Ozone 11 Help

Introduction

Craft the perfect listening experience with Ozone 11, the ultimate collection of mastering tools. Whether you're putting the finishing touches on the next chart-topping hit or producing your first song, Ozone 11 delivers cutting-edge processing and AI-powered workflows. Effortlessly make your tracks release-ready and unlock the full potential of your productions.

Getting Started

Table of Contents

Common Terms

The following terms are used throughout this manual to specify different types of Ozone plug-ins:

1. **Mothership plug-in**: This term is used to describe to the main Ozone plug-in that includes multiple processing modules, a customizable signal chain, Master Assistant, Referencing, Codec Preview, and Dither.
2. **Component plug-in**: This term is used to describe the different Ozone plug-ins that include a single Ozone processing module.
    1. **The following Ozone modules are available as component plug-ins**: Clarity, Dynamic EQ, Dynamics, Equalizer, Exciter, Imager, Impact, Low End Focus, Master Rebalance, Match EQ, Maximizer, Spectral Shaper, Stabilizer, Vintage Compressor, Vintage EQ, Vintage Limiter, and Vintage Tape.

OZONE GLOSSARY

Check out the **Glossary** chapter for additional information about terms used throughout this manual.

Quick Start Suggestions

Not sure where to start? Try using presets or Master Assistant to get going quickly with Ozone.

Using Presets

Ozone includes a wide variety of factory **presets** to help get you started on your master.

In the **Ozone mothership plug-in**, you can try out different mastering chains using **global presets**. Global presets will modify the signal chain and the settings of each module included in the chain. You can use **module presets** to audition different settings in an individual module. You can open the module preset manager by clicking on the preset button in a module’s tile in the signal chain.

In the **Ozone component plug-ins**, you can open the preset manager by clicking the presets button in the plug-in header area. The preset files used by Ozone component plug-ins are the same files that are used by the module preset managers in the Ozone mothership plug-in.

Using Master Assistant

The intelligent Master Assistant feature allows you to quickly to develop a starting point for your master. Master Assistant is only available in the Ozone mothership plug-in. Click on the Master Assistant button in the mothership plug-in header area to open the Master Assistant setup panel. Check out the **Master Assistant** chapter to learn more.

Navigating the Plug-in Interface

The Ozone plug-in interface is divided into the following main sections:

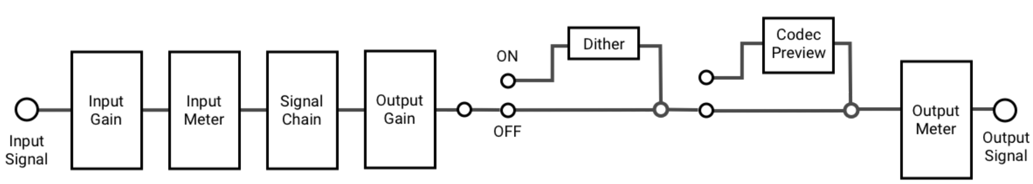
1. **Global Header**: Includes Stem Focus (mothership plug-in only), **Master Assistant** (mothership plug-in only), **Presets**, IPC instance name, Undo History, **Options**, and Help.
2. **Signal Chain**: Adjust the contents and order of the processing chain.
3. **Module Interface**: Controls and meters associated with a specific processing module.
4. **I/O Panel**: I/O (Input/Output) gain and metering, global bypass, and a number of auditioning features.

LEARN MORE

See the **General Controls** chapter for more information about the controls included in the main areas of the Ozone interface.

Signal Flow in Ozone

The following diagram outlines the high-level signal flow in the Ozone mothership plug-in.



The Ozone mothership plug-in includes a customizable signal chain. By default no modules are included in the signal chain.

Plug-in Feature Differences

There are a few features that will only appear in the Ozone mothership plug-in and/or in select Ozone component plug-ins. The following table outlines features that are only available in specific Ozone plug-ins:

Feature	Available in
Master Assistant	Ozone mothership plug-in *only*
Stem Focus	Ozone mothership plug-in *only*
Signal Chain	Ozone mothership plug-in *only*
Referencing	Ozone mothership plug-in *only*
Codec Preview	Ozone mothership plug-in *only*
Dither	Ozone mothership plug-in Ozone Maximizer component plug-in
I/O panel: **Sum to Mono** and **Swap Channels**	Ozone mothership plug-in Ozone Imager component plug-in

Tips for Optimizing Performance

There are a number of factors that can affect performance of an individual plug-in or application and there is a chance you may run into performance issues from time to time. Here are some adjustments you can make to help mitigate performance issues you might encounter:

1. Remove any modules you are not using from the signal chain.
2. Try increasing the Buffer Size setting in your DAW/NLE, if possible.
3. If using the Ozone Equalizer in Digital mode: Try adjusting the EQ buffer size in the **Options** menu for optimal performance.
4. If using the Digital crossover type in any multiband module: Try adjusting the Crossover Buffer Size in the **Options** menu.
5. If possible, try reducing the number of bands enabled in any of the multiband modules.
6. If possible, try using the Stereo channel processing mode rather than Mid/Side, Left/Right, or Transient/Sustain.

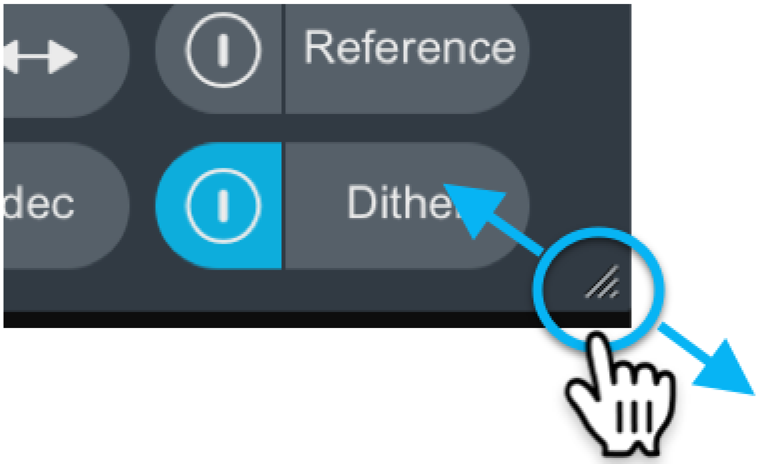
General Controls

Overview

This chapter includes information about general controls and functions you can find in Ozone.

Resizing

You can resize the window by clicking and dragging the bottom right corner of the Ozone interface.



