Jazz drumloops, brush loops, hi-hat and cymbal loops, all laid out so you can program your own beats. Tons of tempos and coverage of even the more obscure Jazz styles such as swing, funk, Latino, African and even some Caribbean! Round it off with some of the sweetest Jazz guitar, double stand-up bass, tons of drum, guitar and bass samples, plus a pile of construction kits covering authentic Jazz! ...and as if that weren't enough, there's a bonus section of suitcase electric piano and Hammond organ on the CD-ROM! Finally, all the building blocks for the genuine Jazz touch!

AKAI S1000.....**Call**

Audio .....**Call**

The Will Lee Bass Library



The disc that started it all! Recorded by the most in-demand bass player working today, "Bass Legend" Will Lee!

The CD-ROM version contains 11 unique, world class, chromatically multi-sampled basses from Will's private collection. Includes harmonics, mutes, slides, drops, thumbs, pulls, x-notes, and effects (as well as sustained notes at multiple dynamics) all chromatically sampled!

The audio version contains 6 basses and a taste of “WillLoops.” This disc has been used on countless hit records and films by producers and programmers too numerous to mention!

AIFF/SampleCell, AKAI S1000, E-mu E4 or Roland 700.....**129.95**

Audio..... **89.95**

Will Lee: Ultra Freakin’ Bass



More than 40 minutes of the most "In The Pocket" bass grooves, riffs and loops ever assembled, as performed by "Bass Legend" Will Lee! Recorded at multiple tempos and keys in an array of styles that will keep you busy making music for a long time.

Bonus: the audio version (2-CD set) also includes 5 new basses from the Will Lee Bass Library, sampled chromatically, featuring Will's most requested bass - the Sadowsky 4-string! So we call this audio disc Will Lee Bass Library Vol. 2. The CD-ROM version is a single disc, just the loops, pre-looped and formatted, ready to eat.

AIFF/SampleCell, AKAI S1000, E-mu E4 or Roland 700.....**214.95**

WAV **114.95**

Audio **89.95**

PRO SAMPLES

Vol. 1
Hip Hop & RnB 1



Hip Hop/RnB style construction kits- the full mix plus the individual samples used in the full mix broken down. Tempo ranges from 70 bpm to 97 bpm. A great selection of material taken from the EastWest libraray Ill Jointz.

EAPS01.....**44.95**

Vol. 2
Hip Hop & RnB 2



Hip Hop/RnB style construction kits- the full mix plus the individual samples used in the full mix broken down. Tempo ranges from55 bpm to 95 bpm. A great selection of material taken from the EastWest library Black Butta 2.

EAPS02.....**44.95**

Vol. 3
Drum & Bass 1



Featuring Drum 'n' Bass construction kits - the full mix plus the individual samples used in the full mix - broken down. Tempo ranges from 160 bpm to 171 bpm. A selection of drum hits: kicks drums, snare drums & hi hats.

EAPS03.....**44.95**

Vol. 4
Percussive Loops



Rhythmic grooves whose feel is described by movie titles, such as The Matirx, Blade Runner, Mission Impossible & Patriot Games. Tempos from 60 bpm to 148 bpm. Includes cymbal hits, effects and low drum hits.

EAPS04.....**44.95**

Vol. 5
Pop & Rock Drumloops



Featuring drum loops in the style of Rock, Pop, Rock-o-billy, Swing & Shuffle. Tempo ranges from 94 bpm to 158 bpm. A great selection of material taken from the EastWest library Steve Smith Rhythmic Journey.

EAPS05.....**44.95**

Vol. 6
Experimental Grooves & FX



Loops broken into categories like Musical loops, Rhythmical Loops, Strange Loops. A good selection of single sounds. Ambient, Musical, Screams, SFX, Spoken Words, Strange single sounds. Some material from Cuckooland Asylum.

EAPS06.....**44.95**

Vol. 7
Future Beats



Featuring FX, Guitar, Radio, Techno, Mad, Tabla , Tambourine Loops. Also "chemical" Pads and Stabs. A great selection of material taken from the Zero-G library Chemical Beats.

EAPS07.....**44.95**

Vol. 8
World Vocals



Vocals in Maani, Sufani, Pardesi, Chamka, Larka, Hek, Maanlo, Kooli, Horbols,Ghata, Sauda, Karnatic, Koka, Beparwa, Daasi, Ishaq, Joom, Heer, Baabla, Rabba, Daya, Vadia and Raanjan. Some from Zero-G's Deepest India.

EAPS08.....**44.95**

Vol. 9
Future Beats 2



Futuristic style loops in construction kit format – range from 100 bpm to 208 bpm, created using raw and organic sources for rhythms and soundscapes. Some material taken from the Zero-G's Fields of Motion.

EAPS09.....**44.95**

PRO SAMPLES

Vol. 10
House



Featuring House influence Drum Loops, Synth Loops, Arp Loops, Bass Loops, Pads, Stabs & Splats. Also includes Guitar,Bass, Weirdo, Buzz Syner, Inta Stab, Elixia Organ, M1 Bass Organ & SY99 Piano Multisamples.

EAPS10.....**44.95**

Vol. 11
Classic Orchestra



String Ensembles, Solo Violins, Violin runs, sordino, long, pizzicato, glissandi, trills, Cello arppegio, Woodwinds, Harp, Gran Casa, Timpani, Tam Tam, Snares, Cymbals. Some from the Peter Siedlaczek Advanced Orchestra.

EAPS11.....**44.95**

Vol. 12
Dance Vocals



Featuring Dance inspired Vocals in the follow styles- Male & Female Spoken Words, Vocal FX, Male & Female Ad Libs, Female Choir. A great selection of material taken from the Best Service library Voice Spectral.

EAPS12.....**44.95**

Vol. 13
Choirs



Ahh Falls, Ahhh, All Shouts, Cluster, Consonants, Ehh, Falls, Female Shouts, Glissandi, Hah, Hee-ahh-ooh, Homm, Hoo, hoo-ahh-ehh, Houu, Laughter, Mhh, Ohh, Rrr, Vvv, Yhh, Zzz, Choir Female Duuh, Mmmh....

EAPS13.....**44.95**

Vol. 14
World Grooves



Latin flavored grooves such as Salsa Brass, Salsa Bass, Salsa Drum, Salsa Groove, Salsa Guitar, Samba Brass, Samba Bass, Samba Drums, Samba Grooves, Samba Guitar, Merengue Brass, Merengue Bass, Merengue Groove.....

EAPS14.....**44.95**

Vol. 15
Dance Drums



All the drums you'll need for your next Dance music production- Dance Bass Drums, Snare Drums, Dance Claps, Dance Hi Hats, Cymbals, Mixed Percussion, Bongo/Congas, Shakers, Tambourines, Toms, and more...

EAPS15.....**44.95**

Vol 16
Guitar & Bass



World class guitars and basses. Stingray Bass, Fender bass bends, Acoustic bends, Acoustic Finger Frets, Acoustic Guitar FX, Acoustic Harmonics, Acoustic Medium / Hard Pick / Pop, Blues Chugs, Hendrix Power 5th, more..

EAPS16.....**44.95**

Vol 17
Guitar Licks



Skilled guitarist Jeff "Skunk" Baxter of Steely Dan lays down grooves entitled Albatross, Black Friday, Slow Blues, Funk, Doobie, Heavy Slide, Hard Rock, Bo Diddley and Acoustic. Some from EastWest library J. B. Skunkworks.

EAPS17.....**44.95**

Vol. 18
Steinway Grand Piano



Featuring 32, 64, 128 and 256 Meg Piano samples of the beloved Steinway Grand Piano. A great selection of material taken from the EastWest library Ultimate Piano Collection.

EAPS18.....**44.95**

PRO SAMPLES

Vol. 19
Pop Brass



Featuring 3 Trumpet Falls Short & Long, 3 Trumpet Sustain, Trombone, 3 Trombones, Tenor Sax, Solo Trumpet, Alto Sax. A great selection of material taken from the EastWest library Quantum Leap Brass.

EAPS19.....**44.95**

Vol. 20
Orchestral Brass



Featuring Solo Trumpets, 3 Trumpets, 3 Trombones, Solo Tuba, 4 Horn Med, 4 Horn Forte, Big Low Stack. A great selection of material taken from the EastWest library Quantum Leap Brass.

EAPS20.....**44.95**

Vol. 21
Drum & Bass 2



Featuring Drum n Bass inspired Drum loops from 150 bpm to 170 bpm, Also includes Bass loops, Bass Notes, Guitars, Keyboard & Synths & Pads. A great selection of material taken from the Zero G library Total Drum n Bass.

EAPS21.....**44.95**

Vol. 22
Hip Hop



Hip Hop loops with titles like Show Me Your Love, Relaxin', Groovin' Down. Also included are Acid Jazz loops, Lo-fi Drum Kit loops, Skratch loops, Background loops, Guitar loops, Wah Guitar loops, Synth lix, Acoustic Bass lix.

EAPS22.....**44.95**

Vol. 23
Trip Hop



Twisted Trip Hop Loops, Bongo & Congo loops, Conga hits, Scratching, Leslie Guitar, Strings, Pads, Acoustic Bass licks, Electric Bass licks, Rhodes, & Single Hits. Some material taken from the Zero G library Twisted City.

EAPS23.....**44.95**

Vol. 24
Breakbeat



Featuring breakbeats with titles like Dopedancer, Tribe Vibe, Ride on Rim, Rock Hard, Hit the Break from superstar drummer Shawn Lee. A great selection of material taken from the Zero G library Planet of The Breaks.

EAPS24.....**44.95**

Vol. 25
Pop & Funk Brass



Brass riffs with titles like Power Tower, JB Soul, Earth Wind & Fire.s Rock n roll Tenor Sax, Baritone Sax, loops. Includes Multisamples: Solo Trumpet, Solo Tenor Sax, Solo Baritone, Trumpet & Tenor Section, Brass Section, more..

EAPS25.....**44.95**

Vol. 26
Nu RnB & Hip Hop



Featuring Hip Hop grooves in the style of Missy Elliot & Busta Rhymes. Loads of the hottest hip hop loops! A great selection of material taken from the Ueberschall library Da Nu RnB Hip Hop.

EAPS26.....**44.95**

Vol. 27
Dancefloor Bass



Featuring tons of single bass hits and multisampled basses both electric and acoustic. A great selection of material taken from the Best Service library XX Large Bass.

EAPS27.....**44.95**

PRO SAMPLES

Vol. 28
Vocal House



Mousse T's brilliant samples. Drumloops, Percussion loops, Guitar Licks, Organ loops, Piano loops, Rhodes licks, Vocal Ad libs, Female Vox, Male FX Vox, Basses, Brass, Guitars, Hits, Organs, Rhodes, Strings, more..

EAPS28.....**44.95**

Vol. 29
Dance Drumloops



Featuring drum loops for use in all musical styles- Dance, Disco, 2 Step, Pop, House. Tempo ranges from 80 bpm – 140 bpm. A great selection of material taken from the Best Service library XX Large.

EAPS29.....**44.95**

Vol 30
World Instruments



Instruments such as Saz 7 Stings, Erhu Plectrum, Erhu loops, Kantele, Mandolin Truxa, Sargon, Tibetan Singing Bells, Singing Bells Loops, Burma Bell, Snake Charmer, Irish Low Whistle, Mexican Piccolo, Ethnic Flute...

EAPS30.....**44.95**

Vol. 31
Rare Ethnic Instruments



Featuring instruments such as Duduk, Erhu, Highland Pipes, Chinese Gongs, Alpen Horn. A great selection of material taken from the EastWest library Quantum Leap Rare Instruments.

EAPS31.....**44.95**

Vol. 32
Ethnic Adventures



Featuring grooves with the essence of Africa, Cuba, India, China, Brazil, Japan, Jungle, Ballad, Havana, Bossa & Bolero. A great selection of material taken from the EastWest library Percussive Adventures

EAPS32.....**44.95**

Vol. 33
Orchestral Loops



Orchestral loops with movie title names - Hitchcock, Psycho, Scwartzeneger, Vertigo, Stravinsky, Predator, Manhattan. A great selection of material taken from the EastWest library Symphonic Adventures.

EAPS33.....**44.95**

Vol. 34
Hip Hop and R'nB 3



Featuring soulful Hip Hop /RnB grooves in construction kit format. Tenpo ranges from 56 bpm to 91 bpm. A great selection of material taken from the EastWest library Black Butta 1.

EAPS34.....**44.95**

Vol. 35
Funk 'n RnB



Featuring funky grooves in construction kit format. Tempo ranges from 65 bpm to 94 bpm. A great selection of material taken from the EastWest library Funky Ass Loops.

EAPS35.....**44.95**

Vol. 36
Chillout



Fantastic array of dreamy atmospherics and FX for use in ambient dance, film soundtracks or futuristic/spacey/trance music. Includes Drones, Analog Echo, Sweeps, White Noise Atmosphere, Ambient Percussion and more...

EAPS36.....**44.95**

PRO SAMPLES

Vol. 37
Dance Synths



Make your next dance track memorable! Acid loops, Bass loops, Piano loops, Moog Bass, Rave Bass, Crunch Bass, House Selection, Synth loops, Garage Organ, Filter Bass, Ethnic Percussion, Afro Drums, Flange Snares, FX...

EAPS37.....**44.95**

Vol. 38
Trance



Essential raw material for tranceheads worldwide. Trance style loops in construction kit format. Tempo ranges from 132 bpm to 142 bpm. A great selection of material taken from the Zero G library Trance Construction Kits.

EAPS38.....**44.95**

Vol. 39
Hip Hop 2



Featuring dope hip hop beats in construction kit format from the streets of NYC. Tempo ranges from 69 bpm to 99 bpm. A great selection of material taken from the Zero G library NY Cutz 2.

EAPS39.....**44.95**

Vol. 40
Breakbeat 2



Featuring the toughest raw breaks on the planet. Tempo ranges from 60 bpm to 100 bpm. A great selection of material taken from the Zero G library Beneath the Planet of the Breaks.

EAPS40.....**44.95**

Vol. 41
Solo Strings



Violin & Viola - pizzicato, pizzicato piano, spiccato piano, spiccato forte, martelato, mezzoforte, tremolo forte & piano. This version features 4 samples per octave.

EAPS41.....**44.95**

Vol. 42
Session Instruments



Featuring the Bongo, Conga, Rhodes, Trumpet, and Saxophone. A great selection of material taken from the Best Service library Xsample Volume 1.

EAPS42.....**44.95**

Vol. 43
Real Drum Kits



Featuring real drum kits- Rock, Jungle, Pop, Hip Hop, Swing, Funk. Including Rock & Pop Snare chromatic. A great selection of material taken from the Yellow Tools library Pure Drums.

EAPS43.....**44.95**

Vol. 44
Real eBasses



Featuring samples of great basses like the Ibanez BTB and Fender Jazz Bass. Included are slaps, pops, slides, ghost notes, harmonics, fingered notes. A great selection of material taken from the Yellow Tools library Pure e Basses.

EAPS44.....**44.95**

Vol. 45
Techno ID



Featuring techno inspired Drum loops, Percussion loops, Bass loops, FX loops, Vox loops, Synth loops, Hooklines. A great selection of material taken from the Ueberschall library Techno Identity.

EAPS45.....**44.95**

PRO SAMPLES

Vol. 46 Flamenco



Recorded exclusively in Spain, this CD captures a wealth of traditional and modern rhythms and flavours played by original respected Flamenco musicians - names such as Ketama, Tino De Geraldo and Pedro Andrea among others. Includes guitar rhythms, Male Cantaor and Female Cantaora (vocals), tapping feet, solo tapping, castanets, shouted vocals and Cajon rhythms. Bring out the gypsy in your music!

"All ... you need to add either a hint of Flamenco to an existing track or construct your own Spanish-style dance floor smash... the feel of true Flamenco music ...." (SOUND ON SOUND, UK).

EAPS46.....**44.95**

The people have spoken!

ProSamples, a series from three of the world's most critically acclaimed Soundware developers — BEST SERVICE, EASTWEST, and ZERO-G — is such a huge success that EASTWEST sent their team back into the studio to churn out five more new volumes. Pro Samples is the fastest selling series in history!!

It is with great pleasure that they introduce the latest Pro Samples offerings, Vol. 46-50. Each new ProSamples volume includes up to seven formats (with the exception of the multisampled instruments, which don't include ACID).

Formats include ACID, AUDIO, AIFF/MAC, AKAI, EXS24, HALion, and WAV/PC.

Vol. 47 Breakbeat 3



Returning to the "rawfunkindirtyassboogiebeats" that have made the Planet Of The Breaks series such a phenomenon, Emre Ramazanoglou delivers a huge selection of fresh and raw grooving breakbeats. Edgy and tough, small, large, fast, fat, lo-fi, hi-fi, mid-fi, broken, smashed, cuts, drops and extreme funk with quintessential quirk! Shawn Lee (Creator of the 5 star, Key Buy Award, Platinum Award 'Planet Of The Breaks' and 'Beneath The Planet Of The Breaks') was blown away when he heard these beats. Return To The Planet Of The Breaks will add that extra dimension of groove to your projects and tracks.

EAPS47.....**44.95**

Vol. 48 Disco Fever



Very hot stuff from smash CD Disco De Luxe - authentically produced for disco/house/frenchhouse/funk&soul productions. inspired by often-sampled disco vinyl for hot club burners. contains construction kits, drumloop kits, strings, brass, flute, guitars, basses, vox, single sounds. Relight your fire!

EAPS46.....**44.95**

Vol. 49 FX_Area



An essential fx library for music, film and webdesign. well-sorted professional effects: button / signals / movements / mechanical gears, data stream, guns, robots, scanners, crafts, clocks.. / electricity, air.. / vox: robots, alien / atmos: explosion, time-trip, underwater space, control-room.. / and even background-loops.

EAPS49.....**44.95**

Vol. 50 Real Guitars



Guitars from Pure Guitars by Yellow Tools. Real Guitars – as if they were played live! Nylon Concert, Steel Pick, Steel Fingered, 12-String, Bass Fingered. The best guitars and basses were used. Offers the user authentic, realistic (unlooped), and dynamic guitar performances. In addition, there are different FX volumes, chords and harmonics.

EAPS50.....**44.95**

RAW

Compatible, convenient and competitively priced, 2rAw is the sample product you've been waiting for! These CDs feature sounds in ReCycle (Rex) files, Reason refill and Acidized wav files for enhanced use with all the hottest Mac and PC music creation software. No matter which program you use, getting the sounds into your project is quick and easy, so you spend less time converting and editing files and more time creating!

Funkmaster



If you want real, pure, original funk, you have to go to the source, James Brown. The Funkmaster Jabo Starks is on more James Brown classics (Sex Machine, Good Foot) than anyone and kicks out some of his funkiest beats, along with a slamming virtual band - including bass, guitar and percussion played by top notch funksters.

This library contains the following 201 Acid/wav loops: Drum, Bass, Guitar, Percussion, 163 Rex2/Refill files- Drum, Bass, Guitar, Percussion, Construction kits, 112 Drum hits and variations of The Jabo Starks Redrum Kit.

Multi-Format..... **44.95**

ExtremeROCKFUNK

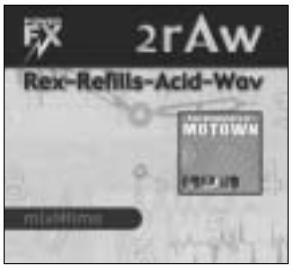


ExtremeRockFunkRage collection is geared to fill the needs of samplists, with some of the extremely powerful and contemporary rock grooves, riffs and loops that are so prominent in much of today's music. The construction kits are played by Unstoppable, one of the LA's most underrated and influential rock trios, with a unique blend of hard rock and pocket funk, PLUS plenty of extra guitar, bass drum and vocal scream riffs with an Extreme attitude.

Compatible, convenient and competitively priced. Feature sounds in ReCycle (Rex) files, Reason refill and Acidized wav files for enhanced use with all the hottest Mac and PC music creation software. Getting the sounds into your project is quick and easy!

Acid/Reason..... **44.95**
Audio.....**89.95**

Drummers of Motown



Add some history to your drum tracks with the drums and percussion from the Drummers of Motown. Pistol Allen, Uriel Jones and Jack Ashford have played on over 350 hit records, the original Hitsville USA groovemasters who supplied the beats behind tracks like I Heard it Thru the Grapevine, Tears of a Clown and My Girl. Now get total control of the these classic loops and rhythms\. Unprecedented control over time stretching loops. 252 Acid/Wav Drum and percussion loops, 248 Rex2/Refill files, 72 Wav file guitar sounds, 1 Motown Redrum Kit (29 Drum hits)

Acid/Reason..... **44.95** **Audio** **89.95**

Dub Tactic Manual



Straight from the tape vaults of visionary producer Bill Laswell, Dub Tactic Manual stokes the dub fires with an awesome collection of Material style dub/ambient elements that transcend musical boundries. Loads of dub bass lines, gritty breaks, earthy percussion and spacey keys to infect your tracks with a serious sound virus. This multiformat disk allows for sample access convenience and tempo control so you can spend your time more creatively. Wicked Material.

258 Acid/wav Bass, Beats, Keys, Percussion , 207 Rex files Bass, Beats, Keys, Percussion

Acid/Reason..... **44.95**
Audio.....**89.95**

Compatible, convenient and competitively priced, these CDs feature sounds in ReCycle (Rex) files,
Reason Refill and ACIDed wav files for enhanced use with all the hottest Mac and PC music creation software

New World Order 1



The New World Order series scans the globe to bring you unique loops from around the world. NWO 1 contains drum and percussion loops and grooves from India, Africa, Cuba, the Carribean, Japan, Ireland, Texas - that's right Texas, and Brazil. Awesome grooves from instruments like the tabla, the talking drum, the daf, the dumbek, the bodhran, the djembe and more. Control the world with these acidized and rex2fied beats.

Multi-Format.....**44.95**

New World Order 2



The New World Order series scans the globe to bring you unique loops from around the world. NWO 2 contains music loops and phrases from instruments like the Balalaika from Russia, the Gombri from Algeria/Morocco, the French Hurdy Gurdy, the Kora from Senegal, Balinese Gamelan, Key Harps from Sweden and different stringed instruments from India and Ireland.

Multi-Format.....**44.95**

Club Spectrum



Lab tools for the Remixologist. Club Spectrum provides a killer variety of the necessary essentials for making dance music. 4 the floor beats, bass lines, synth riffs to create house, trance, techno, electro, retro, drumnbass… all in Acid/wav, and REX2 formats to easily work with a song´s tempo. Mix in time, Mix and refine.

Multi-Format.....**44.95**

Uptown Breaks



PowerFX drops the BOMB!!!! New breakout beats for the NOW feel in R&B/Hip Hop/Nu Dance Pop. Wicked beats and syncopations with variations for butt moving connotations. Ultra modern-futuristically hip, Uptown Breaks is the real deal!

The formats included in this pacakge are Reason, REX 2 & Acidized WAV

Multi-Format.....**44.95**

FX in Time



This collection features loops supplied in Acidized .wav and .rx2 (REX) file formats, as well as a Reason ReFill.

The loops are the result of performances involving hand tools, power tools, garden and household equipment and lots of other mechanical bits and pieces "played" in a musical manner – for anyone who wants out of the ordinary loops and rhythm patterns to give their projects a very unique and distinctive edge. An unusual and creative palette of sounds and textures.

Multi-Format.....**44.95**

Club Track Builders



Six top dance producers from four countries make Club Trackbuilders a unique and versatile tool for anyone creating club music with an edge! House, techno, 2-step, drumnbass styles with beats, bass and synth loops in rex-refill-acid-wav file formats. Spend more time creating great dance music.

Contains: 304 Acid/wav files, bass, beats, rhythm, synth 304 Rex2/Refill files- bass, beats, rhythm, synth files.

Acid/Reason..... **44.95**

Audio.....**89.95**

HARD CORE

Without a doubt, BT is one of the fastest rising artists in today's music scene. He has written the score for the #1 box office hit "Fast and the Furious", as well as scores for "Driven", "Go", and "Under Suspicion", produced N'Sync's single "Pop", released his own album "Movement in Still Life" (which received rave reviews), and most recently released the double re-mix album "R&R".

Breakz From the Nu Skool



The first of two sample CDs in BT's signature collection. Features cutting edge breakbeats hand mangled through hundreds of plug-ins, stomp boxes, pressed to vinyl, you name it! From Kyma to Reason, Breakz from the Nu Skool streams through on all frequencies of subconscious awareness. Best of all this CD is sample accurate! No more flaming as you stack loop upon loop. Drop them into any AKAI compatible Sampler and you're off and sprinting. Also included is a bonus Audio version for fast auditioning. Once you've busted some Breakz from the Nu Skool, you won't look back!

Audio/WAV **89.95**
AKAI and GIGA (Halion compatible).....**224.95**

Twisted Textures



It's time to unleash the power of art, spirit and sound with BT's new sample CD - Twisted Textures. Twisted Textures is a massive two disc collection of sounds, pads, and waveforms that will lead you into all realms of the human condition and beyond the doorways of ordinary reality. Included with these amazing textures is a guide organized by emotion to help you narrow your search as you explore the complexity and depth this CD has to offer. So no matter what project you are working on, Twisted Textures has a sound for you. Suspend time and open your heart as you enjoy Twisted Textures.

Audio/WAV **89.95**
AKAI and GIGA (Halion compatible).....**224.95**

Monster Beats



A double CD set (audio plus WAV CD-ROM) loaded with over 75 minutes of in-your-face Rock, Funk, Fusion and Alternative grooves ranging from 60 to 203 bpm. Recorded and mixed with state of the art recording technology at a number of London's top studios, Monster Beats showcases the talents of an acclaimed session drummer and recording engineer at the top of their game. The drums are played with attitude, but never over-played as on some drum CDs... Monster Beats has a usability that no other loop based CD can beat. With extreme re-mix variations of EACH groove utilising some of the most sought after outboard FX and eq, plus custom analogue modular synthesizers. Complete drum performances, precisely timed Wav files are included - ensuring each and every beat can be looped at any point with exact accuracy!

MONSTER BEATS is produced by Chris Whitten and Peter Henderson and features 89 live drum kit performances based on Chris' favourite tracks, from rock to fusion with a large dose of alternative grooves. Monster Beats features natural born grooves, played with a real human feel serving up clean, gritty and raw beats, including sections without kick drum, solo hi hat and more drum fills than have ever been available before. Sound on Sound recently said, "The loops are tightly played and extremely well recorded, with just the right amount of grit and grime…Producers and composers from all genres looking for something extra to spice up their drum tracks will find plenty to use here".

AUD/WAV **89.95**
Giga **179.95**

Brutal Beats



From the producers of the KEY BUY and award-winning CHEMICAL BEATS! Loaded with the most intense and aggressive drum grooves ever. Recorded and mixed with state of the art recording technology (SSL 9000J) in one of the world's top drum rooms (Townhouse Studio 2's Stone Room in London), Brutal Beats is a tour de force of loud, rocking, edgy loops. With extreme re-mix variations utilizing propriety software and hardware and some of the latest processing devices, it has an edge that no other loop based CD can beat. With precise timing for each and every beat, grooves can be looped at any point with exact sample accuracy!

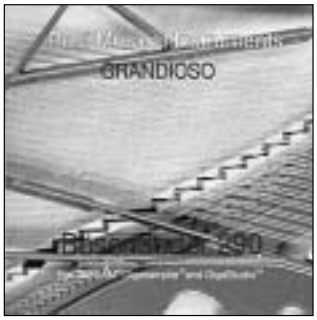
This library features 47 live drum kit performances based on classic 70's rock tracks by Geoff Dugmore, one of Britain's most in-demand session players. The beats include intros, fills, verse & chorus grooves etc. A massive variety of grooves are provided, including straight and shuffle feels, slow blues, odd-time patterns and many more. You will find a huge range of tempos from under 60bpm to over 200!

All performances were recorded with full 24-track digital recording mixed down to stereo. 'Straight' (mildly compressed) and 'Compressed' (pumping) mixes of all performances are provided, with up to TEN stages of digital and valve compression and limiting. Over 100 radically mutated dance remixes are included. The individual hit library is also provided to allow maximum flexibility and creativity.

AUD/WAV **89.95**
Giga **179.95**

PIANO LIBRARIES for GIGA

Post Musical Instruments Grandioso Bosendorfer 290



This Bosendorfer library provides the greatest possible control during the softest pianissimo, through crescendos to the reserves of power needed for the loudest fortissimo.

By utilizing new technology to optimize the mechanical performance of the action, Post Musical Instruments has created a product which leaves the pianist in total control of dynamic response, timbre and touch. PMI has finally captured a grand piano with "breath". They recorded both dry samples and ambient samples. The dry samples are recorded fairly close to the piano strings. The wet samples are recorded at a distance so that the hall acoustic is captured.

You can experience a concert hall type of sound and control the amount of ambience until you play it absolutely dry.

Over 2 GB of samples, up to 16 recorded layers of velocity, separate sustain pedal up and down samples, multiple release samples... separate "dry"and "wet" samples that can be mixed for ultimate ambience control.

Giga **219.95**

Post Musical Instruments Granioso Steinway D



Recorded with the best grand piano they could find: a 1965 premium condition Steinway model D3. It was fully refurbished by Steinway Hamburg in 1999. This piano served the Rotterdam 'DOELEN' concert hall for several decades, where hundreds of famous musicians, from Claudio Arrau to the Rolling Stones, performed for live audiences and broadcast concerts. Up to 6 articulations were captured (PPP, PP, P, MF, F, FF and FFF) for sustain pedal up, sustain pedal down and 4 articulations for the release triggered samples. Samples were recorded using the finest digital equipment available. Prism Sound, a 24-bit ProTools TDM Mix+ system and Waves processing was used. They mapped up to 16 levels of velocity, true multiple velocity release layers, ultimate staccato, and sustained pedal-down samples with a carefully chosen amount of resonance. The library has 5 GB of samples. The end result was tested by several concert pianists, who helped develop a sampled instrument which could meet their highest expectations. There is no doubt in our minds that this is the best sampled piano ever.

Giga **219.95**

Bosendorfer Sampled Piano Library



EASTWEST's BOSENDORFER 275 sampled grand piano features up to eight stereo dynamics per note for unparalleled expression and realism. The Bosendorfer Grand Piano was recorded in Switzerland by the producer of the critically acclaimed FIVE STAR Sample Library THE ULTIMATE PIANO COLLECTION, Olivier Truan. It is a 1995 BOSENDORFER 275 (9 foot 1 inch) Concert Grand built in Austria. It has an extra range down to F0, 92 keys. Almost every note was sampled in 8 stereo velocities, 4 with the pedal up, 4 with the pedal down for unparrelled realism and expression. 736 stereo samples were recorded in total, without any looping (up to 2 minute decays). The piano was digitally recorded and edited direct to disc to achieve the highest possible quality.

Giga **179.95**

Steinway B Sampled Piano Library



EASTWEST's STEINWAY B sampled grand piano features up to six stereo dynamics per note for unparalleled expression and realism. The Steinway B Grand Piano was recorded in Switzerland by the producer of the critically acclaimed FIVE STAR Sample Library THE ULTIMATE PIANO COLLECTION, Olivier Truan. Almost every note was sampled in 6 stereo velocities, 3 with the pedal up, and 3 with the pedal down. 528 stereo samples were recorded in total, without any looping (up to 2 minute decays). The piano was digitally recorded and edited direct to disc to achieve the highest possible quality. Extra care was taken in respect to A/D conversion, recording levels, and dynamic range to guarantee the highest bit-fidelity.

Giga **179.95**

EASTWEST

ADVANCED ORCHESTRA



Advanced Orchestra Set

Peter Siedlaczek optimized this version of the acclaimed ADVANCED ORCHESTRA to take advantage of the incredible power of Gigasampler. Using the 16 lowest keys as control keys users can access gigabytes of sounds instantly and switch between different sound categories in real time!! Advanced Orchestra is new standard for symphonic samples from the producer of the quadruple FIVE STAR sample CDs “Orchestra” and “Orchestral Colors” - Peter Siedlaczek. Advanced Orchestra contains over 5,000 samples of all Instruments and instrument groups, as well as a large symphony orchestra. Besides sustained and staccato notes on two dynamic levels, you will also find a series of new and useful samples which have never been released before. There are numerous characteristics that distinguish this library from others. The most important are: A variety of important and frequently used phrasings and articulations, which cannot be satisfactorily imitated with normal samples, for example, half or whole tone trills, various grace notes, runs, arpeggios, tremolos, crescandi, glissandi and other events playable in any key or pitch. A collection of unconventional, interesting sounds played on the classical instruments such as scratches, whistles, thumps, buzzes and other effects, which can be technically modified in many ways, adding a new color to your music. All of the staccato notes and some other events are sampled twice. If a quick repetition of the same note is needed, you can use them alternately. The result is more natural and not so “mechanical”. The strength of this sound library lies not only in classical music or “real sounding” natural structures and arrangements, but also many other types of music. Especially the complex phrasing and articulations, such as runs, trills or glissandi, are suitable for all types of distortion and unorthodox treatment. Use your imagination and experiment. A short sketch demonstrating the potential of this library - titled ‘A Day in a Philharmonic Hall’ - can be heard on the first disc (and East-West's Demo CD). It consists (with the exception of the drum loop) solely of samples from this library. Using this library, you can now arrange complex and amazingly realistic soundinq structures. More realistic than ever before. The library includes:

String Ensembles
(instruments: 14 violins, 10 violas, 8 celli, 6 basses)

Solo Strings & Effects
(instruments: violin, viola, cello & unconventional sounds).

Woodwinds
(instruments: 3 flutes, solo flute, alto flute piccolo flute, oboe, english horn, 3 clarinets, bass clarinet, bassoon, contrabassoon.

Brass and Solo Clarinets
(instruments: 3 trumpets, solo trumpet, piccolo trumpet, 3 trombones, solo trombone, french horns, solo french horn, tuba and solo clarinets).

Percussion & Harp
(instruments: cymbals a due, cymbal 24″, cymbal 18″, timpani, gran casa (bass drum), snare drum, large tam-tam (gong), thai gongs tuned), chinese opera gongs, triangles, vibraphone, marimba, xylophone, orchestra bells (glockenspiel), tubular bells, harp etc.

Giga **495**

Advanced Orchestra Compact
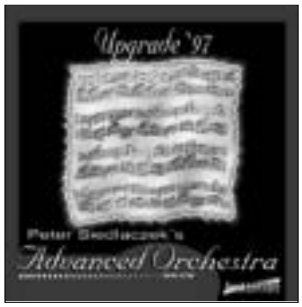


The ever-growing need for orchestral samples and the continuing success of the "Advanced Orchestra “library created a lively demand for an affordable” light" version of this internationally renowned library. The Compact Edition contains – thanks to some clever “compression techniques” – the most useful samples from the “big” library which need much less RAM than the "big" samples do.

Contents: Strings ff & mp (sustained, tremolo, con sordino, staccato, pizzicato), Brass & Woodwinds ff & mp (stacatto, sustained); Orchestral Percussion, Harp.

AKAI S1000 ..... **134.95**

Advanced Orchestra Upgrade



Advanced Orchestra Upgrade is a 3 CD-ROM set that adds functionality to Advanced Orchestra, but also was designed as an independent library.

Upgrade ‘97 contains:

NEW sample categories found in no other libraries.

True Legato Strings, NEW sound qualities, New Instruments, Bass Flute (this “magical” instrument is a must), Harp Single Notes, Percussion II, Hollywood Strings and Main Library Short Loops.

AKAI S1000 ..... **269.95**

ADVANCED ORCHESTRA

Advanced Orchestra Volumes 1-5



Available Individually

The new standard for symphonic samples from the producer of the quadruple FIVE STAR sample CDs “Orchestra” and “Orchestral Colors” - Peter Siedlaczek. Advanced Orchestra consists of 5 CD-ROMS (the Emu format has 6 CD-ROMs) containing 5,455 samples of all Instruments and instrument groups, as well as a large symphony orchestra.

Advanced Orchestra Volume 1

String Ensembles contains: 14 Violins, 10 Violas, 8 Celli, 6 Basses

Also included: 14 Violins, 10 Violas, 8 Celli, 6 Basses in Mono 22 kHz.

Above instruments played Con Sordino, Short detache, Sustained notes, Pizzicato, Col legno, Tremolo, Crescendo, Trills, Grace Notes, Short Chords, Runs (major & minor), Arpeggio (major & minor), Glissando.

Also available in Audio, EMU, GigaSampler and Roland.

AKAI S1000 .....**149.95**

Advanced Orchestra Volume 2

Solo Strings contains: Solo Violin, Solo Viola, Solo Cello

You'll find all the above instruments: Short detache, Sustained Notes, Spiccato, Solo expressivo, Con Sordino, whole tone ascending notes, Trills, Runs (major & minor), Short chords, Arpeggios (major & minor), Slow & Fast Glissando, Tremolo, Pizzicato, Crescendos.

Also available in Audio, EMU, GigaSampler and Roland formats.

AKAI S1000.....**149.95**

Advanced Orchestra Volume 3

Woodwinds contains: 3 Flutes, Flute, Alto Flute, Piccolo Flute, Oboe, Bassoon, Contrabassoon, Solo Clarinet, English Horn

You'll find the above instruments:

Staccato, Grace Notes, Runs (major & minor), Flutter tongue, Sustained notes, Crescendo, Trill, Mordent, Solo Vibrato, whole tones and half tones ascending and descending.

Also available in Audio, EMU, GigaSampler & Roland formats.

AKAI S1000.....**149.95**

Advanced Orchestra Volume 4

Brass & Effects contains: 3 Trumpets, Solo Trumpet, Piccolo Trumpet, Tuba, 3 Trombones, Crescendo, Solo Trombone, Solo French Horn, 4 French Horns,

You'll find the above instruments in: Staccato, Sustained Notes, Fast, slow & short glissando, Trills, Valve flutters, Flutter Tongue crescendos, Lip trills and more!

Also included are effects:

FX "blowing for the wind", FX "shock treatment", FX " Short & Percussive events, FX "unsingable melodies for everyone", FX "clusters & longer events", FX "the wonderful world of animals", FX "breath of a wind player"

Also available in Audio, EMU, GigaSampler & Roland formats.

AKAI S1000 .....**149.95**

Advanced Orchestra Volume 5

Percussion and Harp contains:

Cymbals A Due American Symphonic, Cymbal 24", Cymbal 18", Timpani, Gran Casa, Snare Drum, Orchestra Bells, Large Tam Tam/Gong, Thai Gong, Chinese Opera Gong, Triangles, Xylophone, Vibraphone, Marimba, Harp, Tubular Bells

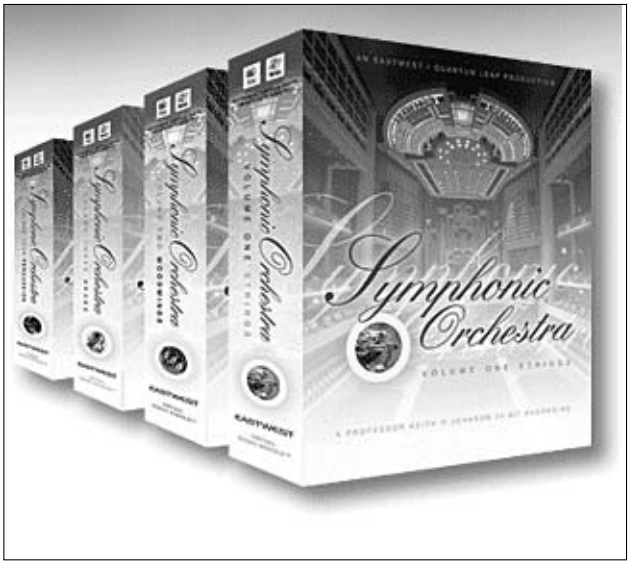
You'll find the above instruments: single crashes, glissando, single strokes, tremolo, crescendos, single hits, strokes, various effects and more!

Also available in Audio, EMU, GigaSampler & Roland formats.

AKAI S1000.....**149.95**

EASTWEST

SYMPHONY



Symphonic Orchestra

A Doug Rogers and Nick Pheonix Production
A Prof. Keith O. Johnson Recording

The EASTWEST / QUANTUM LEAP Symphonic Orchestra was recorded where orchestral instruments sound their most natural - in a 'state of the art' concert hall. Listen to the pure beauty of the "critically acclaimed" Symphonic Orchestra and "state of the art" 125 million dollar Concert Hall that were captured, using custom built recording equipment, by audio legend - Prof. Keith O. Johnson - recipient of two GRAMMYS, plus eight additonal GRAMMY nominations for his 90 plus cassical recordings.

The Complete Symphonic Orchestra Library

Volume One - STRINGS *Volume Two - WOODWINDS*
Volume Three - BRASS *Volume Four- PERCUSSION*

FEATURES

- This is the first 24-bit orchestral sample library to include three simultaneous stereo mic setups (close, stage and hall), so users can mix together any combination of mic positions to control tonality and ambience.
- Concert hall environment offers a surprising amount of tonal control, virtually eliminating the need for EQ; and the ambient control, using any combination of these three stereo tracks, eliminates the need for artificial reverb (which usually sounds artificial).
- Create the sound "you" want. With other orchestral libraries, you get what they give you, and apart from adding reverb and EQ, you have virtually no control over the sound of the instruments. The control of tonality and ambience in the Symphonic Orchestra enables each user to "design" the sound of the orchestra to suit the work that is being created - no other orchestral collection offers this!
- In addition to this unprecedented control of tonality and ambience; the real ambience of the hall, combined with the automatic sample switching of up and down strokes that is built into the NATIVE INSTRUMENTS audio engine and interface (included with each volume of the library), has eliminated the "machine gun" effect when playing fast repetitions.
- Use the stage mics for a typical big Hollywood sound; boost certain instruments by adding a hint of the close mics; add in the hall surround mics to create a surround sound mix, or to add natural hall reverb. Finally, you can say goodbye to lifeless, dry studio recordings.
- All of the players were recorded in position. All of the instruments and sections were chromatically sampled, with multiple dynamics, with extremely dynamic and expressive articulations, and – although the library is huge, it has been programmed to be "extremely easy to use". For example, an efficient method of composing would be to use just the "close mics" or the "stage mics" during the composition stage, adding a combination of the three stereo pairs to adjust ambience and tonality in the final mixing stage.
- Each volume comes as a Virtual Instrument which includes a high-performance Native Instruments interface powered by the efficient KONTAKT audio engine (see page 1099 for details) including Direct-from-Disk streaming for Mac and PC.
- Each Virtual Instrument in the collection combines intuitive handling with excellent sound quality, and functions as a plug-in instrument, without the need for a sampler, in hosts such as Cubase, Digital Performer, Logic, Nuendo, Pro-Tools, Sonar etc.
- Professional sound quality with advanced 32-bit processing; eight-part multitimbrality and 256 voice polyphony allow for lush arrangements.
- Clean and intuitive interface never lets the technology get in the way of your creativity. All important instrument parameters can be immediately adjusted without the need to page, scroll, or fumble through a manual.
- For the sonically adventurous, many sound shaping tools have been built in to the interface. Powerful multi-mode filters, envelopes, and LFOs give a wealth of creative possibilities. Integrated high-quality reverb, chorus, and delay effects are also provided.
- Producers Doug Rogers from EASTWEST and Nick Phoenix from QUANTUM LEAP, have collectively received over 30 international awards for the sample libraries they have created.

Available as individual volumes or as a library.....2695.00

Symphonic Orchestra Vol. 1 Strings



18 Violins: •expressive vibrato p, f •expressive vibrato slow p, f •straight vibrato p, mf, ff upstroke, ff downstroke •lyrical vibrato slow, fast attack •legato vibrato p, mf, f •sordino expressive mp, mf •sordino fast mp, mf •sliding sustain, slow, medium, fast and imperceptible speeds •non-vibrato p, mf •tremolo 1 & 2 •pizzicato, p, mp, mf, f •martele up and down •short martele up and down •marcato short, medium and long •effects

11 Violins: •expressive vibrato dimuendo p, f •expressive vibrato p, f •lyrical vibrato p, f •straight vibrato p, mf, ff •legato vibrato p, mf, f •glissando short, long •trill half step •trill whole step •spiccato •short martele up and down •marcato short and long

10 Cellos: •expressive vibrato, slow and fast attack •straight vibrato p, mf, ff •legato vibrato p, mf, f •sordino, slow, medium and fast attack •crescendo long •sforzando long •pizzicato, p, mp, mf, f •non vibrato •portato short and long •trill half step •trill whole step •tremolo •martele up and down mf, f •short martele up and down mf, f •effects

10 Violas: •expressive vibrato p, f •expressive vibrato slow p, f •straight vibrato p, mf, ff •legato vibrato p, mf, f •martele up and down mf, f •short martele up and down mf, f •marcato short and long

9 Basses: •expressive vibrato, slow and fast •straight vibrato p, mf, f, ff upstroke, ff downstroke •legato vibrato p, mf, f •crescendo long •sforzando short •pizzicato, p, mp, mf, f •martele up and down mf, f •short martele up and down mf, f •marcato short •portato •tremolo •bow slaps •effects

Solo Violin: •expressive vibrato 1, 2, 3 •crescendo •slide •vibrato, p, mf, f •non- vibrato legato, p, f •martele up and down •marcato

Solo Cello: •expressive sustain 1, 2 •sforzando •slide •vibrato legato, p, mf, f •non-vibrato legato, mf •martele up and down •marcato f

Harp: •mp, mf, f •roll finger, mp •harmonics

Multi-Format.....**895.00**

Symphonic Orchestra Vol. 2 Woodwinds



3 Flutes: •vibrato p, mf, f •non-vibrato mf •expressive vibrato •trill half-step •trill whole step •glissando, long and short •grace notes •staccato

C Flute: •vibrato mp, f •non-vibrato mf •expressive vibrato 1 and 2 •staccato p, mf, f •flutter •grace notes •falls

3 Oboes: •vibrato mp, f •non-vibrato mf •expressive vibrato mp, mf •staccato p, mf, f •trill half step •trill whole step •grace notes

Solo Oboe: •vibrato mp, f •non-vibrato mf •expressive vibrato 1 and 2 •staccato p, mf, f •trill half step •trill whole step •falls •octave runs •glissando •grace notes

Alto Flute: •vibrato mf •expressive vibrato 1 and 2 •non-vibrato mf •staccato p, mf, f •glissando

Piccolo Flute: •vibrato mp, f •expressive vibrato •staccato mp, f •glissando •trill half step •trill whole step

English Horn: •vibrato mp, f •expressive vibrato 1 and 2 •non-vibrato mf •staccato p, mf, f •falls •octave runs •glissando •grace notes

Bassoon: •vibrato mf •non-vibrato mf •expressive vibrato 1, 2 and 3 •portato •staccato, p, mf, f •glissando

Contra-Bassoon: •vibrato mp, f •expressive vibrato •non- vibrato mf •portato •staccato mp, f •glissando •grace notes

3 Clarinets: •non-vibrato mp, f •staccato p, mf, f

Solo Clarinet: •non-vibrato mf •expressive non-vibrato 1 and 2 •portato •taccato p, mf, f

Bass Clarinet: •non-vibrato mp, f •expressive non-vibrato •portato •staccato mp, f •glissando

Multi-Format.....**895.00**

Symphonic Orchestra Vol. 3 Brass



6 French Horns: •sustain p, mp, mf, f, fff •staccato p, mf, f, fff • long staccato fffvportato, mf, fff •portato sfz, mf, fff •sforzando f •legato slides, short and long •legato p, mp, mf, f, fff •rips, short 1 and 2, and long •shakes •stop mutes

Solo Horn: •sustain, p, mf.ff •staccato p, mf, ff •portato, mf, ff •sforzando f

4 Trumpets NV: •sustain mp, mf, ff •sustain accent mp, mf, ff •legato mp, mf, ffvstaccato, mp, mf, ffvsforzando short •forte-piano •slides fast and slow

Solo Trumpet (C): •expressive vibrato 1, 2 and 3 •non-vibrato mp, mf, ff •staccato double tongue position 1 p, mf, ff •staccato double tongue position 2 p, mf, ff •sforzando •portato •slides fast and slow

4 Bass/Tenor Bones: •sustain p, mf, ff •sustain accent p, mf, ff •staccato p, mf, ff •portato •sforzando •flutter tongue •crescendo short, medium and long

Solo Bass/Tenor Bone: •sustain p, mf, ff •staccato p, mf, f, fff

3 Wagner Tuben: •sustain mp, f •sustain accent mf •portato fff •rips

Solo Tuba: •sustain mp, f •sforzando •staccato, mp, mf, ff

Multi-Format.....**895.00**

Symphonic Orchestra Vol. 4 Percussion



Percussion: •5 Timpani with Calf Heads •.2 Bass Drums •4 Snares •5 Concert Toms •Funeral Drum •Tenor Drum •Field Drum •4 Anvils •Vibes •Crotales •Glockenspiel •Xylophone •6 Suspended Cymbals •6 Cymbal pairs •Band Cymbal •5 ft. Gong •4 ft. Gong •2 Normal Gongs •Bell Tree •Mark Tree •Orchestral Chimes •Triangle •Lots of toys

Multi-Format.....**445.00**

Prof. Keith O. Johnson has spent over 30 years developing a reputation for innovative thinking, technical achievement and musicianship which has elevated him to a position in the audio industry occupied by only a handful of visionaries. His intensive investigation of electronic behavior and acoustic perception have led most recently to his development (with digital engineer Michael Pflaumer) of the revolutionary High Definition Compatible Digital encoding process, produced and marketed by Pacific Microsonics (recently acquired by Microsoft). His 90-plus recordings have long been considered the standard for high fidelity, and include two GRAMMY award-winners and eight additional GRAMMY nominations.

BEATS

Downtown Beats



If you are looking for the latest Nu Pop/Hip Hop/R&B beats that can be heard on records but not found on any current sample CDs, check out "Downtown Beats". Need the hottest and latest breaks ala Cherion Studios (Britney, N´Sync, BackStreet Boys) or Destiny`s Child, Outkast or Dirty South, the groove is right here. Beats, breakdowns, hits – organized in the meticulous PowerFX way. Wicked beats to write hits on. Plain and simple.

Audio **89.95**

Downtown Orchestra



Downtown Orchestra is the MODERN MASTERWORK sample cd from PowerFX. Strings, brass, woodwinds and percussion loops and samples beautifully written, played and recorded specifically for contemporary production. You get entire orchestral parts and sections, plus a killer set of "low string riffs" that rock the house. The highly inspirational phrases in this offering scream to be turned into your "latest and greatest" composition. Audio + Wav discs included.

Audio/WAV **89.95**

Downtown Strings 1



The finest collection of R&B/Hip Hop strings ever made. This beautifully recorded collection features custom string arrangments written specifically for today`s urban producers. String quartet construction kits, full string ensemble passages, killing upright bass grooves, unique and useful Harp loops and riffs as well as an ample selection of multisampled scales. Real strings from real players for the real deal.

Audio **89.95**

Downtown Strings II



This is an impeccable collection of pristine, 24 bit Wav, quartet arrangements organized into 5 separate categories: Cinematic, Disco, Hip-Hop, Pop & Rock and Modern Classic. Features string quartets containing each instruments part as a separate file as well as an "all" section where the instruments play the piece together.

WAV **89.95**

Nu House



A fantastic collection of tough construction kits and individual samples from Zero-G ready to roll: chunky drums and loops, pounding bass lines, cutting synths and bubbling filters. Thirty construction kits, with tracks up to 3 minutes long, and a second CD full of loops, drums, synths, pads, arps, swells, hits and fx. Rex Files accessed using Reason.

Audio/WAV/Refill.....**89.95**

House Musique



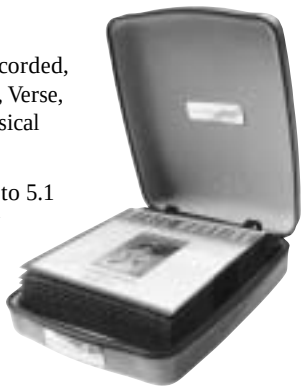
Fresh & funky disco house material. Inspiring, dynamic construction kits. Extra lick & loop material: e-bass (73'stingray), funk & wahwah guitars (65'strat), flute, clavinet (D6), organ (B3), acoustic piano, e-piano (mk1); plus useful drum & percussion collections for exs24, halion & kontakt; all sounds also in .wav on CD2.

Audio/WAV **89.95**

Beats Working – Pro Tools



Zero-G's BEATS WORKING offers an indispensable first-call toolbox of immaculately recorded, highly usable drum performances grouped in versatile song construction sets, with Intro, Verse, Bridge, Chorus, Drops and Fills of every performance, covering a diverse selection of musical styles from Classic Pop to Big Band.



The world's first Professional Drum Performance Library recorded in 10 channels for mixing to 5.1 surround sound (and beyond). Recorded at the legendary ABBEY ROAD Studio 2 in London by top recording and mix engineer Haydn Bendall, and performed by top session drummer Ralph Salmins, the sound and performances are impeccable. Utilising high-end esoteric microphone pre-amps, and a direct signal path to a Euphonics R1, it is hard to imagine a purer, cleaner signal path.

In addition, multiple kicks, snares (both modern and vintage), toms, cymbals and hats were recorded at multiple velocities, giving you the ultimate flexibility to design your own kit.

9 disc ProTools version (24-bit SDII files - 10 channels of audio for each performance) **445.00**

Audio/WAV (ACIDized) 16-bit stereo 2-disc set **116.95**

B&H PHOTO - VIDEO - PRO AUDIO

BREAKS

Escape from the Planet of the Breaks



Shawn Pelton, long time Saturday Night Live drummer and one of the most in demand drummers in the world, teams up with N.Y. guitarist/ programmer Vinnie Zummo, producer of the award winning New York Cutz & New York Cutz Volume 2 - Off Da Hook, and Zero-G to bring you a 3-CD set of the maddest, phattest 'rawfunkindirtyass-boogiebeats' in the universe, including 2 audio CDs and a CD-ROM disc with WAV (Acid ready) files of every sample, cut and ready to burn in your software and sampler.

High-fi, lo-fi, & mid-fi drums, wa drums, sound efx, long loops, short loops, composite tracks, Psychedelia, Funk, R&B, Hip Hop, Jazz, Trance, Trip Hop, Dance, Metal & general madness in one slammin' package to bring the madd edgy taste of New York drumming to your next project.

Whether playing with Shawn Colvin on her two time Grammy Award winning song "Sunny Came Home", David Byrne, Bruce Springsteen or Elton John, Shawn always brings his solid 'funk thang' to a song. Add his bodacious groove to your next track. Taste the flavah.

AUD/WAV **62.95**
Giga **179.95**

Beneath the Planet of the Breaks



More down right dirty beats from the funkster Shawn Lee! Beneath The Planet Of The Breaks is Zero-G's continuing journey to bring you the toughest raw breaks on the planet. As the sequel to Planet Of The Breaks, which swept the board with top awards from every major music magazine, Beneath The Planet Of The Breaks is even better! Not only do you get the baddest loops imaginable, but you also get every loop cut and diced in WAV fromat ready for use in every leading music sequencer package, direct from the CD box! The funk master has returned! Dual format: AUDIO/WAV (Mixed Mode)
I concluded that the beat junky monkeys amongst you wanted more raw breaks(i.e. without additional programming and processing etc.). With that in mind, I've delivered a higher ratio of raw breaks to processed ones. Restriking the balance, if you will. You'll also notice a few more uptempo breaks and more flavas than you can shake a proverbial drum stick at. Two different formats of loops: there are beats that play for approximatey two minutes (these have additonal sonic treatments and so on), and shorter sixteen-bar ones.

AUD/WAV **89.95**
Giga **179.95**

Public Enemy



Welcome to the Sampledome

In 1987, PUBLIC ENEMY redefined rap music and Hip-Hop culture with their explosive debut album, "Yo! Bum Rush The Show". Their subsequent six albums, released over the better part of the next decade, sold millions of copies worldwide. The New York Times named Public Enemy's "It Takes A Nation of Millions To Hold Us Back" to their list of "25 Most Significant Albums of the Last Century", and Entertainment Weekly listed their "Fear Of A Black Planet" album to be one of the most important records of the '90s.
Now, with the release of WELCOME TO THE SAMPLEDOME, a "beats and loops" collection, PUBLIC ENEMY have become the first major Hip-Hop act to release such a project.

The 2- disc set features over two hours of brand new "beats and loops", presented in a "construction kit" format [fully mixed loops and all of the individual samples used to make up the entire loop]. Users can "customize" each loop by using as much or as little of the original.

Audio **89.95**

Return to Planet of the Breaks



Returning to the 'raw-funkindirtyassboogiebeats' that have made the ZERO-G Planet Of The Breaks series such a phenomenon, Emre Ramazanoglu delivers a huge selection of fresh and raw grooving breakbeats. Edgy and tough, small, large, fast, fat, lo-fi, hi-fi, mid-fi, broken, smashed, cuts, drops and extreme funk with quintessential quirk!

Audio/WAV (ACIDized)/ReFill.....**62.95**
Giga **179.95**
Pro Tools **224.95**

DRUMS

Drumkit From Hell



Features Tomas Haake from Meshuggah on Ludwig and Sonor kits. CD-1 contains samples, and CD-2, "Loops From Hell", contains programmed loops.
The sound of the drums is left gratifyingly natural with a degree of ring and ambience following each hit. This makes it easy to treat the sampler output as a set of conventionally recorded drum tracks for the application of gates or EQ. Each drum is multisampled with both left and right hand grip, from soft to hard hits. For example, the snare consists of 10 different hits from soft to hard with both left and right hand grip. In addition to that; 10 hard left hand hits, all with a slightly different approach plus a variety of useful hits such as rolls and sidestick.

All cymbals are also multisampled with different types of hits. Each sample has two different stereo samples, one is the drum itself (with all the other mics turned on), and the other one is the room/ambient mics. With this feature you're able to process the closely miked sounds and the ambient sounds seperately and create a wide variety of general drumsounds. Imagine the possibilities that'll bring to you!!

Multi-format (WAV, LM-4, EXS24, Giga, DR-008, HALion)..... **89.95**

Aerosmith's Joey Kramer Drum Loops



If you are looking for the absolute best pop/rock drumloop collection on the planet - look no further!

JOEY KRAMER "the engine" behind the legendary rock group "Aerosmith," has put together the most comprehensive collection of "Pump & Thump" drum loops and samples available anywhere. Never before has the percussive thunder of a Rock & Roll Icon been offered for use to the general public. Meticulously recorded and painstakingly assembled to offer you the best collection available anywhere of truly usable rock solid drumming. This collection contains no fat! Every loop and every sample are the stuff of rock at it's best. So strap yourself in and collaborate with a master of this genre!

This 2 Audio CD collection features all of the loops and samples mixed both "Dry" on CD-1, and "Wet" on CD-2. Both versions were recorded and mixed by Richard Chycki and produced by Marti Frederiksen.

Audio **89.95**
AKAI (2-CD) **179.95**
Multi Track ProTools or WAV (10-CDs) **299.95**

Skunkworks



SKUNKWORKS features rhythm guitar grooves and solos from legendary Steely Dan guitarist JEFF "SKUNK" BAXTER. Jeff is the recipient of many industry awards, including twenty six gold and platinum records, two Grammys (plus six more Grammy nominations!), and is widely regarded as one of the greatest rhythm guitarists of all time.

The sounds in this collection, as you would expect, are killer! After all, Jeff has been a design consultant for Roland, Guild, Vox, and Gibson - and has his own signature series acoustic guitars. His playing on this collection is impeccable, including super tight rhythm guitar licks that have never been offered before in any sample collection. But that's not all! There are guitar licks for all musical applications from acoustic to heavy metal, and everything in between. An additional BONUS disc of .WAV files is included with the package. How often have you been able to musically collaborate with one of the greatest guitarists of all times? Well, now you can!

Audio/WAV.....**44.95**
AKAI S1000 .....**89.95**

Steve Smith Rhythmic Journey



Features drum loops and individual drum samples from ex-Journey drummer STEVE SMITH, recorded by Aerosmith engineer Kevin Elson at Fantasy Studios in San Francisco. Steve Smith is one of the world's most revered drummers. With the group Journey, Steve established himself as a tour-de-force groove master. RHYTHMIC JOURNEY was co-produced by East West producer Doug Rogers and David Frangioni, who collectively have produced many top-rated sample collections.

Needless to say, the grooves on RHYTHMIC JOURNEY are flawless and sound phenominal - pre mixed by Rogers, Frangioni and Elson. There are also individual sounds for drum replacement or those that want to construct their own grooves using Steve's sounds. The combined experience of Rogers, Frangioni, Elson and Smith has produced one of the best sounding and best performed drum loop CDs of all time. Power on your sampler and get ready, as the journey has only just begun.

Audio/WAV.....**44.95**
UNITY (DS-1 format).....**89.95**
AKAI S1000 .....**89.95**

B&H PHOTO - VIDEO - PRO AUDIO
EQUIPMENT LEASING AVAILABLE

YELLOW TOOLS

Pure Drums



A totally new concept of acoustic instrument samples. Yellow Tools Pure Drums contains 17 different drumkits, dry natural and compressed, with different microphone positions, 16 different samples per note, left - right hand, effect volumes like flams and rolls. Pure Drums is laid out as General MIDI kits and in chromatic scales for pure live performances. All programs were designed as 16mb and 32mb sets.

Kits include: Rock, Pop, Funk, Soul, Jungle, Hip Hop, Hardrock, Ballad, Jazz, Swing, Reggae, Grunge and Street.

AKAI S1000, EXS-24, GigaSampler **134.95**

Pure Guitars



Pure Guitars gives you the utmost control over all aspects of the recorded instruments, enabling them to sound as if they were played live! Featuring: Nylon Concert, Nylon Spanish, Nylon Vibrato, Jumbo, Ovation Piezo, Steel Pick, Steel Fingered, 12-String, Bass Fingered, Bass Aggressive, Bass Short. The best guitars and basses were used, such as Ovation, Martin, Taylor, Lakewood, Hopf, Lauenhardt and Kobs. This library offers the user authentic, realistic (unlooped), and dynamic guitar performances. Eleven different live acoustic guitars and basses; different FX volumes, chords and harmonics.

AKAI S1000, EXS-24, GigaSampler **134.95**

Celtic



The first choice in Celtic Sample CDs! Zero-G's Celtic features great performances and multisamples on a wide range of instruments including Fiddle, Uilleann Pipes, Mandolin, Accordion, Galician Bag, Bagpipes, Whistles, Flutes, Bodhran, Breton Bombarde, Overton, Banjo, 12 string, 6 string, Marching Snare, Pipe & Snare, Skittish Snare, Celidh Snare, and more. Features four full CDs of material (2 audio & 2 WAV)!
A huge range of solo riffs and phrases performed in authentic styles by leading Celtic musicians, together with a set of multisamples for most instruments. This is an absolute must-have collection for film composers or anyone looking for a true Celtic flavour. These samples will inspire you and prove very effective in setting the mood of a track as their authenticity shines through. Add the captivating essence of the Celtic tradition to your music!

Audio/WAV **89.95**
GigaSampler.....**134.95**

Pure E Basses Vol. 1



Pure e-basses vol. 1 features the "Fender Jazz Bass" and "Fender Precision Bass". The Fender Jazz Bass was recorded in the playing styles fingered, slap and pick muted. The Fender Precision Bass was recorded in the playing styles fingered, pick and pick muted.

Pure E Basses Vol. 2



Features the "Ibanez BTB 5-string", "Warwick Streamer Stage 1 fretless" and "Epiphone Jack Cassidy Signature Model". Recorded in fingered, slap and fingered short styles. The Warwick Streamer Stage 1 fretless was recorded fingered. The Epiphone Jack Cassidy Signature Model was recorded fingered muted.

Each playing style contains about 2,000 sound samples. All playing techniques like ghost notes, tappings, hammer-ons, 8th notes, slides slow, slides fast, pop ghost notes, pop slides, pop 8th notes etc. are also playable separately with up to 8 velocity splits per note. For the first time all basses were recorded in 2 different scales! This unique feature enables the choice of the string on which the note should be played! All playing techniques give you the option of combining any of the basses. Plus you get pre-fabricated combinations for each playing style with 8 velocity splits per note. More than 100 different samples per note!!! More than 150 FX per bass (harmonics, fret noises, slides, noises, scrapes etc).

AKAI S5000/S6000, EXS-24, GigaSampler **134.95**

AKAI S5000/S6000, EXS-24, GigaSampler **134.95**

POWER FX

Ambiotica



Ambiotica transcends musical genre and expands new music making possibilities for any REASON user with cinematic pads and soundscapes For the NN19, sonic wizardary for the SubTractor and creative drum programming for the Dr. Rex make this a MUST have sound resource tool.

Includes: • 212 SubTractor Patches • 141 Rex2 files • 116 NN19 Patches • 10 ReDrum Kits.

REASON/REX **89.95**

Clubotica



The ultimate contemporary dance resource for REASON users, this specially created collection is ideal for making the hippest dance tracks or as the ultimate dance remix toolbox. From bombastic beats and basslines to potent pads, patches and FX loops, everything you need to make guaranteed fillers!

Includes: • 212 SubTractor Patches • 141 Rex2 files • 116 NN19 Patches • 10 ReDrum Kits

REASON/REX **89.95**

Retronica



Combine vintage keyboard sounds and killer 70's loops with futuristic electronica elements and techniques. Organize it all for the hottest piece of music software and you get RETRONICA, the new sample CD ROM for Reason. You have the perfect tools to make the best retro/electro/funk/jazz music imaginable!

Features: • 2 Dumloops • Refill Electric Piano loops • Refill Vocoder loops • Upright Bass Refill loops • Redrum kits • Subtractor patches ...and more!

REASON/REX **89.95**

Synthotica



A diverse collection developed by sound designer Magnus Lindberg. Contains rich, beautiful and unique aural textures that will find a home anywhere you need lush synth and sampler sounds – plus a complimentary variety of synth rex loops and programmable drum patches. A whole new world of useful and controllable sounds.

Contains: • 51 SubTractor synth patches • 13 NN19 sampler patches • 300 Dr. Rex files • 8 ReDrum patches

REASON/REX **89.95**

The Electro Jazz Retro Funk Thing



Take a step back to an innovative period in music history with The Electro Jazz Retro Funk Thing. This CD contains a wide range of vintage instruments (such as Wurlitzer & Rhodes pianos) as well as a plethora of slammin' drum loops, colorful effects loops and much more. In addition to offering convenient construction kits, there's a huge variety of tasty Wurlitzer & Rhodes chords that make it easy to get that classic Jazz sound into your project. Also featured are Clavinet loops, vintage synth string and analog pads and drum and cymbal hits to make this CD the ideal! It's a great source of sounds for anyone looking to add authentic Retro or Electro Jazz flavor reminiscent of the sound made famous by the likes of the Headhunters and Herbie Hancock. It's the bomb!

AUDIO **89.95**

The Love & Pop String Thing



A brilliant and unprecedented presentation of Beatles style string quartets, string ensemble pads, loops and 60´s style rock construction kits that are made to fit into all styles of music. Vivid and powerful loops that will instantly send you running to your setup to start kicking out masterpieces or to your wardrobe in search of a flowered shirt.

Wonderfully arranged and beautifully played by the Felix Ochestra in Moscow with string quartets that are presented in construction kits that include the individual cello, viola and violin parts for total user flexibility. Also included: Vintage guitars, basses and drums played by vintage people. It´s the totally cool Love and Pop String Thing.

AUDIO **89.95**

QUANTUM LEAP

'56 Stratocaster



Introducing a next generation sample library from East West and Quantum Leap producer Nick Phoenix on 4 CD-ROMs that captures the pure tone of a '56 strat played through a Super Twin Reverb amp. 3000 chromatic samples were taken using multiple pick-up settings and multiple picking styles at 4 different dynamics. This innovative library features groundbreaking new sampling methods. Features include: realistic chord strumming and chugging, pitchless strum and chugging patches, attitude and pick angle control, true legato notes for flowing leads, gracenotes, slides, harmonics, mutes, pops, vibrato, non vibrato, power chords and FX. The biggest Sampled Instrument ever!! 2.5 Gigabytes!! Please note: This product is HALion PC compatible- will not work with MAC version.

GIGA **269.95**

BRASS



The Definitive 5 CD-ROM Collection of Multi sampled Brass Instruments & Sections, featuring Solo Trumpet, Solo trombone, Bass Trombone, Piccolo Trumpet, Tuba, Baritone, Sax, Tenor Sax, Alto Sax, Soprano Sax, 4 French Horns, 3 Trumpets, and 3 Trombones, meticulously recorded and programmed by Nick Phoenix, who produced the critically acclaimed sample library Quantum Leap Guitar & Bass and Rare Instruments.
The ultimate solution for ORCHESTRAL, POP, JAZZ, BIG BAND, ROCK, MARIACHI, LATIN, R&B, HIP HOP productions. The brass instruments were chromatically recorded through RCA 44 ribbons, Neumann M 49 condensers, Neve and Manley preamps and Apogee Rossetta 24 bit converters in Los Angeles best recording studios.

AKAI S1000, AKAI S5000, EMU EOS, Giga, Kurzweil, Roland and Unity **395.00**

Rare Instruments



Features 16 groundbreaking, ethnic, multi-sampled instruments. Rare Instruments is the most expressive ethnic library ever created, including up to 54 variations for some instruments on 3 CD-ROMs. In some cases the natural range of the original instrument has been expanded, thereby creating instruments that are more playable than the original!! Meticulously recorded in studios, not in some village in Africa on a home recorder; consequently the recording quality and standard of programming is superior to all other ethnic collections. Leap Rare Instruments is the Rolls Royce of ethnic collections!

AKAI S1000, GIGA **445.00**

Voices of the Apocalypse



A breakthrough sample library from East West and Quantum Leap producer Nick Phoenix on 3 CD-ROMs. Voices of the Apocalypse - Classical Choir. Every vowel and consonant soundssampled at multiple dynamics, programmed in a way that allows you to create realistic words, sung in any language. Sounds nothing like any other choir library you've ever heard, because the producer fearlessly went after the full spectrum of a Wagneresque choir. He chromatically sampled a 3 octave range of both men and women in a fabulous sounding hall. Everything from Angelic to downright evil!

GIGA **449.95**

Guitar & Bass



Nick Phoenix is a Los Angeles based session musician. Nick recorded and programmed this huge 600Mb collection himself because he couldn't find any other guitar sample library that enabled him to express himself on a keyboard as a guitarist would on his guitar - the result is a stunning, brilliantly programmed collection that is so realistic you would never know the sounds were coming from a keyboard with your eyes closed. Quantum Leap Guitar & Bass is a revolutionary guitar & bass library comprised of incredibly expressive lead & rhythm patches. This CD-ROM contains the most comprehensive collection of guitar samples ever assembled, as well as nine unbelievable bass patches. Up to six samples are assigned to each note! Grace notes, bends, trills, pops, chugs, slides etc. are all included within each patch making the library very easy to use. Styles include: Blues, Rockabilly, Surf, Pop, Jazz, Steel & Nylon String Acoustics, Psychedelic, Rock, Grunge, Funk, Fretless, Electric Sitar & Hawaiin. Ingenious Programming, SuperB Sound Quality! - There's nothing else like it!

AKAI S1000, Roland, Unity **224.95**

GIGA **269.95**

UEBERSCHALL

Drum n Bass Fast Forward



The next level of Drum n Bass. This package contains more than 20 fresh, new, diverse construction kits that will satisfy every wish - from hard electronic cuts to funky disco kits to jazzy club tracks. Includes additional material-drum loops, bass licks, FX loops & single sounds. This CD is a must for every drum n bass producer!!

Audio/WAV **89.95**

Brazil Electro



TA unique collection of atmospheric modern brazil, bossa styles and samples merged with electronic grooves, dance elements, FX and lo-fi extravaganza. A new generation of loungy easy-listening club music, created to give your production a relaxed hot and jazzy sensation which brazilian music represents best. Live percussion, guitar and key loops.

Audio/WAV **89.95**

da nu RnB Hip Hop



The slammin' second 2 CD set in the series of phat RnB, Hip Hop samples. Da nu RnB Hip Hop Part 2 is packed full of powerful extended song arrangements. Construction kit format, dividing the mix, drum loops, backiing, instruments & phrases and single drum sounds. Extended drum arrangments lasting up to a minute!

Audio (2-CDs) **89.95**
ACID **44.95**

Disco De Luxe



Boogie fever is back and better than ever. A 2 CD Set of construction kits, drum loops, strings, brass, flutes, guitars, basses, vox and single sounds - all to make your music feel like an authentic 70's disco release. Invent your own remix. Samples were inspired by the often sampled disco vinyl records such as Chic, Bee Gees, Donna Summer, ABBA... Kylie Minogue, Modjo and Jamiroquai.

Audio (2-CDs) **89.95**

House Essentials



Ueberschall presents House Essentials, a new masterpiece for House music production containing more than 2,500 fresh samples. Drumloops, drum sounds, bass sounds, funky licks, brass & string sounds, vocals & effects. Doug Laurent and top remixer Risque opened up their exclusive archives - the essential standard for any house and dance producer!

Audio/WAV (2-CDs) **89.95**

Nu Metal



Nu Metal features deep, dark and atmospheric killer material: powerful, dynamic construction kits up to 1 minute in length (A plus B song parts laid out in single blocks), extended live drum loops and breaks, vocals, deeper 7 string basses and guitar riffs (plus breaks), guitar FX. For professional monster productions in the style of Korn, SoulFly, Linkin Park and Papa Roach.

Audio/WAV (2-CDs) **89.95**

Techno Trance Essentials



The standard for techno and dance production. Over 2,595 excellent new sounds (505 drumloops, 540 drum sounds, 210 basses, 490 techno sounds, 620 vocals & 230 effects). You won't find a better collection of techno material around! Produced by Doug Laurent (Honey m, Culture Beat, Front 242, La Bouche, Sash and Yello, to name just a few).

Audio/WAV (2-CDs) **89.95**

Cuckooland Unhinged



This incredible selection of rhythms, textures and FX will have you making music in very new ways. Grooves from 67 to 170 bpm provide rhythmical backing unlike anything you have heard before. Textures and FX are not of this world. This collection is ideal for all - from Trip Hop to Film - new styles of music will definitely evolve from this great CD!

Multi-Format **89.95**

UEBERSCHALL

2-Step Garage



2 Step is one of the most innovative styles in the world. 2 Step unites elements of soulful American House, Dub, Drum&Bass, RnB, Reggae and Breakbeat. This double audio CD contains 30 dance construction kits, more than 130 minutes of content. Features drum arrangements of up to one minute, backings, instrumental loops, vocals, complete mixes, for film and commercials - all 100% copyright cleared and license free. A must for all dance producers!!

Audio.....**89.95**

8-bit Stylez



Low tech sounds for high level music!
Fat! Raw! Incredible! C 64 material remastered for all ya dance projects: Nu RnB, Hiphop, House, Drum & Bass, 2 Step, Techno, Electro, Chill'n tunes, Nu jazz, commercials, jingles, www... 40 Power Construction Kits, wicked drumloops, sharp licks, game music & fx shots, machine vox... plus over 1000 single drum and instrument sounds. All from the legendary SID-chip "MOS 6581".

Audio (2-CDs).....**89.95**

Drum'n'Bass Resonance



The ultimate in construction kits. Your house is gonna quake. This CD contains really cool and tricky drum loops, intense sub-basses and much more. The loops are broken down with the complete mix, then the drum loop, then all of the individual sounds and sub-basses. You get 60 Drum'n'Bass mixes with over 850 drumloops and 750 single sounds. Hot and flexible material. Arranged in order of 1) complete mix, 2) drumloops, 3) single sounds.

Audio.....**89.95**

Houseworx!



Europe's No.1 star remixer and producer Mousse T. (remixed for Quincy Jones, Michael Jackson, Backstreet Boys, and more) has put together the treasures of his work on HouseWorx! Included are drum and percussion loops, vocals, guitars, rhodes, organ and piano licks, FX vocal loops, basses and bass licks, vibes, hits, breaks, synthesizer, sound FX, single instruments and drum and percussion samples.

ACID.....**44.95**
Audio.....**89.95**
AKAI S1000, EXS-24.....**179.95**

Rhodesattacks



Rhodesattacks features the legendary Rhodes Mark 1 Seventy Three - the most sought after electric keyboard of all time! Tons of Licks, some in several versions: Original with wah-wah and as effects. Various playing styles: Groovy, mellow, wacky.

For Drum&Bass, House, Funk, R'n'B, TripHop, Jazz and many other styles. In various tempos. Some BONUS MIDI files are included.

Audio.....**89.95**

Special Sound Effects



Produced with the newest 20-bit super bit-mapping recording technology. Soft atmospheres, Natural FX, Psycho, Horror, Galaxy, Work, Comedy, Computer Sounds & Long Ambience as well as countless so-called Zap FX (short switching sounds), perfectly suited for radio spots, ads and animation. Each recording is indexed and listed alphabetically as well as track by track. Besides Dummy Head and RSS, you get pseudo-4-channel recordings.

Audio.....**Call**

ZERO-G

Jungle Warfare for REASON



Over 1600 DrREX Loops, Over 1000 patches/samples for NN19/NNXT. Over 1.6 Gb of data (raw). The whole Jungle Warfare Series expertly converted to ReFill format, in a single multi-disc set. This library was awarded 5 stars (SOS) and a Key Buy Award (Keyboard USA) - and it's still the world's best full-blown Drum & Bass sample collection. Hardcore guerilla dancefloor ammo for speed freaks. A massive collection of ultra-fast techno & funk grooves, construction kits & samples - lethal dance production weapons in the right hands. If you need the RIGHT material to produce Drum & Bass this blinding collection has to be on your unmissable list.

Includes - Special Drum & Bass loops, Massive drum rolls & mega rhythm fills, Jungle percussion loops and kits of component samples, Intense Jungle pads & chords, Aggressive digital synth pads, Jungle chords and hits, In-yer-face Bass notes, Manic Jungle FX, Nasty vocals. Contains all three original Jungle Warfare releases.

Reason **116.95**

Chemical Beats



'Chemical' beats are loops with masses of attitude, huge drum sounds and FX crunched through advanced signal processors, and massive grooves to kick start your imagination. Years of playing, programming and engineering expertise at your fingertips! To complete this awesome package, there are extra mad percussion loops, thunderous basses, radio loops, cutting edge synth & guitar FX, plus a vast collection of drum & cymbal bits.

All the samples are contained within a single REFILL file, and also provided again on the same disc as separate and individual REX2, NN19, ReDrum Kits and ACIDized WAV files.

Can be purchased in other formats: Chemical Beats Audio CD; Chemical Beats CD-ROM for AKAI S3000 series samplers (includes the Audio CD version free in the package); Chemical Beats GIGA or choose the smaller ProSamples version Volume 7 Future Beats 1 which is in multiple native formats and compatible with all samplers.

Reason .....**89.95** **Akai**.....**179.95**
Audio .....**62.95** **Giga**.....**134.95**

Ambient



Features the combined contents, in full, of both of the original AMBIENT audio sample CD libraries (AMBIENT Vol. 1 and AMBIENT Vol.2).

Over 470 Mb of stunning sounds produced by highly acclaimed British electronic musician, Ian Boddy. An array of dreamy NN19 atmospheric and FX patches for use in ambient dance, trance, film soundtracks or futuristic/spacey/trance music.

Combining the hypnotic charms of Jarre, Floyd, Tangerine Dream and so on with current dance trends, "Ambient" is designed to mellow you out with relaxing washes, laid back rhythms and awesome sound effects.

An impressive inventory of analogue and digital synths were used to create these well organised stereo samples. There are digital drones, pads and effects (all very lush stereo) conjuring up deep space hyperdrives through sinister alien landscapes, etc.

IMPORTANT: "Ambient" can ALSO be used with any software that imports WAV data.

Reason/Audio..... **89.95**

Ambient Textures REASON



Features 1000 brand new patches created in Reason 2.0, including 300 Subtractor patches, 201 Malstrom patches, 259 NN-19 Sampler patches, and 240 NN-XT Sampler patches.

Synthesizers used: Microwave XT, Technics WSA1, Supernova 2 Pro X, Micro Q, Alesis QS

The Producer, Jonathan Heslop, has been creating electronic music for the past eight years. He has composed music for various independent films, computer games and even some patches for the Reason 2 Soundbank. During that time he has created a large number of synth patches, the best of which are on this superb Refill CD.

(Notes: The Malstrom and NN-XT patches only work with REASON version 2.0 or later. All samples on the disc are contained within the single Refill. This sound library was previously available direct from Jonathan Heslop's website (www.funk-station.co.uk) but the Zero-G version is now the only one available).

IMPORTANT: The Rex Files contained in this collection can only be accessed using Reason.

REASON **62.95**

ZERO-G

Trance Construction Kits



This exclusive new REFILL for Propellerheads REASON was produced by Mike Ash, the producer of 'Nu House', and is essential raw material for tranceheads worldwide. From epic soaring melodies to pounding machine rhythms, Zero-G TRANCE CONSTRUCTION KITS delivers up-to-the-minute sounds, including plenty of Dr REX Drumloops, Dr REX Grooveloops, ReDrum Kits, NN19 Patches, etc. It features tons of pumping beats from 132 to 146 bpm, ambient trippiness, bubbling synth lines, percussion, FX, and bass patches.

Also included are lots of patches of trance synths, pads and strings - in fact just about all the mind-blowing sounds, textures and sonic tools you need for use in Ambient, Epic or Hard Trance, Techno and "perfect-for-three-in-the-morning"" dance production. You'd be hard pressed to find a better collection of trance material around at this price. If trance is on your flight plan, this is a fine place to take off.

CD-ROM Reason/Rex **62.95**

Wired - The Elements of Trance



The Reason Refill version of the forthcoming Zero-G plug-in Virtual Sound Module WIRED - THE ELEMENTS OF TRANCE, featuring over 1.2 Gigabytes of samples (raw).

This massive and exclusive sound archive from Nigel Anderson (the producer of "TOTAL TRANCE") provides every constituent you need to make the perfect trance track in Propellerheads' REASON software studio. Includes just about every 4/4 percussion combination that you can imagine. Great acid lines; energy and vibe; easy to edit. The pads, pad loops, synth lines and bass loops will help you muster your last ounce of euphoria. Loads of effect stabs and snare rolls. Huge selection of vital drumkit sounds.

Drum & percussion Loops, Snare Rolls, Acid Loops, Ambient Intro Loops, Bass Loops, FX Loops, Guitar Loops, Piano Loops, Trancy Synth Loops, Tribal Trance Loops, Acid Shots, Basses, FX, Piano, Stabs & Jabs, Sweeps, Groove Loops, Pads, Redrum Kits & Samples, 808 Kit, 909 Kit, House Kits, Tec-Trance Kits, and Percussion Hits.

CD-ROM Reason **116.95**

Creative Essentials



Zero-G's award winning Creative Essentials Series of 31 sample CD volumes have now been expertly converted into ReFill format to take full advantage of the incredible power and flexibility of REASON. This highly acclaimed library is surely the largest single title available in ReFill format today - over 7 Gigabytes of quality inspiration compressed into just 3 CD-ROMs. All 31 collections are included:

•Rhythm, Guitar, FX •Percussion Grooves •Brass Elements •Dance Vocals •Electric Dreamz •Trance Formation •Dream Zone •Spices of India •Guitar Separates •Global SFX •House & Garage Construction •HipHop & Swing Breakdown •Junglw Frenzy •Live Bass Grooves •World Class Breaks •Drum & Percussion Tools •Bass Separates •Rock & Pop Vocals •Dance Drums •Upfront Lead Guitar •Ethnic Flavours •Reggae Connection •Techno Prisoners •Funk Construction • Synth Bass Loops •Vintage Keyboards •Voices of Africa •String Textures •Woodwind & Brass •Orchestral Flavours •Jungle Frenzy 2

Reason..... **134.95**

Jungle Frenzy



All the elements you need for creating tracks in all areas of Jungle production! Breaks run at 160bpm to make cutting up and mixing easy. Features drum rolls & fills, percussion loops & hits, basses, pads and jungle MC-ing, and the latest stabs, pads, noizes and FX - flavours from every tree in the Jungle - Intelligent, rough, hard, steppin' breaks & FX for todays Drum'N'Bass samplists. If you need Jungle samples this is unmissable - it gives you stacks of kicking grooves, a wide range of styles and it's as UP-FRONT as you can get. WAV, AIFF and Red book Audio in MFS format for MAC/PC.

AU/WAV/AIFF..... **26.95**

DANCE MEGA

Dance Mega House



2 CD set includes Audio and a 500MB mixed mode AKAI/EMU CD-ROM for the price of an Audio CD!
Dance Mega House/Dance contains 1300 brand new stereo drum loops plus 350 modern house and dance sounds, synths, house organs, chords, pads, basses, multi-samples and drums. 120-140 bpm.

AUD/AK/EMU..... **89.95**

Dance Mega Jungle Rave



2 CD set includes Audio and a 500MB mixed mode AKAI/EMU CD-ROM for the price of an Audio CD!
Jungle/Rave contains 1200 stereo drum loops plus 250 jungle/drum'n'bass and rave sounds including synths, chords, basses, multisamples, jungle pads and atmospheres.

AUD/AK/EMU..... **89.95**

Dance Mega Synth



2 CD set includes Audio and a 500MB mixed mode AKAI/EMU CD-ROM for the price of an Audio CD!
Dance Mega House/Dance contains 1300 brand new stereo drum loops plus 350 modern house and dance sounds, synths, house organs, chords, pads, basses, multi-samples and drums. 120-140 bpm.

AUD/AK/EMU..... **89.95**

Dance Mega Disco House



1,300 Drum Loops 120-140 BPM and 300 Disco House Sounds! Various synthesizers, hits, basses, funky guitar licks, strings and brass hits, classic disco chords. Volumes and programs have a maximum of 8 megs, filter and envelope parameters are already set. The grooviest classic 70's Disco and funky House sampler collection.

AUD/AK/EMU..... **89.95**

Dance Mega Drum Kits



Dance Mega Drum Kits is the drum CD that no collection should be without. You won't believe how much material the producers have put on this 2-CD Set! Over 5,000 awesome samples for you to use license free in your music production! All samples are laid out in keygroups and programs – ready to be played!

AUD/AKAI..... **89.95**

Dance Mega Hip Hop 2



Guitars and licks, synthesizers, hits, brass, pianos, Rhodes, basses and many other chords and samples. All drum loops are perfectly timed and sorted into bpm groups.The ultimate Hip Hop, Rap, RnB, Funk, Soul and Pop sampler collection. This CD is mixed mode! You get AUDIO and WAV formats, all in one package!

AUD/WAV..... **89.95**

Dance Mega Drums



The most extensive drum collections for all types of Dance Production. 90 complete (but different) drum machines and rare analog rhythm boxes with many sound variations. 1500 of the latest drum sounds (Bass Drums, Snare Drums, Hi-Hats, Percussion, Analog Drums and Percussion) and a 400 MB CD-ROM section. All of the drum machines are laid out in keygroups, ready to be played.

AUD/AK/EMU..... **89.95**

Dance Mega Drums 2



The sequel to the world's greatest drum sample collection. Includes over 4000 different drum, percussion and new drum machine samples. Hundreds of ready to play drumkits for modern dance-music styles: Hip Hop, real Drums, Disco, House, Drum'n'Bass. All samples are sorted and categorized. Laid out in key-groups, programs, presets or volumes - ready to be played!!!

AUD/AK/EMU..... **89.95**

PURE



Pure Hip Hop

Nu school Hip Hop from ZERO-G. Fusing the old with the new, Pure Hip Hop brings you a raw slice of gritty beats, the fattest basses and toughest street vibes to build a Hip Hop masterpiece. Construction kits and loads of extra beats & samples make Pure Hip Hop a deep resource for the Hip Hop connoisseur. This multi-format sample collection gives you the ultimate in sampler and software compatibility – Acid ready WAV files, AIFF files, Emagic EXS24 sampler instruments, AKAI (and compatible) sampler instruments (compatible with Steinberg HALion), Steinberg REX2 Recycled files and of course Audio! BPMs and keys are given to make your life so easy. Pure Hip Hop is a must for all your Hip Hop producers who want to add some new beats to their tracks!

Multi-Format **44.95**



Pure Trip Hop

It's difficult to define trip hop, and most trip hop artists deny they are part of any scene at all. Often described as “instrumental hip hop”, emphasis is on the music itself but every DJ producer has their own way of keeping a track alive so there can never be a distinctive trip hop sound. It takes ideas from Jazz, Blues, Hip Hop, Dub, Ambient, even Classical music, but it's only through the sampler that trip hop happened at all. It vastly expanded the way music can be created. You can play with raw sound and experiment with new found frequencies, tempos and tones to make previously unimaginable music.

Thirty musical pieces divided into their separate elements, plus extra helpings of beat loop samples, bass lines, EP chords and drum hits.

Multi-Format **44.95**

Pure R’n’B



Extensive toolkit for assembling polished RnB including 35 song ready musical pieces divided into their separate elements, plus extra helpings of beat loops, bass notes and EP chords. These quality building blocks are spread over a 2 disc Audio/ CD-Rom which includes Audio/AKAI/ REX/ WAV and are also “Acid ready”!

Features classic sounding soul breaks and exclusive swing and soul loops and samples. These are grooves from the heart of soul and swingbeat. Includes stacks of R’n’B construction kits (grooves plus all their component samples) plus extra drumbreaks, basses, Rhodes piano and organs.

Add these pure R’n’B elements into your mix to give your music a refined edge and put a cool and effortless vibe into your grooves.

Multi-Format.....**44.95**

Pure Mayhem



Pure Mayhem from ZERO-G brings electronic warfare to your sampler. Mangled beats, fuzz arps, technoid FX, PWM, noize and more noize. This is one Sample CD package that brings you mayhem in it's purest form, Pure Mayhem. Pure techno aggression. Pure Mayhem delivers killer Drumloops, Sequences, FX Loops, Guitar loops, Ambient Loops and ..Bass Loops. Everything you'll need to produce your next hit record.

Gives you the ultimate in sampler and software compatibility Acid-ready WAV files, AIFF files, Emagic EXS24 sampler instruments, AKAI (and compatible) sampler instruments (e.g. compatible with Steinberg HALion), Steinberg REX2 Recycled files and Audio!

Multi-Format **44.95**



Pure Brazilian Beats

Enjoy the unmistakable flavor of Latin America in the 21st Century - the feel, the groove, the life. Recorded in Brazil by Duda Moura, Pure Brazilian Beats is an incredible collection of completely authentic Brazilian rhythms with an infectious feel, suitable for anyone with a desire to add rhythmic life to their productions. Features acoustic drum groove/loops including: samba, baião, axé, bossa nova, ijexá, samba reggae, samba rock, batucada, samba do Ilê, pagode, samba enrêdo, samba funk and other Brazilian fusion experiences.

Two Multi-format Sample CDs are included, giving you the ultimate in sampler and software compatibility.

Multi-Format **44.95**



Pure Tabla

An unbelievable range of sonic beats created entirely with Tablas! ZERO-G bring you extreme remixes of incredible Tabla performances, with rhythmic adventures unlike anything you’ve heard before. Whether it’s dance or pop, Pure Tabla will give your rhythmic tracks a cutting edge inspiration. This multi-format sample collection gives you the ultimate in sampler and software compatibility – Acid ready WAV files, AIFF files, Emagic EXS24 sampler instruments, AKAI (and compatible) sampler instruments (compatible with Steinberg HALion), Steinberg REX2 Recycled files and of course Audio! All BPMs and Keys are given where applicable.

Multi-Format **44.95**

ILIO

VIENNA SYMPHONIC LIBRARY

The Vienna Symphonic Library First Edition



Produced by top composers, played by musicians from Vienna's world-renowned orchestras, recorded in 96kHz/24-bit and perfected by 30 highly specialized editing engineers,the Vienna Symphonic Library is a sample library of unprecedented sonic dimension. Recorded in a specially constructed recording stage designed expressly for the rigorous demands of orchestral sampling, the Vienna Symphonic Library is simply unsurpassed in both size and quality. Complete, the library consists of 1.5 million single-note samples and performance elements. Until now, many of these instruments have never been recorded for a sample library. For example, the bass trumpet, contrabass trombone, and contrabass tuba bring closure to the sampled brass category, while the world's lowest-pitched gong, the deepest tubular bells, and others from the large and rare percussion collection owned by the Vienna Radio Symphony Orchestra are captured as well.

FEATURES

"Silent Stage" Recording

- The samples are totally devoid of background noise (80 – 100 dB signal-to-noise) and have absolute clarity, even in the pianissimo performances.
- The "Silent Stage," as it is called, was designed specifically to record samples of orchestral instruments.
- The recording technology used (stereo throughout with full mono compatibility) makes it possible to produce everything from epic Hollywood film music to chamber music arrangements or solo pieces.
- An entirely new component system consisting of so-called Performance Elements means that playing techniques such as legatos, runs and repetitions, which have had limited implementation in previous orchestral sample libraries, are now completely and intelligently integrated to make possible a realism unobtainable until now.
- Special mappings make it easy to adjust the timing and note lengths within the performance elements and even to manipulate these authentic passages in real-time.
- Because each instrument is mapped the same way across its entire range, the user can find any instrument or sound quickly and dependably, despite the vast number of individual instrument and ensemble samples.

Consists of two Components

The Orchestral Cube

- A massive collection of string, percussion, brass and woodwind samples (over 42,000), totaling 44 GBs, the Orchestral Cube contains single note samples with a variety of dynamics options, as well as a variety of multi-note samples including trills and tremolos.

The Performance Set

- 36,000 samples covering over 45 GBs, the Performance Set lets you play real legato in real-time with multiple dynamics.
- Legato performances contain up to 3 dynamic layers with optional vibrato, and the string samples are available with and without slides. Also features repetition performances, available in legato, portato and staccato with tempos ranging from 60 to 200 BPM and appoggiaturas.

Sample content of the Orchestral Cube, Strings, Brass & Woodwinds, Percussion

Single tones with consistent dynamics

- chromatically sampled
- up to 5 dynamic layers
- up to 3 articulations (especially solo brass): soft, normal, marcato
- without vibrato, soft and strong vibrato, progressive vibrato
- different lengths for short notes: 0.1 sec, 0.3 sec, 0.5 sec
- 2 sampled variations of the same note for each short note
- long notes (1 to 20 seconds) with release trigger

Single tones with progessive dynamics

- 3 dynamic intensities slight, e.g. p<mp, f>mf etc. up to 4 dynamic layers
- medium, e.g. pp<mf, ff>mf 2 dynamic layers
- strong, e.g. pp<ff 1 dynamic layer
- crescendos - diminuendos, e.g. p<mf>p
- fp, sfz, sffz
- lengths from 1 to 6 seconds
- with and without vibrato
- Trills slow and fast variations, accelerations
- Tremolos crescendo and diminuendo patches

SYNCLAVIER • PROPELLER

Synclavier Essential Percussion



"Essential" is the key word here. Impactive drums, hits and cymbals, plus popular world percussion like steel drum, bongos and congas, not to mention electric bass, make this disc crucial to your production. Features hundreds of Kicks, Snares and Toms. 1,000+ samples.

CD-ROM Available for AKAI, Kurzweil, Roland, SampleCell **179.95**

Synclavier - Brass and Winds



A complete set featuring Trumpet, Trombone, Tuba, Flugelhorn, 2 and 4 Trumpet unisons, 2 Trombone unisons, French Horn and Saxes. Plus Clarinet, Oboe, Bass Clarinet, Bassoon, English Horn, Flute and Exotic Winds. Multiple articulations make expressive performances a breeze!

CD-ROM AKAI, SampleCell **239.95**

Synclavier Keys and Guitars



A gorgeous Concert Grand Piano with four velocity steps — great for classics or rock. Hundreds of synth sounds from such greats as the MicroWave, Sequential VS, MiniMoog, Oberheim and the monster Synclavier itself. Plus a real Rhodes or Wurlitzer Electric Piano, some rockin' Acoustic and Electric Guitar . . .

CD-ROM AKAI, Kurzweil, Roland SampleCell **239.95**

Synclavier World & Orchestral Percussion



Includes all the percussion one could want, including a full orchestral set complete with vibraphone marimba, bass marimba, xylophone, celesta, gongs, chimes, crash cymbals, timpani, bass and snare drums and more! Plus an encyclopedia of world percussion from Anklungs to Wuhan Gongs. 1,000+ samples!

CD-ROM AKAI, Kurzweil, Roland, SampleCell.....**179.95**

Synclavier Strings



Unprecedented warmth and emotion. Solo violin, cello and double bass samples in bowed, pizzicato articulations; cello and viola sections played as sustains and staccatos; violin section: pizzicatos, tremolos, staccatos and lush sustains. Exceptional harp . . . single note and various arpeggios. Bonus: Human voices and vocal effects.

CD-ROM AKAI, Kurzweil, Roland, SampleCell **239.95**

Propeller Island's Complete Gamelan



Sounds from the edge of the world – allowing you to create liquid metallic textures and intricate rhythms which produce a hypnotic effect on the listener.

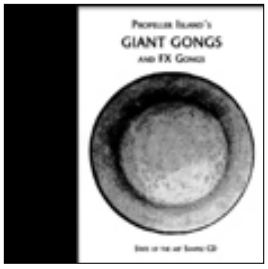
CD-ROM AKAI (Emu, Giga, Software Sampler compatible).....**239.95**

CD-Audio..... **99.95**

Propeller Island's Giant Gongs

Finest quality recordings, made at Propeller Island's studios and the ambient Salvatore Church in Berlin. Pure, unretouched performances; many versions of the gong sounds which have been heavily processed using filters; EQs and wide variety of other-worldly computer enhancements. Rich resource for special ambience!

CD Audio **99.95**



Propeller Island's Legendary M400

A trip back to the 60's and an exploration of the world's first keyboard "sampler." All original sound sets on each key of the instrument.

CD-ROM AKAI (Emu, Giga, Software Sampler compatible) **145.00**

CD-Audio..... **99.95**



SOUNDSCAN



Vol. 01 Hard & Loud Techno

This most original and aggressive sample library is VIOLENT! An overdose of aggressive sounds: hellish patterns, extreme analog sequences full of bite, ultra violent samples, screaming drum & synth loops, tons of massive sound effects. Perfect for all kinds of Techno: Hardcore, Industrial, Big Beat, etc.
448 samples (182 loops & sequences / 111 special FX Licks / 156 single sounds); 56 audio tracks / AKAI partition: 56 volumes / Wav-Acid/Aiff: 382 MB; Tempo: 140 - 180 BPM

USBCDSC01.....**49.95**



Vol. 02 Underground Hip Hop

Created with "underground" in mind. Enormous drum samples (Kick, Snare, Claps, Hi-Hats, FX…), expert and killer beats, MPC loops, city sounds, dark tracks, symphonic impressions, piano phrases, drum fills, breaks, bass, guitar riffs, percussion, various instruments samples… and more. 506 samples (324 expert beats / 127 samples / 55 drums); 50 audio tracks / AKAI partition: 50 volumes / Wav-Acid/Aiff: 509 MB; Tempo: 89 - 97 BPM

USBCDSC02..... **49.95**



Vol. 03 Pure Electro

Exploring the latest trends of dance music. Will kick-start your tracks with a rush of adrenaline. Mouth-watering drums.... classic beatbox tones, dry and processed loops, noisefix section, new versions of the classic sirens to earth-shattering gongs and hits, hypnotic basslines, weird arp sequences, evolving futuristic sound effects... 263 samples (131 drumloops / 21 synths / 19 noise FX / 41 drum sounds); 47 audio tracks / AKAI partition: 47 volumes / Wav-Acid/Aiff: 539 MB; Tempo: 120 - 140 BPM

USBCDSC03.....**49.95**



Vol. 04 Electric & Acoustic Guitar Loops

The most versatile guitar phrases collection. Silky smooth 6 and 12 strings guitar rhythm patterns in many major and minor keys, at different tempi. Numerous variations of Funk and classic wah guitar loops, dream-liquid arpeggios ideal for Folk and Blues, sweet and airy ballad guitars…
319 samples (105 acoustic loops / 214 electric loops); 36 audio tracks / AKAI partition: 36 volumes / Wav-Acid/Aiff: 406 MB; Tempo: 70 - 160 BPM

USBCDSC04..... **49.95**



Vol. 05 Explosive Jungle

Hundreds of jungle loops & breakbeats, most including already trimmed samples of snare, hi-hat and bass drum. Create your own breaks and variations.
No jungle tool would be complete without the distinctive bass and synth sounds: lush, super wide and deep synth strings, minimalist pads, sub-bass, synth bass, sound effects and more.
358 samples (278 drumloops & break / 22 bass / 39 pads / 19 sounds FX); 62 audio tracks / AKAI partition: 54 volumes / Wav-Acid/Aiff: 526 MB; Tempo: 160 - 173 BPM

USBCDSC05.....**49.95**



Vol. 06 Electric & Acoustic Bass

The best bass sounds available today. The acoustic section contains a sensation double basses, finely multi-sampled, with both hard and soft velocities, with or without natural vibrato; multi-samples and hard and soft velocities of Fender, MusicMan and Rickenbacker basses for rock, techno, pop, rap, R 'n B… 698 samples (121 acoustic bass / 387 electric bass / 99 bonus bass / 91 FX notes); 47 audio tracks / Wav-Acid/Aiff : 336 MB / AKAI partition: 30 volumes / 42 multisamples

USBCDSC05.....**49.95**



Vol. 07 Crazy Processed Drumloops

Futuristic loops perfect for producers who want to instill fresh blood into their work: Neurologic, Chaotic, Talkboxed, Rythmabeatz, Dispersonic, Sonic Modular, Vocoded, Fluidity, Alphalayered & Disturbed. A wealth of rare or unknown processing hardware and software — purely rhythmic sound design.
252 samples (84@80 BPM / 84@100 BPM / 84@120 BPM); 34 audio tracks / AKAI partition: 34 volumes / Wav-Acid/Aiff: 490 MB; Tempo: 80 - 100 & 120 BPM

USBCDSC07..... **49.95**



Vol. 08 Jazz & Latin Drum Loops

Filled to the brim with classic jazz, samba, salsa and bossa rhythms that no programming can emulate. Jazz section features essential ride patterns with variations, sensational Jazz "Breakbeats" & brush loops. Latino section - from intimate Bossa and irresistible groove of Zouk to the flourishes of Samba and the rare Songo beats.
188 samples (127 Jazz Drumloops / 61 Latin Drumloops); 34 audio tracks / AKAI partition: 34 volumes / Wav-Acid/Aiff: 525 MB: Tempo: 80 - 240 BPM

USBCDSC08.....**49.95**

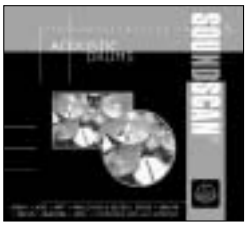


Vol. 09 Old School Keyboards

The finest vintage keyboard samples ever made: electric pianos, Clarinet, organs and other. The electric pianos includes classic "suitcase" instruments and other touring instruments, the Organs section is full of specially made Hammond B3 samples, Farfisa and other classics. Funkiest Wurlitzer and Hohner Clarinet sounds, and much more…
673 samples (280 electric piano / 117 clarinet / 276 organ); 80 audio tracks / AKAI partition: 63 volumes / Wav-Acid/Aiff: 262 MB / 80 multisamples

USBCDSC09.....**49.95**

SOUNDSCAN



Vol. 10 Acoustic Drums

For all musical styles. Many kits that have one thing in common: the sounds were captured without any processing, so that you can apply the right treatment or leave them raw. Processed section features equalized and compressed drums sounds, and additional treatments for up to date sounding kits.
745 samples (496 in drumkits / 249 sorted samples); 58 audio tracks / AKAI partition: 29 volumes / Wav-Acid/Aiff: 583 MB / 18 drumkits (BD, SD, HH, Toms & cymbals)

USBCDSC010 ..... **49.95**



Vol. 11 Dance & Groove Drums

Dozens of kits for many different music styles Dance, Electro, Groove, House, Techno, Trash, Jungle, Disco, Lo Fi, New York & Vinyl… Unique. Samples organized in style-oriented kits, then in separate samples (BD, SD, HH…). Will be instantly useful and should become your favorite modern drums collection.
1895 samples (1132 in drumkits / 763 sorted samples); 91 audio tracks / AKAI partition: 52 volumes / Wav-Acid/Aiff: 137 MB / 130 drumkits (BD, SD, HH…)

UUSBCDSC011 ..... **49.95**



Vol. 12 Analog & Synth Bass

The widest possible selection of classic synthesizers, including analog greats such as MiniMoog, Prophet, Yamaha CS70m, Oscar, Matrix 12, MS 20 and Roland Jupiter 8, Mks 50, SH101 & Juno 106. Digital synths, such as Yamaha DX7, Roland JD800 & 990, Kurzweil K2000, Korg 01/W & Wavestation. Usable and efficient.– 227 samples (120 anal. bass /101 dig. bass); 78 audio tracks / AKAI partition: 39 volumes / Wav-Acid / Aiff: 129 MB / Categories: Funk/Newjack, Disco, Ragga, Sub, Garage, Pop, Acid…

USBCDSC012 ..... **49.95**

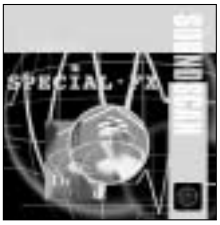
Vol. 13 Ethereal Atmospheres



Writing New Age or Jungle? Scoring to pictures or looking for a great Hip Hop intro? Ethereal Atmospheres offers you tons of original & dreamy textures. Majestic, ultra wide layered pads created using a mixture of tones, breathtaking "chords", ultra spacious sounds directly usable. Bonus: Large selection of ethereal pad sounds from the best analog and digital synths.
177 samples (53 Chords / 44 Stacks / 80 Nice Pads) 46 audio tracks / AKAI partition: 46 volumes / Wav-Acid/Aiff: 503 MB / 46 multisamples set

USBCDSC013 ..... **49.95**

Vol. 14 Twisted Special FX



Uses the latest in sound manipulation software, the finest high-end hardware and the rarest of sound generators. Sounds sensibly grouped into self explanatory categories – quickly locate what you need: Ambient, Mad radio, Sub waves, Electricity, Hits, Meta sounds, Insectronics... Add high quality effects to your productions.
200 samples (150 long / 50 short); 44 audio tracks / AKAI partition: 40 volumes / Wav-Acid/Aiff: 572 MB; Categories: SubWave, Suspense, Ambient Atmospheric, Insectron…

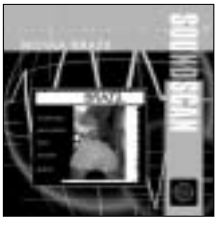
USBCDSC014 ..... **49.95**

Vol. 15 Acoustic Percussions



An outstanding collection of Latin and ethnic percussion, with the highest standard of quality and originality: Anklung, Bell Tree, Bongos, Cajon, Castanet, Chekeres, Chimes, Clave, Congas, Cowbells, Cruche, Cuica, Djembe, Guiro, Maracas, Rainstick, Rattle, Shaker, Step, Tabla, Talking Drums, Tambourines, Timbales, Triangles, Whistle, Wind Chimes, Wood Blocks…
888 samples / 84 audio tracks / AKAI partition: 51 vol. / Wav-Acid/Aiff: 162 MB / 51 Categ: Agogo, Shakers, Congas, Triangle, Wind Chimes.

USBCDSC015 ..... **49.95**



Vol. 16 Bossa Brazil

Straight from Brazil. A superb "bossa" compendium. Drums and percussions, guitar and bass were performed together - sampled separately for maximum flexibility. The drums & percs are divided in sticks, brushes and mallet loops which express well the nuances of bossa performances. The key is given for both the guitar and bass loops.
270 samples (42 drumloops / 228 bass & guitar loops); 66 audio tracks / AKAI partition: 40 volumes / Wav-Acid/Aiff: 503 MB; Tempo : 120 - 140 BPM

USBCDSC016 ..... **49.95**



Vol. 17 Acoustic Keyboards

A full-fledged Piano in the SOUNDSCAN range! A Steinway D and a Yamaha: both available in several memory requirements, from a glorious 32 MB version down to the most exciting 8 MB piano patches you've ever played. A HonkyTonk piano is also included. Bonus: two Harpsichord (French & Flemish) sampled..
592 samples (114 Jazz Piano / 96 Grand Piano / 98 Honky Tonk Piano / 284 Harpsichord) – 14 audio tracks / Wav-Acid/Aiff: 380 MB / AKAI partition: 14 volumes

USBCDSC017 ..... **49.95**



Vol. 18 Funky Vocals

An excellent selection of dry and processed vocals: melodies, phrases and soulful wails; hits and shouting… a broad range of emotions. From flanging effect, to AutoTune, Vocoders, square wave tremolo and other creative tools. Bonus: the biggest and funkiest collection of Talk Box samples "a la" Roger ever done in a sampling product.
647 samples (188 natural / 288 Talk Box / 171 processed); 42 audio tracks / AKAI partition : 30 vol. / Wav-Acid/Aiff: 482 MB; Categories: Ad Lib, Hooks, Ragga, Talk Box, Computer, Autotuned.

USBCDSC018 ..... **49.95**

SOUNDSCAN



Vol. 19 Hip Hop & R 'n B

You think you've heard it all? Think again! Enormous kick, dynamic snares, and a simplicity in programming -the best. Everything delivered in construction kits, which include silky sounds, nasty tones and dark effects...
(This volume is also available in MPC2000 format)
515 samples (123 Timba /46 Teddy / 143 R 'n B / 112 Hip Hop FX/ 91 Hip Hop Melodic)
72 audio tracks / AKAI partition: 36 vol. / Wav-Acid/Aiff: 563 MB; Tempo : 67 - 106 BPM

USBCDSC019 ..... **49.95**



Vol. 20 Fresh Disco House Vol. 1: Drumloops

Dance floor lovers: This Disc kicks off with drum samples, both dry and processed. Samples are brand new – also included: classic TR tones for your convenience. Loops – with a strong Disco House or Latin flavor. For maximum flexibility you'll find variations, separate loops broken down from the full mix, and the individual sounds.
937 samples (350 drumloops & kits / 587 single hits); 73 audio tracks / AKAI partition: 38 volumes / Wav-Acid/Aiff: 426 MB; Tempo: 125 - 129 BPM

USBCDSC020 ..... **49.95**



Vol. 21 Fresh Disco House Vol. 2: Instruments

Basses, guitars, electric piano, clarinet and more. Disco House needs. Breathless variety of sounds: low sub-bass, hits and stabs, suitcase piano, clarinet loops and riffs, funky guitar licks, mouth-tube, octave, vintage wah, bass section, groovy loops, licks, slap, Latin, analog bass, brass riffs, synth hit, soaring leads, pads, piano chords …
704 samples (182 clav. / 217 guitar & bass / 305 misc.); 41 audio tracks / AKAI partition: 41 volumes / Wav-Acid/Aiff: 402 MB

USBCDSC021 ..... **49.95**



Vol. 22 Vintage Guitar Loops

Authentic riffs and patterns using the best guitars and amps going through a vintage trident desk, with no effects or digital processing – to get the warmest guitar sound. Includes blues, rock and classic uses of mildly distorted tube-driven guitar amps. Minor and major patterns, solos, strums and arpeggios...
602 samples (314 Riffs & co / 217 Licks / 71 single notes); 45 audio tracks / AKAI partition: 45 volumes / Wav-Acid/Aiff : 431 MB; Tempo: 63 - 150 BPM

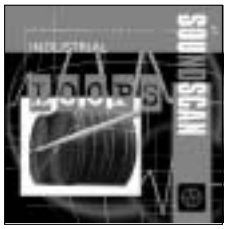
USBCDSC022 ..... **49.95**



Vol. 23 Historical Instruments

A groundbreaking collection of rare vintage instruments often impossible to find in synth or samplers: 5 harpsichord multisamples demonstrating the variety of tones available, Piano Forte, 3 Accordion, Celtic Harp, Viol and baroque flutes are presented here with outstanding quality and a very good usability..
490 samples (187 Harps / 36 Piano Forte / 91 accordions / 64 celtic harp / 89 Viole / 25 flutes) – 18 audio tracks / AKAI partition: 15 volumes / Wav-Acid/Aiff: 407 MB

USBCDSC023 ..... **49.95**



Vol. 24 Industrial Loops

Straight from the weirdest of all build site: large impacts, metal hits and a wealth of superbly sound-designed percussive samples. All those ingredients are mixed with electronic percussions in a tasteful way. Dark Ambience, spacey "electric" patterns, hyperdensified tracks, electro industrial beats… The variety of sounds – difficult to describe with plain words.
406 samples (267 loops / 139 single hits); 62 audio tracks / AKAI partition: 35 volumes / Wav-Acid/Aiff: 511 MB; Tempo: 50 - 202 BPM

USBCDSC024 ..... **49.95**



Vol. 25 Big Beat Volume 1 Drumloops

Distinctive loops make this new, exciting style what it is: acoustic loops with attitude, fills. Furious construction kits full of the most creative stuff. Loops are powerful like never before, from the classics to brand new loops tailored to push the boundaries of Big Beat. Will keep a remixer happy for years.
567 samples (444 construction kits / 123 drumloops); 47 audio tracks / AKAI partition: 27 volumes / Wav-Acid/Aiff: 490 MB; Tempo: 125 - 147 BPM

USBCDSC025 ..... **49.95**



Vol. 26 Big Beat Volume 2 Samples & More

Enhances volume 25. Tons of awesome samples and loops: a comprehensive distortion library, guitar and bass loops, crazy drum fills, separate sounds (hi-hats, claps, kicks, snares, cymbals, noise, vox, textures) and processed samples (brass, piano & organ, alarms, reverse texture, stack sounds).
808 samples (332 drums-percs / 48 Licks / 79 synth ses / 349 others); 43 audio tracks / AKAI partition: 43 volumes / Wav-Acid/Aiff: 513 MB; Tempo: 105 - 130 BPM

USBCDSC026 ..... **49.95**



Vol. 27 Reggae, Ragga & Dub

Drum and instrument loops and samples, effects and special textures. Special care to match the cool of true Ragga, the spaciousness of Dub style, without being victim of hype or harsh sounding recordings. Fills, sequences, construction kits, and a special mention for the outboard effects used. Outstanding quality.
1096 samples (502 construction / 485 drums / 109 sounds); 79 audio tracks / AKAI partition: 59 volumes / Wav-Acid/Aiff: 622 MB; Tempo: 130 - 175 BPM

USBCDSC027 ..... **49.95**

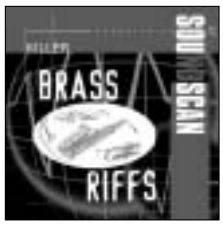
B&H PHOTO - VIDEO - PRO AUDIO
EQUIPMENT LEASING AVAILABLE



Vol. 28 Acoustic & Synth Choirs

These original multi-samples use beautiful children's choirs, male choirs and female choirs. Recorded in Europe, these choirs will bring an organic element of beauty to your musical creations. There is also a section of synth choirs that will lay underneath the most complex of productions and jump right out of your mix!
481 samples (181 children / 83 men & women / 60 women / 158 synth choirs); 61 audio tracks / AKAI partition: 61 volumes / Wav-Acid/Aiff: 425 MB

USBCDSC028 ..... **49.95**



Vol. 29 Killer Brass Riffs

Do you really need more of an explanation? The name says it all! Many producers have waited far too long for a brass disk that is laid out to be useful for any style of music. From trumpet to flugelhorn, sax to trombone, it's all here – with just the right flare to any mix.Bonus section: tons of hits and stabs.
845 samples (533 Riffs / 166 Riffs Solo / 146 hits & stabs); 46 audio tracks o AKAI partition: 46 volumes / Wav-Acid/Aiff: 420 MB; Tempo: 63 - 133 BPM.

USBCDSC029 ..... **49.95**



Vol. 30 Bass Grooves

Features lots of styles and playing techniques, from pop, rock and even hard-rock to funk, from jazz, Latin and soul to R'n B. Over 60 folders – each with an easily identifiable selection of loops, tagged with tempo and key. Lively bass grooves, with all the cool ingredients to make a successful track: fantastic timing, unobtrusive line, excellent sound!
338 samples (14 demos / 219 bass loops / 105 single notes); 64 audio tracks / AKAI partition: 67 volumes / Wav-Acid/Aiff: 558 MB; Tempo: 54 - 135 BPM.