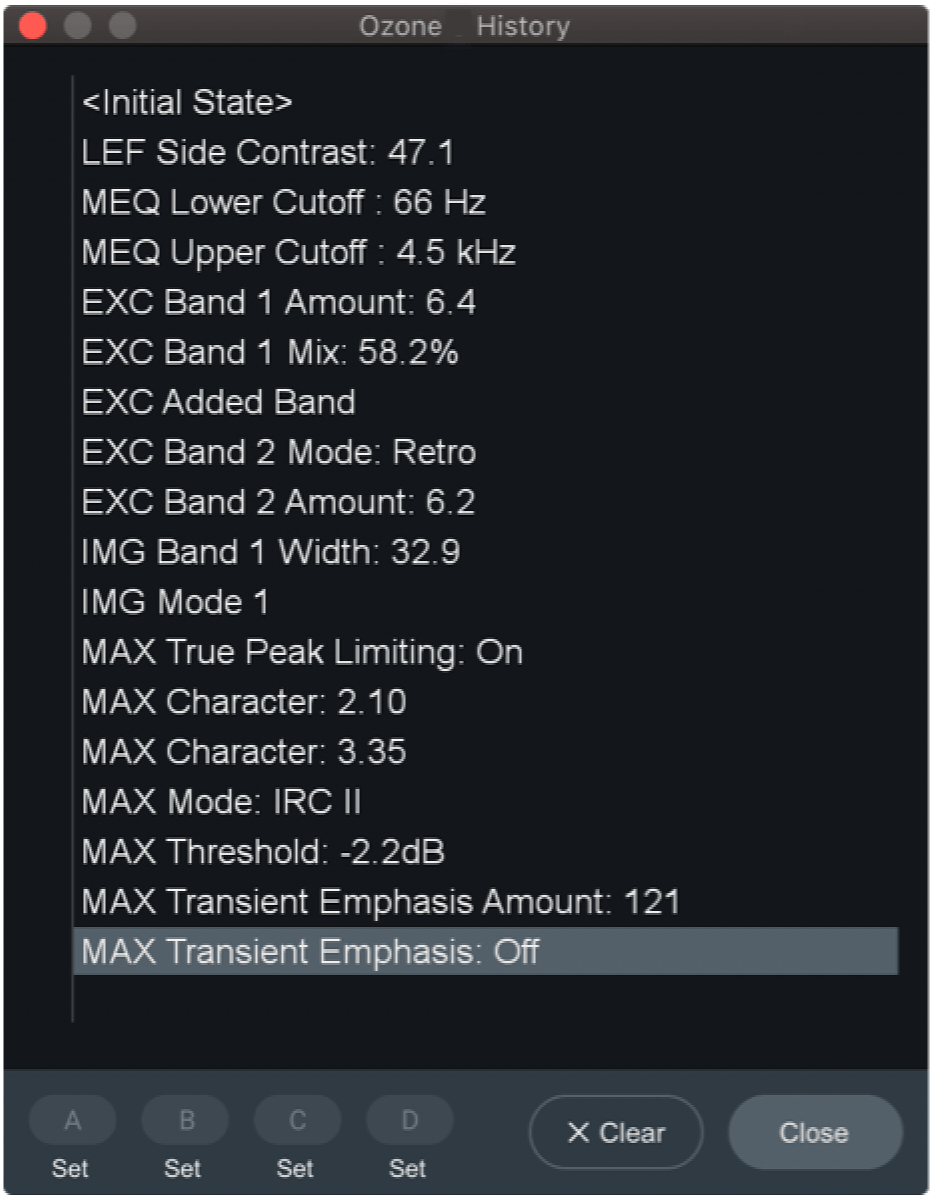
Header controls

The Ozone header area includes the following features:

1. **Stem Focus**: In the default Full Mix option, processing is applied to the full input signal. When Vocals, Bass, or Drums are selected, all modules in the processing chain will be applied only to that stem. *Ozone mothership plug-in only*
2. **Master Assistant View**: Begins analysis or opens the Master Assistant View. *Ozone mothership plug-in only*
3. **Module View**: Returns to the Module View from the Master Assistant View.
4. **Presets** Opens the Preset manager window. The two arrows select the next or previous preset.
5. **IPC Instance Name**: Name of the current instance as it will appear in IPC compatible plug-ins.
6. **Undo**: Reverts the most recent parameter change.
7. **Undo History**: Opens the Undo History window. See the **Undo History** section below for more information.
8. **Options**: Opens the Options window.
9. **? (Help)**: Opens the installed Ozone help documentation in your default web browser.

Undo History

The Undo History window allows you to view a list of recent parameter changes made in the current instance of Ozone, compare different parameter states in the history list, or reset Ozone's settings to a given point in the history list.

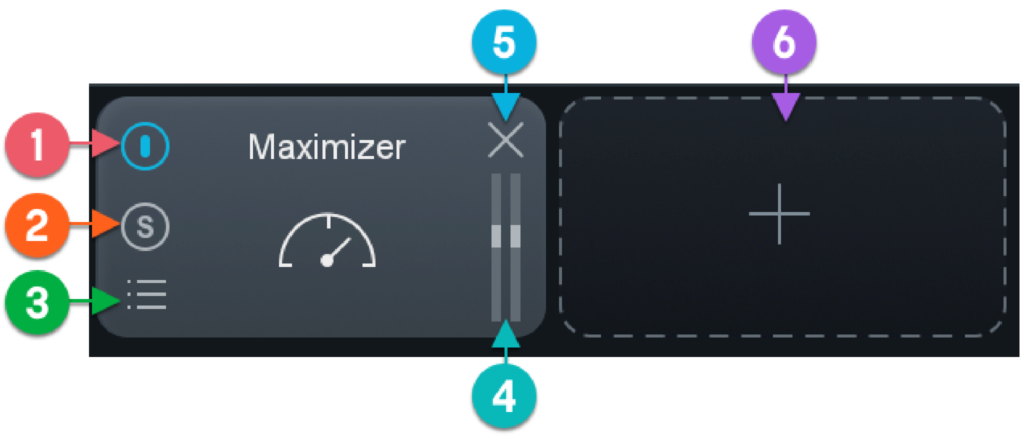


1. **CLEAR**: Removes all actions from the history list. Clearing the history list does not affect parameters, it simply removes existing

entries from the list.
2. **CLOSE**: Hides the Undo History window.
3. **A/B/C/D**: Allows you to set four history snapshot states. Snapshots allow you to quickly toggle between different processing states to compare changes. Assign an event in the history list to a snapshot button by selecting the event in the list and clicking the “Set” button below a snapshot button.

Signal Chain

You can choose modules and adjust their processing order in the signal chain. The following image outlines the controls available in the signal chain area of the Ozone mothership plug-in:



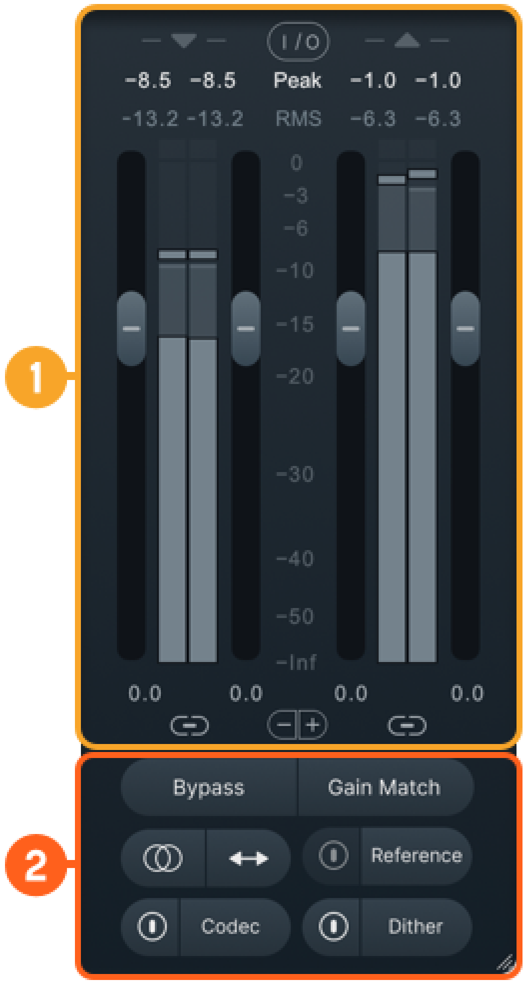
1. **Power**: Enables/disables processing for the associated module.
2. **Solo**: Disables all other module processing and isolates the associated module.
3. **Module presets**: Opens the module preset manager window for the associated module.
4. **Difference meters**: Displays the amount of gain change introduced by processing in the associated module.
5. **x (Remove)**: Removes the associated module from the Signal Chain.
6. **+ (Add)**: Opens the module selection menu.

The following modules are available in the signal chain selection menu:

1. Clarity
2. Dynamic EQ
3. Dynamics
4. Equalizer 1
5. Equalizer 2
6. Exciter
7. Imager
8. Impact
9. Low End Focus
10. Master Rebalance
11. Match EQ
12. Maximizer
13. Spectral Shaper
14. Stabilizer
15. Vintage Compressor
16. Vintage EQ
17. Vintage Limiter
18. Vintage Tape

I/O Panel

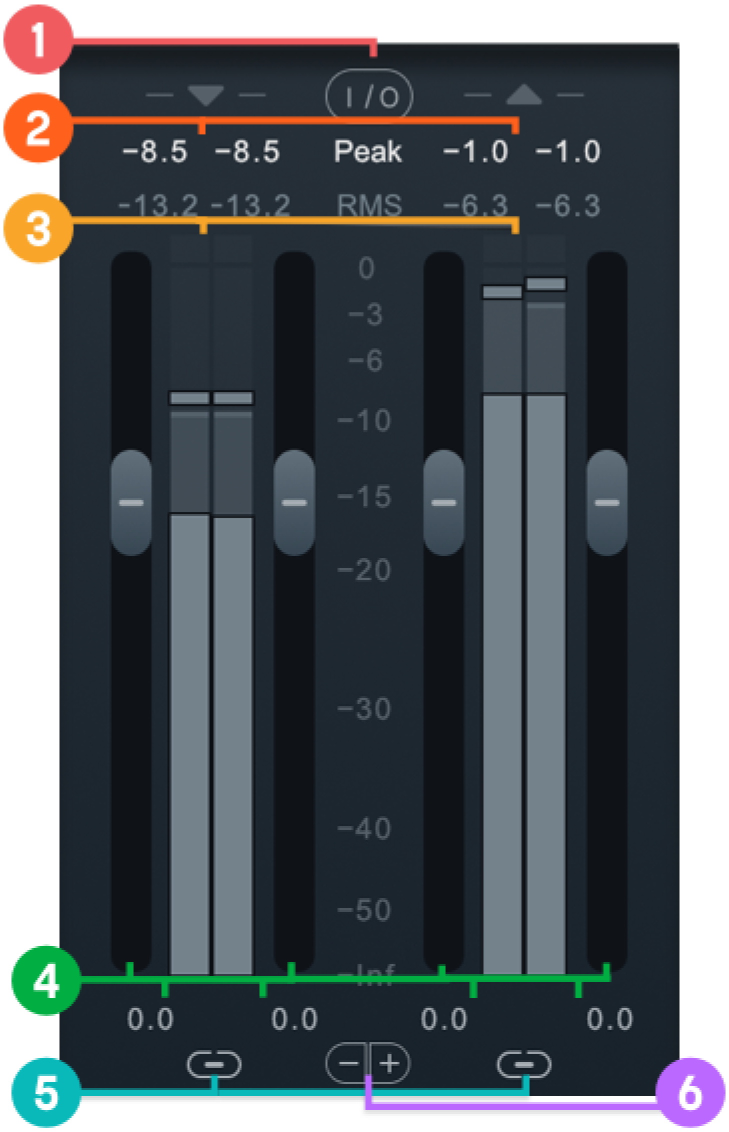
The I/O panel is split into two main sections:



1. **Input/Output Gain and Meters**
2. **Global Processing and Auditioning**

Input/Output Gain and Meters

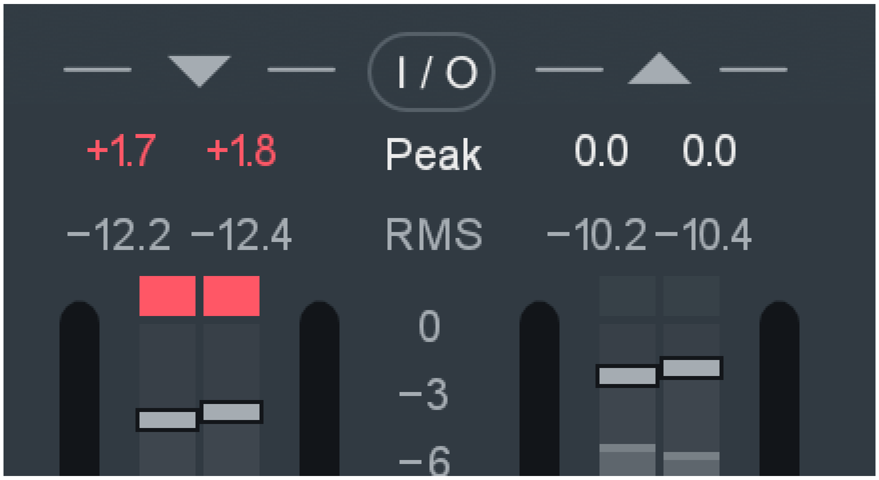
The following image outlines the input and output metering and gain controls available in Ozone.



1. **I/O Meter Options**
2. **Input/Output level readouts**: Shows the current or max input levels, according to the selected meter type.
3. **Input/Output meters**: Displays input/output level information according to the I/O meter option configuration.
4. **Input/Output gain**: Adjusts the amount of gain applied to the input/output signal.
5. **Link**: Links/unlinks adjustments of left and right I/O gain controls.
6. **- | +**: Decreases or increases the resolution of the meter scale.

Clipping Indicators

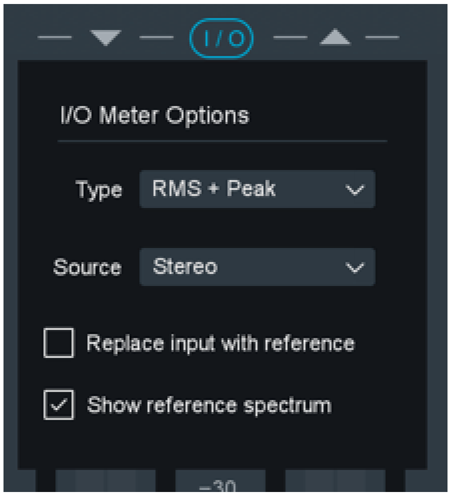
When the input or output signal exceeds 0 dBFS, the clipping indicator box and level readout text above the meters will be displayed in red.



You can click on the red readout text and/or clipping indicator box to reset the clipping indicator.

I/O Meter Options

Click on the **I/O** button above I/O meters to toggle the I/O Meter Options pop-over menu open and closed.



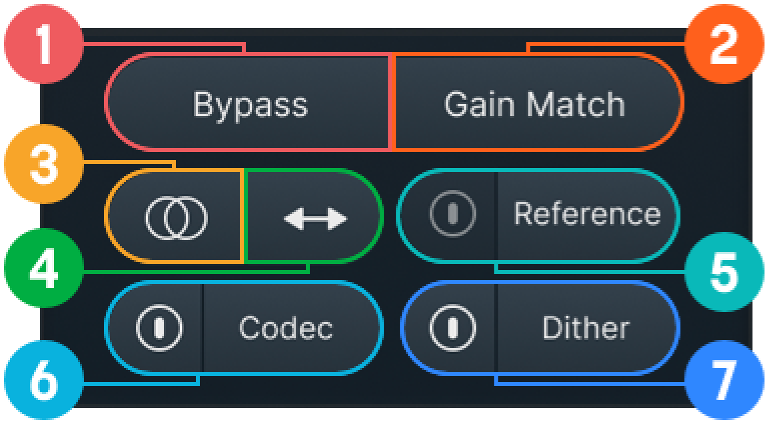
1. **Type**: Selects the metering calculation used to display information in the input and output meters.
2. **Source**: Selects between Stereo or Mid/Side metering displays the input and output meters.
3. **Replace Input with Reference**: When checked, the input meter will show metering data based on the currently selected reference track.
4. **Show Reference Spectrum**: When checked, the selected reference track spectrum will be displayed as a secondary spectrum in all module spectrum views.

ADDITIONAL INFORMATION

1. Reference related options are only available in the Ozone mothership plug-in.
2. Learn more about the individual I/O meter Type and Source options in the **Options** chapter.
3. Learn more about working with reference tracks in the **Reference** chapter.

Global Processing and Auditioning

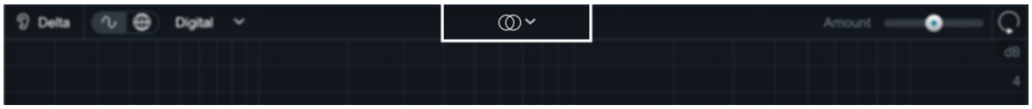
The section located directly below the I/O meters and gain controls includes:



1. **Bypass**: Select to disable all processing in the current instance of Ozone.
2. **Gain Match**: Toggle to enable/disable gain matching of the output signal. Behavior depends on the "**Enable modern bypass gain match behavior**" setting in the **Options** window.
3. **Sum to Mono**: Sums the stereo output channels into a mono signal. Useful for checking mono compatibility. *Available in:* Ozone mothership plug-in & Ozone Imager component plug-in.
4. **Swap Channels**: Swaps the left and right output channel assignments. *Available in:* Ozone mothership plug-in & Ozone Imager component plug-in.
5. **Reference**: Toggle the power button to enable/disable reference track playback. Click the **Reference** button to open the **Referencing** panel. *Available in:* Ozone mothership plug-in only.
6. **Codec**: Toggle the power button to enable/disable Codec Preview. Click the **Codec** button to open the **Codec Preview** panel. *Available in:* Ozone mothership plug-in only.
7. **Dither**: Toggle the power button to enable/disable Dither processing. Click the **Dither** button to open the **Dither** panel. *Available in:* Ozone mothership plug-in & Ozone Maximizer component plug-in.