USBCDSC030 ..... **49.95**



Vol. 31 Rock & R 'n B Miro's Grooves

A hybrid collection containing tons of electric guitar and bass phrases, expertly played by J.M Miro. In order to fully understand the power of this library, you will find a bunch of construction kits produced by the producer of the Soundscan volume 19 Hip Hop / R 'n B – the best selling Soundscan ever.
580 samples (404 loops, riffs & licks / 176 single shots); 50 audio tracks / AKAI partition: 50 volumes / Wav-Acid/Aiff : 480 MB; Tempo: 82 - 127 BPM

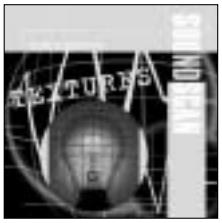
USBCDSC031 ..... **49.95**



Vol. 32 Roots Disco

Break out your fur coats, put on that afro, and get yourself into those tight bellbottom blue jeans once again — the FEVER has just hit! Roots Will take you back to the days of Charlie's Angels and Ponch & John! Drumloops, construction kits, hits and some Starsky & Hutch style bass, strings and guitar! Ccaptures what it took to sound good in the 70s.
774 samples (289 loops, riffs & licks / 425 single shots & co); 58 audio tracks / AKAI partition: 58 volumes / Wav-Acid/Aiff: 600 MB; Tempo: 113 - 150 BPM

USBCDSC032 ..... **49.95**



Vol. 33 Disjointed Textures

Following in the heritage of SoundScan Volume 13 - Twisted Special Effects, Disjointed Textures displays yet another avenue of sound design that's fresh, new, just waiting to be used. From the most beautiful ambience to the most frightening sonic textures, this is a film scene just waiting to happen!
244 samples (152 textures / 53 pads FX / 39 drums FX); 50 audio tracks / AKAI partition: 50 vol. / Wav-Acid/Aiff: 671 MB / Categories: Space Sounds, Pads & FX, Sciences, Vox Loops, Devils…

USBCDSC033 ..... **49.95**



Vol. 34 Burning Grunge Hip Hop

The nastiest, low-down set of hip hop loops available. From construction kits to drum loops to nasty, grungy guitar and bass – sure to put some serious attitude into your mix! Will give your stuff that authentic underground sound your Hip Hop project has been needing. A grab bag of sounds heard nowhere else.
538 samples (164 loops, riffs & licks / 374 single shots & co); 52 audio tracks / AKAI partition: 52 volumes / Wav-Acid/Aiff: 550 MB; Tempo: 88 - 116 BPM

USBCDSC034 ..... **49.95**



Vol. 35 The Dark Side of Trip Hop

These grooves have that unique "laid-back" quality that sets them apart from any other slow loop. Dark Trip Hop is not to be mistaken with anything else: on the production side, each of the kits is instantly identifiable by its fat sound. Inject a loop, and your current mix will feel a lot heavier. New musical style.
246 samples (211 drumloops & kits / 35 special FX); 72 audio tracks / AKAI partition: 72 volumes / Wav-Acid/Aiff: 574 MB; Tempo: 56 - 94 BPM

USBCDSC035 ..... **49.95**



Vol. 36 Percussions Loops

These essential loops have a priceless quality: they are simple! The featured instruments are: Congas, Bongos, Claves, Shakers and tambourines, but also Guiros, Agogos, Djembes, Steel and wood Latin percussion. Latin at heart – a staple of percussion work in contemporary music.
394 samples (96 shakers/43 triangles/ 71 congas,/12 tamb...); 58 audio tracks / AKAI partition: 66 volumes / Wav-Acid/Aiff: 300 MB; Tempo: 60 - 150 BPM

USBCDSC036 ..... **49.95**

SOUNDSCAN



Vol. 37 Pop / Rock Drumloops

Ranging from the most timeless patterns to the really contemporary rock loops. Delivers a wealth of variation for each groove, sufficient to build a complete song, with cymbal add-ons, tom-tom variations, Hi-Hat buildups and adapted fills. Impeccably looped. These eternal classics will be simple essentials for a long, long time.
384 samples (86 main / 174 variations / 174 fills / 75 hits); 22 audio tracks / AKAI partition: 22 volumes/ Wav-Acid/Aiff: 464 MB; Tempo: 96 - 152 BPM

USBCDSC037 ..... **49.95**



Vol. 38 Funk & Groove Drumloops

This library is unique in many ways : one CD is enough to tackle all the known variations of the classic "funky drumming" and provide a wealth of variations for each of the main loops: cymbal, tom-tom, hi-hat alternates, and matched fills. This is timeless funk for all your needs, between 80 and 120 BPM, whether you're into pop/rock, trip-hop, R'n'B....
453 samples (112 main / 163 variations / 62 fills / 116 hits); 28 audio tracks / AKAI partition: 28 vol. / Wav-Acid/Aiff: 584 MB; Tempo: 78 - 128 BPM

USBCDSC038 ..... **49.95**



Vol. 39 Brushed Drumloops

Brushes and Hot Rods, playing groove, trip hop, jazz and latin drum patterns, with comprehensive variation for each f the the main grooves. The slow and dark ambience of the slower tracks are complemented by hellish speed trains and other brush classics. Perfectly matches the sounds of the Soundscan 38 and 39.
309 samples (89 main / 173 variation / 47 fills)
25 audio tracks / AKAI partition: 25 volumes / Wav-Acid/Aiff: 500 MB / Tempo: 72 - 200 BPM

USBCDSC039 ..... **49.95**



Vol. 40 Complete Accordions

The styles grouped here are quite varied, ranging from Canadian folklore to French "bourrée", while also approaching polka, German traditional and Beerfest stuff, even Celtic melodies. Each folder starts with accordion rhythm patterns, followed by themes played unaccompanied in various harmonic combination..
340 samples (180 Chords riffs / 115 Solo / 45 single shots); 59 audio tracks / AKAI partition: 32 volumes / Wav-Acid/Aiff: 500 MB; Tempo : 80 - 195 BPM

USBCDSC040 ..... **49.95**



Vol. 41 Roots Funk

Whether you're producing R 'n B, pop or Dancefloor-oriented tracks, it's all for the groove now! A typically 70's CD, with superb drum loops reminiscent of the era; inspired keyboard parts, and numerous classic talk box effects (also called mouth-tube). To complement bass grooves, guitar loops are present in various forms.
369 samples (32 demos / 337 loops & samples); 63 audio tracks / AKAI partition : 31 volumes / Wav-Acid/Aiff: 550 MB; Tempo : 85 - 137 BPM

USBCDSC041 ..... **49.95**



Vol. 42 Celtic Flavors

Features phrases and samples of all the typical instruments that thrive in Brittany, Ireland and Cornwall: celtic guitar, violin, concertina, acoustic guitars, Irish flute and mandolin - as well as biniou (small bagpipes) and bombarde (french shawm). Styles approached all part of the Celtic tradition. Multi-samples, with biniou, bombarde, celtic guitar and flute.
514 samples (412 loops & riffs / 102 single shots)
52 audio tracks / AKAI partition: 52 volumes / Wav-Acid/Aiff: 498 MB; Tempo: 80 - 160 BPM

USBCDSC042 ..... **49.95**



Vol. 43 Arabian Traditions

This volume is dedicated to the wonderful sound of Arabian voices and instruments. With the included loops and phrases, the traditional melodies of Arabia come alive, allowing you to include that unique ingredient in your productions. Featured instruments are: luth, male and female voices, violin and percussions.
332 samples (289 loops & riffs / 43 single shots)
39 audio tracks / AKAI partition: 38 volumes / Wav-Acid/Aiff: 460 MB; Tempo: 80 - 140 BPM

USBCDSC043 ..... **49.95**



Vol. 44 African Traditions

A truly unique meeting of tradition and more modern parts, this collection focuses on typical sounds from Africa. From Marimba and Kalimba to Djembe and Bongos, the percussions are all oozing with authenticity. You'll also find male and female voices, typical African guitar parts, and many other African music staples.
TBA samples (TBA Riffs & co / TBA Licks / TBA single notes); TBA audio tracks / AKAI partition: TBA volumes o Wav-Acid/Aiff : TBA MB (info. unavaible)

USBCDSC044 ..... **49.95**



Vol. 45 Indian Traditions

The traditional melodies of India come alive, allowing you to include that unique ingredient in your productions. Featured instruments too numerous to list here, from percussions to vocals and instruments. Dive into the magic of Indian Traditions.
391 samples (81 Voices, 51 flutes, 104 Percs loops, 43 Sitar, 19 Papang, 17 Ravanhata, 14 violins, 41 Pungi, 10 Various Instruments, 11 Ambiences)
40 audio tracks / AKAI partition: 40 volumes / Wav-Acid/Aiff: 520 MB;Tempo: 80 - 170 BPM

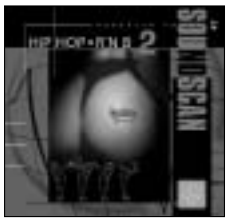
USBCDSC045 ..... **49.95**



Vol. 46 70's Breakbeats

Tons of authenticity and feeling, a great treasure-chest for those modern productions where they can be sliced, diced, slowed down or warped. Sounding just like they were lifted straight off rare and obscure vinyls of the era, 70's breakbeat provides the real thing, with the added work necessary to bring to you these copyright-clean loops
691 samples (512 Breakbeats & loops / 179 single shots); 62 audio tracks / AKAI partition: 62 volumes / Wav-Acid/Aiff : 240 MB; Tempo: 84 - 133,5 BPM

USBCDSC046 ..... **49.95**



Vol. 47 Hip Hop R 'n B v.2

The Team of producers who brought you volume 19 "Hip Hop R 'n B" strike back with this new cool Soundscan! Simplicity in programming will catch your ear first. Such impact with so little clutter is the signature of the best. The construction kits offer you both essential basics, and subtle add-ons to color the basic rhythm section.
561 samples (219 loops, 252 single drums, 61 single shots instruments, 29 demo tracks); 90 audio tracks / AKAI partition: 29 volumes / WavAcid/Aiff: 490 MB; Tempo: 85 - 133 BPM

USBCDSC047 ..... **49.95**



Vol. 48 Ominous Drones

This truly fantastic collection of the deepest textures available today will suggest its numerous ways to you before you've reached the end of the audio Part. The AKAI part features an ingenious mapping: two drones per patch in left hand-right hand keyboard splits. Galactic textures, Low Tones, Mystical, Sub Marine . . .
90 samples & drones; 45 audio tracks / AKAI partition: 45 volumes / Wav-Acid/Aiff : 582 MB (Galactic, Hi Spheres, Low Tones, Mystical, Submarine, Synth, Vox...)

USBCDSC048 ..... **49.95**



Vol. 49 Analog Fantasies

An anthology of classic analog synthesizer multisamples, all ready to play: pads, sweeps and swells, leads, atmospheres, analog effects. All classic sounds, and all you need to quench that thirst for analog synth that have become rare and outrageously expensive collector's items.
727 samples (158 synth patches using 727 samples) / 20 power analog FX); 61 audio tracks / AKAI partition: 61 volumes / Wav-Acid/Aiff: 562 MB

USBCDSC049 ..... **49.95**



Vol. 50 Electric & Acoustic Guitars

The best guitars on the market in the best multi-sampled library! No loops, only the most useful guitar patches. Six and 12 strings acoustic guitars, with Nylon and steel strings. Some typical playing styles, like Bottelneck. Multisamples supplied for soft, hard and muted version of instruments.
1087 samples (586 acoustic / 332 electric / 169 muted) / 50 audio tracks o Wav-Acid / Aiff / EXS24 / Halion / Reason2 / AKAI Z4-Z8/Mach5: 450 MB / AKAI partition: 23 vol.

USBCDSC050 ..... **49.95**



Vol. 51 House Side of 2 Steps

When the energy of House music and the funk of R 'n B meet in a dark alley, the cutting-edge music press gives birth to a new style – "2 steps". An unprecedented merging of styles provides you with this ready-to-use library only Organized in construction kits and ready for immediate use.
935 samples (451 loops, 345 drums, 139 instruments/FX); 92 audio tracks / AKAI partition: 35 volumes / Wav-Acid/Aiff: 470 MB; Tempo: 120 - 137 BPMSciences, Vox Loops, Devils

USBCDSC051 ..... **49.95**



Vol. 52 R 'n B Side of 2 Steps

This exciting new genre allows creative designers and musicians to mix the punch of house music with the atmospheres and originality of Hip Hop/R 'n B. Construction kits contain striking themes and instrument tracks to embellish the super heavy rhythm section. Tempo info extremely accurate.
TBA samples (TBA Riffs & co / TBA Licks / TBA single notes); TBA audio tracks / AKAI partition: TBA volumes / Wav-Acid/Aiff: TBA MB (informations unavaible)

USBCDSC052 ..... **49.95**



Vol. 53 Mediterranean Traditions

Jam-packed. Instrumental loops in all or most major and minor keys of Baglamas, Bozouki, Tzouras and Saz. Plus, additional phrases of Cumbus and Oud. Many instrument multisamples, with variations such as Pizz or Tremolo. Drum and percussion loops,; tempo ranging from 100 to 185 bpm. Gifted local players.
546 samples (172 Riffs & co / 99 drums & percs loops / 275 single notes); 89 audio tracks / AKAI partition: 78 volumes / Wav-Acid/Aiff: 400 MB; Tempo: 60 - 185 BPM

USBCDSC053 ..... **49.95**



Vol. 54 Keyboards Phrases

A very effective way to give atmosphere and life to a modern music track. Seriously talented players teamed up to provide the sweetest collection of Electric Pianos, Clarinet and acoustic piano loops available today. Perfect recordings for modern music production where sound quality is at the foundation of success.
TBA samples (TBA Riffs & co / TBA Licks / TBA single notes); TBA audio tracks / AKAI partition: TBA volumes / Wav-Acid/Aiff: TBA MB (informations unavaible)

USBCDSC054 ..... **49.95**

SOUNDSCAN



Vol. 55 Creative & Dub Drumloops

Enroll an ace drummer, record him with vintage gear and a good dose of attitude, involve some classic processors… and you've got yourself a collection of fantastic drum-loops that will be the basis of excellent future-retro tracks. This collection is as eclectic as it is inspiring: these acoustic groove, jungle and dub loops are anything but basic.
225 samples (acoustic processed drumloops); 65 audio tracks / AKAI partition: 28 volumes / Wav-Acid/Aiff: 460 MB; Tempo: 63 - 160 BPM

USBCDSC055 ..... **49.95**



Vol. 56 Modern Percussion Loops

Bread and butter loops played with the most popular percussion instruments in western music. An essential collection with bongos, congas, djembe, shaker, tambourine, surdo, timbale and triangle. Two tempo groups: from 89 to 100 bpm for funky, groovy tracks; 112 to 133 for dance and pop tracks.
448 samples (bongo, conga, djembe, shaker, tambourine, timbal...); 67 audio tracks / AKAI partition: 39 volumes / Wav-Acid/Aiff: 450 MB; Tempo: 84 - 133 BPM

USBCDSC056 ..... **49.95**

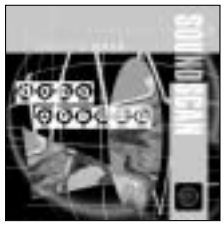


Vol. 57 SFX Toolbox Vol. 1

This release will be followed by many others, so that you may mix and match your sfx collection. Focuses on Ambient sounds. Interior and exterior recordings, plus many other sound effects that allow you to create a virtual scene or dress up a basic video shoot.
187 samples (insects, animals, atmospheres, storm, water, winds, birds, trains, engines, subways, town, vehicles, doors, steps...); 24 audio tracks / AKAI partition: 24 Volumes / Wav-Acid / Aiff: 600 Mb

USBCDSC057 ..... **49.95**

Vol. 58 Drum'n Bass Sensations



This CD revives the genre while producing a fresh new library to work with. All construction kits start with the full loop, followed by individual instrument loops, and a few essential single samples. Recorded with the best studio equipment currently available, and the best and biggest sounding Drum & Bass rhythms ever!
464 samples (256 samples in Construction Kits / 208 single sounds); 80 audio tracks / AKAI partition: 30 volumes / Wav-Acid/Aiff: 440 MB; Tempo: 100 - 178 BPM

USBCDSC058 ..... **49.95**

Vol. 59 Spanish & Gypsy Traditions



Created by natural-born musicians and excellent Gypsy performers. Guitars, vocals, palmas and many other specific loops almost impossible to record in the USA or Japan, for lack of trained players. Impeccable looping; accurate tempo.
626 samples (355 loops/solos / 271 single shots) 61 audio tracks / Wav-Acid / Aiff / EXS24 / Halion / Reason2 / AKAI Z4-Z8 / Mach5 / REX2: 651.9 MB / AKAI partition: 28 volumes; Tempo: 70 - 133 BPM

USBCDSC059 ..... **49.95**

Vol. 60 Asian Traditions



Truthful Asian ambience is really difficult to produce, mostly due to the unavailability of credible loops.
Addresses the need for traditional music from Asia: vocal, percussion and highly exotic instrumental loops. An excellent source of mystery and disorientation.
412 samples (162 loops / 250 single shots) – 38 audio tracks / Wav-Acid / Aiff EXS24 / Halion / Reason2 / AKAI Z4-Z8 / Mach5 / ReX2: 500 MB AKAI partition: 36 volumes

USBCDSC060 ..... **49.95**



Vol. 61 SFX Toolbox Vol. 2

Second volume of our SFX library. Mostly short sounds and household objects. Starts with impacts (metal, glass, woods). Includes ambient sounds like water, doors, appliances, computing and office sound effects. Switches, buttons, plates, dishes – as well as boxes, bottles, money, lighter, matches, etc.
770 samples (impacts); 78 audio tracks / Wav-Acid / Aiff / EXS24 / Halion / Reason2 / AKAI Z4-Z8 / Mach5: 509 MB / AKAI partition: 76 volumes

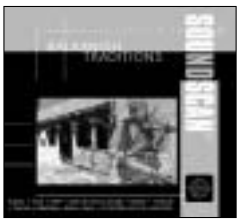
USBCDSC061 ..... **49.95**



Vol. 62 Complete Harp

The coveted harp is used in many classical compositions. Starts with a nice selection of musically inspiring phrases. Benefit from the subtleties of Glissando and Bisbigliando. Mesmerizing and enchanting.
380 samples (36 loops / 35 bisbigliandos /39 chords / 29 cascades / 133 glisses / 108 single shots); 33 audio tracks / Wav-Acid / Aiff / EXS24 / Halion / Reason2 / AKAI Z4-Z8 / Mach5 / REX2: 513,6 MB / AKAI partition: 33 volumes; Tempo: 70 - 140 BPM

USBCDSC062 ..... **49.95**



Vol. 63 Balkanish Traditions

A very original music style. The Balkans' great variety of instruments include Accordion, Double bass, Violin, Saxophone, Derbuka, Strings, Percs, Voices - in musical recordings of Vlaque, Sirba, Paidouchko, Freilakh, Jovamo and many more. Bonus section: instrument multisamples..
626 samples (355 loops & solos / 271 single shots) 61 audio tracks / Wav-Acid / Aiff / EXS24 / Halion / Reason2 / AKAI Z4-Z8 / Mach5 / REX2: 651.9 MB / AKAI partition: 28 vol.; Tempo: 70 -133 BPM

USBCDSC063 ..... **49.95**

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AMERICAN EXPRESS Card

Heart of Africa Vol. 1



An unprecedented sonic expedition into the rich musical heritage of Africa. Discover a fascinating compilation of wonderful instruments, voices, and performances that evoke deep images and primitive emotions. Recorded on location all across the continent.

South African Choirs (phrases & chants): Tribal Choirs (with Chromazones phrases & multisamples); Solo male & female african vocalists (phrases & utterances); Body Percussion, Stomping, Marching, Screams & Jumping rituals; Massive African percussion, fx & ensembles (instrument samples & loops); Extensive collection of Kalimbas, Sansas, M'biras (phrases & multisamples); Marimba, Slit/Log Drums & Balaphone; Bushman instruments; Ethiopian Harps; Berimbau, Mouth bows; Nigerian Udus; Berber yells & Morrocan instruments from N.Africa; S.African Guitar grooves; Tribal remix grooves; Mozambique hunting horns & Pygmy pipes; Village ambiences & SFX and much much more!

CD-ROM AKAI (Emu, Giga, Software Sampler compatible) **285.00**
CD-Audio 2-disc set.....**124.95**

Heart of Africa Vol. 2



Music plays an important role in all cultures across our planet. Yet its heart and soul truly must reside in the great continent of Africa.

Heart of Africa Volume 2 features rare performances recorded live at a two week tribal competition in Kenya. Most of the 8000 participants (primarily children) had traveled many miles by bus to perform their music. From huge feather head pieces and masks to shields and brilliant robes, the outfitting represented serious commitment.

A fascinating compilation of authentic choir and vocal shouts, chants and polyrhythmic percussion ensemble grooves from the Maasai, Kikuyu, Meru, Pokot, and Nandi peoples. A collection of powerfully primal and untamed tribal ensemble phrases & SFX. This volume is an intriguing disc for your next film score, remix or album project.

CD-ROM AKAI (Emu, Giga, Software Sampler compatible) **189.95**
CD-Audio 2-disc Set **99.95**

Heart of Asia



Phrases, Instruments, Utterances from the Far East – the world's first comprehensive collection of Asian instruments and phrases. Digitally recorded in the heart of Asia, this remarkable two-disc set includes over a gigabyte of exotic instruments and performances, including:

Indian, Javanese, and Thai percussion & ensemble grooves; Nepalese, Malaysian and Indonesian percussion ensembles; Complete Javanese Gamelan Orchestra; Chinese, Malay, Nepalese and Indian flute phrases; Indian Tambouras; Chinese & Malay temple flutes; Indian Bamboo Bass Flute phrases; Tibetan Parade Band with Horses; Chinese Cymbals and Gongs; Sitar phrases and samples; Veena, Pipa, Erhu, San Xuan, Luan and Yang Chin; Thai & Malay Orchestra phrases and samples; Chinese Koto instruments, glissandos & phrases; Tibetan children singing; Chinese and Thai Opera divas – vocal phrases; Sanskrit, Hindi, Malay, Javanese, Thai and Mandarin vocal phrases, chants, poetry and speech; Tibetan monks singing in a cave and much, much more!

All samples from this renowned collection are completely authentic, royalty free and record-ready for your next film score, album or remix project..

"... a treasure chest of lovingly sampled performances, plus individual multi-sampled instruments... enchanting... very expressive... long lilting phrases that soar like birds... powerful... heartfelt... intricate rhythms... riveting... gorgeous... evocative... If you're not satisfied to sound like every other new kid in the 'hood, Heart of Asia will definitely give you some cruel tools... a real find." – Keyboard magazine

CD-Audio 2-disc Set (condensed library of phrases only).....**124.95**

CD-ROM 2-disc Set (Roland, AKAI/Emu/ASR, SampleCell, & Kurzweil)
Complete library of phrases & instruments **379.95**

Supreme Beats



A Percussion Library by Bashiri Johnson. Supreme Beats comprises an exhaustive collection of over five hours worth of inspiring acoustic percussion grooves and comprehensive instrument samples. Recorded over six months in top NY and LA studios, the production is state of the art. The Supreme Beats Percussion series is an essential resource for the modern composer, producer and remixer, that is destined to become a classic. It features Bashiri Johnson (Whitney Houston, Donald Fagen, Madonna), Luis Conte (World renown latin percussionist & LA session ace) and Cyro Baptista (Brazilian master), and many others.
The key to Supreme Beats is that each percussion groove is presented in three tempos, first in a full mix, and then with each percussion element solo'ed, allowing for the most flexibility in how the loops can be combined and applied to your music. A full compliment of single hits of all of the instruments are included as well.
In addition to the many traditional and ethnic percussion instruments used on Supreme Beats, unusual live sources such as elevator shafts, breathing, human stomping and many others were creatively incorporated into the grooves, giving these rhythms a truly unique sonic quality.
". .. a feast of human feel, energy and emotion... enough material to feed an army of samplers... one of the most exhaustive and impressive percussion libraries we've seen, and it sounds even better than it looks! ... If we had three thumbs we'd turn 'em all up... Supreme Beats is a winner."
- KEYBOARD Quadruple 5 star review

4 Audio CD Boxed Set African, Contemporary, World, Dance/Hip-Hop..**239.95**
Vol.5 "International" Audio CD.....**99.95**
African/Contemporary CD-ROM **189.95**
World/Dance CD-ROM Roland AKAI/Emu/ASR SampleCell Kurzweil .**189.95**

SPECTRASONICS

BackBeat



If you tired of searching through piles of overcooked, pretentious loops, just to find a solid groove for your song, Backbeat is just what you're looking for...great feeling, rock-steady grooves with attitude, but without overplaying. Producers Eric Persing and Bob Wilson recorded four world-class drummers in numerous legendary studios, with some of the best drum rooms on the planet. The latest technology was used to capture a stunningly Hi-Fi, yet very natural sound.

The featured players include: Gregg Bissonette (Santana, Joe Satriani, Don Henley); Bob Wilson (Liquid Grooves, George Benson, The Crusaders); John Ferraro (Larry Carlton, David Benoit, Albert Lee, Aaron Neville), Eric Boseman (Liquid Grooves, James Newton-Howard)

Create your own complete drum track with Song Construction Elements including Variations, Fills, Fill Bars, Intros, Breakdowns, Ride and Sidestick patterns, add-on cymbals and more. The power, feel and subtlety of a live drummer, combined with all the editing control of your MIDI sequencer results in a quantum leap in what can be achieved with drum loops.

Note: The Backbeat CD-ROM includes a Mac/Windows Data Disc, which contains the Groove Control sequences and high resolution WAVE files of all the loops. You can use the WAVE files for auditioning grooves in Soundfinder, and for direct use in software like Acid, Logic, Cubase, Reason or Pro Tools!

CD-ROM Multi-Disc Set (AKAI/Emu, Roland)
includes Groove Menus and .WAV files **189.95**

CD-Audio (Loops only)..... **99.95**

Retro Funk



The sonic texture of these loops makes them ultra-flexible, bringing massive "vibe" to any modern track. Producer Eric Persing's mixes create a unique, dry sonic attitude in this library, making it ideal for layering with other groove elements. Vintage mics, warm analog tape compression and antique drums all contribute to the funkiness. And with Groove Control, the grooves are at your command, effortlessly blending their timing and human feel to match your music. Turn a GoGo groove into Deep House, a JB Boogaloo into Drum & Bass, a Fatback P-Funk groove into Alternative Reggae, or an R&B stomp into a Pop Ballad!

The featured players on Retro Funk include Gregg Bissonette (Santana, Joe Satriani, Don Henley), Bob Wilson (Liquid Grooves, George Benson, The Crusaders), John Ferraro (Larry Carlton, David Benoit, Albert Lee, Aaron Neville), and Eric Boseman (Liquid Grooves, James Newton-Howard).

Create your own complete drum track with Song Construction Elements including Variations, Fills, Fill Bars, Intros, Breakdowns, Ride and Sidestick patterns, Add-on Cymbals, Claps, Vinyl noise and funky live Percussion loops too! This massive stash of grooves is destined to be a legend.

CD-ROM (AKAI/Emu, Roland, GIGA)..... **189 .95**
includes Groove Menus and .WAV files

CD-Audio (Loops only)..... **99.95**

Bass Legends



A World-Class sampling library featuring three of the greatest bassists on the planet: Marcus Miller, John Patitucci and Abraham Laboriel. Their distinctive performances have been fundamental to literally thousands of great recordings, including the classic albums of musical giants like: Stevie Wonder, Donald Fagen, Chick Corea, Scritti Pollitti, Bryan Ferry, Elton John, Joe Zawinul, Aretha Franklin, Joe Sample, Dave Gruisin, Ry Cooder, Al Jarreau, Robbie Robertson, Quincy Jones, Luther Vandross, George Benson, Michael Jackson, Chaka Khan, David Sanborn, Natalie Cole, Boz Scaggs, Herbie Hancock, Miles Davis, and.....well, you get the idea!

Bass Legends is one of the first sample libraries to offer you both intricately multi-sampled instruments and chromatically sampled grooves and phrases called Chromazones.

For bass sequencing applications, the 650MB CD-ROM collection is recognized worldwide as the most complete and musical Bass Instrument Library on the market. The players' instruments cover a tremendous variety of tones, ranging from 4, 5, 6 & 8 string electrics- to Patitucci's Pollman Acoustic, and were all recorded with the finest in vintage tube technology and captured from within a musical context.

CD-ROM (AKAI [Emu, Giga, Software Sampler compatable], Roland)
basses and grooves **284.95**

CD-Audio (grooves only)..... **99.95**

Symphony of Voices



An extensive collection of voices produced by Eric Persing, with a breathtaking variety of multisamples, phrases and FX from some of the most prestigious singers.

Disc 1 is devoted to richly detailed samples of the 80-voice London Chorale recorded in a large cathedral in England. Variations include multiple dynamics of multisampled ahs, oos, ohs, ees, humming, intense staccato stabs, natural crescendo/decrescendo swells, octaves, moving vowels, vibrato versions and even whistling!

Disc 2 brings you the power of classical soloists, with a wide variety of license-free Operatic Tenor and Soprano phrases (with words and melismatic), in addition to the natural multisamples of each singer. It also contains London Choir FX samples, Major, Minor and Suspended Chords with high & low chord voicings, thick clusters, Twelve-Tone and Ligeti fx, whispering and more!

Disc 3 focuses on an English Boys Choir and a Gregorian Mens Choir. The Boys Choir features multisampled phrases, soft and loud multisampled static vowels and moving vowels, as well as Solo Boy phrases and multisamples. The Gregorian Mens Choir delivers multisampled soft and loud ohs, mms, fifth drones, intervals, multisampled Latin words and ancient Gregorian Chants!

Disc 4 offers an overwhelming variety of Lush Multitracked Pop Stacks, created from over 10,000 perfect vocal performances! Comprehensive multisampled vowels, moving vowels, washes and "scooped" notes. Gorgeous male and female textures will add that smooth silk to your next hit!

4-CD-ROM Set (Roland, AKAI/Emu, Kurzweil, SampleCell)..... **474.95**

B&H PHOTO - VIDEO - PRO AUDIO
ONE HOUR FREE PARKING
AT 349 W. 34th STREET (with purchase of $100 or more)

Liquid Grooves



Grooves that have existed only in your imagination. Fluid rhythms forged by a synthesis of propelling live drum grooves and unusual, organic percussion loops. The live elements, performed by some of L.A.'s most creative session players, have been transformed into a unique sonic texture via ground-breaking signal processing techniques and state of the art instruments such as the Korg Wavedrum.

These highly-original grooves have lots of space and are presented in a Construction Kit format similar to the Supreme Beats collection. Many of the grooves are presented in both "Natural Acoustic" and "Hybrid Remix" versions for a new level of versatility. If you've been frustrated at the lack of slow tempos on most loop discs, this is the library you've been searching for. Tempos start at a laid-back 52 BPM and go to a cruising 126 BPM.

The liquid concept is 2-fold : Sparse "hand-played" grooves that feel great, combined with a focus on uncharted sonic territory.

CD-AUDIO Grooves only.....**99.95**

CD-ROM Complete library
(Roland, AKAI/Emu/ASR, Kurzweil and SampleCell).....**189.95**

Liquid Grooves Groove Control



The unique and inspiring grooves from the timeless library "Liquid Grooves" are now available in a Groove Control version. This revolutionary new method allows you to mix and match grooves in any tempo, and will even follow tempo changes from your sequencer!

Groove Control also allows you to easily change the pattern, feel and pitch of the rhythms to your music simply by using the MIDI editing capabilities of your sequencer. Imagine being able to tune the Wavedrum grooves to the key of your song in real-time, without the rhythm changing! You can also program your own original patterns and fills with the same sounds and mixes from this library!

If you're new to Liquid Grooves, this is the only version you will need. If you're a existing Liquid Grooves CD-ROM user, you can upgrade to the new Groove Control version.

CD-AUDIO Grooves only.....**99.95**

CD-ROM Complete library
(Roland, AKAI/Emu/ASR, Kurzweil and SampleCell).....**189.95**

Burning Grooves



Finally, a drum library that delivers maximum impact. Killer Kits and blazing live drum grooves laid down by L.A. slammer Abe Laboriel Jr. (Seal, SteveVai, Joe Sample) with extreme re-mixes by producer Eric Persing.
The diverse selection of unrelenting alternative, power funk, progressive soul, rock & roll and grungehead loops in a wide variety of tempos (56-215 BPM) puts this collection in a class by itself.
Each groove has a unique mix (edge, hi-fi, ambient, distorted, lo-fi, etc), and some have both HI-FI and LO-FI remixes. It is one of the only libraries to include different time signatures (3/4, 5/4, 6/8, 7/8, 12/8), and those hard to find Tom-Tom and Sidestick grooves. All the loops have been "pre-tweaked" for synchronization at the listed tempo, saving you alot of time.
Burning Grooves offers versions of the grooves played without kick drum and without snare drum for unprecedented flexibility in customizing the loops to your music. This allows you to create your own kick or snare patterns within the loop, breakdowns and makes it easier to layer Burning Grooves with other loops. Separate fills and intro loops are available to help smooth transitions, and create the "live vibe". The fills and grooves don't have any Cymbal Crashes limiting their usefulness. Add your own when you need them.
The CD-ROM version includes "Smokin' Kits" – an entire drum hits library! These are some of the punchiest and most expressive kits ever assembled, since they were sampled within a musical context. If you're looking for attitude, search no further. Where there's smoke, there's fire!

CD-Audio (grooves only).....**99.95**

CD-ROM (Roland, SampleCell, AKAI/EMU/ASR, Kurzweil)
Includes Audio CD. Grooves & Smokin' Kits **189.95**

GROOVE CONTROL

ILIO and Spectrasonics have teamed up to create Groove Control activated libraries that give you independent control over the tempo of live, stereo sampled loops, without affecting pitch. What's more, you can use your existing sampler and sequencer!
Change a straight backbeat groove into a funky shuffle with just a few clicks. If a drummer's performance was a little too laid back for you, simply quantize it a little. Let's say your bass line riff is pushing too hard against the kick drum - just nudge the kicks over until they're "on the money." Or completely change the pattern.
Simply load a groove from the CD-ROM into your sampler, then open the included sequence on your Mac or PC, and you've got Groove Control. Since you're already working in the sequencer of your choice, there's no learning curve. It just works! And since it works inside your sequencer, the grooves automatically follow your tempo changes. Plus, you can use MIDI controllers to automate mix levels, filter sweeps, drum tunings, or you can stack many different layers of grooves on top of each other, automatically locking them to tempo. It all adds up to unprecedented control and inspiring flexibility.
The first Groove Control activated libraries will feature some of the most famous drummers and percussionists in the world - names like Gregg Bissonette, Alex Acuña, Bashiri Johnson, Bob Wilson, John Ferraro, Eric Boseman and many more to come!

How Do You Use Groove Control?

1. Find the groove you want to use.
2. Load the Groove Control version of that groove into your sampler.
3. Load the corresponding Groove Control sequence into your sequencer.

That's it! Once you copy the sequence into your own song, the groove automatically snaps to tempo. Feel free to quantize, alter the feel, change time signature, change drum tunings, or even layer more grooves. Do what ever you want, and your Groove Control groove will follow right along!

SPECTRASONICS

Distorted Reality 1



Produced by Eric Persing, Distorted Reality is a one-of-a-kind library that explores an experimental world of extraordinary colors. This is quite possibly the best selling professional sample library of all time.

Featuring Cutting Edge Textures, FX, Batterie and Loops; from altered 3-dimensional atmospheres transformed via true harmonic Sound Morphing, to incendiary fuzz mutations exploding with corrosive energy. Trance-inducing ambiences, visceral grunge loops, enormous modular drones and astonishing timbral contortions make this an essential resource for the producer, composer, sound designer and industrialist in search of the magic noise!

CD-Audio Selected Library **99.95**

CD-ROM Complete Library (AKAI/Emu, Roland).....**189.95**

Distorted Reality 2



The legacy continues...

An audible feast of colors created with the very latest in sonic sculpturing technology. After three years of intense development, you can be sure that fresh inspiration is in abundance. Spinning metallic textures, singing oceans, vast amounts of unique loops (50-480 BPM!), crushing impacts, glorious pads, crystalline galaxies, subsonic low-end and the many joys of feedback only begin to describe this wonderful collection. Sound Design techniques include: neural network processing, Meta synthesis, physical modeling, transform multiplication, enharmonic and cellular morphing, sonic dispersion, additive, convolution, Hyperprism, transwave cycling, phase vocoding, harmonic stretching, and Virtual Analog synthesis.

CD-Audio Selected Library **99.95**

CD-ROM Complete Library (AKAI/Emu, Roland).....**189.95**

Metamorphosis



Get ready to take it to the next level. Acclaimed producer Eric Persing's newest creation features an entire library of his trademark, cutting-edge loop manipulations. Each one of the hundreds of amazing remix grooves are truly unique in texture and feel. From floating, atmospheric Trip Hop pulsations, to experimental euro-club beats....and all the way to intense, Drum 'N Bass fury, this is an extremely versatile library for creative film composers, producers and remixers. The latest tools like Reactor, Reason and over 100 hardware and software devices were used and abused to great effect in the development of this remarkable library. The processed rhythmic inspirations range in tempos from 50 to 180 BPM and beyond!

For the first time on any library, every groove is presented three different ways: First as a standard loop, then in innovative, tempo-matched Groove Menus which offer amazing real-time groove layering at any tempo, and most importantly in Groove Control activated versions. There are literally thousands of combination possibilities right at your fingertips.

Metamorphosis brings a one-of-a-kind sonic palette to the already powerful combination of Groove Control activated products. Because of the three unique approaches presented, Metamorphosis is much more than simply a collection of inspiring loops — it's an invaluable creative tool for forging your own rhythmic innovations for years to come.

High resolution WAVE files of all the loops are included on the Mac/Windows Data Disc -both for auditioning in Soundfinder, and for direct use in software like Acid, Phrazer, Logic, Cubase, Reason or Pro Tools!

CD-ROM Multi-Disc Set Groove Control
(AKAI/Emu, Roland, GIGA) includes Groove Menus and .WAV files **189.95**

CD-Audio Loops only.....**99.95**

Vocal Planet



An epic five disc CD-ROM set produced by Eric Persing and Roby Duke, with an awesome variety of over 12,000 samples of male and female singers. These incredible one-of-a-kind sounds have all been pre-cleared for your musical use without any additional licensing fees! The main categories on Vocal Planet are:

GOSPEL: A collection of Black Gospel Choir multisamples, shouts, hums, praises and phrases, Urban Youth Choir, Claps, Snaps and Stomping samples and loops, Worship Ambiences, Holy Spirit moaning, soloists and screamin' preachin'!

R&B/DANCE: Greasy multisampled background licks available in Chromazones, Soulful R&B phrases from the top LA singers, Doo-Woppers, Hip-hop and DJ boasts, scorching Dance hooks and shouts, "street corner" acappella groups and hundreds of human percussion grooves!

JAZZ/BLUES: Incredible Jazz vocal ensembles (ala Take-6) -multi-sampled doos, dahs, bahs, accents, sfz, bends and FX- Scat singers, Bass Vocal phrases and multisamples, Sampled Jazz Vocal Chords with intelligent voice leading! Songs and stories from moaning Mississippi Delta Blues men!

VOCAL FX: Wild mouth noises, Rock 'N' Roll screamers, hambone loops, breathing, chaotic utterances, strange vowel drones, sfx, animalistic noises, thousands of vocal percussion samples, processed vocals, yells...

WORLD: Tuvan throat singers, Colombian, Peruvian, Brazilian and Mariachi yelps, Native American chants, Rastafarian toasting, haunting Indian vocalizing, ethereal phrases, ancient sacred singing, Serbian and Middle Eastern wailing, Nordic folklore, Black Sea, Balkan, Zulu chants, Yodelers, Saharan, Macedonian, Qawalli, Himalayan, Gaelic and Celtic folk songs and more.

Includes a Mac/Windows Data CD with a Soundfinder system for quickly finding the right phrase. Groove Control allows you to independently change the tempo, feel and key of more than 175 Human Percussion loops from your sequencer. A bonus Vocal Planet Rebirth mod is included too!

5 CD-ROM Set (more than 8 hours of audio material)
AKAI (compatible with Giga and software samplers), Roland **379.95**

SPECTRASONICS

World renowned composer Hans Zimmer is currently experiencing the most prolific time in his distinguished career. In 1994, Zimmer created the score for the phenomenally successful film, The Lion King, Disney's highest grossing animated film of all time which also proved to be the most successful soundtrack in the history of Walt Disney Records, selling over 12 million copies worldwide.

Zimmer is a pioneer in the use of digital samplers and synthesizers, advanced computer technology, electronic keyboards and their integration with the traditional orchestra for film and television. He began his career as a composer writing jingles and teamed up with Trevor Horn and Geoff Downes as "The Buggles" to produce the worldwide hit, "Video Killed the Radio Star" and subsequent album "The Age of Plastic." Among Zimmer's musical achievements: The Last Emperor, Nature and the Beast, Wonderland, Burning Secret, A World Apart, Rain Man, Black Rain, Driving Miss Daisy, Bird On A Wire, Days of Thunder and Pacific Heights, Green Card, Thelma and Louise, Backdraft, Radio Flyer, Thriller, K2, The Power of One, A League of Their Own, Toys and True Romance, Cool, I'll Do Anything, House of Spirits, Drop Zone and more...

Hans Zimmer Guitars Vol. 1



For the first time, Oscar & Grammy award winning film composer Hans Zimmer releases sounds from his private sample library. These powerful guitar textures are more detailed & expressive than any other Guitar library, with patches containing up to 12-way velocity switching, allowing for such nuances as dynamically-controllable string slides, pops & harmonics.

The versatility of this collection is unmatched, ranging from a wide variety of pristine acoustics, vintage Telecasters, 32 meg leads, to a rare $11,000 Paul Reed Smith collectors edition "dragon" guitar! This outstanding library also includes Guitar grooves presented in every key signature and multiple tempos.

The current version of Zimmer Guitars vol.1 adds three additional new acoustic guitars - sampled without vibrato, in multiple dynamics: a beautiful new Nylon string, 6 string Folk Steel string & the unique Parker "Fly" Electro-Acoustic guitar. Also featured are major & minor acoustic guitar strum samples. These articulations are available in steel and nylon string versions, with both up and down strokes!

Hans Zimmer's relentless pursuit of the ultimate samples for his private library has fueled his rich sonic approach on his scores for: The Lion King, The Rock, Crimson Tide, Broken Arrow, Rain Man, Driving Miss Daisy, Cool Runnings, Black Rain, True Romance, Thelma & Louise, A League of their Own, Toys & many others.

CD-ROM (Emu, Giga, Software Sampler compatible), Roland.....**284.95**

CD-Audio **99.95**

Hans Zimmer Guitars Vol. 2



A new volume of expressive instruments created for Hans Zimmer by veteran sound designer Bob Daspit. Many of the instruments feature velocity switching up and down slides, in addition to a new method for controlling vibrato with dynamics. Among the distinctive sounds are: great new pop sounds including a rich, moody Jazz Guitar - a spirited 1909 Gibson A2 Mandolin - a 1947 Lap Steel Slide guitar for that Ry Cooder vibe - the Coral Electric Sitar, originating in the sixties, and revived by Pat Metheny - the deep twang of the Electric Baritone - and finally, a brand new Steel String Acoustic guitar that will become the most indispensable sound in your collection.

Among the most exotic instruments from South America are the Cuatro, a hollowed-out single piece of wood with 4 sets of strings that create shimmering tones with great sustain - the Charango, made from the body of an armadillo with round, sparkling tones - the Cavaquinho, a high pitched guitar whose characteristic tones possess great depth - and a bold Flamenco Nylon guitar whose tones, tremolo and rasgando will ignite passionate compositions. Perhaps most rare in this collection are the string sounds of the Middle East (created for The Prince of Egypt score), including the Oud, a fretless 11-string guitar with full-bodied round tones - the Saz, a spicy six-string instrument perfect for traditional middle eastern arrangements - the Cumbus, like a 12-string with an Arabic accent - and the unusual Yayli Tambur, a long necked instrument capable of haunting drones and plucky melodies.

CD-ROM Roland, AKAI/Emu, Kurzweil & SampleCell **284.95**

CD-Audio **99.95**

Bizarre Guitar



If you're a fan of the music and textures of Daniel Lanois, Brian Eno, Robert Fripp, Bill Frisell, Edge and Adrian Belew, this is a library you will not want to miss. This unique collection from visionary Guitarist Peter Maunu and Sound Designer Eric Persing, is an ideal companion to the Distorted Reality series.

Bizarre Guitar blazes a new frontier of Sound Design via guitar. It features the extreme contrasts of breathtakingly beautiful ambience beds to searing harmonic slashes, strange funk and unsettling washes. Created with tape loop techniques, e-bow, cutting edge signal processing tricks and the Roland VG-8 guitar system, these textures are truly one-of-a-kind.

The CD-ROM versions include the Audio CD and Groove Control method. This allows you to independantly change the tempo and key of the rhythmic guitar loops, right from your existing sequencer!

CD Audio (w/o Groove Control) **99.95**

CD-ROM with Groove Control (Roland, AKAI/Emu, Kurzweil and SampleCell samplers, with sampler CD-ROM and Audio CD/Mac/PC CD-ROM).....**189.95**

SONIC REALITY

American Heartland



If you need a resource for all things acoustic, then you won't want to miss this library. American Heartland is a tour through the backroads of American culture, overflowing with a variety of earthy stringed, wind and percussion instruments.

These multi-samples were performed by experienced musicians and recorded with authentic vintage equipment to capture the true nature of grass roots music. The most complete approach to presenting these acoustic instruments.

You'll get Mandolin, Moutain Dulcimer, Dobro Slide, Pedal Steel, Banjo, Harmonicas, Pump Organ, Upright Bass, Tack Piano, Drums and more! This disc has lots of multi-sampled notes and chords plus some tasteful licks as well, to add a little cross-country flavor to your music. Perfect for that "unplugged" sound as well as for Folk, Rock, Country, AAA, Blues, commercials, soundtracks – or wherever that old country road takes you.

CD-ROM AKAI Universal (Giga, Software Sampler Compatible), AKAI, Roland, Emu EOS, SampleCell, Kurzweil, Yamaha **189.95**

Interactive Drum Kits



Following numerous requests for a straight-ahead drum disc – kits only, no loops — the producers of this set have obliged with an aggressively purist approach that expands the boundaries of conventional drum kit discs.

Interactive Drum Kits is dedicated to capturing the purity of the world's best studio instruments. Kicks, snares, toms, hats and cymbals, from the vintage Ludwig, and Gretch sets to new top of the line Pearl and Drum Workshop kits, these are extremely realistic drums that you can wack from your keyboard or with your drum pad/triggers.

It takes more than a great drum kit to make a killer CD-ROM. It takes a seasoned performer to massage the best colors out of the instrument. The producers enlisted the talents of Toss Panos (Toy Matinee) and Nick D'Virgilio (Tears For Fears, Genesis) to tune, balance and perform the kits.

One of the unique aspects of this collection is the assortment of interchangeable snare programs. They feature upwards of 20 to 50 hits per snare, from center, to edge, to rim. When you load a single snare menu program, you get all the hits and nuances laid out for you, including drops, flams and ghost notes. It's almost impossible not to get a life-like track out of these drums.

". . . my new favorite drum sample collection…a drum collection that, with a little effort by the user, can be used to create extremely convincing drum tracks… for a drum library with which to create frighteningly realistic drum tracks, you owe it to yourself to audition Interactive Drum Kits & Snares." –KEYBOARD Review, Awarded Key

CD-ROM AKAI Universal (Giga, Software Sampler Compatible), AKAI, Roland, Emu EOS, SampleCell, Kurzweil **189.95**

Concert Grand Piano



Tour the world's top recording studios and you'll notice more and more Yamaha Concert Grands showing up in the corners of these exquisite rooms. Unlike a lot of European and American concert grands, Yamaha pianos have become the choice of many studios and musicians because of their clarity and low-to-high consistency when going to tape.

Sonic Reality has successfully captured these instruments in this very well conceived and executed collection.

A Yamaha C7 recorded in-studio and a CFIIIs 9′ grand recorded on a well-known scoring stage. Pedal down notes with resonating harp included. These pianos respond beautifully to 88 note weighted controllers. Each dynamic is provided in stereo or mono – can be loaded together or individually for maximum flexibility with your setup. A slew of programming options for loud or soft music...useful variety of memory sizes ranging from 10 to 128 megs.

Concert Grand Pianos also boasts the best sustain loops of any piano CD-ROM, with enormous, long sustains that rival the real thing. These pianos play beautifully and will add richness and depth to your tracks.

CD-ROM AKAI Universal (Giga, Software Sampler Compatible), AKAI, Roland, Emu EOS, SampleCell, Kurzweil, Yamaha **189.95**

Interactive String Sections



Beautifully recorded Violins, Violas, Celli and Basses in a variety of playing styles, including legato, marcato, staccato, pizzicato and tremolo for solo, medium and large sized ensembles. Recorded with minimal room ambience, these strings were designed for use with your own external processing — allowing for tremendous flexibility.

You'll get performance oriented styles with separate or integrated arco attack elements, interactive timbre control and several types of alternate double bowing techniques to enhance the realism of your string parts. Combine solo and duo strings to create trios, quartets or quintets. Layer these elements with the full string sections for more texture and nuance. And now, it's now better than ever. Get more sections, articulations and effects variations – and even more detailed programming!

"Well-recorded samples. Ensembles of varying sizes allow building your own string orchestra in modular fashion. Thoughtful programming in the section patches, as well as in the multitimbral setups.

Samples sound great . . . recorded with care. … looping is smooth. . . sound is full without being overblown." –KEYBOARD

CD-ROM AKAI Universal, Roland, Emu EOS, SampleCell, Yamaha **284.95**
AKAI Universal format made for superior compatibility with Giga and Software Samplers

CD-ROM Deluxe Edition AKAI, Emu EOS **379.95**

SONIC REALITY

Modern Songwriter



Modern Songwriter is a complete sonic toolkit of acoustic and electric pianos, pads, strings, basses, synthesizers, brass, guitar, drums, beatboxes, percussion, and organs to inspire your next big hit! The sounds are easily accessible for fast creative performance.

Special features include virtual rhythm guitars that can be strummed in any key and any tempo with a wide variety of chords. Extra "in between" guitar performance elements like note/chord stops, fret slides and other sounds are mapped for use with all the chords when used in MIDI omni mode.

Plenty of GM mapped drum kits and percussion make it simple to swap sets within your songs. Silky pads, a huge variety of classic and synthesized electric pianos, deep basses and a wide variety of musical instruments and effects are packed onto this high performance collection.

An indispensable tool for songwriters, composers, R&B artists, producers and anyone looking for a wide variety of instruments available from just one affordable disc. Shimmering bells, vocal textures and other inspirational sounds. Produced by Dave Kerzner.

CD-ROM AKAI Universal (Giga, Emu EOS, Software Sampler Compatible).....**145.00**

Sonic Traveler



Vintage Timetraveler, American Hearltand, Modern Songwriter, World Traveler in one Super-Value package.

Introducing Sonic Reality's first box set of some of our most popular titles and our latest sounds, designed for easy use with a wide variety of samplers, including AKAI, Emu, Emagic EXS24, HALion, Kontakt, and Gigasampler. For less than the price of three titles, you get four complete Sonic Reality titles, including our highly acclaimed Vintage Timetraveler and American Heartland for a wide variety of classic organs, synths, guitars, basses, drums and other vintage and acoustic instruments.You also get Modern Songwriter which has a little of practically every category of instrument – ranging from grand pianos to strings to pads, electronic instruments and more. Plus - our latest release, World Traveler, with ethnic instruments from all over the globe!

This collection is a must for serious professional and aspiring composers, songwriters, producers and live players. With Sonic Traveler in your arsenal, you're always ready to fly! Previous customers may take advantage of a special upgrade to the full box set. Ask us for details!

Sonic Traveler Box Set AKAI Universal.....**Call**

Vintage Timetraveler



It seems that every time you turn on the radio, you hear a hit song with classic synth, organ and electric piano sounds. Maybe it's because nothing else sounds as rich and musical as these timeless keyboards!

This supersonic collection is packed with the coolest vintage keyboard sounds from the 60s, 70s and 80s. It features B3 Organs, Distorted Leslie Organs, Vox Continental, Farfisa, Optigan, Orchestron, Mellotron, Novatron, Chamberlins, Wurlitzers, Rhodes, RMI, Pianet, Clavinet, Arps, Moogs, Prophets, Jupiters, PPG, OSCar, Gleeman, Modular Synths. Plus, you get tons of far-out extras like Vibrato Fuzz Guitar Stack, Talk Box Guitar, Various Guitar FX, Ricky 4001 Bass, Fender 62 P-Bass, Hofner "Beatle" Bass — and to polish it off, Vintage Acoustic Drum Kits, Classic Drum Machines, and much more.

"How often I've checked out sampled renditions of these and other famous boards, only to be utterly disappointed! Not so with Sonic Reality's "Vintage Timetraveler"… The acoustic drums and drum machine sounds absolutely kill… remarkable…" –KEYBOARD

"... a real toolkit of vintage sounds that isn't confined to the usual classic synths.. Hammond organ samples are lovely... electric bass sounds ... solid and realistic." –SOUND ON SOUND Five Star Review

CD-ROM AKAI Universal (Giga, Software Sampler Compat.) Emu..**189.95**

World Traveler



From the far corners of the globe comes one of the most diverse collections of ethnic instrument samples ever. World Traveler consists of rare playable multisampled instruments that can be used in many different styles of music. You'll get a lot of instruments not available together on any other world instrument libraries, including sounds from the Middle East, the Mediterranean, Africa, East Asia, Australia, Native America and Northern Europe.

This library features ethnic stringed instruments like a Greek Bouzouki, Turkish Saz, Indian Sitar and Tambura, Spanish Nylon Guitars, Russian Balalaika, Mariachi Bass, Finnish Kantele, Zither, Psaltry, Hurdy-Gurdy and Hammered Dulcimers including an Indian Santoor. But, that's just scratching the surface! World Traveler also has African, Latin and Exotic percussion maps, Accordions, Harmoniums and wind instruments like, Australian Didjeridoo, Native American Wood Flute, Pan Flutes and more instruments than we can list here! World Traveler is great for adding exotic textures and ethnic flavor to western music, or for playing in authentic styles of music from around the world.

CD-ROM AKAI Univ. (compat. with Giga and software samplers).....**189.95**

MIROSLAV VITOUS
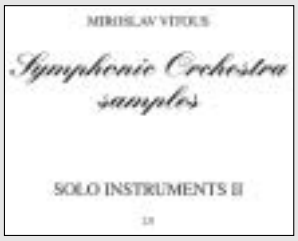
Miroslav Vitous Solo Instruments I



This disc contains Bassclarinet, Bassoon, Basstrombone, Clarinet, Contrabassoon, Trombone, Trumpet, Viola and Violoncello – all in the highest quality. Very expressive. This is the first of two Solo Instrument discs — a must for anyone wishing to emulate orchestral solo performance.

CD-ROM Roland, AKAI, Emu EOS, SampleCell, Kurzweil, Giga native.....**619.95**

Miroslav Vitous Solo Instruments II



This disc contains Alto flute, Contrabass, English horn, Flute, French horn, Oboe, Piccolo, Tuba, and Violin. This is the second Solo Instrument disc, which again contains the most expressive and musical orchestral samples on the market today.

CD-ROM Roland, AKAI, Emu EOS, SampleCell, Kurzweil, Giga native.....**619.95**

Miroslav Vitous String Ensembles

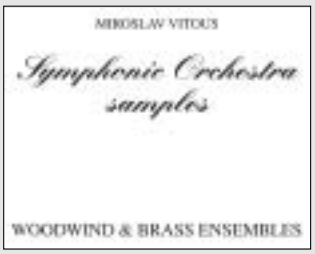
This huge undertaking contains 11 Violins, 23 Violins, 4 Violas, 4 Violoncellos 10 Violoncellos and 9 Contrabasses. Playing styles include Vibrato, No Vibrato, Mute, Detache, Pizzicato, Staccato, Sul Ponty, Sul Ponty Tremolo. Various speeds and velocities for the ultimate performance in String sampling



CD-ROM Roland, AKAI, Emu EOS, SampleCell, Kurzweil, Giga native.....**795.00**

Miroslav Vitous Woodwind and Brass Ensembles

This disc contains 3 Flutes, 3 Bassoons, 3 Clarinets, 3 Oboes, 3 Trombones, 3 Trumpets, and 4 French horns. All the ensembles you will need, to complete the Symphonic Orchestra Samples. Expressive and musical with the highest sound quality.



CD-ROM Roland, AKAI, Emu EOS, SampleCell, Kurzweil, Giga native.....**569.00**

Miroslav Vitous Acoustic Piano



Contains a stereo and mono version of a European Steinway sampled by Miroslav. Concentrates more on the classical sounding, softer touches of the instrument, with a crystal clear high end. Tuned to A-440. Two velocity layers. 23 to 50 mb versions. Easily combine for even larger piano loads. Classic sound!

CD-ROM AKAI, Emu EOS, Roland.....**145.00**

Miroslav Vitous - Classical Percussion and Harp



Classical percussion and harp with a few extras included – too many to list. This disc rounds out the Symphonic Orchestra Samples with beautifully recorded instruments that will cut through your mix with impact. Recorded and edited to achieve the most realistic classical sounds.

CD-ROM Roland, AKAI, Emu EOS, SampleCell, Kurzweil.....**429.95**

MIROSLAV VITOUS

Miroslav Vitous Classical Choirs

At last, the first release from the legendary Miroslav Vitous since his groundbreaking Symphonic Orchestra Samples library. Classical Choirs has all the realism and depth you've come to rely on from this master musician and producer.



Gorgeous multisampled male and female choirs designed for large-scale choral settings, the vocal talent is immense, hailing from the rich classical traditions of Eastern Europe. 16 Men and 16 Women were recorded separately and together as a mixed chorus at the Rudolfinum Dvorak Concert Hall in Prague, with some additional recordings captured in a smaller setting nearby. The recording quality is superb – the talents of the best engineers using fabulous mics and converters to create clear, smooth and quiet recordings.

The men were recorded in three dynamics, ff, mf and mp, singing the long sustained open vowels (Ahs, and Ohs).There are Oos, Ehs and Ees, plus pianissimo and mezzo forte Brumendos (Mms). Additionally there are Oh Expressivos and Staccato Gohs.

The women contributed dynamic renderings of Ahs, Ohs and Oos plus pianissimo and mezzo forte Brumendos, Expressivo Ohs and Staccato Gohs. The mixed choir features Ahs, Ohs, Uhs and Brumendos, in a mixture of octaves and unisons.

Another feature unique to this library is the inclusion of multi-sampled syllables, sung in both Porto (short sustained) and Staccato articulations. These syllables are offered chromatically and include do, re, mi, fa, so, la, si, mu and many more. In the Giga version, these are keyswitched to allow for fast and complex multi-syllabic passages, but in any platform this variety of utterances will provide an unprecedented reality when composing vocal performances.

Miroslav also includes a wide variety of vocal effects, like speech, murmuring, laughing and crying the quality of which you can only get from a well-trained ensemble of this caliber.

CD-ROM (Multi-Disc Set) AKAI (Emu and Software Sampler compat) ..... **Call**

CD-ROM (Multi-Disc Set) Giga native **759.00**

Miroslav Vitous Symphonic Orchestra

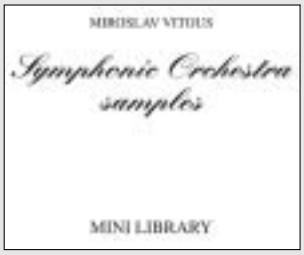


The Miroslav Vitous Symphonic Orchestra is the standard by which all other libraries are compared. It was the first to convincingly capture the nuances of the orchestra through multiple articulations and highly detailed multi-sampling. Vitous applied his famed skill as a musician with his acedemic but practical knowledge of the orchestra to create a masterpiece.

˝The sounds are the result of an intense and uncompromising process of playing and recording ... many hours of editing. Each instrument was played in a variety of ways in the studio and the symphonic hall ... to achieve the spatial positioning of a true symphonic experience. The true blending of harmonies can only come from instruments interacting with one another ... painstakingly achieved after hours of the most demanding recording sessions. I can assure the composer/producer that these orchestra and solo sets are the most realistic and expressive instruments that can be electronically produced.˝ - Miroslav Vitous

CD-ROM 5-disc set AKAI, Roland, Emu EOS, SampleCell, Kurzweil and Giga native .....**Call**

Miroslav Vitous Mini Library



A less expensive option for those who still want the highest quality in orchestral samples. Provides the very basic performances of all the instrumentsin mono, composer banks included. Composers looking for samples to use in their classical music may want to check out the Mother library. Those of you looking for more "basic" orchestral sounds will find lots to choose from in the Mini Disk.

CD-ROM AKAI, Emu EOS, Roland, Kurzweil, SampleCell, Giga, Unity DS-1 ...**335.00**

Miroslav Vitous Master Set



The Master set includes:

- Solo Instruments I
- Solo Instruments II
- String Ensembles
- Woodwind and Brass Ensemble
- Classical Percussion and Harp

CD-ROM AKAI, Emu EOS, Roland, Kurzweil, SampleCell, Giga, Unity DS-1.....**2369.95**

ILIO

Phat Fingers



Produced by Dan Portis-Cathers, this collection delivers hundreds of funkified guitar and bass patterns, slides and fills arranged into over 30 song construction kits. The grooves are incredibly tight, tempo locked and tuned to the key of the song. Most songs consists of bass patterns, slides, fills and some endings, guitar patterns, and occasional lead solos.

Tempos range from a very greasy 75 to a blistering 160 BPM. The grooves are performed by famed axe wielder Shane Roberts, a funk and fusion fanatic who plays regularly with phenoms Stan Lassiter, Reggie Wooten, drummer Chester Thompson and sax player Jeff Coffin.Imagine combining his riffs with Spectrasonics' Retro Funk or Backbeat drum libraries and Memphis Horns!

By far the most ambitious in the Fingerstyles Series, this project includes a wide range of funk and fusion styles – with influences from James Brown and Curtis Mayfield, P-Funk and Soul, to Reggae, New Jack Swing and edgy Rock/Funk Fusion a la Primus. Often hard driving, sometimes smooth and sexy, but always with a lot of phat and grease.

CD-ROM includes Audio CD)
AKAI (Giga, Emu, Software Sampler Compatible).....**189.95**

CD-Audio **99.95**

FingerStyles



The most authentic Acoustic Guitar library ever created, this approach to Acoustic Guitar sample libraries is so useful, that you'll wonder why no one had thought of it before. Produced by Dan Portis-Cathers (Hot Steel Blues, Phat Fingers, Spectrasonic's Heart of Africa), Fingerstyles consists of Guitar Patterns, Melodies, Cadences and Endings arranged in Construction Kits which can be assembled and overlaid to create complete, seamless tracks.

Music styles include New Age, Blues, Folk, Country, Pop, Ballad, Rock and others – nearly 40 compositions in all, each with as many as a dozen strums, riffs, patterns and melodies. Plus you get a variety of additional strummed chords, including extensions, for more flexibility.

Quickly create full-length, ultra-realistic guitar parts. It's the perfect solution for soundtrack composers on a tight schedule, or anyone looking to add rich sounding fingerstyles to their mix. The six string acoustic steel is expertly played and recorded, and each performance is tempo-tuned so you can sequence the patterns with complete confidence.

CD-ROM (includes audio CD)
AKAI (Giga, Emu, Software Sampler Compatible), Roland.....**189.95**

CD-Audio **99.95**

Fretworks



Featuring Legends of the Fretboard, this CD boasts an impressive roster of musicians playing an intriguing variety of stringed instruments, including Hiram Bullock on 6-string, 12-string and slide guitar, Nunzio Signoire on a vintage 6-string Gibson, Mr. Bill Simms on his National Steel, C Nausbaum on mandolin, Nat Harris on guitar and Doug McCaskill on his custom baritone guitar.

This collection is a journey "off the beaten path", capturing the soulful essence of Delta blues, Appalachian folk, and the roots of rock, bluegrass, swing and shuffle. The performances were recorded in a musical context with all of the raw energy and "happy accidents" that make a track live and breath. The phrases are tempo-tuned for easy sequencing, and most are provided in several tempos for flexibility. Divided into four or eight bar patterns and free-style riffs, they can be strewn together in countless ways to create your own unique solos. The vast musical experience represented in these fine performances will bring your tracks to that next level of quality and musicality.

CD-ROM (includes Audio CD) **189.95**
AKAI (Giga, Emu, Software Sampler Compatible), Roland

CD-Audio **99.95**

Hot Steel Blues



Following in the footsteps of the legendary Fingerstyles guitar library, Hot Steel Blues reveals a soulful Blues attitude with low-down and greasy riffs played by legendary guitarist Terry Robb (Steve Miller, Buddy Guy, John Fahey) and produced by Dan Portis-Cathers (Fingerstyles, Phat Fingers, Heart of Africa).

You'll find over 800 phrases divided into Patterns, Melodies and Endings and arranged into Construction Kits that can be assembled and mixed to create complete, seamless tracks. Both "plugged" and "unplugged" styles traverse the entire Blues genre, from Waters to Clapton.

These hi-fi riffs and licks were recorded using top-notch mics and preamps. Both acoustic and electric guitars are featured, including a Martin 000-28, a 1920's Weissenborn lap slide, a Gibson B25 upright slide, a Laguna L-29, a Fender Duo Sonic, and Fender Stratocaster. Both solid state and tube amps were used get that classic, fat and furious sound. This combined with Robb's unbelievably tasty playing makes Hot Steel Blues the most authentic, and musical, Blues guitar library to date!

CD-ROM includes Audio CD:
AKAI (Giga, Emu, Software Sampler Compatible), Roland.....**189.95**

CD-Audio **99.95**

B&H PHOTO - VIDEO - PRO AUDIO

ILIO

Skippy's Noizbox



A new drum loop library from John "Skippy" Lehmkuhl, the creator of the hugely successful Skippy's Big Bad Beats – well-known worldwide for his amazing feel, sound and the astonishing variety of loops his mind can conjure up. Noizbox has more processed elements with an aggressive Euro attitude. Solid and inventive grooves.

More than half of Skippy's Noizbox is over 120 BPM, and features a vastly creative palette of Hi and Mid-Fi sounds, skillfully woven together in Skippy's unmistakable style. Of course, it's also 100% Groove Control activated, so you can manipulate all of this spectacular sound and groove design any way you like!

Every groove is provided with several different mixes; grooves are separated into thorough Construction Kits. When you play all of the elements together, you can fully recreate the Full Mix version of the loop, with no missing parts! This is a rarity for Construction Kit libraries.

If you're not familiar with the work of this self-described loopaholic, you owe it to yourself to check out both Big Bad Beats and Skippy's Noizbox!

CD-ROM Multi-Disc Set (incl. Data CD) AKAI (Emu, Giga, Software Sampler compatible), Roland **189.95**

CD-Audio **99.95**

Skippy's Big Bad Beats



From the mind of groove guru John "Skippy" Lehmkuhl. This funkified library combines old school attitude with totally new grooves, presented in ultra hi-fi sonic fidelity to light your tracks on fire. Not just another remixed loop library, BBB gives you tons of new grooves (66BPM - 145BPM) each with multiple mix variations, including full mix, simple mix, no kick, no snare, high-end only, old record mix and more, so you can actually write a complete song with breaks and turnarounds. And with Groove Control, your options for changing tempo, feel and mix are truly limitless.

The sounds are fresh, fat and in-your-face. The grooves will hit you where it hurts. The loops are perfect for R&B, Nu-School, Hip Hop, D 'n' B, Techno, Acid Jazz, Trip Hop, Modern Break Beats or any style you can dream up when using them with Groove Control. It also features multi-sample menus made with groove slices.

If you're looking for some very 'now'-sounding drum loops and need maximum flexibility combined with easy access, Big Bad Beats will not only deliver but might even represent the future of the sample CD.

CD-ROM (Includes Audio/Data CD) AKAI (Emu, Giga, Software Sampler compatible), Kurzweil, Roland.....**189.95**

CD-Audio (no Groove Control)..... **99.95**

Stark Raving Beats



Here's a wicked set of hard and fast grooves, based on live drum kit performances, mixed and remixed with wildly processed electronics and banging percussion. 100% Groove Control activated for endless tempo, feel and mix possibilities, SRB will bring out the monster in you!

All Go, No Slow! SRB starts at 105 BPM and accelerates to 170. Speed it up further with Groove Control! Every groove has an aggressive style that can burn rubber at any tempo. Great for Progressive, Jungle and D 'n B styles, but use your imagination to dissolve, distort and destroy these gems any way you like.

Hundreds of loops to mix and match millions of ways. Its extreme separation makes it a natural to combine with Skippy's Noizbox, Big Bad Beats, Ethno Techno, or any of the Groove Control activated libraries from Spectrasonics!

Performed and remixed by loopologist Chris O'Brien. SRB fulfills your need for speed and passion for aggression!

CD-ROM Multi-Disc Set (includes Data CD) AKAI (Emu, Giga, Software Sampler compatible), Roland.....**189.95**

CD-Audio **99.95**

Ethno Techno



Produced for ILIO by Eric Persing, this is Bashiri's first sample project since his ground-breaking Supreme Beats percussion library. Features hundreds of astonishing loops from the mind of one of the world's masters of timing and feel.

One of the great features of this collection is its use of custom invented, hand-made instruments with interesting names like Kissing Fish, Double Drink, Tributaries, Galaxy, and Satellite Flashdance. Other off-the-wall sounds include Computer Cans, Styrofoam w/Spring, Plastic Pail, Pipes, Toy Cow, Metal Sheet, and Garbage Can. Combined with ethnic instruments like Bamboo Slits, Mini Dumbek, Udu, Cora, Talking Drum, Nigerian Bata, Surdo, Shakers, Tam Tam, Angklung, Sabar, Waterphone, Mini Djembe, Dun Dun, and Serimbau, this disc is a glorious journey through heart-pounding terrain.

Tempos range from to 70 to 166 BPM. ou get a range of sultry rhythms with origins from all corners of the world, but evolved to a place that is definitely extra-terrestrial.

CD-ROM Multi-Disc Set (includes Data CD) AKAI (Emu, Giga, Software Sampler compatible), Kurzweil, Roland **285.00**

CD Audio 2-Disc Set (no Groove Control) **124.95**

ILIO

Keys To The City



Look no more for an intelligent way to get the hottest studio production elements into your tracks. Keys To The City was created and produced by one of the fastest rising stars of the keyboard, Michael Bearden. This library delivers tons of synth and keyboard grooves organized into multi-tracked compositions. The grooves are provided as complete mixes and individual elements for astounding flexibility – perfect for Dance, Hip Hop, R&B, Pop, Funk and Electronica. The sound is first class. The Bass-ment grooves are all individual bass lines that can be used alone or to accompany Studio loops - which are single layer comps, Duplex - which are dual layer grooves, Triplex, three layer grooves, or Penthouse - which contain four or more layered parts.

Michael's enormous talent has catapulted him to the top of the industry, recording and performing with the likes of Jennifer Lopez, Destiny's Child, Madonna, Michael Jackson, Brandy, Whitney Houston, Lenny Kravitz, Stevie Wonder, Christina Aguilera, Ricky Martin, Babyface, and so many more. He's also frequently called by Paul Shaffer to substitute for him on the Late Show with David Letterman.

CD-ROM Multi-Disc Set (includes Data CDs) AKAI (incl. Acid files Emu, Giga, Software Sampler compatible), Roland/Acid.....**189.95**

CD-Audio.....**124.95**

Digital Boy: Modern Synths and Dance



Jam-packed with over a gigabyte of powerful, evocative sounds, this 2-disc CD ROM set lets you focus on what's important – your music. Modern Synths and Dance is extremely well programmed, for expressive, spine-tingling performances.

Swirling, evolving atmospheric synths and lush, dreamy pads; huge, intense analog/digital hybrid stacks and haunting synth voices; snappy, high-energy reso-synths and dramatic filter sweeps; funky, "old school" analog synths and booty-shakin' basses; "in-yo-face" hits, stabs, and industrial FX; a vast assortment of superb pop synths including bells, brass, clavs, organs, electric pianos, and lead synths; essential acoustic sounds such as guitars, strings, brass, and more; and the most extensive collection of hard-hitting contemporary drum and percussion sounds ever assembled.

Almost all the sounds are extensively multisampled (in stereo of course) making them consistent and playable across the full range of the keyboard. Each sound's envelopes, filters, LFO's, and controllers are expertly programmed for maximum responsiveness and playability.

CD-ROM (Multi-Disc Set) AKAI (Giga, Emu, Software Sampler Compatible), Roland **285.00**

TranceFusion



Fast becoming a cult classic among remixers, electronica gurus, composers and producers, TranceFusion is an indispensable collection of sonic mania for adding drama and intensity to any music.

Trancefusion is hundreds of noise sweeps, spectral climbs, glides, fills and morphs. Other tricks include percussion sweeps, swell-stabs, vocal rantings and big hits to spice up your transitions, plus rave and trance style arpeggios and pads.

The whole purpose of this disc is to energize your tracks. By layering one of these massive glides or climbers at the end of an 8-bar loop, for instance, you can trigger that crucial, thrilling emotional reaction in your audience as they anticipate another slammin' 8-bar beat. Many of the sweeps start softly, climax and fade slowly. Others have hard starts, ramping up to full intensity and stopping cold. Many are pitched glides (a la THX), while others are noise-based or searing reso-sweeps. All of them are thick with movement and intensity.

Build screaming transitions from one groove to the next, to carry the groove into a break or to add some unexpected madness to the mix.

CD-ROM includes Audio CD: AKAI (Giga, Emu, Software Sampler Compatible), Roland **189.95**

CD Audio **99.95**

Analog Meltdown



Here's a new twist on your average synth library. Designed, engineered, blown up and re-animated by mad programmer Kevin Wakefield, Analog Meltdown has the brains of one synth and the body of another. Wakefield adds an array of mod wheels, faders, breath controllers and pedals to each program, making this one of the most playable synth libraries ever.

The sounds are wild and freakish, clamorous and whimsical. They have a unique personality, unpretentious and in-your-face. Analog Meltdown is truly the "Edward Scissorhands" of sample libraries.

A wide assortment of textures – including stunning leads, power pads, FM (no really, it's FM!), plucky sounds, sweeps, bubbly effects, stellar grand prix, electro-shock treatments, sci-fi weapons and transporters, stingers and resonators, and more. Active sounds, designed to be featured and loud.

Caution: When handling the contents of this disc, be prepared to alter the course of your creativity due to the unpredictable effect of these sounds on your psyche.

CD-ROM AKAI (Giga, Emu, Software Sampler Compatible), Kurzweil ..**189.95**

ILIO

Memphis Horns



Wayne Jackson and Andrew Love have dominated the Rock, Soul and Funk scene since the 60's and to this day offer up their multi-tracked sound on tons of Billboard hits. Their list of Gold and Platinum records is impressive: Respect by Aretha Franklin, Dock of the Bay by Otis Redding, Sledgehammer by Peter Gabriel, and hundreds more. Amazing Trumpet, Trombone, Bari and Tenor sound!

Delivers hundreds of phrases in multiple keys and tempos, the phrases are presented both in their original complete form and divided into flexible short phrases, so you can easily integrate them any way you like. Styles range from Slow Soul and Passionate Pop, to Furious Funk.

Includes phrases ranging in tempo from 70 to 140 BPM, each in five harmonically related keys. You also get chromatically multisampled unisons, chords in major, minor, major 7, minor 7 and chord swells. Additionally, there's a great variety of unique brass effects like flutters, Harmon mutes, tone clusters, falls and rips. Andrew also presents some of his famous solo sax work which you won't want to miss!

CD-ROM Multi-Disc Set (includes Audio/Data CDs AKAI (Giga, Emu, Software Sampler Compatible).....**285.00**

CD-Audio/Data CD 2-Disc Set (includes WAV files) **145.00**

Dirk Campbell's World Winds



Acclaimed UK composer and multi-instrumentalist Dirk Campbell brings you the evocative sounds of many rarities like the Bansuri (Indian flute), Turkish and Persian Ney flutes, Kaval (Southeast European flute), African flute, Suling Slendro and Suling Pelog (Javanese flutes), Irish Penny Whistle, Irish Low Whistle, Fujara (Slovakian Contrabass Whistle), Japanese Shakuhachi, Andean Panpipes, Armenian Duduk, Indian Shenai, Egyptian Zummara, Turkish Zourna, Gaida Bagpipes (Greek and Bulgarian, three types), Irish Uilleann pipes (three types), Scots Highland pipes, Medieval Praetorius Bagpipes, Turkish Tulum, Arabian Mezoued, Cretan Askomandura, Czech Dudy, Nafir (Moroccan trumpet) and African Kudu Horn.

Sweet, delicate to eerie, breathy and complex.... evoking the beauty, stillness, depth and mystery of the natural world. In contrast, there are the wild, strange and stirring timbres of ethnic reed instruments, bagpipes and African horns.

For film, TV and media composers, World Music, Rock, Pop and contemporary Classical composers and performers, creative keyboardists and programmers, and anyone with an ear for great instrumental sounds!

CD-ROM AKAI (Giga, Emu, Software Sampler Compatible).....**189.95**

Double Platinum Drums



An unabashed approach to West coast rock drums. Recorded at L.A.'s renowned Sound City, the drums are flawlessly recorded and mixed with both dry and room ambiences, evenly divided between multiple discs.

A jam-packed multi-disc set with over 500 loops in 40 tempo/style combinations (66 BPM to 203 BPM) with Intros, Time (main body), Fills and Endings. Styles include Rock, R 'n B, Pop, Hard Rock, Slow Rock, Metal, Surf, Funk, 3/4 Ballad, Country, Latin, New Orleans... Full fledged multi-velocity Kits include Drum Workshop, Tama, and Remo, while CD-ROM owners also get an extra Gretch kit. Eight different snares to mix and match with, plus a vast collection of brilliant Zildjian cymbals.

Producer John Boylan, Exec Producer Tony Harris, and Engineer Paul Grupp pulled out all the stops (and the mics) to create a clean, impactive, and completely professional sound. Double Platinum Drums is a thrilling rock 'n roll joyride.

CD-ROM AKAI (Giga, Emu, Software Sampler Compat.), Roland, SampleCell, Kurzweil.....**285.00**

CD-Audio.....**125.00**

Double Platinum Rock Piano



Finally, a piano that doesn't require tons of EQ, compression and effects to be heard. This is the industry's one and only true Rock piano, with all of the intrinsically natural brightness and depth you need to speak through your tracks. You've never heard a piano with better blending and stereo imaging.

The piano is a rare and famous vintage Steinway B, custom rebuilt and maintained strictly for studio recording. Its contribution to Rock history is etched in stone, having asserted its presence on multi-platinum records from Elton John and Tom Petty to Carl Perkins and Johnny Cash.

Chromatically sampled (all 88 keys), three velocities, looped and non-looped – an endless variety of patch combinations to choose from.

The samples were recorded straight to digital on a vintage Neve console in Sound City's studio A. Paul Grupp engineered the sessions using his masterful ears and years of experience to get phase-perfect stereo mic placement. Produced by John Boylan and Tony Harris as the best two-fisted piano sound in the world — captured on one easy-to-use CD-ROM.

CD-ROM AKAI (Giga, Emu, Software Sampler Compat.), Roland, SampleCell, Kurzweil **145.00**

GENERAL SOUND EFFECTS

Series 6000 - "The General" World's Largest Digital Sound Effects Library



Recognized all over the world as the most comprehensive sound effects library of its kind, Series 6000 represents the very best in general sound effects libraries. This staggering 40 CD collection of more than 7,500 sound effects (more than 50 hours of recorded sound), reigns supreme as the industry leader in professional audio - you simply cannot get a broader selection or more detailed coverage of the must-have categories required for daily production. Simply put, The General is the standard against which all other sound effects products are measured.

- The General is Sound Ideas' signature library. It offers a broad spectrum of sound effects - including Animals & Birds, Construction, Crowds, Fire, Household, Industry, Military, Sports, Transportation, Office, Weather, and much, much more. The collection also includes long ambience tracks from around the world, Foley sounds, and some truly spectacular audio from four award-winning sound designers.
- Winner of the prestigious Game Developer Hall of Fame Award, The General Series 6000 is the sound effects library that you will return to .... again and again. When you need the ultimate in general sound effects libraries, you need The General.

6001: Aviation #1	6014: Crowds #2	6028: Office
6002: Aviation #2	6015: Fire	6029: Outdoor Sounds
6003: Animals & Birds	6016: Foley: Footsteps #1	6030: Restaurants & Stores
6004: Auto Races & Crashes	6017: Foley: Footsteps #2	6031: Sports
6005: Auto, Ford Escort	6018: Misc: D - H	6032: Misc: H - S
6006: Auto, Olds Cutlass	6019: Helicopters	6033: Transportation
6007: Auto, Lincoln	6020: Household #1	6034: Traffic
6008: Auto, Honda Accord	6021: Household #2	6035: Water
	6022: Industry	6036: Weather: Randy Thom
6009: Auto, Grand Am	6023: Industry & Jungle	6037: Misc: S - Z
6010: Misc: A - C	6024: Marine	6038: Misc: Frank Serafine
6011: Construction #1	6025: Military	6039: Misc: Mike McDonough
6012: Construction #2	6026: Motorcycles	
6013: Crowds #1	6027: Music & Percussion	6040: Misc: Alan Howarth

Series 6000 "The General" (40 Audio CDs) **1275.00**

Founded in 1978 and now located just outside of Toronto, Canada, **Sound Ideas** is the world's largest publisher of sound effects for the professional broadcast, post-production and motion picture industries. We also publish the Westar Music Library, the Sound Ideas Music Library, and The Mix Broadcast Music library, and offer a growing number of products for use in multimedia, game and Web design. Our sound effects and music libraries are used all over the world. Sound Ideas is the exclusive worldwide distributor for a number of custom-created feature film studio sound effects libraries, including: 20th Century Fox, Lucasfilm, Universal Studios, Turner Entertainment, Disney Ideas, Hanna-Barbera, Warner Bros., and Jay Ward Productions – the creators of Rocky & Bullwinkle.

GENERAL SOUND EFFECTS

Series 6000 Extension I



The perfect partner to the incredible Series 6000 Sound Effects Library (and a superb collection in its own right). A stunning collection of 20 CDs, showcasing over 3,200 professional quality sound effects – the result of substantial research, field recording and long studio hours with the efforts of some of the industry's most respected sound recordists and sound designers, this remarkable library is a compilation of today's most in-demand sound effects.

6041: Airplanes: Biplane, Crash, Float, Single Prop, Twin Prop, Four Engine, Squadron, Jet

6042: Airplanes: Jet, Hydraulics, Skid, Airport Ambiences, Autos: 1923 Dodge Paddy Wagon, 1927 Ford Model A, 1928 Lincoln Phantom, Volkswagen

6043: Autos: 1970 Chevy SS, Pickup Truck, Humvee, Auto Crashes, Auto Sounds, Auto Races, Auto Skids

6044: Adding Machines, Air, Alarms, Animals, Antacids, Aquarium Sounds, Archery, Auctions, Axes, Bank Machines, Bars, Barbeques, Basketball, Beads, Bells, Bicycles, Bingo, Birds, Blacksmiths, Blinds

6045: Boats, Bodyfalls, Book and Box Drops, Briefcases, Bubbles, Buckets, Bug Sprayers, Bullfights, Buzzes, Cabinets, Cameras, Canteens, Carpet Sweepers, Carts, Cartoon Sounds, Cash Registers, Casino Sounds, Caves, Chains, Chainsaws, Chairs, Chalkboards, Champagne Bottles, Chandeliers, Cheques, Chests, Chimes, Church Sounds

6046: City and Country Sounds, Clocks, Clothes, Cocktails, Coffee, Compressors, Coins, Construction, Coolers, Cookie Jars, Crashes, Creaks, Credit Cards

6047: Crowds

6048: Digging Sounds, Dirt Falls, Doors, Drawbridges, Drops, Earthquakes, Electricity, Elevators, Escalators, Exercise Equipment, Explosions

6049: Fans, Farm Harvesters, Fences, Fire, Fireworks, Flies, Footsteps, Forest Sounds

6050: Guns

6051: Forklifts, Freezers, Fuses, Garbage Cans, Garbage Trucks, Garden Trimmers, Gas Cans, Gas Stations, Generators, Glass Sounds, Golf, Gongs, Gramophones, Grinders, Gum Ball Machines, Harbour Sounds, Helicopters, Hits, Hockey, Horns, Garden Hoses

6052: Hospital Sounds, Hot Tubs, Hub Turners, Human Sounds, Hydraulic Lifts, Ice, Impact Wrenches, Industrial Sounds

6053: Industrial Sounds

6054: Industrial Sounds, Ironing, Jet Packs, Jewellery Boxes, Jungle Sounds, Knives, Ladders, Lathes, Lawn Mowers, Light Bulbs, Loading Docks, Machines, Mailboxes, Make Up Bags, Marbles, Matches, Medical Bags, Medicine Cabinets, Message Tubes, Metal Detectors, Metal Sounds, Microfilm Machines, Missiles, Motors

6055: Motorcycles, Music Boxes, Newspaper Boxes, Office Ambiences, Oil Cans, Paint Shakers, Paper Shredders, Parachutes, Picnic Baskets, Polishers, Popcorn, Pottery Breaks, Pouring Sounds, Projectors, Pulleys, Pumps, Punch Clocks, Purses, Radiators, Radios, Raffle Sounds, Rain, Rain Sticks

6056: Restaurant Sounds, Robotics, Rock Climbing, Rock Sounds, Roller Coasters, Sci Fi Sounds, Scrapes, Sewing Boxes, Shakes, Sharks, Shaving, Ships, Shipyards, Shovels, Showers, Sirens, Ice Skating Sounds

6057: Ice Skating Sounds, Ski Lifts, Slides, Snow Plows, Snowboards, Soldiers, Spins, Sprinklers, Squeaks, Steam, Stereos, Stopwatches, Stoves, Street Sweepers, Suction, Suits of Armour, Suitcases, Swamp Sounds, Sweeping Sounds, Swimming, Swishes

6058: Switches, Tanks, Tarps, Telemetry, Telephones, Teletypes, Televisions, Telex Machines, Thermos Bottles, Thunder, Ticket Dispensers, Toasters, Toilets, Tool Belts, Toys, Trains

6059: Trains

6060: Tree Mulchers, Twirls, Umbrellas, Vacuum Cleaners, Velcro, Vending Machines, Wagons, Walkers, Walkie Talkies, Washing Machines, Water Sounds, Weather Charts, Whistles, Winches, Winds, Windmills, Wire Fasteners, Wood Sounds, Wool Winders, Wrecking Balls, Wrestling, Zippers

Series 6000 Extension I (20 Audio CDs) **679.95**

Series 6000 Extension II



Ten CDs of more than a 1000 professional, pristine, in demand audio to compliment the 6000 series of libraries. A full range of popular and practical items and ambiences, plus a comprehensive set of snowmobiles and contemporary telephone sounds - cellular, cordless, electronic and pagers.

6061: Air Conditioners, Air Releases, Aircraft Carriers, Airplanes, Alarms, Autos, Axes, Balloons, Barns

6062: Bees, Bells, Boats, Bone Breaks, Can Punctures, City Ambiences, Clocks, Computers, Crashes, Crowds

6063: Crowds, Crunches, Doors, Drops, Electricity, Elevators, Explosions, Film Clappers, Fire, Fireworks, Flame Throwers, Forest Ambiences

6064: Glass Sounds, Golf Carts, Guns, Ricochets, Shot Guns & Silencers, Helicopters, Horses, Hospital Monitors, Human Hits, Ice Picks, Industrial Sounds

6065: Industrial Sounds, Lawn Tractors, Marsh Ambiences, Metal Sounds, Motors, Motorcycles, Music

6066: Parades, Plastic Sounds, Plumbing, Police Scooters, Projectors, Radar, Radios, Rain, Ratchets, Refrigerators, Rips, Rock Sounds, Rockets, Roller Coasters, Room Tones, Scrapes, Shovels, Sirens, Snowballs

6067: A full CD of Snowmobile Sounds - Singles, Groups, Polaris, Vintage & Yamaha

6068: Squishes, Steam, Strobe Flashes, Subways, Suction, Swishes, Telemetry, Cellular Telephones, Novelty Phones, Cordless Telephones, Electronic Telephone Rings & Internal Signals, Teleconference Phones, Pagers

6069: Toboggans, Trains, Trucks, Wagons, Water Sounds

6070: Water Sounds, Weather, Werewolves, Western Ambiences, Whooshes, Wind, Wood Sounds

Series 6000 Extension II (10 Audio CDs) **425.00**

GENERAL SOUND EFFECTS

Series 6000 Extension III

More than 1800 outstanding sound effects ready for action, the Series 6000 Extension III offers an extensive variety of contemporary sounds - full CDs of all terrain vehicles, automobiles and guns, plus spectacular selections of weapons, battles, horror, crashes, smashes, crowds, doors, explosions, hits, vehicles, swishes, whooshes ... and a whole lot more.



6071: All Terrain Vehicles

6072: Auto Plant Ambiences, Auto Horn Dopplers, Auto Races, Passenger Van Sounds

6073: Air, Airplanes, Airport Carts, Animals, Anvils, Applause, Archery, Axes, Bamboo Staffs, Barbeques, Basketball, Battles

6074: Bayonets, Bells, Blinds, Bodyfalls, Boomerangs, Bottles, Boxes, Boxing, Brakes, Bone Breaks, Cameras, Cans, Chains, City Sounds, Command Center Sounds, Compressors, Computer Sounds, Construction, Cowboys, Crashes, Creaks, Crowds

6075: Crowds, Dentist Sounds, Doors, Drops, Electricity, Electronics, Elevators, Exercise Bikes, Explosions

6076: Fans, Faxes, Fire, Fireworks, Flame Throwers, Flares, Food Sounds, Footsteps, Forest Sounds, France Ambiences, Furnaces, Garbage Trucks, Generators, Glass Sounds, Guillotines, Guns

6077: Gun Foley, Gun Holsters, Hits, Horror Sounds, Horse Sounds, Hospital Sounds, Human Hits, Ice Makers, Impact Wrenches, India Ambiences

6078:Industrial Sounds

6079: Italy Ambiences, Knife Sounds, Laboratories, Leather Sounds, Marsh Sounds, Mechanical Latches, Megaphones, Metal Sounds, Missiles, Mortars, Motors, Motorcycles, Music, Paper Sounds, Photocopiers, Pinball Machines, Postage Meters, Power Washers, Prison Guards, Projectors, Paper Punches, Radios, Rain, Refrigerators, Rockets, Roller Blades, Rumbles, Scales, Sci Fi Sounds, Scrapes

6080: Servos, Sewing Machines, Ski Sounds, Slides, Slingshots, Spitballs, Squeaks, Submarines, Swishes, Switches, Swords, Tanks, Tape Recorders, Tarps, Tear Gas, Telemetry, Thailand Ambiences, Throwing Stars, Thunder, Torches, Toys, Garden Trimmers, Trucks, Vacuums, Vending Machines, Walkie Talkies, Water Sounds, Wheels, Whooshes, Winches, Winds, Wood Sounds

Series 6000 Extension III (10 Audio CDs) **425.00**

Series 6000 Extension IV

Another great addition to "The General", Extension IV offers 10 more CDs and over 1,000 sound effects – everything from the practical to the unusual. Includes a full Hum Vee package, a small fleet of airplanes, heavy construction and road equipment, sounds from a variety of industrial environments, solid rock and wood FX, water, rain and liquids, and a host of other spectacular audio.



6081: Air, Airplanes, Alarms, Creatures, Arcade Video Games, Auto Races

6082: Auto Races, Auto Tire Spins, Axe Chops, Battle Ambiences, City Ambiences, Cartoon Voices, Construction Ambiences, Construction Bulldozers, Crashes, Creaks, Crowds

6083: Crowds, Doors, Earthquakes, Electricity, Elevators, Engine Rooms, Explosions, Fires, Fireworks, Forest Ambiences

6084: Furnaces, Glass Sounds, Bullet Impacts & Ricochets, Hits, Hockey Ambiences, Horror & Foley, Heart Monitors, Human Grunts, Ice Sounds, Industrial Ambiences

6085: Idustrial Machines, Marsh Ambiences, Metal Detectors, Marching Bands, Radio Tuning, Rain, Residential & Rest Stop Ambiences, Rock Falls, Room Tones

6086: Sci Fi Alarms, Ambiences, Drones and other Sounds, Stone Scrapes, Shovels, Slides, Snowblowers, Snow Plows

6087: Snow Plows, Telemetry, Telephones, Timers, Traffic Ambiences, Steam Trains

6088: Steam Trains, Tree Falls, Truck Signals, Walkie Talkies, Water Sounds, Wind Ambiences, Wood Sounds

6089: Helicopter Sounds: Astar AS350D, Chinook, Hughes 300, Labrador Search & Rescue, Sikorsky

6090: Hummer H1 SUV: Full Package

Series 6000 Extension IV (10 Audio CDs) **425.00**

GENERAL SOUND EFFECTS

XV Series



XV is a gallery of over 3,000 sound effects covering 60 major categories, drawn from many of Sound Ideas' most popular libraries. XV's unique multi-platform presentation, provides each of the 15 volumes in red book CD audio format, as well as CD-ROM format containing .wav and mp3 files, ready for immediate use in both Mac and Windows systems.

XV-01: Air & Air Valves, Airplanes, Airport Ambiences, Animals

XV-02: Animals, Arrows, Automobiles, Axes, Balloons, Bells, Bicycles, Birds, Boats

XV-03: Boats, Bodyfalls & Bone Breaks, Boxes, Buses, Cameras, Canoes, Cards, Cartoon Sounds, Cases, Cash Registers & Casino Sounds

XV-04: Chainsaws, City Ambiences, Clocks, Coins, Construction, Plastic & Styrofoam Containers, Country Ambiences, Ceiling & Wood Cracks, Crashes, Wooden Crates

XV-05: Crowd Sounds, Debris, Dentist Sounds, Deposit Boxes, Doors

XV-06: Doors, Drops, Earthquakes, Electrical Panels & Electricity, Elevators, Exercise Equipment, Explosions, Fair Ambiences, Farm Tractors, Fire Sounds & Equipment, Fireworks, Flags, Flame Throwers, Footsteps

XV-07: Footsteps, Forest Ambiences, Burning Fuses, Gas Station Sounds, Gavels, Geiger Counters, Glass Sounds, Golf Carts, Gun Sounds, Handcuffs, Heartbeats

XV-08: Helicopters, Horror & Vampires, Hospital Sounds, Household Sounds

XV-09: Household Sounds, Human Sounds, Ice, Impacts, Industry Sounds, Insects

XV-10: Jet Packs, Jet Skis, Jukeboxes, Jungle Ambiences, Kayaks, Ladders, Lighters, Lipstick Tubes, Locks, Manhole Covers, Marsh Ambiences, Matches, Microphones, Military Sounds, Morse Code, Motors

XV-11: Motorcycles, Mountain Ambiences, Music, Newspaper Boxes, Office Sounds

XV-12: Office Sounds, Parades, Pinball, Police Station Sounds, Popcorn Makers, Film & Slide Projectors, Radar Sweeps, Radio Sounds, Restaurant Ambiences, Robot Motors, Rock Falls, Roller Coasters, Room Tones, Safes & Vaults, School Ambiences, Sci Fi Sounds, Sculls, Servo Movements

XV-13: Sirens, Snowmobiles, Sports, Stereo Equipment, Stock Exchange Ambiences, Street Sweepers, Suits of Armour, Swimming, Swishes, Swords & Sword Fights, Tape Cueing, Telemetry, Telephones

XV-14: Teletype Machines, Toboggans, Tombstones, Toys, Traffic Ambiences, Trains, Tree Falls, Trucks, Turnstiles

XV-15: Velcro, Vending Machines, Video Cassette Recorders, Volcanos, Wagons, Walkie Talkies, Water Sounds, Weather, Wheelchairs, Whips, Whistles, Whooshes, Zippers

XV Series (15- Audio CDs & 15-CD ROMs) **674.95**

XV MP3 Series



The XV MP3 Series - two CD ROM discs and the complete contents of the XV Series, available in mp3 files that have been encoded at a 128-bit rate, ready for immediate use in a variety of different audio and multi-media software applications, and fully compatible with both Mac and Windows.

Sound Effect Categories on the XV-MP3-01 disc:

Air & Air Valves, Airplanes, Airport Ambiences, Animals, Arrows, Automobiles, Axes

Balloons, Bells, Bicycles, Birds, Boats, Bodyfalls & Bone Breaks, Boxes, Buses

Cameras, Canoes, Cards, Cartoon Sounds, Cases, Cash Registers & Casino Sounds, Chainsaws, City Ambiences, Clocks, Coins, Construction, Plastic & Styrofoam Containers, Country Ambiences, Ceiling & Wood Cracks, Crashes, Wooden Crates, Crowd Sounds

Debris, Dentist Sounds, Deposit Boxes, Doors, Drops

Earthquakes, Electrical Panels & Electricity, Elevators, Exercise Equipment, Explosions

Fair Ambiences, Farm Tractors, Fire Sounds & Equipment, Fireworks, Flags, Flame Throwers, Footsteps, Forest Ambiences, Burning Fuses

Gas Station Sounds, Gavels, Geiger Counters, Glass Sounds, Golf Carts, Gun Sounds

Handcuffs, Heartbeats, Helicopters, Horror & Vampires, Hospital Sounds, Household Sounds, Human Sounds

Sound Effect Categories on the XV-MP3-02 disc:

Ice, Impacts, Industry Sounds, Insects

Jet Packs, Jet Skis, Jukeboxes, Jungle Ambiences

Kayaks

Ladders, Lighters, Lipstick Tubes, Locks

Manhole Covers, Marsh Ambiences, Matches, Microphones

Military Sounds, Morse Code, Motors, Motorcycles, Mountain Ambiences, Music

Newspaper Boxes

Office Sounds

Parades, Pinball, Police Station Sounds, Popcorn Makers, Film & Slide Projectors

Radar Sweeps, Radio Sounds, Restaurant Ambiences, Robot Motors, Rock Falls, Roller Coasters, Room Tones

Safes & Vaults, School Ambiences, Sci Fi Sounds, Sculls, Servo Movements, Sirens, Snowmobiles, Sports, Stereo Equipment, Stock Exchange Ambiences, Street Sweepers, Suits of Armour, Swimming, Swishes, Swords & Sword Fights

Tape Cueing, Telemetry, Telephones, Teletype Machines, Toboggans, Tombstones, Toys, Traffic Ambiences, Trains, Tree Falls, Trucks, Turnstiles

Velcro, Vending Machines, Video Cassette Recorders, Volcanos

Wagons, Walkie Talkies, Water Sounds, Weather, Wheelchairs, Whips, Whistles, Whooshes

Zippers

XV MP3 Series (2 Audio CDs) **209.95**

SOUND IDEAS

GENERAL SOUND EFFECTS

Series 2000



The Series 2000 Library was the world's first collection of stereo effects to be recorded and mastered digitally. The sound quality is impressive and has set the standard for digital sound effects production worldwide. This collection of 22 CDs is invaluable on its own, or for added power in the studio. Works in perfect harmony with other Sound Ideas sound libraries.

2001: European FX: Ambiences, Crowds, Automobiles

2002: Fire Engine & Police Sirens, Hand Guns, Police Station Ambiences, Rain, Swimming Pools, Traffic

2003: Traffic, European Telephones, Horses, Trains, Restaurants, Fairs, Crowds

2004: Airplanes, Animals, Automobiles

2005: Applause, Crowds, Laughter

2006: Automobiles

2007: Residential Ambiences, Door Bells, Broadcast News, Sports & Weather, Birds, Buses, Cameras, Automobile Skids, Squeals & Crashes, Demolition Sounds

2008: Children, Clocks

2009: Chainsaws, Construction

2010: Dentist Sounds, Doors, Explosions, Fire & Fireworks, Guns, Arcades, Fairs & Roller Coasters

2011: Footsteps - Stone, Concrete, Pavement, Dirt, Grass, Hardwood, Leaves, Linoleum, Metal, Sand

2012: Footsteps - Sand, Snow, Water, Wood, Metal & Garbage, Farm & Tractor Sounds, Babies, Punches

2013: Helicopters, Hospitals, Squeaks, Sirens, Alarms

2014: Household & Industrial Sounds

2015: Motorcycles, Dune Buggies, Lawn Trimmers & Mowers

2016: Boats, Marine Sounds, Newsroom & Office Sounds

2017: Musical & Percussion Instruments, Whistles, Noisemakers

2018: Rain, Thunder, Bar & Restaurant Ambiences, Soda Fountains

2019: Space Sounds, Shopping Malls, Stores

2020: Sports, Military Tanks, Cannons & Battles

2021: Traffic, Trucks

2022: Water, Zippers

Series 2000 (22 Audio CDs) **589.95**

SFX Kit



This massive collection — 20,000 sound files in .wav file format— is especially designed for the game and interactive entertainment industries. Produced by Tommy Tallarico, one of the most successful and accomplished video game composers in history, this essential and wide-ranging library contains 7 CDs of .wav files in 110 different categories, plus a searchable data disc - all configured for easy use with Mac and Windows systems.

TSK-01 (1561 FX / 424 MB): Airplanes, Automobiles, Doors, Machinery, Misc. Transportation, Steam, Trucks

TSK-02 (1725 FX / 512 MB): Alarms & Beeps, Ambiences, Foley, Menu FX, Pickups

TSK-03 (9,073 FX / 509 MB): Bodyfalls, Footsteps, Human Vocals

TSK-04 (2,606 FX / 414 MB): Monsters & Creatures, Mystical & Magic FX

TSK-05 (723 FX / 166 MB): Energy & Forcefields, Misc Sci-Fi, Sci-Fi Loops, Sci-Fi Weapons

TSK-06 Sports and Recreation (2483 FX / 687 MB): Air Hockey, Baseball, Basketball, Billiards, BMX, Boxing, Crowds, Crowds - College, Fishing, Foosball, Football, Hockey, Hockey Horns & Buzzers, Hockey Organ Music, Hockey Vocal, Sounds, Misc Sports, Pinball, Skateboarding, Street Basketball

TSK-07 Weapons & Destruction (1484 FX / 206 MB):Bone FX, Crashes & Smashes, Creaks, Scrapes, & Misc, Destruction, Hits & Impacts, Splats & Squishes, Weapons, Weapons Foley

TSK-DATA: The Data Disc

The SFX Kit Data Disc contains data files in the following four formats: MS Excel, comma separated text, tab delimited text, and a fully functioning version of FileMaker Pro version 5 that offers a searchable interface. Simply use the MS Excel or text files to import the collection's data into your own search software, or fire up the FileMaker Pro version to perform searches immediately. The Data Disc is compatible with Mac and PC. Place the disc in your CD ROM drive and either copy the MS Excel or text files to your hard drive for import, or follow the instructions for setup and use of the FileMaker Pro search software.

SFX Kit (8 CD-ROMs) **589.95**

GENERAL SOUND EFFECTS

BBC SFX Library
British and International Sound Effects



The BBC Sound Effects Library gives you a limitless resource of sound. Meeting the highest industry standards, BBC offers a wide selection of sound effects recorded by BBC's top engineers. The original 40 CD series comprises a full complement of exterior atmospheres, household, interior backgrounds, comedy, fantasy, humor, communications, water, industry, cars, hospitals, equestrian events, livestock, agricultural machinery, horses, dogs, schools and crowds. The second edition, offers an astonishing variety of professional quality sound effects, as well as international sounds and ambiences, from countries in Europe, Asia & South America.

BBC SFX Library 1-40

BBC 01: BBC Sound Effects
BBC 02: Exterior Atmospheres
BBC 03: Household
BBC 04: Interior Backgrounds
BBC 05: Transport
BBC 06: Animals & Birds
BBC 07: Human Crowds, Children & Footsteps
BBC 08: Comedy, Fantasy & Humor
BBC 09: International
BBC 10: Communications
BBC 11: Water
BBC 12: British Birds
BBC 13: Industry
BBC 14: Cities
BBC 15: Natural Atmospheres
BBC 16: Cars
BBC 17: Sport & Leisure
BBC 18: Bang!
BBC 19: Electronically Generated Sounds
BBC 20: Weather I
BBC 21: Weather II
BBC 22: Ships & Boats I
BBC 23: Ships & Boats II
BBC 24: America
BBC 25: Aircraft
BBC 26: China
BBC 27: Babies
BBC 28: Hospitals
BBC 29: Africa: The Human World
BBC 30: Africa: The Natural World
BBC 31: Equestrian Events
BBC 32: Greece
BBC 33: Adventure Sports
BBC 34: Livestock I
BBC 35: Livestock II
BBC 36: Farm Machinery
BBC 37: Horses I
BBC 38: Horses II & Dogs
BBC 39: Schools & Crowds
BBC 40: Spain

BBC SFX Library 1-40 (40 Audio CDs) **495.00**

BBC SFX Library 41-60

BBC 42: The Age of Steam
BBC 43: D.I.Y. and Building
BBC 44: Cats
BBC 45: "Dial 999"
BBC 46: Istanbul
BBC 47: British Birds II
BBC 48: Crowds
BBC 49: Suburbia
BBC 50: France - Paris
BBC 51: London
BBC 52: Exterior Atmospheres II: British Rural BG
BBC 53: India and Nepal: City Life
BBC 54: India, Pakistan and Nepal: Countryside
BBC 55: Footsteps I
BBC 56: Footsteps II
BBC 57: Urban South America
BBC 58: Rural South America
BBC 59: Hungary
BBC 60: The Czech Republic and Slovakia

BBC Library 41-60 (20 Audio CDs) **359.95**

International SFX Library



The renowned International Sound Effects Library has undergone a complete digital remastering, under the careful supervision of Sound Ideas - utilizing the original tape masters. This comprehensive collection features a diverse selection of more than 2,500 sound effects on 28 CDs - more than 35 hours of recorded sound.

Animals ..... 3 CDs
Business ..... 2 CDs
Civic ..... 1 CD
Domestic ..... 1 CD
Entertainment ..... 1 CD
Gimmicks ..... 1 CD
Humans ..... 5 CDs
Machinery ..... 2 CDs
War ..... 1 CD
Nature ..... 1 CD
Ranch ..... 1 CD
Sport ..... 3 CDs
Transportation ..... 6 CDs

International SFX Library (28 Audio CDs) **495.00**

GENERAL SOUND EFFECTS

De Wolfe SFX Library



A very special blend of general and European sounds together in one collection, De Wolfe is an exceptionally well recorded digital sound library that provides over 2000 effects on 23 CDs, and with great new sounds being added to the library on a regular basis, this is a library that can grow on you.

Combined with De Wolfe's wide selection of ambiences, transportation, domestic and comedy sounds are unique sounds like sailing a 18th Century Clipper Ship, the inside ambience of a Zeppelin and Harrier Jets on the attack. The versatility and professional quality of this library makes this collection a must for all audio professionals.

DW 01: Communications
DW 02: Electronic
DW 03: Water
DW 04: Comedy 1
DW 05: Comedy 2
DW 06: American
DW 07: Domestic 1
DW 08: Domestic 2
DW 09: Interior Atmospheres
DW 10: Warfare
DW 11: Industry
DW 12: Cars
DW 13: Farmyard
DW 14: Exterior Atmospheres
DW 15: Period Transport
DW 16: Interior Crowds
DW 17: Period Warfare
DW 18: Human Sounds
DW 19: Horror
DW 20: Construction
DW 21: D.I.Y. 1
DW 22: D.I.Y. 2
DW 23: Wild Animals & Tropical Atmospheres
DW 24: Traffic 1
DW 25: Traffic 2

DeWolfe SFX Library (25 Audio CDs) ..749.95

L2 SFX Library — General Collection from Hollywood



Produced by award-winning Hollywood sound designer Frank Serafine, the L2 SFX Library, is a must for everyone in the sound industry. Mr. Serafine is credited with the development of many digital audio techniques heard in blockbusters such as Star Trek, Lawnmower Man, The Hunt for Red October and Virtuosity. This innovative 10 disc set offers effects in eight exciting categories: animals, electricity and static, industrial (double disc), metal, science fiction (double disc), supernatural, transportation, and water. L2's documentation provides clearly defined titles for each effect as well as more comprehensive descriptions of the sounds.

1: Animals
2: Electricity & Static
3: Industrial #1
4: Industrial #2
5: Metal
6: Science Fiction #1
7: Science Fiction #2
8: Supernatural
9: Transportation
10: Water

L2 SFX Library (10 Audio CDs) 495.00

Audio Pro SFX Library – General European Collection



The Audio Pro Sound Effects Library, produced by Yannick Chevalier of Point 12, offers 18 CDs of an extraordinary variety of sounds: nature, animals, airplanes & airports, trains and stations, transportation, traffic, crowds and industry, leisure & entertainment, tools, offices, communications, cameras, household, clocks, room tones, bells, sports, war, human solo, human groups, music samples, science fiction, cartoons, and footsteps. This library provides a wealth of well-recorded European sounds.

AP1: Sounds of Nature
AP2: Animals
AP3: Airplanes, Airport
AP4: Trains, Station
AP5: Transportation #1
AP6: Transportation #2
AP7: Traffic
AP8: Crowds, Industry
AP9: Leisure, Entertainment
AP10: Tools, Office, Communications
AP11: Household
AP12: Household, Clocks, Tools
AP13: Room Tones
AP14: Bells, Sports, War
AP15: Human: Solo
AP16: Human: Groups, Solo, Crowds
AP17: Music Samples, Science Fiction
AP18: Cartoon, Footsteps

Audio Pro SFX Library (18 Audio CDs) 739.95

COMPACT GENERAL SOUND EFFECTS

SuperSampler



SuperSampler provides an overwhelming array of choice in one neat little package. This broad cross-section of more than 850 sound effects offers content, versatility and coverage - and bursts at the seams with:

•Animals •Automobiles •Background Ambiences •Bells & Whistles •Birds •Business & Telephones •Comedy • Computers & Electronics •Crash & Destruction •Drones •Guns •Hits & Explosions •Human Groups & Solos •Machines & Tools •Monsters •Pass Bys •Percussion •Space •Sports •Switches & Levers •Water •Weapons •Whooshes •Zips •Zaps & Zings

Super Samplers (1 Audio CD)..... **109.95**

Digiffects FX Collection



Designed as an excellent starter package, this collection offers the best from the huge Digiffects library of previously unreleased effects. Consists of 5 CDs with more than 800 FX, and provides a great base of audio for daily production needs. Perfect for small studios and production facilities.

- FX-1: City and Transport
- FX-2: Home and Nature
- FX-3: Industry and Office
- FX-4: Leisure and Sports
- FX-5: Human and Special

Digiffects FX (5 Audio CDs).....**Call**

Digiffects MegaDisc



Over 380 effects from a wide variety of in-demand categories. This CD offers a great general SFX library, all on just one disc. The Megadisc has been tailor-made for use with a sampler and is prepared for easy looping. The package also includes a sound editing guide with tips & practical advice.

Digiffects (1 Audio CD) **109.95**

Mixed Variables



A dynamic addition to your existing general effects collections, Mixed Variables from Crossing Point is a broad spectrum, 3 CD set created by Aarin Richard, one of the industry's top sound designers and an award winning engineer. This incredible collection features a stunning array of unique ambiences, everything from sci-fi to natural settings, plus a great selection of hard effects and Foley. Numerous tracks were recorded with the Neumann KU-100 binaural head (microphone), which creates exceptional stereo imaging and depth for playback over loudspeakers and truly amazing 3D qualities when listened to in headphones - very handy for games or special applications. Other effects are spatially enhanced over two speakers, and all 579 effects were recorded through world class equipment with meticulous attention to sound quality.

MV-01

It's all about ambiences. Create worlds that are mysterious, eerie, haunting, ethereal, serene or foreboding. Sci fi interiors abound with regeneration chambers, rebirthing neuralizers, photon inverters, rotating arms, reactors and pulsing machine rooms. You'll also find yourself in creaking tunnels, wet sewers and with chanting monks, not to mention the very practical selection of natural settings that have been captured on this disc.

MV-02

Wide variety of FX that make great production elements, including whacks, thwacks, crunches, whooshes, fly bys, machines, cars, tools, magic, water, wind, rain, metal, wood, latches, squirts, slurps, aliens, odd flying things, processed effects ... and much more.

MV-03

More hard effects, plus a series of Foley oriented sounds. Many useful sounds here that can just be dropped into your production.

Mixed Variables (3 Audio CDs).....**295.00**

The Best of The Saul Zaentz Film Center



The Best of The Saul Zaentz Film Center is an outstanding collection of more than 200 feature film and general sound effects, representing some of the Center's finest work. This single CD offers a wide ranging selection of transportation, animals, props, Foley, creaks and crashes, explosions, ghosts and zombies, hits and impacts, rumbles, scrapes, tools, water and wind. It's a little bit of the best of everything ...

Based in San Francisco, the Saul Zaentz Film Center has provided filmmakers with first class sound facilities and some of the best sound editors, mixers and Foley artists in the business since 1979. With more than 20 Academy Awards to their credit, Saul Zaentz and his team have produced many memorable major motion pictures, including One Flew Over the Cuckoo's Nest and The English Patient.

•Animals •Biplanes •Bodyfalls •Crashes •Creaks & Squeaks •Riots •Explosions • Impacts •Hits •Rumbles •Water •Whooshes •Wind •Zombies

The Best of The Saul Zaentz Film Center **109.95**

VINTAGE GENERAL SOUND EFFECTS

Series 1000 — World's First SFX Library on CD



This is where it all began at Sound Ideas - the original Series 1000 SFX audio tape library was the first sound effects library of its kind to be made available on CD. Its 28 CDs contain over 3,000 professional quality analog sound effects in a wide variety of categories. This collection from the 1960s and 70s is also a valuable source of vintage archival effects.

1001: Airplanes
1002: Animals
1003: Airports, Archery, Applause, Automobiles
1004: Automobiles
1005: Baseball, Bells, Bicycles, Cameras, Clocks
1006: Birds, Boxing, Buzzers
1007: Children, Computers
1008: Construction
1009: Construction, Crashes, Crowds, Doors, Drawbridges
1010: Crowds
1011: Electronics, Earthquakes, Electricity, Elevators, Exercise
1012: Explosions, Fairs, Farms, Football, Fire, Guns
1013: Footsteps, Foreign Sounds, Gambling, Golf
1014: Helicopters, Horns, Hockey, GlassSounds
1015: Household Sounds
1016: Human Sounds, Jungles, Laboratory, Laughter, Logging
1017: Indians, Industry
1018: Naval Boats, Ships, Motorcycles
1019: Military Battles, Guns, Harps
1020: Musical & Percussion Instruments
1021: Musical & Percussion Instruments
1022: Office, Country Sounds, Parades
1023: Pinball, Post Offices, Projectors, Pumps, Rodeos, Restaurants, Roller Coasters
1024: Service Stations, Sirens, Snowmobiles, SpaceSounds
1025: Stores, Swimming, Subways, Telephones
1026: Rain, Thunder, Tennis, Taxis, Trains, Umbrellas, Vacuum Cleaners, Vendors
1027: Traffic, Trucks, Buses
1028: Water, Whistles, Wind, Zippers

Series 1000 (28 Audio CDs).....**Call**

Vintage SFX from Digiffects

Digiffects presents the Vintage FX Sound Effects Library - a 12 CD cross section of sound effects from the 50s to the 80s. With 1700 professionally recorded sound effects, this collection contains a very broad range of sounds - from insects to big explosions. Originally recorded on Nagra tape recorders these effects have been carefully selected and edited on today's state-of-the-art equipment.



The aim was to select sounds which are now very difficult or impossible to obtain. For example, it's not easy to record a horse and cart rolling through an old town, nor are old cars, trains or machines readily available. If you need sounds to match older movies or audio productions, this is the perfect collection. The sound effects are top quality, they sound great in any audio production, old or new. Many of effects are really unique. Just listen to some of the war sounds or airplanes.

VA1: Vintage City FX - Airports, Traffic, Other Environments

VB1: Vintage Domestic FX - Household Sounds

VC1: Vintage Rural FX - Forests, Weather, Water & Animals

VC2: Vintage Rural FX - More Animals and Tools

VD1: Vintage Transport FX - Trucks, Cars, Race Cars, Motorcycles & Scooters

VD2: Vintage Transport FX - Sirens, Traffic, Steam Trains, Horns, Ships, Motorboats, Sailboats, Airplanes

VE1: Vintage Industry FX - Factories, Workshops, Engines & Equipment, Tools

VE2: Vintage Industry FX - Watermills & Windmills, Equipment & Machines, Stretchers & Carts, Typewriters & Computers

VG1: Vintage Leisure FX - Bicycles, Boats, Swimming, Fireworks, Amusement Parks, Pinball, Cameras & Music

VI1: Vintage Human FX - Babies, Children, Laughter, Voices, Applause, Footsteps, Native Americans, Heartbeats

VJ1: Vintage Special FX - Hits, Breaks, Falls, Squeaks & Demolition, plus Guns, Cannons, War & Fight Sounds

VJ2: Vintage Special FX - Whips, Gas Valves, Fire, Hoses & Water, Alarms, Signals, Feedback & Other Specialty Sounds

Vintage SFX (12 Audio CDs) **449.95**

VINTAGE GENERAL SOUND EFFECTS

The Library — Great Starter Package

If you're in need of a versatile general sound effects collection at a reasonable price, The Library is just for you! This six CD set of more than five hundred sound effects is a great starter package, or a perfect compliment to your own sound library.



Airplanes, Jet, Airplanes, Propeller, Airport, Control Tower, Animals

Automobile Races, DragstersAutomobiles, General

Automobiles, Sports CarsBalloons, Hot Air

Bars & Pubs.....Basketball

Bells.....Bells, Door

Boats, Inboard.....Boats, Outboard

Boings.....Bowling

Buses, City Transit.....Buzzers

Cable Cars.....Cash Registers

Chimes, Wind.....Clocks, Alarm

Clocks, Stopwatch.....Computer Rooms

Construction.....Crowds, Applause

Crowds, Booing.....Crowds, Boxing

Crowds, General.....Doors

Electricity.....Factories

Grocery Store Sounds.....Hand Guns

Harbours & Ports.....Helicopters

Horns.....Household Sounds

Human Sounds.....Industrial Sounds

Junk Yards.....Lawn Mowers

Motorcycle Races.....Motorcycles

Office Sounds.....Oil Wells

Police Radios.....Power Plants

Printing Presses.....Restaurant Sounds

Rifles.....Roller Coasters

Sawmills.....Shot Guns

Sirens.....Stagecoaches

Swimming.....Telephones

Toilets.....Traffic, City

Traffic, Highway.....Trains

Typewriters.....Water Fountains

Water, in Cave.....Water, in Ocean

Waterfalls.....Whistles

The Library (6 Audio CDs.....**209.95**

De Wolfe Classic SFX — General Classic Sound Effects

The De Wolfe Classic Sound Effects Library offers you the chance to experience the sounds of a generation, all digitally remastered for today's audio professional. This classic boxed set provides over 1,000 effects - on six CDs.



De Wolfe Classic FX CD #1

Metal Workshop: Vice, Hammer, File, Hacksaw, Gas Torches, Lathe, Brazing Brass

Woodwork: Screwdrivers, Drills, Gauges, Coping Saws, Handsaw, Tenon Saw, Electric Saw, Electric Jigsaw, Electric Drill, Wood & Jack Planes, Sandpapering, Hammering Nails, Wood Lathe

De Wolfe Classic FX CD #2

Comedy: Music Jingles & Instrument Accents, Whistles, Stings and Musical Transitions, Comedy Effects and Accents, Bounces, Crashes & Smashes, Pops, Whizzes & Whooshes, Squelches & Splats, Cartoon Effects

Toys & Amusements: Train, Tank, Submarine, Space Ship, Robot and Mechanical Toys, Pogo Stick, Slot Machines, Roulette Wheels

Percussive Sounds: Wooden, Metal, Glass, China & Paper Sounds, Lighter Flicks, Money, Scissors, Zippers, Balloons

De Wolfe Classic FX CD #3

Aircraft: Boeing 707, V.C. 10, Boeing 747 Jumbo Jet, Boeing 727, Tristar, Hawker Harrier Vertical Take-Off Jet, Hunterfighter, Formation Flying, Tiger Moth, Beaver, Pioneer, Cessna 175, Westland Scout Helicopter

De Wolfe Classic FX CD #4

Rally Cars: Multiple Engine & Driving FX for eleven different vehicles, ranging from 1000 cc 4 Cylinder to 4000 cc 6 Cylinder, Multiple Car & Hot Rod Pass Bys for Brands Hatch & Silverstone Races

American FX: Crowds, Gambling Ambiences, Games, Pistol & Rifle Ricochets, Indian Attacks, Telephone Sounds, Bar, Restaurant & Party Atmospheres, Rocket Launch & Communications, Vehicles & Traffic, Police Sirens & Cars, Fire Engines & Stations, Niagara Falls

De Wolfe Classic FX CD #5

The Elements: Weather & Natural Reactions, Thunder Claps, Rain, Volcanos, Fires, Wind, Rough Seas, Streams, Rivers & Floods

Wild Animal FX: Lion, Tiger, Leopard, Panther, Cheetah, Grey Gibbons, Monkey House & Chimpanzees, Orangutangs, Gorilla, Deer, Camels, Sea Lions, Crocodiles, Hippos, Flamingos, Elephants, Huskies & Dogs, Bull, Rhinos, Stampede, Jungle, Swamp & Forest Atmospheres

De Wolfe Classic FX CD #6

Domestic: Electric Plugs & Appliances, Cleaning Sounds, Family Pets, Household Sounds, Water & Fixture Sounds, TV Sets & Radios, Light Switches & Doors, Food Preparation

Industry: Fabrication, Factory & Assembly Shop Atmospheres, Machines, Annealing Furnace, Machine Shop Sounds & Atmosphere

Dw Wolfe Classic SFX (6 Audio CDs).....**385.00**

SURROUND SOUND LIBRARIES

Just Boom Trax

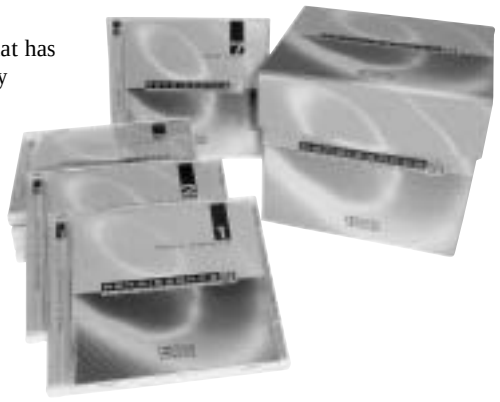


Specifically designed as a sweetener for use in the sub-woofer channel when mixing surround sound productions, Just Boom Trax offers more than 250 tracks of real life, fantasy and utility sound effects, all at low frequencies that allow the audience to feel the sound.

- ◆ Lower the boom and dig deep. Just Boom Trax enhances the depth and impact of your audio with a versatile toolbox of low frequency growls, impacts, hums, booms, throbs, rolls and drones.
- ◆ Also offers sci-fi and fantasy tracks, a set of oscillator tones, low frequency white and pink noise, and a selection of drum hits and synth accents - more audio goodies for your low frequency toolbox.
- ◆ Sounds drawn from real life -
- • aircraft carriers, jets & helicopters
- • boats, ships & submarines
- • automobiles, races & crashes
- • trains, trucks & motorcycles
- • animal growls, footsteps & cries
- • human hits & bodyfalls
- • thunder, wind, earthquakes, avalanches & volcanoes
- • weapons and military battles
- • construction, crashes & hits
- • explosions and impacts
- • doors, industrial & motor sounds, stampedes and the space shuttle's sonic boom.
- ◆ Includes 2 CDs and 2 DVDs of multi-formatted low frequency effects. This collection offers the following six multimedia formats:
- • 44.1k 16-bit - in both .wav and AIFF formats
- • 48k 16-bit - in both .wav and AIFF formats
- • 48k 24-bit - in both .wav and AIFF formats

Just Boom (2 Audio CDs, 2-DVDs.....359.95

Renaissance SFX – Surround Sound Special Effects



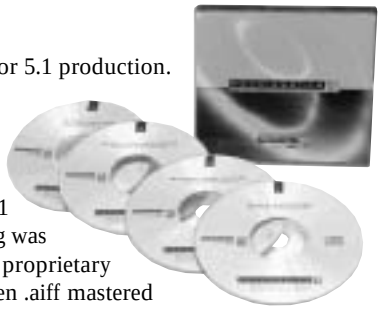
The first sound effects library in the world that has been encoded and produced entirely in Dolby Surround. The Renaissance SFX Library offers awesome fully dynamic sound in a real 3D listening environment. The Dolby Surround encoding process makes it possible for a regular stereo source to deliver all four channels of sound. These sounds extend far beyond simple panning or standard MS techniques - you will hear complete sets of effects that literally move along the speakers and bathe you in true audio from all directions. In addition, you will be able to control the placement and movement of sounds in the different channels in order to recreate just the right audio environment for your productions.

The Renaissance SFX Library can be used in a regular stereo environment, but you will be even more impressed with the way a full surround sound system demonstrates the quality, richness and versatility of these recordings.

CD #1.....World Ambiences
CD #2.....Sounds From Nature
CD #3.....Special Environments
CD #4.....Impact FX
CD #5.....Voice FX
CD #6.....Emotional Voices
CD #7.....Music Tools
CD #8.....Subwoofer FX
CD #9.....Rhythmical Tools
CD #10.....Dramatic String Textures
CD #11.....Strings & Chords

Renaissance SFX (11 Audio CDs).....1195.00

Renaissance 5.1



Renaissance 5.1 is the first sound effects library created for 5.1 production. A perfect complement to the Renaissance SFX library, it offers the best original recordings from the Surround Sound collection, digitally remastered in 24-bit 48 kHz 5.1 surround sound. Renaissance 5.1 uses Digidesign Pro Tools 5.1 name extensions, enabling the Pro Tools 5.1 user to load a 5.1 track as a single audio file. Remastering was accomplished through Renaissance Sound Technology's proprietary psychacoustic software applications, and all files have been .aiff mastered for full PC compatibility as well.

CD #1: World Ambiences:

A selection of ambiences from Asia, Europe and America - markets, villages, a monastery, restaurants, traffic and urban settings.

CD #2: Sounds from Nature:

Ocean and sea sounds, Asian elephants, horses, jungle birds and insects, plus rural sounds from South America and the European Alps.

CD #3: Special Environments:

Percussion, doors & gates, glass & dish sounds, industry & machines, ship sounds, and a rifle range.

CD #4: Subwoofer FX:

Engine subharmonics, LFE impacts, hits, lasers & shots, wind & catastrophic storm subharmonics, earthquake & tremor vibrations, and a variety of waveforms: intermodulated, modulated harmonics, sine harmonics and square harmonics.

Renaissance 5.1 (4-CD ROMs) 395.00

SCI-FI COLLECTIONS

Sci-Fi Series 8000



Sci-Fi 8000 is a 4-CD compilation from Hollywood's best science fiction sound designers. It provides more than 500 contemporary space and futuristic fantasy sound effects and ambiences: everything from alarms, space ships and alien creatures, to hatches and air locks, robotics, rockets, telemetry, weapons and Challenger space shuttle launches.

Among the award-winning sound designers whose talents are featured in this collection are: Randy Thom (Mars Attacks! and Contact), David Yewdall (Fifth Element and Starship Troopers), Mike McDonough (Star Trek: First Contact), and Charles Maynes (Starship Troopers).

This modern collection comes packaged in a special carrying case - a tribute to classic retro sci fi films - and all four CDs have been digitally mastered in stereo. The Series 8000 will take your productions where no man has gone before!

8001: Air Bursts, Alarms, Ambiences

8002: Ambiences, Animals & Creatures, Avalanches, Communications, Battles, Computers, Docking Thrusters, Doors, Drains, Drones

8003: Earthquake, Electricity, Energy Bursts, Explosions, Fires, Fly Bys, Footsteps, Freezer Chambers, Laser Guns & Beams, Phasers, Robotics, Ray & Stun Guns, Dry Ice, Scanners, Servos, Space Rockets & Shuttles

8004: Space Ships, Simulators & Stations, Splats, Squeals, Telemetry, Torpedoes, Vortexes, Weapon Dischargers, Whooshes, Wind, Zips

Sci-Fi Series 8000 (4 Audio CDs).....**359.95**

Sci-Fi Series 8000 Warp 2



Enter the next dimension! Come along for the ride as Series 8000 Warp 2 weaves its way through a whole new galaxy filled with alien, out of this world and futuristic sounds, including:

8005: Air Blasts, Air Locks, Alarms, Ambiences, Animals & Creatures, Battles, Bells, Black Holes, Communications, Computers, Control Rooms, Crowds

8006: Doors, Shields & Hydraulics, Dopplers, Drones, Electricity, Electronic Pulses, Scrambles & Hits, Energy Bursts & Transformers, Explosions, Fly Bys, Force Fields, Laser Guns & Beams, Stun Guns, Electronic & Metal Hits, Laboratory Sounds

8007: Photon Bursts, Plasma Capsules, Power Sources, Robotics, Rockets, Satellites, Scanners, Sensors, Servos, Space Pods, Ships & Stations, Switches, Telemetry, Time Warps, Transporters, Vortexes, Weapons, Whooshes, Zips

Sci-Fi Series 8000 Warp 2 (3 Audio CDs).....**359.95**

Sci-Fi Series 8000 Warp 3

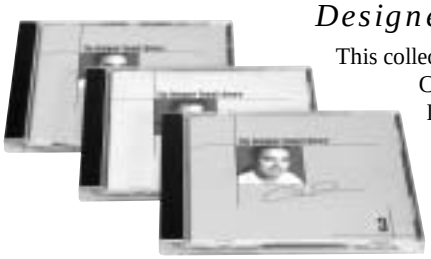


With over 400 sci-fi tracks on 3 CDs, this collection presents a broad galaxy of "out there" audio. Explore the sounds of new life forms, intergalactic transmissions and a selection of drones and deep space ambiences. Warp 3's outstanding contents will compliment journeys to distant stars, alien worlds and the inner space of your imagination.

•extraterrestrial creatures & alien insects •sci-fi ambiences, backgrounds, drones, hums, tones & frequencies •space bells and unusual underwater atmospheres •a full transmission of communications, space buoys, relays & chatter •dynamic energy fields, electric arcs and damaging explosions •weapons, flame throwers and space torpedoes •hits, lasers and whooshes •intergalactic pneumatic delivery systems •radar sweeps, servo movements & sonar pings •tractor beams, transporters and scanners •shuttles, space ships and flying saucers •electronic squeals and telemetry • ethereal space winds

Sci-Fi Series 8000 Warp 3 (3 Audio CDs).....**359.95**

Designer Sound Library — Fantasy, Sci-Fi and Music Accents



This collection of almost 700 sound effects is guaranteed to add a little magic and mystery to your productions. Created by well known sound designers Jonathan Helfand and Clive Smith, The Designer Sound Library offers an imaginative mixed bag of special effects: Fantasy, Science Fiction, Telemetry, Guns, Explosions, Hits, Thunder, Drones, Whooshes, Swishes, Glisses, Pulses, Zips and Sparkles, Complimented by a Variety of Percussion & Musical Stinger Accents.

You'll also discover crystal ice bells & shifting space whirls, glassy ice rain & a fairy dust gliss, plus a full selection of space and fantasy ambiences and orchestral and percussion accents.

Designer Sound Library (3 Audio CDs).....**249.95**

SCI-FI COLLECTIONS

Metropolis Science Fiction Toolkit 1



This Science Fiction Toolkit from the professional sound designers at Futurity contains a vast collection of whooshes, sweeps, releases, scans, accents, alerts, blasts, prompts, bleeps, whirls, flutters, synths, washes, impacts, lasers, sirens, pulses and buzzes, plus a great selection of science fiction ambiences and other electronic tools. Keep the toolkit handy for everything from odd jobs to universal productions.

Toolkit 1.....**109.95**

Metropolis Science Fiction Toolkit 2



Taking design elements to new heights, the Metropolis Science Fiction Toolkit II CD contains over 200 new effects never heard anywhere before. This CD includes great mechanical, ethereal, whoosh, metallic, static, and ambience effects, plus squeaks, scans, scrapes ... and lots more. If you're looking for something off the beaten path, Metropolis II is here to open up a new frontier.

Toolkit 2.....**109.95**

The Big Drone - Wide Variety of Simple and Complex Drones



Mysterious, sweeping, pulsing and enigmatic. The Big Drone - one hundred tracks that are ready to rumble. This two CD set features both simple and complex drones that have been designed to enhance many different kinds of scoring applications.

Big Drone (2 Audio CDs).....**109.95**

JURASSIC DINOSAURS

An out-of-this-world menagerie, Jurassic Dinosaurs provides all kinds of creatures, from tiny to humongous. Included in this jammed CD are all manner of prehistoric, fantasy, alien, monster, supernatural, fairy and robot creatures, with many different varieties of cries and associated sounds.



Some of the effects come from real-life animal sounds, which were morphed into something else. Many others were created in the studio and fine-tuned to provide a full range of possible variations. Additionally, the animal and creature sounds represented here are immensely versatile. Use them for modern day exotic jungle beasts, prehistoric creatures, or futuristic aliens from other planets - the choice is yours!

The creatures represented in this collection include the following:

Organic Creatures, Small to Large:

Mouse, Bird-Like, Chipmunk-Like, Mouse-Like, Dinosaur Hatchling, Bird-Like, Small Animals, Classic Jungle Birds, Small Animals, Laughing Creature, Sheep or Goat-Like, Weird & Humourous Creature, Cat-Like Growls, Roars, Hisses & Snarls, Strange Jungle Animals, Complex Cries, Fluttering, Mournful & Haunted Wails, Heavy Breaths, Elephant-Like Trumpeting, Pig-Like Grunts, Nasal Trumpeting, Complex Dinosaur Battle Screams, Grunting Cries, Hoarse Growls, Alien Cries & Calls, Sneezes & Coughs, Cow-Like Moos, T-Rex Attack and Challenge Cries, Groans & Grunts, Honking Cries, Guttural Rumblings & Roars, Angry Dinosaur Growls & Grumbles

Organic Body & Environment:

Creature Footsteps on Various Surfaces, Impact Tremors, Thundering Stampede, Eating Cracks, Crunches and Snaps, Wing Flapping on Flying Creatures, Flying Insects, Plant Monster FX, Broken Foliage FX

Alien, Robotic & Supernatural Creatures:

Android, Cyborg Language, Help Droidbot, Robot Footsteps and Servo Motor, Robot Start-Up, Overload, Electric Discharges, Radio Control Frequency & Squeaking Joints, Alien Creatures, Dragons, Werewolves, Shape Shifters, Ghost Vortexes, Classic Tinkerbell Winged Fairy

Jurassic Dinosaurs **109.95**

Drone Archaeology



This CD, produced by Numerical Sound, contains 99 stereo drones, ranging from 32 to 64 seconds in length. The drones are all steady in pitch, but have constant movement, thus they are never the same from one moment to the next. In order to aid multimedia and film production, the drones have been classified into seven categories as follows: Nature, Science, Drama, Industry, Voice, Bell and Drones.

Each drone is contemporary and original in nature, offering the film composer and musician an invaluable resource for creating soundtracks, sound beds, and sound design. Used as foundation material, these drones can be processed and transformed in any number of ways to assist the user in the development of their own unique sound.

Drone Archaeology **109.95**

SCI-FI COLLECTIONS



The Science Fiction SFX Library

The Science Fiction Sound Effects Library offers a broad range of futuristic, space, alien and creature sound effects and ambiences. This 3 CD package provides a full audio galaxy, designed for versatile use in a wide variety of science fiction scenarios and sequences.

The collection also contains excerpts from a sci-fi audio drama called Space Fighter Duel, showcasing many of the librarys effects and suggesting editing / mixing possibilities for them. Space Fighter Duel is provided as both a continuous track, and as separate indexes, and its available for use either in its entirety, or separated into specific effects.

Contains exciting and expressive source material in the following categories:

Spacefighter Duel & Selected Elements

Flights & Attacks, Launches, Blastoffs, Flybys & Circling, Explosions, Battle, and a Roasted Toasted Alien

Ambience Effects, Action Effects and Scenes

Space Hangar Ambiences, Spacefighter Strafing Runs & Flybys, Squadron Formations, Moon Landing Sequence, Starship Bridge & Warp Drive Sequence, Space & Blaster Battles, Lift-Off to Orbit Countdown, Re-Entry Sequence, Computer Room, Mad Scientist's Laboratory, Laser Cannon Blasts, Alien Weather & Lightning Bolts, Horror Heartbeat, and Engine Room Ambience

Electronic Effects

Bells, Bubble Up Effects, Communications & Telemetry, Beeps, Radio Bursts, Intercom Static & Space Radio Interference, Space Beacon, Warnings, Hand-Held Computers, Sine Waves, Warble, Electromagnetics, and Space Radio Tuning

Mechanical Effects

Laser Bursts, Pneumatic Sliding Doors, Sinister Pulsating Machines, Vacuum Pump, Mechanical Computers, and Various Machine, Switch & Motor Sounds

Weapons Effects

Vindicator & Scavenger Weapons Fire, Spaceship Weapons Fire, Turbo Laser, Proton Torpedoes, Death Star Explosions, Hand Blasters, Hand Phasers, and other Weapons

Fantastic Creatures

Alien Creature Mumbling, Cute Droid, Mean Little Creature, Alien Intelligence and Computer Speech, The Id Monster, Cantina Creatures, Robot, Gargle & Growl Monsters, Weird Thing, Mad Martian Mouse, White Fang, and other Robots and Alien Radio Drop Ins

Miscellaneous Effects

Ominous Flying Saucer, Vibrations, Laser Tunnel Resonance, Radioactive Emanations, Disintegrator - Integrator Units, Launch & Catapult Effects, Meteor Showers, Personal Extra-Vehicular Thruster Units, Rumbles, Wails, Sirens & Sputters, Garbled or Coded Radio Communications, Phaser and Other Space Sounds

Soundscapes & Music

Solar & Cosmic Storms, Flying Saucer Landings & Take-Offs, Close Encounter with Mothership, Glass Tones, Mental Projection, Dark Swells, Interdimensional Madness, Nightmare Terror & Radioactive Crying, Humourous Zombies, Kaleidoscope, and other Musical Compositions

Science Fiction SFX Library (3 Audio CDs) **249.95**



Radioactive —Eerie and Sci-Fi Waveforms

Created by recording and processing shortwave radio signals as they bounce against each other and the upper reaches of Earth's atmosphere, Radioactive is a sci fi sound effects collection unlike any other. This collection of 200 eerie wave forms, pans, sweeps, phasers, communications, shimmers, bubbles, beeps, static, monsters and apparitions has a unique flavour all of its own, and a nearly limitless range of possibilities for your applications.

- Space Monster Screams
- Phaser Morph
- Shimmering Space Bells
- Ringing Soundscape
- Space Ribbon with Whine
- Time Travel Reverb
- Shuttle Craft Engine Room
- Steady Bubble Gyro Delay
- Alien Tractor Beam Sequence
- Evil Diablo Wave
- Vortex Entry
- Single Ray Gun Blast
- Rumbling Wind Whoosh
- Solar Wind & Waves
- Predatory Black Hole
- Phaser Wave Boomerang
- Electric Tubescape
- Distorted Space Guitar
- Fading Bubble Chirp
- Galactic
- and many others.

Radioactive (2 Audio CDs) **209.95**

EXTREME SOUND EFFECTS

Crash & Burn



This 2 CD collection of over 400 FX offers an outstanding selection of destructive sounds and sequences. Prepare to duck and cover - this library contains disasters, earthquakes and explosions, marine accidents and underwater mishaps. Also included in the collection:

- Wood, Junk, Metal, Plastic & Rock
- Bombs and Timers
- Dynamite, Gas Explosions and Grenades
- Fireballs, Chemical & Petroleum Fires
- Firestorms and Fire Extinguishers
- Forest & Building Fires
- Guns, Bullets & Missiles
- Torches and Ignitions
- Window Breaks & Glass Smashes
- Molotov Cocktails • Rumbles, Blasts, Debris
- Building, Ceiling & Elevator Falls
- Bodyfalls & Body Hits • Twisters

Crash and Burn (2 Audio CDs & 2-DVDs) 215.00

Power Pack from Digiffects



Power Pack from Digiffects is a collection of high powered and superquality processed sounds created to save you valuable time in situations when ordinary effects aren't strong enough. The collection boasts more than 1,000 pre-boosted effects on 5 CDs. Packed with tremendous impact and processed to the limit, the effects are also designed to be combined with each other.

#1- Synthetic Media FX: Tailor-made for commercial use

#2- Synthetic Media FX: Longer and machine sounds

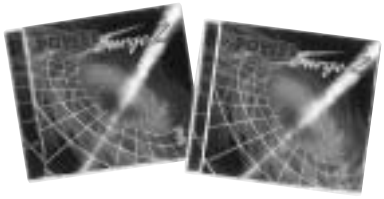
#3- Processed Sound FX: Stronger, tougher and louder

#4- Processed Sound FX: Machine sounds & impacts

#5- Cartoon Sound FX: Comedy & corny sounds

Power Pack (5 Audio CDs & 2-DVDs ....Call

PowerSurge 1



This 2 CD collection is high energy, high voltage and full of potential - your power source for electrical sparks, arcs and zaps, crashes, smashes, hits and punches, large impacts, guns and helicopters, electronic beeps & buttons, telemetry and computers, a full range of explosions, weather and natural disasters, and an entire galaxy of potent sci-fi FX, ambiences, and alternate realities.

PowerSurge 1 (2 Audio CDs).....209.95

PowerSurge 2

PowerSurge 2 provides another jolt of high voltage sound effects to energize your productions. Full strength, more juice, lots of adrenalin and ready to take your productions to their full potential! PowerSurge 2 has just the sizzle required to give you the most bang for your buck!

PowerSurge 2 (2 Audio CDs).....209.95

Larger Than Life



The Larger Than Life SFX Library is an astounding collection of awesome audio effects. We're talking BIG - explosions, gunfire, thunder, horror, crashes, bangs, ...and much more! Created by Sound Dogs, one of Hollywood's most elite production teams, these sound effects are undiluted and full strength.

CD #1 Air Releases, Airplanes, Airport Control Towers, Archery, Augers

CD #2 Automobiles, Auto Crashes & Skids, Boats, Bodyfalls, Bone Breaks, Crashes, Electricity, Explosions, Feedback, Fire Bursts, Flame Throwers, Glass Smashes, Hand Guns, Rifles

CD #3 Body & Face Hits, Squishes, Horses, Industrial Machines, Metal Sounds

CD #4 Radio Static, Rockets, Sci Fi Sounds, Submarines, Sword Battles & Hits, Light Sabres

CD #5 Tanks, Telemetry, Torpedoes, Trains, Volcanos, Water Geysers, Weather, Whooshes, Wood Hits

Larger Than Life (5 Audio CDs) 495.00

The Dark Side of Sound

For things supernatural and spooky. This delightfully dark 2 CD collection contains drama, horror & suspense ambiences, plus a full selection of creatures, monsters, ghosts, vampires, zombies, haunted houses and lots of other creepy effects. The Dark Side of Sound has it all:



- Encounters With Evil
- Dripping Blood
- Ripping Flesh
- Graveyards
- Coffins
- Dungeons
- Guillotines
- Secret Passageways
- Demonic Languages
- Eerie Whispers
- And much more...

The Dark Side of Sounds (2 Audio CDs) 209.95

EXTREME SOUND EFFECTS

Thrillers, Fantasies & Hauntings



Acclaimed Hollywood sound designer John Peters (credits include Mars Attacks and Stargate SG-1) has produced another high anxiety collection of classic sci fi and horror sound effects—Thrillers, Fantasies & Hauntings.

This 3 CD set offers a complete torture chamber full of realistic and over-the-top sounds: ready to inflict pain, scare the living daylights out of your listeners, and send chills up your spine.

TFH-01

Grisly electric chair sequences electrical discharge effects, energy effects, magnetic waves & supernatural force, s energy streams & bolts shot/transmitted, Frankenstein's classic laboratory power generator, flame throwers, stun guns and scanning devices, arrows, knives & axes being shot and thrown, torture devices — whips, the Boot, the Rack, the Pit & the Pendulum, medieval catapult fireball launches, chain effects, bodyfalls, stumbles & footsteps, human screams.

TFH-02

Strange heartbeats, gory body flesh rips, stabs and exploding heads, acid melts of faces, heads & bodies, monsters, creatures, birds & ghosts, dungeon doors creaking and slamming, object impacts, bounces & ricochets, weapons—blasters, lasers, proton, zapper and neutrino guns, plus heavy weapons fire, resonating force fields & radio transmissions, explosions, landslides, avalanches, asteroid pass bys.

TFH-03

Storm rumbles & ominous forces, spooks, haunted houses & poltergeist activity, horror and suspense atmospheres and backgrounds, suspenseful power drones & horror music, metaphysical lightning bolts & thunderclaps, howling, ghostly, wailing & alien wind effects.

TFH (3-Audio CDs).....295.00

Impact Effects



Just a single CD, but it packs a punch! Impact Effects contains 605 tracks, all recorded and designed for maximum effect and impact.

Arrow Impacts into Bodies & Wood
Auto Crashes & Explosions
Ball Falls & Hits
Body Falls & Bone Breaks
Building Crashes & Ceiling Cracks
Bullet Impacts into Bodies, Glass, Metal & Wood
Cabinet, Drawer & Cardboard Crashes
Crashes & Auto Crash Test Dummies
Debris Falls, Earthquakes, Soft Falls
Garbage Drops & Hits,
Gas & House Explosions
Glass Falls, Hits & Smashes
Dull, Heavy, Low & Soft Hits
Human Hits, Junk Crashes
Knife Impacts into Bodies & Wood
Mass Destruction
Metal Crashes, Falls & Hits
Meteor Crashes
Plastic Crashes, Crumples, Falls, Hits & Spins
Rock Falls, Hits & Smashes
Sack Drops & Hits, Subway Crashes
Sword Impacts
Wood Breaks, Crashes, Falls & Hits

Impact Effects 109.95

Impact Effects 2



Another smash hit!!

The second CD in this series contains 700 spectacular impacts of every description. Impact Effects 2 offers a wide variety of objects and surfaces, many different impact styles and literally tons of debris. Here's a sample of the sounds you'll find on Impact Effects 2:

• Crashes, Smashes & Crunches • Scrapes and Spins • Hits, Drops, Falls & Rolls • Spills and Squishes • Breaks, Bodyfalls and Debris • Heavy Thuds, Disasters, Demolitions and Rumbles

And here's a list of things that were hit, dropped, kicked, broken and trashed:

•Airplanes, Autos & Trailers
•Anvils, Hammers & Axes
•Battering Rams
•Books
•Cardboard
•Ceramic Dishes, Pots & Tiles
•Chain Link Fences
•Chimes & Alarm Clocks
•Church Bells
•Computer Keyboards
•Concrete
•Doors
•Drums
•Duffle Bags & Baseball Bats
•Dumpsters
•Eggs
•Garbage Cans & Storage Bins
•Glass, Metal, Rock & Wood Debris
•Glass Panes & Coffee Tables
•Gravel, Dirt, Leaves and Foliage
•Ice
•Leather
•Light Bulbs
•Manhole Covers
•Metal Cabinets
•Metal Pipes & Ductwork
•Pianos & Cymbals
•Plastic Bottles & Jugs
•Pots & Pans
•Propane Tanks
•Rocks & Wood
•Sheet Metal
•Spray Cans
•Styrofoam
•Swords
•Televisions
•Trees
•Wrecking Balls

Impact Effects 2 .....109.95

SPECIALTY SOUND EFFECTS

Thunder



Thunder bursts into your productions with two breathtaking CDs - a compilation of more than 100 thunder and rain effects gathered from some of the world's best storm chasers and professional sound engineers. CD #1 contains single claps, cracks, rumbles, rolling thunder and lightning strikes. CD #2 offers a sky full of three minute storm ambience tracks, ranging from gentle rain showers to exploding thunder and fierce hurricanes.

Thunder (2 Audio CDs) **209.95**

Foley Footsteps

The Foley Footsteps CD is a comprehensive all-digital collection of male and female footsteps, recorded on a variety of different surfaces and walking speeds. Surfaces available include cement, wood, grass, gravel, metal, and snow. Sounds of individual steps, paced steps, shuffles, running, jumping and climbing stairs are included. Each sequence is recorded in its entirety, while each individual footstep is also indexed to expedite the foley replacement process. Take a walk on the wild side with Foley Footsteps.

Foley Footsteps **109.95**

Foley Footsteps Winter

Foley Footsteps Winter focuses on real winter situations, providing a great variety of surfaces, footwear and actions with:

- Real snow: Fresh Snow, Packed Snow, Corn Snow & Wet Snow
- Real surfaces: Linoleum, Metal Grate, Concrete, Wood Floor
- Real footwear: Ski Boots, Overshoes, Hiking Boots, Rubber Snow Boots and Snow Shoes

Over 1,700 winter effects, all digitally recorded and edited at a sample rate of 48 kHz 24-bits. The entire library was recorded on a foley stage wet room, by bringing the facility temperature down to 5°C (42°F) and loading the stage with real Canadian winter snow.

Foley Winter (2 Audio CDs).....**250.00**

Art of Foley



The Art of Foley contains a warehouse full of simulated sounds, both extreme and everyday, for Foley synchronization and audio sound replacement usage. Featuring a great selection of the very best from top Foley artist Dan O'Connell of One Step Up, and many gems from the Sound Dogs archives.

FS-01: Applause, Archery, Automobiles, Bags, Basketball, Bells, Belts, Birds, Bone Breaks, Books, Bottles, Brushes, Buckets, Buses

FS-02: File Cabinets, Cans, Carts, Briefcases & Luggage, Ceramic Smashes, Chains, Chairs, Clicks, Cloth Movements, Clothes, Coins, Computer Keyboards, Construction Ambiences, Crashes, Creaks, Desks, Dishes, Dog Movements, Doors, Drapes & Curtains, Drawers, Drills, Drops, Electricity Sparks, Equipment Rattles, Eyeglasses

FS-03: Body Falls, Fax Machines, Footsteps

FS-04: Footsteps

FS-05: Golf, Glass Sounds, Guns, Hammers, Handcuffs, Hits, Human Horror Sounds

FS-06: Horses, Stagecoaches & Wagons, Everyday Human Sounds, Insects, Kicks, Knives, Ladders, Latches, Leaves, Levers, Lighters, Liquids, Locks, Machines, Matches, Metal Sounds

FS-07: Metal Sounds

FS-08: Motors, Nails, Paper Sounds, Pens, Pencils, Ping Pong, Punches, Rattles

FS-09: Cloth Rips, Rocks, Ropes, Safes & Vaults, Sand, Saws, Scissors, Scrapes & Screeches, Shakers, Shovels, Spins & Twirls, Rubber Squeaks, Flesh Stabs, Swimming & Diving, Swishes, Swords, Tables, Telephones, Turrets, Typewriters, Ventilators

FS-10: Water Sounds, Whistles, Whooshes, Wood Sounds

Art of Foley (10 Audio CDs).....**495.00**



ClickShop

4-CD set of 2,300 FX designed to really turn you on ... and off ... and on again. This collection of switches, buttons, beeps, hits, rips, scrapes, clicks, hums, motors and machines offers a great variety of mechanical and other industrial sound effects and backgrounds - both big and small. Also included is a selection of unprocessed sound effects with a wide range of applications for science fiction, horror, fantasy and cartoon productions.

ClickShop (4 Audio CDs).....**Call**

Emergency



This dramatic 2-CD set of more than 500 sound effects features accident scenes, radio dispatches, alarms, sirens, crowds and equipment sounds - your first response and all right in the heat of the action! Emergency provides realistic effects and ambiences to accompany a full range of police, fire, ambulance, hospital, riot and crime scenes.

• Accidents & Rescues • Fires & Explosions • Riot Scenes & Tear Gas • Ambulances, Fire Engines & Life Saving Equipment • Police Chases & Hostage Scenes • Police Cars, Equipment & Guns • SWAT Teams & Forensic Crime Investigations

Emergency (2 Audio CDs).....**209.95**

SPECIALTY SOUND EFFECTS

Series 9000 Open & Close The Ultimate Door Collection



It's an open & shut case - this is simply the most comprehensive collection of doors, windows, gates, drawers, latches, locks, door bells, and compartments ever assembled! The Series 9000 Open and Close SFX Library collection of 12 CDs, with more than 6200 sound effects, offers only "top drawer" selections: from out of this world space doors... to down to earth household cupboards ... and everything you can imagine in between. If it opens and closes - you'll find it in here!

Series 9000 (12 Audio CDs) **509.95**

Just Noise



Just Noise is a unique collection of vintage media noise, static, buzz and hum. Just right when you need some authentic background crackles, whine, fuzz or hiss, and also ideal as a starting element for sound design. This 2 CD collection contains 140 tracks, representing more than 20 different sources of great old fashioned noise, including:

Need a way to age, weather or add realism to a track? Try some garbled reception, static and hum from classic 1940s and 1950s radios, or a bit of crackling noise from a 1960s tape recorder. Add a bit of transistor buzz, a sprinkle of LP record noise or some pulsing synthesizer pops. You'll also find a great selection of tape hiss, walkie talkie static and TV noise on this 2 CD collection.

- neon signs • open wires • tape recorders
- scanners • LP records • walkie talkies
- televisions • tube & transistor radios
- wax cylinders and gramophone discs
- analog synthesizers • reel to reel / cassette tapes

Just Noise (2 Audio CDs) **209.95**

Series 11,000 Sports



The Sports Series offers extreme closeups of individual hits, impacts and catches, full competitive game action recorded at the sidelines, and the sounds of spectators, crowds and stadiums in a wide selection of sports activities. Recorded with professional, semi-pro and advanced collegiate athletes, it features both indoor and outdoor stadium crowd sounds of up to and more than 100,000 spectators. Five CDs of heart-pounding effects, with more than 30 different sports represented. Included are international favorites like soccer and tennis, recreational sporting activities like skateboarding and golf, and all the power & crunch of the major leagues - hockey, football, baseball and basketball.

S11-01: Badminton, Baseball, Softball, Basketball, Mountain Bike

S11-02: Mountain & Road Bicycles, Billiards, Bowling, Boxing, Croquet, Curling

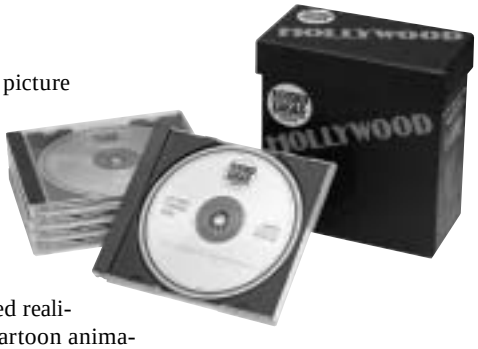
S11-03: Curling, Exercise Equipment, Football, Golf, Gymnastics, Ice Hockey

S11-04: Inline Hockey, Martial Arts, Paintball, Ping Pong, Racquetball, Skateboarding, Ice Skating

S11-05: Cross Country & Downhill Skiing, Snowboarding, Snowshoeing, Soccer, Swimming Dives & Races, Tennis, Volleyball, Wrestling

Series 11,000 Sports (5 Audio CDs) 495.00

Series 4000—Hollywood



If you've ever craved those well-known motion picture sounds for your productions, Series 4000 delivers! 2,200 effects on 5 CDs, you'll find digitally recorded cartoon boinks, zips and pops, created by North America's top cartoon talents. You'll feel chills when you hear the flesh ripping in the horror volume, and you'll want to hit the deck when the Uzi discharges 20 rounds in one second. Whether you want the heightened reality of a foley stage, or absurd sounds of classic cartoon animation, Hollywood is the only choice.

4001 Cartoon: Boinks, Bongs, Dits, Bings, Boings, Bounces, Warbles, Impacts, Cymbals, Gongs, Falls, Squishes, Splats, Drips, Pops, Bubbles, Rattles, Grinds, Squeaks, Bells, Horns, Whistles, Zips, Whizzes, Twangs, Space Sounds, Accents, Crashes

4002 Cartoon: Accents, Drums, Percussion, Cars, Airplanes, Trains, Boats, Motors, Drills, Guns, Birds, Cats, Chickens, Chipmunks, Cows, Dogs, Donkeys, Ducks, Elephants, Frogs, Horses, Monkeys, Pigs, Sheep, Growls

Firearms: Airguns, Hand Guns, Rifles, Shot Guns, 9 mm Uzis, Mini Uzis, M-16s, AK-47s, Machine Guns, Holsters

4003 Cartoon: Laughter, Cries, Yells, Groans, Head Shakes, Vocals, Sneezes, Wheezes, Nose Blows, Gargles, Yawns, Snores, Kisses, Babies, Footsteps, Swishes, Body Punches, Face Punches, Face Slaps, Human Hits, Human Stabs, Whips, Human Bodyfalls, Glass Smashes, Auto Crashes, Metal Crashes, Horror Sounds, Explosions

4004 Music: Great Selection of Cartoon, Circus, and Classical Music

4005 Music: Wide Variety of Fanfares, Jazz, Marches, Rock, Horror and Space Music

Series 4000 Hollywood (5 Audio CDs).....425.00

SPECIALTY SOUND EFFECTS

Dynamic Range



Dynamic Range from Sonic Boon is an all digitally recorded gun sound effects library. It was recorded with an ear towards "bigger than life" dramatic, visual sound - sound so big you don't just hear it, you see and feel it. The collection consists entirely of gun sounds and related effects, including interior and exterior shots, bullet hits, ricochets, silencers and gun handling sounds. Dynamic Range is making noise!

Interior Shots:
Revolvers, Pistols, Rifle, Automatic Weapons, Random Fire

Exterior Shots:
Revolvers, Pistols, Shotgun, Assault Rifles, Automatic Weapons, Random Fire, Special Weapons

Foley /Handling:
Revolvers, Pistol, Rifles, Shotguns,Auto Rifles, Assault Rifles, By Effect

Dynamic Range (2 Audio CDs).....395.00

The Works – Ultimate Industrial Collection



Sonic Boon's The Works is about sounds of motion, action & force. 10 CDs with 2,870 effects, including such categories as pneumatics, buttons, switches, signals, levers, ratchets, gears, friction, stress, servo motors, small motors, small mechanisms, electrical effects, compartments, doors, machines, motors and engines, in addition to a diverse collection of backgrounds: interior work & manufacturing, packing, shipping, conveyors, exterior work, labor, industry, and constant tone environments.

WORKS 1: Pneumatics, Buttons, Switches

WORKS 2: Signals, Levers, Ratchets, Gears

WORKS 3: Friction, Stress, Servo Motors

WORKS 4: Small Motors, Small Mechanisms, Electrical Effects

WORKS 5: Compartments, Doors, Tape Machines

WORKS 6: Machines, Motors, Engines

WORKS 7: Backgrounds: Interior Work & Manufacturing

WORKS 8: Backgrounds: Packing, Shipping, Conveyors

WORKS 9: Backgrounds: Exterior Work, Labor, Industry

WORKS 10: Constant Tone Environments

The Works (10 Audio CDs) 495.00

Blow Tools — Wind and Air Movement Effects



A 12-CD sound effects library containing over 700 minutes of rushing, wailing, blustering, buffeting, groaning, rustling, droning, growling, shrieking and moaning, whistling, rumbling, whispering, sweeping and swooshing, breezing, blowing, howling, gusting and whooshing – you can create effects for any type of wind or air movement, including tunnels, air locks, ducts, objects falling or moving through air, and horror or sci-fi background effects. Each digitally recorded effect is clean and ready to use. Alone or layered — create a custom mood or environmental texture.

- Of the 263 Blow Tools winds, 239 are "perfect loops." The wind effects that are not pre-looped grow out of silence with ramps of varying length, then crescendo to full intensity and decrease to silence again. The wind effects also have a wide variety of character with varying texture, resonance, pitch, range, and activity.
- Blow Tools also includes more than 80 shorter blowing effects of amplitude variation - including flurries, gusts and quick bursts - plus 175 individual swooshes of 34 different types.
- Preview CD and documentation make it quick and easy to scan through the hundreds of wind textures. This special disc contains 5-second samples of each looped effect, plus an entertaining audio demonstration program. Use the Preview CD to determine the exact character or wind type you're looking for, then follow the documentation directly to the library CD containing the full-length effect.

Blow Tools (12 Audio CDs) 539.95

Sounds of War Authentic World War II Sounds



An authentic and historic gallery of period audio - single weapon, equipment and tank effects, aircraft sounds and effects, short scenario ambiences and full battle sequences (including an aerial attack, dogfights and an armoured assault). A complete World War II sound effects archive.

These sound effects had to be tracked down and salvaged from a wide variety of archival sources across Europe. All effects were then digitally enhanced through cedar processing and/or mixed into the longer scenario ambiences by skilled engineers with state-of-the-art equipment.

- A Hurricane fighter with its straining Rolls Royce Merlin engine taking off ...
- Heavy fire of an armored assault by Panzer Grenadiers, firing machine pistols ...
- Stukas screaming down on zig-zagging convoys while Royal Naval gunners throw up a constant wall of anti-aircraft fire
- Flying Fortresses thundering down their runways, hauling their full bombloads into the air, heading for the industrial heartland of Hitler's Germany
- Boots on cobblestones as infantrymen fight through northern France
- Crack of rifle fire and the rattle of heavy machine guns of German defenders

Sounds of War.....109.95

BIRDS & ANIMALS SOUND EFFECTS

Just Birds & Animals



Two fully digital CDs - an outstanding mini collection of isolated individual bird and animal species, plus a variety of key ambiences and backgrounds.

JBA 01:

Alligators, Barn Ambiences, Bats, Bears, Bees, Bird Ambiences, Blackbirds, Budgies, Chachalacas, Chickadees, Chickens, Cockatoos, Crows, Doves, Ducks, Eagles, Falcons, Flamingoes, Geese, Hawks, Hummingbirds, Killdeers, Loons, Macaws, Mockingbirds, Owls, Parrots, Peacocks, Pigeons, Roosters, Seagulls, Sparrows, Starlings, Sun Conures, Turkeys, Vultures, Warblers, Whippoorwills, Woodpeckers, Wrens, Yellowthroats, Domestic Cats

JBA 02:

Country Ambiences, Crickets, Dolphins, Donkeys, Elephants, Piranhas, Flies, Frogs, Goats, Gorillas, Horses, Insects, Katydids, Jaguars, Jungle Ambiences, Leopards, Lions, Marsh Ambiences, Monkeys, Penguins, Pigs, Prairie Dogs, Sea Lions, Sea Otters, Seals, Rattlesnakes, Squirrels, Tigers, Walruses, Whales

Just Birds & Animals (2 Audio CDs) **209.95**

Just Birds & Animals II



With more than 340 sound effects, Just Birds & Animals II adds lots of wildlife to your productions. More animals, birds and natural settings from around the world, including:

- baboons, chimpanzees, gibbons, gorillas, lemurs, monkeys, orangutans
- lions, tigers and bears
- seals and whales
- hippos, hyenas, jaguars, rhinos and wolves
- buffalo, elk, horses and moose
- cows, donkeys, pigs and sheep
- Flocks of birds - buzzards, peacocks, roadrunners and many more
- Wide selection of domestic cats and dogs and great natural ambiences
- Badgers and wolverines, snakes, dragons and Tasmanian devils!

Just Birds & Animals II (2 Audio CDs) **209.95**

Wild World of Animals — Rare, Hard to Capture Animal Sounds



Wild World of Animals features rare, hard-to-capture sounds on 10 CDs with over 850 effects. Everything from aquatic animals (sharks, whales, dolphins, walruses), wild animals (buffalo, alligators, elephants, snakes, tigers, pandas), and birds & insects to domestic animals (cattle, goats, dogs, cats) plus an entire CD of horses!

Wild Animals I:

Buffalo, Camel, Elephant, Elk, Giraffe, Impala, Moose, Panda, Rhino, Wildebeest, Zebra

Wild Animals II:
Bear, Bobcat, Cougar, Leopard, Lion, Ocelot, Panther, Tiger, Wildcat, Jackal, Wolf

Wild Animals III:
Chimp, Gorilla, Monkey, Orangutan, Bat, Rabbit, Agouti, Rat, Frog, Alligator, Crocodile, Snake, Locust, Bee, Fly, Mosquito

Aquatic Animals I: Dolphin, Sea Lion, Seal, Walrus, Shark, Whale

Aquatic Animals II: Whales

Birds I: Birds in different environments; Amazon, Belize, North America

Birds & Insects I: Birds in different environments; Residential, Rural, Africa; Crickets, Cicadas

Birds & Insects II: Marsh and Jungle Ambiences from Around the World

Domestic Animals I: Cattle, Donkey, Goat, Pig, Sheep, Dog, Cat

Horses I: Many Gaits on Various Surfaces

Wide World of Animals (10 Audio CDs).....**495.00**

AUDIENCE REACTION / AMBIENCE

Audience Reactions — Applause, Laughter & Cheering



These two jam-packed CDs contain more than 500 audience reactions - just what you need for sit-com post-production and multimedia applications! Audience Reactions from Sound Ideas is a unique collection with a vast amount of studio audience material, complimented by larger arena & sports stadium crowds, and by children's reactions.

AR-01:

Small & Medium Sized Studio Audiences	Rhythmic Clapping in Unison
Large Stadium Crowds	Stomping & Clapping, Applause & Cheers
Applause, Standing Ovations, Cheers & Whistles	Surges of Applause, Whistling & Foot Stomping
Male & Female Applause	'Bravo!', 'Encore!' & 'Yea!', Rowdy Cheering

Audience Reaction (2 Audio CDs) **215.00**

Audience Reactions II — Vocal and Emotional Reactions

Features a wide range of responses from fresh new audiences: live theatre, various studio groups - and more children too. With a generous mixture of crowds, just male or just female audiences, and groups of children of different ages, Audience Reactions II offers applause, cheering, anger and booing, laughter, screams, and a full palette of vocal and emotional reactions from many different sets of voices. Reserve your seat in the audience today - Audience Reaction II is ready to strut its stuff in sitcoms, multimedia, and post-production of all kinds. Jam packed with more than 500 effects, these two CDs ring with genuine ovations and responses - your ticket to providing professional performances.

AR-02:

Small & Medium Sized Studio Audiences	Thoughtful & Surprised Reactions, Gasps
Large Stadium Crowds	Relief & Disappointed Reactions, Disgusted Reactions
Studio Audience of Children	Hecklers & Shushes, Positive & Negative Reactions
Booing & Hissing, Angry Yelling, Cheers, Laughter	Reactions of Disbelief, Shocked & Surprised Reactions
Laughter & Applause, Sympathetic Reactions	Trumpet Fanfare & 'Charge!', Happy New Year Countdown
Male & Female Laughter & Applause, Amazed Reactions	Screams: Fear, Panic & Adoration, Children's Reactions

AR-03:

Medium Sized Studio Audiences, Applause, Standing Ovation & Re-Seating, Cheers & Whistles, Male & Female Applause, Rhythmic Clapping in Unison, Applause & Cheers, Surges of Applause, Rowdy Cheering, Surprise!', 'Hip, Hip, Hooray!', 'Charge!', 'Go Team Go!', Sympathetic & Disappointed Reactions, Excited & Disbelief Reactions, Frightened, Panic and Horror Screams, Chanting 'Hell No, We Won't Go', Laughter & Applause, Booing & Hissing, Angry Yelling, Disgruntled and Riotous

AR-04:

Small Sized Studio Audiences, Male Only & Female Only Reactions, Studio Audience of Children, Booing & Hissing, Angry Yelling, Cheers, Laughter, Laughter & Applause, Sympathetic Reactions, Male & Female Laughter & Applause, Surprised Reactions, Shocked Gasps, Relief & Disappointed Reactions, Disgusted Reactions, Hecklers & Shushes, Positive & Negative Reactions, Shocked & Surprised Reactions, Happy New Year Countdown, Screams: Fear, Panic & Adoration, Children's Reactions, Chant 'Go Team Go' & 'De-Fence', 'Hip Hip Hooray'

Audience Reaction II (2 Audio CDs) **215.00**

Series 3000 — Ambience I



Series 3000 provides you with a wide range of ambience effects. Every cut is four minutes long to reduce the need for looping and editing. If you're involved in video or film post production, Ambience I is an indispensable creative tool that will save you time and money.

This unique collection of 12 CDs boasts over 200 stereo effects covering the following categories:

3001	City Atmospheres				
3002	Parks, Construction, Harbors	3006	Outdoor Crowds		
3003	Jungles, Swamps, Mountains, Deserts, Forests	3007	Outdoor Crowds	3010	Restaurants, Bars, Offices
3004	Country & Residential	3009	Indoor Crowds	3011	European Ambiences
3005	Highways & Airports	3009	Indoor Crowds	3012	Room Tones

Series 3000 (12 Audio CDs) **425.00**

B&H PHOTO - VIDEO - PRO AUDIO
ONE HOUR FREE PARKING
AT 349 W. 34th STREET (with purchase of $100 or more)

AMBIENCE

Series 7000 — Ambience II



Offering more than 350 contemporary ambiences on 15 CDs, Ambience II provides more variety than ever before. This collection of sounds includes six CDs exclusively devoted to extensive coverage of international ambiences, and also features a "Scenes" CD: a great new Sound Idea. Each track in the library is digitally recorded and all are exactly 3 minutes in length.

7001 Barns, Countrysides, Forests, Jungles, Marshes, Mountains, Oceans, Storms, Winds

7002 Airports, Arcades, Ballrooms, Banks, Bars, Basketball, Beaches, Billiard Halls, Boardrooms, Bowling, Fairs

7003 Bus Stations, Casinos, Cities, Construction, Conventions

7004 Indoor Crowds (Small, Medium & Large), Outdoor Crowds (Medium & Large)

7005 Harbours, Hockey, Hospitals, Household, Industries, Libraries, Lobbies, Markets, Office Foyers, Parades

7006 Parks, Parking Lots, Parties, Playgrounds, Residential, Restaurants, Room Tones

7007 Room Tones, Schools, Service Stations, Ice Skating, Stadiums

7008 Stock Exchanges, Stores, Subways, Swimming Pools, Symphonies, Theatres, Train Stations

7009 Scenes CD: Barns, Battles, Caves, Christmas, Cities, Courtrooms, Dinosaur Parks, Earthquakes, Fight Scenes, Forest Fires, House Fires, Horror, Police Accidents, Steam Train Stations, Western Streets

7010 Foreign Ambiences from: Australia, Austria, Belgium, China, Czech Republic

7011 Foreign Ambiences from: Czech Republic, Denmark, Egypt, England, France

7012 Foreign Ambiences from: Germany, Hong Kong, Hungary, Iceland, India

7013 Foreign Ambiences from: India, Italy, Japan, Korea, Luxembourg, Malaysia

7014 Foreign Ambiences from: Mexico, Nepal, Netherlands, Poland, Russia, Scotland

7015 Foreign Ambiences from: Scotland, Spain, Switzerland, Taiwan, Thailand, Uruguay

Series 7000 (15 Audio CDs) **425.00**

Series 10000 — Ambience III



The Ambience III collection from Sound Ideas provides your passport to 28 different countries and national regions - 300 full length ambience tracks on 14 CDs. Also included: 5 CDs that place you in the heart of the Big Apple - New York City - a comprehensive selection of backgrounds, ambiences and signature sounds.

And, there are more than 60 additional tracks of forest, water, rural, rain, residential, city, construction, industry, traffic, crowd, room tone, office and other backgrounds. In all, the Series 10,000 Sound Effects Library packs in 850 minutes of exceptional ambience sounds.

S10-01 Barns

Foreign Ambiences from: Australia, Belgium, Bolivia, Brazil

S10-02 Cities, Construction, Countrysides

Foreign Ambiences from: Brazil, China

S10-03 Indoor and Outdoor Crowds, Forests

Foreign Ambiences from: Czech Republic, England

S10-04 Harbours

Foreign Ambiences from: France, Germany,

Guatemala, Hawaii, Hungary

S10-05 Industries

Foreign Ambiences from: India

S10-06 Foreign Ambiences from: Italy, Luxembourg, Mexico, Netherlands

S10-07 New York Ambiences - Airport, Bank, Bar, Boat, Bus Terminal, Chinatown, City

S10-08 New York Ambiences - City, Construction, Crowds

S10-09 New York Ambiences - Horse, Hotel, Library, Market, Office, Park

S10-10 New York Ambiences - Park, Restaurant, Store, Subway, Traffic

S10-11 New York Ambiences - Traffic, Train Station

S10-12 Post Offices, Rain

Foreign Ambiences from: New Zealand,

Pakistan, Philippines, Poland

S10-13 Residential, Restaurants, Room Tones, Schools, Traffic
Foreign Ambiences from: Russia, Switzerland, Thailand, Turkey

S10-14 Water Ambiences: Falls, Oceans, Rivers, Streams

Foreign Ambiences from: Turkey, Venezuela

Series 10000 (14 Audio CDs) **425.00**

IMAGING ELEMENTS

Ear Candy I



Looking for something sweet in production and imaging elements? Check out Ear Candy, now available from Sound Ideas. This compilation of over 200 music themes, stabs, stingers, hits, percussion, work parts, lasers, openers, stagers, communications and sci fi FX offers an eclectic and humourous mixture of the routine, the strange and the unusual.

Music Tracks:

- Alternative Jam.....Free Spirited, Open, Raw
- Power Bar .....Raspy, Earthy, Hard Edged
- Beat Break.....Percussive, Confident, Upbeat
- Raw Energy.....Dark, Eerie, Restless
- It's Coming!.....Drone Track
- Space Trek.....Drone Track
- Wide Open Road.....Energetic, Driven, Untamed
- What a Drag .....Mechanical, Competitive, Tense
- Stepping It Up .....Energetic, Forceful, Motivating
- Coming to Get You .....Pulsating, Insistent, Overtaking
- Techno Stomp .....Gritty, Ominous, Futuristic
- Processing.....Robust, Throbbing, Percolating
- Cosmic Query .....Melancholy, Plaintive, Wafting
- Pipe Organ & Stabs.....Amen Finale / Brass Stabs
- Intro.....Synth News Intro
- Rock & Percussion .....Rock Stingers & Percussion

FX:

- Camera Shutter FX & Crowds
- Electricity, Electronics & Explosions
- Fax Machine & Microphone Feedback
- Garburators
- Machine Guns
- Hits & Industrial Sounds
- Radio Static
- Sci Fi Beeps, Hits, Sweeps, Tones & Airlocks
- Laser Shots, Phasers & Zaps
- Sci Fi Malfunctions, Plasma Conduits & Vending Machines
- Sci Fi Death Rays, Vaporization & Overloads
- Sci Fi Ship Tractor Beams, Engine Misfires, Take Offs & Landings
- Alien Howls & Snarls with Ship Landings
- Whooshes
- Radio Communications
- Electronic Drones, Grinds & Beeps
- Robot Sneezes & Warp Bubbles
- Hydraulics, Force Fields & Chirps
- Space Winds, Alien Attacks & Pressure Valves
- Sweeps, Zaps, Short Circuits & Sparks
- Heavenly Bodies
- Tape Recorder Tape Drags & Telephone FX

Ear Candy I **109.95**

Ear Candy 2



The Ear Candy series continues with even more goodies to satisfy your sweet tooth - a host of alternative rock theme music tracks, other music beds, percussion, logos, electronic stingers and accents and sound effects, plus the "ultimate" whoosh collection - 406 elements in all!

- A Selection of Alternative Rock Themes
- Musical Beds, Accents, Logos & Stingers
- Electro-Acoustic Stingers, Logos, Accents & Hits
- Electronic Stingers & Accents
- Electro-Acoustic Whooshes
- Whoosh Effects
- Sound Effects

Ear Candy 2.....**109.95**

Ear Candy 3



The third CD in this series of spectacular and highly practical imaging elements comes all packed and ready to go with 350 tracks in all! Included are a wide selection of special effects, more whooshes, sweeps, zaps and lasers, electronics and communications effects, and a great collection of high impact sound effects, ready for use just about anywhere!

- Special Effects & Hits
- Electronic Stingers & Accents
- Whooshes, Sweeps, Zaps & Lasers
- Electronics & Communication Effects
- Sound Effects

Ear Candy 3.....**109.95**

Ear Candy 4



More than 500 sizzling imaging elements and sound effects! Offers a tasty selection of accents, logos & stingers, communications, drones, hits, lasers, rewinds, scratches, servo motors, special fx, tones & telemetry, and whoosh effects, including:

- Dance and Groove Beds
- Bursts & Flutters
- Crowds
- Drum Hits
- Electricity & Electronic Pulses
- Explosions
- Laser Shots, Bursts and Blasts
- Record Scratches and Groove Noise
- Ripping, Metallic, Snapping, Spiral and Flaming Whooshes
- Audio Tape Rewinds
- Space Drones
- Station IDs
- Stereo Arc
- Telemetry & Beeps
- Transmission Bursts
- Warning & Overload Signals & Sirens

Ear Candy 4.....**109.95**

IMAGING ELEMENTS

Ear Candy 5



Ear Candy 5 packs a punch with a concentrated selection of 250 well produced imaging elements designed to add serious spice to your productions. These evocative, multi-dimensional audio parts will surprise you and challenge your creative side. his cutting-edge CD showcases:

Low in Calories ... High in Impact...

Ear Candy 5 is a definite "must have" for your production menu.

- Beds: 18 beds for news, sports, traffic and weather
- Comedy Stagers: 9 versatile comedy accents
- IDs & Logos: 15 pre-assembled stagers & sweepers
- Telephone Beds & FX • Hits • Audio Punctuators
- Guitar Hits & Stingers • Rewinds & Scratches
- Texture Drones & Stingers • Special F

Ear Candy 5.....**109.95**

Ear Candy 6



Ear Candy 6 - another delicious feast of imaging elements and music tracks, served up just the way you like them.

For starters, Ear Candy 6 opens with more than eighty musical beds, stingers, loops and percussion grooves - a buffet of Dance, Rock and Electronic themes. The second course features over one hundred imaging element house specialties: Dissolves, Electronic Swipes, Sweepers, Whooshes, Lasers, Hits, Zaps, Zips and Special FX.

Ear Candy 6 from Sound Ideas. Two hundred more tracks of hot and tasty imaging elements and vivid musical ingredients - just the right spice for your productions!

Ear Candy 6.....**109.95**

Ear Candy 7



Ear Candy 7 - loaded with 436 new audio ingredients for your productions. This CD is packed with a tremendous variety of dynamic imaging elements, including:

- Music Riffs, Beds, Logos & Segues
- Accents & Enhancers
- Fly Bys, Takeoffs & Landings
- Engines & Machines
- Electronic Alarms, Telemetry, Pulses & Beacons
- Beeps, Bonks & Boinks
- Space Wakes, Lasers & Cannons
- Whooshes & Special FX

Ear Candy 7 from Sound Ideas. Just what your recipe calls for!

Ear Candy 7.....**109.95**

Mix III



The Mix III Broadcast Music Library offers a superb package of almost 1,000 imaging elements of virtually every description. Features a wide variety of music and sound effects tracks – designed specifically for broadcast production use.
This 5-CD set is stuffed full of specialty beds, stingers, intros, segues, fanfares, underscores and drones .... imaging elements to pack a punch and spark new life into your productions.

Contemporary ... Versatile ... Dynamic ...

The Best Value in a Production Element Package

Music Categories:

Mix #21 - Rock Beds, Dance, Hip Hop & Funk, Stingers, Bumpers, Licks and Segues

Mix #22- Logos, FX and FX Music

Mix #23 - Mixed Bag: News, Weather, Sports, Fanfares, Comedy, Country, Jazz, A/C, Electronic, Drama Segues and Drama Underscores

Mix #24 - Stagers, Sweepers, Lasers and Drones

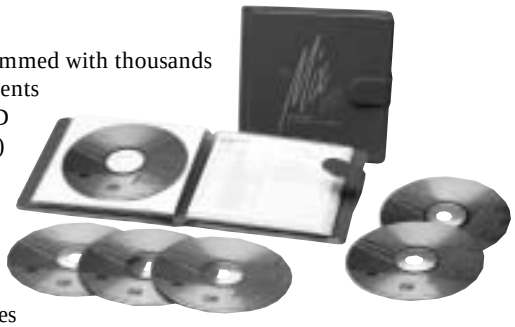
Mix #25 - IDs, Work Parts, MiniBeds and Promo Beds

The Mix Music Library offers you ...

- Broadcast lengths • Royalty-Free purchase • Unique storage binder
- Wide selection of musical styles, organized for easy access
- Full reference catalog with CD track index listing
- Music by a variety of composers

Mix III (5 Audio CDs).....**334.95**

Mix VIII



A 5 CD series jammed with thousands of imaging elements (and a whole CD of sound effects) – especially commissioned from many of the best music talents in the business. Includes the following very popular element styles:

•Hits •Sweepers •Communications •Telephone, Fax & Modem Tones •Sounders •Drones •Record Scratches •Music & Drum FX •Rewinds •Solo Rock Guitar Hits, Riffs, Runs & Licks •Doors •Swipes •Wipes •Dissolves •Ambient Pads •Loops •Whooshes •Electronic Stops, Hits & Kicks •Zips & Zaps

Mix VIII (5 Audio CDs).....**419.95**

IMAGING ELEMENTS

The Big Whoosh!



The Big Whoosh was the first imaging elements packaged ever released. Sound Ideas has now purchased the rights to this CD - an extravaganza of whooshes, production elements, audio signatures and always in demand sound effects for radio, broadcast, multimedia, game developers and Web designers.

Whooshes:
long, medium, short, flanged, whispy, echoing & rumbling
Music Sounders:
pulses, news tags, logos & closers
Space:
star showers, spooky & dangerous FX, encounters, manoeuvers, takeoffs & landings
Plus:
accents, explosions, heartbeats, hits, jail doors, sparkles, telephones, zings, zaps, funny stuff & much more . .

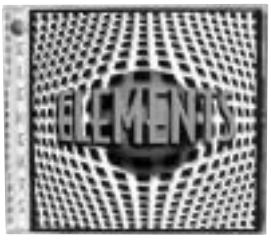
Big Whoosh.....**109.95**

The Big Whoosh 2



Catch the rush. The Big Whoosh Two from Sound Ideas - a special delivery parcel of long, medium and short whooshes, enhancers, phased and alien whooshes, drones, whoosh rewinds and impacts, fly bys, takeoffs & landings, special FX and music logos.

Big Whoosh 2.....**109.95**



Elements

The Elements CD: This single CD of more than 220 digitally recorded elements for radio, television and multimedia includes music beds, logos, hits, accents, explosions, atmospheres, zaps, sizzles, percussion instrument effects, and a series of different fax machine tones. Also featured in the collection are alien and ghost effects, plus suspense, danger and horror elements.

Elements **109.95**

Imaging Accents



Liven up your productions with Imaging Accents - a complete orchestra section of musical flourishes, trills, glisses and fanfares.
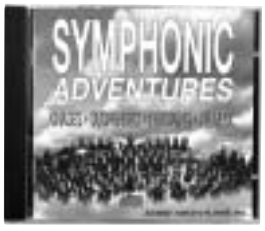
Thirty different percussion instruments, plus drum rolls, shots, rhythms and marches. Everything from bell trees and bongos to whistles, wood blocks and xylophones. And ... many different comedy accents, horns and crashes.

- Bells
- Bell Trees
- Bongos
- Cabassas
- Chains
- Chimes
- Claves
- Concert Bass Drums
- Cowbells
- Cymbals
- Flexitones
- Glockenspiels
- Gongs
- Guitars
- Harps
- Orchestra Bells
- Pianos
- Ratchets
- Rattles
- Shakers
- Spinners
- Spoons
- Tambourins
- Tambourines
- Triangles
- Vibraphones
- Vibraslaps
- Whistles
- Wood Blocks
- Xylophones

Plus ... a complete drum kit - shots, fills, rhythmic accents, tom rolls, snare rolls, jazz and syncopated rhythms, jungle beats, marches, rock rhythms and solos, and a giant selection of timpani rolls, rhythms, fanfares, accents, bounces and wobbles.

And ... comedy accents of all descriptions - chases, circus themes, crashes, spooky effects, horns, marches, mouth harps, trains, trombones, trumpets and tubas.

Imaging Accents (2 Audio CDs).....**Call**



Symphonic Adventures

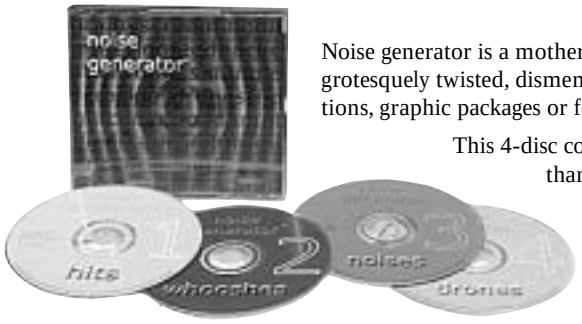
Create your own high quality soundtrack with this professional interactive toolkit! Specially recorded for film, video & multimedia use. BPM (beats per minute) furnished.

Whole orchestra and individual instrument segments include: strings, brass, woodwinds, pianos and percussion.

Designed for music editors and film composers, this CD is a symphonic construction tool kit - any chosen segment can be looped with another, or used as a new start point.

Symphonic Adventures **109.95**

IMAGING ELEMENTS



Noise Generator

Noise generator is a motherlode of organically-based real sound design elements — a huge variety of real sounds grotesquely twisted, dismembered, smacked, & verbally abused. Sounds that would perfectly accent video transitions, graphic packages or footage of exploding livestock – creative, attention getting elements.

This 4-disc collection of sound design elements was designed to sound COMPLETELY different than the tired, old early 90's FM-radio-ish lasers, white noise sweeps and synthesizer stuff that puts most of us to sleep. Our mision was to create a collection of sounds that would stand out and physically knock people off their feet.

Most of the 1,875 sounds on these discs started as real sounds, ie: banging the inside of a dumpster with a hammer, dropping a manhole cover (trust us: that's LOUD) - even stuff like taking microphone feedback, processing the feedback, then feeding it back against itself again. We took these elements and processed the living daylights out of them using digital audio workstations and state-of-the-art digital signal processors (plug-ins and outboard gear). Finally, they were mastered through high-end gear to end up with a ton of sounds that resulted in an incredible sounding product - but incredibly unhappy neighbours.

Comes with 4-color coded, fully-indexed audio discs and a 40-page booklet, organized by type and sound length.

Noise Generator (4 Audio CDs) **995.00**

Slam, Bang, Boom



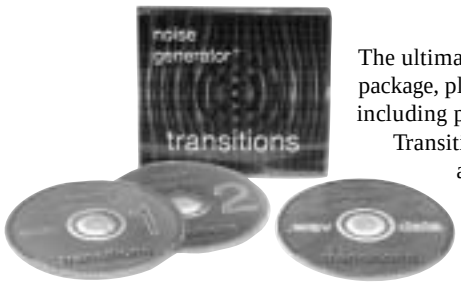
Slam*Bang*Boom is a 2-CD, 1257 element sound design package, plus a corresponding CD ROM of WAV files. Over 89 categories of hits - including explosions, bursts, computer and Internet sounds, flashes, & bowel-loosening thunder and lightning hits, the second set in the 'Noise Generator' collection of sound design products is dedicated to making the world a far more sonically disturbing place.

More hits than Elvis and The Beatles combined! Useful for anything that uses sound - from radio to video to multimedia. Tools for accenting cuts, titles, flashes, exploding watermelons and for getting your audience's attention.

The sounds on this collection come from a variety of sources, some real and some synthesized - mostly designed to sound different than anything else out there.

Comes with 2-color coded, fully-indexed CD, one CD ROM and a 24-page booklet, organized by type and sound length. Contains 16-bit, 44.1k, WAV files and is both Mac and PC-compatible. This CD ROM should also work in AKAI S5000/6000 series samplers with OS1.3 or later as well as any EMU or Kurzweil samplers that can import .WAV files.

Slam, Bang, Boom (4 Audio CDs & 1 CD-ROM) ..... **Call**



Transitions

The ultimate arsenal of whooshes & transitional elements! Transitions is a 2-CD 1229 element sound design package, plus a corresponding CD ROM of WAV files. Crammed with over 100 categories of transitions - including powerups, beams, electrostatics, fly-bys, vinyl sounds, rumbles, evolvers & grainy breakups.

Transitions is the third set in the 'Noise Generator' collections of sound design products, giving producers an edgier, more offensive, in-your-face selection of sound design.

First, a definition: a 'Transition' is defined (in almost 98% of the cases in this collection) as an event with a sloped attack that builds to an abrupt climax - or builds to a climax and then recedes. Good for aurally accenting wipes, fades, dissolves - or gaping holes in your audio that require covering up. Of course, radio can also utilize these sounds for sweeps and transitions, or just make their productions sound very, very different than what's out there now. The sounds come from a variety of sources, some real and some synthesized – designed to sound different than anything else out there.

Comes with 2-color coded, fully-indexed audio CDs, one CD ROM, and a 24-page booklet organized by type and sound length. The CD ROM contains 16-bit, 44.1k, WAV files and is both Mac and PC-compatible. This CD ROM should also work in AKAI S5000/6000 series samplers with OS1.3 or later as well as any EMU or Kurzweil samplers that can import .WAV files.

Transitions (2Audio CDs & 1 CD-ROM) ..... **Call**

FEATURE FILM SOUND EFFECTS COLLECTION

Amadeus Sound Effects Library

Produced by San Francisco's Saul Zaentz Film Center, this outstanding collection of audio from the Academy Award-winning feature film offers a deep selection of effects that won the Oscar for Best Sound, including:

Crowd Sounds: - applause, whispers, yells, laughter, murmurs & movements

Doors:- hits, jails, knobs, knocks, rattles & squeaks

Horse & Carriage Sounds: - walking, whinnies, snorts, pulling carriages & wagons, pass bys, exteriors & interiors

Human Sounds: - yells, whistles, sighs, lunatic babbling, screams, moans, laughs, grunts, chokes, cries, breathing & coughing, ***PLUS—***

• Indoor & Outdoor Ambiences, Thunder, Rain and Wind Machine Effects

• Twangs, Hits & Scrapes • Bass & Timpani Drums

• Props, Live Theatre Foley & Other Theatrical Effects

LucasFilm — Academy Award Winning Sound Effects

Lucasfilm, the world's greatest innovator of sound design and Sound Ideas,the world's largest creator of sound effects have joined forces to produce a unique six CD sound effects collection. Now, for the first time ever, you can have Academy Award winning sound effects originally used in many of LucasFilm's major motion pictures, plus new effects specially created by Sound Ideas for this incredible co-production.

LF01 Animal Sounds
LF02 Elements (Wind, Rain, Fire)
LF03 Industry Sounds
LF04 Aircraft Carrier
LF05 Helicopter & Private Jet
LF06 Porsche 911 & Corvette

Twentieth Century Fox Classic Motion Picture Sounds

Digitally remastered, this 10 CD library boasts over 1,350 sound effects representing years of legendary motion picture history.

TCF 1: Antique Autos, Auto Crashes, Ferraris, Hot Rods, Model Ts, Stock Car Races, Auto Skids

TCF 2: Airplanes, Animals, Arrows, Archery, Crowds, Baseball, Battles

TCF 3: Bells, Bodyfalls, Cartoon Boings, Boxing Crowds, Cameras, Carnivals, Catapults, Crashes, Crowds

TCF 4: Drawbridge, Earthquakes, Electronics, Explosions, Fight Scenes, Bar Fights, Fires & Fire Trucks, Flame Throwers, Professional Football

TCF 5: Glass Smashes, Cannons, Firearms, Handcuffs, Naval Horns, Horses & Stagecoaches

TCF 6: Human Hits, Jail Doors, Jungles, Knife Stabs, Knife Throws, Metal Hits

TCF 7: Motorcycle Gangs, Vintage Motorcycles, Office Ambiences, Parachutes, Police Radios, Ratchets, Record Scratches, Rockets, Sand Storms, Ship Signals, Ship SONAR, Space Shuttles

TCF 8: Air Raid & Ship Sirens, Snowmobiles, Artillery Soldiers, Cavalry Soldiers, Soldiers Marching, Roman Soldiers, Green Hornet Stings, Submarines, Submarine Control Rooms, Submarine Torpedoes, Sword Fights

TCF 9: Military Tanks, Telegraphs, Telephones, Thunder, Period Traffic, Trains

TCF 10: Steam Trains, Trolleys, Military Trucks, Underwater Sounds, Western Streets, Police Whistles

Universal Studios—Classic Motion Picture Sounds

Universal Studios, in cooperation with Sound Ideas, has made available for the first time ever, an awesome collection of over 1,000 sound effects on five compact discs. These are classic sound effects, straight from the heart of some of Universal Studios' most famous and timeless feature film presentations. This amazing collection of sound effects is like getting access to Universal Studios during the shooting of some of their most monumental feature films ever.

US01: Single Prop Airplanes, Biplanes, Triplanes, Twin Prop Airplanes, Four Engine Airplanes, Eight Engine Airplanes, Airport Ambience

US02: Aviation, Animals, Avalanches, Human Bodyfalls, Boxing, Comedy, Crashes

US03: Doors, Electricity, Explosions, Fires, Human Hits, Horns, Knife Throws, Metal Hits, Face Punches, Space Doors, Space Gun Shots, Space Ships, Swords, Telegraphs, Underwater Sounds, Volcanos, Werewolves, Whips

US04: Grenades, Military Cannons, Rockets, Whistle Bombs, Explosions, Depth Charges, Torpedoes, Submarines, Tanks, Swords, Sword Fights, Marching, Hand Guns, Muskets, Rifles, Ricochets, Machine Guns, Military Battles

US05: Steam Trains, Indians, Western Battles, Stagecoaches, Horses Walking, Horses Trotting, Horses Cantering, Horses Galloping, Cavalry Horses

Turner Entertainment— MGM Motion Picture and Cartoon Sounds

Presenting an epic collection of motion picture and cartoon sounds - the Turner Entertainment Co. SFX Library is a superb collection of sound effects originating from their vast collection of MGM major motion pictures, such as: 2001, Dr. Zhivago, Ben Hur, The Dirty Dozen and the classic cartoons of Tom and Jerry. The Library consists of 3 CDs, and contains over 500 sound effects, all digitally remastered. The CDs come in an attractive travel holder, displaying a fascinating collage of authentic old movie stills originating from many of MGM's motion pictures.

TE01: Airplanes, Ambulances, Animals, Cartoon Sounds, Creatures, Monsters, Archery, Auto Crashes, Auto Races, Civil War Battles, Revolutionary Battles, Sci Fi Battles, Western Battles, World War II Battles

TE02: Crowds, Doors, Earthquakes, Electronics, Explosions, Karate, Sci Fi Fights, Hand Guns, Machine Guns, Sci Fi Guns, Gun Silencers, Western Guns, Human Hits, Horror

TE03: Parachutes, Sci Fi Sounds, Ships, Space Ships, Storms, Air Raid Sirens, Sword Fights, Military Tanks, Steam Trains, Time Machine, Volcanoes, Weather, Winds

Disney I.d.e.a.s—Great SFX

A super CD of over 150 digitally recorded sound effects created by the recording professionals at Disney i.d.e.a.s., the elite post-production facility on the backlots of the Disney MGM Studios in Florida. This is a must have CD of ambiences, animals, sports, transportation, machinery, cartoon sounds, and special effects.

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ROYALTY FREE MUSIC

Mix Music Library — Full Length Music Tracks (2- 3 Minutes)

Mix V



Mix V offers over one hundred full length tracks of today's most contemporary and dynamic music. 5 CDs, more than five hours of top quality production music, all full length and royalty-free.

Wide selection - 18 different musical styles, organized for easy access. Music by a variety of composers. Full reference catalog with CD track index listings for all the Mix Libraries. Packaged in a compact storage binder.

Mix #38: Industry, Corporate

Mix #39: Bright, Light Activity, Leisure Romantic, Pastoral

Mix #40: Action, Sports, Pop, Rock, Dance, Urban

Mix #41: New Age, Classical, Christmas & Holidays, Solo Instrument

Mix #42: Country and New Country, Drama, Suspense, Comedy, Cartoon, Jazz & Blues, Children, Military Marches, World Music

Mix V (5 Audio CDs) **419.95**

Mix IX



This 10 CD package offers 250 rhythm tracks in a variety of versatile music categories. Ideal for productions where the music can't overpower an important announcement.

With The Mix IX Rhythm Track Library, you've got first rate music that won't interfere with the voiceover.

$419.95

MIX CD #65: Rock - 25 Rhythm Tracks

MIX CD #66: Dance - 28 Rhythm Tracks

MIX CD #67: Percussion 18 Rhythm Tracks

MIX CD #68: New Country 28 Rhythm Tracks

MIX CD #69: Sports & Industry 28 Rhythm Tracks

MIX CD #70: Romantic & Soft Adult Contemporary 24 Rhythm Tracks

MIX CD #71: Soft Adult Contemporary 27 Rhythm Tracks

MIX CD #72: Adult Contemporary 24 Rhythm Tracks

MIX CD #73: Lite Jazz 24 Rhythm Tracks

MIX CD #74: Jazz 24 Rhythm Tracks

Mix VI



Mix VI features 150 full length and royalty-free music tracks on 5 new CDs. The result of quality contributions from 20 professional composers, Mix VI spans an impressive 13 high demand music categories.

Mix #43: News, Sports, Hard Rock

Mix #44: Dance & Bright, Light Activity

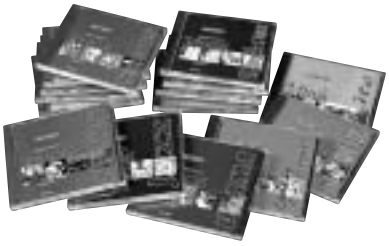
Mix #45: Country, Western, Jazz

Mix #46: Comedy, Children, Solo Piano

Mix #47: Classical, Christmas

Mix VI (5 Audio CDs).....**419.95**

Turner Music Library



Never before commercially available, the Turner Music Library breaks loose with outstanding professional quality production music, exclusively distributed by Sound Ideas:

- Eight versatile categories
- Fully orchestrated themes
- Over fifty world class composers
- Incredible variety of musical styles and selections
- Signatures, logos, openers, underscores, and more

$499.95

Disc	Category	# of Themes	# Of Tracks
TBM-101	Comedy & Cartoon	27	103
TBM-201	Drama 1	28	86
TBM-202	Drama 2	45	70
TBM-301	Jazz	23	44
TBM-401	Lite 1	37	77
TBM-402	Lite 2	13	96
TBM-501	News 1	13	95
TBM-502	News 2	22	88
TBM-503	News 3	19	53
TBM-601	Rock & Dance	21	62
TBM-701	ShowBiz	22	96
TBM-801	Sports 1	9	84
TBM-802	Sports 2	6	93
TBM-803	Sports 3	9	94
TBM-804	Sports 4	10	63

ROYALTY FREE MUSIC

The Mix Music Library — Broadcast Length (30-60 Seconds)

• Broadcast lengths • Royalty-Free purchase • Wide selection of musical styles, organized for easy access
• Full reference catalog with CD track index listing • Music by a variety of composers • Unique storage binder

Mix I

The Mix I Broadcast Music Library offers more than 10 hours of production-ready material, totalling over 1200 tracks plus a full CD of sound effects... all on 10 compact discs. This premier collection contains 2 CDs of Production Elements, including: Logos, Work Parts, Promo ID Beds, Weather Themes, Sports Themes, Stingers, Newscaster Openers, Sweepers, Traffic Themes, Bumpers, Tags, Orchestra Hits & Stabs, Harp Music, Halloween Accents, Circus Accents, and Bugle Calls, Drum Hits & Rolls.



Mix #1: Easy Listening & Classical
Mix #2: Soft A/C & Travel
Mix #3: Sports
Mix #4: Rock, Pop, Dance
Mix #5: Country & New Country
Mix #6: Jazz & Comedy
Mix #7: Christmas & Holidays
Mix #8: Production Elements 1
Mix #9: Production Elements 2
Mix #10: Sound Effects

Mix I (10 Audio CDs).....**419.95**

Mix II



The Mix II Broadcast Music Library presents over 10 more hours of great production music, totalling more than 850 tracks ... all on 10 compact discs.

Mix #11: Easy Listening
Mix #12: Soft A/C & Travel
Mix #13: A/C #1
Mix #14: A/C #2 & News
Mix #15: Rock
Mix #16: Rockalikes, Period Rock, Pop
Mix #17: Dance, Techno, Funk
Mix # 18: Action, Sports
Mix #19: Mixed Bag: Country, Comedy, Electronic, Jazz and Drama
Mix #20: Christmas

Mix II (10 Audio CDs).....**419.95**

Mix III



The Mix III Broadcast Music Library offers a superb package of almost 1,000 imaging elements of virtually every description. Featuring a wide variety of music and sound effects tracks, Mix III was designed specifically for broadcast production use. This 5 CD set is stuffed full of specialty beds, stingers, intros, segues, fanfares, underscores and drones .... and a whole lot more! Spark new life into your productions.

Mix #21: R ock Beds, Dance, Hip Hop & Funk, Stingers, Bumpers, Licks and Segues
Mix #22: Logos, FX and FX Music
Mix #23: Mixed Bag: News, Weather, Sports, Fanfares, Comedy, Country, Jazz, A/C, Electronic, Drama Segues and Drama Underscores
Mix #24: Stagers, Sweepers, Lasers and Drones
Mix #25: IDs, Work Parts, MiniBeds and Promo Beds

Mix III (5 Audio CDs).....**334.95**

ROYALTY FREE MUSIC

Mix IV



The Mix IV Broadcast Music Library offers a host of music themes and sound effects - all royalty free. With more than 14 hours of first class production music and premium quality sound effects on 12 CDs, Mix IV boasts more than 1500 tracks and over 400 themes.

Mix #26 Romantic, Leisure
Mix #27 Bright, Light Activity
Mix #28 Bright, Light Activity
Mix #29 Industry, Drama
Mix #30 Country, Western
Mix #31 Children, Comedy
Mix #32 Action, Sports, Pop
Mix #33 Dance, Urban, Rock
Mix #34 Classical, Holidays
Mix #35 Jazz, Ethnic
Mix #36 Military, Period
Mix #37 Sound Effects

Mix IV (12 Audio CDs) **419.95**

Mix VII



Mix VII is a breath of fresh air for the airwaves, an inspired new mix of sixty, thirty and fifteen second music tracks. With over 1200 tracks on twelve CDs, and a full thirteen hours of music from a variety of composers, Mix VII provides great music in 16 high-demand categories, for any broadcast production need.

Mix #48 Adult Contemporary
Mix #49 Adult Contemporary
Mix #50 Easy Listening, Christmas
Mix #51 Pop, Percussion
Mix #52 Dance, Electronica, New Age
Mix #53 Rock
Mix #54 Sports
Mix #55 Jazz
Mix #56 Industry, Corporate
Mix #57 Country
Mix #58 Comedy, Children
Mix #59 Classical, National, Ethnic

Mix VII (12 Audio CDs).....**419.95**

Mix VIII



Mix VIII is a 5 CD series jammed with thousands of imaging elements (and a whole CD of sound effects) that you can't afford to be without. Especially commissioned from many of the best music talents in the business, this collection's quality is superb and the variety is impressive. Mix VIII includes the following very popular element styles:

• Hits • Sweepers • Communications • Telephone, Fax & Modem Tones
• Sounders • Drones
• Record Scratches • Music & Drum FX • Zips & Zaps
• Rewinds • Solo Rock Guitar Hits, Riffs, Runs & Licks
• Doors • Swipes • Wipes • Dissolves • Ambient Pads
• Loops • Whooshes • Electronic Stops, Hits & Kicks

19.9595.00

Mix X



The Mix Ten Broadcast Music Library is fresh and happening - more than 740 tracks in a great range of must-have categories: Rock, Soft Rock, Dance, Electronica, Solo Guitar, Country, Blues, Jazz, Drama and World music.

Mix X (12 Audio CDs) **419.95**

Ultimate Drums & Bass (Full Length)



Simply the best resource for underscore themes and work parts. This loaded CD contains over 90 tracks of playful, comic, dramatic, world, light attitude, extreme, rock and funk themes, over 50 tracks of riffs and grooves in six different categories, 13 different loopable phrases, and more than 65 broadcast edit tracks in nine categories.....**250.00**

ROYALTY FREE MUSIC

Mix Signature Collection Combo (Full & Broadcast Length)

Looking for a full set of Mix Signature Collections? Your search has ended! The Mix Signature Collection Combo offers the entire series at an unbeatable combo price! The Mix Signature Collection Combo gives you the very best of The Mix - in 12 great music packages, bundled together for easy use and maximum value:

Rock Mix	with Just a Little Attitude	2 CDs
Eazy Rock Mix	Lots of Funk & Soul	2 CDs
Corporate Mix	Music That Means Business	2 CDs
Sports Mix	Pumped, Powerful and Proud	2 CDs
Eazy Listening Mix	Bright, Fresh, Warm and Gentle	2 CDs
Lite & Easy Mix	Relaxed, Carefree & Breezy	2 CDs
New Age Mix	A New Wave for a New World	2 CDs
Country Mix	Great Country & New Country	2 CDs
Lite Jazz Mix	Casual & Contented Moods	2 CDs
Jazz Mix	Jazz & Blues Just for You	2 CDs
Classical Mix	A Varied Classical Selection	2 CDs
Holiday Mix	Traditional & Pop Arrangements	2 CDs



The Mix Signature Collection Combo - 24 CDs, 380 themes, 1,480 broadcast and full length tracks - it's Mix Heaven!

OR—BUY THEM INDIVIDUALLY

Rock Mix

Driven, punchy, kicking and cool - with just a dash of attitude. Rock Mix delivers 33 themes - a solid serving of electric guitar, percussion and saxophone tunes to gear up and drive your productions. Go the distance with this varied collection of heavy, Texas and southern rock, splashed with a dash of surf and blues. The Rock Mix - gutsy, energetic, rugged and rebellious.

Eazy Rock Mix

A little bit hip - a little bit bold; a smattering of funk - a sprinkle of soul! Eazy Rock Mix gives you 33 themes to stroll, saunter, swing and roll positive energy throughout your productions. Featured instruments in this collection include acoustic & electric guitars, piano, organ, saxophone, flute and fiddle, and there's also a smattering of rap and soul to keep it cool. The Eazy Rock Mix - carefree, confident & laid back.

Corporate Mix

Corporate Mix delivers 30 bold, dynamic and dominant themes that feature orchestral, brass and sampled sound arrangements. Up tempo, progressive and ready to work - this is music that means business. Promote your productions with the Corporate Mix - it gets results.

Sports Mix

Sports Mix - a winning combination of 34 themes that breaks a sweat and counts the score. The Sports Mix offers driving, competitive and exhilarating compositions that feature brass, electric guitar and orchestral arrangements. Knock em out with the Mix Signature Sports Mix - an adrenaline rush that really delivers.

Eazy Listening Mix

Something bright & fresh; something warm & gentle. Eazy Listening Mix offers 31 themes - reflective, warm and soothing arrangements designed to relax, revitalize and recharge your batteries. The Eazy Listening Mix features instruments like acoustic & electric guitars, alto, tenor & soprano saxophones, electric and traditional pianos and flutes - a soft set that lets you get mellow, release your cares & unwind.

Lite and Easy Mix

Bright, flowing and carefree. Lite & Easy Mix - a fresh selection of 30 themes designed to keep things light and breezy. This 2 CD set provides a soothing selection of reflective, mellow and gentle compositions, featuring solo and acoustic instrument arrangements. Drift into a daydream with the Lite & Easy Mix.

New Age Mix

2 CDs and 120 tracks of electronic style music that echos the rhythms of a modern age. Pulsing and abstract .... hypnotic and ethereal - the New Age Mix offers a varied sample of several electronic styles in this selection of original synthesized music. Explore new territory with the New Age Mix - music for a new world.

Classical Mix

Refine your productions with this varied classical selection of 23 different public domain compositions, offered in a full compliment of broadcast lengths. Classical Mix graces your productions with popular classical music from Mozart, Bach, Vivaldi, Tchaikovsky, Beethoven, Brahms, Dvorak, Pachelbel and others. Add a little elegance and polish to your productions with the Classical Mix.

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Lite Jazz Mix

Smooth, spirited & carefree - with a casual & contented mood. Lite Jazz Mix presents 34 sophisticated, stylish, breezy and flirtatious themes - ready to sound notes of optimism and introspection. With instrumentation that includes acoustic & electric guitars, alto, tenor & soprano saxophones, pianos, flutes, clarinets, trumpets, brass & trombones, the Lite Jazz Mix has everything you need to accent the hours of early mornings, easy afternoons and late evenings.

Jazz Mix

Mellow, seductive, sleek & cool - jazz & blues just for you. Jazz Mix offers a temptation of 34 different spirited, traditional, big band, swing & quartet jazz tunes, plus a selection of cool, laid back blues - all designed to create productions with classic, sophisticated appeal. The Jazz Mix features a variety of music styles and instrumentations, including trumpets, saxophones, pianos, guitars, drums, vibes, horns, clarinets & trombones - something for every taste and always delicious.

Country Mix

Rugged, lively and down home - great country and new country selections to reflect today's tastes. Country Mix serves up 34 satisfying themes from the heartland, the rodeo, the badlands and the urban frontier. This selection of rustic, freewheeling, leisurely and wide open tunes provides dance, swing, open country and back porch flavours, and features a wide range of styles and instruments, including: steel, electric & acoustic guitars, fiddles, accordions, pianos, mandolins, dobros and harmonicas.

Holiday Mix

Celebrate Christmas, Hanukkah & the New Year with The Mix - a joyous gathering of 37 themes, both old and new. This selection of traditional and pop arrangements features many well-known holiday compositions and a small but satisfying feast of other seasonal music - perfect accompaniments to that special time of the year. Holiday Mix offers different and intimate arrangements of flutes, winds, harps, strings, guitars, pianos and percussion, and also the grandeur of orchestras and big bands.

SONOMIC ON-LINE LIBRARY CARD

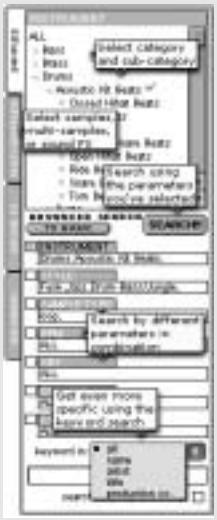
Choose From Over 200,000 Samples and Sound Effects



With the Sonomic On-line Library Card, you can customize your own sound library. Choose from over 200,000 sounds online- the latest loops, hottest sounds, and rarest FX, with thousands more added every week. Audition as many as you want and download exactly the sounds you need. All sounds are online and searchable in seconds. Select only the sounds you want from some of the best sample and sound FX libraries in the world, including Zero G, Sampleheads, Universal Sound Bank, Q Up Arts, Sound Ideas, BBC, Valentino and many more. With regular CDs, you have to pay and hope the disc contains sounds you can use. All of the samples and sound FX at Sonomic are royalty-free, so you only pay once, no matter how many times you use the sounds in commercial projects. Multi-samples are available in EXS-24, Samplecell and Unity formats. All other samples can be downloaded on ACIDized .Wav or .Aiff formats for immediate use with any type of hardware or software.

Choose the card that's right for you: One card offers 50 samples or 20 sound FX, and another card is available that offers 10 samples or 5 sound effects.

- Available 24 hours a day, 7 days a week
- Sonomic's search engine allows you to search by category or keyword. A few mouse clicks and you've created a custom list of the sounds you need.
- Every one of the sounds can be previewed before you buy them, so you always get exactly the sounds you need.
- Sonomic offers only professional-quality sound FX, samples, and multi-samples, which can be downloaded in any standard file format. Sounds are automatically stored in your SOUNDBAY enabling you to store, organize, access and re-download them from anywhere. All of the sounds stored in SOUNDBAY are secure and can be downloaded as many times as you like from wherever you're working.



Three Easy steps:

1) Go to www.sonomic.com and click on Register

2) Enter the Online Library code found on back of card

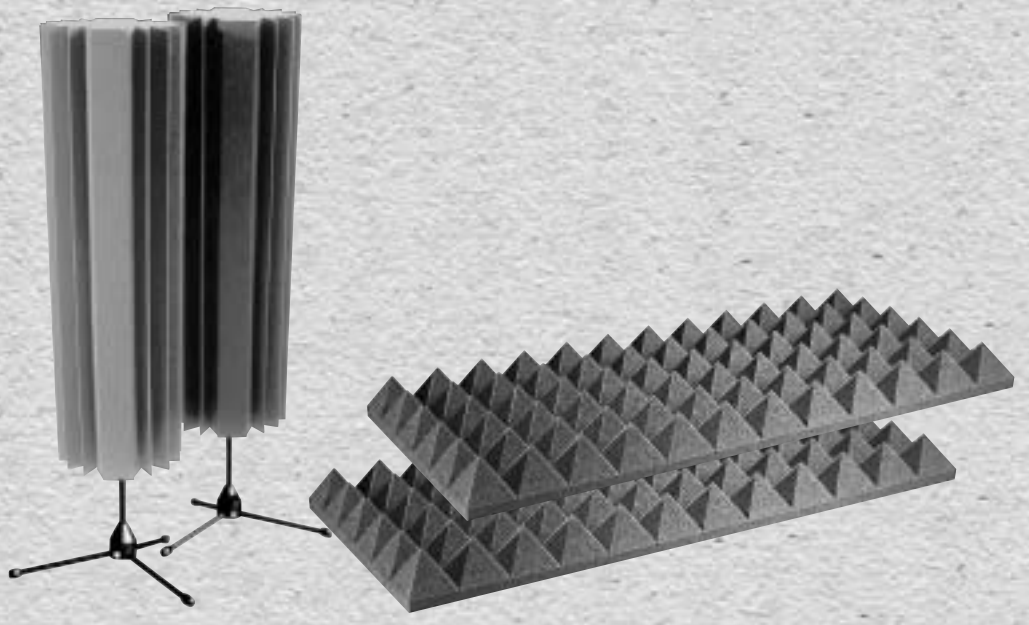
3) Search, preview and download exactly the sounds you need!

SOLC —
Library Card for Fifty (50) Samples OR Twenty (20) Sound Effects Download **29.95**

SOMLC —
Mini Library Card for Ten (10) Samples OR Five (5) Sound Effects Download **7.95**







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The ProfessionalAudio SourceBook

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Raxxess 1414-1432

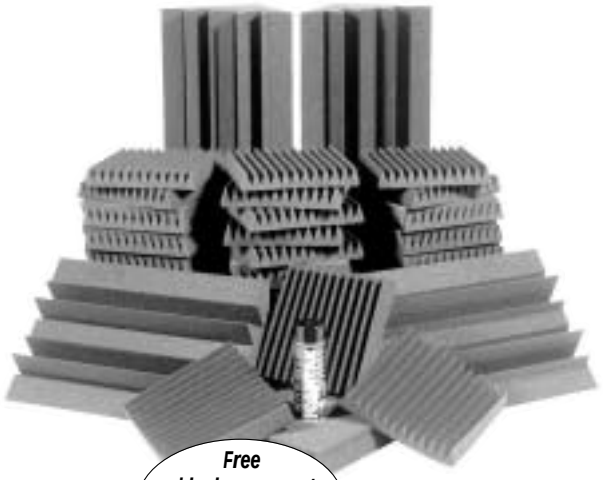
Ultimate.....1433-1439

AURALEX ACOUSTICS

ROOMINATORS

Complete Acoustical Control Kits

People are always looking for better sound. The trouble is, without fixing your room acoustics, none of your equipment will sound as good as it should. Your room won't let your gear's true sound come through. Auralex has taken the guesswork out of acoustics and made it easy and affordable to improve your sound. Auralex offers four Roominators kits that can easily tame the sound problems many people run up against in commonly-sized rooms. Everyone from the novice to the pro can sound their best....no formulas, no mumbo jumbo, no physics required. Now you can have the big studio sound you've been looking for, but didn't know where to find and probably thought you couldn't afford. Auralex has packed the Roominators kits with the same exact features and performance that many famous clients have loved for years.



Free shipping on most Auralex Acoustics products

Alpha 1 Roominator Kit

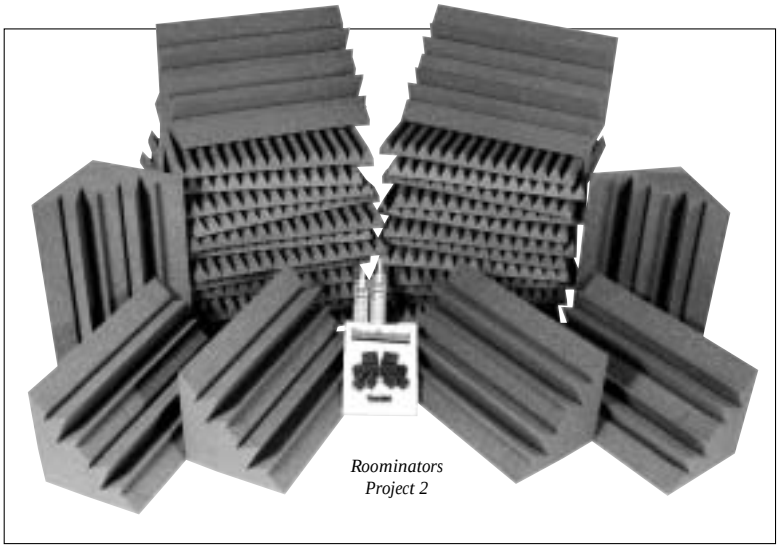
Roominators Kits contain:

Studiofoam 2″ Wedge Panels or Wedgies (Alpha1) to absorb early reflections and kill that hollow, "ringy" room sound.

LENRD Bass Traps to smooth out rooms' inaccurate and excessive low frequencies.

Foamtak or TubeTak Adhesive to mount Studiofoam and LENRDs with minimum hassles.

T'Fusor 3D Sound Diffusors *(Deluxe and Pro Kits only)* for that spacious ambience and wide sweet spot big-name studios have.



Roominators Project 2

- Improve the sound of everything you record, listen to or mix
- Reduce your monitoring volume - yet actually hear better
- Hear more of the subtle sonic details that your room's inaccurate reflections have been burying
- Achieve better sound out of your existing equipment
- Improve your engineering skills, be more creative and get more enjoyment out of your equipment and your room!

Alpha1 Roominators Kit

(64) 1′ x 1′ x 2″ Wedgies
(4) LENRD Bass Traps
(1) Foamtak Spray Adhesive
Alpha 1.....**Call**

Roominators Project2

(24) Studiofoam 2′ x 2′ x 2″ Wedge Panels
(8) LENRD Bass Traps
(5) TubeTak Pro Liquid Adhesive
Project 2.....**Call**

Roominators Deluxe Plus

(24) Studiofoam 2′ x 2′ x 2″ Wedge Panels
(8) LENRD Bass Traps
(6) T'Fusor 3D Sound Diffusors
(2) Foamtak Spray Adhesives
Deluxe Plus.....**Call**

Roominators Pro Plus

(36) Studiofoam 2′ x 2′ x 2″ Wedge Panels
(12) LENRD Bass Traps
(8) T'Fusor 3D Sound Diffusors
(3) Foamtak Spray Adhesives
Pro Plus.....**Call**

Roominators are available in charcoal gray.

AURALEX ACOUSTICS

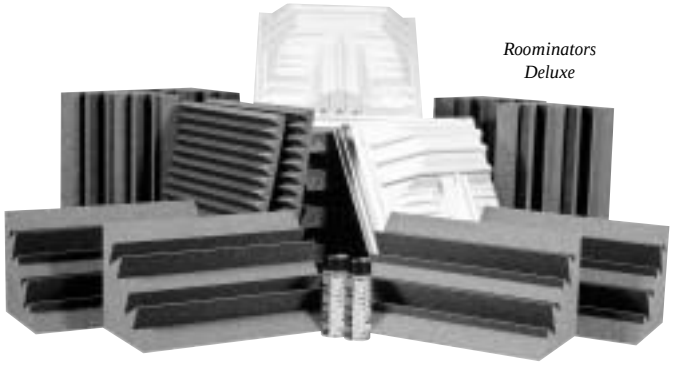
Free shipping on most Auralex Acoustics products

ROOMINATORS

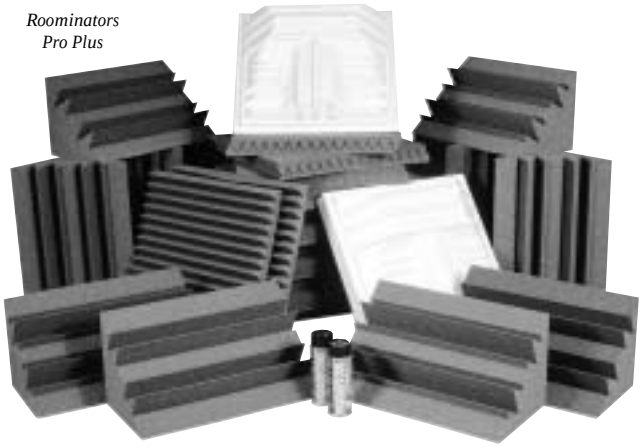
It's never been this easy or affordable to tame the acoustics in your small studio, iso room, voiceover booth or control room. Each kit includes instruction sheets with installation suggestions.

If you're the type who doesn't know a lot about the physics of sound control.....and doesn't WANT to know a lot about the physics of sound control.... then you owe it to yourself to check out all the Roominators kits: from the super-affordable Alpha 1 up to the top-of-the-line Pro Plus Kit. And, if you ever need to, simply add more Roominators kits as your budget and acoustical needs dictate.

See the chart below for some rough guidelines to help steer you toward the kit that's just right for you. And, if your needs dictate, don't forget that you can also "mix and match" your room's solutions by combining more than one Roominators kit (of the same or different types) or by adding some of the other Auralex sound control products.



Roominators Deluxe



Roominators Pro Plus

Roominators Kit	Square Footage (Area)	Shortest Room Dimension
Alpha 1	≤ 100′	6-8′
Project 2	≤ 100′	8-12′
Deluxe Plus	100-250′	10-14′
PRO Plus	250-400′	12-16′

ADHESIVES

TubeTak Pro Liquid Adhesive

TubeTak Pro is a super-strong, permanent bonding adhesive that applies easily with a standard caulking gun or the optional TubeTak Pro Applicator. One tube mounts up to 32 square feet of foam or diffusors, depending on thickness.



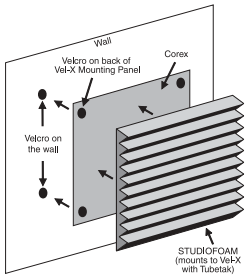
Foamtak Spray Adhesive

The fast and easy way to mount foam products. Unlike TubeTak Pro, which is a more permanent adhesive, Foamtak can be applied lightly so that your foam is removable, or it can be applied heavily for a more permanent bond. Foamtak offers greatly improved long-term bonding power, is guaranteed not to oxidize Auralex acoustic foams, and mounts approximately 96 square feet, depending on foam thickness.

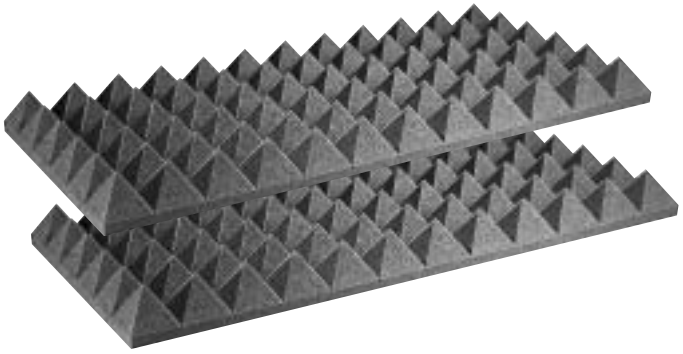


Vel-X Mounting

Vel-X panels are 22″ square corrugated polypropylene panels that temporarily mount Auralex acoustic foam or MetroFusors on walls. The foam is adhered to the Vel-X panels with TubeTak Pro adhesive, then special Velcro dots are used to stick the panels to the wall. The panels are movable at will, allowing you to tune the room for a particular project or take them with you when relocating to a new facility. Vel-X panels are shipped with both TubeTak Pro and special Velcro dots. Minimum shipment is twelve 22 x 22″ panels. Please note that 1″ needs 40 panels per box of foam, 2″ needs 24 panels per box of foam, 3″ needs 16 panels per box of foam and 4″ needs 12 panels per box of foam.



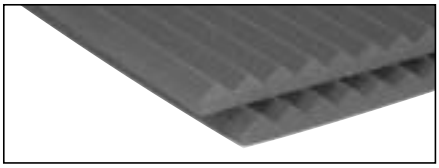
STUDIOFOAM

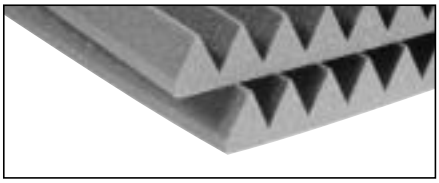


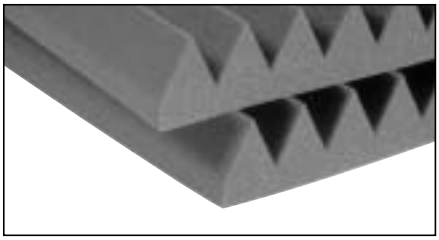
Wedges and Pyramids

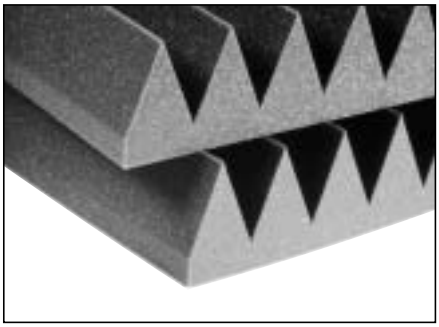
Studiofoam Wedges offer excellent absorption, a sleek, high-tech look and minimal installation seaming. They come in many colors and are available in 1, 2, 3 and 4˝ thicknesses and 2´ x 4´ sheets. Each sheet is precision-cut for clean-looking installations, so you won't find any innies or outies, or have to hassle with out-of-square panels. Consistency, increased flame retardancy, industry-leading absorption and a longlasting formulation add up to make Studiofoam wedges the hands-down value leader in the acoustics marketplace.

Studiofoam Pyramids offer a perfect combination of precision appearance and great sound control. With valleys running in both directions, they offer increased installation versatility (each pyramid is either 2˝ or 4˝ wide, making it simple to cut) minimal seaming, supply extra diffusion due to their 4-sided geometry and give your space tasteful, upscale looks and sound. Studiofoam Pyramids are manufactured in 2˝ and 4˝ thicknesses in 2´ x 4´ panels and are available in many colors. As they have less foam mass than Studiofoam wedges, Studiofoam Pyramids perform about 85% as well as Studiofoam wedges, but still outperform many competing brands.









1" Studiofoam Wedges

Perfect for environments that require good sound control, but where total dryness isn't required. 1˝ Studiofoam works most effectively on mid and high frequency sound waves and may be used to treat walls or, more commonly, ceilings (especially if they aren't parallel to the floor), even when the walls are treated with thicker Studiofoam. 1˝ Studiofoam absorbs as well as some competing 2˝ products, so if your budget is a bit thin, 1˝ Studiofoam might be just the ticket for you.

NRC: *.50* ***Qty:*** *20 panels* ***Coverage:*** *160 sq. ft.* ***Suggested adhesive per box:*** *2 Foamtak or 5 TubeTak*

2" Studiofoam Wedges

2˝ Studiofoam Wedges are Auralex's most popular seller and best overall performer. Use them to treat small- to medium-sized areas including vocal booths, control rooms and studios. They effectively kill standing waves and flutter echoes and, when used in conjunction with bass traps, or broadband absorbers, can effectively tame the full frequency bandwidth in virtually any room. 2˝ Studiofoam is the workhorse of the industry and is your safest bet if you're tuning your room without the help of a professional acoustician.

NRC: *.80* ***Qty:*** *12 panels* ***Coverage:*** *96 sq. ft.* ***Suggested adhesive per box:*** *1 Foamtak or 3 TubeTak*

3" Studiofoam Wedges

Twice as absorbent as 2˝ at 125Hz, 3˝ can do many of the same things 4˝ can do (especially when used with LENRD or Venus Bass Traps) and can provide a well-controlled, more accurate sound in any size room. Recommended for rooms with higher SPLs or more low frequencies such as drum or voiceover booths. Many rooms have low-end problems, so if you step up from 2˝ to 3˝, you will benefit from the extra low-end absorption 3˝ Studiofoam Wedges offer.

NRC: *.90* ***Qty:*** *8 panels* ***Coverage:*** *64 sq. ft.* ***Suggested adhesive per box:*** *1 Foamtak or 2 TubeTak*

4" Studiofoam Wedges

Recommended for medium to large areas like concert halls, gymnasiums and churches, rooms with pronounced low frequency problems or where sonic accuracy is mandatory and maximum absorption is required (e.g. voiceover or drum booths, forensic audio labs and mastering rooms). 4˝ Studiofoam Wedges provide 3x the low-end control of 2˝ and can effectively tame even the worst sonic anomalies. In some instances, using 4˝ Studiofoam can lessen the need for significant dedicated bass trapping.

NRC: *1.10* ***Qty:*** *6 panels* ***Coverage:*** *48 sq. ft.* ***Suggested adhesive per box:*** *1 Foamtak or 2 TubeTak*

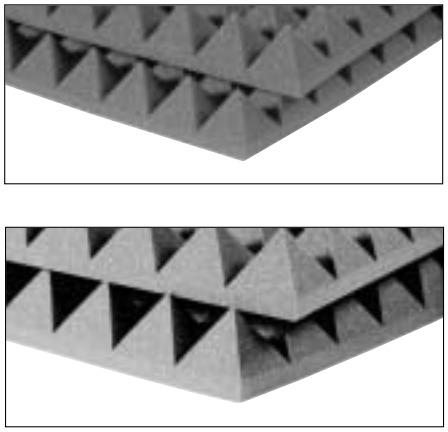
B&H PHOTO - VIDEO - PRO AUDIO EQUIPMENT LEASING AVAILABLE

STUDIOFOAM

2" Studiofoam Pyramids

Use these to treat small- to medium-sized areas including iso booths, control rooms and studios. They effectively kill standing waves and flutter echoes and, when used in conjunction with LENRD or Venus Bass Traps or Sunburst Broadband Absorbers, can tame the full frequency bandwidth in virtually any room. 2″ Studiofoam Pyramids offer a bit of extra diffusion and slightly less absorption than 2″ Studiofoam Wedges, so they yield a less dry-sounding space with a bit more "air.".

NRC: *.70* ***Qty:*** *12 panels* ***Coverage:*** *96 sq. ft.* ***Suggested adhesive per box:*** *1 Foamtak or 3 TubeTak*



4" Studiofoam Pyramids

As with 4″ Studiofoam Wedges, these 4″ Pyramids are recommended for larger spaces, rooms with pronounced low frequency problems or where sonic accuracy is mandatory and stronger absorption is required. Having 4 sides exposed on each pyramid also yields more sound wave diffusion, which is desirable in some spaces. Thus, 4″ Studiofoam Pyramids will yield less overall dryness than 4″ Studiofoam Wedges.

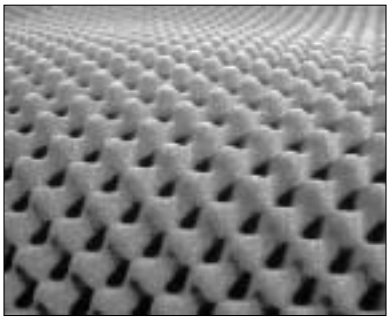
NRC: *.95* ***Qty:*** *6 panels* ***Coverage:*** *48 sq. ft.* ***Suggested adhesive per box:*** *1 Foamtak or 2 TubeTak*

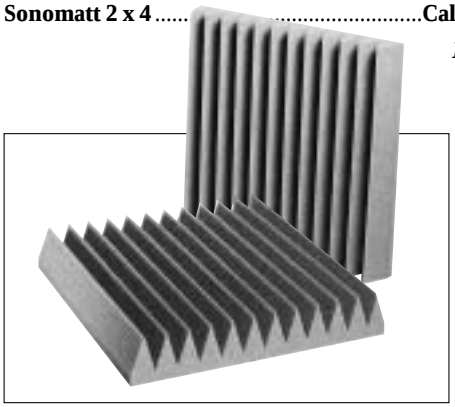
Sonomatt Acoustic Panels

The perfect product for the budget-conscious, yet acoustically-discerning customer. Sonomatt is cut in the industry standard "eggcrate" style to keep its price low and absorbs almost as well as 2″ Studiofoam Wedges, yet outperforms "premium" brands that cost way more. Please note that Sonomatt's dimple pattern will not be square to the edges of the panels, so installing them side by side may not be advised from a visual standpoint. Sonomatt is available in charcoal gray in 2′ x 4′ and 4′ x 8′ sheets and it may be the perfect choice for those who intend to cover their sound control materials with acoustical cloth.

Sonomatt 2 x 4 .....**Call** **Sonomatt 4 x 8** .....**Call**

NRC: *.70* ***Qty:*** *12 2x4's (96 sq. ft.) or 2 4x8's (64 sq. ft.)* ***Suggested adhesive per box:*** *1 Foamtak or 3 TubeTak*





Wedgies

Wedgies are 1′ squares of 2″ thick Studiofoam and are a great solution for spot treating studios, home listening rooms, iso booths and more. With slightly more wedges per square foot than 2″ Studiofoam, Wedgies feature maximized surface area for greater exposure to sound waves. Wedgies are a great solution for small flutter echo problem areas and, when spread apart a little bit, yield beneficial diffusion off their exposed edges. Available in charcoal gray only.