Channel Processing Modes

Ozone offers different channel processing modes that determine how processing is applied within a given module. When a module supports more than one processing mode, a selection menu will appear in the module header area.



The following options are available in the dropdown menu:

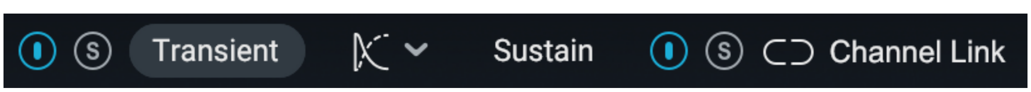
1. **Stereo**: Default processing mode for all modules, including those that do not offer explicit channel processing options. When selected, only one set of controls will be exposed and adjustments will apply to both channels.
2. **Mid/Side**: Applies Mid/Side encoding to the signal coming into the module, allowing you to process the Mid and Side channels independently. One set of parameters applies processing to the encoded Mid channel and the second set of parameters applies processing to the encoded Side channel. The signal is decoded to a stereo signal at the output of the module.
3. **Left/Right**: Allows you to process the Left and Right channels independently by splitting the module input signal into two processing channels. The signal is then 'summed' back to Stereo at the output of the module. One set of parameters applies processing to the Left channel and the second set of parameters applies processing to the Right channel.
4. **Transient/Sustain**: Analyzes the signal coming into the module to isolate transients and separate them from the sustained portion of the signal. The Transient and Sustain can then be processed separately. One set of parameters applies processing to the Transient channel and the second set of parameters applies processing to the Sustain channel. The two signals are mixed back together at the output of the module. If no processing is applied, the Transient and Sustain will mix back together to form the exact same signal.

The following modules offer Mid/Side, Left/Right, and/or channel processing mode options:

Module	Mid/Side	Left/Right	Transient/Sustain
Clarity	X		X
Dynamics	X		
Dynamic EQ	X	X	X
Equalizer	X	X	X
Exciter	X		X
Imager			X
Impact	X		
Match EQ	X	X	X
Low End Focus	X		X
Spectral Shaper	X		X
Stabilizer	X		X
Vintage Compressor	X		
Vintage EQ	X	X	X

Channel Processing Mode Controls

When Mid/Side, Left/Right, or Transient/Sustain mode is selected, the following controls become available in the module header area:



1. **Power Button**: Enables/disables processing in the associated channel.
2. **Solo**: Isolates the output of the associated channel.
3. **Channel selection buttons:** Select the associated channel view by clicking the “Mid” or “Side” buttons (in Mid/Side mode), the “Left”

or “Right” buttons (in Left/Right mode), or the “Transient” or “Sustain” buttons (in Transient/Sustain mode).
4. **Channel Link**: Enables/disables linked parameter adjustments on both channels.

Master Assistant

Table of Contents

Overview

Master Assistant is designed to be your second set of AI-powered ears. It will analyze your track and offer an objective suggestion to help you achieve a professional-sounding master. For a beginner, this will hopefully give you the confidence to share your work with the world. For an expert, we hope that Master Assistant will provide a valuable second opinion and a faster setup.

The Ozone 11 Master Assistant comes with a set of ten genre targets that have been derived from the latest chart-topping hits. When a target is selected, the assistant will match your song’s tonal balance, vocal balance, width, dynamics, and loudness to the average of those songs. You can also generate and save your own custom targets for Master Assistant based on reference song files on your computer. The new Master Assistant overview page provides powerful, high-level control over all of this matching so that you can dial-in the settings to your taste.

Workflow Steps

1. Add Ozone 11 to the Master channel or Stereo Output of your DAW.
2. Click the **Master Assistant Tab** in the header area of the Ozone mothership plug-in to begin the analysis phase.
3. **Analysis**: Master Assistant requires audio input to perform analysis and adjust settings.

    1. Playback your track for **at least 8 seconds** so that Master Assistant has enough time to analyze the input audio. Loop playback in your DAW if you are analyzing a selection that is less than 8 seconds long.
    2. Playback the **loudest portion** of your track to achieve the best results.

4. After analyzing your song, Master Assistant will build a processing chain and switch to the Master Assistant controls view when the processing chain is built.

Master Assistant View

The Master Assistant View provides high-level controls and metering in the following sections:

1. **[Header]**
2. **[Target Library]**
3. **[Tonal Balance]**
4. **[Loudness]**
5. **[Vocal Balance]**
6. **[Extras]**

Header

1. **Master Assistant Tab**: If analysis has already been performed, selecting this tab will show the Master Assistant View.
2. **Module View Tab**: shows the module view where you can adjust, add, or remove modules. Note that if you return to the Master Assistant view after making adjustments in the Module View, those changes will be overwritten by selecting a new Target.
3. **Gain Match**: matches Ozone’s output level to the input level by automatically compensating for any gain changes introduced by the processing.
4. **Global Bypass**: disables all Ozone processing.
5. **Relearn**: launches the analysis phase and sets up new targeted processing for the newly-measured section.
6. **Settings**: shows the Ozone settings menu.
7. **Help**: opens a web browser with the Help Documentation.

Target Library

You can modify the overall target that the Master Assistant will work toward in the Target Library section. Changing the target selection will adjust all Master Assistant processing to work toward matching your track to the selected target. The factory target curves were generated through an analysis of chart-topping hit songs from a range of different genres. The Cinematic target was generated through an analysis of the scores of top box office films.

Each Target is comprised of sub-targets that are associated with different aspects of Master Assistant processing:

1. **Tonal Balance**: the distribution of energy across the audible frequency spectrum.
2. **Vocal Balance**: the difference in loudness between the vocals and the rest of the music.
3. **Stereo Width**: the ratio of mid to side signal power in each band.
4. **Dynamics**: a measure of microdynamics, defined by the ratio of a very short term loudness to short term loudness.
5. **Loudness**: the integrated loudness measured in LUFS.

Targets from Reference files

You can create and manage your own custom reference targets by importing audio files from your computer.

1. **Plus button**: opens a system dialog where you can select audio files on your computer to add to your custom Target Library. Adding or selecting a new reference file target will require a short analysis phase to build a new signal chain.
2. **Trash**: deletes a selected reference file target.

NOTE

Master Assistant will analyze an uploaded reference file and learn the loudest eight seconds. The target will be generated from that loudest section of audio.

CUSTOM VOCAL BALANCE

Master Assistant does not create vocal balance targets from custom references. When a custom reference target is selected, the Master Rebalance module will not be added to the signal chain.

Tonal Balance

The Tonal Balance section has one control:

1. **Equalizer**: scales the gain of all EQ nodes from 0%-200%. This links directly to the EQ global amount control in the upper right of Equalizer 1.
    1. The Equalizer 1 module can also be enabled and disabled with the power button for quick comparisons. When Gain Match is enabled, any gain introduced by processing will be compensated when bypass is engaged.

The Tonal Balance section also features metering that will be familiar to users of our product, Tonal Balance Control. When a Target is selected, you will see this reflected in the blue tunnel. The white line represents your audio’s tonal balance. Increasing the Equalizer, Stabilizer, or Clarity amount should result in your track aligning more closely with the Target tonal balance.

Loudness

The Master Assistant measures integrated loudness during the analysis phase and sets the Maximizer’s Gain to the level necessary to achieve the target loudness for the detected genre (or the loudness of a custom reference).

The Loudness section has one control and a destination selector:

1. **Maximizer**: adjusts the Gain of the Maximizer +/- 4 dB.

    1. The Maximizer module can also be enabled and disabled with the power button for quick comparisons. When Gain Match is enabled, any gain changes will be compensated during this bypass.

The Master Assistant will optimize your audio's output level for two destinations:

1. **Full Scale**: this is intended as a good output level for most masters. The Maximizer's Output Level will be set just shy of full scale at -0.1 dB with True Peak limiting disabled.
2. **Streaming**: this is intended for audio that will be played back with lossy encoding and loudness normalization, as is typical of most streaming services. The Output Level is set to -1 dB with True Peak limiting enabled.

This section also features a scrolling waveform and gain trace that will meter how much gain reduction or boost is being applied by the Maximizer. This is useful to dial in exactly how much limiting is taking place when adjusting the Maximizer slider.