Wedgies, box of 24 .....**Call** **Wedgies, box of 96**.....**Call**

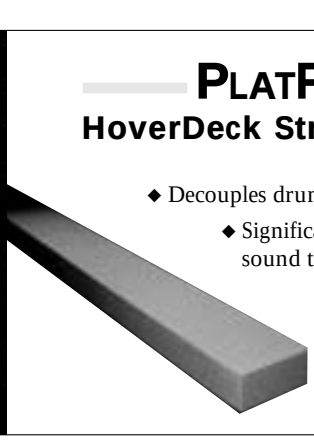
NRC: *.75* ***Qty:*** *24 or 96* ***Coverage:*** *24 or 96 sq. ft.* ***Suggested adhesive per box:*** *1 Foamtak or TubeTak/24 sq. ft. - 1 Foamtak or 3 TubeTak/96 sq. ft.*

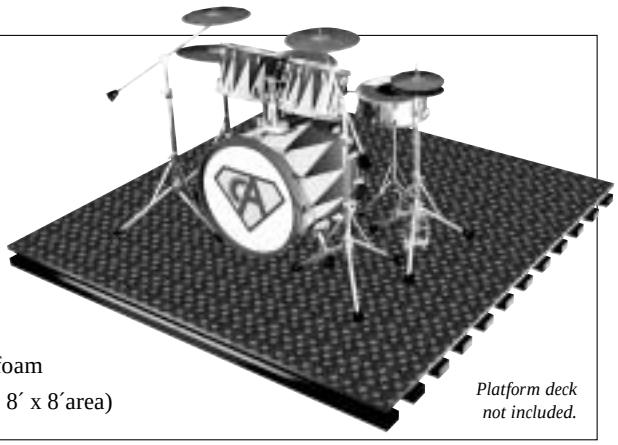
PLATFOAM

PLATFOAM

HoverDeck Structural Isolation System

- Decouples drums/amps/speakers from the room
- Significantly increases isolation, minimizing sound transfer to areas outside the studio
- Improves purity of tone and crispness of instruments
- 2″ x 4″ x 4′ pieces of high density foam
- 24 pieces per box (will float an 8′ x 8′area)





Platform deck not included.

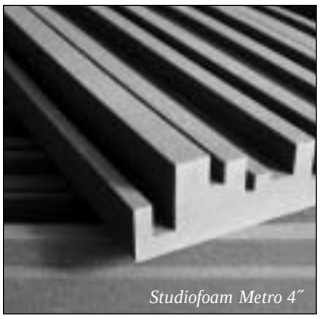
AURALEX ACOUSTICS

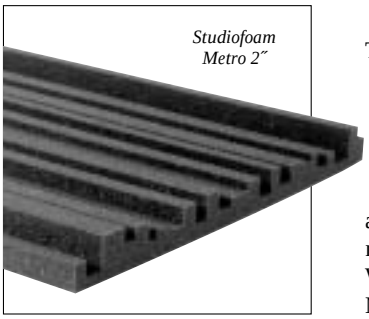
METRO FAMILY

Free shipping on most Auralex Acoustics products

The Metro Family blurs the line between those products that work well acoustically and those that yield the desired aesthetics that users desire. The Studiofoam Metro and the MetroFusor Diffusor afford minimal installation seaming because the left and right edges of the panel are the same height. They lend themselves to a variety of installations that look great and sound quite natural due to Metro's linear — and moderate — acoustical action.







Studiofoam Metro

The differing heights of protrusions on the Studiofoam Metro spreads sonic energy out in the time domain, as well as each absorbing a slightly different slice of the frequency spectrum. It takes slightly different amounts of time for sound waves to travel through the different sections of the Metro, strike the mounting surface and travel back into the room, so wavefronts are in effect softened rather than just being absorbed. This is why a Metro room will retain a bit more "feel" than a room that's treated with a stronger absorber like 3˝ or 4˝ Studiofoam Wedges or Pyramids and why, when coupled with a judicious amount of MetroFusors, a well-controlled, natural-sounding space will result.

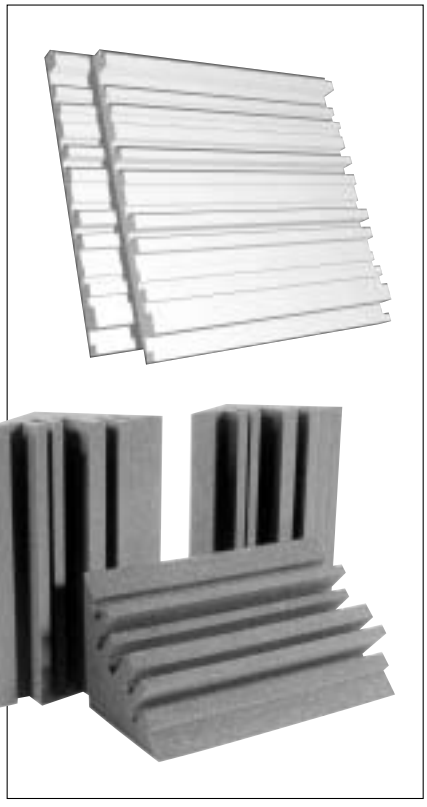
Studiofoam Metro 2˝
NRC: *.70* ***Qty:*** *12 panels* ***Coverage:*** *96 sq. ft.* ***Panel Size:*** *2´ x 4´ x 2˝*
Suggested adhesive per box: *1 Foamtak or 3 TubeTak* .....**Call**

Studiofoam Metro 4˝
Qty: *6 panels* ***Coverage:*** *48 sq. ft.* ***Panel Size:*** *2´ x 4´ x 4˝*
Suggested adhesive per box: *1 Foamtak or 2 TubeTak* .....**Call**

MetroFusor Sound Diffusor

As a companion to the well-received Metro 2˝ Absorber and Metro LENRD Bass Trap, Auralex has developed the MetroFusor. It's custom made from high quality EPS (expanded polystyrene), so it's lightweight, affordable, easy to paint, easy to mount and inexpensive to ship. It features the same distinctive, proprietary surface contour as the Metro Absorber for a clean, consistent, architecturally-pleasing appearance that looks upscale and attractive in even the most highbrow of environments. The MetroFusor's surface contour yields both sound wave refraction AND time domain spread for improved modal response without removing a significant amount of sonic energy from your space. It's perfect for churches, boardrooms, performance venues, studios and more. It's also a significant bass trap below 400Hz, which is quite a desirable attribute (because low frequency diffusion would only muddy your sound).

Quantity: *12* ***Size:*** *2´ x 2´* ***Coverage:*** *48 sq. ft.* ***Suggested adhesive per box:*** *2 TubeTak*



Metro LENRD Bass Traps

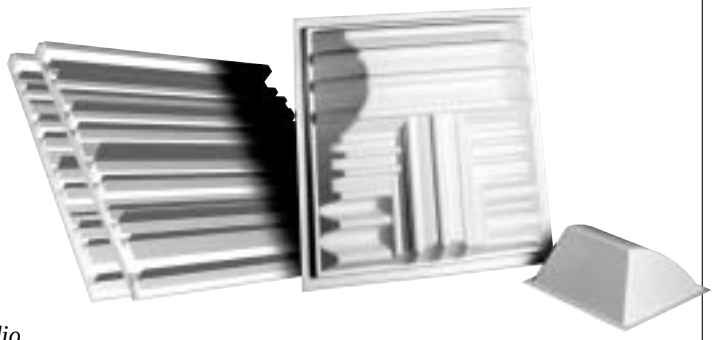
Metro LENRD Bass Traps round out the Auralex Metro family. This triangular-shaped bass trap fits into room corners and/or wall/ceiling junctures perfectly, and provides outstanding broadband absorption (especially at low frequencies). Based on the design of the wildly successful original LENRD Bass Trap, the Metro LENRD gives a fresh look to a reliable, unbeatable sound absorber. Plus, it's available in 12 colors.

All three Metro products work together to provide sound absorption, diffusion and bass trapping, while the "cityscape" cut provides visual continuity to your room. The Auralex Metro family was designed with a professional, architectural look for use in studios, classrooms, home theaters, office buildings, churches or anywhere a more upscale appearance is called for.

Quantity: *8* ***Coverage:*** *16 lineal feet* ***Size:*** *2´ x 1´ x 1´*
Suggested adhesive per box: *1 Foamtak or 2 TubeTak*

DIFFUSORS

In Which Rooms and Which Placement Is Diffusion Appropriate?



First of all, control rooms. Most studio designers will tell you that the front of the room (walls and ceiling) should be absorptive as far back as the engineer. The ceiling from the engineer back can contain a mix of diffusion and absorption, but many top designers feel the rear wall should feature a diffusor array surrounded by broad bandwidth absorption, especially in larger rooms. The side walls and ceiling from the engineer's position on back can be alternately absorptive and diffusive.

Second, live rooms (studios). Personal taste, room size and room function determine how dry (absorptive) or wet (diffusive) a studio should be, but many famous rooms are absorptive in some spots and diffusive in others. Generalizations: rock studios should be more absorptive than classical or jazz studios, and rarely should they be totally "dead." If you've got to be in a wet room, have the liveliness controlled by diffusion to yield a more pleasing and smoother overall room sound and reverb tail (decay).

Third, isolation and voiceover booths. Personal taste and room size come into play here, too. A voice talent sounds great in totally absorptive environments, while a drummer may prefer a combination of diffusion and absorption on all room surfaces with significant bass trapping. When recording sax players, violinists or solo sopranos, you might want some combination of the two; probably more diffusive than absorptive. It's truly your call, but bear in mind that once bad (non-diffused) "room sound" is captured on tape, disk, etc., you can never get rid of it.

Free shipping on most Auralex Acoustics products

T'Fusors

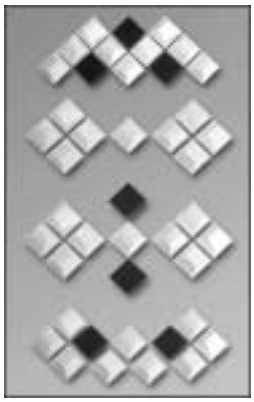


Made from high-impact thermoplastic resin, T'Fusors, which are 23.75″ square, drop perfectly into a suspended ceiling grid, and stand up very well to normal wear and tear. They are lightweight and nest into each other, dramatically lowering shipping costs. T'Fusors mount easily to walls and ceilings using construction adhesive or mechanical fasteners, and are sized to facilitate alternating them with acoustic foam panels for better overall room sound. They have a 1″ ledge in their back that permits insertion of SheetBlok, flat-cut acoustic foam, rigid fiberglass or Auralex Mineral Fiber insulation for greater versatility. They're available in easily paintable white.

MiniFusor Sound Diffusor



Just like its big brother the T'Fusor, the MiniFusor is very versatile and affordable. It's shape lends itself to various wall patterns and its cavity can be filled with acoustic foam or Auralex mineral fiber insulation to provide better diffusion and low frequency trapping. It is made from rigid thermoplastic resin and, like the T'Fusor, is super-easy to mount and paint.
Auralex MiniFusors offer a rare combination of great performance and a great price. So, if you're in the market for diffusors and are on a tight budget, you owe it to yourself to give serious consideration to a MiniFusor array.
MiniFusors are 12″ x 12″ x 5″ and have a quantity of 12 per box (12 sq. ft.).



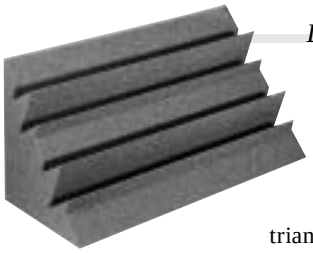
Four MiniFusor array suggestions shown with Wedgies

Where, and In What Quantities, Shoud You Install T'Fusors?

In most small control rooms it is recommended to install a group of at least 4 T'Fusors on the rear wall. Larger control rooms get at least four additional T'Fusors on the rear wall and, optionally, additional units placed on the ceiling from the mix position back. Discretionary placement: side walls behind the mix position, alternating with acoustic foam. Regardless of room size or function, groups of less than 4 T'Fusors are not audibly worthwhile. Also, for T'Fusors to work properly, the pattern should not repeat sideways or vertically in close proximity to itself.

LENRDS

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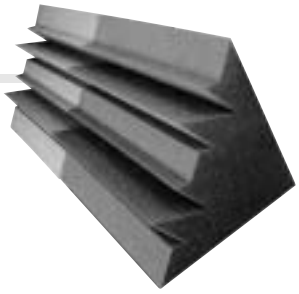
LENRD Bass Traps

Low frequency sound waves are the most prominent and difficult to control, whether you're attempting to block their transmission to a neighboring space or trying to absorb them to clean up the low frequency response. LENRD's triangular shape traps bass buildup by putting a big chunk of Studiofoam right where you need it: in your trihedral corners (where two walls come together and meet the ceiling).

Each LENRD kit contains eight 2′ tall LENRDs, enough to trap 16 lineal feet. If you've got 8′ ceilings, one box would be enough to trap both your room's rear corners floor to ceiling. Or, if space permits, you could trap all four corners of your room from the ceiling down 4′. Due to LENRD's efficiency, you can achieve premium results by utilizing thinner Studiofoam for the bulk of your wall treatment.

Suggested Adhesive: *1 Foamtak or 2 TubeTak per box*
Standard LENRD NRC: *1.35*

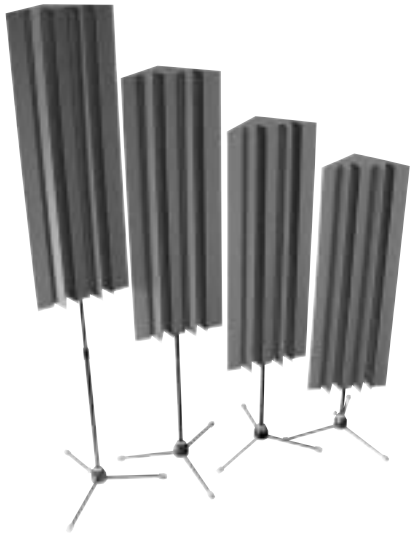
MegaLENRD



At twice the width (extending 2′ along each wall, not 1′ like the standard LENRD), the MegaLENRD is substantial enough to handle low frequency problems in rooms of any size.

Quantity Per Box: *2* ***Colors:*** *All*

Stand-Mounted LENRDs

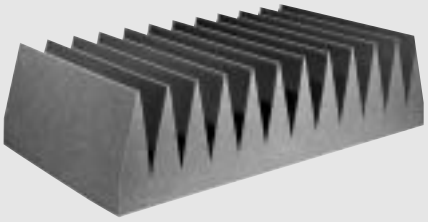


Stand-Mounted LENRDs are 4′ tall instead of the standard 2′, and come complete with the same stands utilized with Sunburst-360s. In "normal" mode, the LENRDs stand about 6′ tall, but in "extended" mode, they can reach heights of nearly 8′ or more. Stand-Mounted LENRDs are the perfect complement to MAX-Wall Modular Acoustical Environment. They're portable, effective, highly affordable and because they're stand-mounted, there's no adhesive to worry about and no wall repair to do if you move and take them with you.

LENRD NRC: *1.35*
Quantity Per Box: *Four 4′ LENRDs complete with 4 MAX-Stands*
Colors: *Charcoal gray, vivid purple and burgundy*

VENUS BASS TRAPS

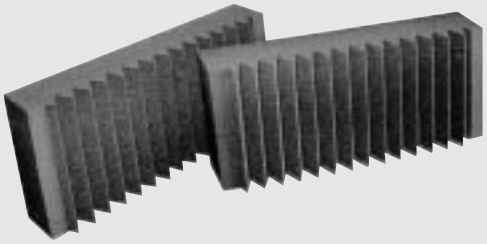
Venus Bass Traps



Many smaller studios and control rooms suffer from low frequency anomalies not only due to their lack of adequate trapping in general, but also due to their dimensions. Therefore, you may have tons of bass one place, but if you move 6″, you've got none at all.

In a studio, it's easy enough to move a performer, amp or drum kit to a different spot. But in a control room, where everything's fixed in location, such standing wave occurrences become tougher to deal with.....and much more important based on the nature of what a control room is for. The Venus achieves a prodigious level of low frequency absorption at your room boundaries—where low frequency problems begin. It exhibits a phenomenal coefficient of 1.63 at 125Hz, and ships in a 2′ x 4′ x 12″ size. It is often cut in half to 2′ x 2′, then paired with a 12′ Auralex CornerFill. Venus not only works well in smaller rooms, it really shines in larger rooms like gymnasiums, houses of worship and multipurpose rooms.

NRC: *1.30 Qty: 2 (16 sq. ft.)* ***Suggested Adhesive Per Box:*** *1 Foamtak or 2 TubeTak*



MegaMAX

Sometimes, studios or tracking rooms have very pronounced bass anomalies that can't be totally mitigated by conventional placement of bass traps. The 2′ x 4′ MegaMAX panels are 9″ thick and have a hole cut through them so that they can be mounted on MAXStands, enabling them to be moved easily to "fine tune" difficult low-end problems. Their portability allows them to be used as low frequency go-bo's. The MegaMAX can be placed in problematic areas unique to particular applications.

NRC: *1.10, 1.32@125Hz* ***Qty:*** *2 (16 sq. ft.) Adhesive and MAX-Stands are optional*

B&H PHOTO - VIDEO - PRO AUDIO
ONE HOUR FREE PARKING
AT 349 W. 34th STREET (with purchase of $100 or more)

SUNBURSTS



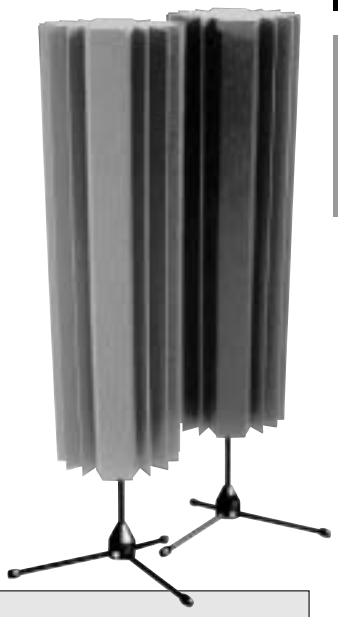
Sunburst Broadband Absorbers

Sunbursts feature a smart and unique male/female configuration that helps solve quite a wide range of acoustical problems. Because they offer a significant amount of both low frequency control in particular and broadband absorption overall, they work well in rooms that don't have corners available for treatment with LENRDs or where you don't need or desire large expanses of absorbent wall or ceiling treatment. (In other words, Sunbursts work well in rooms where you desire control but don't want too dry a sound, or in rooms where you don't have a lot of wall or ceiling space available for treatment.) Sunburst Females are often cut in half and used as "mini-LENRDs" or to frame Studiofoamed areas. They look great and deliver truly linear broadband performance. (They're virtually flat!) Sunbursts contain 4 males and 4 females per box; males are 1´ x 4 ´x 7˝. Instructions and suggestions are included.

NRC: 1.10 *(male yields 1.08 at 125Hz)* ***Size:*** *1´ x 4 ´x 7˝*
Quantity Per Box: *4 males and 4 females (32 lineal feet)* ***Suggested Adhesive Per Box:*** *1 Foamtak or TubeTak*

The Sunburst-360 Eclipse Acoustical Environment

Sunburst-360s are wedge-cut, elongated, stand-mounted absorbers that offer adaptability, portability and affordability. When placed around a vocalist, instrumentalist, voice talent or control room mix position, they form an Eclipse Acoustical Environment, yielding superior broad bandwidth control of reflected sound waves in the nearfield and low frequency waves in the room in general. Sunburst-360s are so linear in their absorption that they exhibit a deviation of only ±1.9 Sabins across the entire test frequency spectrum from 100Hz to 5000Hz. This is the linear absorber the acoustics marketplace has been searching for all these years.
With an Eclipse Acoustical Environment, you can quickly, easily and affordably achieve world-class sound control in any environment. Sunburst-360s are shipped with 2 males and 4 females per box; males are 12˝ x 15.5˝ x 4´. 2 MAX-Stands with 18˝ extensions per box, allowing height of 4´ 6˝ to 9´. Available in charcoal gray, vivid purple and burgundy colors.



BRIEF DEFINITIONS *and* STATISTICS

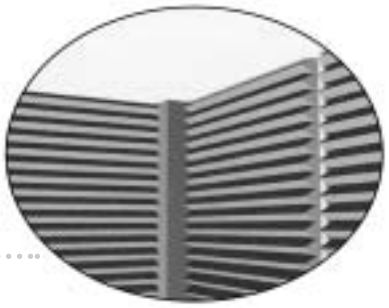
NRC (Noise Reduction Coefficient):
A numerical rating given a material which tells us how much airborne sound the material absorbs. This rating generally applies to soft materials like acoustic foam, fiberglass, fabric, carpeting, etc. but also applies to much lesser degrees to harder materials like brick and drywall. A material's NRC is an average of its absorption at various center frequencies between 125Hz and 4000Hz. The higher the number, the better an absorber the material is.

STC (Sound Transmission Class):
A numerical rating of how effective a material is at blocking the transmission of sound through itself. This rating generally applies to hard materials like rubberized sound barriers, concrete, brick and drywall but also applies to a much lesser degree to softer materials like mineral fiber (although it isn't tested by itself; it's tested as part of a wall system to verify its effectiveness). Virtually every material filters out some of the sound that travels through it, but dense materials are much better at this than are spongy materials

AURALEX ACOUSTICS

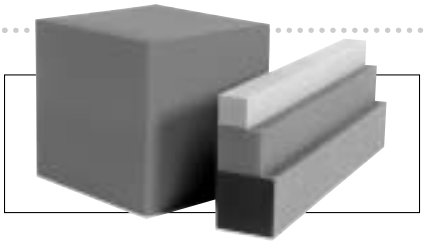
CORNERFILLS

CornerFills



CornerFills are sections of flat-cut Studiofoam that are beneficial in two ways. Aesthetically, they give you smooth, clean edges against which to butt your wall treatments and bass traps for a more professional appearance. Acoustically, they smooth out the excess low frequency energy that congregates in room corners. ***Suggested Adhesive Per Box:*** *1 Foamtak or 1 TubeTak*

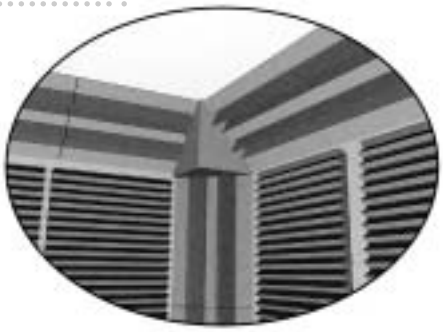
2″ x 2″ x 24″ 36 per box (72 lineal ft.) .....**Call**
3″ x 3″ x 24″ 16 per box (32 lineal ft.) .....**Call**
4″ x 4″ x 24″ 9 per box (18 lineal ft.) .....**Call**
12″ x 12″ x 24″ 1 per box (2 lineal ft.) .....**Call**



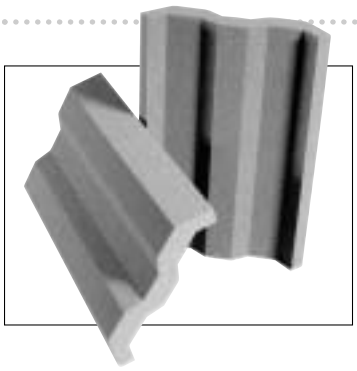
CornerFill Cubes

CornerFill Cubes are a terrific way to finish out trihedral corners where three LENRD Bass Traps come together. Simply install the Cornerfill Cube at the wall/ceiling juncture, then butt your LENRDs up to it for a finished, professional look and tons of additional low frequency absorption. Great for gyms and larger rooms. 12″ x 12″ x 12″. 2 per box (2 lineal ft.).

ATOM-12



This kit consists of four 12″ CornerFill Cubes and 12 LENRDs. Installation's a snap and what you end up with, in each trihedral corner of your room, is a configuration like the one pictured here. Installed in the "usual" way, LENRDs are the most effective small bass traps on the market and have no equal. But when you install them using 12″ CornerFill Cubes, you're really going to have serious low frequency accuracy in your room. All rooms need low frequency smoothing. Trihedral corners (where two walls and the ceiling come together) are the places where physics amplifies bass the most. Given these facts, you can see why the ATOM-12 is a smart way to begin giving your room the bass trapping it needs.



VersaTile

This next-generation broadband absorber combines a visually attractive absorptive panel with specially engineered air cavities that yield extraordinary sonic benefits. The VersaTile can be positioned to bridge a corner, creating a sizable air pocket that delivers great low frequency absorption. It can also be placed on walls and ceilings with either side out, allowing for some sophisticated looks while offering superior absorption characteristics versus thinner, flat acoustic foam panels.
If room boundary space is limited, the VersaTile packs a ton of absorption into a minimal amount of cubic space.

Quantity: *6 or 24 panels* ***Coverage:*** *16 or 64 sq. ft.*

gramma

gramma

GIG AND RECORDING AMP & MONITOR MODULATION ATTENUATOR

The Gramma is used to float an amp or loudspeaker and yields nearly total acoustic isolation, resulting in a purity of tone that has to be heard to be believed.



- Get your own personal amp/speaker riser
- Isolate and decouple your amp and/or monitors from the floor/stage, resulting in much greater separation and clarity
- Makes engineering much easier due to less bleed and resonance
- Dramatically improve audibility on-stage and in the studio
- 23″ x 15″ x 2¾″ LWH
- 1 or 2 GRAMMAs will support most amps, speaker cabinets, subwoofers and monitors

B&H PHOTO - VIDEO - PRO AUDIO

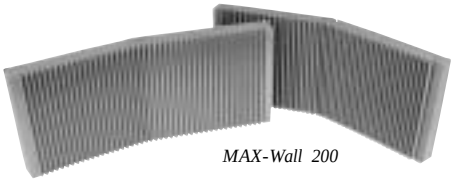
MAX-WALL

MAX-Wall's are everything you've ever wanted from acoustical treatments — affordability, portability, expandability and adaptability. The interlocking MAX-Wall system can grow as your needs change. Add some Sunburst 360s or stand-mounted LENRD bass traps and you've got an incredible amount of acoustical control in a simple, portable package. MAX-Wall's are perfect if you don't want to mount anything to your walls, if your place needs to serve a variety of acoustical purposes or if you're on a budget. They offer great broadband absorption (even at low frequencies) and are incredibly linear, with an overall NRC of 1.05. It offers excellent broadband control of sound waves throughout your room and in the near field, so you can play, record or mix great music or vocal tracks anywhere, anytime.

The MAX-Wall panels are thick enough to yield a pretty high degree of sound containment, so a couple boxes of them can be used to set up a temporary pseudo-iso booth. It won't yield total sound isolation, but will give you enough containment so that you can lay down a clean new track in a hurry and likely not have to monitor on headphones.



MAX-Wall 831



MAX-Wall 200



MAX-Wall 1141VB



MAX-Wall 420



MAX-Wall 633



MAX-Wall 211

MAX-Wall 200
2 - 20″ x 48″ MW-panels

MAX-Wall 211
2 - 20″ x 48″ MW-panels
1 - MAX-Stands
1 - MW-panel with 18″ x 12″ window

MAX-Wall 420
4 - 20″ x 48″ MW-panels
2 - MAX-Stands

MAX-Wall 521
5 - 20″ x 48″ MW-panels
2 - MAX-Stands
1 - MW-panel with 18″ x 12″ window

MAX-Wall 633
6 - 20″ x 48″ MW-panels
3 - MAX-Stands
3 - MW-panels with 18″ x 12″ window

MAX-Wall 831
8 - 20″ x 48″ MW-panels
3 - MAX-Stands
1 - MW-panel with 18″ x 12″ window

MAX-Wall 1141VB
11 - 20″ x 48″ MW-panels
4 - MAX-Stands
1 - MW-panel with 18″ x 12″ window
12 - MW Corner Couplers



MAX-Wall 521

Kits are available in charcoal gray, purple, or burgundy. MAX-Walls can be raised to 8´ by mounting them on MAX-Stands that have an extension. MAX-Clips are included for added support. To match your MAX-Wall, MAXPak Studiofoam features the same wedge-cut design. Corner Couplers are available to join MAX-Walls at 90° angles. (The 1141 VB Kit includes Couplers). The MW-Window Kit can be purchased separately and includes a 48 x 20″ panel with an 18 x 12″ plexiglas-type window.

DESIGNER SERIES

Free shipping on most Auralex Acoustics products

Introducing the Auralex Designer Series

Auralex continues to lead the industry in acoustical performance with innovative products that the market demands. With the introduction of the new DST Designer Series, Auralex now offers DST versions of Studiofoam, LENRD, and Roominators with a more stylish look.

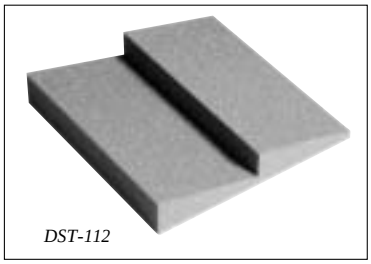
STUDIOFOAM DST

For the design-concious looking for excellent absorption, a sleek, high-tech stylish look and minimal installation seaming, Studiofoam DST is for you! Available in 2 simple shapes, you can easily create a multitude of great patterns that adds a true desgner-touch to your recording, listening or home theater environment at a great value.

DST-112

These 1´x 1´s provide a great design base, and can easily be cut in half to offer exciting combinations.

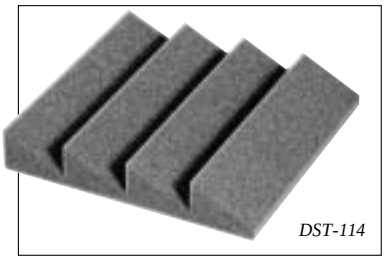
NRC: *0.60* ***Qty:*** *24 or 96 panels*
Coverage: *24 or 96 sq. ft.* ***Color:*** *Charcoal Gray*
Suggested Adhesive Per Box:
1 Foamtak or 3 Tubetak (per 96 sq. ft.)


DST-112

DST-114

The DST-114's are also 1´x 1´s that complement the 112's and other DST Studiofoam products. They can easily be cut in 3˝ shingles, allowing for many options when used singularly or with 112's.

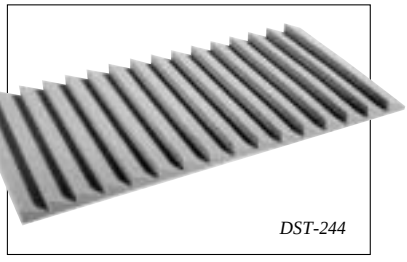
NRC: *0.65* ***Qty:*** *24 or 96 panels*
Coverage: *24 or 96 sq. ft.*
Suggested Adhesive Per Box:
1 Foamtak or 3 Tubetak (per 96 sq. ft.)
Color: *Charcoal Gray, Burgundy, Purple, Forest Green*


DST-114

DST-244

The same cut as the DST-114, but in a 2´ x 4´ x 2˝ Studiofoam panel. The "shingle" contour allows you to easily cut them to create great designs.

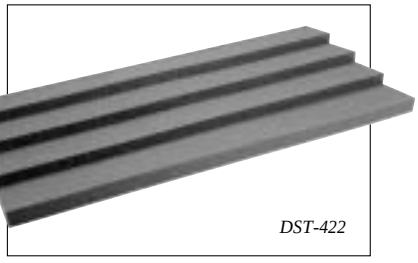
NRC: *0.65* ***Qty:*** *12 2´ x 4´ panels* ***Coverage:*** *96 sq. ft.*
Suggested Adhesive Per Box: *1 FoamTak or 3 TubeTak*
Color: *Charcoal Gray, Burgundy, Purple, Forest Green*


DST-244

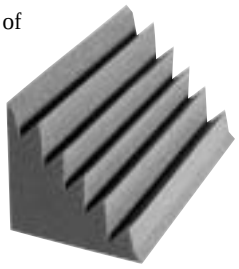
DST-422

The same cut as the DST-112, but running lengthwise in a 4´ x 2´ panel. Create great borders that finish off almost all Auralex products by easily cutting them down to 6˝ x 48˝. If you "don't want it to look like foam", this is the product for you.

NRC: *0.60* ***Qty:*** *6 or 12 - 4´ x 2´ panels*
Coverage: *48 or 96 sq. ft.*
Suggested Adhesive Per Box:
1 FoamTak or 3 TubeTak (per 96 sq. ft.)
Color: *Charcoal Gray, Burgundy, Purple, Forest Green*


DST-422

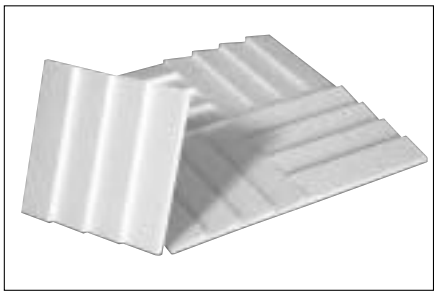
DST LENRD



Enjoy all the benefits of the ever-popular LENRD bass traps in a design that will make any DST install look stellar. When you need some low-end absorption, the DST LENRD is a great design alternative .....**Call**

NRC: *1.20* ***Qty. Per Box:*** *8 (16 linear feet);*
Colors:
charcoal gray, burgundy, purple, forest green
Suggested Adhesive Per Box:
1 Foamtak or 2 Tubetak

DST-R



The newest member of the diffusor family is the Diffusive Sound Tile Reflector, or DST-R. Intended as the perfect compliment to the DST-112 and DST-114 Studiofoam Panels, they can be aligned in a number of ways to create a designer look. Only 1˝ thick, the DST-R is 1´ x 1´. It is the same shape as a DST-114 panel, making it a great option when installation depth is limited, such as a low ceiling. Made from high quality EPS, the DST-R is lightweight, can be installed and painted easily, and is a very low cost diffusive alternative.

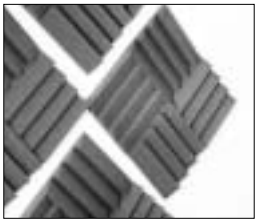
Qty: *36 per box* ***Size:*** *1´ x 1´ x 1˝*
Coverage: *36 sq. ft.*
Suggested Adhesives Per Box: *2 Tubetak*

AURALEX ACOUSTICS

DESIGNER SERIES

ROOMINATORS DST

Roominator-DST Kits span a wide variety of applications, addressing the acoustic and aesthetic needs of the small project studio as well as the most demanding professional mix/mastering environments. Whether it's your first studio, a home theater, or a Pro Tools Suite, the Roominators-DST Kits are designed to dramatically improve your acoustics, and give you a great designer look.







D36 Roominator DST Kit

Perfect for spot treatment. The D36 offers you a starting point to reduce unwanted slap and flutter echo in your room.

The kit includes:

(18) DST-112 panels
(18) DST-114 panels in your choice of their 4 colors
(2) Tubetak Pro Adhesives

D36 .....**Call**

D108L Roominator DST Kit

All the components you need to attack the acoustics of a smaller Project Studio. Excellent for smaller home theaters. Two D108L kits may be appropriate for medium-sized studios and home theaters, as well as live rooms and rehearsal spaces.

The kit includes:

(54) DST-112 panels
(54) DST-114 panels in your choice of their 4 colors
(8) DST-LENRD Bass Traps (charcoal gray)
(6) Tubetak Pro Adhesives

D180L .....**Call**



D240xi

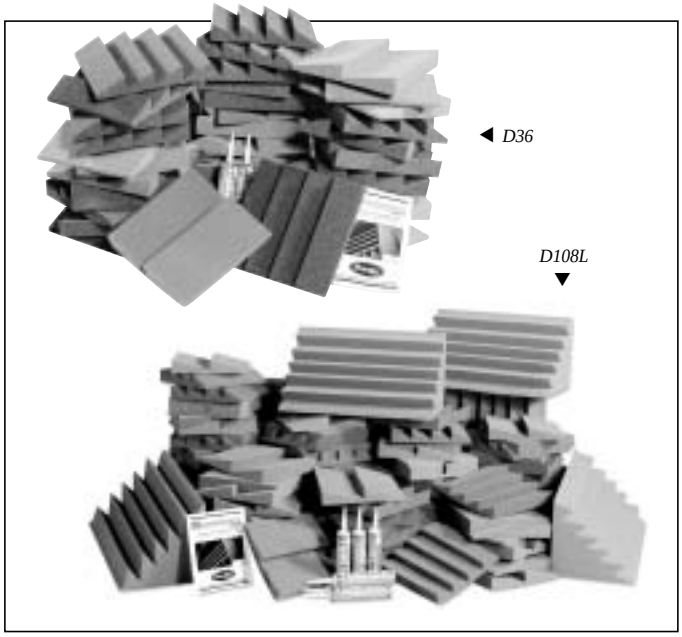
D240xi Roominator DST Kit

The Big-Daddy of Roominator kits! It not only gives you plenty of product for superior acoustic performance, it provides all the components necassary for a fabuluous look. It also includes a Gamma Isolation Riser and 10 MoPad Monitor Isolation Pads to properly isolate a sub-woofer and up to 5 monitors.

The kit includes:

(48) DST-112 panels
(12) DST-422 panels (96 sq. ft. in charcoal gray)
(96) DST-114 panels in your choice of their 4 colors
(16) DST-LENRD Bass Traps (charcoal gray)
(12) T'Fusor 3D Sound Diffusors
(6) Tubetak Pro Adhesives

D240xi .....**Call**



◀ *D36*

D108L ▼

CONSTRUCTION MATERIALS

SheetBlok Sound Barrier



SheetBlok is a dense, limp-mass vinyl material that is about 6dB more effective than solid lead at stopping sound transmission. It acts as a thin, dense sound barrier layer in walls, ceilings or floors and is most effective when used as one component of a multi-layered construction scheme. SheetBlok helps in decoupling (i.e. floating) floors and walls and, while not considered a finish product, is paintable with latex paint and can be the last layer in some cases. It can be used to wrap HVAC ducting, as a vent noise blocker, as a pipe noise insulator or under carpet/carpet pad. It holds up against harsh environments and cuts easily with a utility knife or scissors. SheetBlok can be doubled up, increasing its effectiveness from STC 27 to about 35.

SheetBlok 4′ x 10′ pieces (40 sq. ft.).....**Call**
SheetBlok 4′ x 30′ rolls (120 sq. ft.).....**Call**

Color: *Black or Clear (subject to availability)* ***STC:*** *27*
Thickness: *1/8″* ***Weight:*** *±1#/sq. ft.* ***Service Temperature:*** *-40° to 180° F*
Tensile Strength: *400 psi* ***Tear Strength:*** *70#/inch*
Flammability: *Rated UL94 S.E. “0”; Passes test MVSS-302*
Notes: (1) STC stands for Sound Transmission Class, a rating of how effective a product is at being a sound barrier.

SheetBlok-Plus
Comes with pressure-sensitive adhesive already attached. Available in 4′x30′ rolls only.

U-Boats
Floor Beam Float Channels



Made of a specially-formulated rubber compound, these U-shaped channels are used to support framing members and float (isolate and decouple) them from the surrounding structure. With the help of U-Boats, a floated room features greatly improved transmission loss (isolation) and low frequency definition. It’s easy to figure how many U-Boats to get: simply multiply the total square footage of the area you want to float by two-thirds; the resulting number gives you the approximate amount of U-Boats you’ll need.



U-Boats on 16″ centers will alleviate 50-99% of vibrations above 40Hz. U-Boats on 32″ centers will alleviate 50-99% of vibrations from 20Hz up.

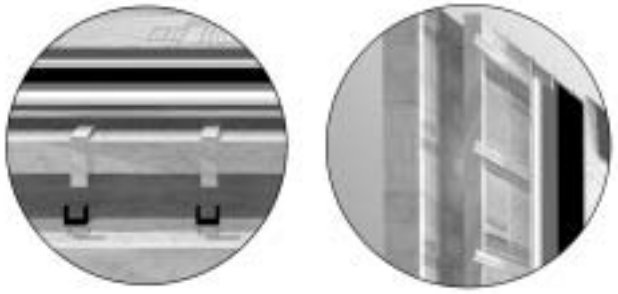
Size: *2⅛″ x 1½″ x 2″ (WHL)* ***Color:*** *Black*
Thickness: *5/16″ and 1/2″ base*
Quantity: *Available in lots of 50 or more*

Mineral Fiber Insulation



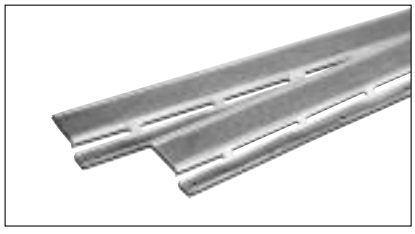
The best way to tighten up a room’s sound and make it more soundproof, is to design it right, build it tight and make the partitions as dense as possible. That’s where specialized Mineral Fiber comes in. It is very dense and is specifically designed for acoustical usage. It’s available in 2″ and 4″ thicknesses, and is designed to offer better bass trapping, overall absorption, thermal characteristics, moisture absorption and sound transmission loss (isolation) than any other product. For those of you in need of absorption, but for whom acoustic foam may not work for whatever reason (flame retardancy, looks, style, etc.), Mineral Fiber covered with specialized acoustical cloth is a viable alternative.

2″ Mineral Fiber Insulation .....**Call**
4″ Mineral Fiber Insulation .....**Call**
Quantity Per Box: *2″ - 6 (48 sq. ft.); 4″ - 3 (24 sq. ft.)*
Flame Retardancy: *Class A*
Advantages vs. Standard Fiberglass: *better sound isolation, better acoustic absorption, lower moisture absorption, better flame spread and smoke density, 1200° burn point vs. 650°*



U-Boats, SheetBlok and Mineral Fiber help make the ideal floating floor

RC8 Resilient Channel



RC8 is a specially-formed, sturdy metal device that, when used to hang drywall (instead of just attaching the drywall to the studs or joists), greatly improves the sound transmission characteristics of the wall or ceiling system. RC8 should be installed horizontally at the bottom and top of your wall, then every 2′ or less in between. Up to 2 layers of 5/8″ drywall and a layer of SheetBlok may be hung on RC8, even on a ceiling.
RC8 is available in handy 8′ lengths that are ground shippable. There are twenty-four 8′ pieces per tube.

MS SERIES

Confidence Monitors



MS2ADV-1 ▲



▼ MS4AD-1

When your livelihood depends on the integrity of an electronic signal, an MS Series Confidence Monitor can give you blissful peace of mind. They provide assurance that analog audio, digital audio, or video signals are present and meet established quality standards before being broadcast, recorded, or otherwise distributed. The various models differ primarily in the types and number of signals they can monitor, though all are housed in compact, single space chassis that occupy minimal rack real estate. Monitoring is accomplished audibly (through speakers or headphones) and visually through meters and video monitors.

FEATURES

They All Feature

- Each model features a pair of built-in highly compact yet full range shielded stereo speakers, along with a Speaker Mute switch.
- An amply powered TRS output is provided for headphone listening. Speakers and headphone amps share a volume slide control, and non-video models have Balance sliders.
- Phase indicator displays the phase relationship of the left and right channels of the monitored source. Indicator glows green to amber under normal conditions, and red (conditions of total phase reversal), helping determine mono-compatibility.
- Each input is routed to a bright, 14-segment LED meter, which may be switched for VU or PPM ballistics and +4 or +8 dBu reference levels with front panel switches.
- Meter brightness may be adjusted with a dimmer control, to suit user taste and variations in ambient lighting.
- All front panel control and indicator labels are rear printed on a heavy duty, wear-free mylar overlay.
- All analog inputs and outputs are high common-mode rejection XLR balanced, and each is provided with a rear panel signal activity indicator for easy identification of system problems. Outputs also have ±6 dB Gain Trim controls.
- They are powered with an included UL-approved external power supply via a DIN 5 connector. For maximum reliability, they feature a clean, modular design with minimal cabling. And all include a Three Year Limited Warranty

Analog/Digital Audio Models

- Models with digital monitoring capability are equipped with an A/D Select button and both front and rear bi-color digital signal activity indicator(s). Digital input(s) accommodate stereo digital signals sampled at up to 96 kHz, in AES/EBU format, via rear panel BNC inputs.
- A/D converters provide up to 8X oversampling, with a dynamic range of 95 dB.

Video Models

- Video models include a 1.8″ LCD video monitor which automatically senses and displays either NTSC or PAL signals.
- Adjustments for color, brightness, contrast, and tint are provided on the front panel.
- Because the speakers in the MS Series are magnetically shielded, no special consideration is required for installation in critical locations near video monitors.

Six Models to fit any application

MS-2A-1: 2-channels analog audio inputs.....**739.95**

MS-2AD-1: 2-channels analog and digital audio inputs.....**909.95**

MS-4A-1: 4-channels analog audio inputs.....**839.95**

MS-4AD-1: 4-channels analog and digital audio inputs **1050.00**

MS-2AV-1: 2-channels analog audio inputs plus video inputs with LCD monitor **1294.95**

MS-2ADV-1: 2-channels analog and digital audio plus video inputs with LCD monitor ...**1399.95**



FURMAN

POWER CONDITIONERS

PL-8 Power Conditioner and Light Module



Featuring triple-mode varistor spike and surge protection, the PL-8 is the perfect accessory to any equipment rack, providing the basic power protection essential to every audio or video system. Install a PL-8 in the top slot of your rack, and the eight switched AC outlets on the rear panel will power up all your equipment while the two slide-out light fixtures provide discreet illumination of the controls for use on stages, in studios, or in any dark place. A dimmer control lets you adjust the lamp brightness to just the right level.

- Protects delicate electronic equipment by combining a high voltage surge and transient suppressor with an RFI/EMI interference filter. The fast-acting suppression circuit responds in nanoseconds, clamping spikes and surges to safe levels. The filter works to prevent electronic noise from fluorescent lights, electric motors and radio transmitters from leaking from the AC line into sensitive analog or digital circuits.
- The PL-8 can absorb a surge current of up to 6500 amperes for 10 ms without damage.
- Two slide-out, swiveling light fixtures for rack illumination.
- The lamps come supplied with 7 watt nightlight bulbs. Bulbs are easily replaced without removing the unit from the rack. (Replacements are readily available and you can even use colored Christmas tree bulbs).
- Master switch controls the eight rear outlets and glows red when the power is on.
- A separate switch for the lamps allows you to check your equipment settings before applying power to it.
- Dimmer control for lamp brightness
- Rated at 15 amps, the front-panel circuit breaker can be easily and quickly reset should the PL-8 be overloaded.

PL-PLUS Ultra-Level Protection with Voltmeter and Lights



An enhanced version of the PL-8, the PL-PLUS offers improved surge protection, RF noise filtering and an AC voltmeter. The 20 LED bar-graph voltmeter reads from 90 to 128 volts in 2-volt steps. The normal range voltages are indicated in green, with moderately and extremely high or low voltages in yellow and red respectively. The voltmeter's accuracy is ±2 volts. It can easily be recalibrated if necessary. The sophisticated RFI filter network offers greatly improved capacity for stopping line noise. This filter incorporates a toroidal inductor and three high-voltage capacitors to achieve a common-mode noise attenuation of over 50 dB at 10 MHz. Finally, the PL-PLUS uses a longer power cable than the PL-8 (10´ heavy-duty 14 gauge power cable).

PM-8 Power Conditioner with Current and Voltage Meters



The model to choose when rack lights aren't needed. The PM-8 has all the power onditioning features of the PL-PLUS, plus it has a ninth unswitched outlet whose front panel location makes it very convenient for equipment in temporary use. It has both a voltmeter and an ammeter. The ammeter reads true RMS current, so it will indicate reliably and meaningfully for all loads, including inductive and capacitive loads such as power amplifiers. (Ammeters that respond only to average AC current may give readings that differ widely from RMS-reading meters.) The PM-8's voltmeter reads from 90 to 135 volts in 5 volt steps. The ammeter reads from 0 to 20 amps, in 2 amp steps.

POWER CONDITIONERS

Metered Six Series



The Metered Six are an extension of Furman's well-respected 15- and 20-Amp power conditioners. The "Metered Six" differ from their siblings by the inclusion of additional filtering for digital equipment, Digital AC voltmeter, and all (except the PL-PLUSD and PL-PROD) feature a Digital AC current meter. The current meter is a true R.M.S. variety, and both meters sport laboratory accuracy. The true R.M.S. (root mean squared) current meter, allows installers to read the total system current draw without the false readings typically experienced due to distorted AC voltages. Additionally, digital AC outlets provide increased filtering and inhibit contamination of neighboring components.

FEATURES

- All units are single 19˝ rack space, and feature an attractive black anodized front panel, and rust proof chromate steel chassis for positive ground contact. They can withstand a one-time power surge of 6500 amps (11,000 amps for Pro models).
- Common and transverse mode noise is greatly diminished through Furman's three stage RFI/EMI filtering. This prevents fluorescent lights, radio transmitters, and similar sources of "electrical pollution" from leaking into sensitive audio/video components.
- Each unit features (4) digital and (4) analog AC outlets, which are activated by either a red backlit power switch, or a blue handled magnetic circuit breaker (Pro Series).
- An independent power switch activates units featuring dual retractable lights. This allows you to carefully inspect an entire equipment rack's settings prior to turn-on.
- All power switches have been chosen to handle high in-rush peaks, typical in many high performance power amplifiers.
- Each features a 10´ heavy-duty power cord and full safety compliance for North America through C-UL listing.
- Metered Six units featuring dual retractable lights come supplied with one 7-watt nightlight bulb per light tube. Replacements are available in 4 to 7-watt sizes at most supermarkets and hardware stores.

PRO Models (Only)

- 20 Amp (2400 Watt) rating handles even the largest amps
- The three PRO units add an "extreme voltage shut-down" circuit. Useful in installations where faulty wiring and accidental 220v AC feeds are common, this circuit senses AC voltages that are either so high or low that operation would be impossible (under 80v AC or over 140v AC). The circuit then immediately shuts down all output power well before damage can occur.
- Additionally, "Protection OK" and "Ground OK" monitor the integrity of the wiring and protection devices.

Metered Six Models

PL-PLUSD: 15 Amp, dual retractable rack lights, digital volt meter **174.95**

PL-PLUSDM: 15 Amp, dual retractable rack lights, digital volt/current meter **189.95**

PM-8DM: 15 Amp, digital volt/current meter .. **174.95**

PL-PROD: 20 Amp, dual retractable rack lights, digital volt meter **299.95**

PL-PRODM: 20 Amp, dual retractable rack lights, digital volt/current meter **308.95**

PM-PRODM: 20 Amp, digital volt/current meter **289.95**

PL-PRO and PM-PRO



Non-Digital 20-Amp Power Conditioners

The predecessors to the Metered Six series, the industry-standard PL-PRO and PM-PRO are almost identical to their respective digital counterparts (PL-PROD and PM-PRODM), except they use LED readouts instead of digital readouts, and they don't include the special outlets with filtering for digital equipment. Like the PL-PROD and PM-PRODM, they feature 20 AMP operation, Extreme Voltage Shutdown, critical functions are monitored by a "smart" microprocessor, and LED status indicators report on problems.

FURMAN

POWER CONDITIONERS/SEQUENCERS

PS-8/PS-8R Ultra-Level Protection with Sequencing



The PS-8 Power Sequencer is needed whenever various kinds of equipment must be powered up or down in groups, rather than all simultaneously. In audio systems, sequenced powering is often necessary to allow turn-on transients from low level amplifiers and processors to settle down before any power amps are turned on, because simultaneous powering would result in a loud, annoying, and potentially destructive "pop" reaching the speakers. And in any large system whose components present an inductive load to the AC line (including electric motors, power supplies, and power amplifiers of all kinds), sequenced powering can avoid excessive inrush currents that cause circuit breakers to trip even though the steady-state currents are not excessive. The PS-8R is identical to the PS-8 except it can be remotely switched on/off from a distance.

- Using them is a simple and inexpensive way to apply and remove power in a controlled, repeatable, foolproof 3-step sequence. Ideal when large installations must be switched on or off by inexperienced personnel.
- There are three outlet pairs labeled Delay 1, Delay 2, and Delay 3, that receive power almost immediately, 5 and 10 seconds respectively after the power switch is turned on. When power is turned off, the sequence is reversed (Delay 3 loses power almost immediately, Delay 2 1 after 5 and 10 seconds. Intervals are preset but can be changed via internal trimpot adjustment.
- In addition to the delayed outlets, a single front panel outlet and a rear panel pair are unswitched.
- They offer all basic power conditioning features including 15 Amp operation, varistor spike and surge protection across all three modes, as well as a sophisticated EMI/RFI filter for blocking radio frequency line noise. Clean, filtered power is provided at all 9 outlets, even the unswitched ones.
- They can detects faults in AC wiring. Green and and red neon indicators light in a specific pattern for normal wiring and in different patterns for various faults.
- If a fault is present, a chart next to the indicators will identify its exact nature.

PS-8R Step-up Feature

The PS-8R is identical to the PS-8 with the addition of a barrier terminal strip on the rear panel that allows a switch (either a momentary or maintained-contact type) to be connected to turn the unit on and off at a distance. If a 3-conductor cable is used, an LED may also be installed at the remote end to indicate that the power is on. The Delay 1 LED on the PS-8R blinks to indicate the remote switch is off.

PS-8.....**212.50** **PS-8R**.....**244.95**

PS-PRO Pro-Level Protection with Sequencing and Remote



The PS-PRO Power Conditioner/Sequencer is capable of powering up a rack full of equipment in a 3-step delayed sequence. The sequence is reversed for power-down. The sequence can be initiated with either momentary or maintained switches, locally or remotely. A duplex outlet is provided for each delay step. A front panel screwdriver adjustment sets the delay time. A locking switch with a removable key is provided for maximum security. The PS-PRO also contains special circuitry to detect additional wiring faults such as reversed hot and neutral, missing ground or neutral, etc. One or more PS-PROs may be installed in remote locations and operated via low-voltage wiring.

PS-PRO Step-up Features:

- 20 Amp (2400 Watt) rating
- Multiple levels of protection components can safely absorb large spikes and provide highly effective RF filtering — the most comprehensive protection available.
- Extreme Voltage Shutdown guards against destructive wiring faults
- Critical functions are monitored by a "smart" microprocessor; LED status indicators report on problems
- Locking switch with a removable key is provided for maximum security.
- Heavy duty 10 foot, 12 gauge line cord

PS-PRO.....**462.95**

AR-1215/1220/1230/PRO

Voltage Regulators

Furman's AR series are voltage regulators designed to protect audio, video, computer and other electronic equipment from problems caused by AC line voltage irregularities—sags, brownouts, or overvoltages that can cause sensitive digital equipment to malfunction, or, in extreme cases, to sustain damage.

They provide a steady, stable 120v AC output. The 15 amp AR-1215, 20-amp AR-1220 and 30-amp AR-1230 accept any voltage from 97v to 141v and transform it to a constant 120V, ±5V. (Voltages beyond that range may also be converted to usable levels, depending on how far out of range they are).

The AR Series has been designed specifically with the unique needs of audio and video in mind. Its technology differs from that of computer-oriented voltage regulators in many important ways. For example they don't use a ferro-resonant transformer, which would be heavy and bulky, radiate a large magnetic field, and be too frequency-sensitive to be usable with generators. Instead, they use a design based on an eight-tap toroidal autoformer. The toroidal design assures minimal leakage of stray magnetic fields.

FEATURES

- Their circuitry monitors the incoming line voltage with each cycle, comparing it to an extremely precise voltage reference, accurate to ±0.15%. If a voltage fluctuation requires that a different tap be selected, the new tap is electronically switched exactly at the zero-crossing, to avoid distorting the AC waveform. (Some commercial voltage regulators using multiple-tapped transformers switch taps at uncontrolled times, thereby creating voltage spikes,often creating clicks that can leak into the audio.) Since the AR Series design is not sensitive to small errors in line frequency, they are ideal for using with e with generators.
- They offer power conditioning that is truly in class by itself, thanks to the quantity, quality and configuration of the overvoltage suppression devices used. These include MOV's and high voltage inductors and capacitors. This unique combination can safely divert large spikes as well as filter audible high frequency noise.
- For added safety and protection, the Extreme Voltage Shutdown function senses dangerously high or low voltages and shuts down the output before any damage is done. The output remains off until the overvoltage or undervoltage is removed, with an LED indicating the shutdown condition. This invaluable feature protects your equipment from accidental connection to incorrect line voltages (for example, 220v when 120v is expected).

AR-1215



The AR-1215 has eight 15 Amp outlets on the rear panel, and one on the front panel. All outlets are regulated, spike-suppressed, and filtered against RFI with a 3-pole filter. There are no controls except the circuit breaker/on-off switch. A bar-graph meter comprised of 10 LED's indicates input voltage, while another LED indicates "In Regulation" status (i.e., that the output voltage is within ±5V of 120V.) Housed in a compact, single-space rack-mount chassis, 1.75″ high and 8″ deep and weighs only 12 lbs.....**459.95**

AR-1220 Step-up Features



The AR-1220 has twelve 20 Amp outlets on the rear panel, and two 15 Amps on the front panel. Housed in a two-space rack-mount chassis, 3.5″ high and 17″ deep, weighs 35 lbs.....**929.95**

AR-1230 and AR-PRO Step-up Features



The AR-1230 has one 30A, NEMA L14-30P twistlock inlet, one 30A NEMA L14-30R twistlock outlet, and six Edison outlets on the rear panel, plus two Edison outlets on the front panel. Housed in a compact, two-space rackmount chassis, 3.5″ high and 17″ deep, 39 lbs. Similarly, the 30 Amp AR-PRO delivers 120v AC ±4% anywhere in the world within ranges of 88 to 134 and 170 to 264v AC, 50/60 Hz. It has an input voltmeter bar-graph and output ammeter bar-graph with 21 LEDs. Monitor indicates low, normal, or high output voltage. 14 regulated, 20A conditioned outlets (2 front, 12 rear). It has a provision for remote turn-on/off; multiple units may be turned on simultaneously or in a delayed sequence. Weighs 50 lbs.

AR-1230 .....**1399.95** **AR-PRO**.....**1799.95**

FURMAN

IT-1220 / IT-1230

Balanced Power



Designed for the most critical, ultra-low-noise installations, the IT-1220 and IT-1230 supply 20 or 30 amps of balanced AC power to recording studios, video or film production facilities, broadcast stations, etc. They drastically reduce hum and buzz caused both by ground currents from power supply filtering and by radiation from supply cables into sensitive lines, particularly low level and unbalanced sources. (They have also been reported to noticeably improve dynamic range and sonic clarity. At the heart of these units is a specially wound and shielded toroidal isolation transformer with a center-tapped secondary. These precision transformers, in conjunction with wideband EMI/RFI filters and output spike protection, make these units excellent power conditioning solutions for any situation.

- The IT 1220 rear panels provide 12 outlets, the front panel provides two. The IT-1230 rear provides ten 20A outlets, one 30A twistlock outlet and one 30-amp twistlock inlet. The front provides two 15A outlets. All outlets are balanced and Ground Fault Circuit Interrupter protected.
- An IT-1220 can be installed in minutes without an electrician, the IT-1230 however, may require an electrician.
- Exclusive "Soft Start" circuit prevents the large inrush surge currents and spikes that would otherwise occur at turn-on and turn-off with such large transformers.
- Accurate, self-checking AC voltmeter measures not only normal voltages, but also flashes eye-catching special pattern alerts for off-scale but not extreme conditions (80-90 or 130-140 volts).
- They provide clean and completely safe power – there is no need to "lift grounds" or compromise the integrity of safety ground wires to achieve hum reduction.
- Extreme Voltage Shutdown circuit shuts-down the unit if exposed to dangerous voltages (like accidental connection to 220v)
- The IT-1220 and IT-1230 occupy three rack spaces (5.25″ high).

IT-1220 .....**1099.95** **IT-1230**.....**1299.95**

IT-REFERENCE

Discrete Symmetrical AC Power Source



The ultimate expression in AC power filtering, the IT-Reference provides transparent, ultra-low noise AC power for the most discriminating audio/video professional.

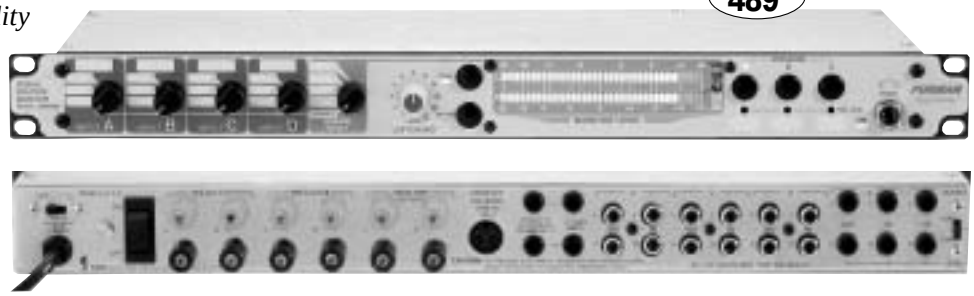
Same features as the IT-1220 plus:

- Discrete 5A windings for each set of outlets. This is critical, as modern video and digital components can couple performance-degrading noise into each other, even if the filtering from the source AC is the best money can buy.
- Exclusive power correction circuitry gives all power amplifiers the ability to reproduce dynamics with extreme power, speed, and clarity that will produce startling three-dimensional images.
- 20 Amp peak power, 12 Amp continous power
- 4 high current ultra low impedance outlets for power amplifiers and powered monitors
- 4 pairs of individually isolated, electrostatically shielded, positive contact AC outlets designed for analog and digital audio and video components

SRM-80A

Signal Router/Monitor

$489.95

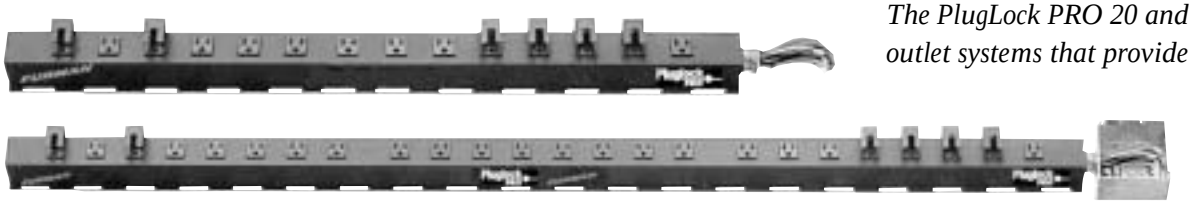


The SRM-80A provides all of the flexibility and capabilities of large format console "Monitor Section" in an easy to use 1U device. Designed to simplify mixdown and dubbing from the single stereo output from DAWs, associated computer sound cards and digital I/Os, the SRM-80A provides signal routing and monitoring for four stereo mixdown devices such as DAT machines, cassette decks, CD burners, MiniDisc recorders, as well as three sets of studio monitors and/or headphone amps. The SRM-80A also simplifies mixdown and copying between different media types. Multiple copies can all be made simultaneously between DATs, CD Burners and other devices, through use of the SRM-80A's "Source Select" switches.

- Use one power amp to drive two speaker pairs, also accommodates one pair of self-powered speakers.
- Monitor using three different speaker pairs, at the same loudness, using exclusive "set-once" level trims.
- Speaker button "A" switches line-level for driving self-powered monitors; B and C use a high power relay to redirect your power amp output to either of the two pairs.
- Gain adjust for each speaker pair allows for equal loudness when switching between them.
- High power headphone amp for private monitoring and mix checking.
- 40-segment high resolution metering for L/R monitor outputs
- VU meter display with peak hold function
- Peak or Average response switch
- Line level can be set to -10 or +4 dBu
- Ground lift switch
- DIM button reduces level by 15 dB during interruptions —an easy way to lower the listening level temporarily without having to adjust the listening level control
- Mono Sum let you check for radio compatibility, stereo imaging, and phase problems
- Optional remote allows five push button functions (MONO, DIM, and SPEAKER SELECT) to be at the listening location.

PGP-20/PGP-60

PlugLock PRO 20 and PlugLock PRO 60 Locking Outlet Systems



The PlugLock PRO 20 and PRO60 are locking outlet systems that provide unprecedented ease, versatility and reliability for professional rack applications. Both feature rugged, all-metal construction, and include a set of Furman's metal PlugLock clamps that securely lock any plug or wall wart into place for reliable operation. The PGP-60 comes with 12 clamps, the PGP-20 comes with 8 clamps.

Rated at 40 amps, the PGP-20 provides 14 Edison outlets (10 on one circuit, four on the other) in a chassis that is only 30″ tall and can be mounted vertically in smaller racks and other locations.

The PGP-60 provides 24 Edison outlets in its 60″ tall chassis, and is designed primarily for vertical mounting in equipment racks. The PGP-60 is rated at 60 amps, divided into three 20 amp circuits to provide flexibility for multiple and varying applications. The included 3/4″ nipple mount allows an electrical junction box to be mounted either at the top or bottom of the PGP-60. Power sequencing relays are also available. A local circuit breaker can be easily added to each of the 20 amp circuits if necessary.

PGP-20.....**88.50** **PGP-60**.....**177.50**

MIDDLE ATLANTIC

EDIT CENTER SERIES

Multimedia Studio Furniture

An elegant, ergonomic workspace for the professional



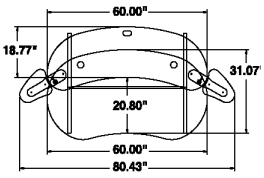
ES+S12D shown with S12 single bay outboard rack without doors

- Suitable for any production or editing environment including project studios, post production editing suites, learning labs, and broadcast facilities
- Factory-assembled systems are available in either a graphite or cherrywood laminate finish
- Essential features that professionals require are built-in — no need to purchase expensive options
- Desk systems are available in two sizes: 84″ and 60″ wide
- Overbridges are capable of accommodating multiple 21″ computer monitors as well as large program monitors
- Two pivoting speaker platforms (standard on all Edit Center overbridges) allow you to place your audio monitors at the optimum listening angle
- Two overbridge styles are available to optimize your system: one features two 4-space rack bays angled for direct viewing, while the other features an open span underneath

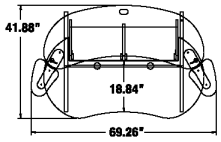
Edit Center with Side Racks — are available with either single or dual 12 space racks, with or without doors. Racks with factory-installed gasketed doors (plexi front and solid rear) along with internal acoustical absorptive material reduce noise escaping from the enclosure by up to 22db. The rear doors features an ultra low-noise cooling fan, a removable filter kit for air intake, and a brush grommet at the door bottom that allows cable exit while maintaining a seal for noise and air flow.

DIMENSIONS

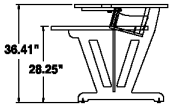
desks shown with monitor platforms at minimum and maximum extensions



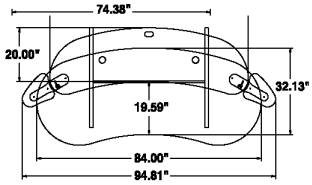
ES top view



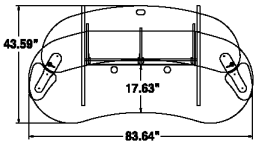
ESUR top view



side view (all desks)



EL top view



ELUR top view

DESKS

ESC: 60″cherry desk with overbridge.....**1460.00**

ESURC: 60″ cherry desk with overbridge with (2) 4 space rackbays ...**1570.00**

ELC: 84″ cherry desk with overbridge.....**1540.00**

ELURC: 84″ cherry desk with overbridge with (2) 4 space rackbays.....**1640.00**

EDSC: 60″ cherry producer/client desk, no overbridge **980.00**

EDLC: 84″ cherry producer/client desk, no overbridge **1180.00**

Also available in graphite color.

DESKS WITH SIDE BAY RACKS

ES+S12D: 60″ desk with overbridge and 1 single bay outboard rack (with front/rear doors, cooling kit).....**2450.00**

ESUR+S12D: 60″ desk with overbridge, two 4 space rackbays and 1 single bay outboard rack **2600.00**

ELUR+S12D: 84″ desk with overbridge, two 4 space rackbays and 1 single bay outboard rack **2700.00**

ELUR+D12D: 84″ desk with overbridge, two 4 space rackbays and 1 dual bay outboard rack.....**3070.00**



ESC



ELUR



ELUR+D12D

B&H PHOTO - VIDEO - PRO AUDIO

ONE HOUR FREE PARKING

AT 349 W. 34th STREET (with purchase of $100 or more)

EDIT CENTER SERIES

Side Bay Racks with Doors, Noise Control and Cooling System

Standard on all Edit System Series Workstations

- Available in single or dual bay configuration, with either a graphite or cherry wood laminate top, complete with casters
- Contoured reversible top matches the desktop, allowing the racks to be positioned at a comfortable angle to the user
- Racks can join the desk flush, or slide under the desk top
- Front rackrail is set-back 2″ for equipment clearance; rear rackrail can be positioned at any of four pre-drilled depths
- An optional laminate shelf can be mounted in place of rackrail
- Gasketed plexiglass front doors and gasketed laminated rear doors control noise and open 95° on concealed hinges
- Rear door features a built-in ultra low-noise fan, a removable filter kit, and a brush grommet at the door bottom that allows cable exit while maintaining a seal for noise and air flow
- Single racks come standard with front and rear door; dual racks include doors on one bay only. Racks can be ordered without doors
- 11-guage, 10-32 threaded front and rear rackrail are set back 2″ for equipment clearance
- All Edit Center racks ship fully assembled

D12D: 12 space dual bay outboard rack, weighs 262 lbs.....**1570.00**
S12D: 12 space single bay outboard rack, weighs 152 lbs **1150.00**

SQUARE-TOP ISOLATION RACKS

Available in 12 and 24 space, these racks are available with a graphite finish top to match graphite Edit Center desks.

S12SDG: 12 space single bay rack; square graphite top, 150 lbs.....**1120.00**
S24DG: 24 space single bay rack; square graphite top, 206 lbs.....**1320.00**

SIDE BAY RACKS WITHOUT DOORS, NOISE CONTROL OR COOLING SYSTEM

Available in graphite or cherry wood laminate top, complete with casters.

S12: Single bay outboard rack (12 rackspaces), 134 lbs.....**695.00**
D12: Dual bay outboard rack (12 rackspaces), 242 lbs **1240.00**

DESKTOP RACK

Provides an extra 8 rackspaces of desktop equipment mounting. Includes graphite top and factory installed 11-guage, 10-32 threaded front rackrail.

DT8G: Table top rack (8 rackspace); weighs 41.3 lbs **250.00**





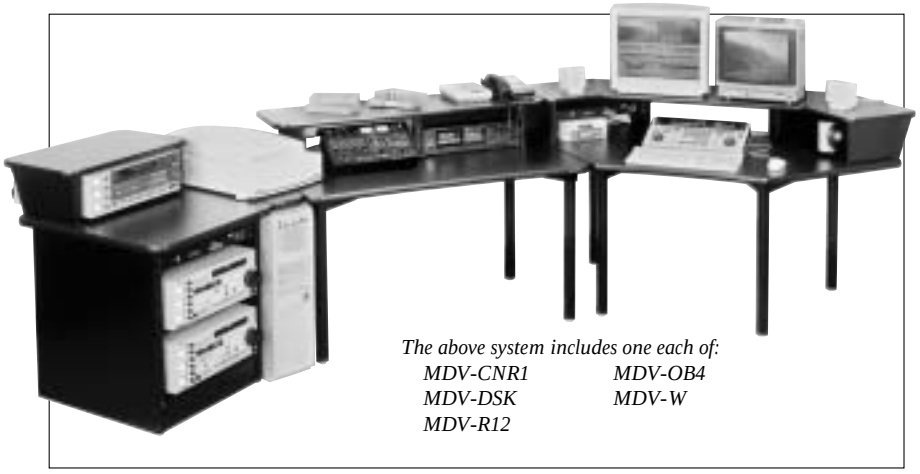
Producer/Client Desks

Introducing an elegant producer/client desk that perfectly matches the existing Edit Center desks and complements any studio environment. Both the 60″ EDS and 84″ EDL models feature the new X-Brace leg system, which provides extreme stability and allows these desks to be used independently with full-circumference seating capability. EDS and EDL desks are available in an attractive marbled graphite or cherry, and can be used with Edit Center side-bay racks.

MULTIDESK VIDEO

Modular Studio Furniture System

- Well suited for project or professional studios
- Modular design allows components to be used together in any combination — you configure a complete system that accommodates the way you work based on your specific requirements
- There are four desk models to choose from, each finished in high-pressure graphite laminate with sturdy steel tube legs and overbridges that can accommodate audio or video monitors
- Optional accessories include a 12RU side rack with matching optional overbridge, as well as a connecting wedge for angling a rack

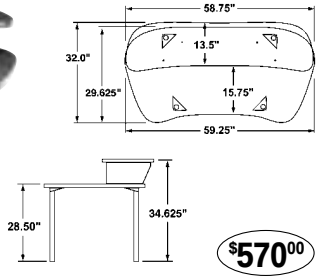


The above system includes one each of:

MDV-CNR1 *MDV-OB4*
MDV-DSK *MDV-W*
MDV-R12

MDV-DL Desk

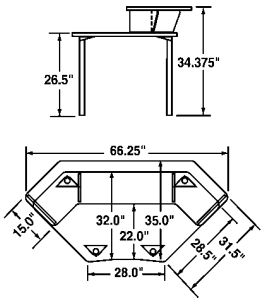




$570.00

- 59″ wide footprint provides lots of workspace
- Overbridge accomodates multiple monitors and is set 6″ above the desktop – the correct height for ergonomic viewing
- Standard 28.5″ desktop height
- Can be used with either the Edit Center line of isolation racks or an MDV-R12

MDV-CNR1 Corner Desk



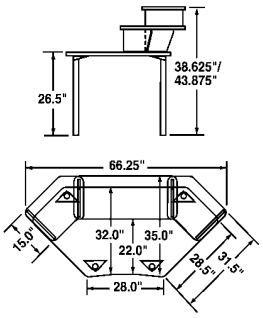


$570.00

- A one-piece overbridge is user-configurable three ways: one 4 space rackbay in center, two 4 space rackbays on sides, or middle supports justified to outside to accommodate mixers up to 32″
- Overbridge provides 7″ inside height clearance
- 28.5″ desktop height

MDV-CNR3 Corner Desk

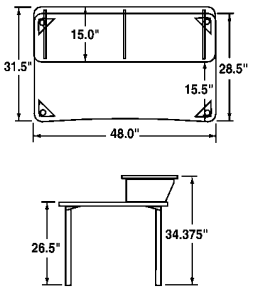




$660.00

- 3-piece overbridge accommodates mixers with tall meter bridges
- The overbridge center section can be adjusted to 11¼″ inside height clearance, or 16½″ inside height clearance
- Center section provides 30″ inside width clearance
- 28.5″ desktop height

MDV-DSK Straight Desk





$380.00

- Overbridge can be user-configured as two 4 space rackbays, or remove center section to accommodate mixers up to 39″ wide
- Overbridge provides 7″ inside height clearance
- 28.5″ desktop height

MULTIDESK VIDEO

MULTIDESK ACCESSORIES



$220.00

MDV-R12 Side Rack

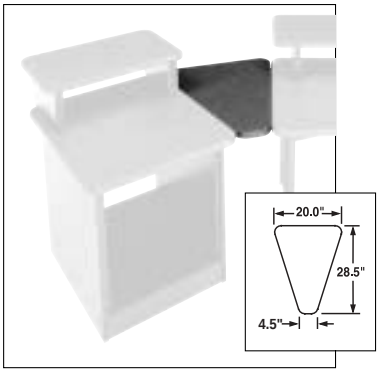
- Add to any MDV desk for additional rackmount capacity and desktop space
- 12 rackspaces
- Accepts optional casters
- 26.5 x 23 x 28.5″ (HxWxD)



MDV-OB4 Overbridge

- Mounts on top of the MDV-R12 rack
- Provides additional 4 rackspaces
- When placed beside any MDV desk, a contiguous surface is maintained
- 7.875 x 23 x 15″ (HxWxD)

$125.00



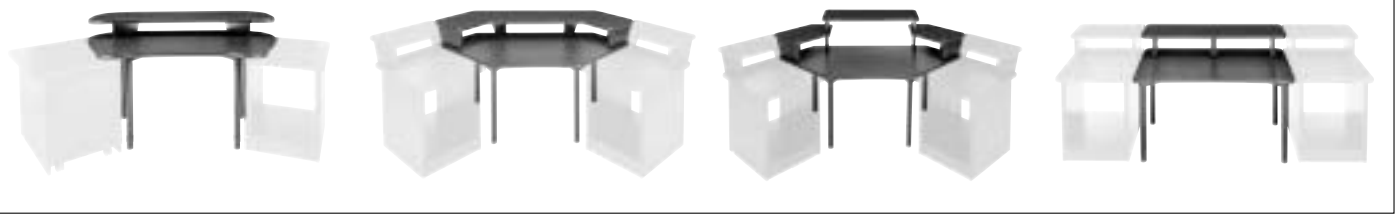
MDV-W Connecting Wedge

- Connect any MDV component at an ergonomic 35° angle
- Mounting hardware is included

$95.00

MDV-CK Caster Kit

- Includes four wheels, two metal caster saddles and mounting hardware. Fits MDV-R12 racks **72.00**



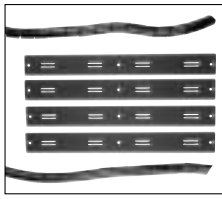
SPEAKER STANDS

Although designed as an accessory for the Edit Center system, these elegant speaker stands can easily integrate into any production environment including the multiDesk Video System.

- 36″ high speaker stands that are optimized to accommodate near and mid-field audio monitors in video and audio production and editing facilities
- High-mass all wood construction for resonance damping
- Includes carpet spikes and rubber feet to accommodate all floor surfaces
- Unique dual cable channels separate audio and AC cables (when using powered monitors), and include a wide molded cover
- 15¾″ x 15¾″ platforms and bases are reversible to show either a graphite or cherry wood laminate surface
- Elegant curved supports are finished in a durable black texture-lac



EC-SPK36 .....**300.00** *a pair*



Cable Management Kits

These convenient cable management kits effectively organize the mass of cable typically found in digital video and audio systems. Kit consists of spiral wrap to contain cables, and flat panels with tie-points to manage cable under-desktops.

CM-L: Cable management kit for 84″ desk system, includes 8 cable managers and 20′ cable wrap **70.00**
CM-S: Cable management kit for 60″ desk system, includes 4 cable managers and 20′ cable wrap **60.00**

STUDIO RACKS



OAK EQUIPMENT RACKS

OBRK Series 18˝ Deep Racks

- Attractive 3/4˝ oak laminate racks blends natural elegance with value-oriented engineering
- Assembles easily with the included concealed 1/4-turn fasteners and threaded inserts
- Factory-installed 11-gauge, 10-32 threaded rackrail
- Available in 8, 12, 16 and 20 rack spaces

OSR Series Sloped Racks

- The same construction as the OBRK series with a gently sloped rackface that facilitates easy equipment viewing
- Included casters enhance mobility
- Available with 16 and 24 rack spaces

OBR SERIES

Part #	Racking Height	Depth	Overall Height	Price
OBRK8	14˝ (8 Space)	18˝	15½˝	**135.00**
OBRK12	21˝ (12 Space)	18˝	22½˝	**145.00**
OBRK16	28˝ (16 Space)	18˝	29½˝	**170.00**
OBRK20*	35˝ (20 Space)	18˝	36½˝	**185.00**

OSR SERIES

Part #	Racking Height	Depth (Top - Bottom)	Overall Height	Price
OSR16	28˝ (16 Space)	19½˝ - 23˝	36½˝	**240.00**
OSR24**	42˝ (24 Space)	19½˝ - 23˝	50½˝	**275.00**

**Accommodates short vertical power strip. ** Accommodates medium and short vertical power strip. To install power strip, quick clips must be screwed to sides, or PB-5A used on optional rear rackrail.)*

BLACK LAMINATE EQUIPMENT RACKS

RK Series (16˝ Deep) and BRK Series (18˝ Deep) Equipment Racks

- Constructed with 5/8˝ furniture-grade MDF board
- High-quality black laminate finish
- Easily assembled with included tool and hardware
- 16˝ deep (RK Series), or 18˝ deep (BRK Series)
- Optional caster kit available (RKW)

Part #	Racking Height	Depth	Overall Height	Price
RK2	3½˝ (2 Space)	16˝	5˝	**85.00**
RK4	7˝ (4 Space)	16˝	8½˝	**92.00**
RK8	14˝ (8 Space)	16˝	15½˝	**108.00**
RK12	21˝ (12 Space)	16˝	22½˝	**115.00**
RK16	28˝ (16 Space)	16˝	29½˝	**125.00**
RK20*	35˝ (20 Space)	16˝	36½˝	**130.00**
BRK8	14˝ (8 Space)	18˝	15½˝	**115.00**
BRK12	21˝ (12 Space)	18˝	22½˝	**125.00**
BRK16	28˝ (16 Space)	18˝	29½˝	**130.00**
BRK20*	35˝ (20 Space)	18˝	36½˝	**145.00**

RKW Wheel kit-2 locking (adds 2¾˝ to height).....**40.00**

**Accommodates short vertical power strip. To install, quick clips must be screwed to sides, or PB-5A used on optional rear rackrail.)*



SRK Rolling Equipment Rack

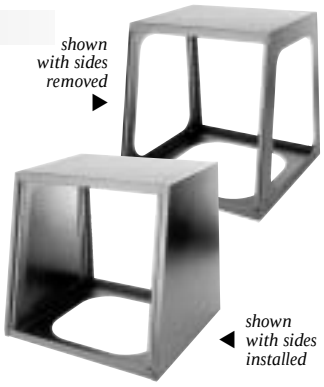
- 16 vertical rackspaces at bottom; 12 sloped rackspaces on top
- 54¼˝ high x 16˝ deep at bottom
- Black laminate finish
- Easily assembled with included hardware
- Factory-installed casters included

SRK racks accommodate medium and short vertical power strip. To install, quick clips must be screwed to sides, or PB-5A used on optional rear rackrail.)

STUDIO RACKS

Slim 2 Series Desktop Turret Racks

Fully Assembled Steel Frame with Laminate Sides



- Ships fully assembled
- Incorporates a sturdy black steel open frame and removable black laminate side panels
- Sloping rack face for easy equipment viewing
- Frames can easily be joined with the open sides and bottom allowing wire passage
- Front and rear 11-gauge, 10-32 threaded rack rail is standard

Part #	Racking Height	Top Depth	Bottom Depth	Overall Height	Price
2-10	17½″ (10 space)	14.1″	18″	19⅜″	**230.00**



KD Black Laminate

- Ships flat and assembles effortlessly with provided hardware
- Constructed from high pressure black melamine
- Sloping rack face offers easy equipment viewing
- Front and rear 10-32 threaded rack rail is standard

Part #	Racking Height	Top Depth	Bottom Depth	Overall Height	Price
2-10M	17½″ (10 space)	14.7″	17.9″	19⅝″	**175.00**
2-6M	10½″ (6 space)	15.9″	17.9″	12¾″	**155.00**

OBR/BRK/RK Series Front Doors



Racking Height	Solid Steel	Plexiglass	Vented Steel
14″ (8 space)	SSDR-8 **80.00**	PSDR-8 **130.00**	VSDR-8 **100.00**
21″ (12 space)	SSDR-12 **82.00**	PSDR-12 **1500.00**	VSDR-12 **120.00**
28″ (16 space)	SSDR-16 **86.00**	PSDR-16 **165.00**	VSDR-16 **135.00**
35″ (20 space)	SSDR-20 **100.00**	PSDR-20 **200.00**	VSDR-20 **150.00**

- Security doors for the OBRK, RK, and BRK series racks
- Protects equipment from tampering
- Standard keylocks, black powder coat finish
- Available in solid steel, plexiglass, and vented steel models

Desktop Rail Racks

Lets you quickly and inexpensively organize desktop processing and test equipment. Offered in 8 and 12 space sizes, both models extend upward at a gentle slope. Overall width is 19⅛″, depth is 11½″.



Part #	Racking Height	Overall Height	Price
DR-8	14″ (8 space)	15¾″	**46.00**
DR-12	21″ (12 space)	22¾″	**50.00**

Computer Enclosures

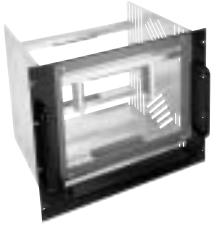
RM-CPU-ATX Rackmount Computer Chassis



- Vibration-isolated bay with a lockable, latched front door
- Rackmounts ATX motherboards
- Accommodates up to 5 drives (three 5¼″ and two 3½″ drives)
- Includes 2 cooling fans + washable filter
- 300 watt ATX Pentium 4 compliant power supply
- 7″ (4 space) height
- Accepts any motherboard up to 12 x 9⅜″

SL-18 Computer Chassis Slide Kit

- Allows you to slide rackmount computer chassis out for servicing
- 18″ extension length
- Sold in pairs.....**75.00**



RM-MM Computer Monitor Rackmount

- Adjustable clamps accommodate most 14″ to 17″ monitors
- Clear plexiglass window protects screen
- Hinged front door provides access to controls
- 15.75″ (9 space)

RM-KB Rackmount Computer Keyboard



- Includes high-quality 101-key AT style keyboard, trackball and padded wrist-rests
- Articulating arm provides effective cable management
- Complete with 12′ keyboard and trackball cables, including PS/2 connectors and a DB9 serial adaptor.
- Hinged cable manager in rear
- Occupies 1.75″ (1 space)

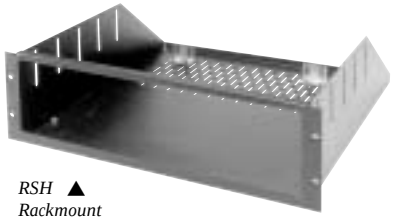


KB-SS Keyboard Sliding Shelf

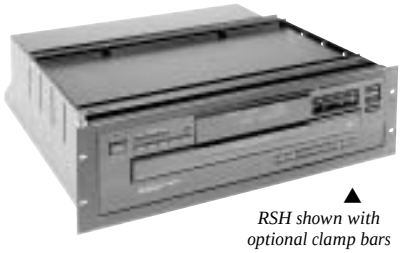
- Rackmount shelf occupies 1¾″ (1 space)
- Accommodates most standard small-pattern keyboards
- Articulating arm provides effective cable management
- Durable black powder coat finish

RSH SERIES CUSTOM RACKMOUNTS

The Professional's Choice for Trimming Non-Rackmount Components



RSH ▲ Rackmount



▲ RSH shown with optional clamp bars

- Install any component (up to 18¾″ wide) in a standard 19″ enclosure without making modifications
- Custom cut for a perfect fit and no gaps
- Over 6,000 component specifications are available in Middle Atlantic's database
- Racks include rear brackets, have a ventilated bottom, and assemble easily in minutes
- Bottom depths of 11½″, 14, 15½″, 17½″, or 20½″ are available
- Individual faceplates are also available in a black textured finish (SFACE), or a black brushed anodized finish (AFACE)
- Optional RSH clamps secure components to the shelf – rubber lining maintains a tight grip without scratching

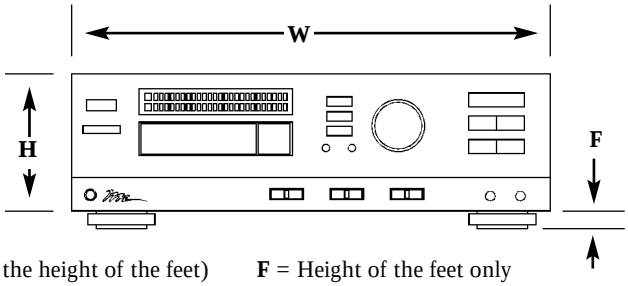
RSH-4S Black textured anodized finish **120.00**

RSH-4A Black brushed anodized finish.....**120.00**



Ordering Instructions

- To see if the component you want to rackmount is available in Middle Atlantic's database of over 6000 models, simply go to ww.middleatlantic.com and search their custom Rack Finder utility or, call us up at B&H and we'll be more than happy to check for you
- In the unlikely event that your component is not in the custom rack database, you can fax the measurements to Middle Atlantic

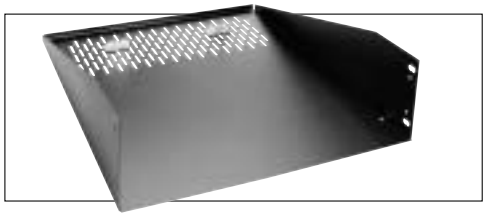


W = Faceplate width **H** = Faceplate height (not including the height of the feet) **F** = Height of the feet only

*** W, H and F dimensions must be within 1/32″. Depth must be within 1/8″*

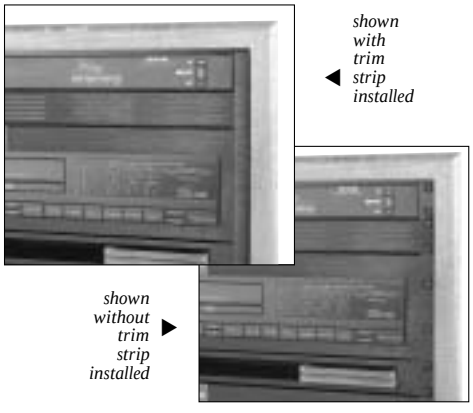
Wide Unit Shelf System

Accommodating components too wide to fit the 17⅜″ inside width of standard rackshelves, these shelf systems fit units up to 22½″ wide, and either bolt into millwork or to the mounting angles of standard rackrail. Occupies three rackspaces, the WUSS system comes complete with a pair of ears, a bottom, and appropriate assembly hardware.



WUSS11.5 11.5″ depth.....**54.00**
WUSS15.5 15.5″ depth.....**58.00**
WUSS17.5 15.5″ depth.....**60.00**
WUSS20.5 20.5″ depth.....**65.00**

Trim Strips



shown with trim strip installed ◀

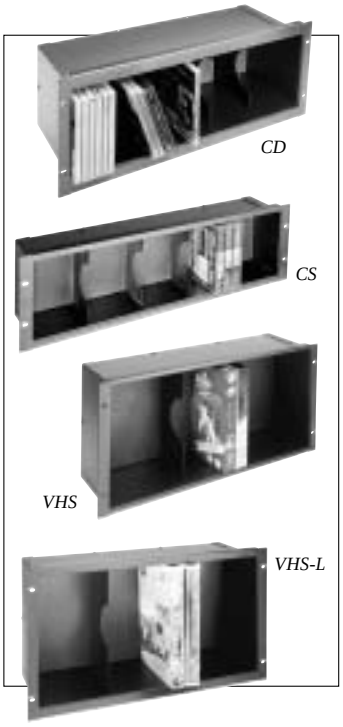
shown without trim strip installed ▶

As a final aesthetic complement to any installation, finishing trim strips offer a quick, clean, and effective method of covering rack screws once the job is complete. Each channel-shaped trim strip measures 45 spaces (78¾″ long), and features internal grooves that fit into the supplied snap washers. Offered in brushed black anodized or black powdercoat finish, trim strips are cut easily with a hacksaw. A cutting service is also available. Included decorative caps trim the ends, if required. Sold individually.

Part #	Description	Attachment Method	Price
TA	brushed black anodized	snap washers (8), end caps (4)	**45.00**
TS	black powder coat	snap washers (8), end caps (4)	**20.00**

** Works with all standard equipment mounting hardware except HW100 and HW500*

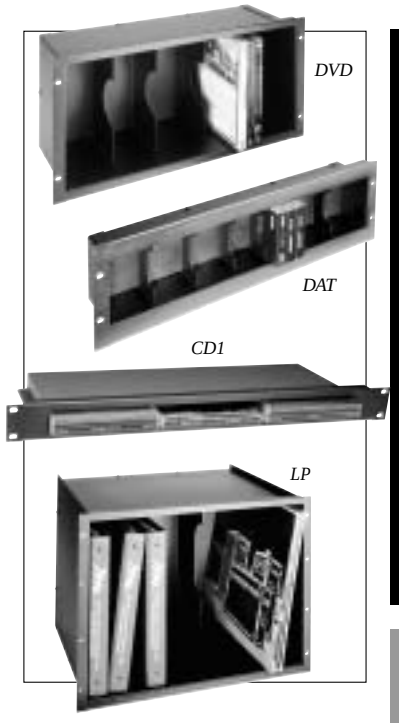
MEDIA STORAGE



Front-Loading Media Holders

Store DATs, CDs, DVDs, VHS tapes, cassettes, LPs and other recorded media in our rackmount media holders. Partitioned to prevent the contents from tipping, each model features a 16-gauge flanged aluminum face and a brushed, black anodized finish. Selected models are available with a textured black powder coat. All stored media protrudes 3/8″ to allow quick retrieval.

Part #	Fits	Height	Finish	Depth	Price
CD	40 CDs	7″ (4 space)	brushed	5¾″	**80.00**
TCD	40 CDs	7″ (4 space)	textured	5¾″	**80.00**
CS	25 audio cassettes	5¼″ (3 space)	brushed	3″	**85.00**
DAT	27 DAT tapes	5¼″ (3 space)	brushed	2 9/16″	**82.00**
VHS	14 VHS tapes in regular case	8¾″ (5 space)	brushed	4 13/16″	**88.00**
TVHS	14 VHS tapes in regular case	8¾″ (5 space)	textured	4 13/16″	**80.00**
VHS-L	13 VHS in padded case	10½″ (6 space)	brushed	5 9/16″	**96.00**
CD-1	6 CDs	1¾″ (1 space)	brushed	5½″	**45.00**
LP	LP's, binders	14″ (8 space)	brushed	12 9/16″	**150.00**
DVD	28 DVDs	8¾″ (5 space)	brushed	5½″	**99.00**



Heavy-Duty D and TD Series Drawers

Fully enclosed with inside dimensions measuring 15⅞″ wide by 14½″ deep, these drawers feature spring-loaded latches and rugged, straightforward construction. Faceplates come with a textured black powder coat (TD) or brushed and black anodized (D) finish.

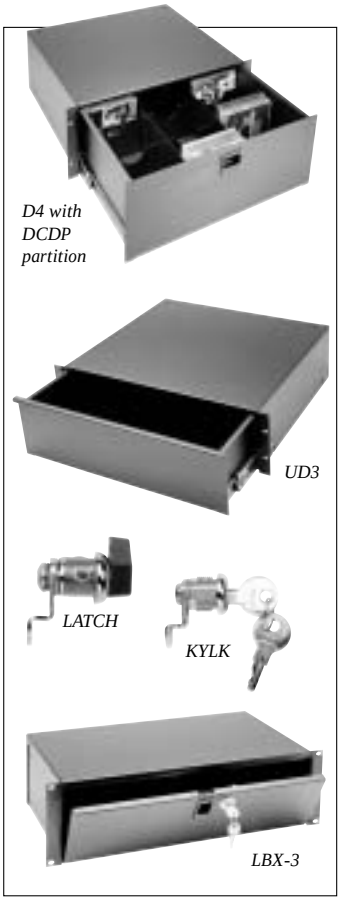
Anodized Finish Part #	Textured Finish Part #	Racking Height
D2 **115.00**	TD2 **110.00**	3½″ (2 space)
D3 **128.00**	TD3 **120.00**	5¼″ (3 space)
D4 **137.00**	TD4 **128.00**	7″ (4 space)
D5 **156.00**	TD5 **145.00**	8¾″ (5 space)

All D and TD Series Drawers are also available with installed keylocks.

Media Partitions for D, TD, and UD Series Drawers

Partition your D, TD, and UD series drawers for media storage. Snap-together fit is also handy for small parts and miscellaneous hardware.

Part #	Fits	Accommodates	Price
DDATP	D2, TD2, UD2	84 DAT tapes, hardware	**28.00**
DCSP	D2, TD2, UD2	58 cassette tapes	**28.00**
DCDP	D4, TD4, UD4	103 compact discs	**36.00**
DVDP	D4, TD4, UD4	40 DVD discs	**30.00**
DVHSP	D3, TD3, UD3	24 VHS tapes	**30.00**



UD Series Drawers

Drawer contents always remain inside with the UD series fully enclosed rackmountable utility drawers. Full extension ball bearing slides smoothly extend a full 14″ to reveal inside dimensions measuring 15⅞″ wide by 14½″ deep. Storage height is only 1/2 less than the actual racking height. Each model comes with a face mounted drawer pull. A user-installed latch and keylock are optionally available. Black powder coat finish.

Part #	Racking Height	Price
UD2	3½″ (2 space)	**92.00**
UD3	5¼″ (3 space)	**98.00**
UD4	7″ (4 space)	**115.00**

Keylocks and Latch

KYLK User-installed UD keylock **18.00**
LATCH User-installed UD latch **13.00**
ACC-KEY Additional set of keys for KYLK ...**29.00**

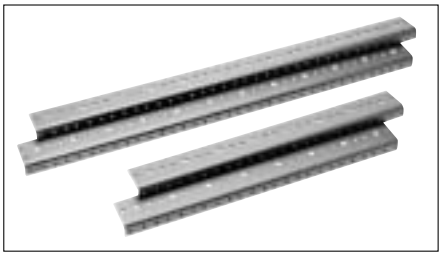
Lockboxes

Safely secure gear in our rackmount LBX series lockboxes. 9″ deep, 3 and 4 space models are offered with a durable black powder coat finish.

Part #	Racking Height	Depth	Price
LBX-3	5¼″ (3 space)	9″	**75.00**
LBX-4	7″ (4 space)	9″	**82.00**

ACCESSORIES

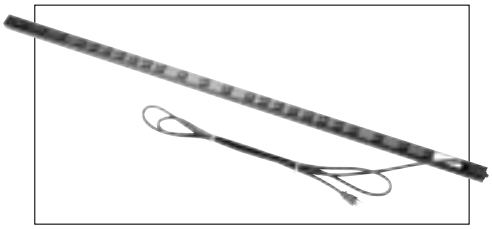
RRF Rackrail



Measuring 1/8″ thick, the all-steel RRF rack rail is threaded for 10-32 screws. With full hole spacing, they are sold in pairs and designed for use with 19⅛″ wide rough openings. Durable black powder coat finish, cutting service available.

Part #	Length	Racking Height	Price
RRF2	3½″	(2 space)	**6.95**
RRF4	7″	(4 space)	**10.95**
RRF6	10½″	(6 space)	**15.60**
RRF8	14″	(8 space)	**17.95**
RRF10	17½″	(10 space)	**21.50**
RRF12	21″	(12 space)	**25.00**
RRF14	24½″	(14 space)	**28.00**
RRF16	28″	(16 space)	**30.00**
RRF18	31½″	(18 space)	**32.00**
RRF21	36¾″	(21 space)	**36.50**
RRF27	47¼″	(27 space)	**44.78**
RRF45	78¾″	(45 space)	**72.50**

High Density 24-Outlet 15 Amp Slim Power Strip



High-density power strip boasts a unique feature in that while containing 24 power outlets, 11 of these outlets have been spaced to accommodate plug-in power supplies, resulting in no wasted outlets! Measuring 1¼″ deep by 1⅝″ wide, this power strip provides maximum useablity while taking up a minimum of space. Ships with 10′ power cord terminated with a NEMA 5-15P plug, 15 amp circuit breaker and mounting system
PD-2415SC-NS.....**96.00**

WL-60 Magnetic Work Light

Designed for use when installing and servicing equipment, the WL-60 provides convenient illumination in rack enclosure interiors. The unit features a strong magnetic base that attaches to any metal surface, and a quadruple-jointed support arm makes it easy to aim the light, which stays where you put it. The WL-60 includes a 60 watt "rough-service" bulb, aluminum shade with on/off switch, and an ultra-long 18′ line cord that eliminates the need for an extension cord.



WL-60.....**58.00**





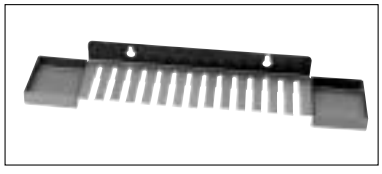
Acoustical Panels

Acoustic wall panels provide effective acoustic treatment for problem rooms. They are faced with 100% polyester fabric, which are easy to clean, and will not rip, tear, or crack like foam products. They feature resin-hardened edges which are beveled for aesthetics. In addition, due to superior materials used, AP series 2″ thick panels offer the same sound absorbing rating as 4″ thick panels. Acoustic panels are available in 4 sizes and two attractive colors, all 2″ thick.

Gray Color Part #	Beige Color Part #	Size	Price
AP24X24G	AP24X24B	24″ x 24″	**118.00**
AP24X48G	AP24X48B	24″ x 48″	**164.95**
AP30X48G	AP30X48B	30″ x 48″	**199.95**
AP30X84G	AP30X84B	30″ x 84″	**318.00**

The Claw

The unique dual-size "claw" configuration of this versatile wall-mount cable hanger provides efficent storage for all cables large and small. Convenient side trays hold small items. 16″ wide x 3½″ deep. Black powder coat finish.....**14.95**





MD-KB Computer Keyboard Tray

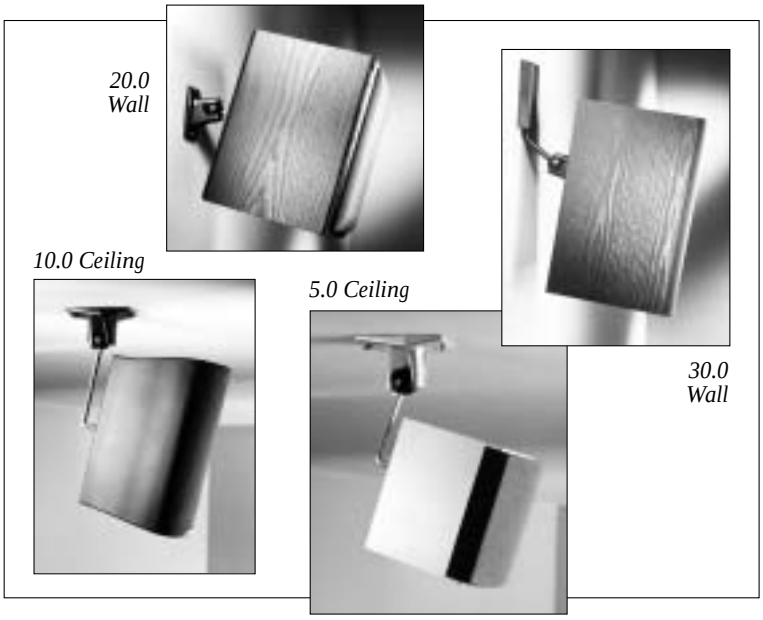
Designed for under-desk use, the MD-KB computer keyboard tray is equipped with a padded wrist support. The unit's built-in mouse tray can be pulled out to either the left or right side and retracts out-of-the-way when not in use. All mounting hardware included. (Does not work with MDV-CNR1 or MDV-CNR3.).....**99.95**

B&H PHOTO - VIDEO - PRO AUDIO

STAINLESS STEEL SERIES

Stainless Steel Speaker Mounts

For over 25 years OmniMount has been an innovator in the field of audio mounts. Today, the OmniMount name is synonymous with superior quality and has become the choice of professional installers and the brand most recommended by speaker manufacturers. Their mounts offer numerous advantages, including durable ergonomic designs, ease of installation and omni-directional positioning.
OmniMount's Stainless Steel series is a unique, top-of-the-line speaker mounting solution assembly, tops for functionality, safety, ease of installation, appearance, and multi-angle useage.



FEATURES

- ◆ Built with stainless steel hardware for strength, durability, and quality. Perfect for outdoor as well as indoor use
- ◆ Critical areas are polymer injected for less resonance, producing a more precise range of sounds
- ◆ All-inclusive combination of threads and/or mounting plates is designed to fit any model of speaker
- ◆ OmniMounts patented ball joint rotates and holds in any direction making it the most flexible solution
- ◆ Unique black, white or gray braided sheath included to hide steel finish if desired



5.0 Wall/Ceiling

- Contains both wall and ceiling mount options
- Drill bit and all necessary mounting hardware included
- Supports speaker up to 5 lbs
- Stainless Steel shaft with 4mm and 5mm thread
- Available in Black, White, or Grey.....**199.95**

10.0 Wall/10.0 Ceiling

- Sold separately as wall or ceiling mount, the 10.0 mounts are designed for bookshelf-size speakers
- Drill bit and all necessary mounting hardware included
- Supports speakers up to 10 lbs
- Stainless Steel shaft with 1/4″-20 thread
- Available in Black or White

20.0 Wall/20.0 Ceiling

- Sold separately as wall or ceiling mount
- Stainless Steel shafts and all necessary hardware included.
- Supports speakers up to 20 lbs.
- Available in Black or White **319.95**

30.0 Wall/30.0 Ceiling

- Sold separately as wall or ceiling mount
- Unique ball and clamp assembly
- Supports speakers up to 30 lbs
- Available in WA, WB, CA, & STMP configurations
- 1/2″-20 thread pattern
- Available in black or white

60.0 Wall/60.0 Ceiling

- Sold separately as wall or ceiling mount
- Includes all necessary mounting hardware
- Supports speakers up to 60 lbs.
- Available in WA, WB, CA, & STMP configurations
- 5/8″-18 thread pattern
- Available in black or white

120.0 Wall/120.0 Ceiling

- Sold separately as wall or ceiling mount
- Includes all necessary mounting hardware
- Supports speaker up to 120 lbs
- Available in WA, WB, CA, & STMP configurations
- 1/2″-20 thread pattern
- Available in black or white

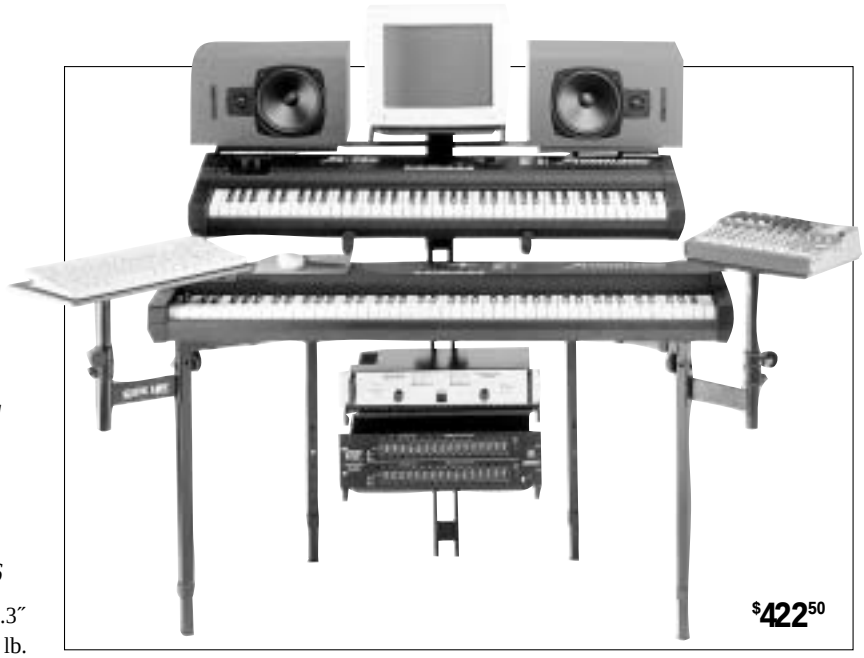
WS-500

All-Steel Modular Workstation System

The arc-welded, all-steel WS-500 is ideal both for the musician who needs a versatile, fully adjustable work place, as well as for home recording, project studios and schools. It features height adjustable shelves and tiers, and is capable of securely holding plenty of audio and computer equipment. All shelves are supplied with soft rubberized cushions that hold the gear firmly in place. Assembly requires no tools.

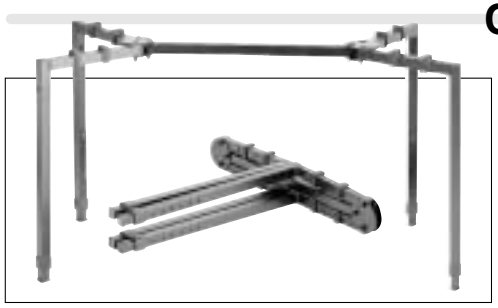
The WS-500 includes: WS-550, WS-561, WS-551, WS-552, WS-553, two WS-554's, QL-648 and QL-636

Height Range.....42 to 49.7″ **Depth Range**.....26.8 to 31.3″
Width Range.....22.7 to 46.4″ **Weight**.....64 lb.



$422.50

COMPONENTS



WS-550
Height Adjustable Base Unit

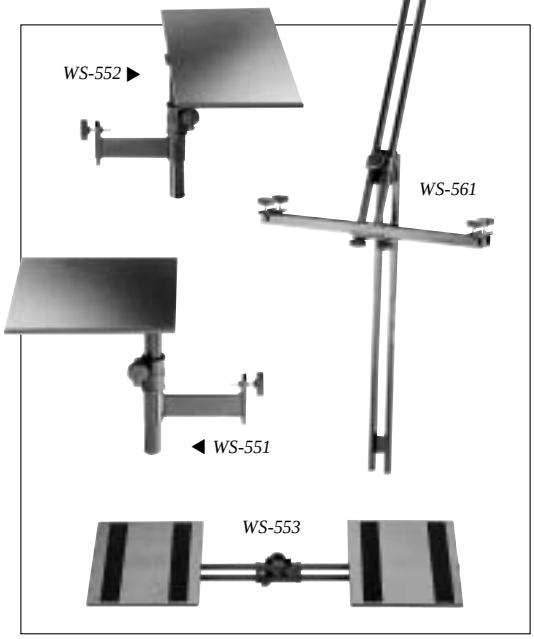
Used as the base for the WS-500 workstation. All legs are independently height adjustable and can be swung into position to conform to any keyboard, synthesizer or mixer.

Height Range.....24.25 to 32.5″ **Folded Height**.....24.6″
Width Range.....29.2 to 46.4″ **Frame Tubing Size**.....30mm
Depth Range.....17.3 to 27.4″ **Weight Capacity**.....250 lb.
Folded Width.....29.2″ **Weight**.....18 lb.

WS-561
Add-on Vertical Back Spine for WS-550

Optional spine kit that converts a WS-550 to a modular workstation starter frame for customized configurations. This allows use of all WS-500 workstation shelves.

Height Range.....42″ to 49.7″ **Width Range**.....27.2″ **Weight**.....14 lb.



WS-551 / WS-552
Utility Shelves

Solid metal, height adjustable, utility shelf can be used for holding mixers, drum machines or multi-track recorders, etc. Shelves can be rotated 360º for exact positioning.

Shelf Depth.....11.8″ (WS-551); 9.8″ (WS-552) **Weight Capacity**.....40 lb.
Shelf Width.....11.8″ (WS-551); 19.6″ (WS-552) **Weight**.....4.4 lb.(WS-551); 5.5 lb.″ (WS-552)
Height Adjustment Range, measured from main tier to top of shelf.....0 to 8″

WS-553
Width Adjustable, 90º Fixed-Angle Speaker Shelves

Arc-welded steel construction with rubber padding. Height-adjustable anywhere along workstation spine with a rubber-coated attaching clamp that won't scratch the stand.

Shelf Depth ...11.8″ **Shelf Width**....11.8″ **Weight Capacity**.....60 lb. (30 lb. each shelf)
Weight....9 lb. **Width Adjustment Range**, measured from outside edge of shelves....26.4″ to 43.3″

WS-500

WS-554
Small, 90º Fixed-Angle Shelf

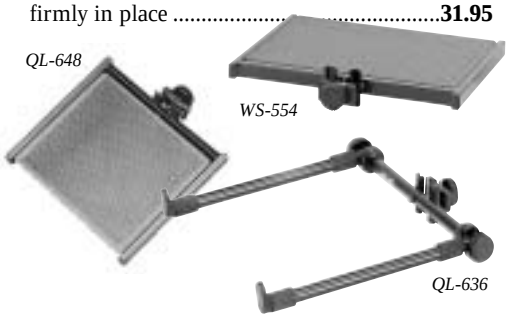
For the lower portion of the vertical back spine. Features a perforated metal surface and a rubber-coated attaching clamp **29.95**

QL-648
Small, 90º Fixed-Angle Shelf

Features a perforated metal surface and a rubber-coated attaching clamp **25.95**

QL-636
Large, Tilt-Adjustable Tier

Delivers 5 usable angle positions: 90º, 81º, 72º 63º, and 54º. Rubber rest pads hold equipment firmly in place **31.95**



TIER AND SHELF SPECS

Model	Width	Depth	Weight Capacity	Weight
WS-554	18.5″	11.8″	50 lb.	4 lb.
QL-648	13″	11.8″	50 lb.	3.3 lb.
QL-636	17.7″	13.8″	50 lb.	3.1 lb.

"H"-STYLE KEYBOARD STANDS

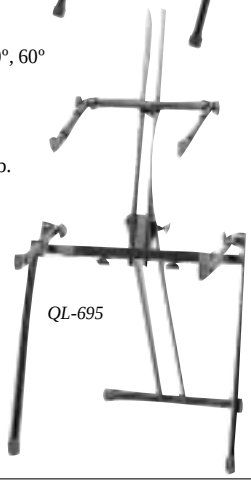
The QL-699 and QL-695 "H" stands offer sit down keyboard players comfort and flexibility. They are great for rehearsal rooms as well as project and home setups. The first tier is tilt and width adjustable to accommodate any keyboard up to 88 notes. Additional tiers are height and tilt adjustable and can hold 61 or 49 note keyboards up to 16.5″ deep. Both models fold flat without for easy transport.



Total Set-up Top to Bottom Height: 53.5″
Folded Height: 58.7″
Total Set-up Front to Back Depth: 29.3″
Side to Side Width Adjustment (first tier): 20.6″ to 31.7″
Tilt Adjustment Angles (first tier): 90º, 80º, 70º, 60º
Tilt Adjustment Angles Additional Tiers: 90º, 81º, 72º 63º, 54º
Weight Capacity (1st tier): 75 lb.
Weight Capacity of Each Additional Tier: 60 lb.
Weight: 25.3 lb. *(QL-699)*; 22 lb. *(QL-695)*



◀ *The QL-690/0, a stand-alone slant-back column display stand accepts the same optional accessories as the QL-699 and QL-695. (53″ set-up height, 23.6″ base set-up depth. Weighs 21 lbs.)*



QL-699.....**219.95** **QL-695** **189.95**

Optional Accessories for the WS-500 and "H" Stands

Tiers

Additional tiers feature rubber rest pads that hold equipment firmly in place and a rubber-coated attaching clamp will not scratch stand.

QL-634 Small Fixed-Angle Tier **22.50**
QL-635 Large Fixed Angle Tier.....**25.95**
QL-636S Short tilt adjustable tier. Usable angle positions are 90º, 81º, 72º 63º, 54º.....**28.50**

Shelves

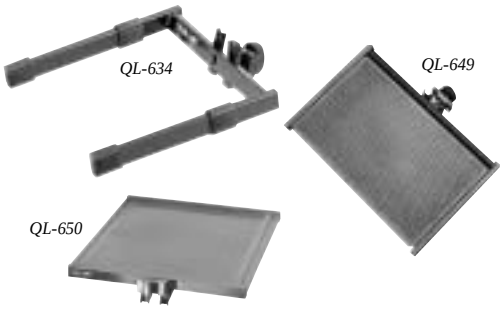
Additional shelves feature a perforated metal surface and a rubber-coated attaching clamp.

QL-649 Large, 90º fixed-angle shelf. Frame size is 30 x 15mm **28.50**

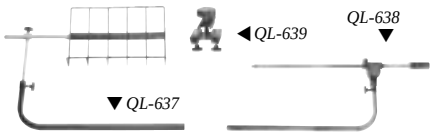
QL-650 Extra large, 90º fixed-angle shelf for computer monitors **37.50**

TIER AND SHELF SPECS

Model	Width	Depth	Weight Capacity	Weight
QL-634	13″	11.8″	50 lb.	2.2 lb.
QL-635	18.5″	11.8″	50 lb.	2.6 lb.
QL-636S	17.7″	10″	50 lb.	2.6 lb.
QL-649	18.5″	11.8″	50 lb.	4 lb.
QL-650	17.7″	17.7″	60 lb.	7.1 lb.



Misc. Furniture



QL-637 Adjustable Sheet Music Holder. Requires QL-639 Clamp. 12″ width; 8.6″ top to bottom height; 2″ bottom lip depth. Weighs 2.4 lb. **29.95**

QL-638 Fully Adjustable Fixed Length Mic Boom. Requires QL-639 Clamp. 24.8″ boom length. Weighs 3 lb. **28.50**

QL-639 Accessory Clamp. Needed to attach QL-637 and QL-638 mic boom. Only one clamp is needed when using both the QL-637 and QL-638.....**19.95**

QUIK LOK

Z SERIES WORKSTATIONS

Z-555 Expandable Music Project Station

The Z-555 workstation delivers sleek, attractive styling that is a perfect compliment to any home or studio environment. All work shelves and crossbars are height adjustable, and the speaker shelves may be mounted inward to save space. Quik Lok offers a wide range of options for musicians to configure their work area. Add computer peripherals, speakers, rack equipment, music keyboards, mixers, etc. The Z-555 has a computer-welded steel frame and 3/4˝ thick laminate work surfaces.



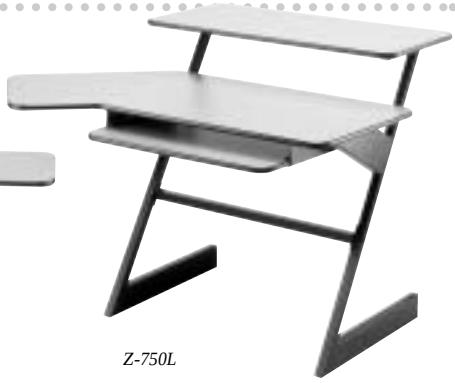
Z-555 SPECIFICATIONS	
MAIN SHELF	
Height*	26.75˝, 27.75˝, 28.75˝
Width	51.2˝
Depth	27.5˝
Thickness	3/4˝
MONITOR SHELF	
Height *	41˝
Width	11.8˝
Depth	11.8˝
Top Crossbar Height	36˝, 37˝, 38˝
FRAME	
Width Monitor Shelves Turned In	55˝
Width Monitor Shelves Turned Out	75.6˝
Depth of Legs	35.4˝
Weight Capacity	See capacity of components

** positions from floor*

Z-Shaped Designer Series Computer Furniture

Ergonomically designed triple-shelf workstations for maximum comfort and efficiency. Sleek, attractive styling adds an elegant look to any work environment. All models feature scratch-resistant 3/4˝ thick laminate shelves, while the computer-welded heavy-duty steel frame provides tremendous weight capacity, accomodating all kinds of computer equipment. Shelves are dove gray with black rubber edge moldings


Z-750R


Z-750L


Z-760


Z-250

Z-750R - Triple Shelf Workstation with 45º Right-Angled Main Shelf **227.50**

Z-750L - Triple Shelf Workstation with 45º Left-Angled Main Shelf **227.50**

Z-250 - Triple Shelf Workstation.....**218.95**

Z-760R - Triple Shelf Workstation with 45º Right-Angled Main Shelf and Height-Adjustable Top Shelf.....**259.95**

SPECIFICATIONS				
Model	**Z-750R**	**Z-750L**	**Z-250**	**Z-760**
MAIN SHELF				
Height*	28.5˝	28.5˝	28.9˝	28.5˝
Width	52.8˝	52.85˝	44˝	52.8˝
Depth	22.2˝	22.2˝	22˝	22˝
Thickness	3/4˝	3/4˝	3/4˝	3/4˝
TOP SHELF				
Height *	36˝	36˝	36˝	34.4˝ - 49˝
Width	38.6˝	38.6˝	38.6˝	38.6˝
Depth	17.1˝	17.1˝	17.1˝	17.5˝
Thickness	3/4˝	3/4˝	3/4˝	3/4˝
PULL-OUT SHELF				
Height*	25.8˝	25.8˝	25.8˝	25.8˝
Width	27.4˝	27.4˝	27.4˝	27.4˝
Depth	14˝	14˝	14˝	14˝
Thickness	3/4˝	3/4˝	3/4˝	3/4˝
FRAME				
Width	33.6˝	33.6˝	33.6˝	33.6˝
Leg Depth	22.3˝	22.3˝	22.3˝	22.3˝

** from floor*

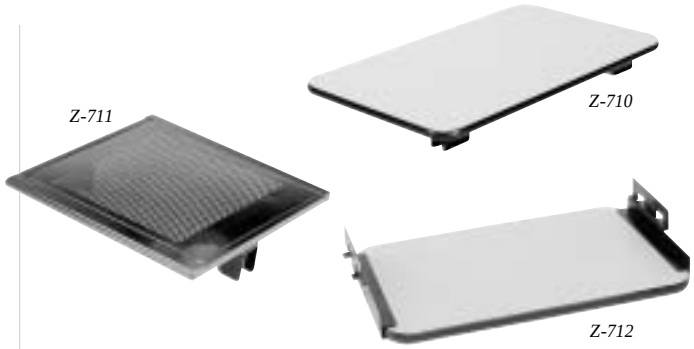
Z SERIES WORKSTATION ACCESSORIES

OPTIONAL SHELVES

Z-710
Mica Shelf
Attaches to the lower crossbar. The Z-710 can hold computer and audio equipment. Off-center design allows shelf to be mounted either closer or further away from user. Available in dove gray color. **49.95**

Z-711
Perforated Metal Shelf
Attaches to the lower crossbar. Off-center design. Available in black color only.....**29.95**

Z-712
Sliding Keyboard Shelf
Mounts to the main tier of the Z-555. Holds all computer keyboards and most 25 to 37 note keyboard controllers. 2.5″ clearance from the top of the main tier to the top of the sliding shelf. Dove gray color. For the Z-555 only **57.95**



SHELF SPECIFICATIONS

Model	Width	Depth	Thickness	Weight Capacity	Weight
Z-710	30″	18.3″	.71″	90 lb.	13.3 lb.
Z-711	13.9″	13″	N/A	75 lb.	3.3 lb.
Z-712	30″	14″	.71″	25 lb.	11.1 lb.

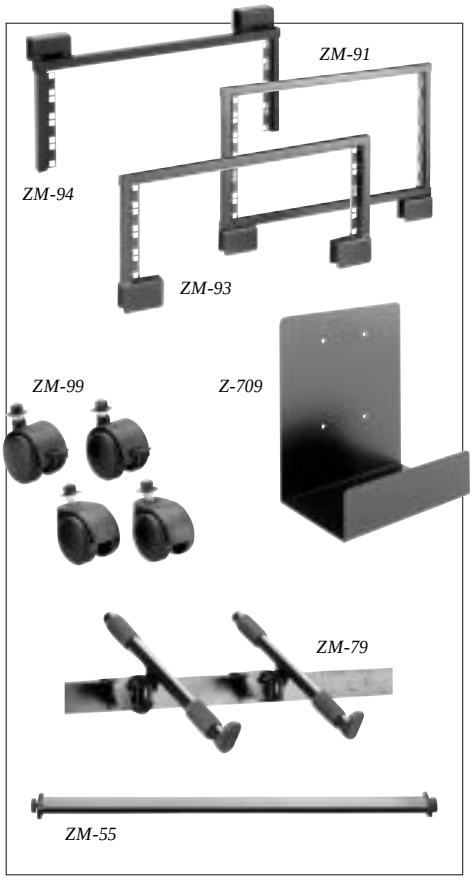
OPTIONAL ACCESSORIES

ZM-55
Lower Crossbar
Used to mount an optional shelf or equipment holder on the lower section of the Z-555. The ZM-55 is 51.1″ wide, has a tube size of 30 x 50mm, and weighs 5.3 lbs. It attaches to the stand frame 13.4″ from the floor. Available for the Z-555 only **44.95**

Z-709
Computer Holder
Mounts virtually any CPU tower to all Quik Lok "Z" stands. The Z-709 is 15.75″ high, 8.6″ wide, 13.8″ deep and weighs 3.3 lbs. It has a weight capacity of 44 lb **39.95**

ZM-97
Tilt & Width Adjustable Tier Set
Holds multi-track recorders, mixers, drum machines, keyboards or midi data input devices on the top crossbar of the Z-555 workstation. The ZM-97 features the "Dino-Bite" locking disc for secure, non-slip tilt adjustments. It's dimensions are: 14.3″ depth; 90º, 80º, 70º, 60º usable angle positions; and 60 lb. weight capacity. It weighs 4 lbs **34.95**

ZM-99
"Z" Stand Caster Kit
2 locking and 2 non-locking casters **19.95**



Rack Equipment Holders
Available in 6-space and 4-space versions, these rack equipments holders and are used to hold power amps, outboard gear, tone generators / samplers or any rack mountable equipment.

ZM-91 - 6-Space Holder
Rests securely on top of the lower crossbar. It cannot be used with the Z-712 pull-out computer shelf.....**34.95**

ZM-93 - 4-Space Holder
4-space version of the ZM-91 .....**33.50**

ZM-94 - 4-Space Holder
Mounts from underneath the "Z" stand crossbar.....**44.950**

RACK HOLDER SPECS

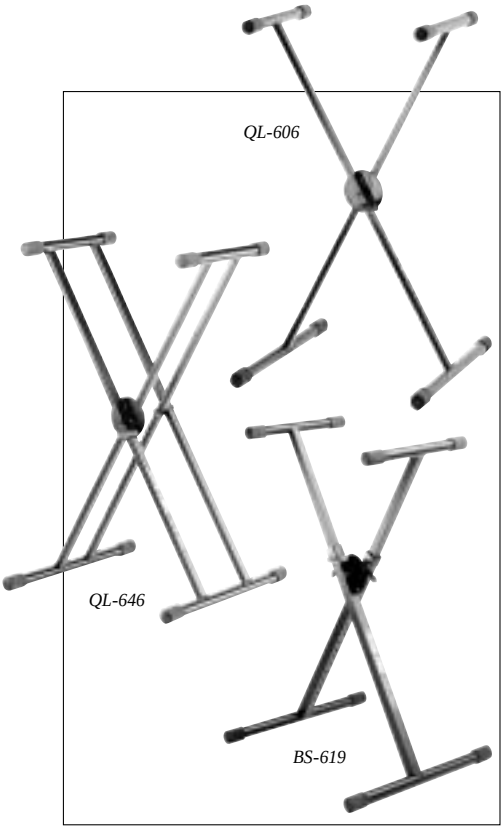
Model	Width	Depth	Weight Capacity	Weight
ZM-91	20.7″	12.6″	120 lb.	4.9 lb.
ZM-93	20.7″	9″	80 lb.	3.3 lb.
ZM-94	20.7″	9″	80 lb.	3.3 lb.

ZM-90
Power Strip Holder
Will secure virtually any non-rackable A/C power strip on the lower crossbar, for neater cable management **19.95**

KEYBOARD STANDS

Keyboard X-Stands with the Quik Lok Locking Mechanism

Keyboard stands that use the "Original" locking disc are by far the most popular among the most demanding professionals. This battle-tested disc system has been accepted as the industry standard since 1978. Musicians around the world have relied upon this line of stands for their durability, ease of use and dependability.



Height and width positions
QL-606 / QL-646
Height: *36.6″, 34.5″, 31.2″, 27.4″, 23″*
Width: *14.4″, 19.8″, 24.5″, 28.7″, 2.3″*
BS-616
Height: *42.6″, 39.5″, 35.4″, 30.6″, 25″*
Width: *16.5″, 22.5″, 27.8″, 32.7″, 36.8″*

X-STANDS			
Model	**QL-606**	**QL-646**	**BS-619**
Leg Depth	20″	20″	19.5″
Tier Depth	13.75″	13.75″	13.75″
Folded Height	38.75″	38.75″	28″
Weight Capacity	180 lb.	200 lb.	180 lb.
Weight	7.3 lb.	10.6 lb.	9.5 lb.
Brace Tube Size	30 x 15mm	30 x 15mm (x2)	25 x 25mm (upper) 30 x 30mm (lower)
Tier Tube Size	30mm	30mm	30mm

QL-606 X-Stand
Single-Brace, Single Tier.....**44.95**

QL-646 X-Stand
Double-Brace, Single Tier....**64.95**

BS-619 X-Stand
Portable Full-Sized Keyboard with Collapsible Telescopic Tiers.....**64.95**

X-SERIES OPTIONAL ACCESSORIES

Tiers

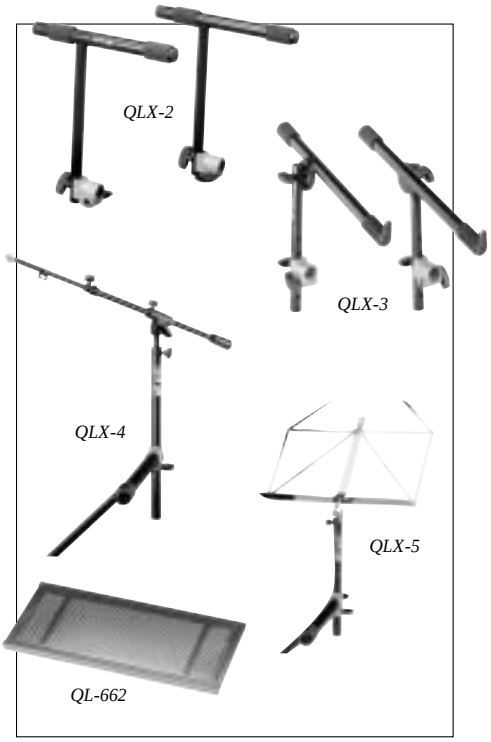
Second Tiers offer 5-position height adjustment and allow a QLX-4 Mic boom or QLX-5 music holder to be added.

QLX-2 Height Adjustable, Fixed-Angle Second Tier - 90º fixed angle allows top keyboard to be moved closer or further away from user.....**29.95**

QLX-3 Height, Tilt and Angle Adjustable Second Tier - with four-position angle adjustment **37.50**

QLX-1 Height, Tilt, Angle and Depth Adjustable Second Tier - with 4-position angle adjustment and 7.6″ depth adjustability.....**Call**

TIER SPECS			
Model	**Depth**	**Weight Capacity**	**Weight**
QLX-2	13.38″	100 lb.	4 lb.
QLX-3	17.3″	75 lb.	5.5 lb.
QLX-1	15.7″	75 lb.	6.6 lb.



QLX-4 Fully Adjustable Telescopic Mic Boom - swivels to any position, five position height adjustments, extends to 40.5″ **29.95**

QLX-5 Fully Adjustable Sheet Music Holder - Five position height adjustment. Folds easily for transport **33.50**

QL-662 Large Metal Shelf - Perforated metal shelf with reinforced steel bracing converts any full-sized X stand or second tier into a convenient table for holding drum machines, multi-track recorders or any kind of table top equipment. The QL-662 can be used on the following X stands and tiers: T-10, T-20, T-14, QLX-11, QLX-21, QLX-2, QL-606, QL-609, QL-646 & BS-619.....**54.95**

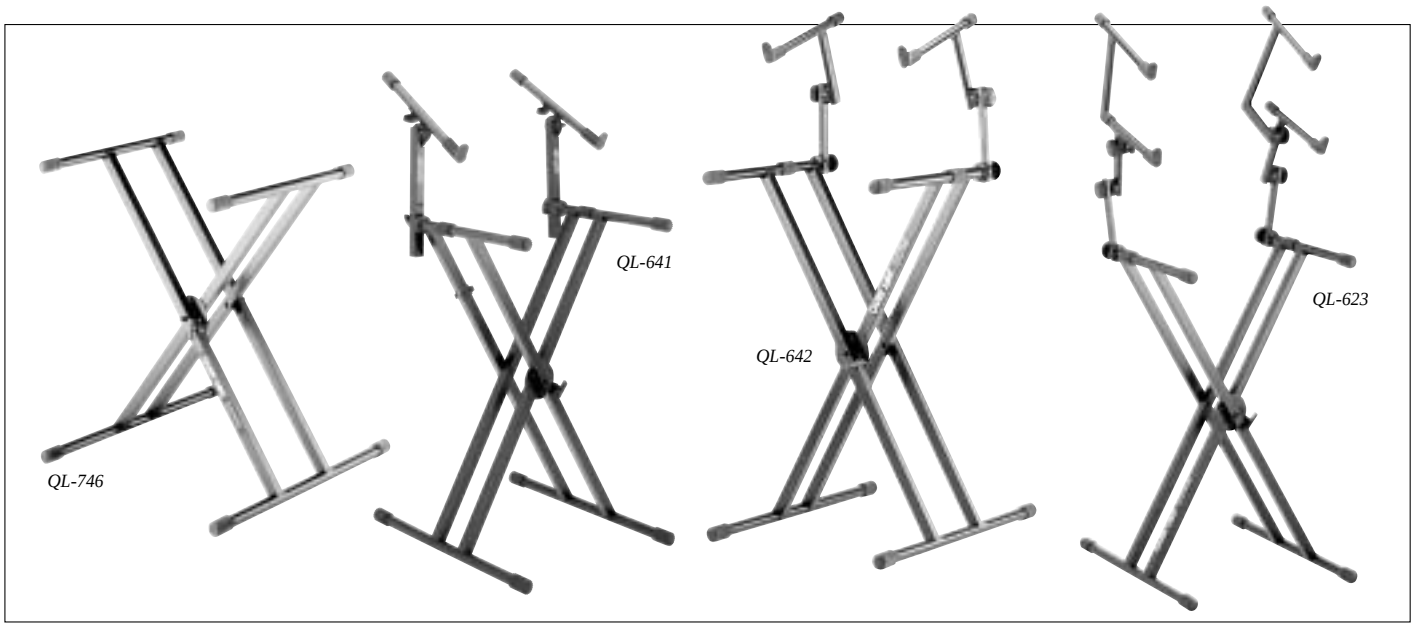
QLX-4, 5 & QL-662 SPECS				
Model	**Width**	**Depth**	**Height Positions**	**Weight**
QLX-4	—	—	11.25″, 10.2″, 8.9″*	2.4 lb.
QLX-5	17.7″	2.3″◊	6.5″ to 11.25″	5.5 lb.
QL-662	28″	15.4″	—	10 lb.

** from top of tier to boom swivel ◊ Boom Lip depth*

KEYBOARD STANDS

Pro Series Heavy-Duty X-Stands with Locking Disc System

The absolute hands-down choice of proffesional musicians the world over because of their adjustability, strength and reliability. All stands feature the "Original" locking disc system and super heavy-duty 30 x 30mm steel tube bracing for extra sturdiness and structural support. The Pro Series stands are capable of holding the heaviest keyboards safely and securely.



PRO-SERIES X-STANDS SPECIFICATIONS				
Model	**QL-746**	**QL-641**	**QL-642**	**QL-623**
Leg Depth	26″	26″	26″	26″
Main Tier Depth	20″	17.9″	17.9″	17.9″
Second Tier Depth	—	15.8″	15.8″	12″
Third Tier Depth	—	—	—	12″
Second Tier Angles	—	90°, 77°, 64°, 51°	90°, 77°, 64°, 51°	90°, 81°, 72°, 63°, 54°
Third Tier Angles	—	—	—	90°, 81°, 72°, 63°, 54°
Folded Height	40.4″	40.4″	40.4″	40.4″
Weight Capacity	260 lb.	265 lb. (overall) 75 lb. (2nd tier)	265 lb. (overall) 75 lb. (2nd tier)	265 lb. (overall) 60 lb. (2nd & 3rd tiers)
Weight	14.5 lb.	20 lb.	20 lb.	24 lb.
Brace Tube Size	30 x 30mm	30 x 30mm	30 x 30mm	30 x 30mm
Tier Tube Size	30mm	30mm	30mm	30mm

Height and width positions

QL-746 / QL-641

Height:
38″, 34.5″, 32″, 27.8″, 23″

Width:
15.5″, 20.9″, 25.9″, 30.4″, 34.1″

QL-642 / QL-623

Height:
38″, 34.5″, 32″, 27.8″, 23″

Width:
15.5″, 20.9″, 25.9″, 30.4″, 34.1″

QL-746 X-Stand
Heavy-Duty Double Brace, Single Tier.
Options: QLX-1 (adjustable second tier), QLX-2 (height-adjustable fixed-angle second tier), QLX-4 (telescopic mic boom), QLX-5 (sheet music holder), and QL-662 (large metal shelf) **72.50**

QL-642 X-Stand - Heavy-Duty Double Brace, with Height and Tilt Adjustable Second Tier.
Options: QLX-3 (adjustable second tier), QLX-4 (telescopic mic boom) and QLX-5 (sheet music holder) **113.50**

QL-641 X-Stand
Heavy-Duty, Double Brace with Height, Depth and Tilt Adjustable Second Tier. Options: QLX-1(adjustable second tier), QLX-3 (adjustable second tier), QLX-4 (telescopic mic boom) and QLX-5 (sheet music holder) **119.95**

QL-623 X-Stand
Heavy-Duty, Double Brace with Tilt Adjustable Second and Third Tier. Options: QLX-4 (telescopic mic boom) and QLX-5 (sheet music holder) **139.95**

STUDIO RACKS and ACCESSORIES

ER ECONOMY RACKS

The simplest of designs, these knock down economy racks are black oak laminate over 5/8″ medium density fiberboard. They provide a very economical yet attractive means for rack mounting audio gear in home studios or smaller installations. Assembly hardware and 11-gauge steel rack rail are provided. An optional swivel base (S-10) allows for 360° rotation for either the ER-8 or ERS-10. Perfect for use on table or desktop. Furniture grade nylon hooded casters are optional on all models.

Model	Rack Unit	Depth	Width	Height	Price
ER-4	4	15.75″	20.50″	8.625″	**69.95**
ER-8	8	15.75″	20.50″	15.875″	**83.95**
ER-12	12	15.75″	20.50″	22.875″	**89.95**
ER-16	16	15.75″	20.50″	29.875″	**97.95**
ER-20	20	15.75″	20.50″	36.875″	**106.95**
ER-8-18	8	18″	20.50″	15.875″	**92.95**
ER-12-18	12	18″	20.50″	22.875″	**99.95**
ER-16-18	16	18″	20.50″	29.875″	**106.95**
ER-20-18	20	18″	20.50″	36.875″	**114.95**
S-10 Swivel base.....**19.95**			**Casters** Set of 4.....**58.95**		

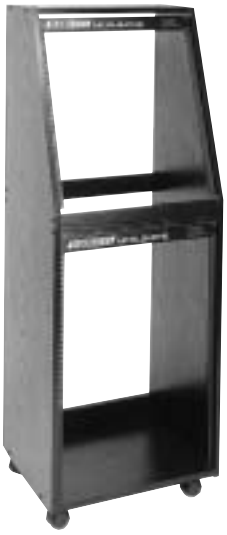


ER SLANTED ECONOMY RACKS



These economy racks are constructed with heavy duty 3/4″ medium density fiberboard, attractively finished with a black oak laminate. All three units are shipped knocked down and assemble in minutes with the hardware kit provided. Rack rail included is of 11-gauge, cold rolled steel. Furniture grade casters are standard equipment on these slant racks. A great combination of form and function.

Model	Rack Spaces	Top Depth	Bottom Depth	Width	Height	Price
SLANTED RACKS						
ERS-10	10	12.625″	16.5″	20.50″	20.375″	**10495**
ERS-18	18	12.5″	18″	20.50″	38.75″	**153.95**
SLANTED over STRAIGHT RACKS						
ER-12-8	20	8.25″	15.35″	20.50″	39″	**153.95**
ER-12-16	28	8.65″	15.75″	20.50″	53″	**195.95**





SCDR Security Door

Steel security door mounts to rack rail. Available in 8, 12, 16 and 20 spaces to match the ER and ERK series racks. Includes key locks and all hardware.

SCDR-8B for ER-8 and ERK-8 .....**58.95**
SCDR-12B for ER-12 and ERK-12 .....**64.95**
SCDR-16B for ER-16 and ERK-16 .....**78.95**
SCDR-20B for ER-20 and ERK-20 .....**89.95**



ERD Economy Rack Door

Attractive smoked 3/16″ plexi door to fit the economy racks. Textured powder coat finish with keyed lock. Attaches easily, and comes in sizes to fit the ER-8, 12, 16 or 20.

ERD-8 for ER-8.....**58.95**
ERD-12 for ER-12.....**81.95**
ERD-16 for ER-16.....**99.95**
ERD-20 for ER-20.....**125.95**

STUDIO RACKS and ACCESSORIES

ERK ELITE RACKS

The Elite Series of fine audio racks are of knock down construction using the highest grade of fused 3/4″ melamine. Available in 4, 8, 12, 16, and spaces with a straight front or 16, 20 and 24 spaces slanted, as well as a 12 over 8 spaces configuration. They come equipped with 11-gauge steel rack rail and the finest knockdown furniture hardware available. Edges are attractively trimmed in solid oak or maple, oiled and rubbed to enhance the warmth of the wood. Available in ebony fleck and maple. Furniture grade casters are standard on 16 spaces and larger.

STRAIGHT MODELS

Model	Rack Spaces	Depth	Width	Height	Height w/casters	Price
ERK-4	4	14″	20.75″	8.875″	—	**1049.95**
ERK-8	8	15.875″	20.75″	16.875″	—	**125.95**
ERK-12	12	15.875″	20.75″	23.875″	—	**139.95**
ERK-16	16	15.875″	20.75″	30.875″	32.625″	**153.95**
ERK-20	20	15.875″	20.75″	37.875″	39.625″	**167.95**

SLANT MODELS

Model	Rack Spaces	Depth Top	Depth Bottom	Width	Height	Height w/casters	Price
ERK-165L	16	15″	18″	20.75″	33.50″	35.25″	**195.95**
ERK-20SL	20	14.25″	18″	20.75″	40.50″	42.25″	**209.95**
ERK-24SL	24	13.25″	18″	20.75″	47.50″	49.25″	**223.95**
ERK-12/8	20	9″	15.875″	20.75″	40.25″	42″	**229.95**



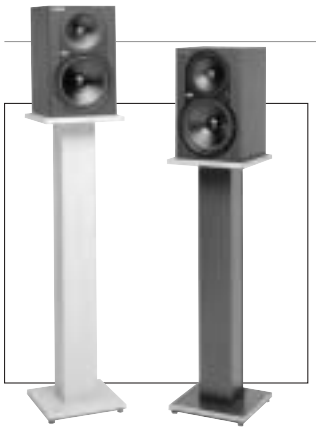
SPEAKER STANDS

ERSS Economy Speaker Stands

These monitor stands are ideal for any studio and are constructed of 3/4″ furniture grade hardboard, finished in a black oak laminate. The speaker platform and base are 12″ square. The bottom is easily removed to allow the 5″ center post to be filled with bagged sand for acoustical de-coupling. Sold in pairs, these stands are shipped knocked down in one box and assemble in minutes. Sold in pairs.

ERSS-36 36″ speaker stands.....*pair* **99.95** **ERSS-42** 42″ speaker stands.....*pair* **107.95**





ESS Elite Speaker Stands

Elite speaker stands are sure to complement the aesthetics of any room or studio. They are attractively constructed from fused melamine. The speaker platform and base are edged with solid oak or maple that is sealed with linseed oil to enhance the warm wood finish. The stands are either 36″ or 42″ tall and both the speaker platform and base are 12″ square. These stands are sold in pairs and are shipped knocked down. Assembly is simple. Bagged sand may be added to the 5 x 5″ center post for acoustical de-coupling. Sold in pairs.

ESS-36B 36″ ebony fleck speaker stands...**129.95** **ESS-36M** 36″ maple speaker stands **129.95**

ESS-42B 42″ ebony fleck speaker stands...**136.95** **ESS-42M** 42″ maple speaker stands **136.95**

RAXXESS

CONFIG-U-RAXX

Modular Workstation Furniture System



The Config-U-Raxx studio furniture system from Raxxess is a completely modular workstation designed to compliment the modern studio environment. The Config-U-Raxx line consists of an array of desktop surfaces, side wings and shelves, as well as add-ons such as doors and keyboard trays.Each component can be easily interfaced with any other component in the system,allowing for an almost endless variety of configurations. Future expansion of the system will be just as easy. A wide range of components allows a system to be configured to suit any room size and shape, while the new height adjustment system allows any desktop surface to be adjusted from 26˝ to 31˝ above the floor. These features make the Config-U-Raxx system just as comfortable for the MIDI composer as it is for the audio engineer.

The Config-U-Raxx line consists of an array of desktop surfaces, side wings and shelves, as well as add-ons such as doors and keyboard trays. Each component can be easily interfaced with any other component in the system, allowing for an almost endless variety of configurations. Future expansion of the system will be just as easy.

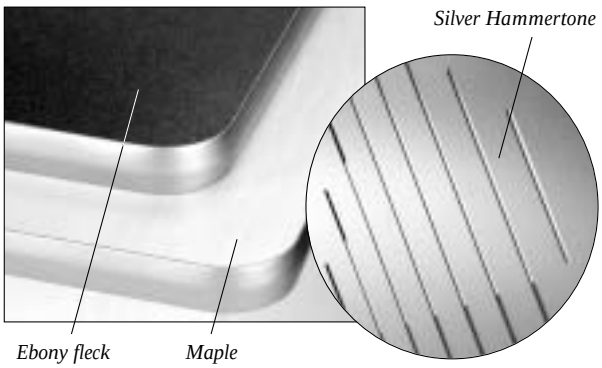


Supported by an all steel substructure, this furniture is designed to provide years of trouble-free service and rock-solid performance. The Config-U-Raxx system is available in maple and ebony fleck finishes with silver T-mold trim. All steel surfaces are coated with an attractive silver hammer tone finish.

FEATURES

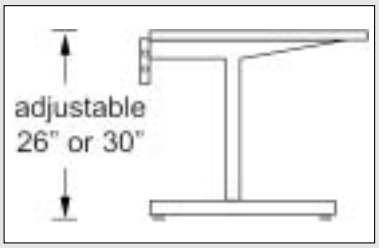
- All steel substructure
- Metal portions finished with an attractive Silver “Hammertone” finish
- Steel side wing racks available in your choice of 2 heights:
- 12 rack spaces - 26˝ height
- 14 rack spaces - 30˝ height
- Maple or Ebony side wing racks available in 14 rack spaces
- All center desks are adjustable in height to fit either height side wings

The Config-U-Raxx line consists of an array of desktop surfaces, side wings and shelves, as well as add-ons such as doors and keyboard trays. Each component can be easily interfaced with any other component in the system, allowing for an almost endless variety of configurations. Future expansion of the system will be just as easy.



Center Desks

Center Desks are the "centerpiece" of your Config-U-Raxx system. They are availble in maple or ebony fleck with silver trim and are height adjustable from 26˝-30˝ for use with 12U or 14U Side Wings (sold seperately). Start by choosing the center desk that best suits your needs.



SCD-36
Straight Center Desk- small



SCD-36B Straight Center Desk- small-ebony.....**269.95**
SCD-36M Straight Center Desk- small-maple.....**269.95**

SCD-42B
Straight Center Desk- Large



SCD-42B Straight Center Desk- large-ebony.....**289.95**
SCD-42M Straight Center Desk- large-maple.....**289.95**

ACD-30-45B and ACD-42-56B
Angled Center Desks







ACD-30-45B Angled Center Desk- small-ebony **279.95**
ACD-30-45M Angled Center Desk- small-maple **279.95**
ACD-42-56B Angled Center Desk- large-ebony **335.95**
ACD-42-56M Angled Center Desk- large-maple.....**335.95**

CRND *Corner Desks*



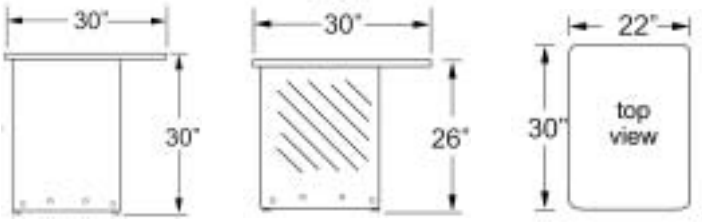
CRND-B Corner Desk-ebony **349.95**
CRND-M Corner Desk-maple.....**349.95**

CONFIG-U-RAXX

Side Wings

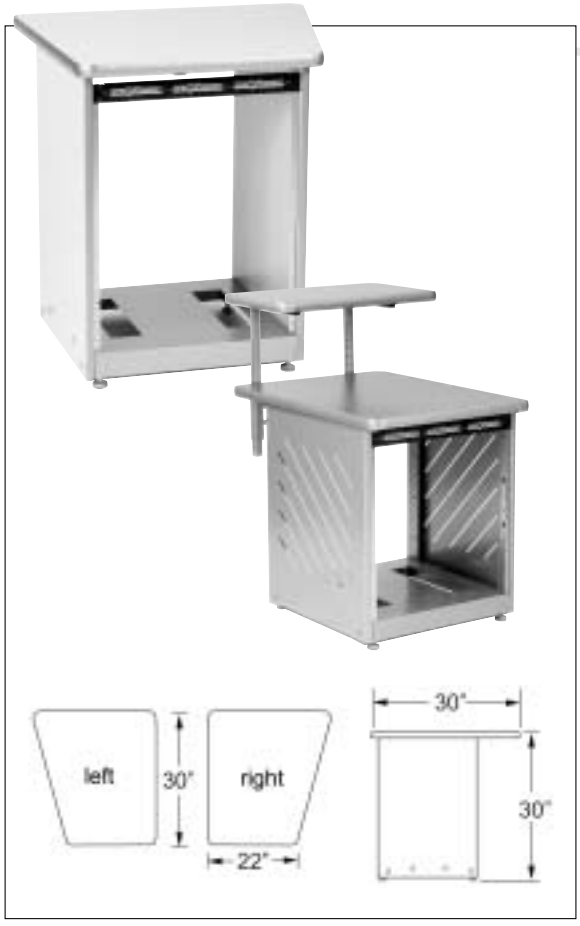
SSW-22

Available as 14U rack with melamine sides in ebony fleck or maple, 14U steel sides with silver hammertone finish and 12U steel sides with silver hammertone finish. 1⅛˝ thick melamine top surface available in ebony fleck or maple with silver trim. Doors and shelves are sold separately.





SSW-22B Straight Side Wing- wood sides-ebony-14U **165.95**
SSW-22M Straight Side Wing- wood sides-maple-14U.....**165.95**
SSW-22-12BS Straight Side Wing- metal sides-ebony-steel-12U ..**179.95**
SSW-22-12MS Straight Side Wing- metal sides-maple-steel-12U..**179.95**
SSW-22BS Straight Side Wing- metal sides-ebony-steel-14U **195.95**
SSW-22MS Straight Side Wing- metal sides-maple-steel-14U....**195.95**



Angled Side Wings

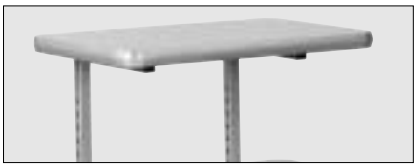
ASW-20-29

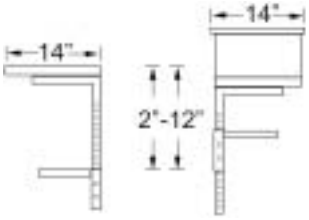
Angled side wings bring your rackmounted equipment closer to your sitting position. Available as 14U rack with melamine sides in ebony fleck or maple, 14U steel sides with silver hammertone finish and 12U steel sides with silver hammertone finish. 1⅛˝ thick melamine top surface available in ebony fleck or maple with silver trim. Doors and shelves are sold separately. Doors and shelves are sold separately. You must specify right or left-hand side when ordering.

ASW-20-29LB Angled Side Wing- left-wood sides-ebony-14U.....**209.95**
ASW-20-29LM Angled Side Wing- left-wood sides-maple-14U **209.95**
ASW-20-29/12LBS Angled Side Wing- left-metal sides-ebony-steel-12U **229.95**
ASW-20-29/12LMS Angled Side Wing- left-metal sides-maple-steel-12U.....**229.95**
ASW-20-29LBS Angled Side Wing- left-metal sides-ebony-steel-14U.....**244.95**
ASW-20-29LMS Angled Side Wing- left-metal sides-maple-steel-14U.....**244.95**
ASW-20-29RB Angled Side Wing- right-wood sides-ebony-14U **209.95**
ASW-20-29RM Angled Side Wing- right-wood sides-maple-14U **209.95**
ASW-20-29/12RBS Angled Side Wing- right-metal sides-ebony-steel-12U **229.95**
ASW-20-29/12RMS Angled Side Wing- right-metal sides-maple-steel-12U...**229.95**
ASW-20-29RBS Angled Side Wing- right-metal sides-ebony-steel-14U.....**244.95**
ASW-20-29RMS Angled Side Wing- right-metal sides-maple-steel-14U **244.95**

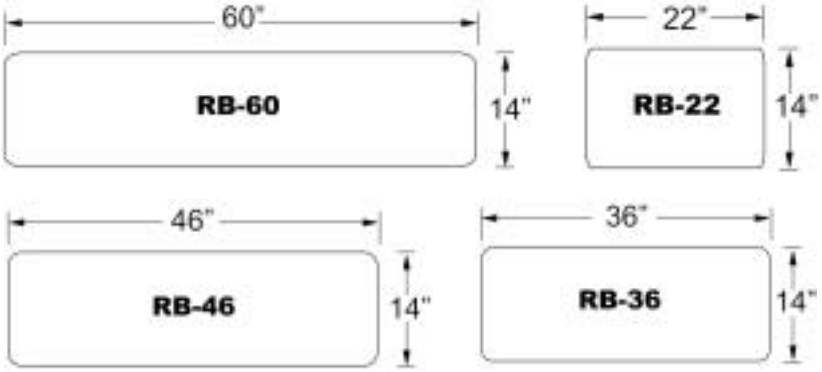
CONFIG-U-RAXX

Shelves





Shelf/rack height is adjustable from 2″ to 12″ above your desktop.

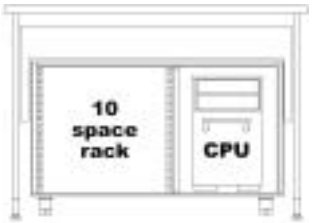


RB-60B Upper shelf for center desks- extra large-ebony.....**139.95**
RB-60M Upper shelf for center desks- extra large-maple **139.95**
RB-46B Upper shelf for center desks- large-ebony.....**119.95**
RB-46M Upper shelf for center desks- large-maple.....**119.95**
RB-36B Upper shelf for center desks-small-ebony **99.95**
RB-36M Upper shelf for center desks-small-maple **99.95**
RB-22B Upper shelf for side wings- incl. hardware-ebony **84.95**
RB-22M Upper shelf for side wings- incl. hardware-maple **84.95**

UCR

Under Counter 10 Space Rack with CP Shelf





UCR shown with SCD-42 (optional)

KBS1 Keyboard Shelf and KBS-2 Keyboard/Mouse Shelf



With the 3-D adjustable shelf, you can put your keyboard exactly where you want it! This shelf slides forward and back, up and down, tilts up and down and swivels left to right. Available in ebony fleck or maple.

KBS-1B Ebony 20″ keyboard shelf for CRND only.....**99.95**

KBS-1M Maple 20″ keyboard shelf for CRND only.....**99.95**

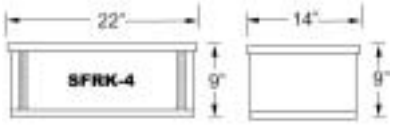
KBS-2B Ebony 28″ keyboard shelf for ACD and SCD only.....**104.95**

KBS-2M Maple 28″ keyboard shelf for ACD and SCD only-maple.....**104.95**

SFRK-4B

4U rack for use in place of RB-22 allowing quick access to most commonly accessed euipment.





SFD

Studio Furniture Door

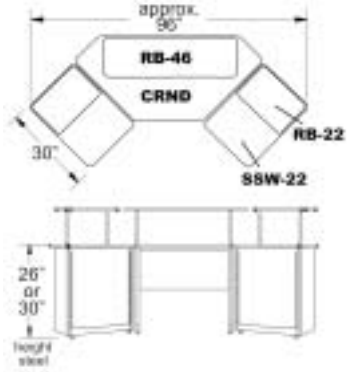
Steel framed Plexi door for SSW and ASW Side Wings (wood sides only)- silver hammertone.



CONFIG-U-RAXX

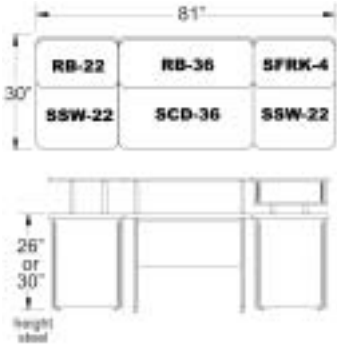
SUGGESTED CONFIGURATIONS





1 - **CRND-B**
corner desk
2 - **SSW-22B**
side wing racks
1 - **RB-46B**
rack bridge
2 - **RB-22B**
rack bridges

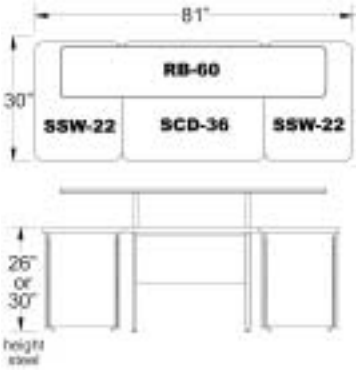




1 - **SCD-36B**
2 - **SSW-22/12BS**
side wing racks
1 - **RB-36B**
rack bridge
1 - **RB-22B**
rack bridge
1 - **SFRK-4B**
1 - **CBS** center bridge
support

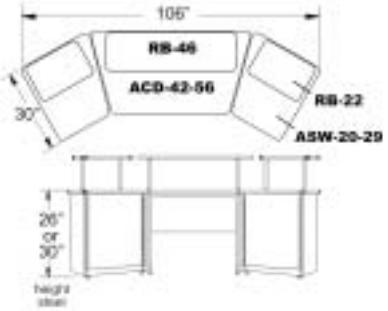


1 - **SCD-36M**
center desk
2 - **SSW-22M**
side wing racks
1 - **RB-60M**
rack bridge



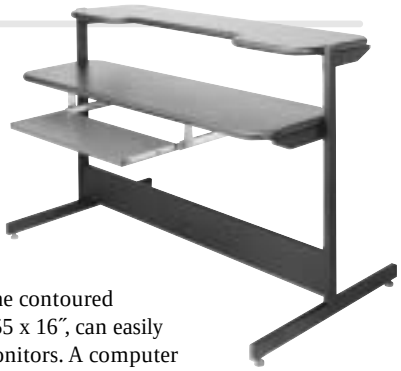


1 - **ACD-42-56M**
enter desk
2 - **ASW-20-29M**
side wing racks
1 - **RB-46M**
rack bridge
2 - **RB-22M**
ack bridges
1 - **KBS-2M**



STUDIO FURNITURE

ETUDE



Perfect as a studio keyboard workstation or learning center, the Etude's side supports are constructed of 1″ by 2″ 16-gauge tubular steel . The fabricated bottom cross member is 1½″ by 6″. The generous 55″ wide and 16″ deep keyboard shelf supports an 88-note keyboard. The contoured upper shelf, which also measures 55 x 16″, can easily accommodate audio and video monitors. A computer keyboard shelf can be mounted to either of the two standard shelves using the hardware provided into pre-drilled holes. Both the keyboard shelf and the optional printer/CPU shelf (not shown) that mounts to the lower portion of the frame are adjustable in 1″ increments up to 5″ **419.95**

Etude Printer/CPU Shelf *(ETUDEPS)* **99.95**

RMPC-4ATX Computer Chassis

- 4 space chassis fits any ATX style motherboard
- 20.3″ deep with 2″ removable handles
- Supplied with a 300w ATX power supply

7 Expansion Slots with Card Braces:

- Includes 2 internal and 1 external 3.5″ drive bays, plus 3 external 5.25″ bays



RMPC-4ATX open



RMPC-4ATX closed

Front Panel Features:

- Switches for power, reset and alarm
- Power and drive activity LED indicators
- Alarms for fan and temperature
- Hinged locking plate for security
- AT style "plug and play" keyboard connector
- 4.5″ filtered intake fan keeps components cool and dust-free.

RMPC-4ATX **$199.95**

ISORAXX



Recent developments in recording technology allow today's audio professionals and enthusiasts unprecedented control over the quality of their work. But along with the benefits provided by these great new products comes additional noise created by fans and hard disk drives. The IsoRaxx is the solution to mechanical noise problems in your studio.

- Keeps equipment safe and quiet — reduces fan and hard drive noise by an average of 21dB
- Electrostatic filter intake keeps dust out
- Equipment is easily visible through the clear lexan panel in the front door
- Separate passages for signal, AC and computer cables
- Dual variable-speed fans
- Key-lockable rear door with removeable hinges ease equipment installation and maintenance
- Constructed of 3/4″ thermally fused melamine
- Comes assembled; they are available in ebony fleck or maple finishes; casters included

	ISR-14-24	ISR-14-30	ISR-20-30
Rack Spaces	14	14	20
Overall Height	31.75″	31.75″	42.5″
Overall Width	25″	25″	25″
Desktop Depth	25.5″	32″	32″
Upper Rack Depth	15.75″	26.25″	26.25″
Middle Rack Depth	17.75″		
Lower Rack Depth	21″	22″	22″
Total Rack Depth w/door	23.625″	30″	30″
Price	$909.95	$979.95	$1119.95

OPTIONAL ACCESSORIES

SFD Studio Furniture Rack Door – For the Lonestar, Golden Gate, Big Apple and Big Easy. Features matching ebony fleck melamine with solid oak trim, a clear lexan door and flush mount hinge system. Keylock included. Can be set for left or right swing and will retrofit to existing furniture systems.....**119.95**



SFD

ERK-4 Four-space elite rack, sits flush on all shelves **104.95**

CBS Center bridge support for heavier loads **39.95**

KBS-1 28″ Keyboard & Mouse Shelf For all workstations. Finished in matching ebony fleck while the front edge is trimmed in solid oak **99.95**

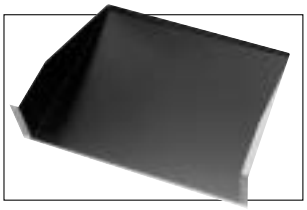


ERK-4

Studio Furniture Casters Fits all workstations except the Daytona **31.95**

STORAGE/SHELVES

UTS Utility Shelves

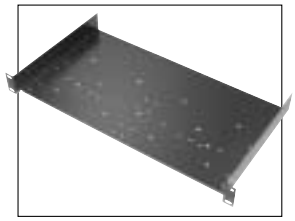


16-gauge, cold rolled steel shelves that allow the rackmounting of otherwise non-rackmountable equipment. One-piece construction. Finished in durable black baked enamel.

UTS-1 11″ x 1.75″ (DxW). 1 space **27.95**

UTS-2 14.50″ x 3.50″ (DxW). 2 space **35.95**

UTS-3 14.50″ x 5.25″ (DxW). 3 space **39.95**



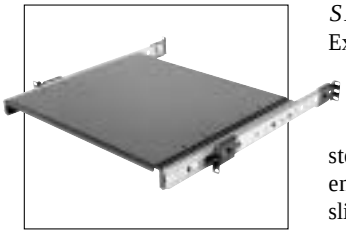
UNS Universal Rack Shelf

One space, 14-gauge shelf has a hole-pattern in the bottom for mounting a variety of half-rack equipment. Comes with two each 6/32, 8/32, and 10/32 machine screws. 10 x 1.75″ (DxW). Finished in durable black baked enamel.....**18.95**

SLE Economy Sliding Shelf



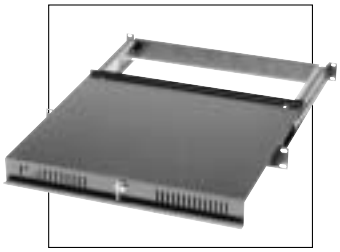
Supports equipment up to 35 lbs. Ears attached to the rear frame allow the unit to be secured to the back of any enclosed rack. 12″ sliders with nylon roller bearings allow a smooth sliding movement. Sliders provide positive lockout, for the black laminated wood shelf **51.95**



SLS Sliding Shelf

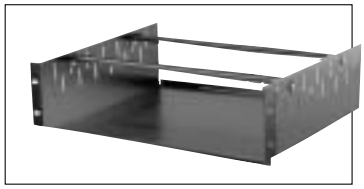
Extra heavy duty shelf occupies just one rack space. Supports up to 100 lbs. with a safety factor of 100% if properly secured. The 14 gauge steel shelf is finished in black baked enamel. Extra heavy-duty ball bearing sliders extend a full 14″ for a usable surface area of 16 x 14″ (WxD), and are equipped with a mechanism that locks the shelf in the out position. Rear rack rail provided.....**125.95**

SLS 1.5 Sliding Shelf



Steel shelf measuring 17.50 x 14.5″ (WxD) locks both in and out. Extending on 14″, full extension, ball bearing sliders and occupying one and a half rack spaces. Supported by an all steel frame and extension ears for rear support.....**97.95**

TR Trap Shelf

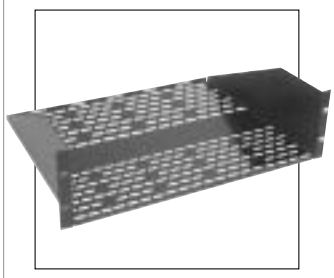


Constructed of 16-gauge cold, rolled steel, it is finished in durable black baked enamel. Comes complete with 2 fully adjustable hold down bars that "trap" equipment. Bars are padded to protect the finish on your gear. Returns on the front and rear edges of shelf provide additional lateral strength.

TR-2 14.25 x 17.25 x 3.50″ (DxWxH). 2 space trap **38.95**

TR-3 14.25 x 17.25 x 5.25″ (DxWxH). 3 space trap **44.95**

TR-4 14.25 x 17.25 x 7″ (DxWxH). 4 space trap **48.95**



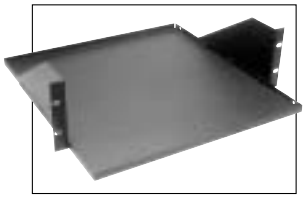
UTVS-3 Vented Shelf

A 16-gauge steel shelf that is finished in black powder coat, handles up to 60 lbs. of equipment and occupies 3 rack spaces. Slotted bottom allows for ventilation and the slotted rear return allows for cable management.

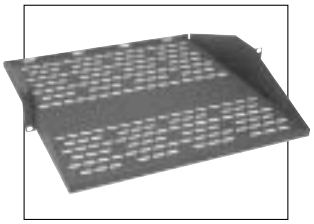
UTVS-3-14 3 space, 14″ deep **38.95**

UTVS-3-18 3 space, 18″ deep **42.95**

ADS-3 Adjustable Shelf



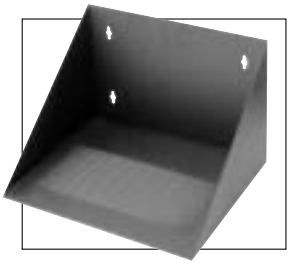
This three space, 16-gauge steel shelf has three position adjustable ears that allows you to move equipment from flush with rack rail to 8″ forward of rail. Its dimensions are 19 x 17 x 5.25″ (WxDxH) **44.95**



VCS-2 Cantilvered Shelf

Occupying two rack spaces, this 16-gauge steel shelf allows you to bring equipment well forward of the rack face. Vented for air circulation, this shelf is 18″ deep and will support up to 50 lbs. It features built-in cable management and is finished in black powder coat **45.95**

WMS Wall-Mount Shelf



Provides a secure, yet inexpensive way to mount non-rack mountable equipment to a wall. With a vented bottom for air circulation and conduit knock-outs for cable passage, this 16-gauge steel, fully welded unit has a 200 pound capacity. Finished in rugged black powder coat, it can also be used to house the ER or ERK series of racks.....**64.95**

B&H PHOTO - VIDEO - PRO AUDIO

STORAGE/SHELVES

ESD - Economy Sliding Drawer
16-gauge steel, fully enclosed construction with a black semi-gloss finish. The 12″ cabinet grade slides allow for full extension. The rackmount drawers feature a laser cut knockout on the front panel for adding an optional key lock, as well as a laser cut knockout in the rear for cable passage if needed.
ESD-2 2 space.....**79.95** **ESD-3** 3 space.....**81.95**
ESD-4 4 space.....**89.95**

SDR - Sliding Rack Drawer
Heavy duty sliding rack drawer available in 2, 3 or 4 spaces. Cabinet grade 14″ full extension sliders provide smooth action. Features include a large flush mount latch, a 13 gauge drawer front and black textured enamel finish. A laser cut knockout on the front panel allows you to add an optional lock, while the laser cut knockout in the rear allows cable passage should you want to store an electronic device in the drawer.
SDR-2 2 space.....**90.95** **SDR-3** 3 space.....**100.95**
SDR-4 4 space.....**111.95**

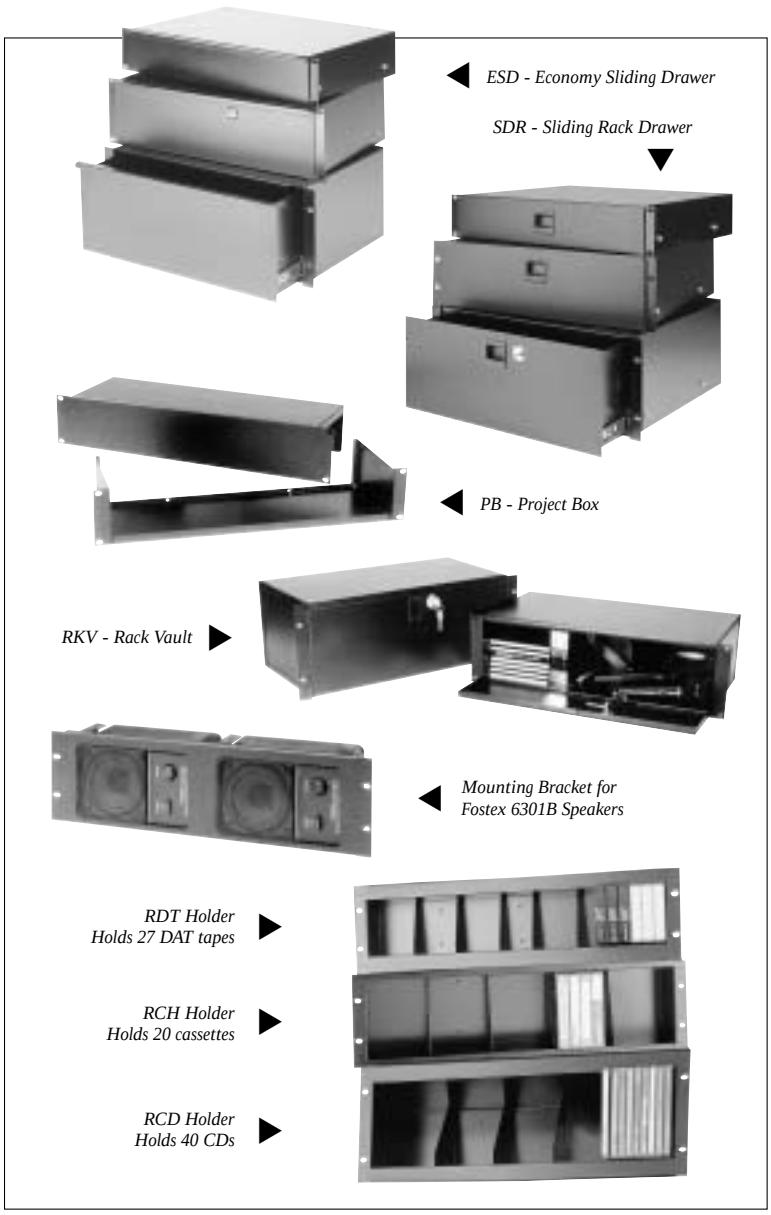
Lock for ESD and SDR Drawers
The ESD and SDR feature laser cut knockouts on the front panel for adding a key lock at any time. *(LOCK)*.....**9.95**

PB - Project Box
These project boxes are of two piece, cold rolled steel construction. Perfect for the do-it-yourselfer. Black baked enamel finish. Available in one or two space configuration. Front to back measures 6″.
PB-1 1 space.....**29.95** **PB-2** 2 space.....**32.95**

RKV - Rack Vault
Secured storage of otherwise loose equipment, tools, tapes, CD's, papers and other personal items; all steel construction; flush mount latch and keyed lock. Perfect for any audio rack, in the studio, on the road or at the installation; black baked enamel finish.
RKV-3 3 spaces. 17 x 9 x 5″ (LWH).....**51.95**
RKV-4 4 spaces. 17 x 9 x 7″ (LWH).....**58.95**

Mounting Bracket for Fostex 6301B Speakers
A kit for rackmounting a pair of the Fostex 6301B speakers. Finished in black powder coat.....**83.95**

Media Holders
Convenient way to store compact discs, DAT tapes, or audio cassettes in you rack.
RDT 3 space. Holds 27 DAT tapes.....**49.95**
RCH 3 space. Holds 20 cassettes.....**51.95**
RCD 4 space. Holds 40 CDs.....**58.95**

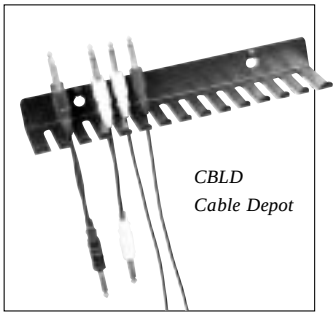


Custom Mounting Kits for Consumer A/V Components



An easy and attractive solution for rack mounting non-rackmountable components. Perfect for custom home installations, office suites or any other location where aesthetics are as important as sound. These kits will give your installation the "custom" look that clients love.
Each kit includes the aluminum faceplate with anodized or textured paint finish, cold rolled steel rack shelf, and rear braces that prevent shifting. Optional clamping bars are available when maximum stability is required.
Call today for more information.

HARDWARE AND ACCESSORIES


CBLD
Cable Depot

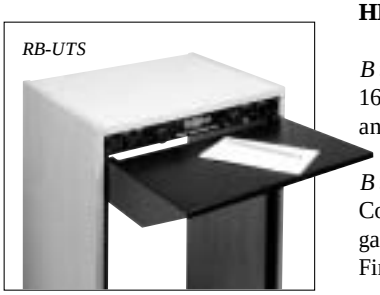
Cable Depot (CBLD)
16-gauge steel cable reorganizer keeps cabling off the floor, out of the way and kink-free. Fourteen fingers extend to hold up to forty cables. Maximum cable diameter is .25″.....**11.95**

Headphone Hangers
A convenient way to hang headphones. The HH-1 comes straight out of the rack while the HH-2 projects out at a right angle, taking the place of any rack screw already in your rack. Simply screw it in until it is in the desired position and tighten the nut.

HH-1 Straight headphone hanger.....**6.95**

HH-2 Right angle headphone hanger.....**6.95**

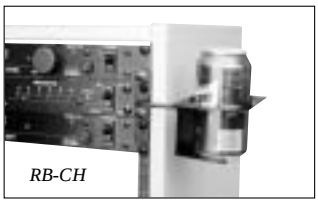

HH-2


RB-UTS

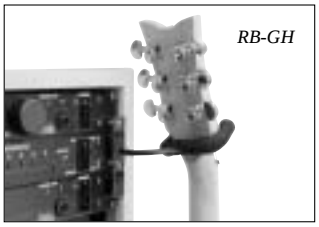
Button Mount Utility Shelf (RB-UTS)
16 gauge cold rolled steel utility shelf that mounts easily to rack buttons and removes just as easily. Flat black powder coat finish.....**28.95**


RB-TC

Button Mount Tool Rack (RB-TC)
Convenient way to keep tools at hand when working on rack or gear. 16 gauge cold rolled steel tool rack mounts conveniently to buttons. Finished in black texture powder coat finish.....**27.95**


RB-CH

Button Mount Cup Holder (RB-CH)
Keep your drink close at hand. 16 gauge cold rolled steel with black textured finish.....**19.95**


RB-GH

Button Mount Guitar Holder (RB-GH)
Button mount guitar holder keeps your guitar where you need it when not being played. Padded to eliminate marring the finish. Removes easily and stores in small spaces.....**17.95**


AR-5

Console Lamp with BNC Connector (CL-1H)
A high-powered 12 volt, hooded mixing console lamp that illuminates the mixer and enhances isolation in low light environments. 12″ sturdy goose neck, BNC connector, and black matte finish.....**17.95**

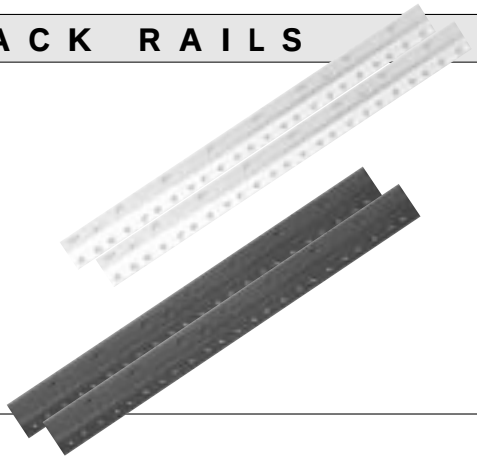
Accessory Riser (AR-5)
A handy little device that mounts under an amplifier handle or on a mic stand. It can act as a third hand or save valuable rack space. Perfect for tuners or other small devices. 3 x 5 x 5″ (HxDxW).....**11.95**


CL-1H

RACK RAILS

RKRL - Rack Rail
11-guage steel full hole rack rail; punched and tapped for 10-32″ screws. Side mounting holes punched every space. Available in all sizes, 2 to 45 space, black baked enamel. Sold in pairs.

GKRL - Galvanized Rack Rail
.075″ galvanized steel rack rail; punched and tapped for 10-32″ screws; full hole rack rail has holes tapped at 5/8″-5/8″-1/2″. Available in any length up to 45 space. Sold in pairs.



RKRL
Rack Rail
$1.35 per space

GKRL
Galvanized Rack Rail
.90 per space

Available up to 45 spaces

HARDWARE AND ACCESSORIES

Pin Torx Security Screws
10-32 x .75″ black oxide pin torx pan head. When security is vital.
PNTX-50 50 per bag.....**12.95**
PNTX-100 100 per bag.....**19.95**

Pin Torx Bit (PNTX-B)
Matching bit for pin torx screws.....**7.95**

Pan Head/Captive (CPSW)
10-32 x .75″ philips pan head screws integrated with nylon cup washers save the user the trouble of fumbling with separate parts allowing for faster installation of equipment.
CPSW100 With washers. 100 quantity. .**14.95**

CPSW500

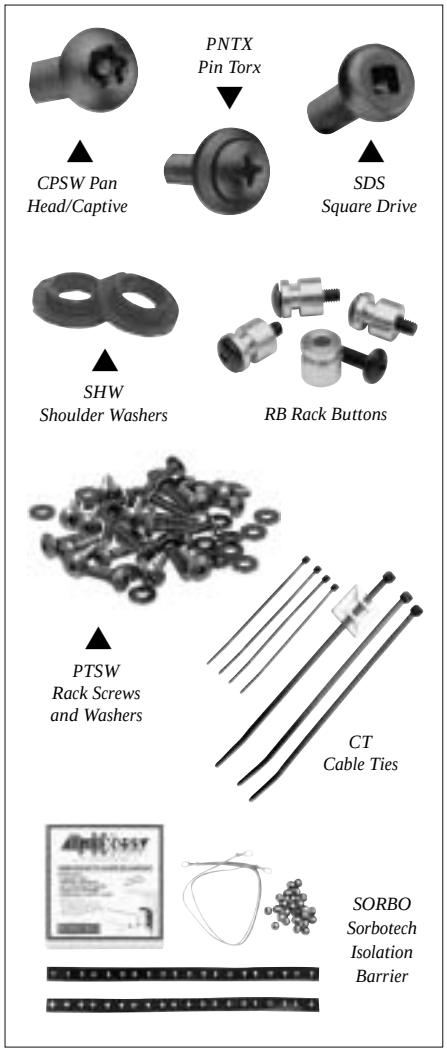


CPSW500 With washers. 500 quantity. Includes fanny pack.....**69.95**

Square Drive Screws (SDS)
When security is a concern. Pan head 10-32 x .75″ long. 100 per bag with washers ...**19.95**

Square Drive Bit (SDB)
Accomodates security rack screws.....**2.00**

Shoulder Washers (SHW)
#10 washers sold in bags of 100. Used to isolate equipment from rack rail to eliminate ground loop problems **6.35**



Rack Buttons (RB)
Machined aluminum cylinders with center through hole to accept 10-32 rack screw. Buttons can be mounted to rack rail without equipment interference and can hold a variety of accessories. Sold in pairs.
1/2″ long **6.35**

Rack Screws and Washers (PTSW)
10-32 x .75″ black oxide, phillips truss head screws with #10 nylon washers to prevent scratching equipment. Available in packages of 25, 50 and 100.



PTSW25 25 peices **4.75**
PTSW50 50 peices.....**7.95**
PTSW100 100 peices **14.95**

CT Cable Ties
100 pieces per bag. Black color.
CT-100-4 4″ cable ties.....**7.95**
CT-100-8 8″ cable ties.....**9.69**
CT-100-8R 8″ reusable cable ties **13.49**
CT-100-12 12″ cable ties.....**14.25**
CTM Cable tie mount; 1/5 x 1/5″ **11.95**

Sorbotech Isolation Barrier (SORBO)
Vibro acoustic damping barrier for six rack spaces of gear, a proven shock absorber and electrical isolator. Kit includes rack rail barrier, shoulder washers and copper braided grounding strip.....**Call**

Rack Mount Power Distribution

The PD-800 is an attractive, high-quality and economical rack mount power distribution solution. The PD-800L features pullout lights with an on/off switch and dimmer control. Both the PD-800L and the PD-800 feature a heavy-duty 12′ AC cable and 9 unevenly placed outlets, so you don't have to give up an extra outlet each time you plug in a wall-wart.

- Long (12′), 14-gauge AC cord
- Nine unevenly spaced outlets conveniently accept wall warts
- Front mounted circuit breaker
- Spike and surge protection
- 15 amp rating ◆ ETL approved
- RFI and EMI surge protection on both sides of the power line
- 14-gauge cold rolled steel faceplate with durable powder coat finish



PD-800 Rack Mount Power Distribution.....**51.95**

PD-800L Rack Mount Power Distribution with pullout lights, on/off switch and dimmer control **84.95**

POWER DISTRIBUTION

SC-6 - Low-Voltage Sequencing Controller



Allows remote sequencing of up to six AC circuits. Using standard CAT-5 cable or terminal strips (12 or 24V), the user can send relay control messages to our PM series power modules. Jumper switches inside each module allow you to assign a circuit value to each module. A trim-pot on the SC-6 allows you to adjust the power on/power off delay time. Four SC-6 units can work in parallel or series. This will allow you to daisy chain multiple sets of sequenced circuits for a cascading array of delayed power-on circuits.
The PM-1 modules are available in both Modular Power Strip and stand-alone versions with your choice of CAT-5 or terminal strip connectors.
A wall-mounted remote on/off switch (SC-6RS) is available for remote activation of the SC-6.

- UL listed power supply
- Connects to other Raxxess power sequencing products CAT-5 cable
- Up to four units may be connected, allowing up to 24 sequence steps
- 24VDC terminal strips provide to interface with existing power products
- Dry-contact closure terminals for remote power sequencing

SC-6 Low Voltage Sequesncing Controller.....**349.95**

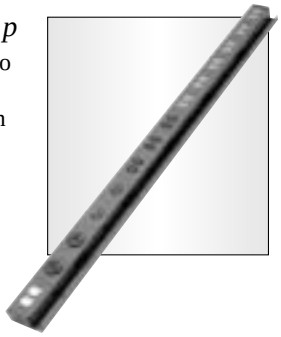
SC-6RS Wall switch for SC-6/SPD-6 remote control.....**45.95**

SC-6RSK Key switch for SC-6/SPD-6 remote control.....**51.95**

STAND-ALONE, SURFACE MOUNT REMOTE POWER MODULES

Model	CAT-5	Terminal Strip		Price
PM-1SRJ-DR15	✓		DR-15	**147.95**
PM-1SRJ-DR20	✓		DR-20	**153.95**
PM-1SRJ-DRIG	✓		Isolated Ground	**181.95**
PM-1SRJ-DRHG	✓		Hospital Grade	**167.95**
PM-1SRJ-L5	✓		L5-30	**173.95**
PM-1SRJ-TT	✓		TT-30	**159.95**
PM-1STS12-DR15		12v	DR-15	**147.95**
PM-1STS12-DR20		12v	DR-20	**153.95**
PM-1STS12-DRIG		12v	Isolated Ground	**181.95**
PM-1STS12-DRHG		12v	Hospital Grade	**167.95**
PM-1STS12-L5		12v	L5-30	**173.95**
PM-1STS12-TT		12v	TT-30	**159.95**
PM-1STS24-DR15		24v	DR-15	**148.95**
PM-1STS24-DR20		24v	DR-20	**153.95**
PM-1STS24-DRIG		24v	Isolated Ground	**181.95**
PM-1STS24-DRHG		24v	Hospital Grade	**167.95**
PM-1STS24-L5		24v	L5-30	**173.95**
PM-1STS24-TT		24v	TT-30	**159.95**

MPS - Modular Power Strip



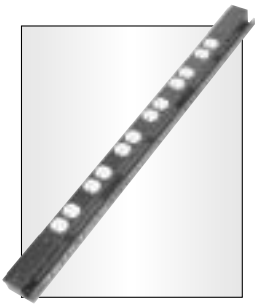
Available in just about any configuration to meet any power distribution/ sequencing needs. The MPS Modular Power Strips can be ordered in an almost infinite variety of configurations; including current levels, receptacle type, remote sequencing modules, color-coded pig tails, etc.

- 32.5″ (7 devices) or 64″ (14 devices)
- 1 to 6 individual circuits
- Mounting plates for 1, 2, 3, 4 or 6 devices
- Choice of receptacles - Edison 15 or 20 amps, TT-30, L5-30, Isolated Ground, IEC 320, Hospital Grade
- 20 and 30 amp sequencing power modules available for MPS with TT-30, L5-30 and duplex receptacles.

MPS-32 32″ Holds 7 receptacles.....**Call**

MPS-64 64″ Holds 14 receptacles.....**Call**

LPPS - Low Profile Power Strip



Available in three sizes with 6, 9, or 12 Edison duplex receptacles in your choice of 15 or 20 amp versions. Up to 3 independent circuits.

- Three lengths available - 34″ (6 duplex Edison), 46.5″ (9 Duplex Edison) or 59″ (12 duplex Edison)
- 1 circuit standard - 2 or 3 optional
- Available in 15 or 20 amp versions

Model	Duplex	Amp	Circuit	Termination	Price
LPPS6-15-1	6	15	1	Cord	**90.95**
LPPS6-15-2	6	15	2	Pigtails	**99.95**
LPPS6-15-3	6	15	3	Pigtails	**104.95**
LPPS6-20-1	6	20	1	Cord	**111.95**
LPPS6-20-2	6	20	2	Pigtails	**118.95**
LPPS6-20-3	6	20	3	Pigtails	**125.95**
LPPS9-15-1	9	15	1	Cord	**111.95**
LPPS9-15-2	9	15	2	Pigtails	**118.95**
LPPS9-15-3	9	15	3	Pigtails	**125.95**
LPPS9-20-1	9	20	1	Cord	**139.95**
LPPS9-20-2	9	20	2	Pigtails	**146.95**
LPPS9-20-3	9	20	3	Pigtails	**153.95**
LPPS12-15-1	12	15	1	Cord	**132.95**
LPPS12-15-2	12	15	2	Pigtails	**139.95**
LPPS12-15-3	12	15	3	Pigtails	**149.95**
LPPS12-20-1	12	20	1	Cord	**159.95**
LPPS12-20-2	12	20	2	Pigtails	**167.95**
LPPS12-20-3	12	20	3	Pigtails	**174.95**

CONVERTA RACKS

Converta Racks

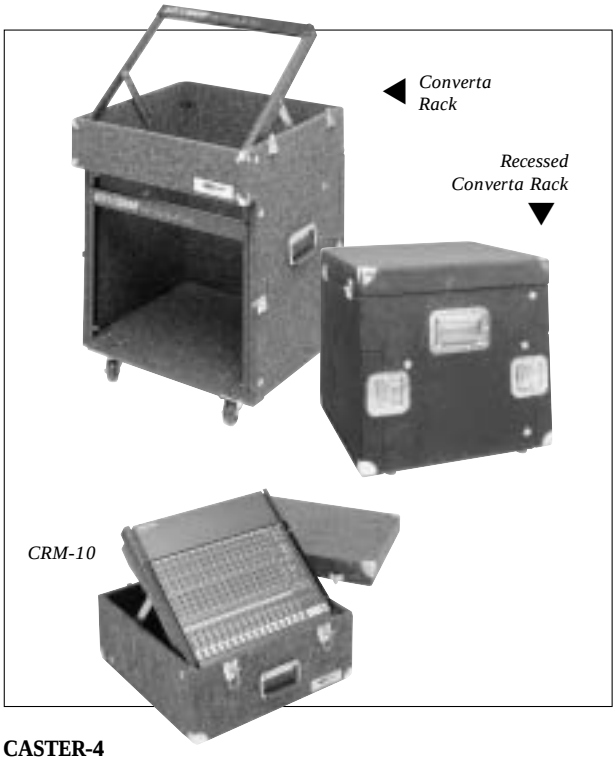
Maximizes rack storage in a minimum of vertical space. Ideal for the DJ or musician who wants to get on and off the job as fast as he can.

- Removable top cover exposes 10 or 12 rack spaces that can be tilted to any of 8 positions from zero to 45° ◆ Available with 6 or 10 vertical rack spaces
- Removable rear cover provides easy access for connecting cables
- Rugged plywood construction ◆ Internally braced
- Heavy-duty butterfly latches, releasable hinges and nickel-plated corners
- Rugged, gray carpeted exterior

Also Available — Recessed Converta Racks

Includes recessed spring-loaded handles and latches, black carpet covering and heavy-duty metal corners.

	CRM-10	CR-10/6	CR-10/10	CR-12/10
Lower Rack Spaces	N/A	6	10	10
Top Rack Spaces	10	10	10	12
Overall Width	20.75″	20.625″	20.625″	20.625″
Lower Rack Depth	N/A	16.125″	16.125″	19.25″
Height of space between top mixer and lower rack	7.25″	5″	5″	5″
Depth of top lid	2.5″	2.5″	2.5″	2.5″
Overall Depth	20.75″	20.125″	20.125″	23.25″
Overall Height	9.75″	18.75″	26″	26″
Price	181.95	230.95	258.95	293.95
Recessed Hardware Version Price	—	CR-10/6R 293.95	CR-10/10R 328.95	CR-12/10R 363.95



CASTER-4
Set of 4 casters for the CR-10/10 and CR-12/10.....**38.95**

CASTER-4LCK
Set of 4 casters with brakes for the CR-10/10 and CR-12/10 ..**45.95**



Elite Converta Racks

The functionality of Converta Racks in a semi-portable furniture style cabinet. Perfect for houses of worship, schools, corporate meeting rooms or any venue where A/V equipment must suit the decor.

- 10 or 16 vertical rack spaces in the lower section
- 10 rack spaces on top anchored by a ratchet system that can be tilted to any of 8 positions from zero to 45° ◆ Constructed of 3/4″ fused melamine
- Locking, smoked lexan front door provides security and visibility
- Heavy-duty key-lockable panel covers the mixer section, retracts on sturdy cabinet sliders and lays flat against the rear of the rack
- Locking rear door with breakaway hinges allows easy equipment installation and maintenance ◆ Includes furniture-grade hooded casters
- Passages in the bottom of the rack allow for easy cable management
- All hardware is brass plated to compliment the cherry finish
- Ships assembled via common carrier only



	Vertical Rack Spaces	Overall Height	Outside Depth	Outside Width	Inside Depth from Rack Rails	Weight	Price
ECR-10/10 CH	10	31″	24″	20.75″	21	100 lbs.	384.95
ECR-10/16 CH	16	41″	24″	20.75″	21	120 lbs.	419.95

COMMERCIAL PRODUCTS

Gangable Audio Rack (GAR)

- Fully welded gangable rack, constructed of 12-gauge (top and bottom) and 14-gauge (side posts) cold rolled steel
- Available in 2 depths, for equipment 22.50″ and 27.50″ deep
- Adjustable 11-gauge steel rack rails, in both front and rear positions
- Punched rack rail supports with multiple cable tie offs
- Vertical cable passageways
- 3 sizes of conduit knockouts in three locations, including the solid roof insert
- Slotted vents at the front frame top
- Durable black powder coat
- Optional locking side panels easily mount via tabs that fit into slots in the frame
- Optional front and rear locking doors for left or right swing, plus perforated solid and fan roof inserts are available.



Model	Panel Space	Depth	Width	Height	Height w/Casters	Price
GAR-35-26	35 space (61¼″)	26″	24″	67.75″	69″	609.95
GAR-35-32	35 space (61¼″)	32″	24″	67.75″	69″	629.95
GAR-40-26	40 space (70″)	26″	24″	76.25″	77.5″	643.95
GAR-40-32	40 space (70″)	32″	24″	76.25″	77.5″	664.95
GAR-44-26	44 space (77″)	26″	24″	83.25″	84.5″	108.95
GAR-44-32	44 space (77″)	32″	24″	83.25″	84.5″	119.95

Side Panels

16-gauge steel side panels, finished in textured powder coat. Available for the GAR and SKR series racks. Elite side panels of ebony fleck laminated hardboard with solid oak edges are also available for the SKR series.



For the SKR Series

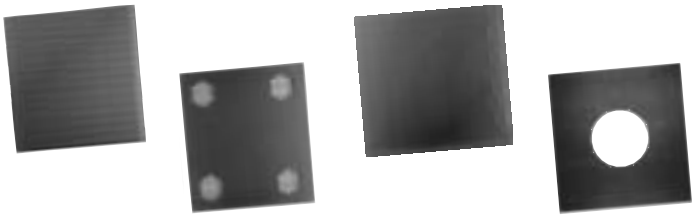
SKRSP-12	powdered coat	12 spaces	35.95
SKRSP-24	powdered coat	24 spaces	58.95
SKRSP-42	powdered coat	24 spaces	94.95
SKRESP-12	ebony fleck	12 spaces	46.95
SKRESP-24	ebony fleck	24 spaces	78.95
SKRESP-42	ebony fleck	42 spaces	118.95

For the GAR Series

GARSP-35-26	powdered coat	35 spaces	124.95
GARSP-35-32	powdered coat	35 spaces	149.95
GARSP-40-26	powdered coat	40 spaces	129.95
GARSP-40-32	powdered coat	40 spaces	164.95
GARSP-44-26	powdered coat	44 spaces	144.95
GARSP-44-32	powdered coat	44 spaces	174.95

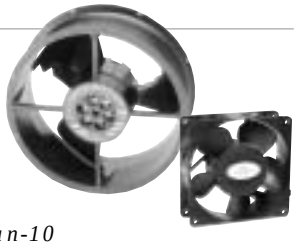
Roof Inserts for GAR Racks

Optional roof inserts for various service applications. All are of 16-gauge cold rolled steel and are finished in a durable black powder coat finish.



GART-10F Houses one 10″ fan **51.95**
GART-4.5F Mounts four 4½″ fans **51.95**
GART-P 66% open area perf pattern **51.95**
GART-S Solid roof insert **51.95**

Cool off the gear in your GAR series racks with these ultra quiet, easy-to-mount 115 volt fan kits. They include unterminated attached AC power cables and mounting hardware.

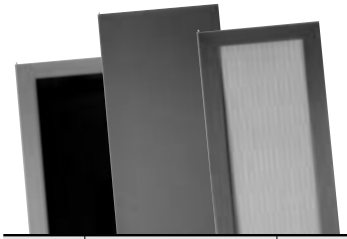


Fan-4

- Mounts easily to FTP roof insert as well as FPL 1, FPL2 and FPL3 rack panels
- Moves 95 cubic feet of air per minute **32.95**

Fan-10

- Mounts easily to FTP-10 roof insert
- 10″ ball-bearing fan 550 cubic feet of air per minute
- Free air rating of 49dBA **146.95**



Enclosure Doors

Available for KAR, GAR, and SWR racks; in solid 16-gauge steel, perforated (66% open area) or in smoked plexi. Keyed lock installed. Finished in a black powder coat.

For Door	Perforated Doors	Price	Plexi Doors	Price	Solid Doors	Price
GAR-35	GARD-35P	139.95	GARD-35PLX	195.95	GARD-35	369.95
GAR-40	GARD-40P	209.95	GARD-40PLX	398.95	GARD-40	146.95
GAR-44	GARD-44P	224.95	GARD-44PLX	424.95	GARD-44	154.95
KAR-18	KARD-18P	139.95	KARD-18PLX	204.95	KARD-18	92.95
KAR-24	KARD-24P	159.95	KARD-24PLX	249.95	KARD-24	109.95
KAR-35	KARD-35P	179.95	KARD-35PLX	359.95	KARD-35	118.95
KAR-40	KARD-40P	209.95	KARD-40PLX	426.95	KARD-40	124.95
KAR-44	KARD-44P	209.95	KARD-44PLX	426.95	KARD-44	132.95
SWR-10	SWRD-10P	118.95	SWRD-10PLX	174.95	SWRD-10	64.95
SWR-18	SWRD-18P	139.95	SWRD-18PLX	204.95	SWRD-18	94.95
SWR-24	SWRD-24P	154.95	SWRD-24PLX	239.95	SWRD-24	99.95
SWR-35	SWRD-35P	179.95	SWRD-35PLX	364.95	SWRD-35	124.95

COMMERCIAL PRODUCTS

Enclosed Audio Rack (KAR)



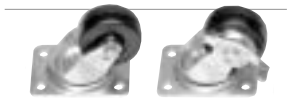
- Welded, 16-gauge cold rolled steel construction with fully adjustable front rack rail
- Front adjustable 11-gauge rack rail
- 16-gauge steel rack rail supports
- Available in 2 depths, for equipment up to 20″ and 25″
- Variety of conduit knockouts are located at the top and bottom rear
- Punched rack rail supports with multiple cable tie offs
- Perforated sides provide ventilation
- Optional front and rear locking doors are easily removable for left or right swing
- A removable 3 space solid roof insert allows for interchangeable panels for fans or passive cooling
- Black powder coat finish

Model	Panel Space	Depth	Width	Height	Height w/Casters	Price
KAR-18-22	18 space (32″)	22″	21⅞″	36″	37½″	294.95
KAR-18-28	18 space (32″)	22″	21⅞″	36″	37½″	308.95
KAR-24-22	24 space (42″)	22″	21⅞″	46½″	48″	314.95
KAR-24-28	24 space (42″)	28″	21⅞″	46½″	48″	335.95
KAR-35-22	35 space (61¼″)	22″	21⅞″	65¾″	67¼″	342.95
KAR-35-28	35 space (61¼″)	28″	21⅞″	65¾″	67¼″	378.95
KAR-40-20	40 space (70″)	20″	21⅞″	74¼″	76″	378.95
KAR-44-22	44 space (77″)	22″	21⅞″	81½″	83″	389.95
KAR-44-28	44 space (77″)	28″	21⅞″	81½″	83″	419.95



Set of 4 Levelers (FL)

Fit all GAR and KAR racks. Adapter plate included....**25.95**



Set of 4 Casters

Hard rubber, 2.5″, ball bearing swivel caster; 175 lb. capacity.

CASTER-CM **45.95**

CASTER-CMLCK
Locking caster **51.95**

Tie Bars

19″ rack mountable 1/4″ diameter rods. Efficient horizontal cable management.
TBO offset.....**4.50**
TBS straight.....**3.95**

77″ Lacer Strip



11-gauge steel; mounts vertically to rack rail supports for cable management. Can be cut to any length.
LCS offset.....**14.95**

Sectional Wall Rack (SWR)



- Fully welded 12 and 14-gauge construction, internal bracing, reinforced hinges and integral center section supports for maximum strength
- Adjustable 11-gauge steel rack rail, ventilated sides and top and bottom conduit knockouts
- Keyed locks between the center and rear sections
- Center section and optional door can swing left or right
- 6″ square knock-out for cable passage
- Tie off strips for cable management
- Attractively finished in durable black powder coat
- Optional locking rod and locking door available
- Hinging system eases installation and provides greater security

Model	Panel Space	Center Depth	Rear Depth	Height	Price
SWR-10-12	10 space (17½″)	12″	4½″	21¾″	272.95
SWR-10-17	10 space (17½″)	17″	4½″	21¾″	286.95
SWR-18-12	18 space (31½″)	12″	4½″	35¾″	370.95
SWR-18-17	18 space (31½″)	17″	4½″	35¾″	412.95
SWR-24-12	24 space (42″)	12″	4½″	46¼″	440.95
SWR-24-17	24 space (42″)	17″	4½″	46¼″	510.95
SWR-35-17	35 space (61¼″)	17″	4½″	65½″	636.95

Wall Mount Rack (WMK)

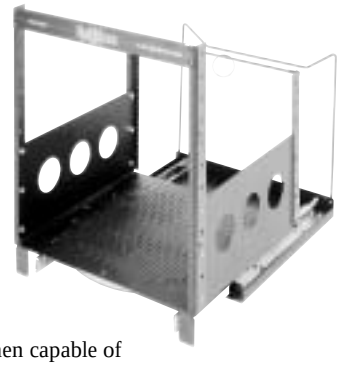


- Compact audio enclosures for smaller installations
- 16-gauge steel, finished in durable black powder coat
- 11-gauge rack rails
- Top and bottom conduit knockouts and slotted sides for passive cooling
- Key hole slots on rear panel allow for secure attachment to wall
- WMK-10 utilizes a rear mounting pan with a laser knockout for cable passage
- Mounting pan facilitates servicing equipment and uses a locking rod for real rack security
- Front locking door can swing from left to right and is 1.5″ deep

Model	Panel Space	Width	Depth	Height	Price
WMK-7	7 space (12¼″)	22¼″	17″	15½″	153.95
WMK-10	10 space (17½″)	22¼″	17¼″	20¼″	244.95

COMMERCIAL PRODUCTS

Rotating Rack System *(ROTR)*



Perfect for installing in existing cabinetry or new millwork, the Raxxess Swivel Rack system comes in sizes from eight to twenty four rack spaces. Available in a two or four slide formats capable of supporting 150 and 300 pounds respectively. The rack extends a full 20″ from its frame and is then capable of rotating 90° in either direction, allowing unobstructed access to components regardless of their depth. The system locks in the closed position with security panel and screws provided. Easy and effective cable management system and eight-space rear rack rail are included.



ROTR-2
2 sliders, specify height.....**335.95**

ROTR-4
4 sliders, specify height.....**384.95**

Skeleton Rack *(SKR)*



- A "Bare Bones" approach to racking audio equipment
- Constructed of four pieces of 11-gauge steel rack rail (cut to any length up to 42 spaces 73.50″) and a top and bottom pan of 16-gauge cold rolled steel
- Surprisingly strong, yet assembles in less than five minutes with a standard wrench
- Optional side panels are available (12, 24 and 42 spaces) in either 16-gauge steel with a black powder coat finish or Elite panels in ebony fleck or mahogany laminate with solid oak edging

Model	Panel Space	Depth	Width	Height	Height w/Casters	Price
SKR-12	12 space (21″)	20″	19¼″	26⅜″	28⅜″	97.95
SKR-24	24 space (42″)	20″	19¼″	47¼″	49¼″	125.95
SKR-42	42 space (73½″)	20″	19¼″	78¾″	80¾″	153.95

Captive ROTR Rack *(ROTR/CAPTIVE)*

Installing and maintaining equipment in traditional commercial racks can be a time consuming and tedious process. While moving a rack around on casters can provide some degree of accessibility, Raxxess has a better way. Designed to save you time and money on service calls by enabling unprecedented access to installed equipment, the new Captive ROTR rack is an idea whose time has finally come. Available in 35 rack spaces, the Captive ROTR rack consists of a rollout 22″ deep rack frame on heavy-duty 4″ casters and a 16-gauge welded steel housing. A ball-bearing swivel base sits between the caster base and the reinforced 12-gauge rack frame allowing 90-degree rotation when the rack frame is at full extension. Heavy duty 28″ full-extension telescoping sliders allow the rack frame to move smoothly in and out of the housing. Our unique "Omega" shaped, 3″ wide cable passage cutout in the roof combined with ample cable tie-off points in the rack allow for easy cable management.



CPROTR-35 35 spaces **1119.00**
CPROTR-42 42 spaces **1189.00**



Security Door

Button mount security door attaches directly to rack rail. Can be removed with ease and no tools.

SCDR-8B
8 space button mount.....**58.95**

SCDR-12B
12 space button mount.....**64.95**

SCDR-16B
16 space button mount.....**77.95**

SCDR-20B
20 space button mount.....**89.95**

PANELS

FPL Fan Panel
Heavy duty, 16 gauge steel; three space panels: punched to accommodate 4½″ diameter fans; textured finish.

FPL-1 Holds 1 fan **21.95**
FPL-2 Holds 2 fans.....**23.95**
FPL-3 Holds 3 fans.....**25.95**

VPL Vent Panel
For passive cooling of racks; .125″ aluminum. Black brushed anodized.

VPL-1 1 space...**10.95** **VPL-2** 2 spaces..**13.95**
VPL-3 3 spaces **14.95**

AFG Aluminum Flanged Panel
Attractive 1/2″ flanged for additional strength; sturdy .060 aluminum; easily machined. Black brushed anodized.

AFG-1 1 space..**12.95** **AFG-2** 2 spaces..**14.95**
AFG-3 3 spaces..**15.95** **AFG-4** 4 spaces..**19.95**

SFT Steel Flat Panel
Sturdy 14-gauge cold rolled steel; perfect for use where economy is the key; smooth finish.

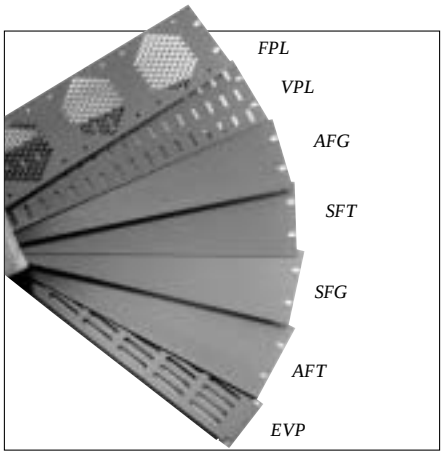
SFT-1 1 space .....**5.95** **SFT-2** 2 spaces.....**6.95**
SFT-3 3 spaces....**8.95** **SFT-4** 4 spaces.....**9.95**

SFG Steel Flanged Panel
16-gauge, cold rolled steel; 1/2″ flanged for additional strength. Perfect for use where economy is the key; textured finish.

SFG-1 1 space.....**8.95** **SFG-2** 2 spaces ..**10.95**
SFG-3 3 spaces .**11.95** **SFG-4** 4 spaces ..**13.95**

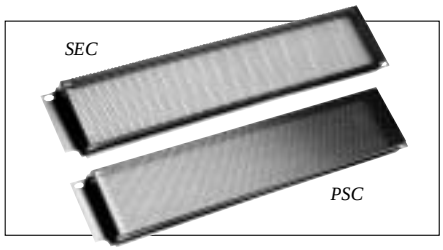
AFT Aluminum Flat Panel
.125″ aluminum; easily machined. Black brushed anodized.

AFT-1 1 space.....**11.95**
AFT-2 2 spaces.....**14.95**
AFT-3 3 spaces.....**15.95**
AFT-4 4 spaces.....**17.95**



EVP Economy Vent Panel
Inexpensive flanged panel for passive cooling; 16-gauge, cold rolled steel; smooth finish.

EVP-1 1 space.....**7.95** **EVP-2** 2 spaces ..**10.95**
EVP-3 3 spaces **13.95**



Perforated Steel Security Covers
One-piece 16-gauge perforated steel covers protect equipment from tampering. 1″ offset allows knob clearance. Neoprene gasket edges prevents scratches. Screw notches allow removal of the cover without removing the gear. Black baked enamel finish. The SEC series has 5/32″ holes that provide a clear view of secured equipment. The PSC series has 1/16″ holes creating an obstructed view.

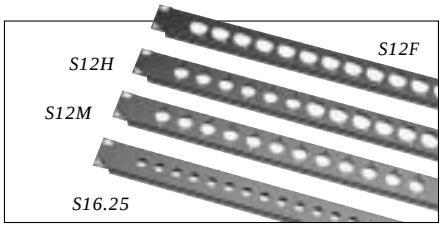
SEC-1 1 space....**15.95** **PSC-1** 1 space....**15.95**
SEC-2 2 spaces ..**16.95** **PSC-2** 2 spaces ..**16.95**
SEC-3 3 spaces ..**18.95** **PSC-3** 3 spaces ..**18.95**

Perforated and Fine Vent Panels
One piece 16-gauge steel construction; finished in black baked enamel. The 5/32″ perforations in the PVP series allow optimum air circulation, while the 1/16″ perforations of the FVP provide a more limited view.

PVP-1 1 space .....**9.95** **FVP-1** 1 space.....**9.95**
PVP-2 2 spaces..**10.95** **FVP-2** 2 spaces ..**10.95**
PVP-3 3 spaces..**12.95** **FVP-3** 3 spaces..**12.95**

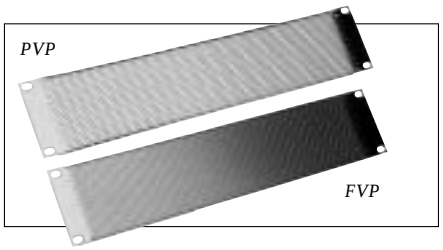
Single Space Connector Panels
Constructed of 16-gauge cold rolled steel and flanged for strength.

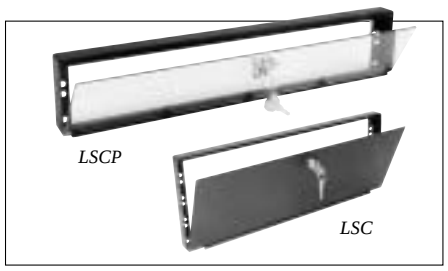
S12F for 12 female XLR's **7.95**
S12H for 12 male XLR's **7.95**
S12M for 6 male and 6 female XLR's **7.95**
S16.25 for sixteen 1/4″ connectors.....**7.95**



Hinged Steel Flanged Panels
16-gauge flanged steel panel with textured finish. Hinged to allow access to your rack via the front panel. Perfect for bringing electrical wiring and connectors to the front of the rack.

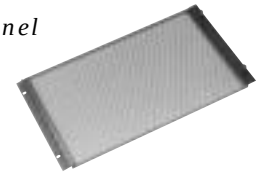
SFG-2H 2 spaces .....**Call**
SFG-3H 3 spaces .....**Call**
SFG-4H 4 spaces .....**Call**





Locking Security Covers
The ultimate in security for rack-mounted devices. Heavy gauge steel frame construction, with either a black steel or see-through plexi front panel. The eight-space version is available with 2, four space plexi or steel panels. Perforated available as an option.

LSC-2 2 space, black steel.....**28.95**
LSC-4 4 space, black steel.....**32.95**
LSCP-2 2 space, plexi **35.95**
LSCP-4 4 space, plexi **39.95**
LSCP-8 two 4 space, plexi **58.95**

Hinged Vent Panel


A hinged vent panel that opens side to side, left or right. Captive screws thread directly into rack rail; constructed of 16 gauge, perforated 5/32″ steel. Occupies 6 rack spaces; finished in black baked enamel.

HVP-6.....**38.95**

MODULAR PANELS

SCSI Module

Fits SCSI connectors 25 and 50 pin.

DB25-1 ..... **7.95** **DB25-2** **7.95**
DB25-3 ..... **7.95**

Canare BNC Module

Fits Switchcraft, Neutrik (FP, LNEFP), Cannon.

XLR-1M ..... **7.95** **XLR-2M** **7.95**
XLR-3M ..... **7.95** **XLR-4M** **7.95**
XLR-5M ..... **7.95** **XLR-6M** **7.95**

RCA Module

With 1/4″ hole fits RCA jacks and LEDs.

RCA-1 ..... **7.95** **RCA-2** **7.95**
RCA-3 ..... **7.95** **RCA-4** **7.95**
RCA-5 ..... **7.95** **RCA-6** **7.95**
RCA-7 ..... **7.95** **RCA-8** **7.95**

Phone Module

With 3/8″ hole for phone jacks and F connectors.

PH-2 ..... **7.95** **PH-3** **7.95**
PH-4 ..... **7.95** **PH-5** **7.95**
PH-6 ..... **7.95** **PH-7** **7.95**
PH-8 ..... **7.95**

XLR Female Module

Fits Switchcraft, Cannon, Neutrik (FP, LNEFP, Speakon), Whirlwind WC3F-M.

XLR-1F ..... **7.95** **XLR-2F** **7.95**
XLR-3F ..... **7.95** **XLR-4F** **7.95**
XLR-5F ..... **7.95** **XLR-6F** **7.95**

XLR Male Module

Fits Switchcraft, Neutrik (FP, LNEFP), Cannon

XLR-1M ..... **7.95** **XLR-6M** **7.95**
XLR-1M ..... **7.95** **XLR-6M** **7.95**
XLR-1M ..... **7.95** **sXLR-6M** **7.95**

Universal

Fits Switchcraft XLR male and female and Neutrik FDL, MDL, FP and MP.

UN-2 ..... **7.95** **UN-3** **7.95**
UN-4 ..... **7.95** **UN-6** **7.95**

Elco Module

Fits 38, 56, 90 and 120 multipin Elco connectors.

EL-1-38 ..... **7.95** **EL-2-38** **7.95**
EL-1-56 ..... **7.95** **EL-2-6** **7.95**
EL-1-90 ..... **7.95** **EL-2-90** **7.95**
EL-1-120 ..... **7.95**

Multipin Whirlwind Module

WH-39 Fits Whirlwind 39-pin connectors **7.95**
WH-61 Fits Whirlwind 61-pin connectors **7.95**
MP-190 For use with modular panels in a standard 1900 box **7.95**

Midi/Duplex Module

Fits standard MIDI connectors.

MIDI-1 ..... **7.95** **DP-1** **7.95**
MIDI-2 ..... **7.95** **DP-2** **7.95**
MIDI-4 ..... **7.95**

Banana Module

Fits dual universal banana jacks for speakers.

BAN-1 ..... **7.95** **BAN-2** **7.95**
BAN-3 ..... **7.95** **BAN-4** **7.95**
BAN-6 ..... **7.95**

Hubbel Twistlock Module

HUB-1 ..... **7.95**

Neutrik NL8MPR Module

NL-8 ..... **7.95**

Modular Wall Box

Modular panels allow users to create a custom connector system using stock panels. Available as a 19″ rack frame (one piece construction) or as a 2, 4 or 6 panel attractive wall box. Panels are aluminum and measure 3½ x 3½″. Wall boxes (MWB) are of 16-gauge steel construction.

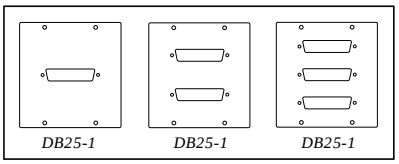
MCP - Modular connector panel, aluminum **7.95**
MWB-2 Holds 2 modular panels **23.95**
MWB-4 Holds 4 modular panels **29.95**
MWB-6 Holds 6 modular panels **33.75**

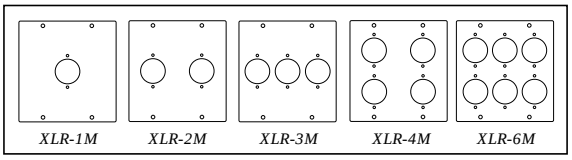
MCF - Connector Frame

Holds 5 panels **13.49**

Modular Blank Panel

3.5″ square aluminum panel with no connector punchout. Fits MCF or MWB **7.95**

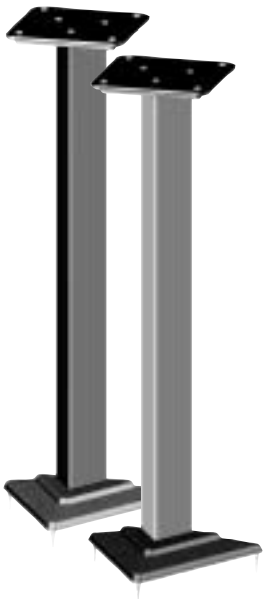




STUDIO MONITOR STANDS

Monitor Speaker Stands

Ultimate Support monitor stands feature cabinet-quality top plates, cable management, metal base spikes, and the ability to add shot or sand internally. They are also great for living rooms to get those expensive speakers up to ear level and produce the sharpest and clearest sound possible.



Cabinet Quality Medium-Density Fiberboard Top Plate

Place monitors on rubber pads that isolate the monitor cabinet from vibration.

Cable Management System

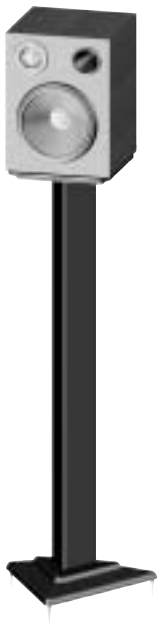
Two cable channels run inside the column to separate balanced or unbalanced cables from the power cable. Just lay them in the open-faced channels and snap on the backing. Wrap extra cable on pegs hidden beneath the stand.

Stable Stand

The aluminum column filled with shot or sand combined with the cast-zinc base, creates a bottom-heavy, ultra-stable stand.

Metal Base Spikes

To bypass the sound-dampening effect, metal base spikes go through the carpet and padding to rest on the subfloor. For hard surfaces use enclosed rubber feet.



All Studio Monitor Stands are sold in pairs, have a 14″ base, a 9 x 9″ top plate and a load capacity of 75 lbs.

	Height	Weight	Color	Price
MS-12B2	12″	5.7 lbs.	Black	**Call**
MS-12P2	12″	5.7 lbs.	Pewter	**Call**
MS-18B2	18″	6.4 lbs.	Black	**Call**
MS-18P2	18″	6.4 lbs.	Pewter	**Call**
MS-24B2	24″	7.1 lbs.	Black	**Call**
MS-24P2	24″	7.1 lbs.	Pewter	**Call**
MS-30B2	30″	7.8 lbs.	Black	**Call**
MS-30P2	30″	7.8 lbs.	Pewter	**Call**
MS-36B2	36″	8.45 lbs.	Black	**149.00**
MS-36P2	36″	8.45 lbs.	Pewter	**149.00**
MS-45B2	45″	9.55 lbs.	Black	**159.00**
MS-45P2	45″	9.55 lbs.	Pewter	**159.00**

Choosing a Stand Height

When seated in the mix position, measure the distance from your ears to the floor and choose the stand height (36″ or 42″) that will place the monitor tweeters at ear level. Low frequency sound waves are large and fill a lot of space. High frequencies come from smaller, direct waves. Studio monitors are designed so that on-axis response (direct parallel path from monitor to ears) yields the most accurate sound.

KÖNIG & MEYER X-STYLE KEYBOARD STANDS



E-Z Adjust X-Stand

QUALITY GERMAN-ENGINEERED X-STANDS

E-Z Adjust X-Stand (18940B)

This sturdy, medium-duty steel frame has a durable black finish. It features an easy adjust button, which allows you to simply open the stand to the desired height and release. The stand locks firmly into place. Push the button to close the stand.

Performer X-Stand (18962B)

This light-duty X-stand features a pin system which allows the stand to open to a pre-set height every time. A dual-position center point allows a greater range of height adjustment.

	Set-Up Height	Weight	Load Capacity	Price
E-Z Adjust	18-39″	11 lbs.	165 lbs.	**49.00**
Performer	25-35″	7 lbs.	90 lbs.	**35.00**



Performer X-Stand

SPEAKER STANDS



TS-80 shown with speaker

TS-80 Series Speaker Stands

These high-quality aluminum and steel tripod speaker stands feature a built-in socket adapter that accommodates both 1⅜″ and 1½″ speaker sockets. A glass-reinforced telescoping collar has a die-cast metal clamp for maximum security and durability. The secure leg collar slides smoothly on the tube for easy set-up.



- Glass reinforced polycarbonate fitting at the leg juncture uses a metal clamp over the fitting to minimize over-tightening and breakage
- The stands raise and lower with the turn of a knob, locking tightly for secure support. Metal clamps minimize over-tightening and breakage





- The rubber footcaps are held in place by both internal and external friction so they stay on gig after gig
- The connector tube, which telescopes up to accommodate 1⅜″ speaker sockets and down for 1½″ sockets. Locks firmly into desired position. Minimizes the need for adapters. The telescoping tube lock prevents accidentally pulling out the center tube

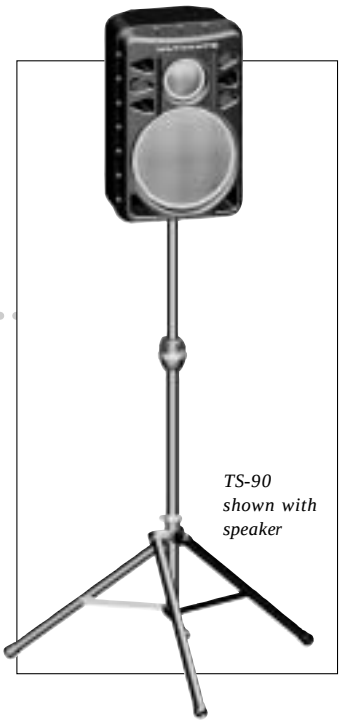
Telescoping and Leg Collars

Die-cast metal clamps fit snugly over glass-reinforced polycarbonate fittings creating the strongest and most durable telescoping collar and leg collar possible. Since the handknob actually tightens the metal clamp and not the plastic directly, these collars aren't easily overtightened, snapped or cracked!

TS-90 Series Speaker Stands

A step-up from the TS-80, the TS-90 tripod speaker stands features a telescoping tube lock that increases safety and prevents accidently pulling the center tube. It has an internal friction mechanism with three settings. In the "RAISE" setting, the mechanism grips and holds the tube, allowing you to incrementally pull the center tube up to the required height. "LOCK" the collar in place for extra security. In the "LOWER" setting, TeleLock eases off and assists lowering the center tube. This unique design safely takes the strain off you!





TS-90 shown with speaker

TRIPOD SPEAKER STANDS											
	TS-70B	**TS-80B**	**TS-80S**	**TS-88B**	**TS-88S**	**TS-88BL***	**TS-90B**	**TS-90S**	**TS-99B**	**TS-99S**	**TS-99BL***
Height	50″ - 77″	44″ - 79″	44″ - 79″	62″ - 110″	62″ - 110″	62″ - 110″	44″ - 79″	44″ - 79″	62″ - 110″	62″ - 110″	62″ - 110″
Weight	5 lbs. 4 oz.	6 lbs. 4 oz.	6 lbs. 4 oz.	8 lbs.	8 lbs.	9 lbs.	6 lbs. 4 oz.	6 lbs. 4 oz.	8 lbs.	8 lbs.	9 lbs.
Base Diameter	40″	47″	47″	62″	62″	62″	47″	47″	62″	62″	62″
Load Capacity	100 lbs.	150 lbs.	150 lbs.	150 lbs.	150 lbs.	150 lbs.	150 lbs.	150 lbs.	150 lbs.	150 lbs.	150 lbs.
Folded	41½″ x 5″	43½″ x 5½″	43½″ x 5½″	57½″ x 5½″	57½″ x 5½″	57½″ x 6″	43½″ x 5½″	43½″ x 5½″	57½″ x 5½″	57½″ x 5½″	57½″ x 6″
Color	Black	Black	Silver	Black	Silver	Black	Black	Silver	Black	Silver	Black
Price	**60.00**	**85.00**	**80.00**	**105.00**	**94.50**	**129.50**	**105.00**	**96.00**	**125.00**	**110.00**	**149.00**

*The TS-88L and TS-99BL are Leveling Leg Tripods. The leg adjusts in length to level the stand on uneven surfaces.

B&H PHOTO - VIDEO - PRO AUDIO

SPEAKER STANDS

MUSIKA TS-70B ECONOMY SPEAKER STAND

The Musika TS-70 tripod speaker stand was designed to bring Ultimate quality to the masses with a more affordable speaker stand. Although it is not a heavy duty stand like the TS-80 or TS-90 series, it still brings quality and innovation to this segment.



- Glass reinforced telescoping collar secures the telescoping tube without slipping. A safety pin is also included for extra security



- Glass reinforced tripod leg fitting secures the base of the tripod. By using a glass reinforced plastic ultimate tripods are stronger and more durable that the comepition



- Like all tripod speaker stands from Ultimate, the TS-70 includes an adapter for 1½″ speaker sockets as well as 1⅜″ sockets

MOUNTING BRACKETS

BMB-200K Large Mounting Bracket
- External 4 x 6″ bracket securely mounts any speaker cabinet with a flat bottom on an Ultimate tripod (1½″ tube)
- Glass-reinforced polycarbonate fitting holds 100 lbs.
- All hardware included for installation

(ULBMB200K).....**16.80**

TMB-400K Slide Mounting Bracket
- Low-profile external 4 x 4″ bracket securely mounts any speaker cabinet with a flat bottom on an Ultimate tripod (1½″ tube)
- Two-piece design: the plate attaches to the bottom of the speaker, and the slide-in base connects securely to the tripod
- All hardware included for installation
- Strong glass-reinforced polycarbonate fitting holds 100 lbs.

(ULTMB400K).....**12.00**

TAD-100 Adapter Stud (ULTAD100)
- Converts a 1½″ tripod top tube to mount a speaker cabinet with a 1″ inside diameter socket
- Strong glass-reinforced polycarbonate fitting holds 100 lbs **11.25**



TAD-138 Adapter Stud (ULTAD138)
Same as above but converts 1½″ top tube to 1-1⅜″.....**10.00**



TSM Series Socket Mounting Brackets
- Internal 4″ diameter bracket securely mounts to most any speaker cabinet with a flat bottom on an Ultimate tripod
- All hardware included for installation

TSM-150K (ULTSM150K)
1½″ cavity. Glass-reinforced polycarbonate fitting holds 100 lbs.....**12.00**

TSM-150MK (ULTSM150MK)
1½″ cavity. Metal fittings hold 150 lbs....**14.25**

TSM-138MK (ULTSM138MK)
1⅜″ cavity. Metal fittings hold 150 lbs....**13.50**

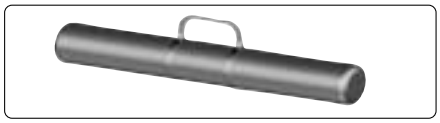
BAGS



BAG-80 Zippered Tote Bag (ULBAG80)
Holds one TS-80 or TS-90 series tripod stands. 5.9 x 5.9 x 48″ **19.95**



BAG 80D Zippered Tote Bag (ULBAG80D)
Double bag—holds two TS-80 or TS-90 series tripod stands. 5.9 x 11.8 x 44.5″ ..**35.00**



BAG-88 Zippered Tote Bag (ULBAG88)
Holds one TS-88 or TS-99 series tripod stands. Also holds a TS-88BL or TS-99BL tripod. 5.9 x 5.9 x 63″ **24.95**



BAG-88D Zippered Tote Bag (ULBAG88D)
Double bag—holds two TS-88 or TS-99 series tripod stands. Also holds two TS-88BL or TS-99BL tripods. 5.9 x 11.8 x 63″ **34.95**

KEYBOARD STANDS

IQ X-Stands

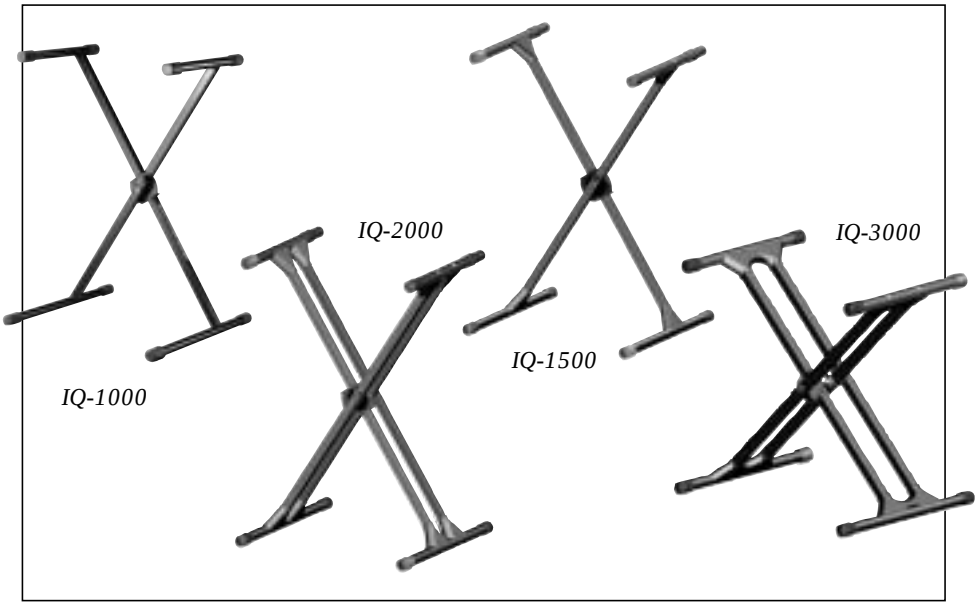
These high-performance stands offer many features including:

Memory Lock
Set the stands to different positions - memory lock opens to your choice every time.

Stable Stand
Large footprint and tubing make these stands extremely stable. Adjustable end-caps stabilize the stand on uneven surfaces.

Cable Management
Clips throughout stands and tiers.

Reinforced Connectors
The IQ-1500, 2000 and 3000 stands have connections between the leg braces and top/bottom tubes that increase strength and durability over welded x-stands.



IQ X-STANDS				
	IQ-1000	**IQ-1500**	**IQ-2000**	**IQ-3000**
Set-up Height	26½″ - 36″	26½″ - 36″	26½″ - 36″	25″ - 37″
Weight	11 lbs.	12 lbs.	15 lbs.	22 lbs.
Load Capacity without Tiers	100 lbs.	100 lbs.	150 lbs.	300 lbs.
Load Capacity with Tiers	100 lbs. distributed	100 lbs. distributed	150 lbs. distributed	300 lbs. distributed
Stand/2nd Tier/3rd Tier (lbs.)	75/25/–	75/25/–	75/50/25	125/100/75

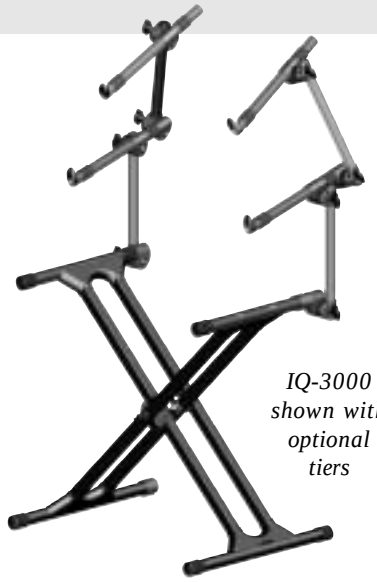
Accessories

IQ-200 Set-Position Second Tier
Second tier holds a 2nd keyboard. For the IQ-1000.....**29.95**

IQT-100 and IQT-200 Tiers are height, depth, width and angle adjustable. Height and depth from 1″ to 12″, tier angle and width in 7.5° increments. Tiers come with a 3-year warranty.

IQT-200 Adjustable Tier
Used as a 2nd tier on the IQ-1500; a 2nd and 3rd tier on the IQ-2000; and a 3rd tier on the IQ-3000....**42.00**

IQT-100 Adjustable Tier
Used as a second tier on the IQ-3000 .....**49.00**



IQ-3000 shown with optional tiers

IQ-1000

- 14″ support arms, 20″ leg lengths
- 100 lb. load capacity
- Memory Lock system with 4 height adjustment settings
- Off-set leg braces make more leg room and space for pedals
- Adjusting end caps
- Accomodates IQ-200 set-position second tier

$35.00

IQ-1500

Same as above, plus–

- 16½″ support arms, 20″ leg lengths
- Reinforced connectors between leg braces and top/bottom tubes increase strength and durability over welded x-stands
- Accomodates optional IQT-200 fully-adjustable second tier

Call

IQ-2000

Same as above, plus–

- Sturdy, double brace stand
- 150 lb. load capacity
- Accomodates 2 optional IQT-200 fully-adjustable second and third tiers

$54.00

IQ-3000

Same as above, plus–

- 19½″ support arms, 22″ leg lengths
- 300 lb. load capacity (for the lowest 2 settings, 25″ and 27″, load capacity is 250 lbs.
- Memory Lock system with 9 height adjustment settings
- Accomodates optional IQT-100 and IQT-200 fully-adjustable second and third tiers

Call

KEYBOARD STANDS

Apex AX-48BP Column Keyboard Stand

The Apex column keyboard stands have set the standard in the industry for performing musicians. Ingenious design makes them incredibly easy to set up, adjust and transport.



- Sleek, hollow column becomes its own carrying case. Sets up in seconds with no tools
- Sturdy column stand is stable, even with large 88-note keyboards
- Arms firmly support the keyboard yet detach and slide into the top of the column for easy transport. Includes two column tiers that are expandable to three with Apex accessories
- Legs slide out into place for quick set-up or store neatly inside column base for transport.
- Durable black anodized aluminum looks great on any stage.
- 46″ height with angle adapter. It weighs 19 lbs., has a load capacity of 125 lbs per tier; 220 lbs. total. The collapsed length is 52″
- Comes with two standard clamps and two sets of 13″ tribar arms $162.00

AX-48BP Apex Plus
Same as above, includes an Apex Mic Boom plus a Super Clamp and 18″ long tribars to accommodate a larger keyboard **218.95**

AX-48BLX Apex Deluxe
Comes with one standard clamp, 13″ tribar arms one super clamp and 18″ long tribar arms to accommodate a larger keyboard .....**Call**

apex accessories

Tribars and Clamps
For keyboard tier assembly.
Requires 2 tribars and 1 clamp.

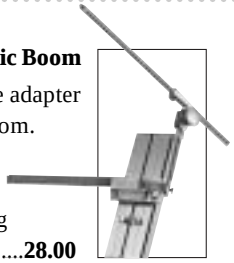
TBR-130
Standard Tribar (13″)....*each* **9.75**

TBR-180
Long Tribar (18″) .....*each* **9.75**



CMP-480
Standard Clamp. For use with standard 13″ tribars only .....**Call**

CMP-485
Super Clamp. For use with 18″ long and standard tribars ...**21.00**



MIC-480 Mic Boom
Microphone adapter with 24″ boom. Includes hardware for attaching to the stand **28.00**



BAG-88
Zippered tote bag. Fits entire APEX column into tote bag for transport **24.95**



Deltex DX-48B Keyboard Stand

An economical yet high quality choice for musicians with lighter keyboards.