Vocal Balance

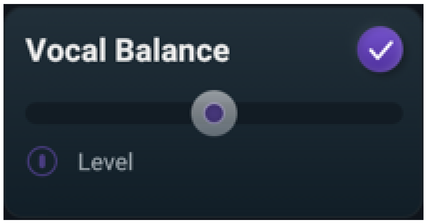
To perform a vocal balance, the Master Assistant will use AI to separate the vocal from the rest of the music during the analysis phase. It measures the integrated loudness of both the vocal and music. Master Assistant will then apply the Master Rebalance module to adjust your vocals towards a target loudness difference between the vocal and music.

The Vocal Balance section has one control:

1. **Level**: adjusts the gain the Master Rebalance module, focused on vocals.

    1. The Master Rebalance module can also be enabled and disabled with the power button for quick comparisons.

If Master Assistant measures the vocal/music balance and determines that it is well balanced for your genre target, it will show a check mark and not add Master Rebalance to the chain.



If Master Assistant does not detect a vocal during analysis, it will not add Master Rebelance and not show a check mark.

Extras

The Extras section has four controls:

1. **Dynamics Match**: scales all of the per-band Impact module amounts from 0% to 100%. This control maps directly to the global amount in the upper right corner of the Impact module.

    1. The Impact module can also be enabled and disabled with the power button for quick comparisons. When Gain Match is enabled, any gain changes will be compensated during this bypass.

To perform a dynamics match, the Master Assistant will analyze the microdynamics in your audio across four separate bands. It will then use the Impact module to adjust the microdynamics to match the target.

1. **Width Match**: scales all of the per-band Imager amounts from 0% to 100%. This control maps directly to the global amount in the upper right corner of the Imager module.

    1. The Imager module can also be enabled and disabled with the power button for quick comparisons. When Gain Match is enabled, any gain changes will be compensated during this bypass.

To perform a width match, the Master Assistant will analyze the balance of mid and side information in your audio across four separate bands. It will then use the Imager module to adjust the side channel to match the selected target. Note that Master Assistant may enable Recover Sides at different amounts based on the selected Target. It will not enable Stereoize. If the low frequency band of the analyzed audio is sufficiently narrow, Master Assistant will not attempt to widen the bass to match a selected target.

1. **Clarity Amount**: controls the amount of Clarity processing. This links directly to the main amount control in the Clarity interface. Clarity will apply a Tilt value that is appropriate for the target genre. If a reference file target is selected, the Master Assistant will analyze its genre and apply that genre's tilt.

    1. The Clarity module can also be enabled and disabled with the power button for quick comparisons. When Gain Match is enabled, any gain changes will be compensated during this bypass.

2. **Stabilizer Amount**: controls the amount of Stabilizer processing. This links directly to the main amount control in the Stabilizer interface. Stabilizer will match its target to the selected genre. If a reference file target is selected, the Master Assistant will generate a custom "Assistant" target for Stabilizer based on the tonal balance of the imported reference file.

    1. The Stabilizer module can also be enabled and disabled with the power button for quick comparisons. When Gain Match is enabled, any gain changes will be compensated during this bypass.

Clarity

Table of Contents

Overview

Clarity uses hundreds of bands to maximize the spectral power of your audio. This is achieved by adaptively contouring the spectrum towards flat noise. Clarity will tame resonances and unmask background content. This can effectively “pull the blanket off” dull mixes without sounding harsh.

In scientific, blind listening tests we have also found that the Clarity processing can increase the perceived loudness of a master at the exact same integrated LUFS.

The Clarity interface includes the following sections:

1. Module Header
2. Controls
3. Difference Meter and Action Region

Module Header

The module header includes the following controls:

1. **Delta**: Monitors the difference in the signal before and after the Clarity module.
2. **Channel Processing Mode**: Selects the Channel Processing Mode used in the Clarity module.

    1. The Clarity module supports **Stereo**, **Mid/Side**, or **Transient/Sustain** mode.
    2. See the **General Controls** chapter for more information.

3. **Reset**: Returns all module controls to their default values.

Controls

The Clarity module includes the following controls:

Amount

Scales the gain of Clarity's spectral processing. At 100, the maximum gain that Clarity can possibly boost or cut is 6 dB.

Tilt

Adjusts the flat target that Clarity is contouring towards. At the default 0 dB/oct, the target is pink noise. This can be adjusted in dB/oct where positive values are brighter and negative values are darker. For example, tilting +3dB/oct will target white noise. A Tilt of -3 dB/oct will target brown noise.

Attack

Adjusts the amount of time (in milliseconds), for Clarity to react to the incoming signal.

Release

Adjusts the amount of time (milliseconds) for Clarity to return to a baseline of no gain change.

Difference Meter and Action Region

Displays the tonal changes that the Clarity processing is applying. This is similar to an EQ where frequency is measured from left to right and gain is up and down. Different from an EQ, Clarity will adapt its spectral contour over time.

The Clarity processing can be constrained to a particular frequency area with the action region handles. The action region allows for processing between 300 Hz and 20kHz

Dynamics

Table of Contents

Overview

You can use this module to shape the dynamics of your mix, with up to four bands of analog-modeled compression and limiting.

1. Module Header
2. Views
3. Controls and Meters

Module Header

The module header includes the following controls:

1. **Delta**: Monitors the difference in the signal before and after the Dynamics module.
2. **View Selector**: Switches between the Crossover Spectrum, Gain Reduction Trace, or Detection Filter views. See the **Views** section below for detailed descriptions of the different meter views.
3. **Link Bands**: When enabled, all band specific control adjustments will be linked relative to their current settings.
4. **Channel Processing Modes:** Selects the channel processing mode used by the Dynamics module. Dynamics supports **Stereo** and **Mid/Side** mode. See the **General Controls** chapter for more information.
5. **Auto Gain** and **Adaptive Release**:

    1. **Auto Gain**: When enabled, make-up gain is automatically calculated and applied to the output signal to compensate for level differences introduced by dynamics processing. The automatic gain control calculates the RMS levels of the input and output signals independently for each Dynamics crossover band. Gain is automatically applied to the output signal based on the RMS level difference between the input and output signals. **Auto gain and manual global/per-band gain adjustments can be applied simultaneously.**
    2. **Adaptive Release**: When enabled, automatically adjusts the Release time of the Compressor based on the peak factor of a signal. If a **transient signal** is detected, the Release time is scaled to be *shorter* in order to reduce pumping. If a **sustained signal** is detected, the Release time is scaled to be *longer* in order to reduce distortion.

6. **Learn**: Enables automatic crossover point placement. When active, crossover points will be moved to minima detected in the frequency spectrum of your track. Learn will automatically disable itself when crossovers are set to their ideal frequency values. Learn is not available in single band processing mode.
7. **Reset**: Returns all module controls to their default values.

Views

You can toggle between the different views using the view selector buttons:

The Dynamics module includes the following views:

1. **Crossover Spectrum**
2. **Gain Reduction Trace**
3. **Detection Filter**

Crossover Spectrum

All multiband modules in Ozone support up to four processing bands. You can create new processing bands and manage multiband crossovers in the Crossover Spectrum view. **Note**: crossover cutoff frequencies are not shared or linked across multiband modules.

DYNAMICS GAIN REDUCTION/INCREASE IN CROSSOVER

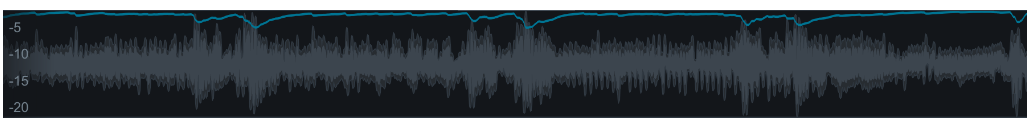
The horizontal lines drawn in band colors display gain reduction/addition occuring in the associated processing band. This is only shown in the Dynamics module crossover view.

Adjusting crossovers

1. **Add band**: Hover your cursor over the crossover spectrum view and click the **+** button to add a new crossover point.
2. **Enable/disable band**: Click the **power** button to enable/disable processing in the associated band.
3. **Solo band**: Click the **S** button to isolate playback of the associated band.
4. **Remove band**: Hover over a band in the crossover view and click the **x** that appears to remove the band.
5. **Adjust crossover cutoffs**: Click and drag a handle to adjust the crossover frequency. You can also double-click on a crossover handle and enter the value manually in the inline edit field that appears.

Gain Reduction Trace

Displays a scrolling waveform with a superimposed trace reflecting the amount of gain reduction applied over time.