- Sleek triangular column design with a durable black anodized finish.
- Sturdy column holds up to two keyboards
- Arms firmly support the keyboard yet detach for easy transport
- Folding legs detach and store in the optional tote bag
- Durable black anodized aluminum looks great on any stage.
- 47″ height; weighs 14 lbs, it has a load capacity of 50 lbs. per tier (100 lbs. total). It's collapsed length is 45″

$105.00

DX-48BP Deltex Plus
Includes a DX-48B plus the BAG-88 tote bag.....**Call**

deltex accessories

BAG-88 Zipper Tote Bag
Heavy-duty nylon tote bag holds one Deltex stand. Dimensions are 10½ x 60½″ **24.95**



ULTIMATE

KEYBOARD *and* HOME STUDIO STANDS



Apache Triple Tier A-Frame Stand

The Apache Triple Tier frame is a versatile option for stage or studio. Each tier is height and tilt adjustable, while the back legs can be positioned vertically to optimize valuable studio space. It is available in 5´ and 4´ widths, and comes with three tiers, three pairs of 14⅛˝ support bars and a tote bag. (The top tier on the 60˝ wide Apache has 20˝ support bars.) This frame is easily expandable with a wide variety of accessories.

	AP-22BPT	AP-26BPT
Height	60˝	60˝
Width	48˝	60˝
Load Capacity per Tier	150 lbs.	150 lbs.



Studio Organizer

Designed to keep your gear within arm's reach. Components are height and tilt adjustable and the modular design is expandable with a variety of accessories. The kit includes a large laminated worktable, top-tier with two 14⅛˝ support bars and two half-tiers with 8¾˝ support bars.

HS-26BP STUDIO ORGANIZER					
Height	**Worktable**	**Footprint**	**Weight**	**Load Capacity**	**Price**
27½˝ - 32˝	16˝ x 35½˝	19˝ x 38˝	21 lbs.	150 lbs. total distributed	**175.00**

THINKERTOYS

Customize your existing stand or build your own stand from the ground up. Thinkertoys provide a means of connecting Ultimate aluminum tubing with fittings, connectors, swivels, casters, foot caps, shelves, hand-knobs, speaker mounts, lighting attachments, legs and more. If you can't find a support to match your needs for instruments, computers, and equipment, ThinkerToys are for you.

Aluminum Tubing

Available in one foot increments up to 12 feet in silver or black anodized finish.

12067 Standard Silver Tubing
1½˝ diameter; .049 thickness.....**3.00**

12068 Standard Black Tubing
1½˝ diameter; .049 thickness.....**4.00**

13523 Small Diameter Tubing
1⅜˝ diameter; .049 thickness **6.00**

12069 Large Diameter Silver Tubing
1⅝˝ diameter; .058 thickness **4.00**

12070 Large Diameter Black Tubing
1⅝˝ diameter; .058 thickness **5.00**

12074 Thick Walled Silver Tubing
1½˝ diameter; .125 thickness.....**5.00**

12075 Thick Walled Black Tubing
1½˝ diameter; .125 thickness.....**5.95**



Single T-Frame Leg System

KRB-220B - Base - 22˝ wide.....**24.95**

Single A-Frame Leg System

CM-12BS - 42˝ leg. Comes with one pivot fitting and leg brace.....**Call**

Half-Tier HT-15B

The HT-15B half-tier holds smaller gear off to the side of the stand. It includes one T-fitting and two short Support bars. It has a load capacity of 15 lbs.....**24.95**

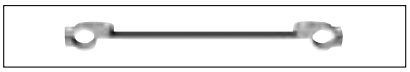
Universal Support Bars

Includes a universal fitting and rubber pads.

USB-160 - Short (8¾˝).....**9.95**
USB-220B - Standard (14⅛˝).....**12.95**
USB-222 - Long (20˝) **14.95**

KEYBOARD *and* HOME STUDIO STANDS

THINKERTOYS



Bracing

Aluminum flatbar with universal fittings.

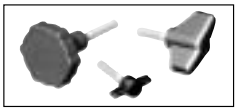
BRC-140B - 1″ flatbar; 14″ long **12.95**
BRC-190B - 5/8″ flatbar; 19″ long.....**12.95**

End Caps, End Plugs



Support Tiers offer added stability. Tiers come with two T-fittings.

FCP-150 - Foot cap for 1½″ tubing .....**.95**
ECP-150 - End cap for 1½″ tubing.....**.50**
ECP-162 - End cap for 1⅝″ tubing.....**2.00**
ELP-150 - End plug for 1½″ tubing.....**.50**
ECP-151 - End plug for thick-walled 1½″ tubing .....**.50**
ECP-162 - End plug for 1⅝″ tubing .....**.50**



Fastners

For use with T-fittings and collars.

TSS-150 - T-screw and spacer, 1/4-20 ...**1.00**
ZKB-175 - 1.75″ length handknob with washer. Use on TS-80/88 stands.....**3.50**
ZKB-137 - 1.37″ length handknob with washer **3.50**
HEX-100 - Hex nut, 1/4-20.....**.20**
SHS-100 - Socket screw, 1″, 1/4-20.....**.20**

Tools

BAL-100 Ball driver 3/16 .....**Call**
CUT-100 Tubing cutter.....**Call**
AWR-316 Allen wrench 3/16 .....**1.00**

Boom Jointers



Connects two standard tubes in line.

BJR-150B Black color ..**17.50**
BJR-150 Silver color.....**16.00**



CLA-200 Clip Assembly

The CLA-200 holds notes, sheet music, etc. Mounts on 1½″ tubing.....**70.00**

CSB-180 Computer Support Bar



Attaches to horizontal tiers, extending forward from the underside of the tier. Holds computer keypads. Sold as single bar (12¾″).....**10.00**



CST-150 Caster Assembly with Swivel

Mounts in the end of standard tubing where adjustable eangles of the caster are desired **24.95**

EXT-222 Pin Extension Pair

Slide over existing support bar pins for securing front slanted instruments .....**Call**

Microphone Mounts



MIC-150 - Microphone End Mount. Adapter mounts mic booms or goosenecks to the end of standard tubing **19.95**

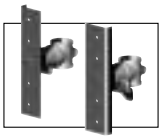


MTM-587 - Microphone T-Mount. Adapter mounts mic booms or goosenecks along a standard tube.....**24.95**

PVF-150



Connect two tubes at various angles.....**8.00**

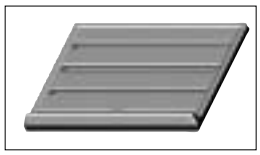


Rack Rail Pairs

For mounting standard 19″ rack equipment. hardware included.

RL-05B - 5 space rack rail pair .....**Call**
RL-10B - 10 space rack rail pair .....**Call**

Shelves



Lightweight black polymer shelves support smaller equipment on keyboard tiers.

SHF-185 - 13½ x 18½″ **34.95**
SHF-415 - 13½ x 41½″ **56.00**

Swivel Joints



SJI-150 - Swivel joint, tube-to-tube. Forms an adjustable joint between two 1½″ tubes.....**29.95**



SJT-150 - Swivel joint, tube-to-fitting. Extends a half-tier to the side of an A-Frame stand. Regardless of position, the SJT-150 keeps the HT-15B on a horizontal plane **24.95**

Velcro

VCA-100 - Adhesive velcro-hook (per foot).....**Call**
VCA-200 - Adhesive velcro-loop (per foot).....**Call**
VCS-712 - Velsro strap pair. Used for cord control.....**4.00**

TCR-150 Telescoping Collar



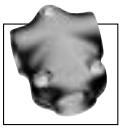
Used to telescope 1½″ standard tubing tubing out of a 1⅝″ tube.....**6.00**



TFG-150 T-Fitting

Forms a rigid 90° joint between two standard 1½″ tubes **5.00**

HTG-150 Hatched T-Fitting



Two-piece T-fitting is strong and convenient for adding items to existing stands.....**6.00**



ATG-150

Adjustable T-Fitting

Set any angle in 7.5° increments.

ATG-150 - For 1½″ tubing.....**14.95**
ATG-133 - For 1½″ and 1¼″tubing.....**14.95**
ATG-125 - For 1¼″tubing **14.95**

UNF-150 Universal Fitting



Used to attach flatbar to standard tubing **4.00**

B&H
PHOTO - VIDEO - PRO AUDIO

The Professional Audio SourceBook

Section 14b
Cables

Introduction to Cables.....1487

Hosa.....1488-1501

Monster Cable.....1502-1511

Proco.....1512-1523

CABLES

Choosing the Right Guitar Cables

A guitar cable is the primary cable connecting an electric guitar's output to its amplifier's input or to the input on the first effect pedal in the musician's pedalboard. Guitar cables are also used to hook basses to bass amps and keyboards to mixers. The broad category for these types of cables is "instrument". Guitar cables are constructed using single-conductor audio cable (also called coax) with an overall shield, terminated in 1/4″ phone plugs.

There is a wide variety of guitar cables because there is a wide variety of guitarists. It is almost impossible to get two guitarists to agree on what a truly great "sound" is. Even if they did, one of them would change his/her mind by midnight. Therefore guitar cables come in a wide variety of lengths, each with different standards for reliability, shielding, sonic quality, flexibility, appearance and price.

The primary guitar cable is the most abused on stage (besides the lead singer's microphone cable). Therefore, it must be built to withstand extreme trauma during performance. Also, in acts where appearance is critical, it also has to look like a million bucks. Although the patch cables used between the effect pedals and between the last effect and the amplifier (or pre amp) need to sound good and be very quiet, the stress on them is not as severe as the stress on the primary cable. Light duty patch cables can be used here without much fear of failure.

Because of the placement of the output jack on some guitars, a right angle plug may be needed, especially if your output is on the face of your guitar.

Here are the three most important rules governing guitar cables:

- Buy the shortest cable you can live with
- If it doesn't have a copper tip on the connectors, don't buy it
- Find a cable that is as flexible and tangle-free as you need for your performance.

The wire in your cable must be flexible and sturdy. "Thin" is generally not as reliable as "thick" cable, and thin is generally not as flexible as thick. The things that matter most in guitar cable construction are strength, flexibility, sonics, shielding and appearance. Strength in a guitar cable's wire is derived from the size (gauge) of the center conductor, the type and percentage of coverage of shield, and the thickness and material making up the jacket.

Guitar cables carry minuscule signals that must be kept away from noisy light dimmers and audio frequency interference that can jump into a cable's signal path at any time. The cable's shield helps to prevent this. Braided shields work better than spiral shields.

Choosing the Right Mic Cables

In pro audio, microphones are low impedance (Lo-Z) and terminate in 3-pin XLR connectors:

- Pin 1 (X)ternal Shield
- Pin 2 (L)ive Hot (+)
- Pin 3 (R)eturn Cold (-)

Another configuration includes an XLR female (the output of all professional mics is a 3-pin XLR male) connector to a 3-pin mini male (1/8″ or 3.5mm) connector for inputs to laptop computers and other devices where space is at a premium. Most professional mixer's microphone inputs are designed with "balanced" circuits to help decrease or eliminate noise and unwanted radio frequency interference (RFI).
Lo-Z Microphone cables can also be wired "unbalanced" and "Hi-Z" high impedance) microphones are available, for high impedance sound systems. Hi-Z cables allow the user to plug Hi-Z microphone directly into the input of, say, a guitar amp or the input of a Hi-Z mixer. Unbalancing a balanced microphone by using an unbalanced cable allows it to sometimes be used in the input of a Hi-Z mixer. This does not always work, depending on the input impedance of the mixer. When this does not work, a Lo-Z to Hi-Z transformer must be placed in line at the end of a standard Lo-Z mic cable. These commercially available transformers make the proper change from XLR to 1/4″ for you.
Microphone wire consists of a twisted pair of copper conductors (typically 22 - 24 AWG — American Wire Gauge). These conductors are covered with one of three types of shielding: braided, spiral (also called "serve" shield), and foil shielding which includes a drain wire. Foil shields work great in snakes, but prove to be unreliable in cables designed for portable use. Braided shield is best for mic cables and spiral is a little more flexible and less expensive than braid.

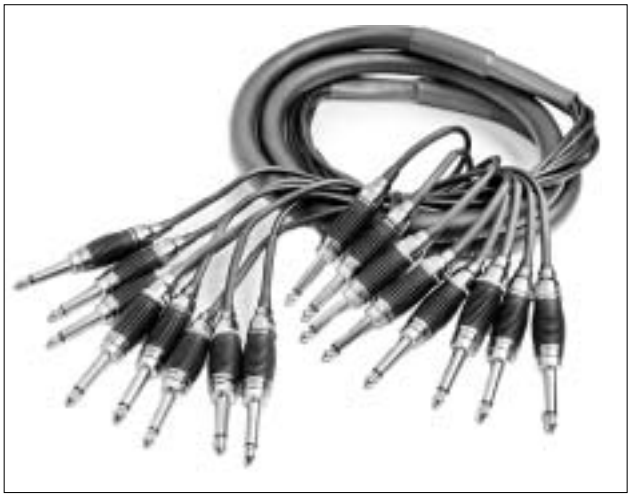
Choosing the Right Speaker Cables

When buying speaker cable, there are two basic things to remember: Less is best. Buy the shortest cable possible for the application. More is best. Buy the largest gauge speaker cables you can afford (the smaller the gauge number, the bigger the wire).

The three basic types of connectors used for speakers in live performance are: 1/4″ connectors (the same ones used in your guitar's output), dual banana plugs, and Neutrik Speakons, wonderful connectors designed specifically for speaker applications. There are other methods of connecting amplifiers to speakers (spade lugs, bare wire) but these three connectors, 1/4″ phone plugs, dual banana plugs and Speakons are the standard.

MICROPHONE CABLES

The Hosa Philosophy



Since they started making cables almost 20 years ago, Hosa's philosophy has been that less-expensive doesn't have to mean "cheaper", and that it simply isn't necessary to connect your equipment with "NASA-approved" gold-plated cables thick enough to tow an eighteen-wheeler. Meaning, if you pay more for a cable, your music won't necessarily sound better. So, if you're like the rest of us and can't afford to spend a week's salary for a few cables, then look no further than the regular Hosa brand for professional cables that work as hard as you do. In the regular Hosa line, strict attention is paid to the basics of cable design and construction. Hosa makes sure that cables are well-shielded from external sources of radio-frequency and current-induced noise and that they're of the proper gauge and of high copper purity to ensure low capacitance. In addition, they use correct soldering techniques to eliminate "cold-solder" joints which could cause intermittent connections or "ground hums". And finally they make sure that the jacket and head-shell/strain-relief combination are rugged enough to withstand the rigors of the intended application. Hosa provides all of these benefits in every cable and backs them up with a Limited Lifetime Warranty. However, for those desiring the ultimate in purity of conductors and esoteric dielectric materials, and are unencumbered by budgetary restraints, Hosa offers their very-high-end Zaolla Silverline line of ultra pure, solid silver interconnects for S/PDIF, microphone, guitar, and audio equipment hookup.

Premium Quad-Type

Four inner wires are twisted together in pairs to form the "hot" and "cold" conductors. These are surrounded by a braided shield. Excellent EMI/RFI rejection.



Model #	Length		Price
CMI-110BLK	10′	4 Inner Conductors + Shield	**14.95**
CMI-125BLK	25′	4 Inner Conductors + Shield	**19.95**

Budget Hi-Z Microphone Cables

Two inner wires form the "hot" and "cold" conductors. These are surrounded by a braided shield. Quality on a budget.



Model #	Length		Price
MCH-105	5′	XLRF to 1/4″ Phone	**9.95**
MCH-110	10′	XLRF to 1/4″ Phone	**10.95**
MCH-125	25′	XLRF to 1/4″ Phone	**14.95**

Budget Microphone Cables

Two inner wires form the "hot" and "cold" conductors. These are surrounded by a braided shield. Quality on a budget.



Model #	Length		Price
MCL-105BLK	5′	2 Inner Conductors + Shield	**10.95**
MCL-110BLK	10′	2 Inner Conductors + Shield	**12.95**
MCL-125BLK	25′	2 Inner Conductors + Shield	**16.95**
MCL-150BLK	50′	2 Inner Conductors + Shield	**25.95**
MCL-1100BLK	100′	2 Inner Conductors + Shield	**39.95**

Mic Cables with Black XLR Connectors



Model #	Length		Price
MBL-105	5′	Black XLRF to Black XLRM	**9.95**
MBL-110	10′	Black XLRF to Black XLRM	**11.95**
MBL-125	25′	Black XLRF to Black XLRM	**15.95**

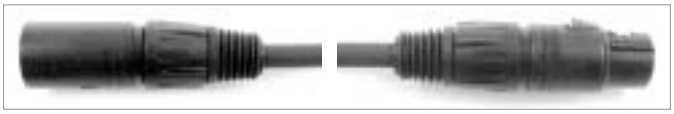
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MICROPHONE CABLES

Microphones require cables that can protect their very low-level output signals and deliver them to the console at full frequency response and without adding noise. Hosa's CMK and MMK-Series cables are chosen every day by professionals who know an amazing value when the see one. Featuring Neutrik-brand XLR connectors, oxygen-free copper conductors in your choice of 20 or 24 gauge (AWG), dense braided shielding for strength and EMI/RFI noise rejection, and tough, protective neoprene-type jacketing, these mic cables get the job done right.

Choose the CMK or CXP-Series for single cable runs. The 20-gauge conductors are incredibly rugged and road-worthy. Use the MMK or MXP-Series 24-gauge cables if you intend to bundle multiple cables together. Their smaller diameter helps keep the bundled runs more flexible.

MICROPHONE CABLES



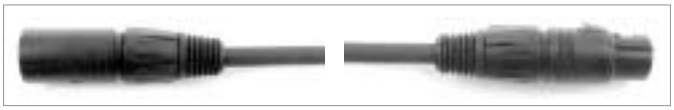
Model #	Length		Price
CMK-003	3′	20 AWG, Neutrik Connectors	**15.95**
CMK-005	5′	20 AWG, Neutrik Connectors	**16.95**
CMK-010	10′	20 AWG, Neutrik Connectors	**18.95**
CMK-015	15′	20 AWG, Neutrik Connectors	**20.95**
CMK-020	20′	20 AWG, Neutrik Connectors	**22.95**
CMK-025	25′	20 AWG, Neutrik Connectors	**24.95**
CMK-030	30′	20 AWG, Neutrik Connectors	**26.95**
CMK-050	50′	20 AWG, Neutrik Connectors	**37.95**
CMK-075	75′	20 AWG, Neutrik Connectors	**43.95**
CMK-100	100′	20 AWG, Neutrik Connectors	**54.95**

MICROPHONE CABLES for HIGH-Z MICS



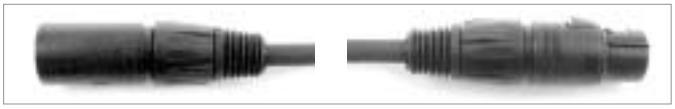
Model #	Length		Price
CXP-003	3′	20 AWG, Neutrik XLRF to 1/4″Phone	**13.95**
CXP-005	5′	20AWG, Neutrik XLRF to 1/4″Phone	**14.95**
CXP-010	10′	20 AWG, Neutrik XLRF to 1/4″Phone	**16.95**
CXP-015	15′	20AWG, Neutrik XLRF to 1/4″Phone	**18.95**
CXP-020	20′	20 AWG, Neutrik XLRF to 1/4″Phone	**19.95**
CXP-025	25′	20 AWG, Neutrik XLRF to 1/4″Phone	**22.95**
CXP-030	030′	20 AWG, Neutrik XLRF to 1/4″Phone	**24.95**
CXP-050	50′	20 AWG, Neutrik XLRF to 1/4″Phone	**34.95**
CXP-075	75′	20 AWG, Neutrik XLRF to 1/4″Phone	**39.95**
CXP-100	100′	20 AWG, Neutrik XLRF to 1/4″Phone	**49.95**

MICROPHONE CABLES



Model #	Length		Price
MMK-003	3′	24 AWG, Neutrik Connectors	**13.95**
MMK-005	5′	24AWG, Neutrik Connectors	**14.95**
MMK-010	10′	24 AWG, Neutrik Connectors	**16.95**
MMK-015	15′	24AWG, Neutrik Connectors	**17.95**
MMK-020	20′	24 AWG, Neutrik Connectors	**19.95**
MMK-025	25′	24 AWG, Neutrik Connectors	**21.95**
MMK-030	30′	24 AWG, Neutrik Connectors	**24.95**
MMK-050	50′	24 AWG, Neutrik Connectors	**29.95**
MMK-075	75′	24 AWG, Neutrik Connectors	**36.95**
MMK-100	100′	24 AWG, Neutrik Connectors	**44.95**

MICROPHONE CABLES for HIGH-Z MICS



Model #	Length		Price
MXP-003	3′	20 AWG, Neutrik XLRF to 1/4″ Phone	**11.95**
MXP-005	5′	24AWG, Neutrik Connectors	**12.95**
MXP-010	10′	24 AWG, Neutrik Connectors	**14.95**
MXP-015	15′	24AWG, Neutrik Connectors	**15.95**
MXP-020	20′	24 AWG, Neutrik Connectors	**17.95**
MXP-025	25′	24 AWG, Neutrik Connectors	**19.95**
MXP-030	30′	24 AWG, Neutrik Connectors	**21.95**
MXP-050	50′	24 AWG, Neutrik Connectors	**29.95**
MXP-075	75′	24 AWG, Neutrik Connectors	**34.95**
MXP-100	100′	24 AWG, Neutrik Connectors	**44.95**

GUITAR & INSTRUMENT CABLES

WOVEN SINGLES

5 Patterns (Wild and Colorful, or Dark and Sparkling)



Model #	Length		Price
3GT-18C1	18′	Cloth Woven Jacket, Blue/White/Black	**18.95**
3GT-18C2	18′	Cloth Woven Jacket, Blue/Green/White	**18.95**
3GT-18C3	18′	Cloth Woven Jacket, Red/Green	**18.95**
3GT-18C4	18′	Cloth Woven Jacket, Black/Gold	**18.95**
3GT-18C5	18′	Cloth Woven Jacket, Black/Red	**18.95**

CLASSIC CLOTH-WOVEN TWEED JACKET

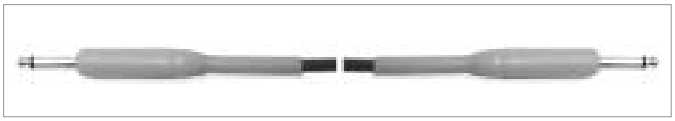
(Metal Headshells, Heat-Shrink Tubing)



Model #	Length		Price
GTR-518TWD	18′	Straight to Straight	**18.95**
GTR-518RT	18′	Straight to Right-Angle	**19.95**
GTR-518RR	18′	Right-Angle to Right-Angle	**19.95**

PRO GUITAR CABLES

(Metal Headshells, Heat-Shrink Tubing)



Model #	Length		Price
GTR-415	15′	Exterior Heat Shrink	**10.95**
GTR-420	20′	Exterior Heat Shrink	**12.95**
GTR-425	25′	Exterior Heat Shrink	**14.95**

TRADITIONAL PRO STYLE GUITAR/INSTRUMENT CABLES

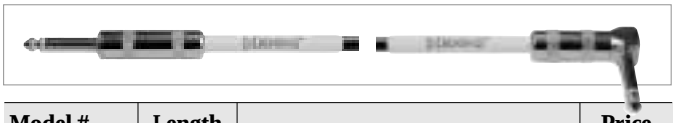
(Metal Headshells, Heat-Shrink Tubing)



Model #	Length		Price
GTR-205	5′	Traditional Pro Guitar/Instrument	**8.95**
GTR-210	10′	Traditional Pro Guitar/Instrument	**9.95**
GTR-218	18′	Traditional Pro Guitar/Instrument	**10.95**
GTR-225	25′	Traditional Pro Guitar/Instrument	**13.95**

TRADITIONAL PRO STYLE 1 OR 2 RIGHT ANGLES

(Metal Headshells, Heat-Shrink Tubing)



Model #	Length		Price
GTR-205RT	5′	Right-Angle to Straight	**8.95**
GTR-210RT	10′	Right-Angle to Straight	**9.95**
GTR-218RT	18′	Right-Angle to Straight	**10.95**
GTR-225RT	25′	Right-Angle to Straight	**13.95**
GTR-205RR	5′	Right-Angle to Right-Angle	**9.95**
GTR-215RR	10′	Right-Angle to Right-Angle	**10.95**
GTR-218RR	18′	Right-Angle to Right-Angle	**11.95**
GTR-225RR	25′	Right-Angle to Right-Angle	**14.95**

INSTRUMENT CABLES

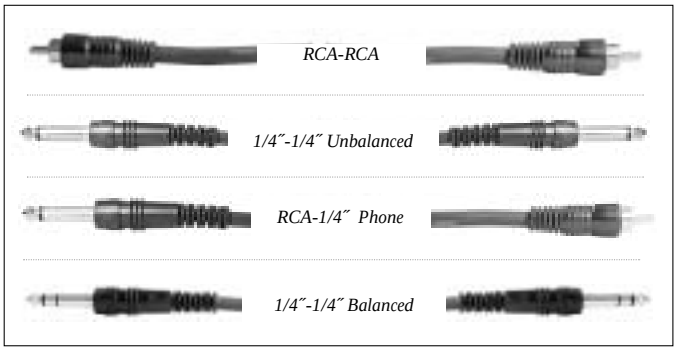
(Metal Headshells, Spring-Steel Strain Relief)



Model #	Length		Price
ISC-105BLK	5′	Black 1/4″ to 1/4″	**8.95**
ISC-110BLK	10′	Black 1/4″ to 1/4″	**9.95**
ISC-118BLK	18′	Black 1/4″ to 1/4″	**8.95**
ISC-118BLU	18′	Blue 1/4″ to 1/4″	**8.95**
ISC-125BLK	25′	Black 1/4″ to 1/4″	**11.95**
IRG-101	1′	Right Angle Plugs	**4.95**
IRG-103	3′	Right Angle Plugs	**5.95**

SINGLE & DUAL AUDIO CABLES

Hosa is world famous for precision-tooled injection-molded terminations, and with good reason. They ensure a tight, positive, reliable connection while providing excellent signal transmission at an unbelievable price. Dense spiral shielding allows flexibility and keeps out unwanted interference, for noiseless audio. Thick protective cable jacketing, your choice of balanced or unbalanced wiring, all the right connectors: Look no further for your studio or stage rig. Hosa has it all.



Single Audio Cables

RCA - RCA CABLES

Model #	Length		Price
CRA-103	3′	RCA-RCA	**3.95**
CRA-105	5′	RCA-RCA	**3.95**
CRA-110	10′	RCA-RCA	**4.95**
CRA-115	15′	RCA-RCA	**5.95**
CRA-120	20′	RCA-RCA	**7.95**

1/4″ - 1/4″ UNBALANCED CABLES

Model #	Length		Price
CPP-101	1′	1/4″ Phone - 1/4″ Phone	**3.49**
CPP-103	3′	1/4″ Phone - 1/4″ Phone	**3.95**
CPP-105	5′	1/4″ Phone - 1/4″ Phone	**4.95**
CPP-110	10′	1/4″ Phone - 1/4″ Phone	**5.95**
CPP-115	15′	1/4″ Phone - 1/4″ Phone	**6.95**

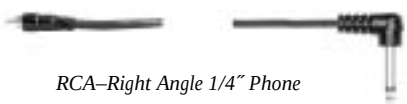
RCA - 1/4″ PHONE CABLES

Model #	Length		Price
CPR-103	3′	RCA-1/4″ Phone	**3.95**
CPR-105	5′	RCA-1/4″ Phone	**4.49**
CPR-110	10′	RCA-1/4″ Phone	**5.95**
CPR-115	15′	RCA-1/4″ Phone	**6.95**

1/4″ - 1/4″ BALANCED CABLES

CSS-103	3′	Stereo 1/4″ Phone-Stereo 1/4″ Phone	**5.95**
CSS-105	5′	Stereo 1/4″ Phone-Stereo 1/4″ Phone	**6.95**
CSS-110	10′	Stereo 1/4″ Phone-Stereo 1/4″ Phone	**7.95**
CSS-115	15′	Stereo 1/4″ Phone-Stereo 1/4″ Phone	**8.95**
CSS-125	25′	Stereo 1/4″ Phone-Stereo 1/4″ Phone	**7.95**

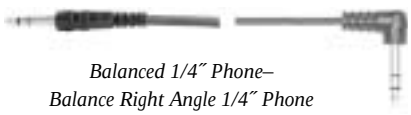
RIGHT ANGLE CABLES



RCA–Right Angle 1/4″ Phone



1/4″ Phone–Right Angle 1/4″ Phone



Balanced 1/4″ Phone–
Balance Right Angle 1/4″ Phone

RIGHT ANGLE CABLES

Model #	Length		Price
RPG-103	3′	RCA–Right Angle 1/4″ Phone	**3.49**
RPG-105	5′	RCA–Right Angle 1/4″ Phone	**3.95**
RPG-110	10′	RCA–Right Angle 1/4″ Phone	**5.95**
RPG-115	15′	RCA–Right Angle 1/4″ Phone	**6.95**
CPG-103	3′	1/4″ Phone–Right Angle 1/4″ Phone	**3.95**
CPG-105	5′	1/4″ Phone–Right Angle 1/4″ Phone	**4.49**
CPG-110	10′	1/4″ Phone–Right Angle 1/4″ Phone	**5.95**
CPG-115	15′	1/4″ Phone–Right Angle 1/4″ Phone	**6.95**
CSR-103	3′	Balanced 1/4″ Phone–Balance Right Angle 1/4″ Phone	**6.49**
CSR-105	5′	Balanced 1/4″ Phone–Balance Right Angle 1/4″ Phone	**6.95**
CSR-110	10′	Balanced 1/4″ Phone–Balance Right Angle 1/4″ Phone	**7.95**
CSR-115	15′	Balanced 1/4″ Phone–Balance Right Angle 1/4″ Phone	**8.95**

CONSUMER DUAL AUDIO CABLES

INJECTION-MOLDED HEADSHELLS IN YOUR CHOICE OF NICKEL OR GOLD CONTACTS

DUAL MOLDED NICKEL RCA - RCA CABLES



Model #	Length		Price
CRA-201	3.3′	RCA-RCA	**5.95**
CRA-202	6.6′	RCA-RCA	**6.95**
CRA-203	9.9′	RCA-RCA	**8.95**
CRA-204	13.2′	RCA-RCA	**7.95**
CRA-206	20′	RCA-RCA	**10.95**

DUAL MOLDED GOLD RCA - RCA CABLES

Model #	Length		Price
CRA-201G	3.3′	RCA-RCA	**7.49**
CRA-202G	6.6′	RCA-RCA	**8.49**
CRA-203G	9.9′	RCA-RCA	**9.95**
CRA-204G	13.2′	RCA-RCA	**8.95**

DUAL MOLDED NICKEL RCA - RCA CABLES

SAVE ON A 20-PC. BAG



Model #	Length		Price
CRA-201BAG	3.3′	1/4″-1/4″	**2.95**
CRA-202BAG	6.6′	1/4″-1/4″	**2.95**
CRA-203BAG	9.9′	1/4″-1/4″	**3.95**
CRA-204BAG	13.2′	1/4″-1/4″	**4.50**



DUAL MOLDED NICKEL RCA - 1/4″ CABLES



Model #	Length		Price
CPR-201	3.3′	RCA-1/4″	**6.95**
CPR-202	6.6′	RCA-1/4″	**7.95**
CPR-203	9.9′	RCA-1/4″	**8.95**
CPR-204	13.2′	RCA-1/4″	**8.49**
CPR-206	20′	RCA-1/4″	**10.95**

DUAL MOLDED NICKEL 1/4″ - 1/4″ CABLES



Model #	Length		Price
CPP-201	3.3′	1/4″-1/4″	**6.95**
CPP-202	6.6′	1/4″-1/4″	**7.95**
CPP-203	9.9′	1/4″-1/4″	**7.95**
CPP-204	13.2′	1/4″-1/4″	**8.95**
CPP-206	20′	1/4″-1/4″	**11.95**

DUAL MOLDED BALANCED TRS 1/4″-1/4″ CABLES

Model #	Length		Price
CSS-201	3.3′	1/4″-1/4″	**8.95**
CSS-202	6.6′	1/4″-1/4″	**9.95**
CSS-203	9.9′	1/4″-1/4″	**10.95**
CSS-204	13.2′	1/4″-1/4″	**12.95**

DUAL RIGHT ANGLE CABLES

Model #	Length		Price
CGA-301	3.3′	Dual Right Angle RCA to Right Angle RCA	**5.95**
CGA-302	6.6′	Dual Right Angle RCA to Right Angle RCA	**6.95**
CGA-303	9.9′	Dual Right Angle RCA to Right Angle RCA	**8.95**
CGA-201	3.3′	Dual Right Angle RCA to Straight RCA	**5.95**
CGA-202	6.6′	Dual Right Angle RCA to Straight RCA	**6.95**
CGA-203	9.9′	Dual Right Angle RCA to Straight RCA	**8.95**
CRG-201	3.3′	Dual, RCA (M) to RCA with Female Junction*	**6.95**
CRG-202	6.6′	Dual, RCA (M) to RCA with Female Junction*	**7.95**
CRG-203	9.9′	Dual, RCA (M) to RCA with Female Junction*	**9.95**

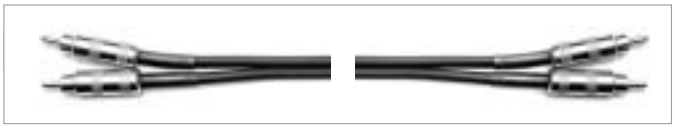


** Allows "Piggy-back" Inputs*

PROFESSIONAL DUAL AUDIO CABLES

SERVICEABLE TERMINATIONS IN GOLD OR NICKEL

PRO DUAL NICKEL-PLATED RCA - RCA CABLES



Model #	Length		Price
CRA-403	3′	RCA-RCA	7.95
CRA-405	5′	RCA-RCA	8.95
CRA-410	10′	RCA-RCA	7.95
CRA-415	15′	RCA-RCA	8.95
CRA-420	20′	RCA-RCA	9.95

PRO DUAL NICKEL-PLATED 1/4″ - 1/4″ CABLES



Model #	Length		Price
CPP-403	3′	1/4″-1/4″	8.95
CPP-405	5′	1/4″-1/4″	9.95
CPP-410	10′	1/4″-1/4″	8.95
CPP-415	15′	1/4″-1/4″	9.95
CPP-420	20′	1/4″-1/4″	11.95

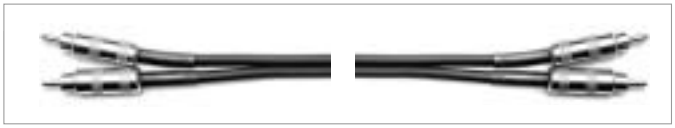
PRO DUAL GOLD PLATED RCA - RCA CABLES

Model #	Length		Price
CRA-403G	3′	RCA-RCA	7.95
CRA-405G	5′	RCA-RCA	8.95
CRA-410G	10′	RCA-RCA	9.95
CRA-415G	15′	RCA-RCA	10.95
CRA-420G	20′	RCA-RCA	11.95

PRO DUAL GOLD PLATED 1/4″ - 1/4″ CABLES

Model #	Length		Price
CPP-403G	3	1/4″-1/4″	8.95
CPP-405G	5′	1/4″-1/4″	8.95
CPP-410G	10′	1/4″-1/4″″	9.95
CPP-415G	15′	1/4″-1/4″	11.95
CPP-420G	20′	1/4″-1/4″	12.95

PRO DUAL NICKEL-PLATED RCA - RCA CABLES



Model #	Length		Price
CRA-403	3′	RCA-RCA	7.95
CRA-405	5′	RCA-RCA	8.95
CRA-410	10′	RCA-RCA	7.95
CRA-415	15′	RCA-RCA	8.95
CRA-420	20′	RCA-RCA	9.95

PRO DUAL NICKEL-PLATED BALANCED TRS 1/4″-1/4″



Model #	Length		Price
CSS-403	3′	1/4″-1/4″	9.95
CSS-405	5′	1/4″-1/4″	10.95
CSS-410	10′	1/4″-1/4″	12.95
CSS-415	15′	1/4″-1/4″	13.95
CSS-420	20′	1/4″-1/4″	14.95

PRO DUAL GOLD PLATED RCA - RCA CABLES

Model #	Length		Price
CRA-403G	3′	RCA-RCA	7.95
CRA-405G	5′	RCA-RCA	8.95
CRA-410G	10′	RCA-RCA	9.95
CRA-415G	15′	RCA-RCA	10.95
CRA-420G	20′	RCA-RCA	11.95

PRO DUAL GOLD-PLATED BALANCED TRS 1/4″-1/4″

Model #	Length		Price
CSS-403G	3	1/4″-1/4″	11.95
CSS-405G	5′	1/4″-1/4″	12.95
CSS-410G	10′	1/4″-1/4″″	14.95
CSS-415G	15′	1/4″-1/4″	15.95
CSS-420G	20′	1/4″-1/4″	16.95

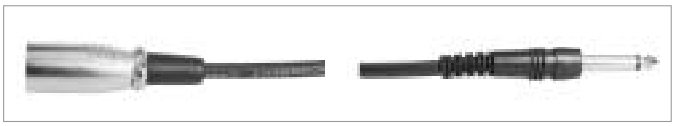
XLR CABLES — Standard

XLR (M) - RCA (M)



Model #	Length		Price
XRM-102	2′	XLR (M) to Molded RCA (M)	8.95
XRM-103	3′	XLR (M) to Molded RCA (M)	8.95
XRM-105	5′	XLR (M) to Molded RCA (M)	9.49
XRM-110	10′	XLR (M) to Molded RCA (M)	795
XRM-115	15′	XLR (M) to Molded RCA (M)	8.95
XRM-120	20′	XLR (M) to Molded RCA (M)	9.95

XLR (M) - UNBALANCED 1/4″ PHONE



Model #	Length		Price
PXM-102	2′	XLR (M) to Molded TS 1/4″ (M)	8.95
PXM-103	3′	XLR (M) to Molded TS 1/4″ (M)	9.49
PXM-105	5′	XLR (M) to Molded TS 1/4″ (M)	9.95
PXM-110	10′	XLR (M) to Molded TS 1/4″ (M)	8.95
PXM-115	15′	XLR (M) to Molded TS 1/4″ (M)	9.95
PXM-120	20′	XLR (M) to Molded TS 1/4″ (M)	10.95

BALANCED XLR (M) TO TRS 1/4″ PHONE

Model #	Length		Price
STX-102M	2′	Balanced, XLR (M) to TRS 1/4″ (M)	8.95
STX-103M	3′	Balanced, XLR (M) to TRS 1/4″ (M)	9.95
STX-105M	5′	Balanced, XLR (M) to TRS 1/4″ (M)	7.95
STX-110M	10′	Balanced, XLR (M) to TRS 1/4″ (M)	9.95
STX-115M	15′	Balanced, XLR (M) to TRS 1/4″ (M)	10.95
STX-120M	20′	Balanced, XLR (M) to TRS 1/4″ (M)	12.95

BALANCED XLR (M) TO TRS 1/4″ PHONE

Model #	Length		Price
XLR-102	2′	Balanced XLR (F) to XLR (M)	11.95
XLR-103	3′	Balanced XLR (F) to XLR (M)	12.95
XLR-105	5′	Balanced XLR (F) to XLR (M)	14.95

XLR (F) - RCA (M)

Model #	Length		Price
XRF-102	2′	XLR (F) to Molded RCA (M)	8.95
XRF-103	3′	XLR (F) to Molded RCA (M)	9.49
XRF-105	5′	XLR (F) to Molded RCA (M)	9.95
XRF-110	10′	XLR (F) to Molded RCA (M)	8.95
XRF-115	15′	XLR (F) to Molded RCA (M)	9.95
XRF-120	20′	XLR (F) to Molded RCA (M)	10.95

XLR (F) - UNBALANCED 1/4″ PHONE

Model #	Length		Price
PXF-102	2′	XLR (F) to Molded TS 1/4″ (M)	8.95
PXF-103	3′	XLR (F) to Molded TS 1/4″ (M)	9.95
PXF-105	5′	XLR (F) to Molded TS 1/4″ (M)	7.95
PXF-110	10′	XLR (F) to Molded TS 1/4″ (M)	8.95
PXF-115	15′	XLR (F) to Molded TS 1/4″ (M)	9.95
PXF-120	20′	XLR (F) to Molded TS 1/4″ (M)	10.95

BALANCED XLR (F) - TRS 1/4″ PHONE

Model #	Length		Price
STX-102F	2′	XLR (F) to Molded TRS 1/4″ (M)	7.95
STX-103F	3′	XLR (F) to Molded TRS 1/4″ (M)	7.95
STX-105F	5′	XLR (F) to Molded TRS 1/4″ (M)	8.95
STX-110F	10′	BXLR (F) to Molded TRS 1/4″ (M)	9.95
STX-115F	15′	XLR (F) to Molded TRS 1/4″ (M)	11.95
STX-120F	20′	XLR (F) to Molded TRS 1/4″ (M)	12.95

STANDARD XLR CABLES



XLR (F) - RCA (M)



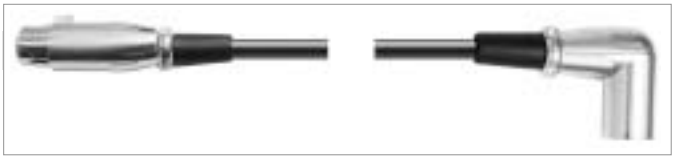
XLR (F) - Unbalanced 1/4″ Phone



Balanced XLR (F) - TRS 1/4″ Phone

XLR CABLES — Professional

BALANCED XLR - RIGHT ANGLE XLR



Model #	Length		Price
XRR-103	3′	Bal. XLR (F) to Rt. Angle XLR (M)	**9.95**
XRR-105	5′	Bal. XLR (F) to Rt. Angle XLR (M)	**10.95**
XRR-110	10′	Bal. XLR (F) to Rt. Angle XLR (M)	**12.95**
XRR-115	15′	Bal. XLR (F) to Rt. Angle XLR (M)	**13.95**

BALANCED XLR - RIGHT ANGLE XLR



Model #	Length		Price
XFF-103	3′	Bal. XLR (M) to Right Angle XLR (F)	**9.95**
XFF-105	5′	Bal. XLR (M) to Right Angle XLR (F)	**10.95**
XFF-110	10′	Bal. XLR (M) to Right Angle XLR (F)	**12.95**
XFF-115	15′	Bal. XLR (M) to Right Angle XLR (F)	**13.95**

XLR (M) - RCA (M)

Model #	Length		Price
XRM-302	2′	Pro XLR (M) to Serviceable RCA (M)	**7.95**
XRM-303	3′	Pro XLR (M) to Serviceable RCA (M)	**7.95**
XRM-305	5′	Pro XLR (M) to Serviceable RCA (M)	**8.95**
XRM-310	10′	Pro XLR (M) to Serviceable RCA (M)	**9.95**
XRM-315	15′	Pro XLR (M) to Serviceable RCA (M)	**10.95**
XRM-320	20′	Pro XLR (M) to Serviceable RCA (M)	**11.95**

XLR (F) - RCA (M)

Model #	Length		Price
XRF-302	2′	Pro XLR (F) to Serviceable RCA (M)	**8.95**
XRF-303	3′	Pro XLR (F) to Serviceable RCA (M)	**8.95**
XRF-305	5′	Pro XLR (F) to Serviceable RCA (M)	**8.95**
XRF-310	10′	Pro XLR (F) to Serviceable RCA (M)	**9.95**
XRF-315	15′	Pro XLR (F) to Serviceable RCA (M)	**10.95**
XRF-320	20′	Pro XLR (F) to Serviceable RCA (M)	**12.95**

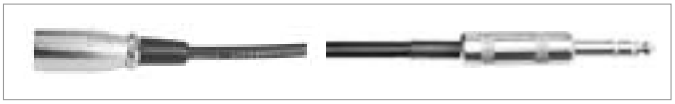
XLR - UNBALANCED TS 1/4″

Model #	Length		Price
PXM-302	2′	Pro XLR (M) to Serviceable TS 1/4″ (M)	**7.95**
PXM-303	3′	Pro XLR (M) to Serviceable TS 1/4″ (M)	**7.95**
PXM-305	5′	Pro XLR (M) to Serviceable TS 1/4″ (M)	**8.95**
PXM-310	10′	Pro XLR (M) to Serviceable TS 1/4″ (M)	**9.95**
PXM-315	15′	Pro XLR (M) to Serviceable TS 1/4″ (M)	**10.95**
PXM-320	20′	Pro XLR (M) to Serviceable TS 1/4″ (M)	**11.95**

XLR - UNBALANCED TS 1/4″

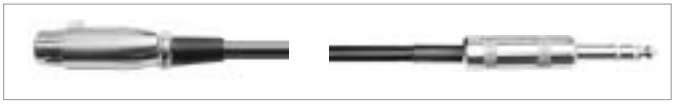
Model #	Length		Price
PXF-302	2′	Pro XLR (F) to Serviceable TS 1/4″ (M)	**7.95**
PXF-303	3′	Pro XLR (F) to Serviceable TS 1/4″ (M)	**8.95**
PXF-305	5′	Pro XLR (F) to Serviceable TS 1/4″ (M)	**8.95**
PXF-310	10′	Pro XLR (F) to Serviceable TS 1/4″ (M)	**9.95**
PXF-315	15′	Pro XLR (F) to Serviceable TS 1/4″ (M)	**10.95**
PXF-320	20′	Pro XLR (F) to Serviceable TS 1/4″ (M)	**11.95**

BALANCED XLR - TRS 1/4″



Model #	Length		Price
STX-301M	1′	Pro Bal. XLR (M) to TRS 1/4″ (M)	**8.95**
STX-302M	2′	Pro Bal. XLR (M) to TRS 1/4″ (M)	**8.95**
STX-303M	3′	Pro Bal. XLR (M) to TRS 1/4″ (M)	**9.95**
STX-305M	5′	Pro Bal. XLR (M) to TRS 1/4″ (M)	**10.95**
STX-310M	10′	Pro Bal. XLR (M) to TRS 1/4″ (M)	**11.95**
STX-315M	15′	Pro Bal. XLR (M) to TRS 1/4″ (M)	**12.95**
STX-320M	20′	Pro Bal. XLR (M) to TRS 1/4″ (M)	**14.95**

BALANCED XLR - TRS 1/4″



Model #	Length		Price
STX-301F	1′	Pro Bal. XLR (F) to TRS 1/4″ (M)	**9.95**
STX-302F	2′	Pro Bal. XLR (F) to TRS 1/4″ (M)	**9.95**
STX-303F	3′	Pro Bal. XLR (F) to TRS 1/4″ (M)	**10.95**
STX-305F	5′	Pro Bal. XLR (F) to TRS 1/4″ (M)	**10.95**
STX-310F	10′	Pro Bal. XLR (F) to TRS 1/4″ (M)	**12.95**
STX-315F	15′	Pro Bal. XLR (F) to TRS 1/4″ (M)	**13.95**
STX-320F	20′	Pro Bal. XLR (F) to TRS 1/4″ (M)	**15.95**

SPEAKER CABLES

Speakon-Equipped 2-Conductor Speaker Cables

SPEAKON - SPEAKON



Model #	Length		Price
SKT-203	3′	Speakon to Speakon, 12 AWG x 2	**29.95**
SKT-205	5′	Speakon to Speakon, 12 AWG x 2	**29.95**
SKT-210	10′	Speakon to Speakon, 12 AWG x 2	**34.95**
SKT-215	15′	Speakon to Speakon, 12 AWG x 2	**37.95**
SKT-225	25′	Speakon to Speakon, 12 AWG x 2	**44.95**
SKT-250	50′	Speakon to Speakon, 12 AWG x 2	**64.95**

SPEAKON - BANANA



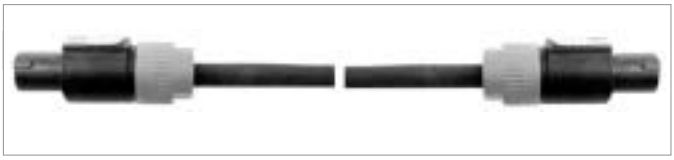
Model #	Length		Price
SKT-203BN	3′	Speakon to Banana, 12 AWG x 2	**22.95**
SKT-205BN	5′	Speakon to Banana, 12 AWG x 2	**24.95**
SKT-210BN	10′	Speakon to Banana, 12 AWG x 2	**29.95**
SKT-215BN	15′	Speakon to Banana, 12 AWG x 2	**34.95**
SKT-225BN	25′	Speakon to Banana, 12 AWG x 2	**39.95**
SKT-250BN	50′	Speakon to Banana, 12 AWG x 2	**59.95**
SKT-2100BN	100′	Speakon to Banana, 12 AWG x 2	**99.95**

SPEAKON - 1/4″

Model #	Length		Price
SKT-203Q	3′	Speakon to 1/4″ Phone , 12 AWG x 2	**22.95**
SKT-205Q	5′	Speakon to 1/4″ Phone, 12 AWG x 2	**24.95**
SKT-210Q	10′	Speakon to 1/4″ Phone, 12 AWG x 2	**29.95**
SKT-215Q	15′	Speakon to 1/4″ Phone, 12 AWG x 2	**34.95**
SKT-225Q	25′	Speakon to 1/4″ Phone, 12 AWG x 2	**39.95**
SKT-250Q	50′	Speakon to 1/4″ Phone, 12 AWG x 2	**59.95**
SKT-2100Q	100′	Speakon to 1/4″ Phone, 12 AWG x 2	**99.95**

Speakon-Equipped 4-Conductor Speaker Cables (Suitable for Biamping)

SPEAKON - SPEAKON



Model #	Length		Price
SKO-403	3′	Speakon to Speakon, 14 AWG x 4	**29.95**
SKO-405	5′	Speakon to Speakon, 14 AWG x 4	**29.95**
SKO-410	10′	Speakon to Speakon, 14 AWG x 4	**34.95**
SKO-415	15′	Speakon to Speakon, 14 AWG x 4	**37.95**
SKO-425	25′	Speakon to Speakon, 14 AWG x 4	**44.95**
SKO-450	50′	Speakon to Speakon, 14 AWG x 4	**64.95**
SKO-475	75′	Speakon to Speakon, 14 AWG x 4	**84.95**
SKO-4100	100′	Speakon to Speakon, 14 AWG x 4	**109.95**

SPEAKON - BANANA

Model #	Length		Price
SKO-403BN	3′	Speakon to 2 Banana 14 AWG x 4	**29.95**
SKO-405BN	5′	Speakon to 2 Banana 14 AWG x 4	**29.95**
SKO-410BN	10′	Speakon to 2 Banana 14 AWG x 4	**34.95**
SKO-415BN	15′	Speakon to 2 Banana 14 AWG x 4	**37.95**
SKO-420BN	20′	Speakon to 2 Banana 14 AWG x 4	**39.95**
SKO-425BN	25′	Speakon to 2 Banana 14 AWG x 4	**44.95**
SKO-450BN	50′	Speakon to 2 Banana 14 AWG x 4	**64.95**
SKO-475BN	75′	Speakon to 2 Banana 14 AWG x 4	**89.95**
SKO-4100BN	100′	Speakon to 2 Banana 14 AWG x 4	**109.95**

SPEAKON - 1/4″ PHONE

Model #	Length		Price
SKO-403Q	3′	Speakon to Two 1/4″, 14 AWG x 4	**29.95**
SKO-405Q	5′	Speakon to Two 1/4″, 14 AWG x 4	**29.95**
SKO-410Q	10′	Speakon to Two 1/4″, 14 AWG x 4	**34.95**
SKO-415Q	15′	Speakon to Two 1/4″, 14 AWG x 4	**37.95**
SKO-420Q	20′	Speakon to Two 1/4″, 14 AWG x 4	**39.95**
SKO-425Q	25′	Speakon to Two 1/4″, 14 AWG x 4	**44.95**
SKO-450Q	50′	Speakon to Two 1/4″, 14 AWG x 4	**64.95**
SKO-475Q	75′	Speakon to Two 1/4″, 14 AWG x 4	**89.95**
SKO-4100Q	100′	Speakon to Two 1/4″, 14 AWG x 4	**109.95**

B&H PHOTO - VIDEO - PRO AUDIO

SPEAKER CABLES

Clear Jacket - Lets the Copper Shine Through

1/4″ - 1/4″ (JUMBO HEADSHELL 1/4″ PHONE PLUGS)

Model #	Length		Price
SKM-203	3′	Clear Jacket 1/4″ to 1/4″, 12 AWG x 2	**11.95**
SKM-205	5′	Clear Jacket 1/4″ to 1/4″, 12 AWG x 2	**12.95**
SKM-210	10′	Clear Jacket 1/4″ to 1/4″, 12 AWG x 2	**1595**
SKM-215	15′	Clear Jacket 1/4″ to 1/4″, 12 AWG x 2	**19.95**
SKM-220	20′	Clear Jacket 1/4″ to 1/4″, 12 AWG x 2	**22.95**
SKM-225	25′	Clear Jacket 1/4″ to 1/4″, 12 AWG x 2	**25.95**
SKM-230	30′	Clear Jacket 1/4″ to 1/4″, 12 AWG x 2	**29.95**
SKM-250	50′	Clear Jacket 1/4″ to 1/4″, 12 AWG x 2	**39.95**
SKM-275	75′	Clear Jacket 1/4″ to 1/4″, 12 AWG x 2	**54.95**
SKM-2100	100′	Clear Jacket 1/4″ to 1/4″, 12 AWG x 2	**69.95**

1/4″-1/4″ (STANDARD HEADSHELL 1/4″ PHONE PLUGS)

Model #	Length		Price
SKM-603	3′	Clear Jacket 1/4″ to 1/4″, 16 AWG x 2	**9.95**
SKM-605	5′	Clear Jacket 1/4″ to 1/4″, 16 AWG x 2	**7.95**
SKM-610	10′	Clear Jacket 1/4″ to 1/4″, 16 AWG x 2	**9.95**
SKM-615	15′	Clear Jacket 1/4″ to 1/4″, 16 AWG x 2	**10.95**
SKM-620	20′	Clear Jacket 1/4″ to 1/4″, 16 AWG x 2	**12.95**
SKM-625	25′	Clear Jacket 1/4″ to 1/4″, 16 AWG x 2	**13.95**
SKM-630	30′	Clear Jacket 1/4″ to 1/4″, 16 AWG x 2	**14.95**
SKM-650	50′	Clear Jacket 1/4″ to 1/4″, 16 AWG x 2	**21.95**
SKM-675	75′	Clear Jacket 1/4″ to 1/4″, 16 AWG x 2	**29.95**
SKM-6100	100′	Clear Jacket 1/4″ to 1/4″, 16 AWG x 2	**39.95**

BLACK "ZIP CORD"

Model #	Length		Price
SKZ-603	3′	1/4″ to 1/4″, 16 AWG x 2	**9.95**
SKZ-605	5′	1/4″ to 1/4″, 16 AWG x 2	**7.95**
SKZ-610	10′	1/4″ to 1/4″, 16 AWG x 2	**8.95**
SKZ-615	15′	1/4″ to 1/4″, 16 AWG x 2	**9.95**
SKZ-620	20′	1/4″ to 1/4″, 16 AWG x 2	**10.95**
SKZ-625	25′	1/4″ to 1/4″, 16 AWG x 2	**11.95**
SKZ-630	30′	1/4″ to 1/4″, 16 AWG x 2	**13.95**
SKZ-650	50′	1/4″ to 1/4″, 16 AWG x 2	**18.95**
SKZ-675	75′	1/4″ to 1/4″, 16 AWG x 2	**22.95**
SKZ-6100	100′	1/4″ to 1/4″, 16 AWG x 2	**29.95**

12-AWG Tough Jacket

1/4″ - 1/4″ (JUMBO HEADSHELL 1/4″ PHONE PLUGS)

Model #	Length		Price
SKJ-203	3′	Rugged Jacket 1/4″ to 1/4″, 12 AWG x 2	**14.95**
SKJ-205	5′	Rugged Jacket 1/4″ to 1/4″, 12 AWG x 2	**15.95**
SKJ-210	10′	Rugged Jacket 1/4″ to 1/4″, 12 AWG x 2	**19.95**
SKJ-215	15′	Rugged Jacket 1/4″ to 1/4″, 12 AWG x 2	**24.95**
SKJ-220	20′	Rugged Jacket 1/4″ to 1/4″, 12 AWG x 2	**27.95**
SKJ-225	25′	Rugged Jacket 1/4″ to 1/4″, 12 AWG x 2	**32.95**
SKJ-230	30′	Rugged Jacket 1/4″ to 1/4″, 12 AWG x 2	**3495**
SKJ-250	50′	Rugged Jacket 1/4″ to 1/4″, 12 AWG x 2	**49.95**
SKJ-275	75′	Rugged Jacket 1/4″ to 1/4″, 12 AWG x 2	**69.95**
SKJ-2100	100′	Rugged Jacket 1/4″ to 1/4″, 12 AWG x 2	**84.95**

14-AWG Tough Jacket

1/4″ - 1/4″ (STANDARD HEADSHELL 1/4″ PHONE PLUGS)

Model #	Length		Price
SKJ-403	3′	Rugged Jacket 1/4″ to 1/4″, 14 AWG x 2	**13.95**
SKJ-405	5′	Rugged Jacket 1/4″ to 1/4″, 14 AWG x 2	**14.95**
SKJ-410	10′	Rugged Jacket 1/4″ to 1/4″, 14 AWG x 2	**19.95**
SKJ-415	15′	Rugged Jacket 1/4″ to 1/4″, 14 AWG x 2	**21.95**
SKJ-420	20′	Rugged Jacket 1/4″ to 1/4″, 14 AWG x 2	**24.95**
SKJ-425	25′	Rugged Jacket 1/4″ to 1/4″, 14 AWG x 2	**29.95**
SKJ-430	30′	Rugged Jacket 1/4″ to 1/4″, 14 AWG x 2	**29.95**
SKJ-450	50′	Rugged Jacket 1/4″ to 1/4″, 14 AWG x 2	**44.95**
SKJ-475	75′	Rugged Jacket 1/4″ to 1/4″, 14 AWG x 2	**59.95**
SKJ-4100	100′	Rugged Jacket 1/4″ to 1/4″, 14 AWG x 2	**6495**
SKJ-4150	150′	Rugged Jacket 1/4″ to 1/4″, 14 AWG x 2	**Call**

16-AWG Tough Jacket

1/4″ - 1/4″ (STANDARD HEADSHELL 1/4″ PHONE PLUGS)

Model #	Length		Price
SKJ-603	3′	Rugged Jacket 1/4″ to 1/4″, 14 AWG x 2	**10.95**
SKJ-605	5′	Rugged Jacket 1/4″ to 1/4″, 14 AWG x 2	**11.95**
SKJ-610	10′	Rugged Jacket 1/4″ to 1/4″, 14 AWG x 2	**14.95**
SKJ-615	15′	Rugged Jacket 1/4″ to 1/4″, 14 AWG x 2	**15.95**
SKJ-620	20′	Rugged Jacket 1/4″ to 1/4″, 14 AWG x 2	**18.95**
SKJ-625	25′	Rugged Jacket 1/4″ to 1/4″, 14 AWG x 2	**19.95**
SKJ-630	30′	Rugged Jacket 1/4″ to 1/4″, 14 AWG x 2	**22.95**
SKJ-650	50′	Rugged Jacket 1/4″ to 1/4″, 14 AWG x 2	**29.95**
SKJ-675	75′	Rugged Jacket 1/4″ to 1/4″, 14 AWG x 2	**39.95**
SKJ-6100	100′	Rugged Jacket 1/4″ to 1/4″, 14 AWG x 2	**49.95**

MIDI CABLES

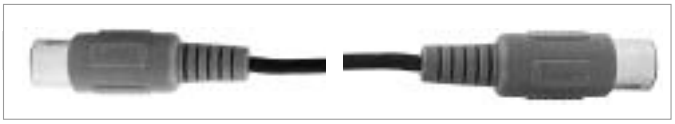
All Hosa MIDI cables are wired 5-pin discrete (all five pins are connected). Perfect for all MIDI transmissions, including Sysex.

PREMIUM MIDI CABLES (BLACK)



Model #	Length		Price
MID-403	3′	Premium MIDI Cable, Metal Plugs	**9.95**
MID-405	5′	Premium MIDI Cable, Metal Plugs	**7.95**
MID-410	10′	Premium MIDI Cable, Metal Plugs	**9.95**
MID-415	15′	Premium MIDI Cable, Metal Plugs	**10.95**
MID-420	20′	Premium MIDI Cable, Metal Plugs	**12.95**
MID-425	25′	Premium MIDI Cable, Metal Plugs	**14.95**

STANDARD MIDI CABLES*



Model #	Length		Price
MID-301	1′	Standard Molded MIDI Cable, Black	**3.95**
MID-303	3′	Standard Molded MIDI Cable, Black	**4.95**
MID-305	5′	Standard Molded MIDI Cable, Black	**5.95**
MID-310	10′	Standard Molded MIDI Cable, Black	**7.95**
MID-315	15′	Standard Molded MIDI Cable, Black	**9.95**
MID-320	20′	Standard Molded MIDI Cable, Black	**8.95**
MID-325	25′	Standard Molded MIDI Cable, Black	**10.95**

** Cables are also available in Red, Blue, Orange and White. Order colors by adding a three-letter suffix. Example: MID-310RED, MID-310ORG, MID-310BLU, MID-310WHT*

DUAL MIDI CABLES*



Model #	Length		Price
MID-201	3.3′	Dual Molded MIDI Cable	**7.95**
MID-202	6.6′	Dual Molded MIDI Cable	**9.95**
MID-203	9.9′	Dual Molded MIDI Cable	**10.95**
MID-204	13.2′	Dual Molded MIDI Cable	**12.95**

** Black only with color-coded ends*

RIGHT-ANGLE MIDI CABLES (BLACK)



Model #	Length		Price
MDR-103	3′	Molded MIDI cable, right-angle plugs	**4.95**
MDR-105	5′	Molded MIDI cable, right-angle plugs	**5.95**
MDR-110	10′	Molded MIDI cable, right-angle plugs	**7.95**
MDR-115	15′	Molded MIDI cable, right-angle plugs	**9.95**



MID-812
8-element MIDI snake

MIDI SNAKE & COILED MIDI CABLE

Model #	Length		Price
MID-812	12′	MIDI Snake, 8-Element, Color Coded	**49.95**
MID-325C	25′	Coiled MIDI Cable, Black	**14.95**



MID-325C
Coiled MIDI cable

FIBER OPTIC & DIGITAL CABLES

Fiber Optic Cables - Use with ADATs and other Toslink-equipped recording gear

OPM-SERIES PREMIUM TOSLINK CABLES



Model #	Length		Price
OPM-303	3′	Toslink-Toslink Fiber Optic Cable	**14.95**
OPM-305	5′	Toslink-Toslink Fiber Optic Cable	**16.95**
OPM-310	10′	Toslink-Toslink Fiber Optic Cable	**24.95**
OPM-315	15′	Toslink-Toslink Fiber Optic Cable	**27.95**
OPM-320	20′	Toslink-Toslink Fiber Optic Cable	**32.95**
OPM-330	30′	Toslink-Toslink Fiber Optic Cable	**39.95**
OPM-350	50′	Toslink-Toslink Fiber Optic Cable	**54.95**

OPT-SERIES STANDARD TOSLINK CABLES



Model #	Length		Price
OPT-102	2′	Toslink-Toslink Fiber Optic Cable	**7.95**
OPT-103	3′	Toslink-Toslink Fiber Optic Cable	**8.95**
OPT-106	6′	Toslink-Toslink Fiber Optic Cable	**11.95**
OPT-110	10′	Toslink-Toslink Fiber Optic Cable	**17.95**
OPT-113	13′	Toslink-Toslink Fiber Optic Cable	**19.95**
OPT-117	17′	Toslink-Toslink Fiber Optic Cable	**24.95**
OPT-130	30′	Toslink-Toslink Fiber Optic Cable	**29.95**
OPT-150	50′	Toslink-Toslink Fiber Optic Cable	**39.95**

OPQ-SERIES TOSLINK - SONY-TYPE CABLES

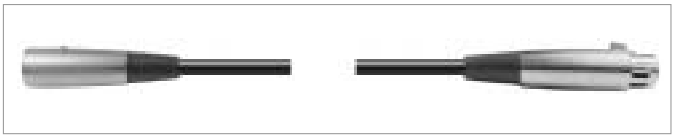
Model #	Length		Price
OPQ-205	5′	Toslink to 3.5mm Fiber Optic Cable	**11.95**
OPQ-210	10′	Toslink to 3.5mm Fiber Optic Cable	**17.95**
OPQ-215	15′	Toslink to 3.5mm Fiber Optic Cable	**22.95**

OMM-SERIES 3.5MM - 3.5MM CABLES

Model #	Length		Price
OMM-305	5′	3.5mm to 3.5mm Fiber Optic Cable	**16.95**
OMM-310	10′	3.5mm to 3.5mm Fiber Optic Cable	**24.95**
OMM-315	15′	3.5mm to 3.5mm Fiber Optic Cable	**29.95**

Digital Cables on Belden 110Ω Cable

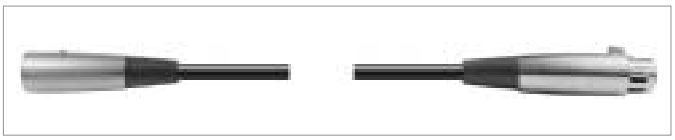
PREMIUM AES/EBU CABLES



Model #	Length		Price
AES-003	3′	AES/EBU, XLRM to XLRF	**14.95**
AES-005	5′	AES/EBU, XLRM to XLRF	**15.95**
AES-010	10′	AES/EBU, XLRM to XLRF	**21.95**
AES-015	15′	AES/EBU, XLRM to XLRF	**26.95**
AES-020	20′	AES/EBU, XLRM to XLRF	**32.95**

Digital Cables on Hosa 110Ω Cable

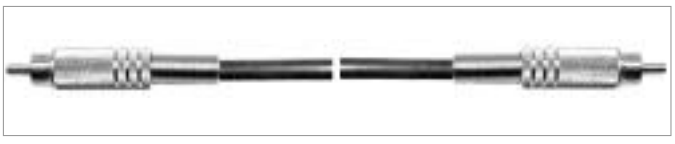
BUDGET AES/EBU CABLES



Model #	Length		Price
AES-003	3′	AES/EBU, XLRM to XLRF	**14.95**
AES-005	5′	AES/EBU, XLRM to XLRF	**16.95**
AES-010	10′	AES/EBU, XLRM to XLRF	**21.95**
AES-015	15′	AES/EBU, XLRM to XLRF	**26.95**
ES-020	20′	AES/EBU, XLRM to XLRF	**32.95**

Digital Cables with Gold-Plated RCAs

S/PDIF 75Ω COAX CABLES



Model #	Length		Price
DRA-501	3.3′	Double-Shielded, Gold-Plated RCAs	**9.95**
DRA-502	6.6′	Double-Shielded, Gold-Plated RCAs	**11.95**
DRA-503	9.9′	Double-Shielded, Gold-Plated RCAs	**13.95**
DRA-504	13.2′	Double-Shielded, Gold-Plated RCAs	**15.95**
DRA-506	20′	Double-Shielded, Gold-Plated RCAs	**17.95**

8-CHANNEL AUDIO SNAKES

BALANCED XLR - XLR

SERVICEABLE TERMINATIONS, NUMBERED



Model #	Length		Price
XLR-803	9.9′	XLR (M) to XLR (F)	**84.95**
XLR-805	16.5′	XLR (M) to XLR (F)	**99.95**
XLR-807	23′	XLR (M) to XLR (F)	**109.95**

PREMIUM* UNBALANCED 1/4″ PHONE - XLR

SERVICEABLE TERMINATIONS, NUMBERED



Model #	Length		Price
PXM-803MTL	9.9′	1/4″ Phone (TS) to XLR (M)	**54.95**
PXM-805MTL	16.5′	1/4″ Phone (TS) to XLR (M)	**64.95**
PXF-803MTL	9.9′	1/4″ Phone (TS) to XLR (F)	**64.95**
PXF-805MTL	16.5′	1/4″ Phone (TS) to XLR (F)	**69.95**

PREMIUM* UNBALANCED RCA - XLR

SERVICEABLE TERMINATIONS, NUMBERED



Model #	Length		Price
XRM-803MTL	9.9′	RCA to XLR (M)	**54.95**
XRM-805MTL	16.5′	RCA to XLR (M)	**64.95**
XRF-803MTL	9.9′	RCA to XLR (F)	**64.95**
XRF-805MTL	16.5′	RCA to XLR (F)	**74.95**

**Exclusive rubber-grip 1/4″ phone plug, ultra-flexible snake cable*

BALANCED 1/4″ (TRS) PHONE - XLR

MOLDED 1/4″ TERMINATIONS, SERVICEABLE XLRS, COLOR CODED



Model #	Length		Price
STX-802M	6.6′	1/4″ Phone (TRS) to XLR (M)	**54.95**
STX-803M	9.9′	1/4″ Phone (TRS) to XLR (M)	**59.95**
STX-804M	13.2′	1/4″ Phone (TRS) to XLR (M)	**64.95**
STX-805M	16.5′	1/4″ Phone (TRS) to XLR (M)	**74.95**
STX-807M	23′	1/4″ Phone (TRS) to XLR (M)	**84.95**
STX-802F	6.6′	1/4″ Phone (TRS) to XLR (F)	**59.95**
STX-803F	9.9′	1/4″ Phone (TRS) to XLR (F)	**64.95**
STX-804F	13.2′	1/4″ Phone (TRS) to XLR (F)	**69.95**
STX-805F	16.5′	1/4″ Phone (TRS) to XLR (F)	**74.95**
STX-807F	23′	1/4″ Phone (TRS) to XLR (F)	**89.95**

BUDGET UNBALANCED 1/4″ PHONE - XLR

MOLDED 1/4″ TERMINATIONS, SERVICEABLE XLRS, COLOR CODED

PXM-803	6.6′	1/4″ Phone (TS) to XLR (M)	**49.95**
PXM-805	6.6′	1/4″ Phone (TS) to XLR (M)	**54.95**
PXF-803	6.6′	1/4″ Phone (TS) to XLR (F)	**64.95**
PXF-805	6.6′	1/4″ Phone (TS) to XLR (F)	**69.95**

BUDGET UNBALANCED RCA- XLR

MOLDED RCA TERMINATIONS, COLOR CODED



Model #	Length		Price
XRM-803MTL	9.9′	RCA to XLR (M)	**54.95**
XRM-805MTL	16.5′	RCA to XLR (M)	**64.95**
XRF-803MTL	9.9′	RCA to XLR (F)	**64.95**
XRF-805MTL	16.5′	RCA to XLR (F)	**74.95**

8-CHANNEL AUDIO SNAKES

PREMIUM* BALANCED 1/4″ TRS - 1/4″ TRS

SERVICEABLE TERMINATIONS, NUMBERED



Model #	Length		Price
CSS-802MTL	6.6′	1/4″ Phone to 1/4″ Phone	**49.95**
CSS-803MTL	9.9′	1/4″ Phone to 1/4″ Phone	**54.95**
CSS-804MTL	13.2′	1/4″ Phone to 1/4″ Phone	**59.95**

BUDGET BALANCED 1/4″ TRS - 1/4″ TRS

MOLDED TERMINATIONS, COLOR CODED

CSS-801	3.3′	1/4″ Phone to 1/4″ Phone	**34.95**
CSS-802	6.6′	1/4″ Phone to 1/4″ Phone	**39.95**
CSS-803	9.9′	1/4″ Phone to 1/4″ Phone	**39.95**
CSS-804	13.2′	1/4″ Phone to 1/4″ Phone	**49.95**
CSS-805	16.5′	1/4″ Phone to 1/4″ Phone	**54.95**
CSS-807	21′	1/4″ Phone to 1/4″ Phone	**64.95**

PREMIUM* UNBALANCED 1/4″ (TS) - 1.4″ (TS)

SERVICEABLE TERMINATIONS, NUMBERED



CPP-802MTL	6.6′	1/4″ Phone to 1/4″ Phone	**49.95**
CPP-803MTL	9.9′	1/4″ Phone to 1/4″ Phone	**49.95**
CPP-804MTL	13.2′	1/4″ Phone to 1/4″ Phone	**54.95**

BUDGET UNBALANCED 1/4″ (TS) - 1.4″ (TS)

MOLDED TERMINATIONS, COLOR CODED

CPP-801	3.3′	1/4″ Phone to 1/4″ Phone	**22.95**
CPP-802	6.6′	1/4″ Phone to 1/4″ Phone	**29.95**
CPP-803	9.9′	1/4″ Phone to 1/4″ Phone	**34.95**
CPP-804	13.2′	1/4″ Phone to 1/4″ Phone	**34.95**
CPP-805	16.5′	1/4″ Phone to 1/4″ Phone	**39.95**
CPP-807	23′	1/4″ Phone to 1/4″ Phone	**59.95**

BUDGET UNBALANCED RT ANGLE 1/4″ (TS) - 1/4″ (TS)

MOLDED TERMINATIONS, COLOR CODED



Model #	Length		Price
CPG-802	6.6′	Rt. Angle 1/4″ Phone to Straight	**29.95**
CPG-803	9.9′	Rt. Angle 1/4″ Phone to Straight	**34.95**
CPG-804	13.2′	Rt. Angle 1/4″ Phone to Straight	**34.95**

PREMIUM* UNBALANCED 1/4″ (TS) - RCA

SERVICEABLE TERMINATIONS, NUMBERED

Model #	Length		Price
CPR-802MTL	6.6′	1/4″ Phone (TS) to RCA	**44.95**
CPR-803MTL	9.9′	1/4″ Phone (TS) to RCA	**49.95**
CPR-804MTL	13.2′	1/4″ Phone (TS) to RCA	**54.95**

BUDGET UNBALANCED 1/4″ (TS) - RCA

MOLDED TERMINATIONS, COLOR CODED

CPR-802	6.6′	1/4″ Phone (TS) to RCA	**24.95**
CPR-803	9.9′	1/4″ Phone (TS) to RCA	**29.95**
CPR-804	13.2′	/4″ Phone (TS) to RCA	**34.95**
CPR-805	16.5′	1/4″ Phone (TS) to RCA	**39.95**
CPR-807	23′	1/4″ Phone (TS) to RCA	**54.95**

PREMIUM* UNBALANCED RCA - RCA

SERVICEABLE TERMINATIONS

CRA-802MTL	6.6′	RCA to RCA	**44.95**
CRA-803MTL	9.9′	RCA to RCA	**44.95**
CRA-804MTL	13.2′	RCA to RCA	**49.95**

BUDGET UNBALANCED RCA - RCA

MOLDED TERMINATIONS

CRA-802	6.6′	8-Ch. Snake, RCA to RCA	**23.95**
CRA-803	9.9′	8-Ch. Snake, RCA to RCA	**27.95**
CRA-804	13.2′	8-Ch. Snake, RCA to RCA	**34.95**
CRA-805	16.5′	8-Ch. Snake, RCA to RCA	**34.95**
CRA-807	23′	8-Ch. Snake, RCA to RCA	**49.95**

AUDIO CABLES

Cables-the Final Link

Monster Cable's legendary microphone and musical instrument cables, found in many of the world's top recording studios, are used religiously by engineers, producers and musicians alike. In fact, Monster's cables are acknowledged as the reference standard for some of today's hottest artists. These professionals recognize that the quality and sound of their cables play a crucial role in delivering the very best possible sound reproduction. And in the highly competitive world of music, Monster gives the artist and recording engineer that extra edge to turn an ordinary performance into something extraordinary.



How Monster Cables Maximize the Sonic Performance of Musicians and Recording Professionals

ENGINEERED FOR BETTER SOUND

While other cables can smother sound, increase distortion and flatten bass, Prolink cables utilize unique patented technologies such as multi-gauge Bandwidth Balanced wire networks and Time Correct windings to transfer complex music signals between components with greater accuracy and proper phase alignment.



Monster's MicroFiber dielectric employs an intricate process of winding an air-core fiber around each individual conductor to reduce dielectric losses and electromagnetic distortions. MicroFiber increases dynamic range, extending high frequencies and reducing background noise.

In addition, Monster's MultiTwist construction, ultra-dense shielding and conductive polymer dielectrics are used to eliminate hum, interference and handling noise. Wider frequency response and more natural reproduction of harmonics are just some of the improvements musicians discover with Monster's Prolink microphone and studio interconnect cables.

SNAKE, PATCHBAY, SPEAKER AND INTERCONNECT CABLES FOR A COMPLETE HOOK-UP

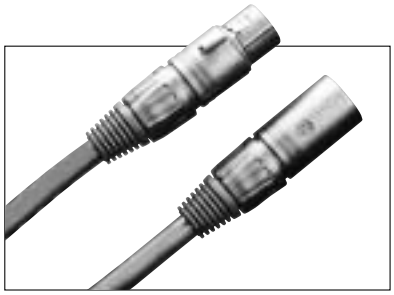
In addition to their outstanding mic and instrument cables, the ProLink line features high performance snakes, patchbay cables and speaker cables, as well as line-level audio, video and digital interconnects for every possible hook-up. Monster recognizes the importance of a high integrity, long lasting connection. Their connectors are designed for maximum signal transfer and the ability to withstand and the rigors of stage and studio environments. Monster's ProLink cables deliver sonic refinements to improve the quality of every note you play, every track you lay down, and every recording you monitor. Plug in and listen. You'll hear the difference with Monster Cable

PROLINK MICROPHONE CABLES

Standard 100 Microphone Cable

An economical introduction to Monster high performance microphone cables, the high clarity Standard 100 is an entry-level microphone cable that delivers low noise and exceptional performance.

STANDARD 100			
Model #	**Length**		**Price**
S100-M-10	10′	precision XLRs	**21.95**
S100-M-15	15′	precision XLRs	**24.95**
S100-M-20	20′	precision XLRs	**29.95**
S100-M-30	30′	precision XLRs	**39.95**
S100-M-50	50′	precision XLRs	**59.95**



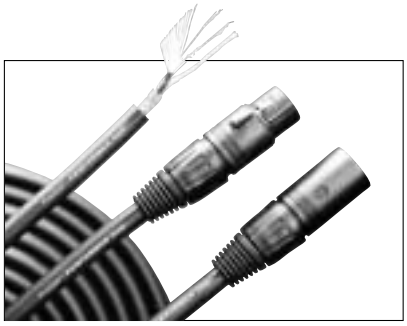
- Special Prolink windings engineered for low noise and improved signal transfer at a budget price
- Basic Bandwidth Balanced (twisted-pair) construction for clarity, low noise and smooth frequency response.
- High density (87% coverage) copper-braided shield reduces outside RFI (interference) and and EMI (hum) for a lower noise floor and increased resolution
- Duraflex jacket offers maximum flexibility, durability and reliability—performance after performance
- Heavy duty black Neutrik XLR connectors provide durability with a professional look

Performer 500 Microphone Cable

A high quality, high resolution professional mic cable, the Performer 500 offers clear, dynamic sound for recording and professional performance applications on a budget.

- Two Time Correct multiple gauge (high and low frequency) wire networks for even frequency response and natural sound reproduction
- Bandwidth Balanced construction for controlled impedance and optimum signal transfer
- MicroFiber dielectric increases clarity and transient response time
- High density (95% copper) braided shield reduces outside interference

PERFORMER 500			
Model #	**Length**		**Price**
P500-M-5	5′	gold contact XLRs	**29.95**
P500-M-10	10′	gold contact XLRs	**39.95**
P500-M-15	15′	gold contact XLRs	**49.95**
P500-M-20	20′	gold contact XLRs	**59.95**
P500-M-30	30′	gold contact XLRs	**79.95**
P500-M-50	50′	gold contact XLRs	**119.95**
P500-M-100	100′	gold contact XLRs	**219.95**
P500-M-250	250′	bulk spool	**2.00 ft.**

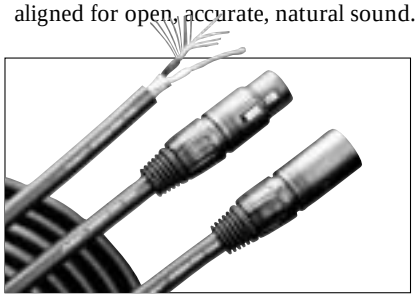


- Duraflex jacket offers maximum flexibility and durability
- Carbon-infused polymer shielding around conductors minimizes handling noise
- Heavy duty black Neutrik XLRs provide durability and a professional look

Studio Pro 1000 Microphone Cable

Studio reference, ultra high resolution microphone cable for professional recording and performance applications.

- Three Time Correct multiple gauge networks for extreme accuracy in imaging, depth and soundstage. Signals are phased-aligned for open, accurate, natural sound.



- Bandwidth Balanced construction for controlled impedance and optimum signal transfer.
- MicroFiber dielectric reduces high frequency loss and increases transient response time
- MultiTwist™ construction with dense 97% copper braided shielding to reduce the effects of EM and RF interference.
- Carbon-infused polymer minimizes handling and vibration noise.
- Heavy duty black Neutrik XLRs with 24k gold contacts.

STUDIO PRO 1000			
Model #	**Length**		**Price**
SP1000-M-5	5′	gold contact XLRs	**49.95**
SP1000-M-10	10′	gold contact XLRs	**74.95**
SP1000-M-15	15′	gold contact XLRs	**99.95**
SP1000-M-20	20′	gold contact XLRs	**129.95**
SP1000-M-30	30′	gold contact XLRs	**189.95**
SP1000-M-50	50′	gold contact XLRs	**299.95**
SP1000-M-250	250′	bulk spool	**6.00 ft**

INSTRUMENT CABLES

Performer 100 Series Instrument Cables

Maximize your playing style with high performance, entry level Performer 100 Series instrument cables. Priced like ordinary instrument cables, the affordable Performer 100 Series offers improved durability, dynamic range, and sonic quality over ordinary instrument cables.

- Precision coaxial design and heavy gauge construction for accurate frequency response and wide dynamic range
- Dense 90% coverage spiral and carbon-infused insulating shields reject RFI and EMI for a lower noise floor and increased resolution
- Carbon-infused dielectric reduces microphonic pops and cracks for a noise-free cable design.
- IsoTec vibration technology further reduces handling noise
- Tight-gripping strain relief to hold cable securely
- Extra-flexible Duraflex outer jacket for superior reliability and cut resistance — performance after performance
- For increased conductivity they incorporate durable and corrosion-resistant 24k hard gold contact 1/4″ connectors

100/500/1000 SERIES STRAIGHT 1/4″ PLUGS



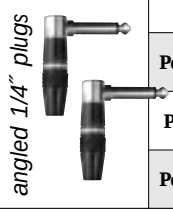
	8″	18″	3′	6′	12′	21′	30′
Performer 100	S100-I-.75 **9.95**	S100-I-1.5 **12.95**	S100-I-3 **14.95**	S100-I-6 **16.95**	S100-I-12 **19.95**	S100-I-21 **24.95**	
Performer 500 (Jazz)		P500-IJ-1.5 **27.95**	P500-IJ-3 **29.95**	P500-IJ-6 **39.95**	P500-IJ-12 **44.95**	P500-IJ-21 **59.95**	P500-IJ-30 **74.95**
Performer 500 (Rock)	P500-IR-.75 **19.95**	P500-IR-1.5 **24.95**	P500-IR-3 **29.95**	P500-IR-6 **34.95**	P500-IR-12 **39.95**	P500-IR-21 **49.95**	P500-IR-30 **59.95**
Performer 500 (Bass)	P500-IB-.75 **19.95**	P500-IB-1.5 **24.95**	P500-IB-3 **29.95**	P500-IB-6 **34.95**	P500-IB-12 **39.95**	P500-IB-21 **49.95**	P500-IB-30 **59.95**
Studio Pro 1000	SP1000-I-.75 **29.95**	SP1000-I-1.5 **39.95**	SP1000-I-3 **49.95**	SP1000-I-6 **69.95**	SP1000-I-12 **99.95**	SP1000-I-21 **149.95**	SP1000-I-30 **199.95**

100/500/1000 SERIES ANGLED TO STRAIGHT 1/4″ PLUGS



	12′	21′	30′
Performer 100	S100-I-12A **19.95**	S100-I-21A **24.95**	
Performer 500 (Jazz)	P500-IJ-12A **44.95**	P500-IJ-21A **59.95**	P500-IJ-30A **74.95**
Performer 500 (Rock)	P500-IR-12A **39.95**	P500-IR-21A **49.95**	P500-IR-30A **59.95**
Performer 500 (Bass)	**P**500-IB-12A **39.95**	P500-IB-21A **49.95**	P500-IB-30A **59.95**

100/500/1000 SERIES ANGLED 1/4″ PLUGS



	8″	18″	3′
Performer 500 (Jazz)	P500-IJ-.75DA **24.95**		
Performer 500 (Rock)	P500-IR-.75DA **19.95**	P500-IR-1.5DA **24.95**	P500-IR-3DA **29.95**
Performer 500 (Bass)	P500-IB-.75DA **19.95**		

Designed with the Musician in Mind...

Monster's Performer 500 Series of instrument cables are specifically engineered for the style of music you play. Style-specific cable designs unlock the full potential of your instrument's tonality and feel, and add a signature-quality to your personal sound. For example, Monster Rock brings out the bite in your guitar, while Monster Jazz enhances deep, full bodied tones. Whether you play rock, jazz, bass, or keyboards, Monster has a cable designed to enhance your sonic signature.

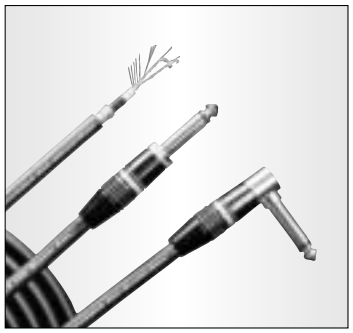
For the ultimate in transparency and accurate audio reproduction, there is Monster's Studio Pro 1000 line-the mic and instrument cables that are so popular with top professionals around the world, both in the studio and during the live performance. On a budget? Monster Standard 100 instrument, mic and speaker cables deliver improved performance at an affordable price.

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INSTRUMENT CABLES

Performer 500 Series Instrument Cables

••• They All Feature •••



- Multi-gauge Time Correct windings for phase aligned signals with a sharp edge delivers natural reproduction of midrange frequencies.
- MicroFiber inner dielectric insulation provides better isolation of magnetic fields for faster transient response time, clear highs and vivid harmonics.
- High density 95% coverage copper braided shield and conductive polymer insulator for superior RFI and EMI rejection for a lower noise floor and increased resolution.
- Carbon-infused dielectric reduces microphonic cracks or pops for a noise-free cable design.
- Iso Tec vibration isolation technology further reduces the effects of handling noise.
- Extra-flexible Duraflex outer jacket for superior reliability and cut resistance— performance after performance.
- Durable & corrosion-resistant 24k hard gold plated connectors with strain relief for a strong, high integrity connection.

PERFORMER 500 SERIES ROCK INSTRUMENT CABLE

High performance lead/rhythm guitar and effects cables delivers aggressive sonic character and dynamics for rock musicians. Delivers more bite and aggressive sound qualities than ordinary instrument cables.

PERFORMER 500 SERIES BASS INSTRUMENT CABLE

Powerful low end response and extended dynamic range from a cable specially designed for the bass player. This maximum performance bass guitar cable incorporates two thick bass frequency conductors to improve voltage and current transmission for maximum punch and an extended low end.

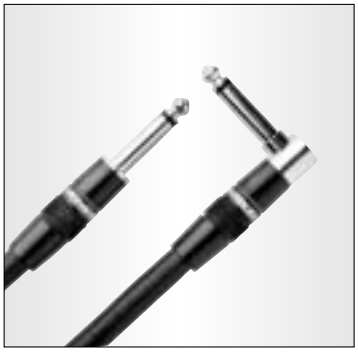
PERFORMER 500 SERIES JAZZ INSTRUMENT CABLES

Specially designed smooth, rich, high quality sounding instrument cable for jazz and acoustic guitars. Emphasizes the natural timbre and complex tones generated by your instrument. Same features as above, PLUS—

- Bandwidth Balanced construction delivers a warm, robust tonal character.
- Dual multi-gauge (high and low frequency) Time Correct wire networks for rich harmonic texture and accurate timbre reproduction
- MultiTwist (very tightly winds negative and positive conductors together) impedance balanced construction for accurate signal transfer and superior noise rejection.

Studio Pro 1000 Instrument Cables

Monster's state-of-the-art reference instrument cable, Studio Pro 1000 cables are used by top recording artists to get every nuance of timbre and tone from their instruments. Utilizing patented Time Correct windings, multiple gauge networks, MicroFiber, and Bandwidth Balanced construction, these cables deliver unmatched detail, clarity and depth. Signals are phase-aligned as they travel through the conductors, resulting in a surprisingly accurate, open sound. Built with an extra-dense braid shield and twisted-pair design, Studio Pro 1000 is ultra quiet and noise free. Heavy duty 24K gold-plated plugs with strain relief ensure the best signal transfer and a positive connection year after year.

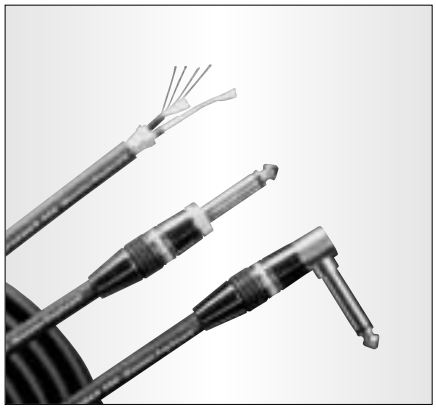


- ◆ Multiple gauge high and low frequency wire networks for accurate, natural sound reproduction.
- ◆ MicroFiber dielectric-wrapped wire networks for the fastest transients, maximum clarity, definition and presence.
- ◆ MultiTwist impedance-balanced construction for maximum signal transfer and noise rejection.
- ◆ Double shielded with 97% coverage copper braid and carbon-infused dielectric for superior noise rejection.
- ◆ 24k hard gold contact, heavy duty, high-current connectors are durable and corrosion resistant.
- ◆ Brings out the live, full sounds from digitized or harsh signals for a more natural and open keyboard feel.

INSTRUMENT CABLES

Monster Keyboard Instrument Cables

Specifically designed instrument cable for analog or digital keyboards, sound modules, DSPs and samplers. Brings out the live, full sounds from digitized or harsh signals for a more natural and open keyboard feel.



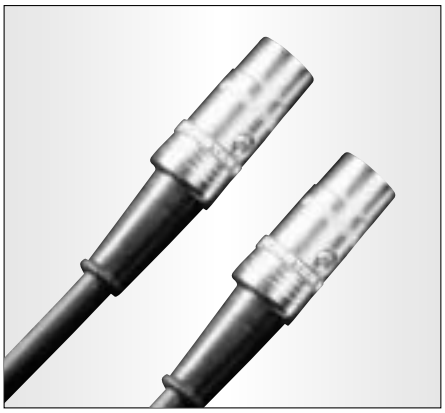
- Specially designed conductor layering in multiple helical bundles for improved high frequency signal alignment
- Bandwidth Balanced controlled impedance technology for optimized electronic interfacing and full, rich tonal quality
- Multiple gauge high and low frequency wire networks for accurate, natural sound reproduction
- Dense copper braid and carbon-infused insulating shields for maximum protection from electromagnetic noise
- IsoTec vibration isolation technology eliminates cable microphonics
- Corrosion resistant 24k hard gold plated connectors with strain relief for increased conductivity and a strong connection
- MicroFiber dielectric core for accurate lows and extended frequency response

Straight 1/4˝ Plugs

P500-IK-3 3´.....*ea.* **29.95**
P500-IK-6 6´.....*ea.* **34.95**
P500-IK-12 12´.....*ea.* **39.95**
P500-IK-21 21´.....*ea.* **49.95**

Angled to Straight 1/4˝ Plugs

P500-IK-12A 12´.....*ea.* **39.95**
P500-IK-21A 21´.....*ea.* **49.95**



Performer 500 Digilink MIDI

Flexible, high speed MIDI cable for error-free performance. Offers improved electronic instrument timing, fewer data errors, smoother control, and tighter music.

- Monster's precise Current Mode construction for high speed MIDI data transfer with fewer errors, even over long runs.
- Engineered to control current and voltage transients for reduced jitter and edge distortion, and minimum delay.
- Sturdy nickel-plated connectors with strain relief for strong, reliable,and long-lasting connections in the studio or on the road.
- Great for in the studio or on the road.
- Wired for standard MIDI. (3 Pins Hot)

P500-MIDI-3
3´ - 5 pin DIN .....*ea.* **16.95**

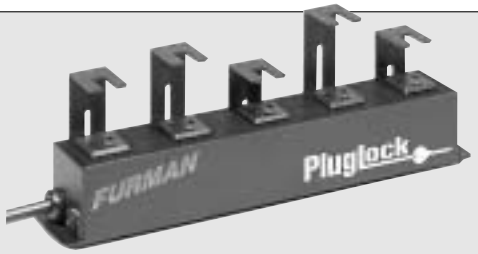
P500-MIDI-6
6´ - 5 pin DIN .....*ea.* **18.95**

P500-MIDI-12
12´ - 5 pin DIN .....*ea.* **24.95**

P500-MIDI-25
25´ - 5 pin DIN .....*ea.* **34.95**

FURMAN PlugLock

PlugLock is an inexpensive outlet strip that holds your wallwarts securely in place. Equipped with a heavy duty 14 AWG cord, five 15-amp receptacles and five locking clamps with knurled thumbscrews, PlugLock ensures that securing wallwarts and plugs will never be a problem again.



- Outlets spaced widely enough for five bulky transformer "wall warts"
- Clamps securely lock wall warts into place
- Mounts in the back of your rack
- Heavy duty 5´ three-conductor cord
- Circuit-breaker protected
- All steel construction
- Three Year Warranty
- Dimensions: 12.6 x 2.3 x 2.5˝; weighs 2 lbs 10 oz
- Mounting hole spacing: 11.61˝
- Clamps adjust from 0.4˝ to 2.1˝ above outlet

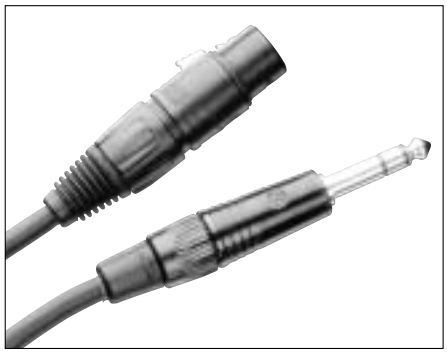
B&H PHOTO - VIDEO - PRO AUDIO
ONE HOUR FREE PARKING
AT 349 W. 34th STREET (with purchase of $100 or more)

STUDIOLINK 500 INTERCONNECT CABLES

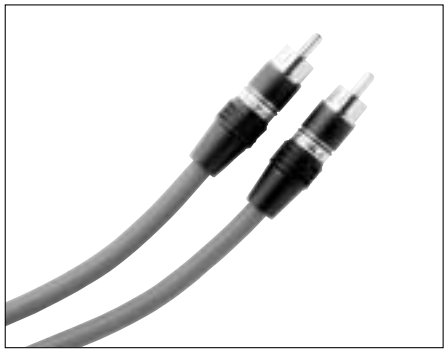
High Performance, Great Sounding Audio Interconnect Cables

The StudioLink 500 series provides excellent sonic performance in professional and home studio applications.

- Multiple gauge high and low frequency wire networks for accurate, natural sound reproduction
- Precision wound Time Correct construction provides extremely accurate musical reproduction
- Open soundstage, precise imaging, tight bass and smooth highs
- Ultra-flexible, durable jacket for ease of use and prolonged life, even in the tightest situations
- Easily distinguishable colors (red, blue, and black) for quick assignment identification
- Available with Gold Plated RCAs or 24k hard gold contact 1/4″ connectors in several configurations









STANDARD 1/4″ PLUGS

Model #	Length	Price
SL500I-ST-.5	.5 Meter	14.99
SL500I-ST-1	1 Meter	19.95
SL500I-ST-2	2 Meter	24.95
SL500I-ST-3	3 Meter	29.95
SL500I-ST-4	4 Meter	34.95
SL500I-ST-6	6 Meter	44.95

1/4″ TO 1/4″ PLUGS

Model #	Length	Price
SL500I-M-.5	.5 Meter	12.95
SL500I-M-1	1 Meter	14.95
SL500I-M-2	2 Meter	19.95
SL500I-M-3	3 Meter	24.95
SL500I-M-4	4 Meter	29.95
SL500I-M-6	6 Meter	39.95

RCA TO 1/4″ PLUGS

Model #	Length	Price
SL500I-CR-1	1 Meter	14.95
SL500I-CR-2	2 Meter	19.95
SL500I-CR-3	3 Meter	24.95
SL500I-CR-4	4 Meter	29.95
SL500I-CR-6	6 Meter	39.95

XLR CONNECTORS

Model #	Length	Price
SL500I-X-1	1 Meter	19.95
SL500I-X-2	2 Meter	24.95
SL500I-X-3	3 Meter	29.95
SL500I-X-4	4 Meter	34.95
SL500I-X-6	6 Meter	44.95

1/4″ TO RCA PLUGS

Model #	Length	Price
SL500I-CM-1	1 Meter	14.95
SL500I-CM-2	2 Meter	19.95
SL500I-CM-3	3 Meter	24.95
SL500I-CM-4	4 Meter	29.95
SL500I-CM-6	6 Meter	39.95

1/4″ STEREO TO XLR MALE

Model #	Length	Price
SL500I-CMX-1	1 Meter	19.95
SL500I-CMX-2	2 Meter	24.95
SL500I-CMX-3	3 Meter	29.95
SL500I-CMX-4	4 Meter	34.95
SL500I-CMX-6	6 Meter	44.95

1/4″ STEREO TO XLR FEMALE

Model #	Length	Price
SL500I-CFX-1	1 Meter	19.95
SL500I-CFX-2	2 Meter	24.95
SL500I-CFX-3	3 Meter	29.95
SL500I-CFX-4	4 Meter	34.95
SL500I-CFX-6	6 Meter	44.95

RCA TO RCA PLUGS

Model #	Length	Price
SL500I-R-1	1 Meter	14.95
SL500I-R-2	2 Meter	19.95
SL500I-R-3	3 Meter	24.95
SL500I-R-4	4 Meter	29.95
SL500I-R-6	6 Meter	39.95

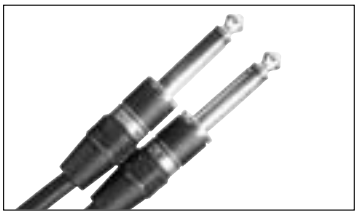
PROLINK SPEAKER CABLES

Standard 100 Speaker Cable

Entry-level speaker cables for improved sound at an affordable price. Excellent general purpose speaker cable.

- Magnetic Flux Tube construction and special cable windings for natural music reproduction
- Impressive clarity, bass response and dynamic range in a compact design
- Durable UL-listed jacket for studio in-wall installation or stage use
- 24k gold-tipped 1/4″ connectors or gold Monster Tips (bananas) for durability and improved signal transfer.

	3′	6′	10′	20′	40′	50′
Straight 1/4″ Plugs	S100-S-3 **12.95**	S100-S-6 **14.95**	S100-S-10 **19.95**	S100-S-20 **29.95**	S100-S-40 **44.95**	S100-S-50 **54.95**
Straight 1/4″ to Dual Bananas	—	—	S100-S-10MT **19.95**	S100-S-20MT **29.95**	—	S100-S-50MT **54.95**
Dual Bananas to Dual Bananas	—	—	S100-S-10DMT **39.95**	—	—	—



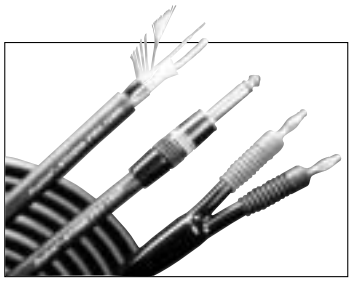
Performer 500 Speaker Cable

One of Monster's most advanced, compact, high performance speaker cable. Designed for self-contained head/speaker combinations, PAs and club applications, the Performer 500 speaker cable provides excellent soundstage and tonal balance at an affordable price.

	3′	6′	10′	20′	40′	50′
Straight 1/4″ Plugs	P500-S-3 **19.95**	P500-S-6 **24.95**	P500-S-10 **29.95**	P500-S-20 **39.95**	P500-S-40 **59.95**	P500-S-50 **69.95**
Straight 1/4″ to Dual Bananas	—	—	P500-S-10MT **29.95**	P500-S-20MT **39.95**	—	P500-S-50MT **69.95**
Dual Bananas to Dual Bananas	—	—	P500-S-10DMT **29.95**	P500-S-20DMT **39.95**	—	—

- Special Time Correct windings for accurate music reproduction
- Magnetic Flux Tube for improved bass response and more natural sonic characteristics
- Increased conductor area transfers all the amplifier's power to speakers
- Durable UL-listed jacket for studio in-wall installation or stage use
- 24k gold-tipped 1/4″ connectors or gold Monster Tips (bananas) for durability and improved signal transfer.





Studio Pro 1000 Speaker Cable

Monster's advanced Time Correct speaker cable is designed for professional audio applications including important stage and studio mains and monitor hook-up.

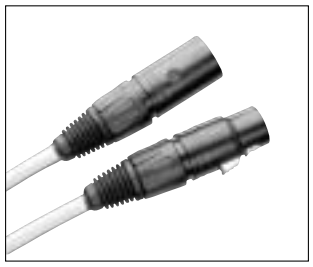
	3′	6′	10′	20′	50′
Straight 1/4″ Plugs	SP1000-S-3 **29.95**	SP1000-S-6 **34.95**	SP1000-S-10 **39.95**	SP1000-S-20 **59.95**	SP1000-S-50 **119.95**
Straight 1/4″ to Dual Bananas	—	—	SP1000-S-10MT **39.95**	SP1000-S-20MT **59.95**	SP1000-S-50MT **119.95**
Dual Bananas to Dual Bananas	—	—	SP1000-S-10DMT **39.95**	SP1000-S-20DMT **59.95**	—

- Precision Time Correct windings with separate wire networks give excellent sound-stage for superior speaker performance
- Heavy gauge Bass Control Conductor for impressive low frequency punch and dynamics
- Magnetic Flux Tube improves bass response and natural sonic characteristics
- Large conductor area provides high current delivery with greater clarity, smooth sound, and natural tonal balance
- Durable UL-listed jacket for studio in-wall installation or stage use
- 24k gold-tipped 1/4″ connectors or gold Monster Tips (bananas) for durability and improved signal transfer

PROLINK DIGITAL INTERCONNECT CABLES

Monster offers a full line of high resolution low loss digital cables for all applications. These include AES/EBU, S/PDIF and high performance fiber optic cables.

StudioPro 1000 Silver Digital AES/EBU Cables



The professionals choice for the most accurate, musical sounding AES/EBU digital audio connections.

- Polished silver plated OFC conductors for fastest rise time and detailed, transparent sound
- Low capacitance, precision 110-ohm construction for reduced jitter and accurate signal transfer even over long runs
- Dense shielding resists leakage and outside interference
- Heavy duty black Neutrik XLRs
- Close-tolerance impedance and extra-wide bandwidth exceeds AES/EBU specification for 110-ohm digital systems

SP1000-AES-.5M .5 Meter - precision XLRs.....**34.95** *ea.*
SP1000-AES-1M 1 Meter - precision XLRs.....**49.95** *ea.*
SP1000-AES-2M 2 Meter - precision XLRs.....**64.95** *ea.*
SP1000-AES-4M 4 Meter - precision XLRs.....**99.95** *ea.*
SP1000-AES-250 250 ft. bulk spool, 5.75´ .....**Call**

Interlink LightSpeed 200

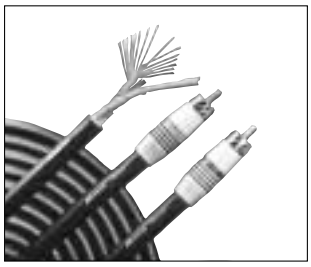


Higher technology fiber optic cable delivers the most accurate bitstream transfer for the clearest, smoothest possible sound from CD Players, MiniDisc, DVD Players, outboard DACs, and Digital Satellite Receivers

- Monster designed pressure connection keeps fiber in perfect alignment with components
- Metal shell connector for lifelong durability
- Graded index optical fiber greatly reduces jitter for enhanced clarity and definition
- Spring-loaded connector keeps fiber in optical alignment for maximum data transfer
- For use with DVD, CD, MiniDisc, DACs, S/PDIF, and Dolby Digital audio applications

ILS200-1M 1 meter fiber optic toslink-to-toslink .....**59.95** *ea.*
ILS200-2M 2 meter fiber optic toslink-to-toslink .....**79.95** *ea.*
ILS200-4M 4 meter fiber optic toslink-to-toslink ..... **99.95** *ea.*

M Series Silver (M1000D) Digital S/PDIF Cables

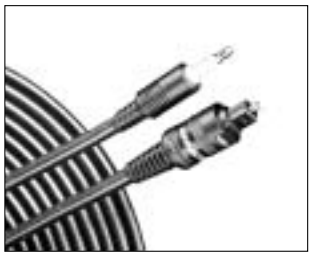


Low attenuation, precision S/PDIF digital coaxial interconnect cables.

- Accurate impedance matching design for detailed, more natural sounding music
- Low capacitance with fast rise time construction reduces jitter
- Extra-wide bandwidth for accurate signal transfer and smooth, accurate music reproduction
- Gas-injected dielectric prevents bit errors, even over long runs
- Double shield prevents digital emissions
- Patented 24k gold Turbine RCA connectors with Split-Tip center pin for a superior construction
- Duraflex jacket for flexibility and long-lasting good looks

M1000D-1M 1 Meter Piece - 3.28´ .....**55.00** *ea.*
M1000D-2M 2 Meter Piece - 6.56´ .....**80.00** *ea.*
M1000D-4M 4 Meter Piece - 13.12´ .....**90.00** *ea.*

Interlink LightSpeed 100



High performance fiber optic cable with special polished low-loss connections for CD, MiniDisc and DAT Players

- Monster's precise fiber optic connection delivers higher sonic performance than standard cables
- Tuned spectral attenuation for optimum signal admittance and low loss
- High velocity of propagation for lower time smear
- Special low-loss fiber delivers clean, clear sound with enhanced transparency
- Available in standard fiber optic to mini fiber optic configurations

ILS100-1M 1 meter fiber optic toslink-to-toslink.....**39.95** *ea*
ILS100-2M 2 meter. fiber optic toslink-to-toslink.....**44.95** *ea.*
ILS100-4M 4 meter fiber optic toslink-to-toslink..... **59.95** *ea.*
ILS100-20FT 20´ fiber optic toslink-to-toslink .....**74.95** *ea.*
ILS100TM-1M 1 meter fiber optic toslink-to-mini optical ....**39.95** *ea.*
ILS100TM-2M 2 meter fiber optic toslink-to-mini optical ....**44.95** *ea.*

MONSTER CABLE

PROLINK 500 SNAKE CABLES

The reference standard for studio and sound reinforcement multi-channel cable, StudioLink 500 Snake cables deliver the ultimate clarity and consistent quality resolution, in multichannel connections to consoles, tape machines, workstations, and effects racks. Available in 4, 8, 16 and 27 channel configurations.

- Two multiple gauge Bandwidth Balanced wire networks for clarity and punch with extended frequency response
- Precision-wound Time Correct construction provides extremely accurate musical reproduction
- Extra fine copper stranding for improved signal transfer and wide bandwidth frequency response



- Tightly twisted-pair construction with 100% shield and drain for each channel rejects external noise and crosstalk
- Each channel is individually jacketed, color-coded, and numbered
- Durable, flexible outer jacket UL-listed for in-wall installations
- Ready for balanced or unbalanced termination schemes

StudioLink 500 4-Channel Snake Cables

SL500-S4-M-1	1 Meter - 1/4″ to 1/4″ plugs	**49.95** *ea.*
SL500-S4-M-2	2 Meter - 1/4″ to 1/4″ plugs	**59.95** *ea.*
SL500-S4-M-3	3 Meter - 1/4″ to 1/4″ plugs	**69.95** *ea.*
SL500-S4-M-4	4 Meter - 1/4″ to 1/4″ plugs	**79.95** *ea.*
SL500-S4-SR-1	1 Meter - 1/4″ Send & Return	**59.95** *ea.*
SL500-S4-SR-2	2 Meter - 1/4″ Send & Return	**69.95** *ea.*
SL500-S4-SR-3	3 Meter - 1/4″ Send & Return	**79.95** *ea.*
SL500-S4-SR-4	4 Meter - 1/4″ Send & Return	**89.95** *ea.*
SL500-S4-X-1	1 Meter - XLR to XLR plugs	**89.95** *ea.*
SL500-S4-X-2	2 Meter - XLR to XLR plugs	**99.95** *ea.*
SL500-S4-X-3	3 Meter - XLR to XLR plugs	**109.95** *ea.*
SL500-S4-X-4	4 Meter - XLR to XLR plugs	**119.95** *ea.*
SL500-S4-CMX-1	1 Meter - 1/4″ Stereo to XLR Male	**89.95** *ea.*
SL500-S4-CMX-2	2 Meter - 1/4″ Stereo to XLR Male	**99.95** *ea.*
SL500-S4-CMX-3	3 Meter - 1/4″ Stereo to XLR Male	**109.95** *ea.*
SL500-S4-CMX-4	4 Meter - 1/4″ Stereo to XLR Male	**119.95** *ea.*
SL500-S4-CFX-1	1 Meter - 1/4″ Stereo to XLR Female	**89.95** *ea.*
SL500-S4-CFX-2	2 Meter - 1/4″ Stereo to XLR Female	**99.95** *ea.*
SL500-S4-CFX-3	3 Meter - 1/4″ Stereo to XLR Female	**09.95** *ea.*
SL500-S4-CFX-4	4 Meter - 1/4″ Stereo to XLR Female	**119.95** *ea.*
SL500-S4-ST-1	1 Meter - 1/4″ Stereo to 1/4″ Stereo	**99.95** *ea.*
SL500-S4-ST-2	2 Meter - 1/4″ Stereo to 1/4″ Stereo	**109.95** *ea.*
SL500-S4-ST-3	3 Meter - 1/4″ Stereo to 1/4″” Stereo	**119.95** *ea.*
SL500-S4-ST-4	4 Meter - 1/4″ Stereo to 1/4″ Stereo	**129.95** *ea.*
SL500-S4-250	4 Channel Snake - 250′ spool	**2.95 per ft**

StudioLink 500 8-Channel Snake Cable

SL500-S8-M-1	1 Meter - 1/4″ to 1/4″ plugs	**99.95** *ea.*
SL500-S8-M-2	2 Meter - 1/4″ to 1/4″ plugs	**114.95** *ea.*
SL500-S8-M-3	3 Meter - 1/4″ to 1/4″ plugs	**129.95** *ea.*
SL500-S8-M-4	4 Meter - 1/4″ to 1/4″ plugs	**144.95** *ea.*
SL500-S8-SR-1	1 Meter - 1/4″ Send & Return	**119.95** *ea.*
SL500-S8-SR-2	2 Meter - 1/4″ Send & Return	
SL500-S8-SR-3	3 Meter - 1/4″ Send & Return	**149.95** *ea.*
SL500-S8-X-1	1 Meter - XLR to XLR	**179.95** *ea.*
SL500-S8-X-2	2 Meter - XLR to XLR	**199.95** *ea.*
SL500-S8-X-3	3 Meter - XLR to XLR	**219.95** *ea.*
SL500-S8-X-4	4 Meter - XLR to XLR	**239.95** *ea.*
SL500-S8-CMX-1	1 Meter - 1/4″ Stereo to XLR Male	**179.95** *ea.*
SL500-S8-CMX-2	2 Meter - 1/4″ Stereo to XLR Male	**199.95** *ea.*
SL500-S8-CMX-3	3 Meter - 1/4″ Stereo to XLR Male	**219.95** *ea.*
SL500-S8-CMX-4	4 Meter - 1/4″ Stereo to XLR Male	**239.95** *ea.*
SL500-S8-CFX-1	1 Meter - 1/4″ Stereo to XLR Female	**179.95** *ea.*
SL500-S8-CFX-2	2 Meter - 1/4″ Stereo to XLR Female	**199.95** *ea.*
SL500-S8-CFX-3	3 Meter - 1/4″ Stereo to XLR Female	**219.95** *ea.*
SL500-S8-CFX-4	4 Meter - 1/4″ Stereo to XLR Female	**39.95** *ea.*
SL500-S8-ST-1	1 Meter - 1/4″ Stereo to 1/4″ Stereo	**199.95** *ea.*
SL500-S8-ST-2	2 Meter - 1/4″ Stereo to 1/4″ Stereo	**214.95** *ea.*
SL500-S8-ST-3	3 Meter - 1/4″ Stereo to 1/4″ Stereo	**229.95** *ea.*
SL500-S8-ST-4	4 Meter - 1/4″ Stereo to 1/4″ Stereo	**244.95** *ea.*
SL500-S8-250	8 Channel Snake - 250 ft spool	**4.80**

PROLINK 500 SNAKE CABLES

StudioLink 500 DA-88 Interconnect Cables

SLDA88-MX-3 3 Meter - D-Sub to MXLR.....**199.95** *ea.*
SLDA88-MX-5 5 Meter - D-Sub to MXLR.....**224.95** *ea.*
SLDA88-FX-3 3 Meter - D-Sub to FXLR .....**199.95** *ea.*
SLDA88-FX-5 5 Meter - D-Sub to FXLR .....**224.95** *ea.*
SLDA88-ST-1.5 1.5 Meter - D-Sub to 1/4" Stereo.....**174.95** *ea.*
SLDA88-ST-3 3 Meter - D-Sub to 1/4" Stereo.....**199.95** *ea.*
SLDA88-ST-5 5 Meter - D-Sub to 1/4" Stereo.....**224.95** *ea.*

StudioLink 500 16-Channel Snake Cables

SLADAT-X-3 3 Meter - Elco to XLR (8m, 8f) .....**399.95** *ea.*
SLADAT-X-5 5 Meter - Elco to XLR (8m, 8f) .....**449.95** *ea.*
SLADAT-ST-1.5 1.5 Meter - Elco to Stereo 1/4" plugs .....**349.95** *ea.*
SLADAT-ST-3 3 Meter - Elco to Stereo 1/4" plugs .....**399.95** *ea.*
SLADAT-ST-5 5 Meter - Elco to Stereo 1/4" plugs .....**449.95** *ea.*
SL500-S16-250 16 Channel Snake - 250 ft spool.....**8.30 ft.**

StudioLink 500 27-Channel Snake Cable

SL500-S27-250 27-Channel Snake (250 ft. Spool).....**12.60 ft.**



STUDIOLINK 500 PATCHBAY CABLES

High performance, great sounding, durable patchbay cables, the StudioLink 500 series provides excellent sonic performance and consistent quality sound in professional and home studio patch applications.



- Two multiple-gauge Bandwidth Balanced wire networks for clarity and low end punch with extended frequency response.
- Precision wound Time Correct construction for extremely accurate musical reproduction.
- Open soundstage, precise imaging, tight bass and smooth highs.
- Ultra-flexible, durable jacket for ease of use and prolonged life, even in the tightest situations
- Five easily distinguishable colors (black, blue, green, yellow, and ivory) for quick assignment identification for 8 pack TTS.
- Two easily distinguishable colors (black and blue) for quick assignment identification for certain 8 pack TTS.
- Available with heavy duty, high conductivity TT, long-frame, or 297-style stereo 1/4" connectors.
- Available with heavy-duty, high conductivity TT, XLR's or 1/4" TRS connectors
- Prolink Studiolink 500 TT available in 6 colors for quick cable identification when multiple cables are used: black, red, blue, green, yellow, and ivory.

SL500-P-1TT8 8 pk - 1′ - 12″ - TT plugs.....**109.95** *ea.*
SL500-P-1.5TT8 8 pk - 1.5′ - 18″ - TT plugs.....**114.95** *ea.*
SL500-P-2TT8 8 pk - 2′ - 24″ - TT plugs.....**119.95** *ea.*
SL500-P-3TT8 8 pk - 3′ - 36″ - TT plugs.....**129.95** *ea.*

SL500-P-6TXM 6′ - 72″ - TT to XLR male.....**24.95** *ea.*
SL500-P-10TXM 10′ - 120″ - TT to XLR male.....**29.95** *ea.*
SL500-P-20TXM 20′ - 240″ - TT to XLR male.....**39.95** *ea.*

SL500-P-6TXF 6′ - 72″ - TT to XLR female.....**24.95** *ea.*
SL500-P-10TXF 10′ - 120″ - TT to XLR female.....**29.95** *ea.*
SL500-P-10TXF 20′ - 240″ - TT to XLR female.....**39.95** *ea.*

SL500-P-6TTRS 6′ - 72″ - TT to Balanced 1/4″ .....**24.95** *ea.*
SL500-P-10TTRS 10′ - 120″ - TT to Balanced 1/4″ .....**29.95** *ea.*
SL500-P-20TTRS 20′ - 240″ - TT to Balanced 1/4″ .....**39.95** *ea.*

MICROPHONE CABLES

StageMASTER Series (SoniMaxer, AudioFlex)

Designed and built by Neutrik in China, StageMASTER's low-impedance mic cables are available in 3 to 50-ft. lengths and offer a 3-year warranty. An overmolded boot adds an excellent strain relief. Spiral shield coverage is 78% or better. The AudioFlex series is an inexpensive yet reliable (3-year warranty) low-impedance mic cable featuring Amphenol IDC-type XLR connectors with 224SM small diameter wire. Also available with black Amphenol XLR connectors (AXMB).