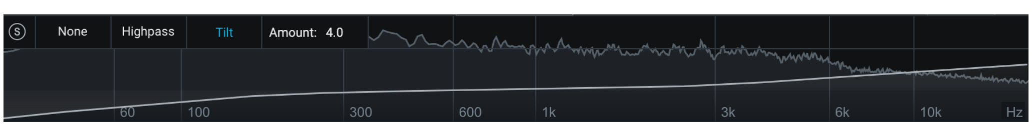


GAIN REDUCTION IN MULTIBAND

The gain reduction trace shows information for one band at a time. The trace color matches the color of the band it is metering.

Detection Filter

This view includes a spectrum analyzer and detection input filter controls. The detection filter allows you to apply a filter to the signal used by the level detector to trigger dynamics processing.



The following controls are available in the Detection Filter view:

1. **None**: Disables the detection input filter. This is the default setting.
2. **Highpass**: Enables a highpass filter. You can adjust filter slope by clicking the filter node and dragging the node handles horizontally.
3. **Tilt**: Similar to the THRUST circuit found on API compressors, Tilt mode preserves low frequencies using a high-frequency weighted filter curve.

    1. **Amount**: Controls the Tilt filter slope. Adjust Amount by clicking and dragging over the readout or by double-clicking on the readout and entering a new value in the inline edit field.

4. **Solo**: Isolates playback of the detection input signal after filtering.

Band Control Views

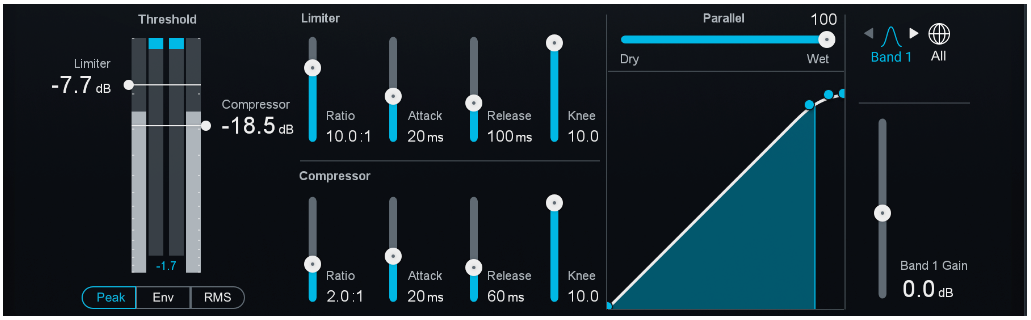
There are two different views for band controls in the Dynamics module panel:

1. **Selected Band View**
2. **All Bands View**

You can toggle between these views using the **Band View Selectors** located on the right side of the Dynamics module controls panel.

Selected Band View

Provides a focused view of the selected processing band. This is the default band view selection.

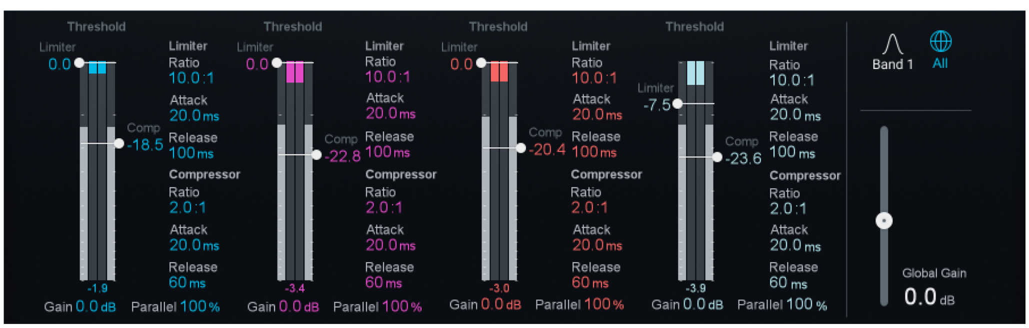


You can change which band is displayed using the following methods:

1. **Band View Selector arrows**: Click the arrow buttons to the left/right of the Band View button to switch between band views.
2. **Select band in Crossover Spectrum**: Click on a band region in the **Crossover Spectrum** to show the associated controls in the Selected Band View.

All Bands View

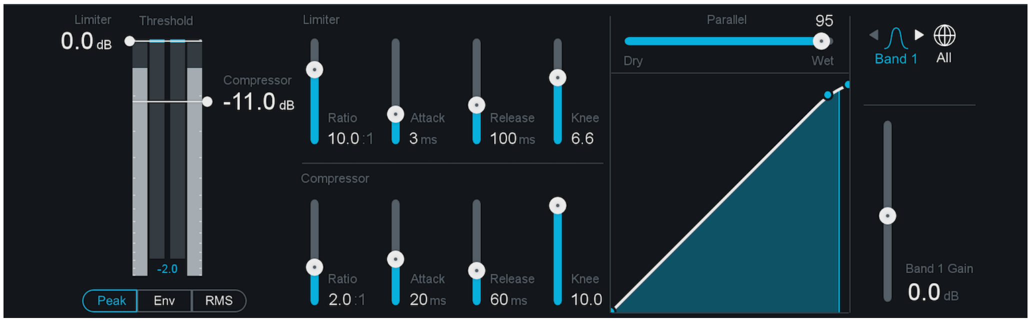
This view offers a consolidated overview of key controls for all four bands. If a band is inactive, the controls will appear greyed out in this view.



> **NOT ACCESSIBLE IN ALL BANDS VIEW**
>
> The **Knee** (Limiter/Compressor) control, global **Level Detection Mode**, and the **Dynamics Curve Meter** are hidden from the **All Bands View**, switch to the Selected Band View to access these features.

Controls and Meters

The following sections outline the controls and meters included in the Dynamics module.



1. **Level Detection Mode**
2. **Threshold**
3. **Ratio**
4. **Attack**
5. **Release**
6. **Knee**
7. **Parallel**
8. **Dynamics Curve Meter**
9. **Band View Selectors**
10. **Global and Band Gain**

Level Detection Mode

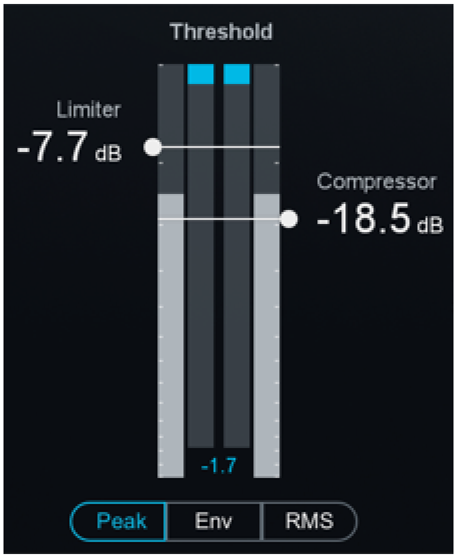
Determines how the input signal level is measured by the Dynamics level detection circuit. **This is a global setting, shared by all bands in multiband mode.**

1. **Peak**: Uses the peak level of the incoming signal. In general, this setting is useful when you are trying to even out sudden peaks in your music.
2. **Env**: (*Envelope*) Uses the average level of the incoming signal evened out across the frequency spectrum. Similar to RMS mode

but with some advantages. Unlike RMS, Envelope mode produces even levels across all frequencies and will not produce the aliasing or artifacts that RMS detection can cause.
3. **RMS**: Uses the average level of the incoming signal. RMS detection is useful when you are trying to increase overall level without changing the character of the sound.