LIFELINES WITH THREE-PIN XLR CONNECTORS								
	3′	**5′**	**10′**	**15′**	**20′**	**25′**	**50′**	**100′**
SoniMaxer	SMM-3 **7.99**	SMM-5 **8.99**	SMM-10 **38.49**	SMM-15 **9.99**	SMM-20 **11.99**	SMM-25 **13.99**	SMM-50 **15.99**	— —
AudioFlex	AXM-3 **Call**	AXM-5 **Call**	AXM-10 **Call**	AXM-15 **Call**	AXM-20 **Call**	AXM-25 **Call**	AXM-50 **Call**	AXM-100 **Call**
AudioFlex	AXMB-3 **Call**	AXMB-5 **Call**	AXMB-10 **Call**	AXMB-15 **Call**	AXMB-20 **Call**	AXMB-25 **Call**	AXMB-50 **Call**	AXMB-100 **Call**

Excellines Series (ExtraFlex)

Designed to provide all the clarity and power you need for great sounding vocals, the ExtraFlex cable offers superior transient response for a quick, yet natural sound. With genuine Neutrik XLR connectors and ArmorFlex outer jacket, this cable is built for years of rugged use. 24-guage, high-purity copper conductors, 92% spiral copper shield and high-dielectric insulator. 10-year warranty. For Hi-Z applications, the ExtraFlex Hi-Z cable is available with nickel shell silver contact 3-pin XLR to 1/4″ BigFoot phone plug. Also available with Hi-Z to Lo-Z transformer. Excellines come with a 10-year warranty.



EXCELLINES SERIES								
	3′	**5′**	**10′**	**15′**	**20′**	**25′**	**50′**	**100′**
ExtraFlex	EXM-3 **17.99**	EXM-5 **16.49**	EXM-10 **17.99**	EXM-15 **19.99**	EXM-20 **19.99**	EXM-25 **22.99**	EXM-50 **29.99**	EXM-100 **49.95**

ExtraFlex Hi-Z

Designed to provide all the clarity and power you need for Hi-Z applications, this cable offers superior transient response for a quick, yet natural sound. Pin 2 is hot as standard, and pin 3 is also available. With high quality connectors and ArmorFlex outer jacket, this cable is built for years of rugged use.

- Nickel shell silver contact XLR -female to 1/4″ BigFoot Phone Plug
- 20 AWG Pro Co 120S1 Wire in Black • 55% Coverage Copper Spiral Shield

ExtraFlex Hi-Z with Transformer

Nickel shell silver contact XLR -female to Neutrik 1/4″ transformer with 224S spiral shield wire

ExtraFlex Hi-Z		
	20′	**25′**
ExtraFlex Hi-Z	EXH-20 **18.99**	–
	EXHP3H-20 **18.99**	

ExtraFlex Hi-Z w/Transformer		
	20′	**25′**
ExtraFlex Hi-Z with Transformer	EXHT-20 **49.95**	EXHT-25 **51.95**

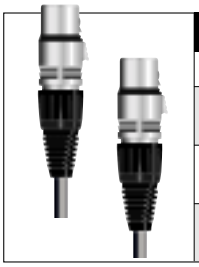
MICROPHONE CABLES

Lifelines (Ameriquad, Mastermike)

AmeriQuad are studio reference microphone cables with quad construction for great reliability. Featuring hum-bucking bi-wire geometry they deliver ten times better noise rejection (20dB of additional noise reduction) than 2-wire configurations for crisp, clean, reliable sound. Uses 4 x 24 AWG Proco 424B "AMERIQUAD" wire with a 96% copper braided shield and Neutrik gold-contact, black-shell XLR connectors.

MasterMike microphone cables provide outstanding low noise performance in a highly durable cable. Featuring Proco's exclusive ArmorFlex outer jacket, this highly flexible cable uses a 2-conductor, 23 AWG Pro Co 223B "DynaMike" Wire in black with a 94% braided shield. Extremely rugged, MasterMike cables are built to withstand on-the-road abuse. Uses Neutrik silver contact, black-shell XLR connectors. Also available with integrated On/Off switch on female connector. LifeLines cables come with a 20-year warranty.





LIFELINES WITH THREE-PIN XLR CONNECTORS								
	5´	**10´**	**15´**	**20´**	**25´**	**30´**	**50´**	**100´**
Ameriquad	AQ-5 **29.95**	AQ-10 **34.95**	AQ-15 **38.95**	AQ-20 **42.95**	AQ-25 **44.95**	AQ-30 **49.95**	AQ-50 **64.95**	AQ-100 **79.95**
MasterMike	M-5 **19.99**	M-10 **22.95**	M-15 **24.95**	M-20 **27.95**	M-25 **29.95**	M-30 **34.95**	M-50 **44.95**	M-100 **74.95**
MasterMike with on/off switch				MSW-20 **33.49**	MSW-25 **37.49**		MSW-50 **59.49**	



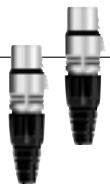
Guardian Series (Merlin)

Watertight with Kevlar reinforced wire...bi-wire technology, bullet-proof and incredibly flexible! Stainless steel shell, heavy-duty gold contact three pin waterproof XLR connectors with 424BXXK quad wire with Kevlar.

Quite possibly the world's best microphone cable, Merlin provides all the clarity and power you need for great sounding vocals. Kevlar reinforced for unmatched strength and dependability, Merlin cables utilize Proco's low-noise, hum-bucking AmeriQuad bi-wire geometry which reduces noise up to 20dB over 2-wire designs. Custom watertight Neutrik XLR connectors feature a machined stainless steel housing with gold contacts.

MERLIN THREE-PIN WATERPROOF XLR CONNECTORS				
15´	**20´**	**25´**	**30´**	**50´**
ME-15 **59.95**	ME-20 **64.95**	ME-25 **69.95**	ME-30 **72.95**	ME-50 **69.95**

Stainless steel shell, heavy-duty gold contact three pin waterproof XLR connectors with 424BXXK quad wire with Kevlar.



SPEAKER CABLES

Lifelines

Perfect for players on a budget, Excellines speaker cable is engineered to flawlessly deliver the full range and power provided by your amplifier. Encased in Proco's ArmorFlex outer jacket for durability and tangle-free flexibility, this cable is terminated with heavy-duty connectors (1/4″ phone plugs, banana plugs, or Neutrik Speakon connectors) for years of trouble-free use. Includes a 10-year warranty.



14 AWG SPEAKER CABLE

1/4″ - 1/4″

Model #	Length	Price
S14-3	3′	14.95
S14-6	6′	16.95
S14-10	10′	16.49
S14-25	25′	23.95
S14-50	50′	34.95

BANANA - BANANA

Model #	Length	Price
S14BB-3	3′	15.95
S14BB-6	6′	16.49
S14BB-10	10′	16.95
S14BB-25	25′	24.95
S14BB-25	50′	34.95

SPEAKON-SPEAKON

Model #	Length	Price
S14NN-3	3′	19.95
S14NN-6	6′	21.95
S14NN-10	10′	24.95
S14NN-25	25′	29.95
S14NN-50	50′	39.95

1/4″ - BANANA

Model #	Length	Price
S14BQ-3	3′	16.95
S14BQ-6	6′	17.95
S14BQ-10	10′	17.95
S14BQ-25	25′	24.95
S14BQ-50	50′	34.95

1/4″ - SPEAKON

Model #	Length	Price
S14NQ-3	3′	17.95
S14NQ-6	6′	18.49
S14NQ-10	10′	19.95
S14NQ-25	25′	24.95
S14NQ-50	50′	37.95

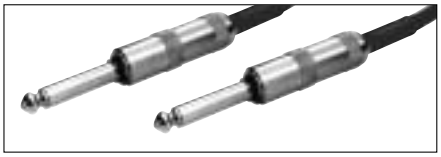
SPEAKON-BANANA

Model #	Length	Price
S14BN-10	10′	19.95
S14BN-25	25′	24.95
S14BN-50	50′	37.95

16 AWG SPEAKER CABLE

1/4″ - 1/4″

Model #	Length	Price
S16-3	3′	14.95
S16-6	6′	15.95
S16-10	10′	16.95
S16-25	25′	18.95
S16-50	50′	24.95



BANANA - BANANA

Model #	Length	Price
S16BB-10	10′	16.95
S16BB-25	25′	19.95
S16BB-50	50′	24.95

1/4″ - BANANA

Model #	Length	Price
S16BQ-3	3′	14.95
S16BQ-6	6′	15.95
S16BQ-10	10′	16.95
S16BQ-25	25′	19.95
S16BQ-50	50′	24.95

1/4″ - SPEAKON

Model #	Length	Price
S16NQ-3	3′	15.95
S16NQ-6	6′	16.95
S16NQ-10	10′	17.95
S16NQ-25	25′	21.95
S16NQ-50	50′	29.95

SPEAKON - BANANA

Model #	Length	Price
S16BN-10	10′	17.95
S16BN-25	25′	21.95
S16BN-50	50′	29.95

SPEAKON - SPEAKON

Model #	Length	Price
S16NN-3	3′	18.95
S16NN-6	6′	19.95
S16NN-10	10′	20.95
S16NN-25	25′	24.95
S16NN-50	50′	29.95

SPEAKER CABLE

Lifelines (PowerPlus)

PowerPlus speaker cables use Proco's low capacitance, low inductance PowerPlus wire, made of high-purity, stranded copper for a crisp, clean sound. They also features specially designed oxygen-free ShowSaver's brand speaker connectors and Proco's exclusive ArmorFlex outer jacket (black), providing you with a durable, yet highly flexible cable. Available with phone or banana plugs, or Neutrik Speakon connectors. 20-year warranty.



12 AWG SPEAKER CABLE

1/4″ - 1/4″

Model #	Length	Price
S12-3	3′	18.95
S12-6	6′	19.95
S12-10	10′	22.95
S12-25	25′	32.95
S12-100	100′	59.95

1/4″ - BANANA

Model #	Length	Price
S12BQ-10	10′	24.95
S12BQ-25	25′	34.95
S12BQ-50	50′	49.95
S12BQ-100	100′	64.95

BANANA - BANANA

Model #	Length	Price
S12BB-10	10′	24.95
S12BB-25	25′	34.95
S12BB-50	50′	49.95
S12BB-100	100′	64.95

1/4″ - SPEAKON

Model #	Length	Price
S12NQ-10	10′	24.95
S12NQ-25	25′	34.95
S12NQ-50	50′	49.95
S12NQ-100	100′	64.95

BANANA - SPEAKON

Model #	Length	Price
S12BN-10	10′	24.95
S12BN-25	25′	34.95
S12BN-50	50′	54.95
S12BN-100	100′	79.95

SPEAKON - SPEAKON

Model #	Length	Price
S12NN-10	10′	27.95
S12NN-25	25′	37.95
S12NN-50	50′	54.95
S12NN-100	100′	64.95

11 AWG (4-CONDUCTOR) SPEAKER CABLE

1/4″ - SPEAKON

Model #	Length	Price
S114NQ-6	6′	49.95
S114NQ-10	10′	49.95
S114NQ-25	25′	64.95
S114NQ-50	50′	94.95
S114NQ-100	100′	169.95

BANANA - SPEAKON

Model #	Length	Price
S114BN-6	6′	44.95
S114BN-10	10′	49.95
S114BN-25	25′	69.95
S114BN-50	50′	104.95
S114BN-100	100′	179.95

SPEAKON - SPEAKON

Model #	Length	Price
S114NN-6	6′	54.95
S114NN-10	10′	59.95
S114NN-25	25′	64.95
S114NN-50	50′	104.95
S114NN-100	100′	179.95

11 AWG SPEAKER CABLE

Pro Co's 13-guage 8-conductor cable was designed to fit into Neutrik 8-conductor Speakons, the largest guage of stranded wire that Pro Co could practically fit into these wonderful speaker connectors.

S138NN-10	10′	69.95
S138NN-25	25′	94.95
S138NN-50	50′	119.95
S138NN-100	100′	209.95

SPEAKER CABLES

Fat Max

Fat Max is the ultimate speaker cable. A massive 8-gauge cable consisting of four 11-gauge conductors in a special bi-wired geometry, Fat Max is designed to deliver the maximum damping factor for tight, chest-slamming lows and a wide, clearly defined audio spectrum. The unique AmeriQuad geometry reduces crossover network interaction and cable inductance, resulting in less stray capacitance for the cleanest possible high-frequency response and superb sound stage. Designed for a lifetime of use and abuse, Fat Max features sonically welded terminations and the best connectors made today (unbreakable 1/4″ gold plated, copper tipped phone plugs, banana plugs, or Speakon connectors).



12 AWG SPEAKER CABLE

1/4″ - 1/4″

Model #	Length	Price
FM-3	3′	59.95
FM-6	6′	64.95
FM-10	10′	69.95
FM-25	25′	69.95
FM-50	100′	109.95

1/4″ - BANANA

Model #	Length	Price
FMBQ-3	3′	49.95
FMBQ-6	6′	54.95
FMBQ-10	10′	59.95
FMBQ-25	25′	64.95
FMBQ-50	50′	104.95

BANANA - SPEAKON

Model #	Length	Price
FMBN-3	3′	49.95
FMBN-6	6′	49.95
FMBN-10	10′	52.95
FMBN-25	25′	59.95
FMBN-50	50′	94.95

1/4″ - SPEAKON

Model #	Length	Price
FMNQ-3	3′	54.95
FMNQ-6	6′	59.95
FMNQ-10	10′	64.95
FMNQ-25	25′	69.95
FMNQ-50	50′	99.95

BANANA - BANANA

Model #	Length	Price
FMBB-3	3′	44.95
FMBB-6	6′	49.95
FMBB-10	10′	49.95
FMBB-25	25′	74.95
FMBB-50	50′	94.95

SPEAKON - SPEAKON

Model #	Length	Price
FMNN-3	3′	49.95
FMNN-6	6′	49.95
FMNN-10	10′	64.95
FMNN-25	25′	64.95
FMNN-50	50′	94.95

RMS1A *Monitor Speaker Switch Box*

Designed with the small recording studio in mind, the RMS-1A is a stereo speaker selector ideally suited to the use of "nearfield" monitors or headphones in addition to large main speakers, all selectively driven from a single power amp. The rugged, black-epoxy-painted steel chassis is perfect for free-standing use or custom installations.



- The RMS-1A routes the outputs of a stereo amplifier to one of three pairs of speakers.
- Switching logic allows fast and easy "A-B" comparison between any two pairs of speakers.
- The heavy-duty toggle switches safely carry loads as high as 1800 watts.
- All inputs and outputs are via standard color-coded 5-way binding posts, allowing the connection of spade lugs, banana plugs or stripped wire ends for cabling from amplifiers and speakers.
- In addition, the ALT B outputs are wired through a standard 1/4″ stereo jack for easy connection of a Proco HJ-6 Headphone Junction Box using a single cable. Epecially useful in personal studios where the "control room" doubles as the "vocal booth"!

B&H PHOTO - VIDEO - PRO AUDIO

INSTRUMENT CABLES

*An innovator and leader in the cable business for over 20 years, Proco pays very close attention to every aspect of what goes into the products they sell you. Everything from solder to stranding, plugs to packaging is scrutinized before a cable is allowed to leave the factory. Every cable is built to rigid specifications. Each connector comes from the finest manufacturers of audio connectors in the world. Proco offers four cable lines categorized by quality—**StageMaster** (good), **Excellines** (better), **LifeLines** (best) and **Guardian** (amazing). Most of what distinguishes one from the other is the type of cable used and the warranty.*

Guardian

Incredible sound in an indestructible cable – simply the best cable made today. Backed by a lifetime warranty - any excuse, even abuse.

LifeLines

Professional grade cables constructed of high-quality components for outstanding performance in a durable cable. Lifeline cables are backed with a 20-year warranty.



Excellines

An EXCEL-lent mid-grade cable, these are a great choice for light-duty professional and amateur applications. Excellines come with a 10-year warranty.

StageMaster

Proco's most economical cable line, StageMasters deliver respectable performance at an exceptional price. Backed with a 3-year warranty.

INSTRUMENT CABLES

Guardian

	Shield Type	% Coverage	Conductors	Silent Plug	Contacts	Right Angle Plug	Warranty
Black Jack	Spiral	90%	1	No	Gold	No	Lifetime
Defender	Braided	96%	4	No	Gold	Yes	Lifetime
Sir Tweed	Spiral	90%	1	No	Gold	Yes	Lifetime
Sir Tweed Woody	Spiral	90%	1	No	Gold	Yes	Lifetime

LifeLines

	Shield Type	% Coverage	Conductors	Silent Plug	Contacts	Right Angle Plug	Warranty
Excalibur	Spiral	90%	1	No	Nickel	No	20 Year
LineCable	Braided	94%	2	No	Nickel	Yes	20 Year
StarGuards	Spiral	90%	1	Yes	Nickel	No	20 Year

Excellines

	Shield Type	% Coverage	Conductors	Silent Plug	Contacts	Right Angle Plug	Warranty
Excellines	Spiral	55%	1	No	Nickel	Yes	10 Year
SilentKnight	Spiral	55%	1	Yes	Nickel	No	10 Year

StageMaster

	Shield Type	% Coverage	Conductors	Silent Plug	Contacts	Right Angle Plug	Warranty
Champion	Spiral	55%	1	No	Nickel	No	3 Year
Crusader	Spiral	55%	1	No	Nickel	No	3 Year

INSTRUMENT CABLES

StageMASTER Series (Champion and Crusader)

Providing the "best-bang-for-the-buck". StageMaster cables utilize new product development and manufacturing procedures with new parts selection and certification to offer improved quality with dramatically lower prices. Perfect for practice, rehearsal, or the learning musician, the cables are constructed from quality materials and built to withstand everyday use. StageMASTER instrument cables are available in two versions: Champion and Crusader.

Each are 1/4˝ instrument cables with unbalanced tip sleeve configuration for guitar, synths, samplers and effects processors. The Crusader series is the best cheap guitar cable on earth and is good for patch work as well. The Champion series are inexpensive and reliable 1/4˝ guitar/unbalanced patch cables, with heat shrink on each end for strain relief.



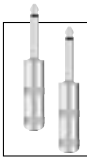
STAGEMASTER SERIES STRAIGHT 1/4˝ PLUGS

	1´	2´	3´	5´	10´	15´	18.5´	25´	30´
Champion				SLG-5 **6.99**	SLG-10 **7.99**	SLG-15 **8.99**	SLG-186 **9.99**	SLG-25 **10.99**	
Crusader	SEG-1 **7.19**	SEG-2 **7.39**	SEG-3 **7.59**	SEG-5 **7.99**	SEG-10 **9.49**	SEG-15 **Call**	SEG-186 **10.49**	SEG-25 **13.99**	SEG-30 **17.99**

EXCELLINES Series (Excellines, Silent Knight)



Engineered to accurately reproduce every sound you create, Excellines cables provide outstanding sound quality and durability. The world standard "workhorse" cable, Excellines feature Proco's 120S1 single-shielded 26-strand, 20-guage high-purity copper conductor with a 100% coverage extruded barrier PVC shield plus 55% coverage copper spiral shield to flawlessly deliver the full range of your signal with superior transient response. Available with G&H's 1/4˝ BigFoot phone plug and RightFoot right angle plugs. For those who want to switch their guitars on stage and live, without the pop, the Silent Knight includes all the features of the Excelline with Proco's specially engineered silent plug.



G&H 1/4˝ BigFoot straight phone plug to 1/4˝ BigFoot straight phone plug.

EXCELLINES CABLES WITH STRAIGHT PHONE PLUGS

1´	2´	3´	5´	10´	15´	18.5´	25´	30´
EG-1 **11.19**	EG-2 **11.69**	EG-3 **12.09**	EG-5 **12.59**	EG-10 **13.49**	EG-15 **14.99**	EG-186 **15.29**	EG-25 **17.09**	EG-30 **17.99**

Excellines Angled to Straight 1/4˝ Plugs

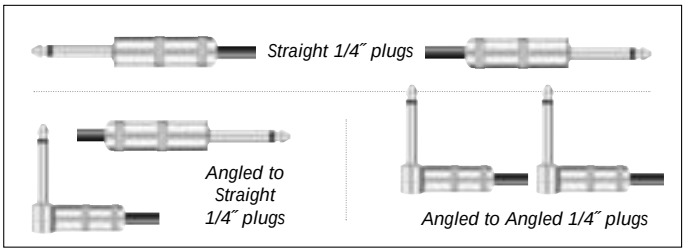
EGL-10 10´ .....**16.49** **EGL-186** 18.5´ **19.99**

Excellines Angled to Angled 1/4˝ Plugs

EGLL-06 .5´.....**16.49** **EGLL-1** 1´.....**16.49**

Silent Knight Straight 1/4˝ Plugs

EG-10 10´ .....**15.49** **EG-15** 15´ **17.49**
EG-20 20´ .....**19.99** **EG-25** 25´ .....**21.99** **EG-30** 30´.....**24.49**

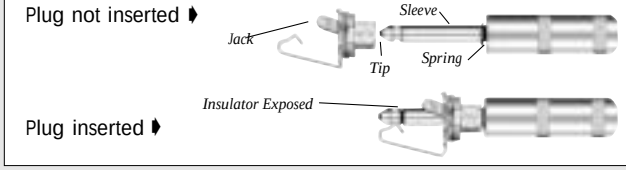


Silent Plugs

Proco's Excellines ¼˝ instrument cables are available with a Silent Plug which automatically breaks the contact between the tip and the sleeve of the ¼˝ jack when unplugging a cable from active equipment.

HOW THE SILENT PLUG WORKS

Before inserting the plug, a spring keeps the sleeve and tip in contact. When inserted in a jack, the pushing action slides the sleeve back, breaking the contact between the tip and sleeve.



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INSTRUMENT CABLES

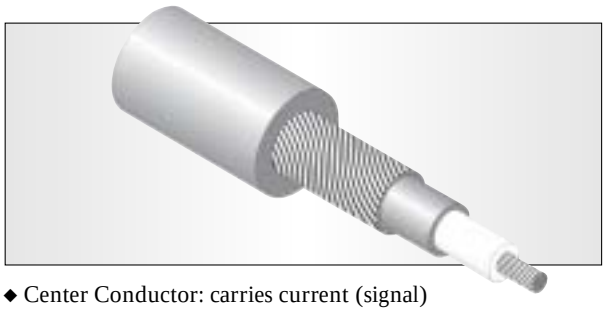
LifeLines Series (Excalibur, StarGuards, LineCable)



Ideal for musical instruments and unbalanced equipment hookup and patching, ***Excalibur*** *and* ***StarGuards*** *both use Proco's low-capacitance 20-gauge 120SX wire with a 90% spiral and 100% semi-conductor barrier shield to provide a clean, low-noise performance. In addition, the Excalibur instrument cables feature Proco's exclusive ArmorFlex oversized outer jacket for increased mechanical strength and improved flexlife, as well as gorgeous high luster 1/4˝ nickel phone plugs and barrel. The* ***StarGuard*** *comes with a BigFoot silent plug that allows you to change your electric guitar to acoustic/electric guitar, silently! (Excalibur cables are also available with or without BigFoot silent plugs).*

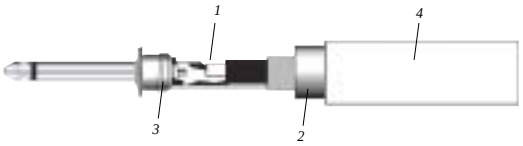
The original ground lifted instrument to amp cable, the ***LineCable*** *series features a telescoping shield configuration whereby the shield is lifted at the instrument and tied at the amp for lower noise. The LineCable series also incorporate 's 223B wire with bi-wire technology. Normally found in low-impedance mic cables and balanced line instrument connections, this two-conductor shielded wire helps reduce noise from single coil pickups. The cable offers 94% coverage copper braid shield and is also available with right-angle plugs.*

Anatomy of The Instrument Cable



- Center Conductor: carries current (signal) from the source (instrument)
- Insulation: separates the conductor from the shield
- Extruded Barrier: reduces handling noise
- Shield: helps to prevent outside interference (from lights or AC) from entering the audio signal
- Outer Jacket: for protection and appearance

The LineCable Process



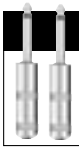
LineCable instrument cables utilize a unique construction process to seal inner conductors along with the solder connection, in an inert plastic resin. The cable is reinforced with two layers of thermal compression sleeving for protection and the ultimate in noise rejection.

1. Solder connections are permanently encapsulated for extra strength and insulation

2. Heat-shrink tubing is applied over the plug's strain relief for added reinforcement at the flex point

3. Metal bonding cement is applied to the plug's threads to weld the metal parts together

4. An outer layer of protective heat-shrinkable tubing is applied to complete the double-sleeved strain relief

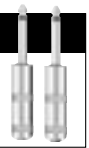


EXCALIBUR NICKEL PHONE PLUGS				
10´	**15´**	**20´**	**25´**	**30´**
XG-10 **39.95**	XG-15 **41.95**	XG-20 **44.95**	XG-25 **45.95**	XG-30 **49.95**



STARGUARDS WITH SILENT PLUGS				
10´	**15´**	**20´**	**25´**	**30´**
SG-10 **24.99**	SG-15 **27.99**	SG-20 **31.49**	SG-25 **34.99**	SG-30 **37.99**

LINECABLE STRAIGHT PLUGS						
5´	**10´**	**15´**	**18.5´**	**20´**	**25´**	**30´**
LC-5 **14.99**	LC-10 **20.99**	LC-15 **25.49**	LC-186 **28.99**	LC-20 **Call**	LC-25 **34.99**	LC-30 **39.999**



LINECABLE STRAIGHT & ANGLED PLUGS					
10´	**15´**	**18.5´**	**25´**	**30´**	
LC-10L **24.99**	LC-15L **27.99**	LC-186L **31.49**	LC-25L **34.99**	LC-30L **37.99**	



INSTRUMENT CABLES

Guardian Series *(Black Jack, Defender, Sir Tweed)*

Black Jack – *Elegant and Bullet-Proof*

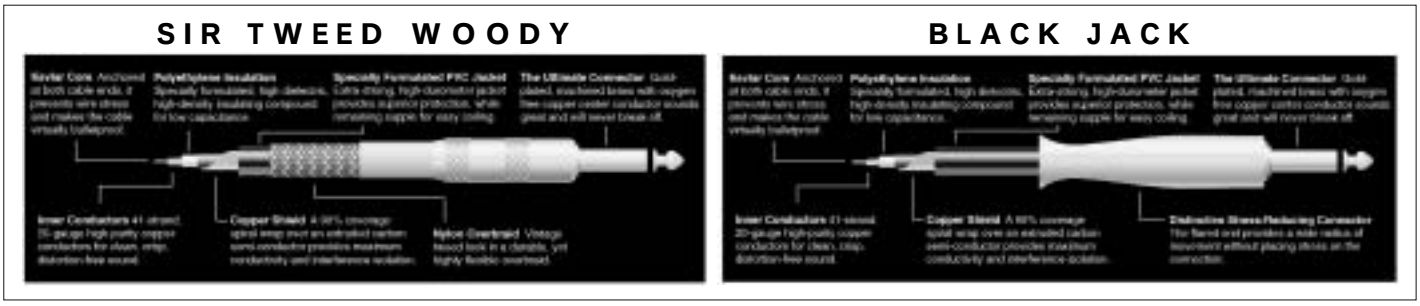


If you're in need of an elegant, great-sounding instrument cable that will last a lifetime, Black Jack is the one! Kevlar reinforced for unmatched strength and dependability, Black Jacks are low inductance, ultra-flexible cables which provide tight bass, unmasked midrange and clean, open highs. Unlike highly-resistive steel inserts and tubing in other products, Black Jacks feature custom G&H connectors with gold plated, machined brass components and an oxygen free copper center conductor which will never break off.



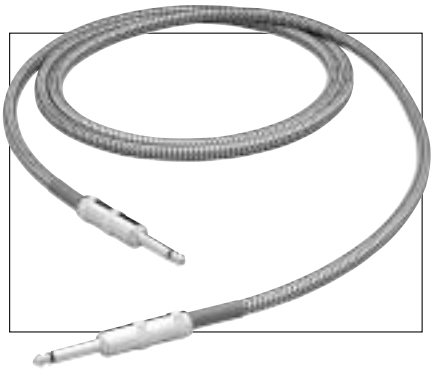
Defender – *The Humbucking Guitar Cable*

Designed to be the last instrument cable you'll ever need, Defender is the best sounding, most reliable cable you can buy today. Utilizing Proco's hum-bucking AmeriQuad bi-wire geometry, which reduces noise up to 20dB over standard designs, Defender can quiet even noisy single coil and hum-bucking guitar pickups. Kevlar reinforced for unmatched strength and dependability, this cable features the unique Lifelines construction process and is terminated with gold-plated, machined brass tip and sleeve with an oxygen free copper conductor.



Sir Tweed, Sir Tweed Woody –

Vintage look... unsurpassed flexibility



Great looks, great sound and indestructible construction- Sir Tweed is the choice for guitarists who want the best cable available. With a true nylon overbraid and Kevlar reinforced core, Sir Tweed is built for a lifetime of use and abuse. The 41-strand center core conductor and copper shielding provide superior signal flow and noise rejection with low capacitance. Sir Tweed is terminated with the best connectors made; gold plated machined brass with a solid, oxygen free copper conductor in the center that will never break off.

With the same great specs as the original Sir Tweed, the Sir Tweed Woody features a solid-wood plug barrel for added distinction. The 41-strand center core conductor and copper shielding provide superior signal flow and noise rejection with low capacitance.

INSTRUMENT CABLES

Black Jack

- Kevlar core anchored at both cable ends prevents wires stress and makes the cable virtually bulletproof. Kevlar strands intertwined with the center conductor provide unbelievably strong strain relief and prevents cables from stretching and breaking over time. A 20-ft. cable without Kevlar, can actually stretch about 6″ over a few years.
- Standard instrument cable uses a 20 guage center conductor with 26 strands of 34 AWG wire. Black Jack uses 41 strands of 36 AWG high purity copper wire, slightly thicker than a hair on your head, for clean, crisp, distortion-free sound.
- The design also ensures superior flexibility, reducing leakage capacitance to offer stable impedance and better sonic performance, especially in the top-end.
- 90% coverage spiral wrap over an extruded carbon semi-conductor provides maximum conductivity and interference isolation.
- Specially formulated Polyethylene insulation not only contributes to a cables flexibility and durability, it also significantly contributes to its sonic quality. Guardian cable's jacket, insulation layers and semiconductor shield layer are optimized for optimal sonic performance — a tight low end, unimpeded highs and imperceptible phase shift.
- Specially formulated, extra-strong PVC jacket provides superior protection, while remaining supple for easy coiling
- Distinctive flared end provides a wide radius of movement without placing stress on the connection.
- 1/4″ gold-plated, machined brass connector with oxygen-free copper center conductor sounds great and will never break off.

The Benefits of Kevlar

There are typically two places where a guitar cable will break. One is at the connector. Constant tugging and stretching (and trying to whip a cable out of the amp from 5 feet away) will put continuous stress at the delicate solder points. Having Kevlar strands intertwined with the center conductor provides an unbelievably strong strain relief. The other place most prone to abuse is about two inches down from the connector. By wrapping their cable around their strap button night after night, an an unbelievable amount of stress is put on a very small section of the cable, stretching it over time and eventually breaking it. The unstretchable Kevlar will keep the cable from stretching and breaking over time.

A by-product of the Kevlar which isn't as apparent is that it actually keeps the cable from becoming noisier as it becomes older. How? A normal cable itself will actually stretch over time. Over the course of a few years a 20ft guitar cable may actually stretch about 6 inches. Besides the obvious stretching of the center conductors, the shielding will also stretch and, therefore, become thinned out. A cable with a shield covering 95% of the wire may only have a coverage of 92% after a period of time, increasing the cable's susceptibility to noise.

Defender *(Same as Black Jack except)*

- Designed to be the last instrument cable you will ever need—ideal for use with noisy single coil pickups.
- AmeriQuad Conductor configuration--24-gauge, high-purity copper conductor are bi-wired in two pairs to reduce inductance and cancel noise for crisp distortion-free sound—reduces noise up to 20dB over standard designs
- Organic Fiber – Low-dielectric insulation holds the critical geometry of the conductor to minimize capacitance and reduce interference
- Specially braided copper shield provides 96% coverage for the best possible reduction of exterior noise
- Lifelines construction

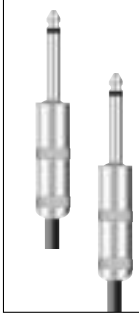
All Guardian Cables include a Lifetime Warranty – any excuse...even abuse

Sir Tweed and Sir Tweed Woody

(Same as Black Jack except)

- Vintage tweed look in a durable, yet highly flexible overbraid. The rich-looking nylon overbraid matches exactly the color and heritage of the 1960's vintage amplifier coverings
- Sir Tweed terminates with gold handles, Sir Tweed Woody (TW01) has oak wooden handles and is plain with no grooves. Sir Tweed Woody (TW02) has oak wooden handles and grooves.

GUARDIAN SERIES STRAIGHT 1/4″ PLUGS



	10′	15′	20′	25′
Black Jack	BJ-10 **Call**	BJ-15 **Call**	BJ-20 **Call**	BJ-25 **Call**
Defender	DF-10 **54.95**	DF15 **59.95**	DF-20 **64.95**	DF-25 **69.95**
Sir Tweed	TW-10 **54.95**	TW-15 **59.95**	TW-20 **69.95**	TW-25 **74.95**
Tweed Woody	TW01-10 **Call**	TW01-15 **Call**	TW01-20 **Call**	TW01-25 **Call**
Tweed Woody with Grooves	TW02-10 **Call**	TW02-15 **Call**	TW02-20 **Call**	TW02-25 **Call**

PATCH CABLES

Standard, Dual, Y and MIDI Patch Cables

Proco produces patch cables for nearly every application imagineable, in dual, "Y" and single-line configurations as well as speciality cables for MIDI and computer applications.



MIDI+

- Solid-shell DIN connectors only
- Excellines 424F (four 35 gauge connectors, 100% foil shield) digital transfer wire
- 5-pin DIN to 5-pin DIN plug (all five pins connected)
- Available in lengths from 3 to 50 feet.
- 10 Year Warranty

MIDI-3	3′	14.99
MIDI-5	5′	14.95
MIDI-10	10′	16.49
MIDI-15	15′	16.49
MIDI-20	20′	17.95
MIDI-25	25′	19.95
MIDI-50	50′	27.95

Single Unbalanced Patch (P) Cables

Proco patch cables were designed to connect unbalanced signals in today's sound and recording systems. They are constructed using Proco 120SM ArmorFlex single conductor spiral shield wire with ShowSavers 1/4″ plugs and/or Proco HD (heavy duty) RCA connectors. Available in QR (1/4″ to RCA) and RR (RCA to RCA) configurations.

- Reliable interconnects for line-level patching.
- 1/4″ BigFoot Phone Plug
- Heavy Duty RCA Phono Plug
- 20 AWG Proco 120SM wire in black
- 100% coverage extruded barrier PVC shield plus 70% coverage copper spiral shield
- 10 Year Warranty



RR - RCA to RCA		
PRR-3	3′	13.95
PRR-5	5′	14.49
PRR-10	10′	15.95
PRR-20	20′	16.95
QR - 1/4″ to RCA		
PQR-3	3′	14.95
PQR-5	5′	15.95
PQR-10	10′	16.95
PQR-20	20′	17.95

Dual Unbalanced Keyboard (DK) Patch Cables

Dual keyboard cables are designed to connect today's stereo keyboard outputs to mixers for stage and recording use. They are constructed of two single conductor cables "twinned" together for ease of use and are offered with ShowSavers 1/4" phone plugs or Proco HD RCA male ends to accommodate a variety of input and outputs. Available in QQ (1/4″ to 1/4″), RR (RCA to RCA) and QR (1/4″ to RCA) configurations.



- High quality, "low crosstalk", stereo interconnects
- 1/4″ BigFoot Phone Plug
- Heavy Duty RCA Phono Plug
- Color-coded Left/Right
- 2 x 20 AWG Proco 2120SM parallel construction wire in black
- 100% coverage extruded barrier PVC shield plus 70% coverage copper spiral shield
- 10 Year Warranty

QQ - 1/4″ to 1/4″		
DK-3	3″	21.95
DK-5	5′	22.95
DK-10	10′	25.95
DK-20	20′	29.95
QR - 1/4″ to RCA		
DKQR-3	3′	20.95
DKQR-5	5′	21.95
DKQR-10	10′	24.95
DKQR-20	20′	29.95
RR - RCA to RCA		
DKRR-3	3′	19.95
DKRR-5	5′	20.49
DKRR-10	10′	22.95
DKRR-20	20′	27.95

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ONE HOUR FREE PARKING
AT 349 W. 34th STREET (with purchase of $100 or more)

PATCH CABLES

Military Spec PJ (Phone Jack)-Series Patch Cables

The phone company invented the phone plug in 1885, a long time ago. They are built differently than the 1/4″ connectors you are used to seeing in audio, but perform exactly the same function in military type jackfields. These 1/4″ tip/ring/sleeve (TRS) mil spec cables use Neutrik's especially designed connectors and Proco's 424BK wire to make these sturdy, long-life assemblies. These connectors are typically used in theatres and Broadway shows and are not commonly used in the portable world.



- Premium quality cable assemblies for military-type jackfields.
- Neutrik NP3TB 1/4″ Military Spec TRS Phone Plugs
- 4 x 24 AWG Proco 424B "AMERIQUAD" Wire in Black, Brown, Red, Orange, Yellow, Green, Blue, Violet, Grey, and White
- 96% Coverage Copper Braid Shield
- 10 Year Warranty

PJ-2	1/4″ to 1/4″	2′	29.95
PJ-3	1/4″ to 1/4″	3′	31.95
PJXF-2	1/4″ to XLRF	2′	31.95
PJXF-3	1/4″ to XLRF	3′	32.95
PJXM-2	1/4″ to XLRM	2′	31.95
PJXM-3	1/4″ to XLRM	3′	32.95
PJBQ-2	1/4″ to 1/4″ TRS	2′	31.95
PJBQ-3	1/4″ to 1/4″ TRS	3′	32.95

TT (Tiny Telephone or "Bantam" Plugs) Cables

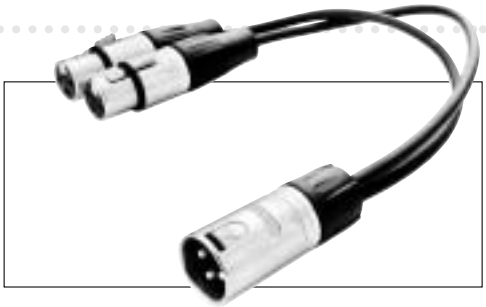
Most professional recording studios TT TRS patchbays because they offer 96 patch points in a single-space 19″ rack panel. Proco's TT cables are ShowSavers Bantam plugs and Proco miniature wire to perform this important patching function. Use for high density needs where rack space is at a premium. Available in TT to TT versions, and also TT to XLR male and female connectors or 1/4″ TRS phone plugs.

- Premium quality cable assemblies for Bantam/Tini-telephone jackfields.
- Neutrik NP3TT 4.4mm TRS Bantam/"Tini-Telephone" Plugs
- 2 x 24 AWG Proco 224SM miniature mic cable with spiral shield (in Black)
- 80% Coverage Copper Braid Shield
- Standard lengths include 2 and 3′ lengths
- 10 Year Warranty



TT-2	1/4″ to 1/4″	2′	24.95
TT-3	1/4″ to 1/4″	3′	26.95
TTXF-2	1/4″ to XLRF	2′	25.95
TTXF-3	1/4″ to XLRF	3′	26.95
TTXM-2	1/4″ to XLRM	2′	25.95
TTXM-3	1/4″ to XLRM	3′	26.95
TTBQ-2	1/4″ to 1/4″ TRS	2′	25.95
TTBQ-3	1/4″ to 1/4″ TRS	3′	26.95

"Y" Cords



Proco makes standard "Y" cords, the most basic of which are constructed with one 1/4″ phone plug (male or female) to two 1/4″ phone plugs (male or female). The same concept can be built using male and female XLR connectors and RCA phono plugs. Uses 2120SM or 224SM Proco patch wire with spiral shield (100% coverage extruded barrier PVC shield plus 70% coverage copper spiral shield) and ShowSavers plugs, Neutrik XLRs and Proco HD (Heavy Duty) RCA connectors. 10-year warranty.

YGQF2Q-1: 1′ 2120 SM patch cable with a 1/4″ female inline phone jack to two ShowSavers 1/4″ phone plugs

YGQ2QF-1: 1′ 2120 SM patch cable with a ShowSavers 1/4″ phone plug to two 1/4″ female inline phone jacks.

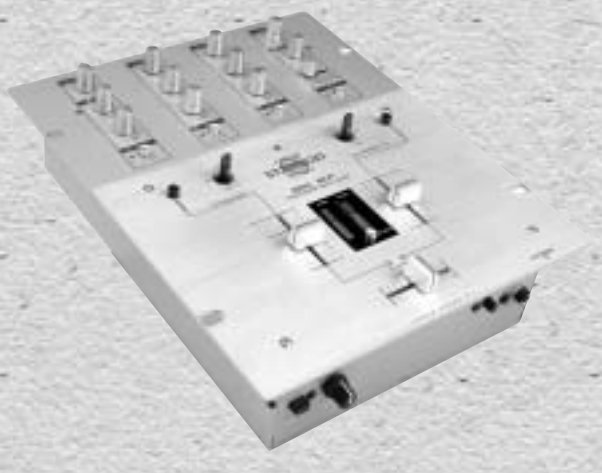
YMXF2XM-1: 1′ 224SM patch cable with a Neutrik XLR female to two Neutrik XLR males

YMXM2XF-1: 1′ 224SM patch cable with a Neutrik XLR male to two Neutrik XLR females

YGQ2R-3: 3′ 2120SM patch cable with a ShowSavers 1/4″ phone plug on one end to two Proco HD RCA phono plugs on the other.

YGQ2R-5: 5′

YGQ2R-10: 10′




Pioneer


Numark
Numark


DENON
DN-D3500
DENON
DN-D3500
DENON
DENON

B&H

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The ProfessionalAudio SourceBook

Section15

DJ Equipment

Denon 1526-1529

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Grundorf.....1534-1535

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Shure 1554

Stanton 1555-1565

Tascam 1566-1569

Vestax.....1570-1579

DN-X400

Professional Digital/Analog DJ Mixer



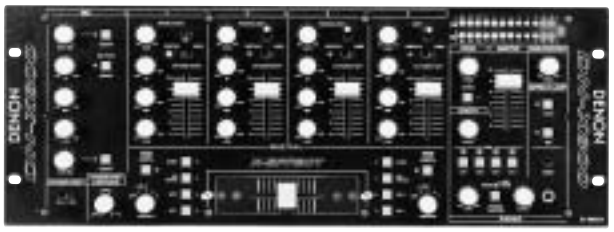
Based on the layout and technology of Denon's DN-X800 digital mixer, the DN-X400 takes full control of advanced mixing and performance features. With two digital outputs, the DN-X400 makes it possible to record directly to the Denon DN-C550R dual CD recorder/player, as well as mark tracks on-the-fly with the Track-Mark function. Also included are Input Gain controls, along with HI, MID and LOW tone controls (-26dB ~ +10dB) on all four main channels, including the mic channel, for further manipulation and tweaking of input signal.

- 10 inputs for connecting to multiple devices- 8 line inputs, including 2 phono connections, and 2 balanced mic inputs (1/4″ TRS) with Post on/off switches.
- A balanced Send/Return effects loop provides even more flexibility when connected to outboard effects and other devices.
- Six outputs include 2 digital outs (coax and optical S/PDIF), balanced Master and Zone, and unbalanced Master and Tape outputs.
- A subwoofer output is provided along with a control for crossover frequencies to be sent to the sub for greater sonic precision.
- High-quality VCA style 45mm slide faders for precise control
- Cross fader assignment selector and contour level controls allow for smooth and stable transitions between program material, while Cue and Program pan controls give the user even greater sonic separation capabilities.
- 3-band line EQ (-26dB – +10dB) and 3-band mic EQ (±12db)
- Cross and channel fader-start trigger is possible and compatible with Denon CD players
- For easy visibility and instant recognition, a split cue H/P monitor and display with a Zone meter adjustment are included.
- Subwoofer output can also be used to remotely trigger any lighting system that accepts an audio input.

Designed with today's cutting edge DJ requirements in mind, the DN-X400 and DN-X800 offers high-quality playback along with a multitude of input and output options in a single four rack-space unit.

DN-X800

Professional Digital/Analog DJ Mixer with X-Effect



The DN-X800 steps up (no Track Mark) from the DN-X400 with full control of advanced mixing capabilities and performance features found on Denon's DN-1800F, DN-2100F and DN-2600F CD players. X-Effects such as Denon's Hot Start, Fader Start, Brake and Sampler functions can be controlled through through the DN-X800.

- The DN-X800 features 14 inputs including 4 digital S/PDIF, as well as 8 line and 3 phono connections. 2 mic inputs (balanced and unbalanced with Post on/off switch) and a balanced Send/Return effects loop provides even flexibility when connected to outboard effects and other devices.
- Eight outputs include 2 digital S/PDIF, balanced Master and Zone, and unbalanced Master, Booth and Tape outputs.

What is X-Effect?

X-Effect means enables you to manipulate the special features of Denon's Double CD Players from the DN-X800. Now you can trigger the Sampler feature of the DN-2600F, and the Hot Start & Brake effects of both the DN-2600F and DN-2100F, which means you don't need to use both hands to operate the CD player and Cross Fader at the same time.

- Sampler Trigger (DN-2600F)
- Brake/Platter-S Effect Trigger for Hot Start (DN-2100F/2600F)
- Hot-Starts Trigger (DN-2100F/2600F)
- Simple (5v pulse) Fader-Start Trigger (DN-1800F/2100F/2600F)

DN-S5000

Tabletop CD Player

The DN-S5000 offers a number "firsts" for the digital DJ. A "motor driven active platter" provides the ultimate feel for true vinyl emulation, while the unique "Alpha Track" feature lets you can access different tracks from the same CD simultaneously. You can even "mix" the tracks. And in the "Hot Disc" mode, you can eject a disc and it will still play for another 35 seconds, while you cue up your new disc.
The DN-S5000 will let you control many aspects of your performance as if you had two decks playing, when in fact, you're only running the one. Combine two DN-S5000s and now you have a true digital powerhouse! Using the DN-X800 (connected through the built-in X-Effect Ports) you can connect two DN-S5000 and gain additional fader control, exchange Memo data and control various play modes of the two decks. Of course, there is an array of digital and analog inputs/outputs, as well as numerous dedicated controls.



FEATURES

- Motor-driven active platter for true vinyl emulation
- Alpha Track (2 in 1) allows independent and simultaneous playback of another track from the same disc, mix tracks with one drive instead of 2
- Hot Disc lets you eject a disc during play and have it continue for another 35 seconds, and you can loop within Hot Disc memory
- 3-way Scratch lets you apply scratch effects by selecting Main, Alpha or Sampler Track as the source
- Mirror Mix lets you select the same track in Main and Alpha Track and play it after a specified time delay
- Clear Scratch Disk allows you create your own custom slip mat designs
- Two DN-S5000s can be connected to a DN-X800 Mixer via the X-Effect ports and operated by the faders. Two DN-S5000s can also view each other's Memo data (when connected via X-Effect I/O ports)
- Relay Play, Program Relay Play, Random Relay Play are possible when two DN-S5000s are connected together
- Independent digital outputs (coax, S/PDIF) for Main and Alpha Track. Constant digital output at 44.1kHz is possible even when Playback Speed is changed
- With digital connection all effects can be output
- Dedicated Alpha Track Fader Start port
- Manual input of BPM values (*AUTO BPM and Manual TAP are also possible)
- Switchable display of Main and Alpha Track times
- 100 mm long slider for playback speed control
- Four Hot Start sounds can be played
- Four seamless loops can be made
- A-B Splice removes unwanted sections of a track seamlessly (2 points per track)
- The A & B points for the seamless loop, sampler loop, and splice points can easily be adjusted using the Scratch Disc
- Six Stutter Cue Play Points
- Two samplers (approx 15 seconds in 20Hz - 20kHz, 16-bit, 44.1kHz quality)
- Recessed connecting terminals on the back panel to avoid damage
- Up to 5,000 loops and cues can be stored internally and/or copied to another DN-S5000. Memo from a DN-D9000 or DN-2600F can also be copied to a DN-S5000
- Instant Start (0.2 sec)
- CD-R/RW disc playback
- Program/Random Play - program up to 25 tracks on a CD for playback and the programs for up to six discs can be stored in memory
- Random Play of CDs loaded in both drives is also possible
- Pitch Control: ±4%, 10%, 16%, 24%, 100%
- Shock Proof Memory - Main 16 seconds, Alpha Track 16 seconds

*Customizable Preset Functions (*Default)*

- Auto Cue: ON*/OFF
- Pitch Range: ±4 / ±10* / ±16 / ±24%
- ALPHA-Track Output: Main out* /(Alpha) Monitor out
- Platter: Turn round* / Turn stop
- Platter Mode Memo
- Auto Load : OFF* / 10 / 30 / 60 seconds
- EOM : OFF /10*/15/20/30/60/90 seconds
- Digital Output Level : Normal* / -6 dB Down
- Fader : PlayCue* / Play/Pause
- Power On Play : ON / OFF*
- A3/4 MODE : Sampler*/4H/S/Alpha Track H/S
- Single Continuous : Single* / Continuous
- Elapsed/Remain: Elapsed* / Remain
- Program All Clear: YES/NO*
- Memo All Clear: YES/NO*
- Memo All Copy: YES/NO*
- Preset Clear: YES/NO*

DENON

DN-2600F

DJ EQUIPMENT

1528

Dual CD Player



A top-of-the line DN-2600F incorporates two Hot Starts per drive-perfect for instantaneously changing a track or Cue point at the touch of a button. The same Hot Start buttons can also be used to control the machine's Stutter' mode, which plays momentary audio as the buttons are pressed down and then recues to the original point when they are released. An 'Effector' mode enables DJs to bring a whole new level of expression to their performances by offering fully customizable frequency filter, reverb and flanger effects. The latter two of these effects may be synced to the music's beats per minute count via the Tap/BPM button, producing reverb decays and flange sweeps that coordinate perfectly with the tempo of the playback material.

FEATURES

- ◆ Up to 16 seconds of CD sound can be sampled and played back at any time, even when the source CD has been removed. This sample can be played or looped in forward or reverse direction. Pitch, output level and key adjustment functions can all also be used to manipulate the sample.
- ◆ The DN-2600F offers a collection of turntable effects to the digital domain:
- – Real-time 'Digi-Scratch' effect allows DJs to simulate analog vinyl scratching on the fly.
- – "Platter Start" and "Platter Stop" emulates the effect of starting and stopping playback on a turntable.
- – Rotating the jog dial in "Platter Hold" mode duplicates the sound of speeding up or slowing down a record by changing the player's pitch control.
- – Brake effect, Key Control and Key Adjust are controllable via the jog dials.
- – Pitch Bend may also be activated via the jog dials (±32%), as well as on dedicated buttons (±18%) and sliders.
- ◆ Most buttons including Cue, Play/Pause, Loop A1, A2 and B, and Open/Close are backlit for maximum visibility and ease of use in low light environments. In addition, the large fluorescent tube displays and numerous LEDs give the DJ a clear indication of the player's status at a glance.
- ◆ Takes A-B Seamless Looping one step further by offering two completely independent loops per drive. 'Next Track Reserve' function provides seamless track jumping.
- ◆ Extensive Custom memory allows users to store Hot Starts, A-B Loop points and Pitch percents on up to 300 tracks for later recall.
- ◆ Rotary Track Selector/Data Master knobs enable users to perform a wide variety of functions quickly and intuitively.
- ◆ Large jog dial/shuttle wheels offer smooth searching, scanning and effects controlling.
- ◆ Drives 1 and 2 are equipped with independent Fader Start input
- ◆ Pitch range of ±10% or ±16% adjustable in 0.1% steps
- ◆ Ten seconds of 'shock-proof' memory per drive guards against interruptions due to external mechanical shocks such as bumps, deal for nightclub-installed applications.
- ◆ Playback modes include Program, Continuous, Single and Random Play
- ◆ Auto cue to music eliminates silent portion at the beginning of CD tracks

DN-D4000 *Double CD/MP3 Player*



The affordable DN-D4000 offers extensive MP3 support including VBR and CBR, folder/file search, ID3-Tag display (Artist, Album & Title) among a host of innovative features:

- CD Text display
- Track and disc memo functions (300 tracks per drive, 10 discs per drive).
- ±4%, ±10% or ±16% pitch control
- 2 seamless loops per drive and 2 hot starts and stutter cue points per drive.
- ±18% pitch bend
- S/PDIF digital output
- 50mm jog/shuttle wheel
- Fader start, Instant start
- 10 sec. CD /100 sec. MP3 shock proof memory
- Relay, Program, Repeat Play, Power on Play
- 2-way Pitch Bend -by Jog Wheel and buttons

B&H PHOTO - VIDEO - PRO AUDIO EQUIPMENT LEASING AVAILABLE

DN-9000

Dual CD Player



The latest in Denon's innovative line of DJ products, the DN-9000 offers many proprietary and "world's firsts" for professional CD players. These include ALPHA Track Play/Mode, A-B Slice function, the ability to create four Hot Start points per drive (Start Point A1, A2, A3 and A4), and the addition of up to 6 Stutter Cue play points. Besides the full sampling capabilities, the DN-D9000 has a built-in Effector function, providing 6 effects (up to four at a time), including Delay, Flanger, Filter (2 types), and Transform (2 types), all fully interruptable instantaneously using the Effector's Bypass button.

FEATURES

- ALPHA Track Play/Mode allows independent and simultaneous playback of two tracks from the same disc, allowing manual mixing of two tracks (Sub and Main), using only one deck.
- This ALPHA track playback allows for mixing of the Sub and Main tracks, as well as an integrated ALPHA track Pre-Monitoring function for monitoring the Sub track during Main track playback in real-time.
- A-B Splice function removes unwanted sections of a track seamlessly on the fly
- Not only can you scratch in real-time, you have the ability to scratch your Samples too with choice of directional sound movement. (Forward/Reverse)
- You can create four Hot Start points per drive (Start Point A1, A2, A3 and A4)
- 6 Stutter Cue play points
- 2 samplers per drive of the unit (approximately 15 seconds, CD quality).
- Create up to four seamless Loops within a track or disc, which can further be adjusted via the jog wheel during Loop playback.
- Constant Digital Out (44.1 kHz with subcode) even when playback speed is changed. And since the DN-D9000 can be connected to the DN-X800 DJ mixer over a digital interface, you won't lose any sound quality when you record your own CDs.
- Auto/Manual BPM Counter (count music BPM manually by simply tapping a button)
- CD-R/RW Disc Playback
- Activate any of the Sampler, Effector, or Scratch special effects at any time during playback. Jump among the three of them, or activate all three at once.
- Built-in Effector function provide 6 effects (up to four at a time), including Delay, Flanger, Filter (2 types), and Transform (2 types), all fully interruptable instantaneously using the Effector's Bypass button.
- Includes 5 Platter modes:

- Drag-S: Gradually raises playback speed to normal speed after you start playback.
- Brake: Gradually lowers playback speed to a stop when the Pause button is pressed.
- Echo: If Echo is ON when Pause is pressed, the playback sound is repeated over and over as it fades out to a stop.
- Reverse: Playback shifts into reverse when the Parameter knob is pressed during playback.
- Dump: Starts reverse playback for a specified time, then returns to normal playback.

- Denon's newly developed digital drives provide premium performance. However, for field service & instant maintenance, Denon offers user replaceable drives—never any downtime!
- Ergonomically designed, large 70mm illuminated jog wheel helps give a more precise scratch and solid feel to the unit, also for use with the Platter Hold function, allowing for extreme pitch bends of ± 100%.
- Memo function, for storing up to 5,000 points: Cue point, Playback Pitch, Pitch Range, Pitch ON/OFF, Key Adj ON/OFF, and Seamless Loop A and B data can be stored independently for Normal and track playback. This data can also be copied to another DN-D9000. MEMO from a DN-2600F can also be copied to a DN-D9000.
- Program Play (up to 25 tracks on a CD) and programs for up to 6 discs can be stored in memory. Random Play of CDs loaded in both drives is also possible.
- Single and continuous mode Relay Play: shifts playback to the CD in the other drive after the track or after playback of the the entire disc in the current drive ends.
- Pitch Control

– Main/Alpha Track: Max. ±100% (select from ±4, 10, 16, 24, 100%; ±4%: 0.05% step, ±10, 16, or 24: 0.1% step, ±100%: 1% step)

– Sampler 1/2: ±24%, 0.1% stepPitch Bend

- ±12% Pitch Bend (when variable pitch range is ±4 or 10%), ±26% (when variable pitch range is ±16 or 24%), ±100%(when variable pitch range is ±100%)

GEMINI

TURNTABLES

XL Series

The XL-Series professional manual turntables feature a start/stop function and run at 33 and 45 RPM. Headshell and a slipmat are included.



XL-120 and XL-200
Belt Drive Manual Turntables

- ±6% (XL-120) or ±10% pitch control
- CN-25 cartridge included
- 33/45 RPM
- Anti-skate feature, strobe illuminator
- S-shaped tone arm
- Feather light touch start/stop function
- Removable headshell & slipmat, dust cover

XL-300 and XL-500 II
Direct Drive Manual Turntables

- ±10% pitch control (with pitch bend on the XL-500 II
- Anti-skate feature, strobe illuminator
- 33/45 RPM, S-shaped tone arm
- Feather light touch start/stop function
- Removable headshell & slipmat, dust cover
- XL-300 includes CN-25 cartridge, XL-500 II is available with cartridge (**XL-500IIc**)

PT Series

High-torque direct drive quartz lock profesional turntables feature locking reverse button, pitch bend, arm-lift height adjustment, up to 7 grams of tone arm pressure, strobe illuminator, and included headshell.



PT-2100
Direct Drive Manual Turntable

- ±10% pitch control
- Soft-touch start/stop button



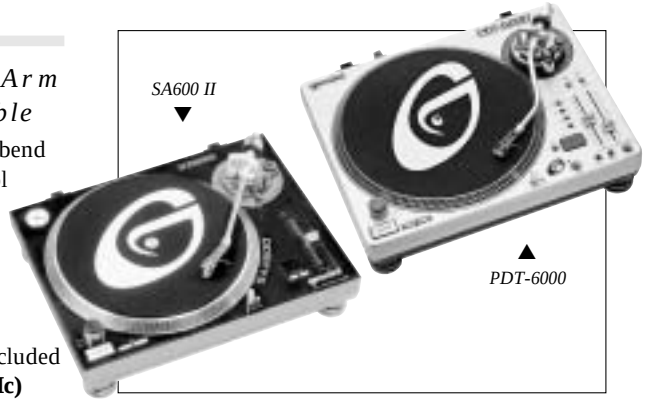
PT-2400
Direct Drive Manual Turntable

Same as the PT-2100 plus-

- LCD shows speed, pitch & platter direction
- 33, 45 and 78 RPM
- Pop-up target light
- Anti-skate adjustment

SA600 II
G-Force Straight Tone Arm Direct Drive Turntable

- ±10% pitch control with pitch bend
- Feather-touch start/stop control
- Pop-out Target light
- Strobe illuminator
- Anti-skate adjustment
- XLR lamp adapter
- Dual volt capability
- Angled Headshell & Slipmat included
- Available w/cartridge **(SA-600IIc)**



PDT-6000
Digital High Torque Turntable

- Digital direct drive manual turntable
- Super high torque motor
- Master tempo function
- Key controller
- Torque and brake sensitivity adjustment
- Pitch range of ± 4%, 8%, 16% or 35%
- BPM/pitch LCD display
- Reverse platter lock
- Digital and line level outputs

CD PLAYERS

PROFESSIONAL TOP LOAD CD PLAYERS

CDJ-15
DJ CD Player

- Instant start
- ±16% variable pitch control
- Frame accurate search
- Cue to music
- Single auto cue/continuous play button
- Mechanical anti-skip protection

CDJ-20
DJ CD Player with Anti-Shock

Same as the CDJ-15 plus-

- ±16% variable pitch control w/pitch bend
- 10-40 second anti-shock memory
- Direct access to track and cue point
- Jog Wheel

CD-400X
CD Player with Digital Effects

Same as the CDJ-20 plus-

- DSP w/ zoom, echo, filter and brake effects
- ±10% variable pitch control w/pitch bend
- Seamless looping, Fly cue
- Top-load with 10-40 second anti-shock
- S/PDIF digital output (BNC)
- Master tempo (change the tempo without affecting the pitch)
- Large jog wheel for seach, pitch bend and effects

SCRATCH PROFESSIONAL FRONT LOAD CD PLAYER

CD-1800X
CD Player with Scratch Effects

Same as the CDJ-400X plus-

- Real-time scratch effects
- 4 instant start cue buttons
- Pitch range of ± 4%, 12% and 16%
- Robo Start (relay play) 2 players needed
- Slot type CD Drive
- Direct access track selection & Auto BPM

19˝ RACKMOUNT PROFESSIONAL CD PLAYER

CDX-601
Single Transport CD Player

- Looping
- Auto-calibration
- Digital output
- Automatic disc calibration system (advance reads CDR's and adjusts the player for format problems)
- Instant start
- Frame accurate
- Jog wheel
- Digital output
- Robo start (relay play)
- Fly cue and pitch bend
- Pitch range of ± 4%, 8% and 16%
- Multi-speed jog search wheel

19˝ RACKMOUNT DUAL CD PLAYERS

CDX-602
Dual Transport CD Player

Same as the CDX-601 plus-

- Dual CD player

CDX-802
Dual Transport CD Player

Same as the CDX-602 plus-

- Seamless looping
- Anti-shock memory

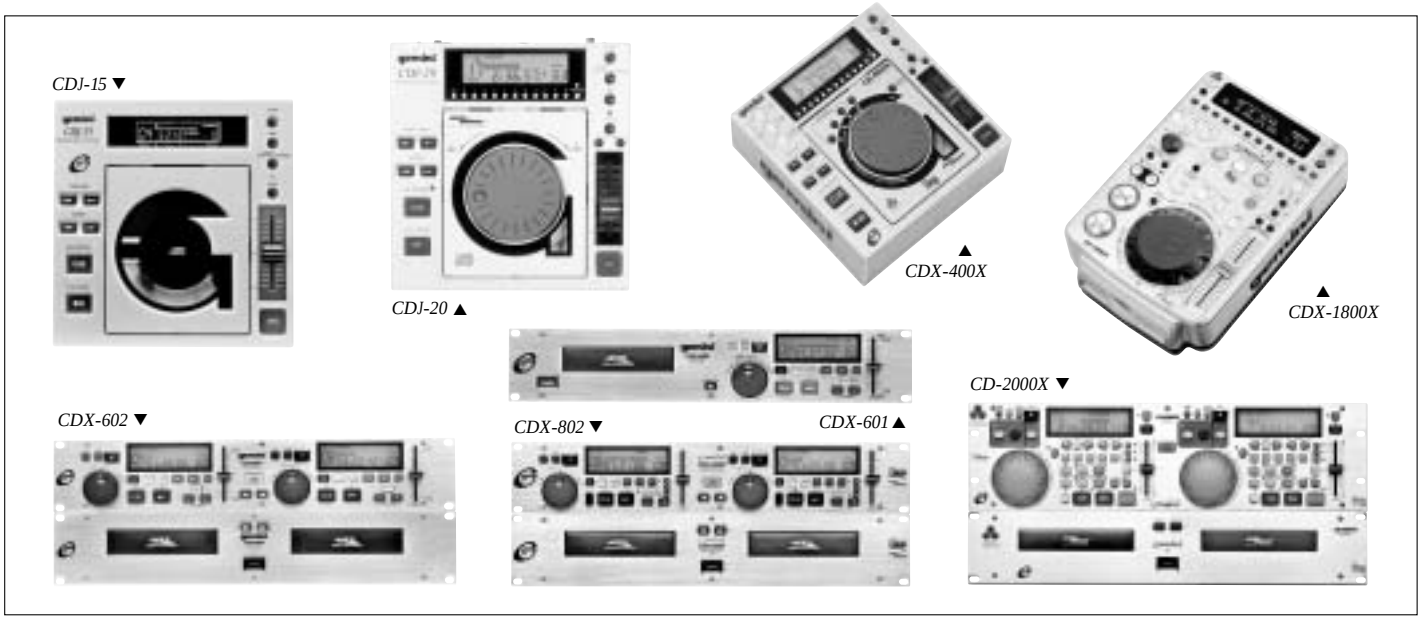
CD-2000X
3U Scratch CD Player

Same as the CDX-802 plus-

- Real time scratch effects
- Instant start
- Frame accurate
- Digital signal processing with zoom (flange), filtering, echo, reverse and braking
- 6 Instant start cue buttons per transport
- 2 Seamless loops per transport
- 8 second digital sampler per transport
- Anti-shock memory
- Master tempo (change the tempo without affecting the pitch)
- BPM counter
- 4%, 12%, 24% or 100% pitch range
- Large jog wheel for pitch bend, effects and digital scratching

PCD-10-1 One CD connecting cable .....**Call**

PCD-10-2 Two CD connecting cables.....**Call**



MIXERS

19″ RACK MOUNTABLE STEREO PRE-AMP DJ MIXERS

PMX-350 - 6.5″ DJ Mixer
- 2 channel stereo with gain & master controls
- 2 phono and 3 line inputs
- Mic input
- Removable RAIL-GLIDE crossfader
- Cue section with level control

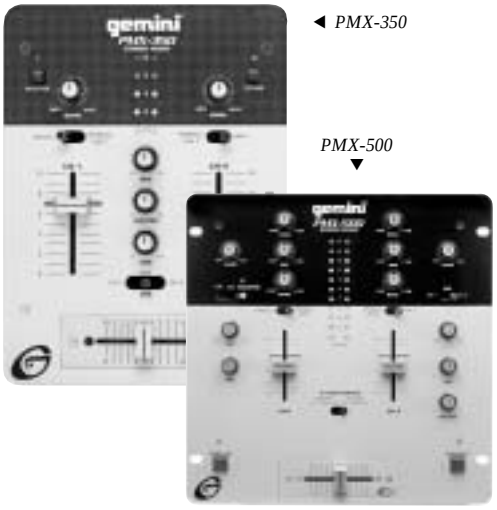
PMX-500 - 10″ DJ Mixer
- 2 stereo channels with bass, mid and treble EQ
- 3 phono/line convertible inputs
- Mic input
- Removable RAIL-GLIDE fader
- Cue with cue/program pan control
- Master and recorder outputs
- Punch in buttons
- Hampster switch

PMX-1100 - 19″ Full Size DJ Mixer
- Preamp, full size stereo mixer for the all purpose DJ
- 4-stereo channels with bass & treble EQ
- 3-phono/line convertible, 4 line inputs, 2 mic inputs
- Mic talkover feature
- Master and recorder outputs
- Easy access mixer with slide faders
- Removable RAIL-GLIDE crossfader
- State-of-the-art, push button cueing with cue/program pan control

PMX-1400 - 19″ Full Size DJ Mixer
Same as the PMX-1100 plus-
- 3-phono/line convertible, 5 line inputs, 2 mic inputs
- Dual 7-band graphics EQ
- Booth and record outputs
- Balanced and unbalanced master outputs

PMX-1800 - 19″ Full Size DJ Mixer
Same as the PMX-1400 plus-
- 6 sound effects (UFO, TRON, Blast, Siren, Gas, Machine Gun & Emergency) plus Echo section

PMX-2400 - 19″ Full Size DJ Mixer
Same as the PMX-1800 plus-
- 24 second digital sampler
- 5 memory banks
- Battery backup safeguards samples



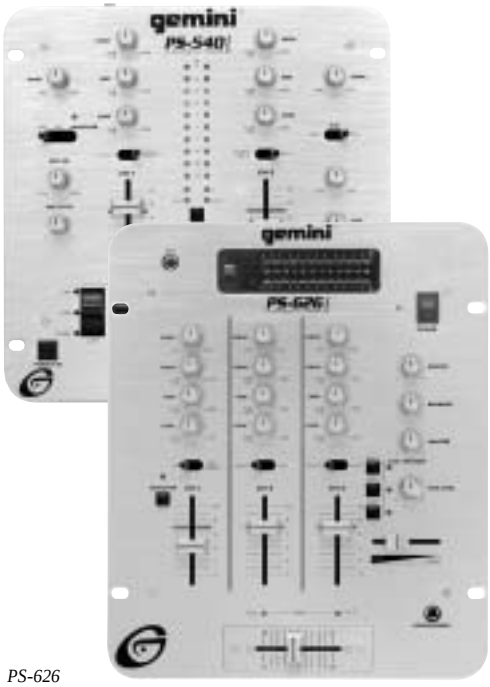
◄ *PMX-350*

PMX-500 ▼

PMX-1100



PS-450



PS-626

PLATINUM SERIES STEREO PREAMP DJ MIXERS

PS-540i -10″ Professional DJ Mixer
- 2 stereo channels with gain controls
- Kill switches for Bass, Mid and High frequencies
- 3 phono/3 line and 1 mic input with talkover
- Removable RAIL-GLIDE fader with curve control
- Punch-in buttons

PS-626i -10″ Professional DJ Mixer
- 3 channels with gain controls
- Bass, Mid and High with -32dB cut
- 3 phono/3 line and 1 mic input with talkover
- Beat sync indicators
- Removable RAIL-GLIDE fader with curve control
- Curve control and hampster switch
- Booth and balanced outputs
- State-of-the-art push-button cueing with cue/program slide control

PS-676i -10″ Stereo DJ Mixer
Same as the PS-626i plus-
- 12-second digital sampler with pitch control

PS-700i - Full Size 19″ Stereo DJ Mixer
- 4 channels with gain, bass, mid and high tone controls
- -32dB cut for Bass, Mid and High on each channel
- 3 phono/8 line and 3mic inputs with talkover
- Effects loop
- Balanced and unbalanced master outputs
- Master, booth and record outputs
- Dual mode display
- Assignable removable RAIL-GLIDE fader
- Balance Control,

PS-900i - Full Size 19″ Stereo DJ Mixer
Same as the PS-700i plus-
- 24 sec. digital sampler with 5 memory banks consisting of 2x2 sec, 2x4 sec and 1x12 seconds

MIXERS/CD PLAYERS

CS-19/CS-19R
19˝ Professional Stereo Club Preamp DJ Mixers

- 60mm channel slide controls (CS-19) or rotary channel controls (CS-19R)
- 5 Channels with gain control
- 4 Phono / 8 Line / 3 Mic inputs with talkover
- Bass, Mid, Treble with -32dB cut
- Master level attenuation
- Auxiliary & loop input / output on top panel
- Outputs: Main, 2 Zones & Booth
- Push button cueing with cue program rotary control
- LED meters for each channel
- Crossfader curve switch
- Balanced XLR & 1/4˝ outputs
- Assignable PRO RAIL-GLIDE crossfader

CS-19 .....**399.00** **CS-19R**.....**399.00**

KM-130
19˝ Rackmount Karaoke Mixer

- Voice & Music Echo
- Voice Cancel ◆ 3 Mic Inputs
- Fully Adjustable Digital Delay
- Mic EQ With Sweepable Midrange
- Voice Cancel for Karaoke CDs

KM-130 .....**99.95**

FX-7000
10˝ Professional FX DJ Mixer

With 24 preset DSP effects and over 1300 parameter settings, the FX-7000 isn't just another mixer, it's a creative tool. Perfect for competition or hip hop jocks.

- 3 stereo channels
- 3-phono, 5-line, 2-mic inputs with Talkover and EQ
- 24 Preset DSP Effects with over 1,300 Parameter settings
- Effect/Dry mix
- Rail-Glide 45mm removable crossfader
- Crossfader reverse (Hamster) switch
- 2 position cross fader curve control
- Balanced outputs, gold plated RCA I/O
- Cut for low, mid and high on each channel
- Cue split w/individual channel cue buttons
- Rotary master and zone output controls

FX-7000 .....**299.95**

MP-3000X
Dual Pro MP3 CD & Audio CD Player

The future of DJing, the MP-3000X offers all the regular features of CD players (Pitch, Loop, Cue & Pitch Bend) as well as Gemini's exclusive Automatic Disc Calibration System (ADCS) which will automatically set up & adjust the laser to the way the CD was recorded.



- All functions available to mix with MP3 discs: cue, pitch, loop, and bend
- Anti-Shock buffer memory
- Robo-Start
- High-visibility blue LCD display
- Loop with real-time loop editing
- Pitch Control with selectable +/- 4%, 8%, 16% range
- Pitch Bend buttons
- Decodes compression up to 320 Kbps, better than any MP3 player out there
- Search Mode selector for music search or track search
- Inner/Outer Jog Wheel for frame-accurate music
- ADCS Automatic Disc Calibration System search and for ultra-fast track search

DJ KITS



First Mix 3.0



Renegade



Mobile Pak 201

Disc O Mix 4.0
Professional Table-Top CD Turntable DJ Package
Includes 2 CDJ-15 CD players, a PMX-350 mixer, PM-100 powered monitor speakers, EM-50 headphones, GM-26 mic and a DJC-1040AR case.....**399.95**

First Mix 3.0
Level 1 Belt Drive Turntable DJ Package
Includes 2 XL-120 Belt-Drive Manual Turntables with CN-25 cartridges, a PMX-250 mixer and EM-50 headphones.....**199.95**

Scratchmaster 4.0
Level 3 Direct-Drive Turntable DJ Package
Includes 2 XL-300 turntables with CN-25 cartridges, a PMX-500 pro mixer, PM-100 powered monitor speakers, EM-50 headphones and a GM-26 microphone **399.95**

Mobile Pak 201
Professional Dual CD-Player Mobile DJ Package
Includes a CDX-602 CD player, PMX-1100 Preamp Mixer and an MRC-2 Case **499.95**

Renegade
Level 2 Belt-Drive Turntable DJ Package
Includes 2 XL-200 turntables with CN-25 cartridges, a PMX-350 mixer, PM-100 powered monitor speakers and EM-50 headphones.....**299.95**

DJ EQUIPMENT CASES

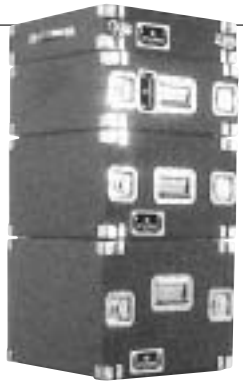
Tables

These adjustable tables are an excellent portable solution for setting up your mobile DJ equipment.



AT-5422 Adjustable Table
Fold out adjustable legs adjust from 26½″ up to 39½″.
The tabletop surface dimensions are 54 x 22″ and it weighs 32 lbs.**119.95**

AT-6022 Adjustable Table
The same adjustment dimensions as the AT-5422, but with tabletop dimensions of 60 x 22″ and a weight of 36 lbs **125.95**



ER-Equipment Racks

These racks feature a 16½″ body depth not including front and rear lids.

Model	Spacing	Dimensions (HWD)	Weight	Price
ER-2S	2-space	5¼ x 20¾ x 16″	16 lb.	**88.95**
ER-4S	4-space	8¾ x 20¾ x 20¼″	23 lb.	**103.95**
ER-6S	6-space	12¼ x 20¾ x 20¼″	26 lb.	**115.95**
ER-8S	8-space	15¾ x 20¾ x 20¼″	30 lb.	**126.95**

Numark CD Mix 1 or 2 Cases

NUM-CD1/S Carpet Style Case
Holds one Numark CD Mix 1 or 2 player. Rear spacing accomodates power supply and cables. Surface hardware. Dimensions: 7¾ x 20½ x 17⅜″ (HWD). Weighs 18 lbs.....**119.95**





NUM-CD1/R Carpet Style Case
Holds one Numark CD Mix 1 or 2 player. Rear spacing accomodates power supply and cables. Recessed hardware. Dimensions: 7¾ x 20½ x 17⅜″ (HWD). Weighs 18 lbs **139.95**

F-NUM-CD1 Flite Style Case
Holds one Numark CD Mix 1 or 2 player. Rear compartment to accommodate the power supply and cables. Cable port in rear for easy passage of wires. Dimensions: 9 x 20¾ x 17½″ (HWD). Weighs 19 lbs.....**149.95**

Mobile DJ Cases



DCV-84
Holds 84 CD jewel cases or 252 view packs. 3 rows wide. Surface hardware dimensions: 6¾ x 18½ x 13½″ (HWD). Weighs 8 lbs **69.95**

DCV-112
Holds 112 CD jewel cases or 336 view packs. 4 rows wide. Surface hardware dimensions: 6¾ x 24½ x 13½″ (HWD). Weighs 10 lbs **69.95**

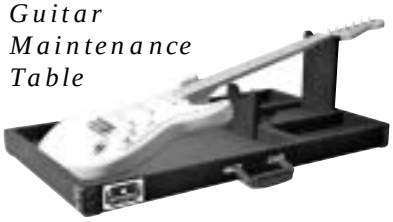
DCV-168
Holds 168 CD jewel cases or 504 view packs. 6-rows wide. Surface hardware dimensions: 6¾ x 36¼ x 13¾″ (HWD). Weighs 15 lbs **99.95**

DCV-310c
Holds 310 CD jewel cases or 930 view packs. 7 rows wide. Recessed hardware with casters on one end for easy transport. Dimensions: 7¼ x 48 x 18¼″ (HWD). Weighs 35 lbs **189.95**



GMT-001

Guitar Maintenance Table



Lower neck support for electric guitars, upper neck support for acoustic guitars. Compartments for holding tools and small miscellaneous parts. Dimensions: 2¼ x 32¼ x 18¼″. Weighs 3 lbs **64.95**

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DJ EQUIPMENT CASES

TL-Series CD Workstations

The TL-Series is a portable CD workstation case which accommodates the mobile DJ with a well-loaded CD setup. This series comes with a trap door on the back for full access to wires and hook-ups. Surface mounted hardware is standard.



TL-252
Bottom: 2-space rack. Center slant: 5-space rack. Top: 2-space rack. Dimensions: 16¼ x 20½ x 16¾″ (HWD). Weighs 19 lbs **164.95**

TL-253
Bottom: 3-space rack. Center slant: 5-space rack. Top: 2-space rack. Dimensions: 17¾ x 20½ x 16¾″ (HWD). Weighs 21 lbs **163.95**

TL-363
Bottom: 3-space rack. Center slant: 6-space rack. Top: 3-space rack. Dimensions: 20½ x 20½ x 19″ (HWD). Weighs 24 lbs **189.95**

TL-484
Accommodates a full-loaded CD workstation. Optional CD carrying lid makes for a portable and compact set up. Bottom: 4-space rack. Center slant: 8-space rack. Top: 4-space rack. Dimensions: 26½ x 20½ x 23″ (HWD). Weighs 31 lbs **208.95**

TL-484CD Same as above with CD carrying lid **239.95**

OU-1284CT
OU Series Rack

This case features a front lid which has fold out legs to convert it into a table that mounts on either the right or left side of the case - adding mork work area. 12 space bottom, 8 space slant, 4 space top. Dimensions: 43 x 203/4 x 231/4. Weighs 84 lbs. Accepts an additional optional table.....**413.95**



Club Racks



Grundorf Club Racks feature pressure-fit front and back covers. The CR-Series are simple and economic racks which are available in various rack mount spacings. These are equipped with all black hardware including plastic stackable corners and a port in each cover to aid in removing and/or running cables.

Model	Spacing	Dimensions (HWD)	Weight	Price
CR-2	2-space	5 x 21¾ x 18¼″	10 lb.	**59.95**
CR-4	4-space	8½ x 21¾ x 18¼″	13 lb.	**69.95**
CR-6	6-space	12 x 21¾ x 18¼″	17 lb.	**89.95**
CR-8	8-space	15½ x 21¾ x 18¼″	21 lb.	**109.95**

Top-Load Racks



Lightweight, yet durable, Top-Load Racks are constructed with multi-layer plywood, finished with metal hardware, metal rack rails and a carpeted exterior. Front, rear and top latchable covers. Top-load racks are available in different body depths. Short depth racks feature a 16½″ body depth; 8-space top slant rack; 4″ high front panel with surface hardware. Deep depth racks feature a 20¼″ body depth; 10-space top slant rack; 4″ high front panel with surface hardware.



Model	Spacing	Dimensions (HWD)	Weight	Price
TLR-8DR	8-space	23¾ x 22 x 24½″	53 lb.	**269.95**
TLR-12DR	12-space	30¼ x 22 x 24½″	62 lb.	**299.95**
TLR-16DR	16-space	37⅜ x 22 x 24½″	71 lb.	**329.95**
TLR-4SS	4-space	16 x 21¾ x 20⅛″	29 lb.	**189.95**
TLR-6SS	6-space	19½ x 21¾ x 20⅛″	33 lb.	**193.95**

NUMARK

TURNTABLES & CARTRIDGES

FUSION SERIES TURNTABLES

TTX1

- Direct drive ultra-high torque open motor (+3.7 kgf-cm peak)
- Straight and S-shaped tone arms included
- "Solid-Core" steel top, rubber base
- Interchangeable pitch fader and button cartridges for club and battle style
- Detachable power and audio connections
- Removable Target light with super-bright white LED
- Blue illuminated display with 10, 20 and 50% pitch range, 33, 45, 78 RPM, reverse, adjustable brake and startup times
- Digital out, line output, key lock

TT200

- Direct drive super high torque motor
- Straight and S-shaped tone arms
- Battle and lub style design
- Steel top, rubberized base, aluminum platter
- User-replaceable pitch fader, removable aluminum light
- Detachable power and audio connections
- 33, 45, 78 RPM, reverse



TT1650

- Direct drive high torque motor
- Battle and cub style design
- Removable target light
- Aluminum platter
- Detachable power and audio connections
- 33, 45 RPM



TT1600

- Efficient belt drive motor
- Battle and club style design
- Detachable power and audio connections
- 33, 45 RPM

CARTRIDGES

CC-1
General Use Cartridge

- Plug-in silver design
- 18 grams, 6v output

CC-1 RS Replacement stylus for CC-1

CS-1
Club Cartridge

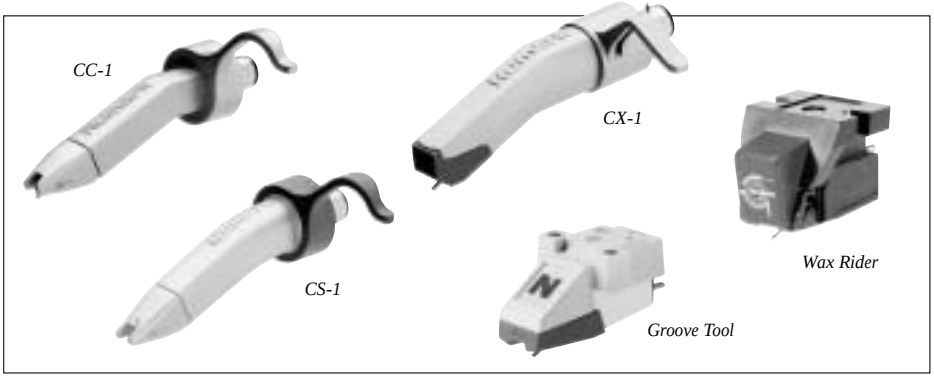
- Plug-in design with rubber grip handle
- 18 grams, 6v output

CS-1 RS Replacement stylus for CS-1

CX-1
Scratch DJ Catridge

- Plug-in rugged design; Silver with blue tip
- 18 grams, 6v output

CX-1 RS Replacement stylus for CX-1



Groove Tool
Headshell Mountable Cartridge

- .6 mil spherical diamond tip cartridge with 2.5g to 3.5g tracking force

Groove Tool GTRS 2 extra styli

Wax Rider
Headshell Mountable Cartridge

- .6 mil spherical diamond tip cartridge with 2.5g to 4.5g tracking force

Wax Rider WRRS 2 extra styli

SCRATCH MIXERS

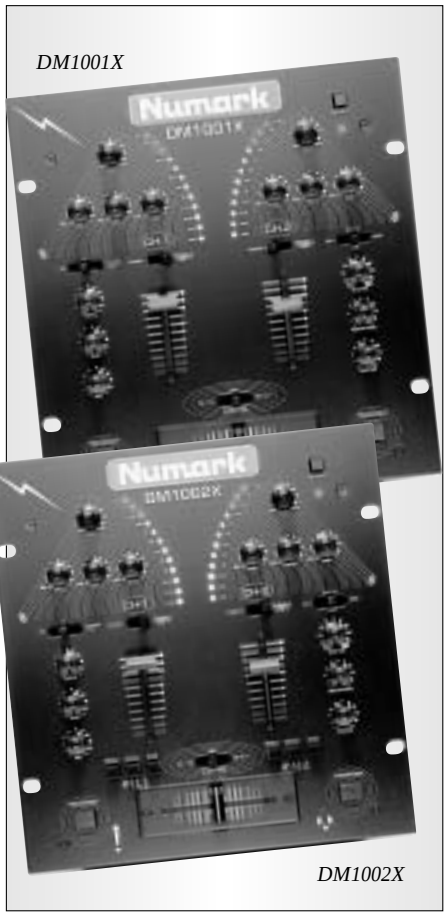
DM950
2 Channel 8″ Mixer
- 2 phono, 2 line and 1 Mic input
- Master and record outputs
- Channels contain 2-band rotary EQ, gain

DM1001X
2 Channel 10″ Mixer

Same as the DM950 PLUS–
- 4 phono, 2 line and 1 Mic input
- Master and record outputs
- 3-band EQ and PFL meters
- Transform buttons
- Replaceable crossfader with slope and "hamster" control

DM1002X
2 Channel 10″ Mixer

Same as the DM1001X PLUS–
- -35 dB KILLS switches for bass, mid and treble



DM1001X

DM1002X


DM3001X ◄

► *DM3002*

DM3000X
3 Channel 10″ Mixer
- 3 phono, 3 line and 1 Mic input
- Master with pan, record, PFL send for sampling, fader start
- Channels contain gain, 3-band rotary EQ
- Assignable crossfader with hamster and slope adjust

DM3001X
3 Channel 10″ Mixer

Same as the DM3000X PLUS–
- Master, Booth, Record and PFL cue send for sampling outputs
- Bass, Middle, and Treble KILL switches
- Unique punch IN/OUT transform buttons, panning

DM3002X
3 Channel 10″ Mixer

Same as the DM3001X PLUS–
- PFL meter, post process loops

MATRIX 2
2 Channel 10″ Mixer
- 2 phono, 2 line and 1 Mic input
- Master, Record and Zone outputs
- Fader start crossfader with extra wide knob
- Channels have Gain, 3-band EQ, and PFL meters
- 1/4″ and 1/8″ headphone jacks

MATRIX 3
2 Channel 10″ Mixer

Same as the MATRIX 2 PLUS–
- 2 phono, 6 line and 1 Mic input
- Master Balanced, Record and Zone outputs
- Process loop with blend control
- VCA crossfader with assign, slope, hamster, fader start, transform buttons, kill switches
- Channels contain long-life input faders, 3-band EQ, gain panning, 8 position toggles and PFL meters

PDM01
24-bit Digital 2 Channel Mixer
- 24-bit digital 2 channel mixer with 2 phono, 3 line and 1 Mic input
- Master balanced, booth, optical and coax digital outputs
- Digital, replaceable and interchangeable VCA faders with slope and reverse controls
- Digital crossfader with 8 functions
- Channels contain dual mode tone controls and filter controls
- Mini crossfader cueing with headphone tone control
- Master balanced, booth, optical and coax digital outputs


Matrix 3

NUMARK

CD & MP3 PLAYERS

TABLETOP CD PLAYERS

Axis CD players have a recessed rear panel, all rubber buttons and are CDR compatible. They offer relay play, 6-way fader/remote start/pause or start/recue.



AXIS 2

- Large interactive pitch wheel, long pitch slider
- 10 sec. anti-shock, true buffered instant start
- ±8/16% pitch, sleep mode, digital output
- BPM counter, separate relay play



AXIS 8

- Real time scratching on 130mm jog wheel
- Direct access of 12 DSP effects
- ± 6, 12, 25 and 100% pitch control on 100mm fader
- KEY lock at any tempo up to ±100%!
- Beatkeeper with auto-synchronization
- 2 complete seamless loops with 3 hot stutter starts
- Reverse and key change
- Adjustable startup and braking speeds
- MIDI in/out for effects, clock, interlocks, start/stop
- User programmable custom presets
- 48 seconds of buffer memory for anti-shock
- Transport bay illumination light
- Software is updateable via CDR

RACK MOUNT CD PLAYERS



CDN 15

- Single rack CDR compatible CD player
- Interactive pitch bend/search wheel with finger grip
- +/-8 or 16% pitch ranges
- Sleep mode to save laser life
- Direct track access with sequence programming
- Power switch protector, All rubber buttons
- Transport bay illumination light
- Digital output



CDN 25

Same as the CDN 15 PLUS–

- Dual rack mountable



CDN 33

Same as the CDN 25 PLUS–

- 20 second anti-shock
- True buffered instant start
- Light up play/cue buttons
- Additional rotary track access
- Relay play, 6-way fader/remote start/pause or start/recue



CDN 88

Same as the CDN 33 PLUS–

- Real time scratching
- Reverse, up to ±%100 pitch
- KEY lock (master tempo with key control) up to 100%!
- Built in Beatkeeper with auto-mixing and synchronization
- 2 seamless loops with 3 hot stutter starts
- Many DSP interactive wheel effects
- ± 6, 12, 25 and 100% pitch control
- User programmable custom feature set
- 48 seconds of buffer memory

CDN 90

Same as the CDN 88 PLUS–

- Adjustable startup and braking
- Saves 3,000 cue points in memory
- MIDI in/out
- User upgradeable
- Sleep mode

MPCD3 MP3
MP3 CD Player



- Single rack mount player plays MP3 files from CDR and audio CDs
- 999 track index
- Direct track access with sequence programming
- Interactive pitch bend/search wheel with finger grip
- +/-8 or 16% pitch ranges
- Digital output

DJM-300/DJM-300S

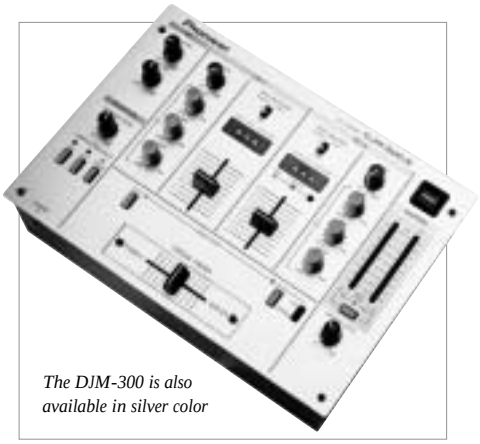
Professional DJ Mixer

The DJM-300 is a DJ mixer that will unleash your full creative potential. Its Auto BPM Counter, separate EQ and Fader Start Play function allow you to perform better and more easily than ever. And a host of other great functions add to the flexibility. Together with a CDJ-700 Series CD player, The DJM-300 is the ideal house or mobile music system that the professionals have been craving.



FEATURES

- Auto BPM Counter accurately measures the beats per minute of almost any kind of dance music automatically. This function includes a digital indication that allows you to confirm the tempo of the current selection at a glance and takes the guess-work out of comparing the beats of different selections. Additionally, the counter works equally well when monitoring the CD player and other digital sources or analog sources such as a turntable.



The DJM-300 is also available in silver color

- The DJM-300 is equipped with separate 3-band (High/Mid/Low) EQ that allows the DJ to adjust the output of each channel independently at the turn of a dial. By enabling attenuation of up to -26dB, the system makes it possible to effectively silence unwanted channels or adjust each channel's output level exactly as desired. A separate +12dB boost capability for each channel is included allowing you to add emphasis to particular sections.
- Separate Trim controls allows you to individually raise or lower the input level for each channel - enabling almost effortless mixing of selections recorded at different signal levels
- Connecting the DJM-300 to the CDJ Series allows you to take advantage of Fader Start Play. By increasing the Fader, on the mixer, the player will automatically override the Pause control and start the music immediately. Fader Start Play also works during Cross Fading to bring out the best of sampler-style music.

The DJM-300 combines high-sound quality with industrial-use durability. Pioneer has employed industrial-use components for the Faders, Volume controls and other rotary controls, and the Cross Fader is designed to be easily replaced from the top panel. Regardless of the sound source, the DJM-300 reproduces all original input sounds faithfully and without unwanted distortion.

- When used with the CDJ Series, the Fader or Cross Fader controls can also return you automatically to a designated cue point if desired. By fading Cross Fader back and forth or Fader up and down quickly, you can give the "essence" of scratching a certain point on a vinyl record.
- The special equalizer controlling the microphone input means you can set the EQ to HIGH to give your voice greater clarity and presence, or to LOW to enable a variety of microphone sound presentations. Moreover, the EQ can also be used to compensate for feedback or burst sounds.
- Using the rear-mounted switches, the rear PHONO inputs can be used as line-inputs to allow the connection of a number of sound-source combinations. In short, the DJM-300 allows even DJs who don't use an analog turntable to make flexible use of four separate sound source inputs.
- During normal operation, the DJM-300's level indicator displays the Master output level. When the Master/Channel Level button is pressed, the input level of each channel can be displayed in turn for quick confirmation of the current channel level input status.

HEADPHONE MONITORING SWITCHING

The headphone monitor switching function makes sound source monitoring more convenient and flexible than ever by allowing you to switch easily between all-channel stereo, all-channel mixed stereo and all-channel left/right distributed monitoring.

CH1	CH2	MASTER	MONITORING SOUND
On	Off	Off	CH1 Stereo
Off	On	Off	CH2 Stereo
Off	Off	On	Master Stereo
On	On	Off	CH1+CH2 Mix, Stereo
On	Off	On	L=CH1 Mono / R=Master Mono
Off	On	On	L=CH2 Mono / R=Master Mono

PIONEER

DJM-600

DJ EQUIPMENT

1540

Professional DJ Mixer

The DJM-600 offers many innovative DJ-friendly features which will make it the standard in clubs. It is for artists, for those serious about music, for those serious about their mixes! Complete with an impressive arsenal of powerful effects, sampling functions, automatic beat synchronization, a choice of cross fader curve patterns... and superb sound quality — the DJM-600 is set to create the ultimate high!

To ensure the widest possible music-making flexibility, The DJM-600 is equipped with an industry-first auto beat sampler, which easily and accurately samples phrases in a BPM-linked operation along with an improved auto beat effector. In combination with Pioneer's CDJ series, this advanced mixer gives DJs everything they need to make party music really come alive.



FEATURES

Auto Beat Sampler

This DJ function allows you to sample your favorite sources and play them back easily at the touch of a button. The BPM is measured digitally and recorded automatically exclusively for designated beats (1/2/4/8/16 beats, maximum interval: 8 seconds). If set to the Stand-By mode when no sound is present, the function automatically starts and stops recording in response to the presence or absence of the input signal and achieves the correct sampling with perfect timing. What's more, the user can edit the end position, making possible a wide diversity of playback effects by making use of three playback mode options and the cross fader start.



- *Single Play* - plays back the recorded sound once only when the button is being pressed
- *Loop Play* - plays back the recorded sound repeatedly when the button is pressed
- *Stretch Loop Play* - extends or shortens the recorded sound in accordance with the mixed track's BPM and plays back the sound repeatedly

Auto Beat Effector

Linked to the BPM, the DJM-600 can add effects to each CH, MIC and MASTER independently. The user can produce music that makes full use of high-level DJ techniques by performing quick setting with the Beat Unit Switch button, adjusting the precise time parameters, or controlling the balance between the effect sounds and the original sounds by varying the mix ratio. Moreover, even when using the fader, it is possible to leave just the effect sounds with or without linking to BPM.

- *Delay* – produces repeat beat effect
- *Echo* – produces repeated sounds, which fade to create an echo effect
- *Auto Pan* – automatically throws sounds between left and right speakers
- *Auto Trans* – periodically cuts the sound
- *Auto Filter* – creates effects through manipulating sound frequencies
- *Flanger* – produces an electronic wave effect
- *Reverb* – produces a range of reverberation effects
- *Pitch Shifter* – manipulates music pitch within ±1 octave
- *Send/Return* (External effects input/output) – connects available peripherals (effectors, samplers, etc)

Auto BPM Counter

Equipped with a counter that automatically measures and digitally displays a track's BPM (measurement range: 70-139/91-180/71-180/Manual), the DJM-600 provides a visual confirmation of a tracks tempo to allow the user to merge one song into another smoothly. Also, BPMs of tracks that are difficult to measure can be manually input by tapping on the TAP button.



Fader Start Play

- When connected to the CDJ Series of CD players, the DJM-600 can begin playback of a track using the cross fader or the channel fader
- Mix tracks without creating spaces by setting the stand-by track's cue point and performing a single-action cut-in
- Since it returns to to the cue point via the back fader alone, sampler-style playback is also possible
- Start/back play of sources recorded using the built-in sampler is also possible using the cross-fader

DJM-600

Channel Faders

- Easily replaceable faders (5 Faders)
- Extremely durable and reliable touch faders

Cross Fader

- 3 selectable crossfade curve patterns
- Fader Curve echo effect enables smoother mixing

Headphone Output with Separate Level Control

- Assign any channel to the headphone out
- Mono Split Mode lets you create separate monitor and cue mixes by assigning each input to the left or right side of the headphone
- The balance between the monitor and cue channels is adjustable

Equalizers

- Equipped with a separate Hi/Mid/Low 3-band graphic equalizer for each channel
- Allows the stress of the instrument sounds to be adjusted within the range -26dB to + 12dB



DJM-600 faders

More Features

- Talk Over button mutes all levels apart from the microphone
- Rec Out terminal
- Four line connections
- Two Mic inputs
- Peak Level meter allows the user to instantly determine the source input condition for each channel to make possible certain operation without loss time
- Independent monitor selection switch will monitor any channel input or effect independently or simultaneously
- Booth Monitor level adjusts the volume for a separate DJ booth monitoring system
- Balanced output provides a direct connection to amplifier
- Top panel phones output
- Separate Trim controls for each channel

DJM-500 Professional DJ Mixer

This is the mixer top DJ's have been waiting for. Explore a new world of sound by linking effects to the beat with the Auto BPM function. The DJM-500 and CDJ Series make the perfect system for creating the ultimate in club or house party music. The DJM-500 features a BPM level display, a Peak Level meter that is equipped with 15-bit LED indicators for all channels and a Fader Start function. The 3-band equalizer corresponds to the HI, MID, and LOW channels and the attenuation level serves as a kill function which can decrease the level to –20 dB. The DJM-500 delivers a variety of effects such as delay, echo, auto pan, flanger, reverb, and pitch shifter that can be imposed on all channels, the microphone, and master.



MIXER FEATURES

	DJM-600	DJM-500
Mic Inputs (XLR / ¼″ inputs)	2 (1/1)	2 (1/1)
Total Channels	4	4
CD/Line Inputs	5	4
Turntable Inputs	3	3
Fader Start & Back Cue Play	yes	yes
3-Position Selectable		
Fader Curve	yes	no
Auto BPM Counter	yes	yes
Beat Sync Effects	yes	yes
Delay	yes	yes
Echo	yes	yes
Auto Pan	yes	yes
Auto Transformer	yes	no
Auto Filter	yes	no
Flanger	yes	yes
Pitch Sifter	yes	yes
Reverb	yes	yes
Send/Return	yes	yes

	DJM-600	DJM-500
Direct Beat Select	yes	no
Effects Level & Depth Volume	yes	no
Beat Sampler	yes	no
Single Play	yes	no
Loop Play	yes	no
Stretch Play	yes	no
Fader Sample Start Play	yes	no
EQ Minimum Level	yes	no
Headphone Monitoring		
(Cue/Master)	yes	no
Headphone EQ	no	yes
Talk Over	yes	no
Outputs:		
RCA Master Outputs	1	1
RCA Booth Master Outputs	1	1
XLR Master Outputs	1	1
1/4″ Master Outputs	0	1
RCA Record Outputs	1	0

PIONEER

DJM-3000

DJ EQUIPMENT

1542

Professional DJ Mixer



The DJM-3000 is Pioneer's first Pro DJ rack mount mixer in a 19˝ format. This re-sizing and rack-mounting facility, coupled with numerous feature improvements, gives users even more scope for creativity in their sets.
The ability to mix tracks spontaneously and seamlessly is central to a DJ's success and accordingly, this unit answers the need for the highest possible levels of flexibility.
Together with effects such as Delay, Echo, Auto Pan, Flanger, Pitch Shifter Reverb, Auto-Transform and Auto-Frequency Filter, the DJM-3000 incorporates an industry-first: Effect Mix Mode. It utilizes 3 effects (Echo, Zip and Roll) to transition from one song to the next in a BPM linked operation. The Roll Mix Effect effectively creates a drum build before segueing into the next song.

FEATURES

Flexible Functions

- A BPM counter automatically measures and digitally displays a track's tempo from 70-180 BPM, allowing for instant synchronization of BPM and effect
- For tracks that are difficult to measure, the Tap button allows DJs to manually input the BPM and confirm via the Beat Indicator that the beat is linked to the Auto BPM.
- Linked to BPM, the DJM-3000 can add effects to each input (Channel, Mic and Master) independently. The DJ can perform quick settings with the Beat Unit Switch button and adjust the precise time parameters, or control the balance between the effect sounds and the original sounds by varying the mix ratio. Even when using the fader, it's possible to leave just the effect sounds without linking to BPM.
- Equipped with a separate 3-band equalizer (HI/MID/LOW) for each channel. This arrangement allows the stress of the instrument sounds to be adjusted within the range -26dB to +12dB.

ROTARY FADERS



Now you can change the faders on the DJM-3000 to rotary faders. This option should be installed by a Pioneer-approved service center.

Key Features

- Digital Output. Most mixers only have analog output, so this feature increases the options by allowing DJs to connect to a digital PA system, MD or DAT player, or any other device with digital input.



- Effect Mix Mode. Easily and accurately effects phrases in a BPM-linked operation.

Effects Section

— *Delay* - Produces repeat beat effect
— *Echo* - Repeated sounds that fade for echo effect
— *Auto-Pan* - Automatically switches sounds between left and right speakers
— *Auto-Transformer* - Cuts sounds at requested beat
— *Auto Filter* - creates effects by manipulating sound frequencies
— *Flanger* - Produces an electronic wave effect. Reverb: a range of reverberation effects
— *Reverb* - A range of reverberation effects.
— Pitch Shifter - Manipulates music pitch within +/- 1 octave
— *Send/Return* - Connects available peripherals (external effects input/output)

Beat Effects

— Effects dial for complete control over effect intensity
— Effects range enables sound effects on 1/4, 1/2, 1/1, 2/1, 4/1, 8/1 beats

Auto BPM

- Beat measurement from 70-180 BPM
- Automatic synchronization between BPM and effect
- Beat Indicator shows when the required beat is linked to Auto BPM

Additional Features

- Adjustable Talk Over level control - mutes all levels apart from microphone
- Mic Switch (Mic Off/Mic On/Talk Over)
- 7 Line Inputs
- 4 Turntable Inputs
- Switchable Phono/Line Inputs
- 4 Fader Start Inputs
- 3 Mic Inputs (1 x Neutric, 2 x 1/4˝)
- Headphone Monitor Split/Headphone auditioning
- Recording outputs
- Reinforced headphone jacks
- Frequency Response of 20Hz-20kHz (CD)
- Signal to Noise Ratio 87db (CD)
- Total Harmonic Distortion Rate 0.02% or less (CD)

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CDJ-1000MK2

Digital Vinyl Turntable



When Pioneer released the CDJ-1000, it quickly became an industry standard in clubs, bars and studios around the world. Its overall performance, feel, sound and memory features gave professional DJ's the ability to push their skills to the limit and harness the freedom and flexibility of digital technology—vinyl feeling and sound without vinyl. The CDJ-1000MK2 adds delay-free scratching, "wide" pitch control and a full-featured digital output to the world's largest touch sensitive jog dial, allowing you to treat a CD the same way you'd treat vinyl.

FEATURES

Performance Control

- Hot Cue (sampler) memorizes up to 3 cue point's per CD and can also be used like a mini sampler for vocals, stabs, or breaks
- Instant reverse with no loss of tempo or pitch - can also be used with the Hot Cue and the Loop Sampler
- Real time seamless loop provides a simple way to set a loop while a track is playing
- Loop In/Out Adjust adjusts the In and Out points of a loop stored in memory. Reloop instantly takes you back to the beginning of the loop stored in memory.

Memory

- Store Wave data as well as cue and loop points in internal memor, or on a supplied 16MB removable memory card (MMC). Scroll through saved cue and loop points.

CD Control

- Cue your CD's with a large 7″ diameter touch-sensitive jog dial designed to emulate a vinyl turntable
- Pitch Bend can either speed up or slow down the tempo of the music. Jog Mode functions with or without touch sensitivity.
- Scratch Play - in Vinyl mode, pressing down on the jog dial stops the music and allows you to scratch or cue your track by rotating the jog dial in either direction.
- Vinyl Speed Adjust Touch/Brake allows adjustment in the speed at which the CD slows down until it stops.
- Vinyl Speed Adjust Release/Start adjusts the speed at which the CD returns to normal playback speed.

Tempo

- Master Tempo locks the pitch of the track even when you change the speed
- Tempo Control with 100mm precision linear slider that is coupled with an 0.02% incremental display panel (at ±6%)
- DJ's can use the ±100% pitch control to create new sound effects from opposite ends of the sound spectrum and also to BPM match completely different tracks. This is in addition to the ±6 ±10 and ±16% ranges.
- Reads CD, CD-R and CD-RW discs

Cue Functions

- Auto Cue finds the beginning of a track even if it differs from the track's frame
- Manual Cue points can be set anywhere on the CD using either the jog dial or manual search buttons. The cue point is automatically stored in the internal memory or removable memory card (MMC).
- Real Time Cue allows a cue point to be set on the fly and stored into internal memory
- A Cue Marker is set automatically within the center display to visually represent the cue point relative to the jog dial's location.

Additional Features

- Start a track instantly by using the cross fader while connected to any DJM Series mixer. By sliding the fader back, it will return to the previously selected cue point
- Connecting two CDJ-1000's (via 3.5mm mini-plug cable) allows one to be put on standby while the other plays. Relay Play automatically starts the second player enabling them to play 'back to back' to give you endless music play.
- Legato link reproduces sound frequencies above 20kHz, which are normally lost during CD-format playback
- Sound jumps or skipping is prevented by using a 16-second shock-proof memory for normal CD play (minimum of 10 seconds if scratching and reading wave data at the same time in Vinyl mode).
- Oil-dampened floating suspension prevents sound skipping, even when subjected to impact during use or from vibrations from high volume music and low frequency bass.
- Full-featured S/PDIF digital output ensures that all digital features are transmitted, even scratching and master tempo.

CDJ-800 Digital Vinyl Turntable

A cheaper version of the CDJ-1000MK2, the compact, silver-colored CDJ-800 features Pioneer's world-renown jog-wheel, "wide" ±100% pitch control, S/PDIF digital output and storage for 500 cue point on board.

- Touch sensitive jog wheel enables a track to return instantly to a cue point
- Touch release dial allows CDs to be slowed, stopped and relaunched at different speeds
- Auto Beat Loop features four timing buttons to instantly create perfect 1, 2, 4, 8 beat loops. Also allows instant 1/1, 1/2, 1/4, 1/8 beat cuts of the initial loop.

CDJ-100S



Performance CD Player

A CD turntable with endless sound creation and remix possibilities, the CDJ-100S allows creative freedom to develop your own individual style. Turn the Jog Dial like an analog turntable, and club sound resounds. You can operate it with a close-to-vinyl turntable feel. Enjoy a wide variety of DJ performances combining analog touch operability with versatile digital effects

FEATURES

- Built-in Digital Jog Break provides analog DJ effects at the turn of the Jog Dial. Each of three buttons provide a different remix assist function (Jet, Zip, Wah)
- Slot-In disc-loading slot mechanism makes it fast and easy to play, change and remove discs from the player
- Large-diameter Jog Dial allows you to manipulate your music like an analog turntable by rotating the dial to find the precise entry point for tracks
- High-precision Tempo Control allows exact control of track speed. The digital display with a 0.1% unit readout provides simplified control with higher accuracy
- 2 Tempo Control ranges – either ±10% (in 0.1% steps), or +10 to -16% (in 0.2% steps)
- Quick Start feature produces instantaneous (0.01 second or less) music start that virtually eliminates time lag
- Master Tempo changes the music tempo without changing the pitch
- An analog bar graph gives elapsed and remaining playing time, and provides a visual readout of the track playback progress - comparable to the moving needle position on an analog record
- Full cueing functions such as Cue POint, Back Cue, Auto Cue and Cue Point Sampler which allows playing up to ten minutes of sound from a preselected cue point
- Relay Play - if you connect 2 CDJ-100S decks, when playback of a track comes to an end on one deck, standby mode on the second deck is released instantly beginning playback on that deck
- Oil-damped floating suspension protects against vibrations and impacts
- Fader Start/Fader Back Cue Play - when the CDJ-100S is used with the DJM-600, DJM-500 or DJM-300 mixer, you can start the music with a fader or cross fader, or use them to return to the cue point



CDJ-700S Professional CD Player

With the help of top DJs, the CDJ-700S has been specifically designed for today's professional club performers. It answers all their needs and concerns, so that they can focus 100% on what really matters: their performance and interaction with the crowd. Now there's absolutely no excuse for failing to exploit all the advantages and creative possibilities of the CD format. The CDJ-700S offers revolutionary and innovative features which make it the sandard in clubs.

- Jog Dial for a close-to-vinyl feel
- Playing Address provides an at-a-glance indication of elapsed playing time
- Tempo Control
- Master Tempo locks in the musics pitch even when the tempo changes
- Quick Start function
- Seemless loop, Real-time loop and Loop-Out adjust functions

CDJ-500II Ltd Professional CD Player

The layout, controls and feel are the same as the conventional turntable design, but the digital CD functions put it light years ahead. Incorporating the best of both the digital and analog worlds puts the CDJ-500II Ltd in a clas of its own.



- Digital features include Master Tempo, Loop functions and cue functions
- Quick Start, Playing Address
- Sensitive Jog Dial
- Fader Start/fader Back Cue Play
- Relay Play

CMX-3000



Twin CD Player

The CMX-3000 extends Pioneer's legacy of award-winning Pro DJ products, by bringing the twin CD player platform not only within the scope of the mobile DJ, but also to small clubs, music bars and the serious bedroom DJ. Stylishly designed with rubberized buttons and a black carbon finish, the CMX-3000 boasts an unbelievable feature set with Wave Display, Onboard Memory and Hot Cues. Additionally, features such as Pitch Bend buttons and Scratch Mode make the CMX-3000 the next generation twin CD player.

FEATURES

- The CMX-3000 reads the track volume level and reproduces the data as a Wave display, which is patented by Pioneer. A visual mapping of each track is represented in the display panel allowing the DJ to quickly find breaks in the tracks and use it as mix point or end point in the same way that you can on an analog record - by checking the position of the needle in the groove.



- A 1000 CD internal memory permits the DJ to store three cue and loop points per CD, as well as Wave data for up to 20 tracks per CD. Stored data can also be copied to other CMX-3000 CD players.
- High-quality Scratch mode allows scratching, scratch cueing and backspinning
- The two-piece jog dial allows greater visibility of the display. The dot matrix display delivers clear, bright status information. The buttons are made from hard wearing rubber and are illuminated for ease of use.
- Emergency Loop feature sets a four-beat seamless loop at the touch of a button. Indispensable if a track is about to end and the DJ has failed to cue up the next track.
- Cue & Loops can be saved to any of the three banks available for each CD and can be recalled at the touch of a button
- BPM Sync button synchronizes the BPM between CD 1 and CD 2, making beat matching easy and ideal when overdubbing on a track
- Pitch bend function changes the music tempo depending on the direction of Jog dial rotation and speed of movement. Pitch bend can be achieved by using either the Pitch Bend buttons or the Jog Dial
- Slot loading eases CD changing and will also reduce the risk of accidental damage to the player
- CD track title information can be read on the display
- Pioneer's legendary Master Tempo feature offers ±6%, ±10%, ±16% and WIDE (±100%) ranges
- Current track BPM Display
- High quality 60mm tempo sliders offer high specification and better quality (adjustment in ± 6% is 0.05%)
- Real-time Seamless Loop simplifies setting and releasing loops. This lets you set a loop while you are playing a track. A loop can also be set at the end of the track so that the track does not stop. In addition, an ADJUST mode has been added to enable one-touch adjustment to the Loop-out point and facilitate loop operation.
- Autocue changes the sound level to suit the style of music
- 15 second shock proof memory
- Plays commercial CDs, CD-R and CD-RW formats
- Legato Link technology allows the DJ to reproduce sound frequencies above 20 kHz, which are lost during normal CD-format playback
- Quick Start (less than 0.01 sec)
- Super fast track search /super fast song search
- Auto BPM counter/Tap BPM counter
- Time Mode (Remain/Elapse)
- Manual Cue / Auto Cue / Real-time Cue / Back Cue / Cue Point Sampler
- Fader Start
- Relay Play
- Digital Outputs

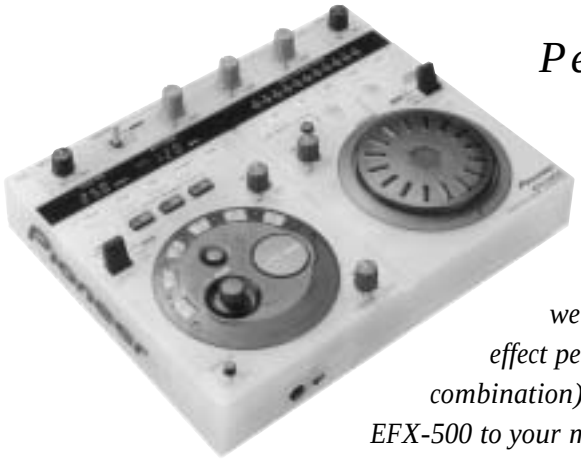


PIONEER

EFX-500

DJ EQUIPMENT

1546



Performance DJ Effector

Designed from a DJ-perspective and with the feel of an instrument, the EFX-500 effects box is guaranteed to make your mixes more dynamic. Offering a user-friendly top panel operation, and a comprehensive range of beat effects, a powerful three-band isolator and even Midi compatibility, it will allow you to scare your crowd with a succession of weird and wonderful sounds. Further evolved Beat Effects deliver real-time effect performance with a Jog Dial, allowing 25 different effects (5 x 5 combination). Expand and explore the world of digital sound by connecting the EFX-500 to your mixer, player, and electronic instruments!

FEATURES

- Separate 3-band graphic EQ (Hi, Mid, Low) cuts sounds up to -oodB and almost completely silences the sound of instruments in each band
- Isolator lever switch with momentary ON & LOCK ON method. (Removing your finger when in Momentary position returns the switch to OFF. When in LOCK ON position, it continues to be ON.)

Beat Effects



— The Auto Beat Effect button produces tempo-synced effects (1⁄4, 1⁄2, 3⁄4, 1⁄1, 2⁄1, 4/1) – Delay, Echo, Auto Pan, Auto Flanger and Auto Transformer

— Depth and Mix knobs change the depth of effects and the balance of the original sounds and effect sounds, respectively

— Set the beat by various methods such as Auto, Tap, msec, BPM and Midi Beat

- Effect lever switch with Momentary ON & LOCK ON
- BPM counter measures the beats per minute and turns it into a digital indication
- You can set the BPM instantly by tapping the button with your finger.
- Effect Frequency allows you to select Hi, Mid and Low frequency to produce effects. You can combine the bands as you like - for example, producing effects only in the Mid (picture), or, conversely, selecting the Low and Hi for effects with the original sounds in the Mid.
- Digital Jog Break can produce new digital sounds on the spot. A turn of the Jog Dial controls 5 effects: Jet, Zip, Wah, Ring and Fuzz. Set the strength and depth of effects as well as the balance of the original sounds and effect sounds. A Jog Break Meter optically indicates the motion of the JOG DIAL and the change of sounds, allowing you to visually confirm the motion.
- The **Jet** function adds a short delay. Turning the jog dial creates wave sounds, with the flanger effect made by using 2 analogue turntables.
- **Zip**: Change the pitch of reproduced sounds without changing the tempo. Lowering of the pitch produces sounds resembling those made by slowing an analogue turntable.
- The **Wah** function changes cut-off frequency. You can make the sound of a certain instrument strong or weak, making it feel like the sound is moving.
- **Ring**: Modulation of sounds produces a tone like the ring of a bell, providing effects like cosmic sound or a robot voice.
- **Fuzz** is a distortion effect on sounds.