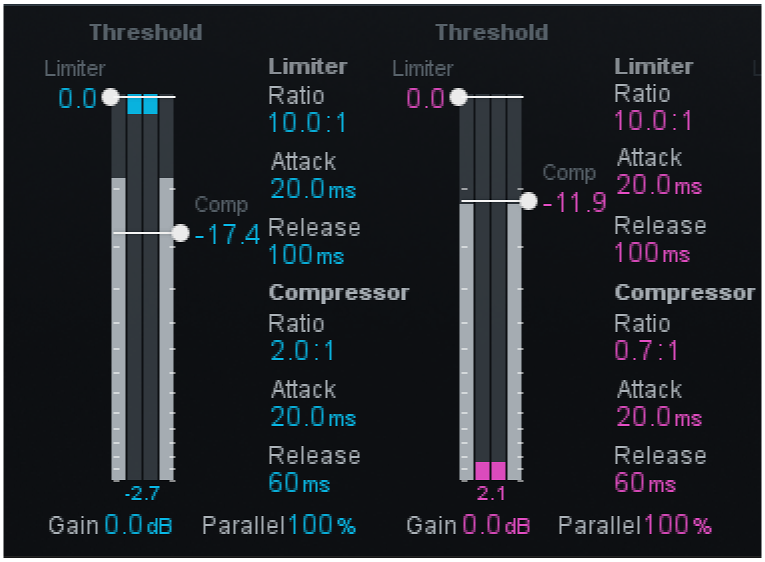
Threshold

Determines the signal level at which the compressor or limiter begins processing. The handle on the left controls the **Limiter Threshold**, the handle on the right controls the **Compressor Threshold**.



Threshold Input Meter

Displays the input level to the level detection circuit (outer meters, displayed in grey) alongside the gain change applied as a result of the dynamics processing (inner meters, displayed in the corresponding band color).



Gain reduction is drawn from the top to the bottom of the meter, gain increase is drawn from the bottom to the top of the meter.

Ratio

Determines the amount of gain reduction applied to a signal when the threshold is crossed.

Compressor Ratios

1. Compressor Ratio **Default**: **2:1**.
2. Compressor Ratio **Range**: **(10):1** to **30:1**. *Note*: (10):1 is shown as 0.1:1 in All Bands View.
3. **Positive Compressor ratios** result in *downward compression*. Downward compression reduces the level of signals that reach the threshold and leaves signals below the threshold unaffected.
4. **Negative Compressor ratios** result in *upward compression*. Upward compression raises the level of signals that fall below the threshold and leaves signals above the threshold unaffected. Upward compression can gently raise levels instead of pushing down peaks.

Limiter Ratios

1. Limiter Ratio **Default**: **10:1**.
2. Limiter Ratio **Range**: **(2.5):1** to **30:1**. *Note*: (2.5):1 is shown as 0.4:1 in All Bands View.
3. **Positive Limiter ratios** result in *downward compression (limiting)*. Higher ratios, particularly 10:1 or greater, result in limiting (more extreme downward compression).

> **LIMITER BEHAVIOR AT MAXIMUM RATIO SETTING**
>
> The Limiter in the Dynamics module is similar to the response of a limiter in the analog domain. When the Dynamics Limiter Ratio is set to 30:1, some overshoot may still occur because it does not function as a Brickwall limiter.

4. **Negative Limiter ratios** result in *upward expansion*. Upward expansion raises the level of signals that exceed the threshold. Upward expansion can be used to add punch to dull mixes or emphasize the beat in rhythm-heavy music.

Attack

Determines the amount of time, or how quickly it takes (in milliseconds), for the dynamics to react and for the signal to become fully compressed after exceeding the threshold level.

Release

Adjusts the amount of time (milliseconds) it takes for the dynamics processor to return to unity gain (recovers gain) when the input signal falls below the threshold.

Knee

Adjusts a range around the threshold that determines how abruptly processing is applied to a signal as it approaches the threshold.

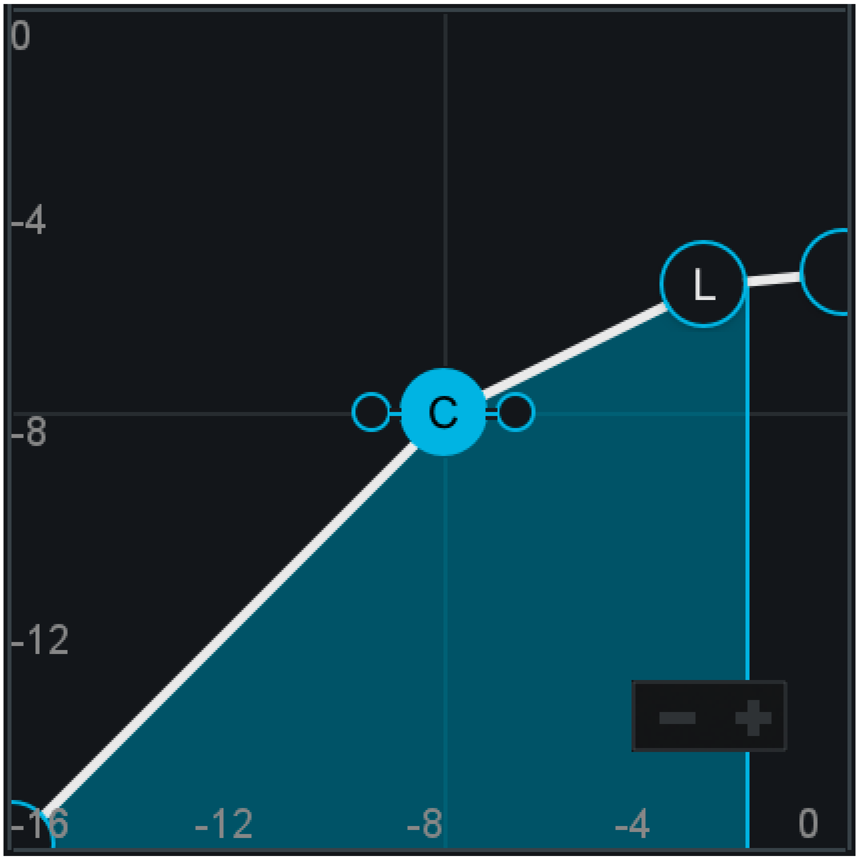
1. Higher values create a “soft knee” effect, which gradually introduces processing as the signal approaches the Threshold level. This provides subtler, more natural sounding compression.
2. Lower values create a “hard knee” effect, which abruptly begins processing when the signal crosses the Threshold level. This offers more aggressive sounding compression and is often used on individual tracks, such as kick or snare drum.

Parallel

Adjusts the amount of the “dry” (unprocessed) signal to mix with the “wet” (processed) signal. This technique is also known as “parallel compression”. This control is available for each band individually.

Dynamics Curve Meter

The Dynamics Curve Meter is an interactive plot of the Threshold, Ratio, and Knee controls as they relate to input and output levels. The y-axis represents output level (dB) and the x-axis represents input level (dB). You can zoom in/out by clicking the +/- buttons that appear in the bottom right when hovering your cursor over the meter. Signal activity is drawn below the curve in the associated band color.



Dynamics Curve Meter Nodes

1. **C** (Compressor): This node is associated with the **Compressor Threshold** and **Knee**. Moving left/right will affect Compressor Threshold. Adjusting the node handles will affect Compressor Knee.
2. **L** (Limiter): This node is associated with the **Limiter Threshold**, **Limiter Knee**, and **Compressor** Ratio. Moving left/right will affect Limiter Threshold. Adjusting the node handles will affect Limiter Knee. Moving up/down will affect **Compressor** Ratio (positive Ratios only).
3. **Leftmost and Rightmost nodes:** These nodes indicate the amount of gain reduction/increase applied to signals above/below the threshold.

Band View Selectors

Switch between **Selected Band View** or **All Bands View**.

Use the arrow buttons to the left and right of the Band # button to switch between available band views.

Global and Band Gain

Adjusts gain after Dynamics processing, useful for compensating for level differences caused by Dynamics processing. You can make adjustments to Global Gain and Band Gain simultaneously.

1. **Band Gain**: Adjusts the amount of gain applied to the associated band after Dynamics processing. When working in the **Selected Band View**, the Gain slider for the currently selected band is located on right side of the Dynamics module controls panel. When working in **All Bands View**, a Gain control is located directly below the threshold meter for each band.
2. **Global Gain**: Adjusts the amount of gain applied to the output signal of all bands after Dynamics processing. The Global Gain control is located on the right side of the Dynamics module controls panel when **All Bands View** is selected. The Global Gain slider is hidden when you are working in the **Selected Band View**.

Dynamic EQ

Table of Contents

Overview

The Dynamic EQ can be very useful in controlling specific frequencies in your mix that are too loud, with a degree of precision not possible with a static EQ.

There are three main areas of the Dynamic EQ module interface:

1. **Module Header**
2. **Spectrum View**
3. **Dynamic EQ HUD Controls**

Module Header

The module header includes the following controls:

1. **Delta**: Monitors the difference in the signal before and after the Dynamic EQ module.
2. **Global Filter Mode**: Selects the type of filter algorithm used to process all bands in the Dynamic EQ module.