Hold Function

- With the function on, the present state can be maintained if you stop turning the JOG DIAL. With the function off, it returns gradually to the state of the original sounds.

Jog Dial Memory Function

- This is a new function to memorize and reproduce the motion of the Jog Dial. When you turn the Jog Dial and simultaneously press the Memory button, the motion of the Jog Dial is memorized for as long as 8 seconds. While pressing the PLAY button, you can replay the memorized action in all the 5 effects as many times as you want.

Other Functions

- The EFX-500 is equipped with Midi I/O terminals for controlling the Clock signal of Midi instruments. The BPM of input analogue sound signal is measured automatically, converted to Midi-Clock signal, and output so that the beats of connected MIDI instruments can be synchronized.
- The Effect Monitor System monitors each of Beat Effect, Digital Jog Break, and master sounds. The volume & meter controls input/effect output levels. The auto-lighting, button indicates Rear Panel the setting of each effect. The compact size makes this machine portable

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DMP-555

Digital Media Player with MP3 and SD Technology



The DMP-555 is compatible with CD audio and MP3 formats and will playback from both CD and SD cards. Since these are capable of storing a huge number of MP3 tracks and loops, the DMP-555 allows DJs to enhance their sets and mixes with the latest music from around the world. When the slim, compact and portable SD Card is used for mixes, the DMP-555 effectively becomes a sampler. This ground-breaking innovation is possible because MP3 loops can be BPM sync'd to the CD output and then fired on cue. In addition, the Master Tempo control can be used to slow down or speed up SD audio, without the pitch being affected.

CD ripping, loop editing and music file management are simple operations with the bundled DJ Booth software. And because the application simulates a set-up using two Pioneer CD players and a mixer, it's easy to master the playback, mixing and editing functions in no time at all. It's even possible to control the software from the DMP-555 itself, making DJ Booth the most innovative and user-friendly application available to today's digital DJs.

FEATURES

- Supports the SD Audio format of the powerful, removable SD card
- Onboard MP3 capability
- Compatible with CD-DA, CD-R, CD-RW and SD Card MP3 compression.

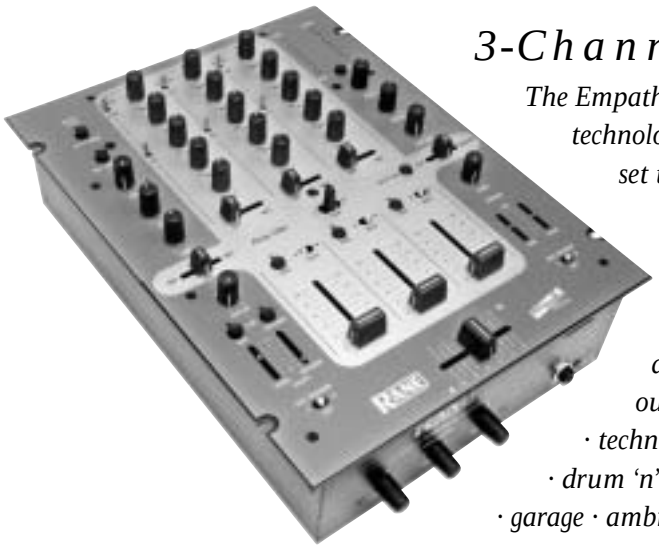


- Playback from both CD and SD card sources. Mix MP3 loops and samples from the SD Card with CD-DA audio
- Separate and dedicated CD and SD card control panels for each source during Dual Synchronous Play
- 3-Band Isolator allows you to cut out any one of the three frequencies (low: 280Hz and below, mid: 280Hz-6Khz, high: 6Khz and above) from either the CD or SD card
- Adjustment of SD card BPM for dual play
- Adjust the pitch of a loop or sample on the SD card during Dual Synchronous Play
- Mixes SD card audio output with your CD audio playout during Synchronous Play
- Match and lock SD card BPM to CD play-out BPM, at the touch of a button during Synchronous Play
- Navigate easily through SD Card contents. One Track Loop Set, Continuous loop of audio track on SD Card or CD.
- Scratch effect
- Master tempo
- Tempo select +/- 6%, +/- 10%, +/- 16%
- Loop/Reloop; Loop Out adjust
- CD text + large screen display
- CD and SD card audio level & BPM meters
- CD eject lock
- Eject and Power button guards
- Front loading
- Digital output
- Anti-shock memory
- Legato Link conversion
- Relay play
- Audio output; Card monitor output
- Fader start
- USB 1.1 interface
- Auto cue and manual cue
- Time mode
- CD / SD source select
- 100mm diameter jog wheel

DJ Booth Software

- CD Ripping - encodes to MP3 format from CD audio and saves to PC hard disk. Input all track information and encode with a constant or variable bit rate.
- Music Management - group by genre, BPM, edit track information and import music files
- DJ Play - simulation of 2 x CDJ players and a mixer
- Edit and create new loop samples, saved to PC hard disk
- Copy audio files to SD card via USB
- Internet - access to music website and online support at the click of a button
- Control both virtual CDJ players, in DJ Play and Create modes, from the DMP-555
- English, French, German, Italian, Spanish, Dutch, Traditional Chinese and Japanese
- Compatible with Windows 98SE / Me / 2000 / XP Home and Professional operating systems

EMPATH



3-Channel 10˝ Format DJ Mixer

The Empath mixer combines the vision of Grandmaster Flash and Rane technology. Compact size, exceptional performance and advanced features set the Empath mixer apart. The Empath is the most able 3-channel, 10˝ format mixer available. As the name implies: understanding, awareness and sensitivity to the needs of DJs. Empath features Accelerated-Slope, full-cut, 3-band tone controls. Dry/Wet pan on Mic, Channels 1, 2 and 3. Four Penny and Giles faders, automatic Flash-Cue, cue and master metering and a host of outputs. Meets the requirements of all DJ genres: trance · progressive · techno · electro · jungle· breaks · funk · hip hop · scratch · house · disco · drum 'n' bass · acid · live music · goa · dance hall · freestyle · reggae · dub · garage · ambient · gabber · rap

FEATURES

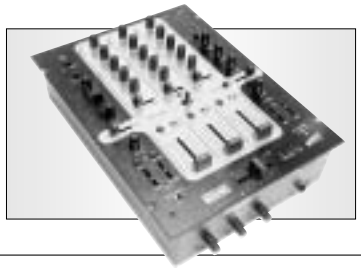
Features Important to DJs

- When traveling, finding power supply adapters is difficult. A universal, 100 to 240v AC, 50/60 Hz internal power supply allows using the mixer anywhere in the world without adapters.
- Auto-Gain prevents performance-wrecking swings in program level when changing sources. The music is never too soft or clipped.
- The FlexFX external effects loop with individual Dry/Wet pan controls for Mic, Input Channels 1, 2 and 3, provides mixing style versatility.
- Smooth, accurate and reliable mixing is made possible by employing the most respected name in audio faders — Penny & Giles faders are simply the best money can buy.
- Two kinds of fader caps are provided for different mixing preferences: soft rubber caps for smooth mixing, and hard plastic caps for fast mixing.
- Cue selection for a three-channel mixer can be slow and complicated. Flash-Cue allows the DJ to rapidly select Crossfader A-Cue-sum or Crossfader B-Cue-sum with the flick of a switch.
- Two-band headphone tone controls allow the DJ to match their mixer to their headphones.
- Nothing is more frustrating than arriving at a club and finding their jacks don't match. Systems may have XLR, TRS or RCA connectors. The Empath provides all three.
- More headphones are compatible with the Empath. It provides both 1/8" and 1/4" headphone jacks, simultaneously usable.
- Two assignable CD triggers work with any CD player with fader start ability.
- The Accelerated-Slope, full-cut, 3-band tone controls are the most flexible and highest quality available.
- Crossfader assign switches for each Input Channel allow any program to be mixed on the A-side, B-side or Post crossfader.
- Level meters view Stereo Cue, Stereo Master Mix, Split Cue or Main Output signals..



GMF Signature Edition

For the mobile DJ who requires the best, Rane offers the autographed Grandmaster Flash edition of the Empath mixer. Functionally and electrically identical to the original blue and silver unit, but with the following features:



- Brushed gold and black Lexan faceplate
- Deluxe carrying case with room for cords, headphones and phono cartridges

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TTM 54i

Performance Mixer

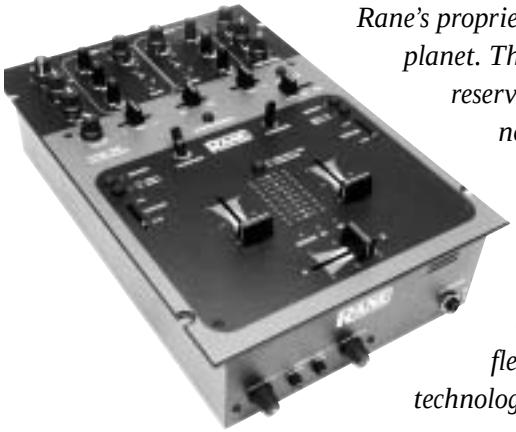


The TTM 54i offers more features, higher performance and more reliability than any performance mixer, from any manufacturer, at any price. The TTM 54i moves the art form to a new level. A precision Active Crossfader with continuously adjustable Contour and "Hamster" reversal controls, provides the lowest feed-through, lowest noise, and most responsive control in the industry, all with unprecedented reliability. The front panel "Hamster" reversal and switchable Contour controls give Active Program Faders ultimate flexibility. A high performance Accelerated-Slope, band-split 24 dB per octave tone controls provide more than 40 dB of Bass and Treble cut without affecting the vocal midrange. Lexan face plate insures sharp appearance for years to come.

- Assignable Effects Loop allows a stereo effect to be inserted in Pre-Fader Program 1, 2, or Master at the press of a switch.
- Independent 18 dB/octave rumble filters for each phono input.
- Balanced mic input with Bass and Treble tone controls and independent Effects Loop.
- Two phono and two line inputs with high quality, long-life clickless transform selector switches.
- Balanced 1/4″ TRS and unbalanced RCA master outputs.
- Dual mono 10-segment level meters with peak hold, indicate level for Programs 1/2.
- Headphone monitor for Master program allows performance rehearsal with Master level turned down.
- Stereo Headphone Cue pans between Program 1 and 2.

TTM 56i

Performance Mixer with Non-Contact Crossfader and Channel Faders



Rane's proprietary magnetic fader is the fastest, most accurate, and longest lasting on the planet. The design uses non-contact magnetic position sensor technology previously reserved for the most demanding aerospace and industrial applications. No travel noise - No bleed - Ever! Unlike optical non-contact faders, the magnetic fader is impervious to smoke, moisture, temperature and aging. The electrical performance is totally unaffected by use. The mechanical life of the magnetic fader exceeds 10 million operations. These magnetic faders know exactly what your hand is doing! A proprietary curve-translation module converts your hand motion into precise audio level control. You get more flexible curve selection and fader assignment options than ever before. This technology is used for both Channel Faders and the Crossfader.

- 3-Band Accelerated-Slope Full-Cut EQ
- Crossfader control includes Reverse and Mode switches and continuously adjustable Contours for A-and B-side. In Mode 1, the Contour control adjusts from smooth blend to extremely fast cut. Select Mode 2 for center-cut effect (PGM 1 and PGM 2 off in the center) with full range Contour control.
- Channel Fader controls include Reverse and Mode switches and continuously adjustable Contours. In Mode 1, the Contour control adjusts from smooth blend to extremely fast cut. Mode 2 is for Left/Right pan effect with Contour control. Channel Reverse switch allows left and right hand Channel Faders to control PGM 1 or 2.
- Two 10-segment meters with peakhold
- FlexFx effects loop allows assigning post-fader Program 1, 2 or both to the effects loop. A Wet/Dry pan control gives total control of how much processed signal is in the mix. With the effects loop post-fader, reverb & other delay effects respond to both Channel Fader and Crossfader operation.

MP 2

Mobile Club Mixer

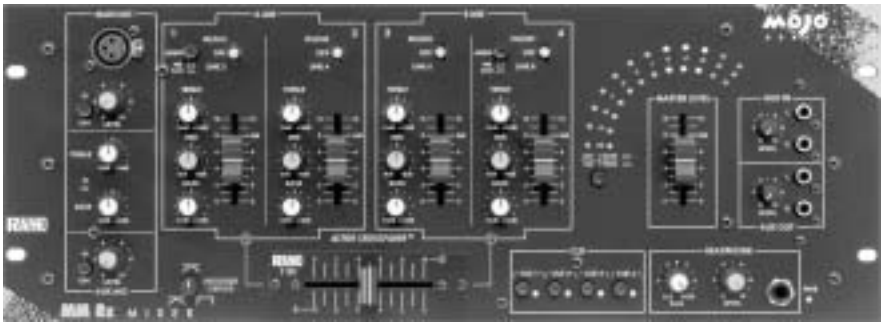


Compact and lightweight, the MP 2 is designed specifically for the on-the-go discriminating DJ. But don't let its petite appearance fool you. The MP 2 is packed with all the necessary features for a great performance and perfect mix.

- Each Program bus selects any of the four stereo inputs. Any input can appear on either side of the Crossfader!
- Two selectable phono/line inputs and two stereo line inputs for each of the two Program buses. Each Program bus has its own Bass and Treble EQ.
- Long lasting, noise-free 60mm active crossfader
- Outputs include a Tape Out with Pre/Post Mic sum, a balanced auxiliary output with level control for booth monitoring or feeding an additional zone and a balanced house feed.
- Balanced main mic input with full-range fader control.
- Additional features include Cue Engage on each bus, headphone slide fader that pans Cue to Master and a powerful low-noise headphone amp with level control.

MM 8z

Club Mixer



The MM 8z features eight stereo inputs. Each of the four assignable buses contains its own 3-band EQ, allowing separate equalization of different sources. The four faders are grouped in two pairs, allowing a custom mix of two input sources on each side of the crossfader. Two Pre/Post crossfader assign switches add flexibility. The first allows summing A Mix Channel 1, Pre or Post crossfader. The second allows summing B Mix Channel 4, Pre or Post crossfader. The Crossfader Contour control allows adjusting the "shape" of the Crossfader response from a gentle curve for smooth, long running fades, to the steep pitch required for top performance cut and scratch effects.

Active Crossfader Technology (MM 8z, MP 22z and MP 24z

Active Crossfader technology combines state of the art voltage controlled amplifier design with a professional quality crossfader control. This combination sets new standards for performance, reliability and serviceability. Virtually all crossfader noise is eliminated. Channel to channel crosstalk is greatly reduced and the off isolation of the faded channel is greatly increased.

Active Crossfader technology dramatically increases the service life of the crossfader. In the unlikely event of crossfader failure, there is no loss of signal. If a crossfader becomes rough or noisy, it may be "hot-swapped" during a performance with no interruption of the audio signal. Simply use the input faders to set the audio levels while the crossfader is out of service. The crossfader may be replaced with either a 60 mm version or a 45 mm version, your choice.

- Each of the four Phono (RIAA) inputs switch to line level via rear panel switches
- Main and Aux Mic inputs route through Bass and Treble controls
- The main program signal includes a stereo Master Loop for equalization or processing
- Main Mic input has an Effects Loop
- Tape output can be switched to exclude the Master Loop signal and record the unprocessed program signal
- Powerful headphone cueing system allows one or more channels to be cued at one time
- Metering can be switched to monitor stereo main output or mono cue on one side and mono program on the other
- Recessed top mounted jacks provide room for cabling in tight DJ coffins or racks without sacrificing its compact 4″ depth

MP 22z

Club Mixer



The MP 22z features six stereo Inputs with individual Trims. Each of the two assignable mixing buses contains its own defeatable four band EQ, allowing separate equalization for different sources. All of the fader controls feature protective dust dams. The Crossfader Contour control allows adjusting the "shape" of the Crossfader response from a gentle curve for smooth, long running fades, to the steep pitch required for top performance cut and scratch effects.

- Six stereo inputs with trims
- Front panel replaceable Active Crossfader
- 'Dipless' 60mm Crossfader; optional 45mm Crossfader
- Mic inputs have Bass and Treble controls as well as an Effects Loop. The MP 22z also features a second Remote Mic/Line Input and a stereo Aux Line Input.
- Each of the two phono (RIAA) inputs can be switched to line level use via rear panel switches
- Master outputs include a stereo Master Loop for equalization or processing, as well as two separate Zone Outputs. The Tape output and Zone 1 can be switched to include the Main Mic signal with the program signal.
- Versatile Program/Cue Metering and Headphone Cueing. The metering can be switched to monitor stereo program or mono cue on one side and mono program on the other.
- Recessed top mounted jacks provide room for cabling in tight DJ coffins or racks without sacrificing its compact 4″ depth.

MP 24z

Club Mixer



The Rane MP 24z is the standard for night club mixers. Featuring four stereo input mixing channels, each supplied with a selection of several phono and line inputs, the MP 24z easily suits the needs of the most complex audio/ video installations. A total of six stereo line inputs are available plus three stereo phono inputs.

- Front panel replaceable Active Crossfader
- 'Dipless' 60mm Crossfader; optional 45mm Crossfader
- Handles two mics simultaneously. Both mics may be contoured by the 3-band Mic EQas well as processed through a Mic Loop jack. On the rear, an XLR low-Z Main Mic Input with switchable Phantom Power, and a 1/4″ Hi-Z Aux Mic Input are both mixable.
- Each of the phono inputs can be converted to line inputs by moving internal jumpers.
- The Output section provides a Master Level fader, four band Program EQ (independent of the Mic EQ), two auxiliary output controls (Zone and Booth), a stereo 12-segment Peak Program Meter and full Headphone Control monitoring and cueing system.
- Besides all of the normal inputs/output, the rear of the MP 24z features a transformer isolated Light Control output withl evel control, balanced XLR Main Outputs, Program Equalizer EQ Range switch (±4 or ±8 dB) and a System Mono switch.
- Sliders are studio-quality as found on high quality audio mixing consoles.
- Another internal jumper allows defeating the Booth Ducker circuit when the mic is activated — this feature has been requested by those wishing to use the Booth Output as a second record output or second zone.
- Another jumper allows the Mic section to be mixed to the Tape Outs — the default setting is "music only".

MP 44

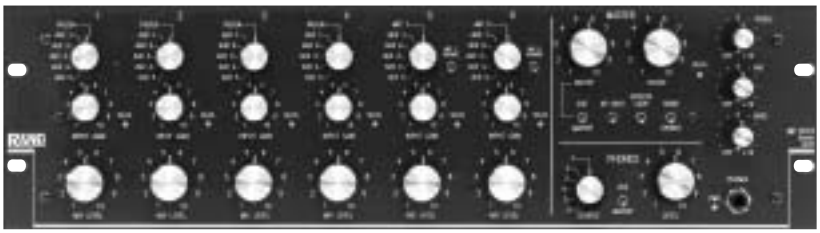
DJ Mixer



The MP 44 is a four-bus slide fader style mixer with a host of features to please the club DJ, club owner and the sound system installer. For the DJ 3-band Accelerated Slope full-cut tone controls on each bus, FlexFX flexible effects loop, A-side, B-side or Post crossfader assign on each input bus, VCA's on the crossfader and channel faders and 2 automatic crossfader CD triggers. For the club owner a remote DC master level volume control. For the installer: a high performance master limiter on booth and master outs, automatic emergency page input and the remote DC master level control. The MP 44 is housed in a 5U chassis.

- Two Auxiliary Inputs are provided. Aux Input A is unbalanced, while Aux Input B is balanced. Each Input features off to +6 dB Level controls.
- Two Aux outputs (balanced and unbalanced) with up to +6 dB level controls.
- Each output may be assigned Pre- and Post-Master Effects loop or Post Master Effects loop plus Mic. These features provide the flexibility to use the Aux outputs for recording or additional zones.
- High performance VCA level controls are used for House and Booth level, providing very low travel noise and long life. A Master left/right balance control and recessed Master Mono switch with LED indicator are provided.
- High-performance Master Limiter with true rms detectors limits House and Booth maximum levels. The rear panel threshold control range is ±20 dBu. Threshold indicators are on the front and rear panels.
- Applied signal on the Emergency Page Input automatically mutes all program material at House and Booth Outputs. The Emergency Page level is set independently using a recessed rear panel control.
- Two CD Trigger Outputs are provided via 3.5mm (1/8″) mini TRS jacks. Outputs are normally high with 50 ms active low trigger pulses. Start is on tip and stop is on sleeve. Ring has no connection. The CD Trigger Enable switch is located on the front panel.

MP 2016 *Rotary Mixer*



The MP 2016 is a rotary-controlled mixer featuring six input channels, Booth and House outputs, Pre-Effects Tape outputs, Post-Effects Tape outputs with level control, switchable Effects Loop, and Headphone cueing. Input Channels 1 through 4 feature high quality RIAA phono preamplifiers which may be switched to line input via rear panel switches. Input Channels 5 and 6 each provide a balanced mic preamp with Input Gain trim, Mic/Line switch, two-band EQ, Mic Engage switch and an internal Booth Ducking enable jumper. In addition to the six dedicated preamplifiers, all six Input Channels may select any of the five stereo Auxiliary Inputs.

- Each Input Channel provides a six-position source selector, Input Gain control for matching levels, Signal/ Overload indicators and a studio-grade Master Mix control.
- The Booth Output may select Master or Cue as its source.
- Master Mix circuit provides Signal/Overload indicators, Mono switch and high-pass/ rumble filters. In addition, the MP 2016 features patent pending high performance threeband, full-cut, Accelerated-Slope tone controls.
- An ultra-low noise, high power headphone amplifier also allows monitoring the Master Mix or Cue Mix.
- Exceptionally ergonomic, intuitive layout packaged in a compact 19″, 3U rack mount chassis, a scant five inches deep.

DJ MIXER COMPARISON CHART

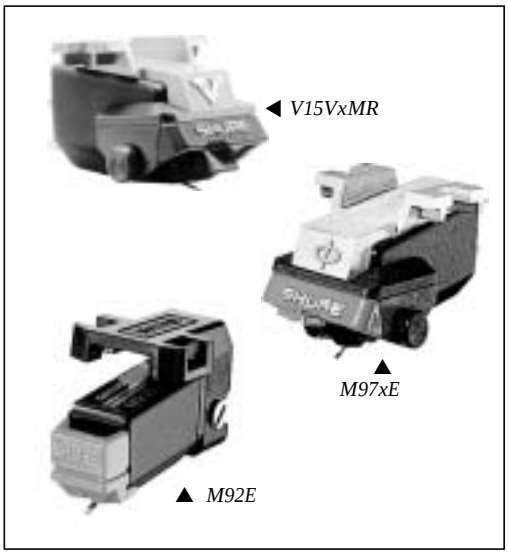
	Features	MM 8z	MP 22z	MP 24z	MP 44	MP 2016 / XP 2016
Stereo Inputs	Phono (RIAA)	(4) RCA	(2) RCA	(3) RCA	(4) RCA	(4) RCA
	RIAA Defeat (Line Input Option)	External	External	Internal	External	External
	Line	(4) RCA	(4) RCA	(5) RCA; (1) 1/4″	(4) RCA	(5) RCA
	Aux (Direct to Master Out)	Yes	Yes	No	Yes	No
	Total Inputs	8 + 1	6 + 1	9	8 + 1	9
Mic Inputs	Main (XLR)	Front & Rear	Front	Rear (XLR + 1/4″)	Rear	Rear
	Phantom Power	No	No (+15 VDC)	Yes(+12 VDC)	Yes	No
	Rear Aux / Remote / Wireless	1/4"	1/4"	1/4"	XLR + 1/4"	XLR
	Mic EQ	2-band	2-band	3-band	3-band	2-band
	Mic Effects Loop	Main Only	Main + Aux	Main + Aux	Main + Aux	No
	Mic-Activated Ducker (Booth)	No	No	Yes	Yes	Yes
Faders	Crossfader Contour Control	Yes	Yes	No	Yes	Yes (XP 2016)
	Input Pre-Post Crossfader Assign Switches	Yes	No	No	Yes	Yes (XP 2016)
	Input / Master Fader Travel	(5) 45 mm	(3) 60 mm	(5) 60 mm	(5) 60 mm	(9) Rotary
	Type	EQ-style	Pro-style	Studio-style	Studio-style	Studio-style
Equalization	Main Mix	-	-	(1) 4-Band	-	(1) 3-Band
	Source Buses	(4) 3-Band	(2) 4-Band	-	(4) 3-Band	(6) 3-Band (XP 2016)
	EQ Bypass	No	Yes	Yes	No	No
	Selectable Boost/Cut Range	No	No	Yes	No	No
Metering	Number of LEDs & Shape	(12) Arc	(12) Vertical	(12) Horiz.	(10) Horiz.	(10) Horiz. (XP 2016)
	Simultaneous Master and Cue	Yes	Yes	No	Yes	No
	Meter Sensitivity Control	No	No	Yes	Yes	No
Headphones	Master-Cue Panning	Yes	Yes	Yes	Yes	No
	Mono Cue Left -					
	Mono Master Right	No	Yes	Yes	No	No
Stereo Outputs	Main Out 1	Bal 1/4"	Bal 1/4"	Bal XLR	Bal XLR	Bal XLR
	Main Out 2	No	No	Unbal 1/4″	Unbal RCA	Unbal RCA
	Max Out Gain Reduction Control	No	No	Yes	Limiter	No
	Zone 1 (or Aux)	Unbal RCA	Unbal RCA	Unbal 1/4″	Bal 1/4″ + Unbal RCA	No
	Zone 2 (or Booth)	No	Unbal RCA	Unbal 1/4″	Bal 1/4″	Unbal RCA
	Zone without EQ	No	No	Int. Jumper	No	No
	Mic To Zone or Aux	Yes	Ext. Switch	Int. Jumper	Yes	No
Tape Output	Pre/Post Main Effects Loop Choice	Yes	No	Yes	Yes	Yes
	Mic & Aux to Tape Out	No (Use Aux)	Ext. Switch	Int. Jumper	Front Switch	Yes
Miscellaneous	Switchable Main Effects Loop	No	No	Yes	Yes	Yes
	Lighting Control Output	No	No	Yes	Yes	No
	System Mono Switch	No	No	Yes	Yes	Yes
	Rack Height x Depth	4U (7″) x 4″	4U (7″) x 4″	4U (7″) x 5¼″	5U (8¾″) x 5¼″	3U (5¼″) x 5¼″ (each)

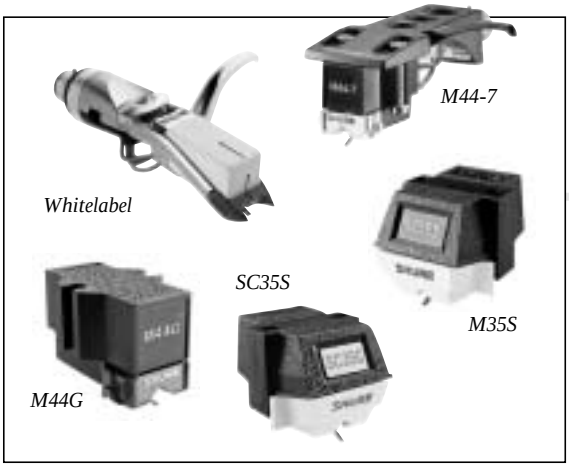
AUDIO CARTRIDGES

Hi-Fi/Home Audio Cartridges

Shure Hi-Fi cartridges are distinguished for their high groove tracking ability and warm, natural sound reproduction. Each cartridge comes with a user guide, mounting hardware, stylus guard and warranty. All models employ the low-mass, high output moving magnet structure invented by Shure.

	V15VxMR	M97xE	M92E	M78S
Features	Audiophile	Audiophile	Value	78 RPM
Stylus Mass	Ultra low	Very low	Low	Low
Sound Accuracy	Ultra high	Very high	Very good	Good
Record Wear	Ultra low	Very low	Low	Low
Tone Arm Mount	Standard 1/2″	Standard 1/2″	P with standard adapter	Standard 1/2″
Diamond Stylus Tip	Micro-ridge	Elliptical	Biradial	Spherical
Side/Front Radii	0.15 x 3.0 mil	0.2 x 0.7 mil	0.4 x 0.7 mil	2.5 mil
Optimum Tracking Force	1 gram	1.25 grams	1.25 grams	1.5 to 3 grams
Price	**325.00**	**89.00**	**35.00**	**79.95**





DJ Record Needlz

An extensive range of DJ Needlz for scratching, mixing and spinning in any performance situation. Each versatile needle features the unique sound characteristics developed by Shure for low record wear, high skip resistance, and reliable record cueing.

All DJ Cartridges Feature:

- Low mass, high output moving magnet construction invented by Shure.
- User guide, mounting hardware, stylus guard, and warranty.
- Wide-diameter, high strength stylus assemblies for dependable back cueing.
- Spherical diamond tips that ride gently over a large footprint in the record groove.
- Cutaway grips for high visibility of the irradiant orange stylus tip.
- Standard 1/2″ mounts or integrated mounts compatible with all leading DJ turntables.

	Whitelabel	M44-7/M44-7-H	M44G	M35S	M35X	SC35C	M25C
Features	Club/Party	Turntablist	Club/Party	Dance	House/Techno	Standard	Fundamental
Application	Mix, Spin	Scratch	Scratch, Mix, Spin	Mix, Spin	Mix, Spin	Scratch, Mix, Spin	Mix, Spin
Skip Resistance	High	Ultra high	Very high	High	High	Very high	High
Sound Emphasis	Solid drop bass, flat mids, accented highs	Big bass	Detailed highs & lows	Rich, warm, tight in all frequencies	Extreme clarity, deep drop bass	Clarity thru frequency range	Solid bass, clear mids
Output	High	Ultra high	Very high	High	High	High	High
Record Wear	Very low	Ultra Low	Ultra low	Low	Very low	Low	Low
mV RMS[1]	6.0 mV	9.5 mV	6.2 mV	5.0 mV	6.0 mV	5.0 mV	5.0 mV
Tracking Force	1.5 to 3.0 grams	1.5 to 3.0 grams	.75 to 1.5 grams	3.0 to 4.0 grams	1.5 to 3.0 grams	4.0 to 5.0 grams	1.5 to 3.0 grams
Frequency Response[2]	20 - 20,000 Hz	20 - 17,000 Hz	20 - 19,000 Hz	20 - 20,000 Hz	20 - 20,000 Hz	20 - 17,000 Hz	20 - 18,000 Hz
Stylus Radius	0.7 mil	0.7 mil	0.7 mil	0.7 mil	0.7 mil	0.7 mil	0.7 mil
Price	**109.95**	**76.95 / 99.95**	**66.50**	**49.95**	**49.95**	**39.95**	**34.95**

1. at 5 cm per second peak velocity 2. essentially flat from:

TURNTABLES

Completely reinventing the wheel is not an easy job, but someone's got to do it. So starting with the industry standards, Stanton has rebuilt the turntable, kept all of the desired features that today's DJs need and even included some from the future. So whether you're looking for a straight-arm or S-arm turntable, analog or digital output or the ability to change speeds without changing the key, Stanton has got a turntable for every DJ. Once you put your hands on these decks, you'll never look at turntables the same way again.



STR8-30
Stanton's entry-level direct-drive turntable. It comes with a Stanton 500 B cartridge pre-mounted on a headshell.



STR8-50
Belt-drive turntable with 500 AL II cartridge pre-mounted on a headshell.



STR8-60
Direct-drive turntable. Includes 500 AL II cartridge pre-mounted on a headshell.



STR8-80
Direct-drive digital output turntable. Includes 520SK cartridge pre-mounted on a headshell.



STR8-90
Based on the popular STR8-100. Although completely analog, this turntable still outperforms traditional S-arm turntables.



ST-100
S-arm version

STR8-100
Straight-arm version



	STR8-30	STR8-50	STR8-60	STR8-80	STR8-90	STR8-100 / ST-100
Turntable Type	2-speed full manual	2-speed full manual	2-speed full manual	3-speed full manual	3-speed full manual	3-speed full manual
Driving Method	Direct Drive	Direct Drive	Direct Drive	Direct Drive	Direct Drive	Direct Drive
Motor	8 pole, 2 phasebrushless DC motor			16 pole, 3 phase brushless DC motor		
Turntable Platter	Aluminum die-cast 330mm diameter					
Speed	33⅓, 45 rpm	33⅓, 45 rpm	33⅓, 45 rpm	33⅓, 45, 78 rpm	33⅓, 45, 78 rpm	33⅓, 45, 78 rpm
Pitch Control	+/- 10%	+/- 10%	+/- 10%	+/- 8%, +/- 16%, +/- 25 selectable		
Starting Torque	more than 1 Kgf-cm	more than 1 Kgf-cm	more than 1 Kgf-cm	more than 2.2 Kgf-cm	more than 2.2 Kgf-cm	more than 2.2 Kgf-cm
Wow and Flutter	less than .15% WRMS	less than .25% WRMS	less than .15% WRMS	less than .20% WRMS	less than .20% WRMS	less than .20% WRMS
S/N Ratio	55dB (DIN-B)	55dB (DIN-B)	55dB (DIN-B)	50dB (DIN-B)	50dB (DIN-B)	50dB (DIN-B)
Needle Pressure Range	0-4g	0-4g	0-4g	0-4g	0-4g	1.5-2g
Dimensions (WxHxD)	17¾ x 5 x 13¾″	17¾ x 5 x 13¾″	17¾ x 5 x 13¾″	17¾ x 5 x 14⅝″	17¾ x 5 x 14⅝″	17¾ x 5 x 14⅝″
Weight	7.75 lbs.	19 lbs.	19 lbs.	19 lbs.	19 lbs.	19 lbs.

MASTER SERIES CARTRIDGES

Stanton has served the styli industry since 1946 and has been at the leading edge of technology and development both in the broadcast studio and recording industry as well as the club and professional DJ.
They are the leader in high-quality phono cartridges and are available for many professional applications including club mixing, broadcast and scratching as well as production/remix use.

Groovemaster II RM

With an elliptical super high polished diamond. Ideal for club mixing, broadcast and production/remix use.

Groovemaster II PRO

Hand-built "best-of-the-best" DJ cartridge. The spherical tip has a Super High Polish diamond tip to reduce vinyl burn and give longer tip life. No other DJ cartridge in the world can make these claims. The gold finish and gold connector sets you apart from the crowd in appearance and performance. Packaged with an extra stylus.

Trackmaster II SK

With a spherical super high polished diamond, the Trackmaster II SK satisfies the needs of a scratch / hip-hop DJ. It includes a free stylus.

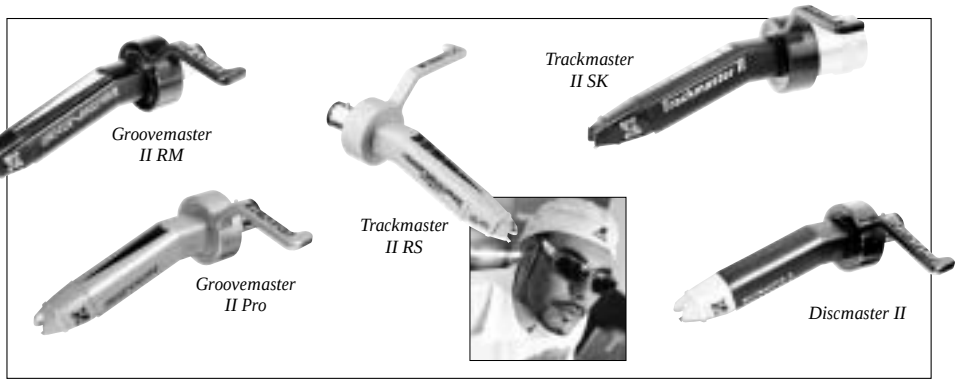
Trackmaster II RS

Roger Sanchez Signature Model

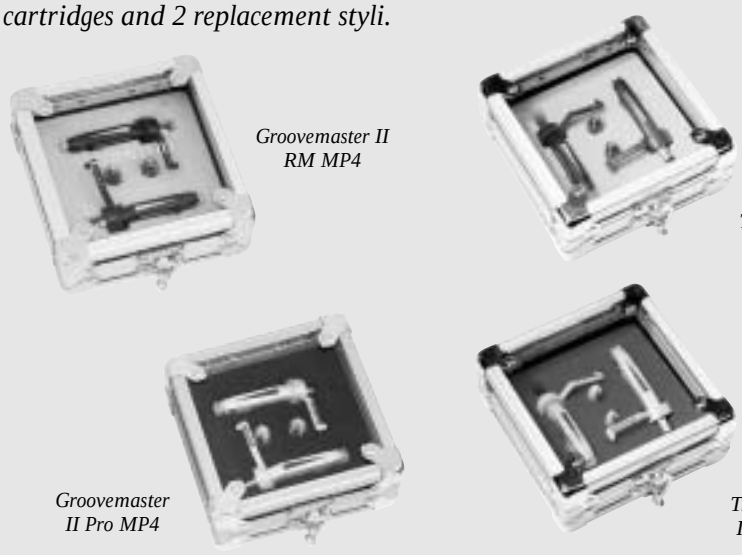
Designed based on Roger Sanchez' specifications. Ideal for club mixing and production/remix use. This model carries an eliptical super high polished diamond and includes a free stylus.

Discmaster II

Designed exclusively for the competition scene. Fat, funky and fabulous, the Discmaster II dazzles with its superior groove holding capabilities, as well as its high output and terrific bass performance. It operates at relatively light tracking forces (2 to 5 grams, optimal at 4 grams), instead of the very heavy forces required by others, so it won't wear out your records.



The matched pair (MP4) versions of Master Series cartridges come in a small aluminum case with a clear plexi-glass top, and include 2 matched cartridges and 2 replacement styli.



	Groovemaster II RM	Groovemaster II Pro	Trackmaster II SK	Trackmaster II RS	DiscMaster II
Frequency Response	20-20kHz	20-20kHz	20-20kHz	20-20kHz	20-20kHz
Output @ 1kHz	7.0mV	7.0mV	7.0mV	7.0mV	7.0mV
Channel Balance	within 2dB	within 2dB	within 2dB	within 2dB	within 2dB
Tracking Force	2-5 grams	2-5 grams	2-5 grams	2-5 grams	2-5 grams
Stylus Tip	.4 x .7mil elliptical	.7mil spherical	.7mil spherical	.4 x .7 elliptical	.7mil spherical
Replacement Stylus	G II - RM	G II - Pro	T II - SK	T II - RS	DM-II

B&H PHOTO - VIDEO - PRO AUDIO
EQUIPMENT LEASING AVAILABLE

STANDARD MOUNT CARTRIDGES

500 AL II

A professional industry standard cartridge. It provides a balance of economy, reliability, ruggedness and optimum performance. From studio use to the mobile scene, it is built for the most discriminating professional.

500 AL II Twin
Package of two 500 AL II cartridges.

505 SK II

A budget-priced scratching cartridge with a super sturdy stylus that has been coupled with pumped-up output for the deepest ottom and in-your-face sound.

505 SK II Twin
Package of two 505 SK cartridges.

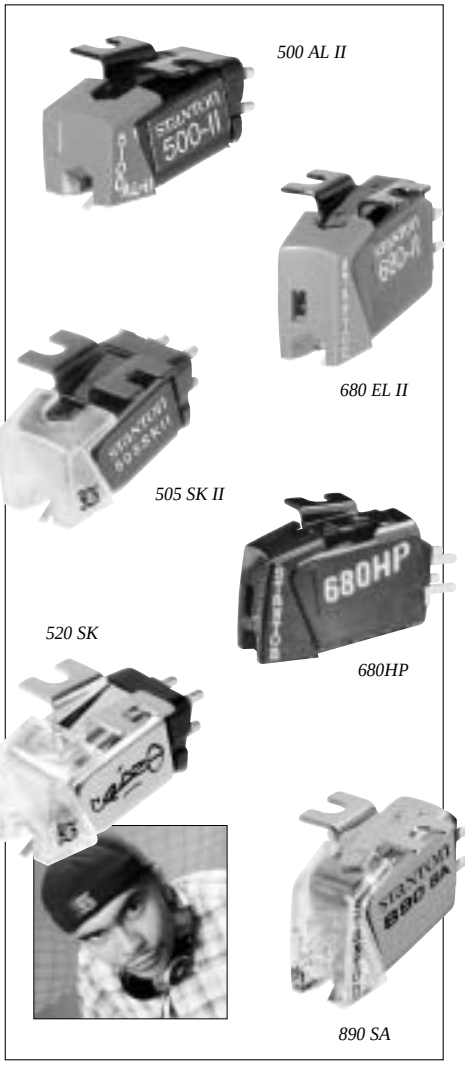
505 SK II H4 Twin
Twin pack of 505 SK cartridges mounted on H4 headshells. Includes 2 extra styli.

520 SK DJ Craze Signature Model

DJ Craze, DMC World Champion 3 years running, has selected this cartridge as the all time best competition cartridge. The sturdy suspension couples the thick wall cantilever to allow for high tracking forces and the grounded cartridge body minimizes hum and delivers great quality sound.

520 SK Twin
Package of two 520 SK cartridges. Includes 2 extra styli.

520 SK H4 Twin
Twin pack of 520 SK cartridges mounted on headshells. Includes 2 extra styli.



680 EL II

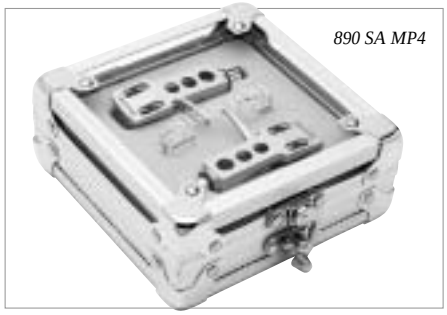
Part of Stanton's popular 4 coil standard mount stereo cartridge line. It continues the renown Stanton ruggedness balanced with total musical responsiveness.

680HP

Designed especially for club use, the 680HP delivers pumped up output and suspension and a wide-band frequency response that gives you high energy record playback and the extra punch you need to quake the dance floor.

890 SA

A high end performance cartridge for the Scratch Artist. With the collaboration of DJ Focus and DJ B-Side, Stanton has improved and reissued one of the best cartridges from thier past.



890 SA MP4
Matched pair versions of the 890 SA in a small aluminum case with a clear plexi-glass top. Includes 2 matched cartridges mounted on headshells and 2 replacement styli.

	500 AL II	505 SK II	520 SK	680 EL	680HP	890 SA
Frequency Response	20-17kHz	20-17kHz	20-17kHz	20-18kHz	20-18kHz	20-20kHz
Output @ 1kHz	4.6mV	6.0mV	6.0mV	3.9mV	7.0mV	6.5mV
Channel Balance	within 2dB	within 2dB	within 2dB	within 2dB	within 2dB	less than 2dB
Tracking Force	2-5 grams	2-5 grams	2-5 grams	2-5 grams	2-5 grams	2-5 grams
Stylus Tip	.7mil spherical	Spherical .7mil	Spherical .7mil	.3 x .7 elliptical	.7mil spherical	.7mil spherical

REPLACEMENT STYLI

D-8900 SA
for 890 SA.....**Call**

D-8900 SA DP
For 890 SA. Three-pack.....**Call**

D-6800 HP
For 680 HP.....**Call**

D-6800 EL II
For 680 EL II. .....**Call**

D-6800 EL II DP
For 680 EL II. Three-pack .....**Call**

D-5200 SK
For 520 SK.....**Call**

D-5200 SK DP
For 520 SK. Three-pack.....**Call**

D-5100 SK II
For 505SK II.....**Call**

D-5100 SK II DP
For 505SK II. Three-pack.....**Call**

D-5100 AL II
For 500 AL II.....**Call**

D-5100 AL II DP
For 500 AL II. Three-pack.....**Call**

H4S Headshell
Silver color. Replacement .....**Call**

STANTON

FINAL SCRATCH

Future DJ Technology with a Blast from the Past!



Final Scratch is the best and only real bridge for the professional and home DJ to enter and control the digital world. By using any standard turntable/mixer setup, a DJ can now manipulate digital music the way the has always done it - by hand and by vinyl. Imagine having an endless supply of records to spin all night long… you can even needle drop MP3 files until your hands go numb! Think of Final Scratch as your never ending record crate - without having to break your back to carry them all. Simply load any digital audio file onto your computer - then pick and choose from your own playlist. Save and load any production, remix or edit that you have created and play it at that very moment or take it to a gig the same night without having to cut an acetate or make test pressings.

FS10K Package
Includes Final Scratch software, two encoded vinyl records, and the ScratchAmp to connect the computer to your existing turntable/mixer setup

FS10K System
Supports 2 turntables, includes the Final Scratch software, the ScratchAmp, and 2 FS vinyl records



SOFTWARE

The Final Scratch software allows the user to save the audio files in various playlists, called Record Boxes, as well as to search and assign the audio files to either turntable. The software provides a visual representation of the music, needle position for each audio file, and remaining time of each audio file.

RECORDS

The vinyl records included in Final Scratch contain digital information, and are used to mix digital audio files like mp3, .wav, .aiff and audio CD. All mixing is done directly from the turntables, including seeking, pitch shifting, cueing, spinning up and down and even scratching. The records can be used in combination with traditional analog records when mixing your set.

HARDWARE

The ScratchAmp acts as a powerful sound card to connect all the components together to form the system. It provides 2 sets of phono inputs and outputs and 2 sets of line outputs. The ScratchAmp processes the signal from the Final Scratch records and sends it to the computer via USB.



System Requirements:

500 MHz or faster Intel compatible CPU, 128 MB RAM, IDE hard disk, 2.5 GB of available hard disk space (500MB for operating system, approximately 2GB for 200 seven-minute songs), USB port, CD Rom player boot enabled, a traditional DJ setup of 2 turntables and 2-channel mixer with phono/line select

CD PLAYERS

S-250 Table-Top CD Player

Stanton's first table-top CD player comes with fader start and digital outputs. The control panel and LED screen are at the front of the unit, just where DJs need them. It features:

- Instant Start
- Digital outputs
- Fader start
- Pitch bend
- Selectable pitch control [+/-8%, 12%, 16%]

S-550 Professional Dual CD Player

Offering an improved jog-wheel, the S-550 can read CD-R's with 8x oversampling and 16% pitch bend. It is karaoke compatible and features:

- S/PDIF digital outputs
- Auto Cue function / Instant Start
- Auto Cue Pitch Fader (+10)
- Selectable pitch range: +/-8%, +/-12%, +/-16%
- Manual Pitch Bend (up to +16)
- Relay Playback (Flip Flop)
- Fader Start outputs
- LCD screen displays minutes, seconds, frames, pitch, and looping
- 4 speed scan
- 20 track program play single or continuous



S-250

S-650 Professional Dual CD Player

Loaded with tons of features that are designed for today's professional DJs.

- Buffer Memory
- Anti-shock
- Seamless Loop
- Digital Outputs
- Relay Playback (Flip Flop)
- Fader Start (cables included)
- Instant Start
- Up to +/- 16% Pitch Bend
- Multipurpose Jog Wheel
- Auto Cue eliminates dead space
- Pitch Display
- 8x Over sampling
- 30 Programmable Play Tracks on each side
- 6 Different Speed Scans
- 8cm CD Singles Compatible
- Instant Cue point recall
- 60 seconds transport protection (open tray closes after 60 sec.)
- Reads CD-R
- Repeat one, Repeat all, Single, Continue, Sleep Modes

S-700 Professional Dual CD Player

All the features of the S-650 plus–

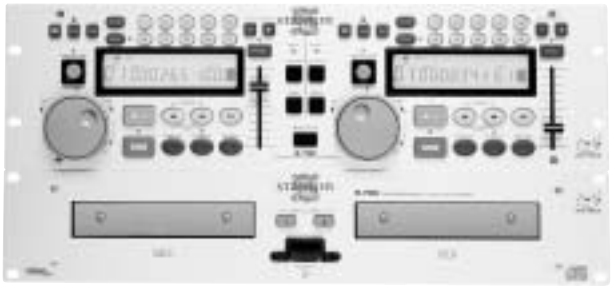
- Seamless Cue (SQ)
- Turntable brake and motor off effects
- Reverse play
- BPM counter
- Direct track access
- 10 Programmable Cue Points on each side
- Selectable Pitch Control (+/-8%, 12% or 16%)



S-550



S-650

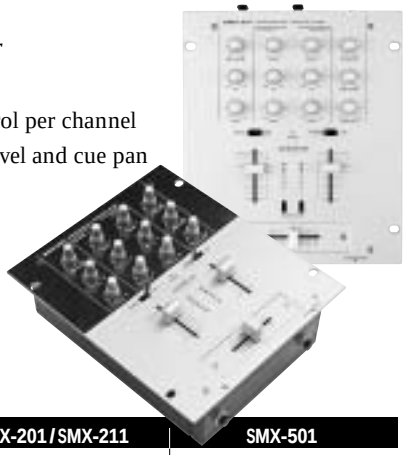


S-700

	S-250	S-550	S-650	S-700
Output Level	2.0V +/- 1 dB	2.0V +/- 1 dB	2.0V +/- 1 dB	2.0V +/- 1 dB
Frequency Response	20Hz - 20 kHz +/- 1 dB	20Hz - 20 kHz +/- 1 dB	20Hz - 20 kHz +/- 1 dB	20Hz - 20 kHz +/- 1 dB
Quantization	16 bit	16 bit	16 bit	16 bit
De-emphasis Response	+ 0.5 dB	+ 0.5 dB	+ 0.5 dB	+ 0.5 dB
Channel Separation	86 dB	86 dB	86 dB	86 dB
T.H.D.	0.02%	0.02%	0.02%	0.02%
S/N Ratio	88 dB	88 dB	88 dB	88 dB
Dynamic Range	88 dB	88 dB	88 dB	88 dB
Power Consumption	25w	25w	25w	25w
Rack Mountable (Player/Control Unit)	19″ (2U); (2U)	19″ (2U); (3U)	19″ (2U); (2U)	19″ (2U); (3U)
Player/Control Unit Dimensions (WxHxD)	19x3.5x10″; 19x3.5x1.6″	19x3.5x10″; 19x5.25x1.6″	19x3.5x10″; 19x3.5x1.6″	19x3.5x10″; 19x3.5x1.6″
Player/Control Unit Weight	11 lbs.; 4.5 lbs.	11 lbs.; 4.5 lbs	11 lbs.; 4.5 lbs	11 lbs.; 4.5 lbs.

CLUB & MIX MIXERS

SMX-201
DJ Mixer



- 2 line and 2 phono inputs
- 2 band EQ and Gain control per channel
- Cue section features cue level and cue pan
- Mic input with 2 band EQ
- User replaceable faders
- Compact size

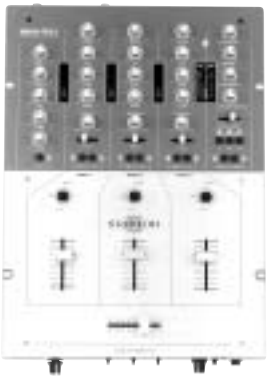
SMX-211

Same as the SMX-201 with Crossfader Start

	SMX-201 / SMX-211	SMX-501
Line Inputs	2 (RCA), -10dB/47k Ω	2 (RCA), -10dB/10k Ω
Phono Inputs	2 (RCA), -50dB/47k Ω	2 (RCA), -50dB/47k Ω
DJ mic Input	1 (1/4″), -50dB/6k Ω	2 (1/4″), -50dB/100k Ω
Master Output	1 (RCA), -10dB/47k Ω	2 (XLR/RCA)
Mic Tone	+/-10dB (Hi, Low)	
Tone Control	+9/-26dB (Hi, Mid, Low)	+9/-40dB (Hi, Mid, Low)
Headphone Output	1 (1/4″), -3dB/33Ω	2 (1/4″ and 1/8″)
Frequency Response	20 Hz - 20 kHz, +/- 2dB	20 Hz - 20 kHz, +.5/-2dB
Gain Control	0-20dB	
S/N Ratio	Less than 75dB	Better than 100dB
T.H.D.	Less than 0.2%	Less than 0.05%
Dimensions (WxHxD)	10½ x 9 x 3½″	14 x 10.25 x 4.75″
Weight	6.6 lbs.	9.25 lbs.

SMX-501
Hybrid 3-Channel "Techno/Scratch" Mixer

This "one of a kind" mixer combines the best of both worlds and bridges the gap between Battle & Club DJs.



- 3 channel scratch mixer
- Assign effects loop to each individual channel, plus the mic
- Custom High Quality Penny and Giles Crossfader with VCA curve adjustment and crossfader assign
- Assignable Crossfader (A, B, Bypass)
- VCA Crossfader curve control & reverse
- Curve control and reverse on Linefaders
- Each channel features a gain control, slider Pan, and a versatile 3 band EQ with complete KILL
- Balanced XLR master output (unbalanced RCA also provided)
- Booth output
- Rotatable 3 Position phono/line switch with flash feature
- Separate channel input and master output LED meters
- Headphone mute
- Fully assignable Send/Return connections for external effects
- Pre Fader Listen (PFL) of channels and effects loop with Cue Pan control
- 3 line, 3 phono, and 2 mic inputs
- Mic channel features gain controls and a 3 band EQ
- Channel fader reverse switch ◆ User replaceable faders

Penny & Giles Faders

P&G faders have been used in broadcast and recording consoles such as SSL, Neve, Mackie (Logic Control) and many others for many years. They are considered by many of the world's top professional audio and recording engineers to be the highest quality faders in the world. Until recently, they were considered to be too expensive for DJ mixers. Stanton approached P&G to make custom faders for its mixers, such as the SA-12, SA-8, and SMX-501. Other companies have now followed (Rane, Allen & Heath, and Vestax), but they are not the same custom P&G faders that Stanton uses.

Focus Fader V2 Digital Optical Crossfader

Improving on the original and world's first optical fader design that brought the industry and art to a new plateau, the Focus Fader V2 is truly curve adjustable to accommodate to any style of DJ artist. If your style requires a smooth fade for long mixes or a razor sharp cut-off for precise scratching, V2 is the answer.
No more bleeding, no more static, no more wasted time, just hours of practice enjoyment and flawless performance. The Focus Fader V2 is highly advanced and opens many doors for innovations in DJ equipment technology. Fitted with an array of optical sensors and microprocessor controlled to eliminate the need for contacts or graphite material, the V2 will outlast any standard graphite or conductive plastic fader on the market.

Focus Fader V1 Digital Optical Crossfader

The Focus Fader V1 is the world's first true optical crossfader for scratch artists. Without the standard graphite contacts found on most faders, the V1 offers no no resistance at all which allows for quicker movements while scratching. The optics provide a true ON/OFF cut instead of the sharp fade found on all other battle mixers (which simulate a cut).
Since there are no contacts, it also means that there will never be any degredation of the audio signal due to the crossfader
Focus Fader V1 is available for SK-1, SK-2, SK-4, SK-5 and SK-6

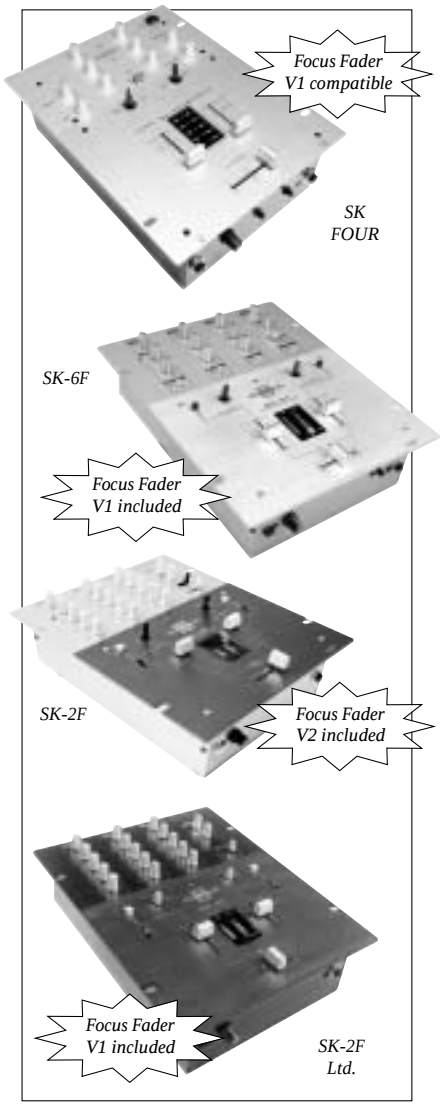
SK SERIES MIXERS

SK FOUR

- 2 line and 2 phono inputs
- 2 band EQ per channel
- Smooth crossfader with VCA system
- Crossfader reverse and curve adjustment with bi-color LED
- Unique PROTEKT panel hides all screws underneath the top panel
- Master / Cue monitor select ◆ Headphone mute
- 2 headphone inputs (1/4″ and 1/8″ minijack)
- TRS balanced Master output
- 8 position input select

SK-6F

- PROTEKT panel hides screws underneath the top panel for hassle-free fader operation
- Includes both the Focus Fader V1 and the CF-SK6 super smooth crossfader with photo-coupler system and curve adjustment
- Master/cue switch
- Dual Headphone Jacks (1/4″ and 1/8″)
- Program reverse function reverses the input signals of channels 1 and 2
- Reverse function for crossfader and each channel fader with 2 color LED
- 2 line and 2 phono inputs. Gain and 2 band EQ (+9/-26dB) on each channel
- Effects send/return for external effects. Effects are post-crossfader. ON/OFF switch for each channel plus MIC and Cue
- 3-way, 8 position line/phono toggle switches with FLASH feature
- Mic input with volume control and 2 band EQ (+10/ -10 dB)
- Headphone input with volume control and cue pan fader, plus headphone mute
- Master output (TRS balanced 1/4″ and unbalanced RCA)



SK-2F

- 10″ 2-channel battle mixer with 2 line and 2 phono inputs
- Includes Focus Fader V1 and the CF-SK2 long body crossfader
- Reverse function and curve adjustment for crossfader and each channel fader with 2 color LED
- 3-way, 8-position line/phono toggle switches with FLASH feature
- PROTEKT panel hides screws underneath top panel for hassle-free fader operation
- Channel faders are close together for easy one-handed dual fader use
- Gain, 3 band EQ (+9/-26dB) and Pan control on each channel
- Headphone output with volume control, cue pan fader, Master/Cue select and headphone mute function
- Master output (TRS balanced 1/4″ and unbalanced RCA)

SK-2F Ltd.

- An upgrade to the standard SK-2F. It has been fitted with the new Focus Fader V2 Digital Optical Crossfader with adjustable curve control, the OS2 optical scratch switches, and a second headphone output (3.5mm mini jack).
- The OS2 is the first phono/line switch that users a fader as the user interface. This allows the same hand/wrist movement to be used when using the crossfader and the OS2, which translates into efficiency in scratching performance. It's flat handle and soft-slide action makes it easy to perform any existing techniques including the Crab Scratch.

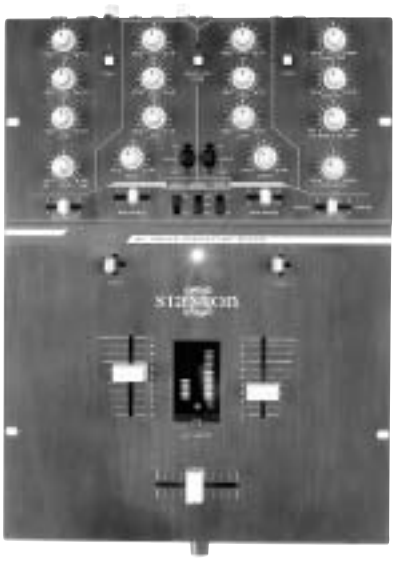
	SK FOUR	SK6F	SK-2F	SK-2F LTD
Channels	2	2	2	2
Crossfader	CF-SK4	CF-SK6	CF-SK2/CF-F1	CF-F2
CF-F1 Compatible	no	yes	yes(included)	no
CF-F1-SK5 Compatible	yes	no	no	no
CF-F2 Compatible	no	no	no	yes
Line Fader		LF-SK6	LF-SK2	
Toggle Switch	8 position	8 position/flash	8 position/flash	OS2 - Optical Scratch Switch
EQ	2 band	2 band +9/-26dB	3 band +9/-26dB	3 band +9/-26dB
Headphone Mute	yes	yes	yes	yes
Adjustable Curve on CF	yes	yes	yes(CF-F1 not adjustable)	yes
Adjustable LF	no	no	yes	yes

STANTON

SA SERIES MIXERS

SA-8 DJ Focus Signature Mixer

"From the mind of Focus" comes the SA-8, the ultimate performance mixer. Never before has so much scrutiny been put into the layout of a scratch mixer. Each feature was carefully arranged and placed according to a DJs natural hand movement (notice where the gain is). The SA-8 offers features for every type of DJ, either strictly scratch, live performance, production, or mix. This "studio-style" scratch mixer (designed by Focus) gives you the technology of a professional console mixer compacted and ergonomically arranged in a format that is preferred by top DJs.



- Individual REC / FX and Direct outputs (each turntable signal can be sent separately though 4 direct outputs located on back of mixer)
- Custom designed P&G (Penny & Giles) crossfader (additional CFF2crossfader included) and custom designed P&G linefaders
- One of-a-kind knobs and faders design to fit the DJ fingers
- Post Fader Cue so a DJ can listen to his scratches in the headphone while using the crossfader as normal
- 3-way adjustable curve control on linefaders
- Autocue eliminates the need for touching the cue pan (switching from Ch.1 to Ch. 2)
- Adjustable Rumble Filter allows the DJ to eliminate (or add) low end frequencies
- Booth Output
- An ergonomic layout so there is no bumping or cluttering of knobs during use
- OSII Optical Switch (for "noise-free" transforms)
- Mono/Stereo Switch per channel
- Program Reverse
- 2-Dual headphone outputs (2¼″ and 2⅛″ minijack) with headphone mute
- Phono / Line / FX Toggle Switch
- Multiple Outputs: 2-RCA (Unbalanced), 2¼″ Direct Out (Balanced), 1¼″ Master (Balanced), Plus FX Send (¼″) ·Multiple Inputs: 2-RCA (Phono), 2-RCA (Line), 1-RCA (Aux), Plus FX Return (¼″)
- Screwless faceplate



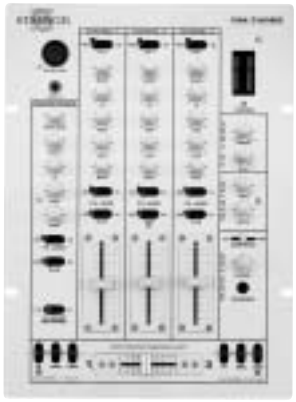
SA-12 DJ Craze Signature Mixer

"It's only the 3x champion, DJ Craze signature mixer!" This battle mixer (designed for the champ) has all the flexibility and quality of a club mixer but has all the requirements of today's Turntablist. The SA-12 incorporates a user-friendly removable effect module that any DJ can install [one of three sound modules available]. These modules can then be assigned to either channel (1/2), the microphone, and even the main mix.

- Custom P&G (Penny & Giles) faders with curve adjustment
- OS2 Optical phono/line switches with LOCK feature (to avoid mistakes)
- Removable effects module (MOD-1) includes pitch shifter, flanger, and delay all with parameter and mix controls
- Effects assignable to channels 1 and 2, master output, and mic/line 3
- Footpedal output for effects module
- 3 position cue select Pre CF Cue / Post CF Cue / Master
- 2 line, 2 phono, and 1 mic/line (switchable) inputs
- 3-band EQ, Gain, and slider-style pan control per channel
- Crossfader and Program faders reverse switches & curve adjustments
- Program reverse
- "Soft/Quiet-Start" feature to avoid pops / noise when mixer is powered on
- 2 headphone inputs (1/4″ and 1/8″ minijack) with headphone mute
- TRS balanced Master output with output trim control
- Mono-stereo switch
- Dimensions: 14.9 x 10.5 x 4.3″, 8 lbs.

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ONE HOUR FREE PARKING
AT 349 W. 34th STREET (with purchase of $100 or more)

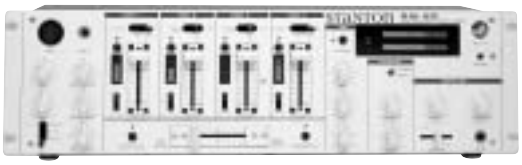
RM-3S 3-Channel DJ Mixer



Based on the popular RM-3 the RM-3S keeps most of the original layout and design but with improved features and better control. A redesigned fader allows a smoother more gradual curve, and the added Cue Pan fader lets you monitoryour mix through the headphones.

- 6 line, 3 phono, and 2 mic inputs
- 60mm channel faders
- Assignable AUX send/return
- Gain, 3 band EQ (+9/-26dB), and Pan control on each channel
- High quality professional crossfader with reverse function
- Mic channel features volume controls and 3 band EQ (+12/-12 dB)
- Cue section features Pre Fader Listen function with volume control and Cue-Pan fader
- 3 band frequency isolators for channels 1 and 3 featuring three-way toggle switches with FLASH function
- Rec output
- User replaceable faders

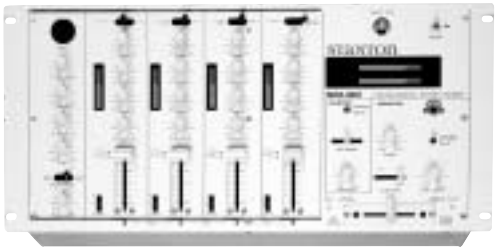
RM-50
4-channel Stereo Preamp Mixer



Designed for small clubs and mobile DJ applications where space and budget are limited. But don't let it's compact three 19˝ rack-space size fool you, the RM-50 is packed with features.

- 9 line and 4 phono inputs, 4 mic inputs (combo and 1/4˝ connectors)
- 3 band master EQ with complete kill (+9/-60 dB)
- Fader start
- 2 position toggle switch on channels 2 and 3 for easy transforming
- Mic channel features 2 band EQ, independent level controls, and VCA controlled autotalk function
- Fully assignable crossfader (A, B, bypass)
- Balanced XLR master output, Booth output
- Master section with level and balance controls and mono/stereo switch
- Cue section features cue pan and level control
- Input and output LED meters
- 12V light BNC input

RM-80
4-channel Stereo Preamp Mixer



- 9 line and 4 phono inputs
- Mic channel features 2 band EQ, independent level controls, ON/OFF switch, and VCA controlled autotalk function
- 3 mic inputs (combo and 1/4˝ connectors)
- Gain and 3 band EQ (+9/-26 dB) per channel
- Fader start
- Sub-mic on channel 4 enables use of 3 band EQ and panning
- Fully assignable crossfader (A, B, bypass)
- Beat indicators
- Balanced XLR master output, Booth output
- Master section with level and balance controls and mono/stereo switch
- Cue section features, stereo/split function, cue pan and level control
- Input and output LED meters
- 12V light BNC input

RM-100
4-channel Stereo Preamp Mixer



A step-up from the RM-80, the RM-100 adds an effects loop which lets you easily add effects or sampling to your mix. It has an Effects send and return assignable to master out, each channel (including all mics), and headphones for monitoring. Cue section features stereo/split function, cue pan and level control. Cue pan fader is assignable to booth output for in-booth monitoring. Sub mic on channel 4 enables use of 3-band EQ and panning.

VRM-10

DJ Dream Mixer

The VRM-10 is like no other DJ mixer in the world. From it's valve preamp and built-in filter to its high quality faders, this mixer is any DJ's dream come true.



- The VRM-10 features 8 line and 4 phono inputs on 4 input channels. Each input channel can be assigned to either side of the crossfader, or the crossfader can be bypassed altogether. A crossfader curve adjustment is provided to allow the user to gradually change from a sharp curve for scratching, to a long fade for mixing. 2 mic inputs are provided on a single channel. Each mic has its own gain control and share a 3 band EQ.
- Wide selection of outputs to choose from. Master 1 (balanced XLR) and Master 2 (unbalanced RCA). Each Master output features its own level control. Master 1 also features a balance control. There are 2 headphone outputs as well as a booth output. Each channel features a Pre Fader Listen button to assign each channel to the headphone output independently. If none of the PFL buttons are pressed, the master output will automatically be assigned. A S/PDIF digital output is also provided.
- Each of the VRM-10's channels feature a gain control and an extremely versatile 3 band parametric EQ with midrange frequency adjustment
- There are 16 on board DSP effects including 2 hall reverbs, 3 room reverbs, 2 plate reverbs, and a spring simulator; 4 stereo delays; 4 chorus / flanger effects. There are 5 parameter controls including level, feedback, speed, factor, and depth. The speed controls the frequency of the LFO (0.1-20Hz), depending on the depth setting. The on board effects can be assigned to each channel or the master output.

Titanium Pak

Titanium Pak is the top of the line DJ Package for the professional DJ looking to upgrade his or her system.



Includes:

- [2] STR8-100 turntables
- [1] SK-6F two channel mixer
- [2] 520SK cartridges
- [1] DJ PRO 60 stereo headphones
- [2] Slipmats

** PLUS a free copy of TMW2002 [Turntable Mechanic Workshop 2002]*

Skratch Pak II



Includes:

- A pair of STR8-60 straight arm direct drive turntables premounted with 520SK cartridges
- A SMX-201 professional turntablist mixer featuring 2 band EQ and Gain control per channel
- a pair of DJ PRO 60 stereo headphones
- 2 Slipmats
- 2 Dust Covers
- All connecting cables

**plus a free copy of the TMW2002 [Turntable Mechanic Workshop 2002]*

Platinum Pak II

Recommended by DJ Craze



Includes:

- [2] STR8-80 turntables
- [1] SK-6F two channel mixer
- [2] 520SK DJ Craze model cartridges
- [1] DJ PRO 60 stereo headphones
- [2] Slipmats
- [2] Dust Covers

**PLUS a free copy of TMW2002 [Turntable Mechanic Workshop 2002]*

Spin Cycle Pak



Includes:

- [2] STR8-30 turntables
- [1] SMX-201 two channel mixer
- [2] 500b cartridges. Already mounted on Stanton Headshells, the 500b provides the perfect balance of reliability, ruggedness and optimum performance.
- [1] DJ PRO 10 stereo headphones - with uniques earcup and headband
- [2] Slipmats

B&H PHOTO - VIDEO - PRO AUDIO

HEADPHONES

DJ Pro 2000 S

- High-quality stereo headphone designed for the professional DJ
- Rotating ear-cup allows for flexible positioning and single sided or shoulder rest wearing styles
- Detachable 10′ coiled cord for safe storage and easy replacement
- 90º (elbow) 1/4″ connector
- Integrated mono or stereo monitoring
- Wide frequency response 20 Hz-30 KHz for outstanding sound reproduction
- Closed ear-cup design for sound isolation
- 50mm drivers for clear low end reproduction and superior power handling
- Convenient folding design for ease of storage and transportation
- 3.5mm mini-jack adaptor and carrying pouch included







DJ Pro 1000 MKII S

- Enjoy the combined benefits of light weight comfort and accurate wide-frequency sound reproduction
- 50mm high efficiency, heavy duty drivers deliver full deep bass
- Straight or coiled removable cord versions aid in field replacement
- 20-20kHz frequency response
- Detachable 10′ cord with 1/4″ adapter
- Includes a free carrying pouch

DJ Pro 500 MC

- Single side, closed back design with a heavy duty driver with bass boost
- For the professional DJ who needs hands-free operation with a mic
- Adjustable metal headband is protected with a soft vinyl covered foam pad
- Color-coded dual connection 12′ cable has two 1/4″ plugs - mono for the mic and stereo for the headphone
- Low impedance 200Ω dynamic mic
- Includes a free carrying pouch

DJ Pro 300

- Lightweight single-sided headphone
- High efficiency 40mm heavy-duty drivers
- Detachable 10′ cord with 1/4″ adapter
- Frequency response 20-20kHz
- Sensitivity 100dB @ 1mW
- Includes a free carrying pouch

DJ Pro 60

- Compact, lightweight stereo headphone with a comfortable self-adjusting head-band and exceptional low end response
- Transducers 40mm drivers
- 3Hz-28kHz frequency range
- Sensitivity 102 dB
- Impedance 50Ω
- 8.5′ cable length
- 80mW maximum power input
- 1/4″ stereo plug with 3.5mm mini jack adapter
- Weight (without cord) .4 lb.
- Includes a free carrying pouch

DJ Pro 50

- High quality stereo headphones at an entry level price
- 40mm super thin driver with Neodymium magnet
- 20 Hz - 20 KHzfrequency response
- Sensitivity: 106dB SPL at 1KHz
- 9′ cord with 1/4″ adaptor

TASCAM

CD-302

DJ EQUIPMENT

1566

Professional Dual CD Player



The CD-302 isn't for every DJ...just the ones that want to stand out from the crowd. The reason is simple, feature for feature, the CD-302 is the most powerful, creativity-inspiring dual CD DJ tool available today. Whether you're into juggling, scratching or beat mixing, the CD-302 isready to handle your best moves. It's like having two separate decks, each with its own controller.

You get features like beat matching that automatically shows the tempos of the discs in both decks, 10-second sampling on each unit, and the ability to seamlessly loop material as long as your audience can take it.

The brake feature lets you make the smoothest segues between songs, and the instant start feature lets you cue your material before anyone can leave the dance floor. Go crazy with ± 32% tempo adjustments that won't change the pitch, or ± 32% pitch adjustments that don't change the tempo. In other words, the CD-302 lets you make the music all your own.

FEATURES

- Two high-quality precision CD decks in a single rack-mountable unit. The CD players are independent—one CD can be cued up while the second is playing.
- Incorporates a built-in sampler that can loop a section of audio in either Play or Cue mode. Samples can also be reversed.
- Pitch and tempo can be changed independently and adjusted over a ±32% range.
- A unique feature of the CD-302 is the ability to analyze both the tempo and downbeats of two different CDs and then adjust the pitch of the cued CD to match the tempo and downbeat of the playing CD without affecting the pitch. This allows the DJ to perform flawless crossfades from one CD to another during performances."
- A built-in RAM buffer allows for a 10-second shock buffer and instant start
- Great sounding analog-style scratching with the large JOG wheel on the remote.
- 100 mm tempo slider
- Preset and cue memory locations may be stored and recalled using a 10-key pad.
- The CD-302 is housed in a 2U, 19″ rackmount enclosure. The remote is housed in a 3U, 19″ rackmount enclosure which can also be mounted flat in a DJ coffin case.
- CD Control Module includes enough RAM for unlimited looping and 10 seconds of sampling. The CD Control Module also includes Auto Cue, Sleep Function, Tap Tempo Button, Pitch Slider and intuitive Pitch Bend using the JOG Wheel.
- Sample up to 10 seconds of audio and play it back forward or reversed. The sampled sound can be played back when the source CD is still in Play or Loop mode.
- Dedicated per-deck remote control units (detachable and rack-mountable)
- Key control function allows changing key without affecting the original tempo (in percentage or semitone steps)
- Automatic BPM (beats per minute) count function and BPM counter. TAP function for manual BPM settings
- Original Key function allows tempo change while maintaining the original key
- Bend function allows adjusting of beats manually between the two decks
- Brake function allows braking in exactly the same way as on a record turntable
- Scratch function allows scratch play
- Voice Reduce function lowers the level of the vocal part
- Up to 1,000 track memories (memorized in 5 banksof 200 tracks each, Bank A to E)
- Output level control in the digital domain (±6 dB, in 1.0 dB steps).
- Elapsed and song remain time display
- Auto cue function (-72, -66, -60, -54, -35 dB)
- Auto ready function to cue up following tracks
- EOM function (10/15/20/30/60, or 90 sec)
- Timer play for deck 1 followed by deck 2.
- Relay play allows consecutive playback using both decks (as one deck stops, the other can be set to start automatically).
- RCA unbalanced outputs for each deck

X-15 / X-17

Professional Mobile/Club DJ Mixers

Designed for professional mobile DJs, the X-15 and X-17 mixers are innovative DJ tools that combine powerful features, excellent sound quality, great reliability and high value. While both the X-15 and X-17 are four-channel rackmount DJ mixers created specifically for mobile DJs, the X-17 provides additional advanced features making it excellent for club DJing as well.

The X-15 offers eight stereo inputs (2 phono, 6 line), while the more flexible X-17 provides four switchable phono/line inputs as well as four dedicated line inputs. Both mixers have two mic inputs, balanced XLR and unbalanced RCA master outputs and 3-band EQ on each channel. You'll also get a set of pro features like fader start on all four channels, dedicated booth and record outputs and a replaceable crossfader. Plus, the X-15 and X-17 are intended to easily fit into a compact mobile DJ system, with standard 19˝ 4U rackmount enclosures.



X-15

X-17



In addition to the features shared with the X-15, the X-17 adds a three-bank, pitch-controllable sampler that stores and recalls sample files with an innovative onboard front-panel Compact Flash card slot. Additionally, the X-17 offers other professional features like front-panel aux input and output jacks with individual level controls, crossfader reverse controls, an effects send/return interface and more. Also, the X-17's flexible design allows for its four channels to source from four switchable phono/line inputs as well as four dedicated line inputs, making it an excellent choice for both mobile DJing as well as club DJ work.

X-15 Features:

- Four-channel 19˝ rackmount mobile DJ mixers
- Balanced XLR and unbalanced RCA master outputs
- Eight stereo channel sources (two stereo phono inputs, six line inputs)
- 3-band EQ per channel
- Fader start on all four channels
- Discrete booth outputs and record outputs
- Two mic inputs
- Replaceable crossfader

X-17 Step-up Features:

- Eight stereo channel sources (four switchable phono/line stereo inputs and four dedicated stereo line inputs)
- 3-bank sampler with pitch control and compact flash slot to save and recall samples
- Separate stereo aux input w/ level control
- Discrete aux outputs, booth outputs and record outputs
- Booth dimmer
- Effects send/return

TASCAM

XS-3 / XS-4 / XS-8

DJ EQUIPMENT

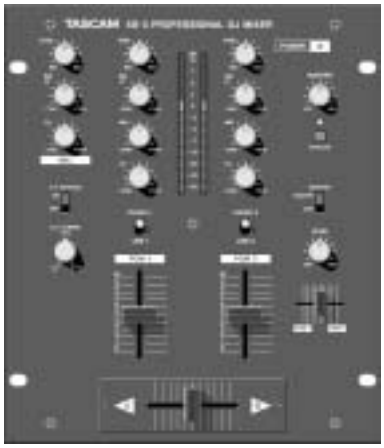
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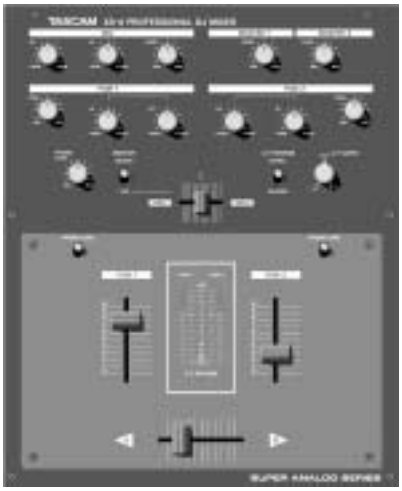
Scratch DJ Mixers

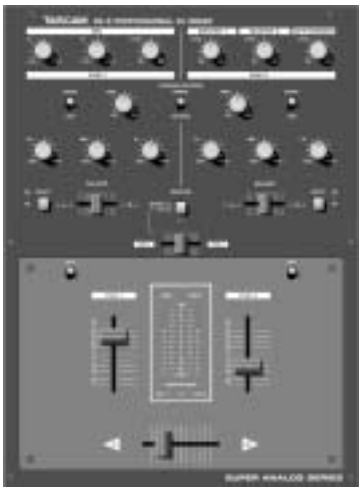
The Tascam XS-3, XS-4 and XS-8 offer affordable mixing solutions for performance-oriented scratch DJs.The XS-3 is an affordable two-channel mixer designed for a wide variety of DJ environments. It offers three-band EQ, a host of inputs and outputs including line and phono inputs on each each channel, balanced and unbalanced master outputs, a recording output, an effects input, a mic input, crossfader start capabilities, adjustable crossfader curve selection and a crossfader reverse control.

The XS-4 is designed as a 2-channel battle mixer with an uncluttered, logical control surface. It offers a 2-band EQ and more sophisticated audio interfacing than the XS-3, including master outputs on balanced XLR jacks. The XS-4 offers all the features and ergonomic design desired for serious scratch DJing with the sound quality and reliability for which Tascam is well known.

The XS-8 is the top of Tascam's scratch DJ mixer line. It is a 2-channel scratch mixer that features a fully adjustable crossfader and channel fader curves, crossfader and channel fader reverse controls, a 3-band EQ, interfacing for external effects sends and returns, discrete level controls for stereo outputs, dedicated EQ controls for the mic input and more.







XS-3 Features

- Two-channel multi purpose DJ mixer
- Four stereo inputs (two phono, two line)
- Balanced 1/4″ TRS and unbalanced RCA master outputs
- 3-band EQ per channel
- Crossfader start and crossfader reverse with adjustable crossfader curve
- External effects send/return, Mic input, Record output
- Replaceable crossfader

XS-4 Features

- Two-channel "Super Analog Series" scratch DJ mixer
- Channels source four stereo inputs (two phono, two line)
- Separate external stereo line input
- Balanced XLR and unbalanced RCA master outputs with separate level controls
- 2-band EQ per channel
- Crossfader reverse with adjustable crossfader curve
- Mic input, Record output
- Replaceable channel faders and crossfader

XS-8 Features:

- Advanced two-channel "Super Analog Series" scratch DJ mixer
- Four stereo inputs (two phono, two line)
- Balanced XLR and unbalanced RCA master outputs with separate level controls
- 3-band EQ - channel balance controls
- Crossfader reverse with adjustable crossfader and channel fader curves
- Program reverse - Channel cut switches
- External FX send/return - footswitch inputs
- Session input, Mic input, Separate external stereo input, Record output
- Replaceable channel faders and crossfader

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X-9

Professional Performance DJ Mixer



The X-9 is a tool for serious DJs that combines a highly flexible 4-channel mixer with a dual effects processor and a dual sampler. The result is a creative powerhouse that helps to add originality and inspiration for every mix.

Designed specifically for high-end creative DJs, the X-9s control surface is laid out to combine ultra-high performance and intuitive ease of use. Its multitude of analog and digital inputs allows for incredible flexibility for interfacing with a wide variety of DJ gear such as vinyl turntables, dual CD decks, external effects devices and other audio tools.

To achieve the ultimate in creative control of your mixes, the X-9 includes a wide variety of high-quality built-in effects. You can add reverb, delay, echo, flanging, auto panning, pitch control, a transformer effect and a low pass filter to any audio signal that passes through the X-9, with 30 spaces to store your settings for later use.

The X-9 also features parametric EQ an essential control feature for exciting custom mixes. All EQ settings are also storable and recallable. Also, using the X-9s sampling function, you can select a sampling source from Input 1 through 4, from the master output, the microphone input, or both sides of the crossfader. Along with its customizable fader start function, which can be assigned to several controllers on the mixer, the X-9 offers all the features you ever wanted in a DJ mixerand a few you never knew you needed until now. Whether youre combining it with TASCAMs CD-302 Dual CD Decks or any other audio playback tool, the X-9 represents a giant leap forward in setting the standard for DJ mixing in the 21st century.

FEATURES

- Mixing and audio manipulation tool for professional DJs
- 4-channel stereo inputs on line and phono inputs
- Dual sampler
- 2 coaxial digital inputs (channels 1 and 2), 2 external line inputs (channels 3 and 4)
- 2 built-in digital effects processors with reverb, delay, echo, flanger, auto pan, pitch control, transformer and low pass filter; 30 storable effects settings
- Fully-parametric 3-band EQ on each input channel with 100 storable patterns
- 3-stage adjustable input fader curve, Input fader reverse, Cross fader reverse, Cross fader curve control
- Microphone input with 2-band EQ
- Low/Mid/High Cut control for both cross fader sides
- Coaxial digital Master output with pre/post master selection in the menu
- Booth output, analog balanced Master output with attenuator, analog unbalanced Master output with attenuator and pre/post master fader switch
- Headphone outputs on front and top panels, with 3-stage headphone EQ
- 4 fader start/stop connectors
- Optional MIDI In/Out/Thru connectors
- External stereo effect send/return connectors
- 2 assignable foot switches connectors

	XS-3	XS-4	XS-8	X-9
Channels	2-channel	2-channel	2-channel	4-channel
Stereo Inputs	4 (2 phono, 2 line)	4 (2 phono, 2 line)	4 (2 phono, 2 line)	4-channel stereo inputs on line and phono inputs
Outputs	Balanced 1/4″ TRS and	Balanced XLR and unbalanced RCA master outputs with separate level controls		Booth output, analog balanced Master output with attenuator, analog unbalanced Master output with attenuator and pre/post master fader switch
EQ	3-band EQ per channel	2-band EQ per channel	3-band EQ per channel	3-band EQ per channel

TURNTABLES

PDX-2000

Super High Torque DJ Turntable with Anti Skipping Tonearm

A totally new turntable designed strictly for professional DJs. The silver body design uses super high-density molded ABS plastic which diminishes vibrations. Its high-torque quick-response DC motor, heavy-duty Anti Skipping Tonearm System (A.S.T.S.) and great versatility through various braking, start and reverse features make this a perfect turntable for any DJ. Even the hardest scratching won't fault the powerful PDX-2000. With two pitch controls it is possible to adjust the pitch by up to ±60%. Achieve the most extreme effects and sounds — and ensure the best possible performance every time with the PDX-2000.



PDX-2000 shown with optional $75 DSC-2000 dust cover

- ASTS Floating-Suspension Tonearm system coupled with Tracking Hold reduces needle skip and jump
- Speed Adjustment allows manipulation the start and stop speed independantly. Instant start/stop or slow wind-up/wind down gives DJs the ability to be creative
- High-torque direct-drive DC motor enables a start speed of 0.5 seconds to reach normal RPM
- Power switch for "Motor off-on" effects
- Ultra Pitch fader controls the pitch range by ±50% and then, combined with Fine Pitch fader, control of ±60% is possible
- Non skipping plus feedback prevention
- Special 100mm fine pitch control fader
- Super direct drive 33/45 rpm
- Adjustable start and break time
- Reverse button makes the platter spin backward at one easy touch

PDX-d3s

Digitally Controlled Turntable

The PDX-d3s offers advanced mixing performance. The reverse switch, selectable pitch range, motor off switch and joystick for pitch bend control give DJ's unlimited new techniques.



- Synchronize feature allows you to control more than 2 turntables with a single operation (optional cable required)
- The stick controller gives instant speed adjustment of ±50% and pitch bend of ± 1, 2, 3 or 6% (selectable). There's no need to touch the turntable for cue point or pitch bend anymore
- Reverse mode feature works for all features such as pitch bend, speed control, etc.
- Selectable range of pitch control allows wider or finer pitch control with 3,6, and 12% of range

BDT-2500

DJ Turntable with Line Out



A high-quality phono amplifier allows the BDT-2500 to connect directly to a line input jack of a mixer/pre-amplifier without the grounding cable usually required with turntables.

- Includes an internal phono amp bypass switch for use when connection to phono input is preferred or required
- Two pitch controls: Main Pitch and Fine Pitch, allow you to set the speed from 16 rpm to 98 prm with no gaps
- Static-balanced "S" shape tone arm system maximizes its traceability and durability
- Unique heavy-duty metal chassis minimizes the feedback and vibration

(Please note: The BDT 2500 is not designed for scratch play)

PREMIUM SERIES MIXERS

R-1 Premium

19″ Rack Mountable Mixer



The R-1 is the superior sonic solution for professional house DJs, installations and studios. The R-1, with its rotary style dials, is created using only the highest quality components. Just wait till you hear its superb sound quality - it's one of the most incredible sounding mixers ever produced! Designed with advice from the world's top club DJs, the R-1 features a built-in 4-band isolator and a 3-band booth EQ, helping you boost your creativity.

- 5 channel mixer features 5 phono and 5 line inputs
- The ALPS custom-made volume pot generates perfect torque and great durability
- 4-way Equalizer/ Isolator - allows DJs to totally separate sounds with the use of the Vestax "Infinity Cut"
- Dedicated 3-band EQ provided for Sub master output enables perfect monitoring in the DJ booth
- Hand-made line transformer - superb quality, generating a warm, thick sound
- Insert type effect loop - for easy connection of external EQs and other effects on individual programs
- Signal/Peak LED on each channel

R-1 SPECIFICATIONS
Frequency Response: 20Hz - 20kHz ±1dB
S/N Ratio: >75dB
THD: <0.005%
Dimensions: 19 x 7 x 5.2″ (WxDxH)

- Special selected rotary input fader
- *Inputs:*
  5-**Phono** (RCA)
  5-**Line** (RCA),
  5-**Effect Return** (1/4″)
- *Outputs:*
  1-**PA Master Out** (XLR balanced),
  1-**Line Out** (1/4″ unbalanced),
  1-**Sub Master Out** (1/4″ unbalanced),
  1-**Record Out** (RCA),
  5-**Effects Send** (1/4″),
  1-**Monitor Headphone Out** (1/4″)

MCR-1

Crossfader Unit for the R-1 Premium



The MCR-1 is a Crossfader Unit for the Vestax R-1 Premium which enables you to assign 5 PGM channels into the crossfader. The MCR-1 uses only the highest audio quality components, so you can add this unit without reducing the R-1's sound quality. The MCR-1 includes an original superior grade PCV crossfader developed exclusively by the Vestax engineering team to provide super long life.

- Cross Fader Assign (A/B) Switch is used to assign the signal from each of the PGM channels to either side of the crossfader
- The detachable cross fader mixes the signals assigned by Cross Fader Assign Switch
- The Monitor Select Switch is used to select the cue signal to be monitored by headphone from PGM channels 1-5 or MASTER
- H.P. LEVEL (Monitor Level) controls the volume level of the headphones
- Headphone Jack is used to connect the headphones with impedance from 8-150 ohm
- 1/4″ unbalanced line output
- IN / SEND (1/4″ unbalanced); OUT / RTN (1/4″ unbalanced)

VESTAX

PROFESSIONAL SERIES MIXERS

PMC-46MKII

19˝ Rack Mount
5-Channel Rotary Style Mixer



The PMC-46MKII was designed with advice from professional DJs in New York. The "curve" of the rotary dial is the most important feature for professional House DJs, and the Vestax PMC-46MKII offers the best House "curve" in the world.

- 4/6 channel mixing instrument with 4 phono and 8 line inputs.
- Advanced mixing facilities through Aux send – return and Submaster
- VCA controlled Crossfader
- Special cue-signal monitoring through new option meter.
- Special selected rotary input fader
- Extremely variable mixing facilities
- Replaceable VCA crossfader with slope
- LED beat indicator
- 1 AUX send and 1 Effects loop
- 2-band EQ per channel
- Input/output level meter
- Balanced XLR and 1/4˝ outputs
- Champagne gold color



PMC-400
Similar to the PMC-46MKII, but with slide faders instead of rotary faders.....**Call**

PMC-250

Ultra Slim 19˝ Rack Mount House/Mobile DJ Mixer



Ideal for 19˝ rackmount requirements, the PMC-250 is only 3RU high. It features a rotatable connector panel for either table top or installation purposes, and has balanced XLR outputs with a -10dB attenuation switch. The mixer is primarily a two channel but has the facility for a third auxilary input, switchable between phono and line input levels. The main two channels feature rotary three band fixed EQ controls with -85 and +4dB cut and boost. Short 5mm faders above these controls offer the "kill" effect without the occasional clicks that can occur with switches. The mixer uses very high quality components and has features to please DJs and sound engineers alike. Input faders and the crossfader are user replaceable

- 3-band frequency isolators are provided on both main music channels, allowing you to boost, cut or fade frequency ranges with maximum control for truly active mixing



Rear view of the PMC-250

- A –10/0db output level switch is provided for professional installation use
- Fader-type cut switch enables smooth and dynamic frequency cut-off without the annoying "clicks" found on regular "kill switches"
- Optional rotary faders can be fitted in place of the standard 45mm faders on the input channels for that "New York" style rotary effect in the mix
- Balanced XLR and 1⁄4˝ jacks are provided for easy connection
- The location of the rear connector panel can be selected from either the rear or on the base of the mixer. This allows the PMC-250 to be used on a tabletop or securely fixed to a vertical rack without wasting rack space on cable connectors
- An Aux (phono/line switchable) input is provided for third turntable, CD, MD, etc.

PROFESSIONAL SERIES MIXERS

PMC-55

Professional 4-Channel Mixer

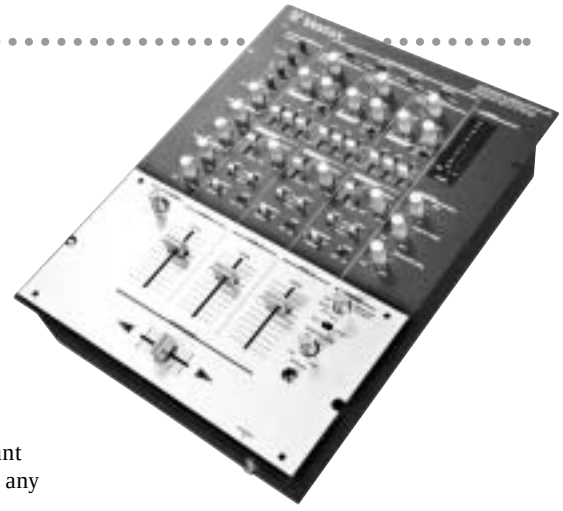


The PMC 55 is the mixer of choice of many top DJ's and clubs due to it's flexibility and high quality components. Pro users of this model include Ritchie Hawtin, Carl Cox and Alex Patterson of The Orb. The PMC 55 is a 12 input, 4 channel mixer with a stereo FX loop, dual balanced XLR outputs, booth output with assignable source, VCA crossfader with curve control and user-replaceable input faders with a rotary "dial fader" option. Each channel features individual gain, metering, pan, 3-band EQ with bypass switch, FX send and insert point. The maximum gain and EQ boost of each channel can also be limited by internal switches, preventing any possible distortion to the master output stage of the mixer. The master output section has a -10dB and -20dB attenuation switch to prevent any further increase in output gain before the amplifier, making the PMC 55 the obvious choice for large nightclubs and sound engineers alike.

- Powerful 4/7 channel mixer with 4 phono and 8 line inputs
- Various effect processing facilities on each channel
- Special programmable protect functions for save PA operations
- Faders use a replaceable 60mm input fader and a 60mm crossfader
- Line inputs and effects connection on front panel for "visiting" players and mobile effects
- An LED indicator has been provided for each channel's CUE switch
- The booth output can be selected from CUE, Master 1 or Master 2 for flexible monitoring
- The AUX (FX) loop can be selected from either pre or post fader for different effect possibilities
- The 3-band EQ has a boost/cut range of -24/+6dB. This enables greater cut of each bandwidth, and less possibilities of distortion through EQ boost
- Champagne gold color

PMC-37 Pro

High-Quality Multi-Genre Mixer



This incredible mixer offers the matrix input assign system, which allows instant selection of multiple sound sources for any input channel. With isolators on each channel, built-in sweep effects, PCV input faders and PCV crossfader, the PMC-37 Pro is in high demand worldwide. Incredibly, one turntable can be assigned to every program channel.

- High-quality VCA circuitry provided optimum S/N, warm fat sound and broadcast-quality audio
- 3/5 channels (3 stereo, 2 mono) with 4 phono and 4 line inputs
- 2 Mic inputs with 2-band EQ and AUX send/return
- 2 effect loops with pre/post switches on each channel
- Matrix input assign system allows instant selection of multiple sound sources for any input channel
- Unique program assign switches for each stereo channel
- Dynamic dual peak/sweep filter on each program extend creative possibilities
- 3-band isolator fader with function on each stereo channel
- 4-position Cue Monitor selector dial allows you to switch between Cue PGM, Master, AUX1 and AUX 2
- Special cue-signal monitoring through option meter
- User replaceable input fader and crossfader

SAMURAI SERIES MIXERS

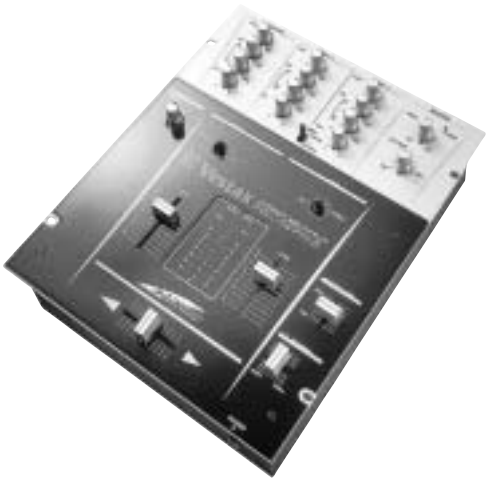
Featuring the world's first super-smooth, no-maintenance digital crossfader, the Samurai series of mixers offer exclusive circuitry that gives you quick and easy control over the shape of the crossfader curve. The selectable base curves gives the DJ access to 8 crossfader curves on each Samurai selection. And best of all, due to the magnetic field sensor of the crossfader, Samurai series mixers are liquid-resistant and 100% maintenance free.

PMC-07ProD



The Battle Mixer we all love – the PMC-07Pro – joins the Samurai range as the PMC-07ProD with the slashing sound of Vestax digital crossfader technology. The PMC-07ProD features a Session Mix function that allows team DJs to connect multiple mixers simultaneously. The Master output is TRS balanced for automatic connection to balanced PA systems. 3-Band Slider EQ faders on each channel with +6dB/-24dB

- Master output is TRS balanced for automatic connection to balanced PA systems
- 3-band slider EQ faders on each channel with +6dB/-24dB
- Crossfader Curve selection and shaping
- Crossfader is resistant to liquid spills
- Deep burgundy and silver finish
- Input/Output: 2-Phono (RCA), 3-Line (RCA), 2-Mic (Phone), 1-Master (RCA, TRS)



PMC-05ProD

As the standard for hip-hop scratch mixing, the PMC-05ProII has been loved by so many hip-hop artists around the world. There have been many pretenders, but the PMC-05Pro is still king. Now, due to the addition of the new Vestax digital crossfader with built-in curve control, the PMC-05ProD Samurai blasts off into a new Solar System, where only the true pioneers of today's sound will go.

- 2/4 channel mixer with 2 phono and 2 line inputs
- Digital crossfader with user designable curves
- The standard for hip-hop/scratch mixing
- Exclusive double panel system with 576 different settings of fader and switch modes
- Switchable input fader curve
- User replaceable input fader
- Monitor crossfader for PFL mixing
- Studio type PCP layout

PMC-06ProD



The incredibly popular PMC-06Pro has been updated with the addition of a digital crossfader to create the PMC-06ProD. This scratch mixer fits between two turntables with minimum distance in between. All faders and switches are laid out for perfect one-hand operation.

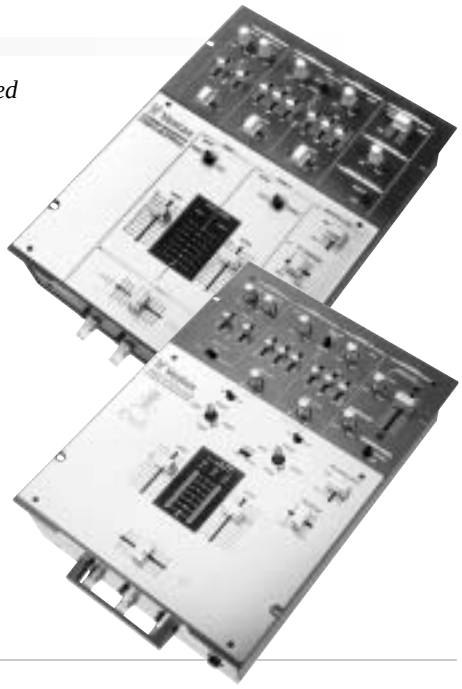
- Top line hip-hop mixer with "Hamster" switch
- Digital crossfader with up to 32 crossfader curves
- 2 channel mixer with 2 phono and 2 line inputs
- Exclusive double panel system with 576 different settings of fader and switch modes
- Switchable input fader curve
- Super slim mixer fits between turntables
- User replaceable Input fader and cross fader
- Monitor crossfader for PFL mixing
- Studio type PCB layout
- Deluxe front panel design

PCV / PROFESSIONAL SERIES

PMC-07Pro

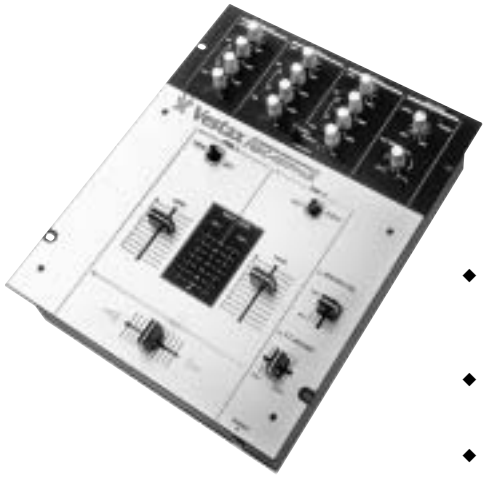
Extended EQ, session mix, mic input with 2-band EQ, input fader curve and many other extended features make the PMC-07Pro the choice for aggressive creative music makers.

- 2/4 channel mixer with 2 phono and 2 line inputs
- Crossfader reverse switch and curve control knob are provided not only on the crossfader, but also on the input faders
- Double panel design with hidden features
- 3-band equalizer, program and crossfader reverse switches for top performance
- Curve adjustments and reverse functions for input and crossfaders
- 8-position input selector switch
- User-replaceable input and cross fader
- Monitor crossfader for PFL mixing
- Studio-type PCB layout



PMC-07Pro I.S.P.

Developed by the world-famous Invisibl Skratch Piklz, the PMC-07Pro I.S.P. inherits all the advanced features from the PMC-07Pro flagship peformance mixer with the addition of quadraphonic (4-channel) output. This allows DJs to expand their performance to a 4-dimensional level with the cue stick control. The beats and scratches can fly around the room, from front to back and left to right.



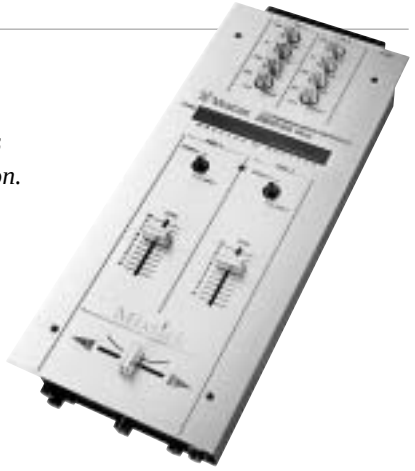
PMC-05ProII

The successor of the world-standard professional battle mixer, the PMC-05Pro II is used by many of the worlds professional battle and hip-hop DJs. Its photo coupler cross fader system with curve control, double panel system and cross fader reverse switch make it the one and only professional battle mixer you will need for top performance.

- 2/4 channel mixer with 2 phono, 2 line and 2 Mic inputs which are equipped with a 2 band equalizer. (Hi and Lo +/- 12dB
- A cross fader reverse switch is provided to enable quick crab scratch technique
- Eight position input selector switch
- Photo Coupler operation of the crossfader allows crossfader Curve adjustment
- Double Panel design with hidden features
- Switchable input fader curve
- User-replaceable input and cross fader
- Monitor crossfader for PFL mixing

PMC-06ProA

The PMC-06ProA professional DJ scratch mixer has a "shaped-up" style to fit between two turntables with minimum distance in between. All faders and switches are laid out for easy one-handed operation. A double panel system and photo coupler cross fader system are also featured for professional use. Photo coupler controlled cross fader with curve adjustment.



- Super slim mixer fits between turntables
- 2 channel mixer with 2 phono and 2 line inputs
- Double panel design with 576 different settings of fader and switch modes
- Switchable input fader curve
- User-replaceable input fader and cross fader
- Monitor crossfader for PFL mixing
- Studio type PCP layout
- "Hamster" switch (reverses the direction of cross fader)

VESTAX

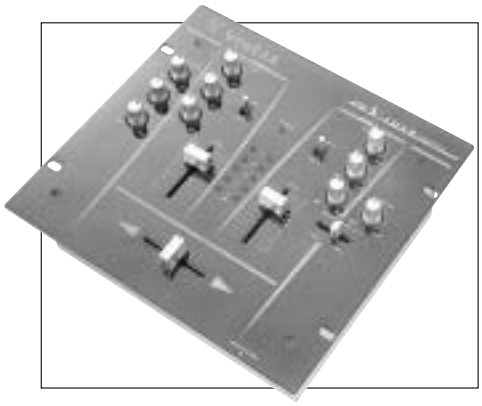
PCV SERIES MIXERS

DJ EQUIPMENT

1576

PCV-002U *Entry-Level Mixer*

Compact body size and extended features makes this unit suitable for scratch, mobile and hobby use.



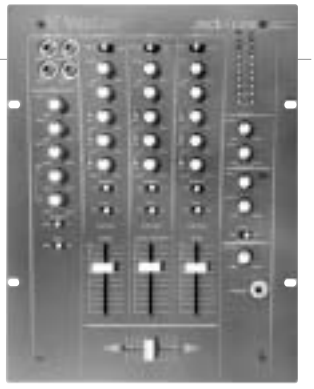
- Multi use facilities: Mobile, bedroom and hobby
- 3 band full Isolator with the renowned Vestax sound on channel 1+2
- portable and rugged metal chassis with ironblue frontpanel
- 3 channel mixer; 2 phono + 2 line inputs and microphone input on rear panel
- 3- band full Isolator for hi-mid-low frequencies on channel 1+2
- Special crossfader for monitoring channel 1 +2
- Vestax double panel design with user upgradeable PCV-crossfader



PCV-150 *19˝ Rack Mountable Mixer*

An all-genre mixer for DJs who require multi-mixing features. Featuring a PCV crossfader, it is the correct choice for professional quality and long-term durability - at a sensible price. Striking metal blue panel design.

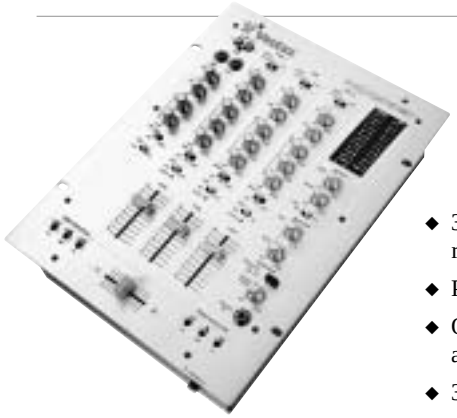
- Powerful 4/7 channel mixer with 3 phono and 5 line inputs
- 3-band "Super Cut" equalizer on each program channel
- Various professional connection possibilities through switchable XLR plus 3 more outputs
- Transformer and assign buttons for ultimate scratching
- Special split master cue system for every application
- PCV-crossfader; metal blue panel design



PCV-175 *Professional Techno/Trans Mixer*

This flexible, popular mixer is highly suitable for techno, trans and hardcore mixing. The European styling of this mixer, coupled with its flat panel faceplate was designed with speed mixing in mind. It features a 45mm PCV crossfader and 60mm vertical faders.

- 3/6 channel mixer with 3 phono, 5 line and 4 mic. inputs
- 3-band equalizer with cut switches
- Pre fade monitoring on each channel
- 10-segment stereo LED level meter
- Effect switches for each channel
- Replaceable PCV faders and crossfader
- Metal blue front panel



PCV-275 *Professional Mixing Controller*

This techno/trance mixer offers many features, such as PCV fader technology, 3-band isolator switches, selectable AUX send and option cue meter.

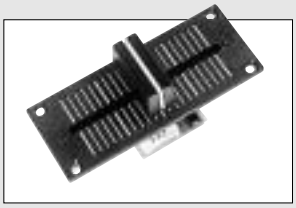
- 3-band isolator switch eliminates each frequency range completely
- Pre/post selectable AUX send switch
- Option meter allows the DJ to visually and audibly check the levels and accuracy of a "mix"
- 3 phono and 6 line inputs
- User replaceable 60mm input faders and 45mm crossfader make "on-site" repairs very easy
- Accepts optional rotary input volume controls
- Line 4 input connector socket is situated on the top panel for the easy addition of a MD/DAT /tape /CD input during performance

B&H PHOTO - VIDEO - PRO AUDIO · HSH · DJCOVER NOVUS · AMERICAN EXPRESS Card

FADERS



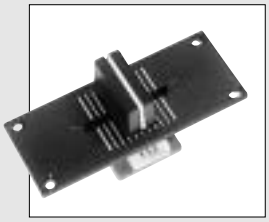
VCC-3W Wood Cartridge Case
Holds 3 stylus with headshellls

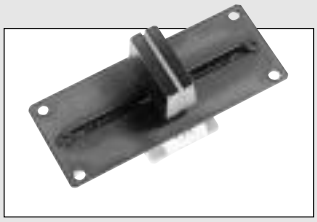


CFR-US
Replacement Crossfader
45mm crossfader for PMC-20SL, 09, 005, 05MK3, 05T, 20, 26, 17, 15SL, 15, 05FX, 17A, 27MK2, 15MK2, 06T, 005AMK2, MW-3000R, 3000, StreetMaster-5005

CFR-S Replacement 20mm Short Crossfader
Fits all mixers with CF-R

CF-PCV Replacement Long Life Crossfader
For PMC-05, 06, 07, 07 ISP, and all other PCV Series mixers





CF-40 Crossfader
For PMC-400, 46

CF-46MKII Crossfader
For PMC-46MKII

CF-50 Crossfader
For PMC-50

DCR-1200

3-Band Frequency Isolator



The original frequency isolator for house, techno and many other styles of mixing. Sound ranges are divided into three groups: High, Mid and Low. The DCR-1200 allows separate volume control of each range.

- Stereo 3-band 5 frequency sound control isolator
- Sound ranges are divided into High, Mid and Low groups
- DJs have total control over the volume of each range individually
- Using the on/off switches, DJs can invert the right and left sound and erase the center-sound instantly
- Input/Output: 1-Line (phone), 1-Master (phone), 1-Divided Hi/Mid/Low Output (phone)
- Frequency antiphase controls for additional performance
- Fast action trough large aluminium knobs
- Creative sound part for any DJ set-up
- Antiphase controls to cut off selected sounds or voices
- Low and high frequency outputs

DCR-1200 Pro

4-Band Frequency Isolator



Get total creative freedom with Hi/Mid-Hi/Mid-Low/Low frequencies. Two frequencies can be chosen on Mid-Hi and Mid-Low. The 4-band control allows DJs to divide the frequency and control the music more precisely.

- Stereo 4-band 6-frequency sound control isolator
- The cut-off frequency can be selected from 2 different frequencies enabling DJs to create various kinds of frequency control
- XLR balanced (+4dB) and 1/4″ unbalanced (-0dB) connections are provided for various occasions and connections
- Large aluminium control knobs enable easy and accurate setting
- AC power supply (no adaptor) creates wider dynamic range and lower noise
- Quick on-off action can be made with bypass switch 19" 1RU rack mount configuration
- Creative sound part for any DJ set-up
- Frequencies switchable for extended usage
- Superb also in master output line

HALF RACK EFFECTOR SERIES

DPH-X1

The DPH-X1 is a stereo phaser with a 12-step phase shifting circuit that directly targets specific "DJ" frequency ranges. Using a combination of phase mode and phase shift controls, 18 superb phase sounds are quickly accessible.



- 12-step phase shifting circuit allows DJ to create an extremely wide range effect. From a natural wave to deep modulation, the possibilities for sound manipulation are extreme
- Bypass/Effect Crossfader allow you to instantly switch or move between dry sound or effect sound in one smooth "crossfade" action



DWG-X1

Utilizing proprietary Vextax Filter technology, the DWG-X1 offers sensational Wah effects that can be applied to a wide range of frequencies: from crisp top end to fat bottom end. The "color" of the filter sound can be uniquely manipulated by an extremely high quality 5-band graphic filter fader with Low/Mid-Low/Mid/Mid-High/High controls.

- Super Phasing effects through 12 stage circuit
- Creative sound part for any DJ set-up
- Phono and line inputs
- Extreme space and deep modulation effects will emphasize the audience

DFG-X2

This super compact frequency isolator separates frequencies perfectly in three parts: Hi, Mid and Low. Features a high spec phono amplifier, allowing you to easily connect a turntable directly to the DFG-X2.



- Line and phono inputs
- Fully 3- band frequency controlling filter bank
- 3-band stereo isolator for flexible usage
- Creative sound part for any DJ set-up



DDG-X2

Finally, a performance Digital Delay featuring a large BPM counter display. Send perfectly timed delayed signals, flying around the dance floor, quickly and easily by just tapping in the tempo. The Tap Function allows instant synchronization of the delay effect with the music source down to extremely fine increments.

- Half-rack 2U size
- Large 7-segment LED display shows BPM & delay status
- Height designed to fit snugly between turntables
- Grounding Post for direct turntable connection
- Simple, but complete, control over the dry and effected signals



DGE-X2

Remix artists will fail in love with the DGE-X2. Unique among graphic equalizers, the DGE-X2 allows you to modify the sound source in real-time, targeting particular frequency ranges.

- This unique 6-band eliminator combines advanced EQ and isolator functions to give you amazing cut control over your 60Hz, 160Hz, 500Hz, 1.6kHz, 5kHz and 16kHz frequencies.
- The no boost cut only vertical faders can be adjusted real-time and the whole effects setting can be bypassed returning to the original sound of the input source quickly and easily

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VIDEOS, CD PLAYERS, TABLE

Tokyo Competition 1998



Tokyo Competition 1999



Tokyo Competition Volume I 2000



Tokyo Competition Volume II 2000



DJ Extravaganza Volume I 2001



DJ Extravaganza Volume II 2001





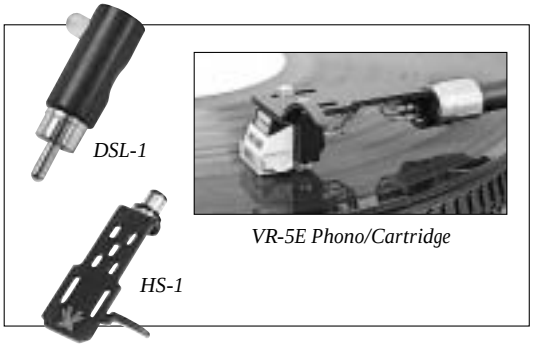
DJT-1400 Professional DJ Table

The DJT-1400 aluminum DJ table is both lighter and more compact than other DJ tables. It ships flat, yet takes mere minute to assemble.

- The aluminum foot in the truss structure provides great support and reduces the weight of the whole body
- Easy to carry around and transport
- 1.3˝ thick, 4.6˝ wide hardboard guarantees high stability for any DJ play
- Superb high lacquer finish

VR-5E Phono/Cartridge
A pro DJ phono/cartridge that combine stracking accuracy and durability. The elliptical stylus is good for all styles **48.00**

HS-1 BLK
Black headshell **24.99**



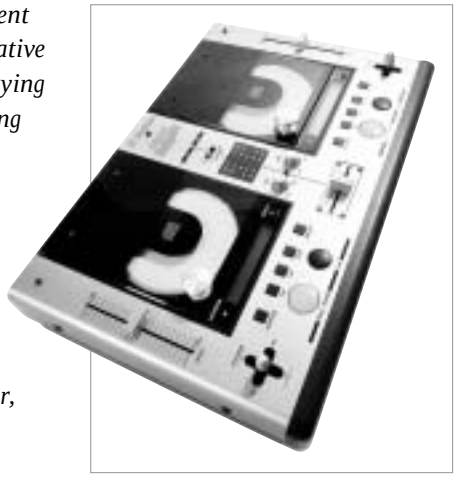
DSL-1
Detachable Stylus light.....**24.99**

AC-12A Power Adaptor
For PMC-15, 06T, 07 Pro and 25.....**24.99**

DC-15A Power Adaptor
For 05, 06, 170A and 03 .....**24.99**

CDX-16 Dual CD Mixing Console

The CDX-16 is a fun, intelligent and elegantly designed alternative to traditional CD players. Playing one or two CDs is easy, creating your own mix is simple. The combination of a shockproof mechanism and 10 second buffer memory make the CDX-16 able to be used in any application, be it club, mobile or professional studios. The built in crossfader, eliminates the need for an external mixer.



- Each CD can be monitored through the headphones allowing the user to practice the mix first. Last “stop” point can be recalled as a “cue” point instantly for quick mix situations.
- +/- 8% pitch control faders are provided for each disc for quick accurate beat matching
- Dual Function joystick is provided for easy cueing and pitch bending
- Single/All repeat switch enables repeat play of either one song or the whole CD