    1. **Analog**: Selects minimum-phase IIR (infinite impulse response) filters.
    2. **Digital**: Selects linear-phase FIR filter types for processing in the Dynamic EQ module. FIR filters retain the phase of the original signal, but are more expensive in regard to CPU usage.

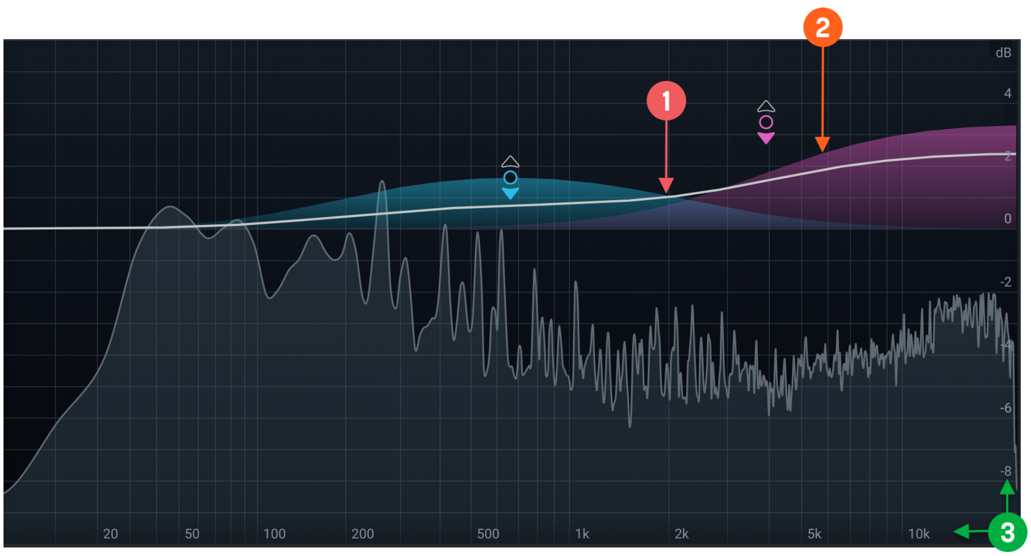
3. **Channel Processing Mode**: Selects the channel processing mode used by the Dynamic EQ.

    1. The Dynamic EQ supports **Stereo**, **Mid/Side**, **Left/Right**, and

Transient/Sustain modes.
2. See the **General Controls** chapter for more information.
4. **Reset**: Returns all controls in the module to their factory default values.

Spectrum View

The Dynamic EQ module includes a spectrum analyzer with the following features:



1. **Composite curve**: The combined filter response of all enabled bands. Represented by a thick white line drawn across the spectrum. Each band contributes to the overall shape of this curve.
2. **Filter response curve**: The filter response of the currently selected node. Represented by a highlighted line and filled area under the band which appears when you select a node. If there are no nodes

selected, the filter response curve is hidden and you will only see the composite curve.
3. **Meter Scales**: there are three different scales in the Dynamic EQ spectrum:

    1. **Dynamic EQ Gain Scale**: (dB) The filter gain scale is located on the **right** side of the spectrum view. **Range**: +15 dB to -30 dB.
    2. **Spectrum Frequency Scale**: (Hz) Horizontal scale displayed along the **bottom** of the spectrum view. **Range**: 1 Hz to 22,050 Hz.

SPECTRUM SCALES

1. The scales for the filter gain and the spectrum magnitude are different, by design. If they were made to match, you wouldn't see enough of the spectrum for it to be useful.
2. See the **Options** chapter for more information about spectrum scale options.

Alt-Solo

You can use the alt/option key when clicking on a node or anywhere in the spectrum to momentarily solo a specific frequency region. When you release the mouse click, alt-solo will be disabled.

1. **Band Solo**: Hold the **alt** or **option** key and click on an EQ node. This is equivalent to pressing the Solo button in the HUD.
2. **Alt-Solo in Spectrum**: Hold the **alt** or **option** key and click anywhere on the frequency spectrum to solo frequencies surrounding the location of the cursor. You can adjust the bandwidth of the alt-solo filter in the Options window.

Working with Dynamic EQ Nodes

The following sections describe how to add, adjust and remove Dynamic EQ nodes.

Add Nodes

1. Hover your cursor over the composite curve, click the **+** button that appears to add a new band to that frequency position.
2. Press `command+ return` (Mac) or `ctrl+ return` (Windows) to add a new node to the center of the frequency spectrum.
3. Double-click anywhere in the spectrum to add a new node at the frequency location of your cursor.

DEFAULT FILTER SHAPE ASSIGNMENT BASED ON FREQUENCY

Default filter shapes are assigned to nodes depending on their initial frequency value.

1. From 20 Hz to 100 Hz: **Baxandall Bass**. If a Baxandall Bass filter already exists in the curve, the default filter shape will be Proportional Q.
2. From 100 Hz to 8 kHz: **Proportional Q**.
3. From 8 kHz to 20 kHz: **Baxandall Treble**. If a Baxandall Treble filter already exists in the curve, the default filter shape will be Proportional Q.

Adjusting Nodes

The following methods can be used to move/adjust filter nodes:

1. Click and drag a node up and down to adjust gain. Click and drag a

node left and right to adjust frequency.
2. Hold the `shift` key while clicking and dragging a node **to lock the movement** to the horizontal axis (for frequency) or vertical axis (for gain).
3. Click and drag the handles that appear on the left/right side of a selected node to adjust the Q/Slope value.
4. Press the **up** or **down** arrow keys to adjust the gain of a selected node. Press the **left** or **right** arrow keys to adjust the frequency of a selected node.
5. Hold `shift` while using the arrow keys **to make coarse value adjustments**.
6. Hold `command` (Mac) or `ctrl` (Windows) while using the arrow keys **to make fine value adjustments**.
7. Double-click on a node to reset all band parameters to their default values.

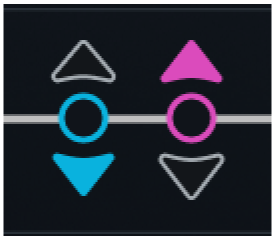
Remove Nodes

The following methods can be used to remove filter nodes:

1. Select a node and click the **X** button in the HUD to remove it.
2. Click and drag to lasso select multiple nodes. Press `delete` or `backspace` key to **remove all selected nodes**.
3. Hold `Shift` and click on individual nodes to select multiple nodes. Use the `delete` or `backspace` key to **remove all selected nodes**.

Set Dynamic Mode Direction

The arrow buttons directly above and below a node indicate the direction the filter will move when triggered. The filled arrow button indicates the currently selected direction.



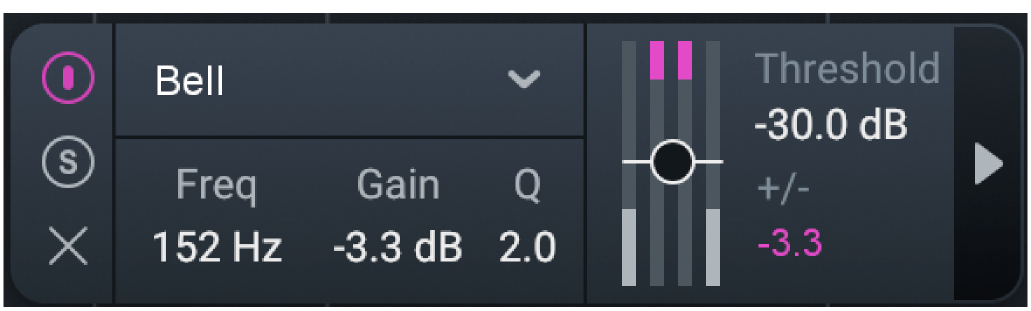
Dynamic Mode Direction

Filters will behave as follows in the different modes:

1. **UP**: When a signal exceeds the threshold, the filter will move upwards.
    1. **Positive Gain**: Using the Up trigger mode will move the filter from the center line toward the static node (boost when triggered).
    2. **Negative Gain**: Using the Up trigger mode will apply the full negative gain until it is triggered-when triggered it will move toward the center line.
2. **DOWN**: When the signal exceeds the threshold, the filter will move downwards.
    1. **Positive Gain**: Using the Down trigger mode will move the filter from the static boost value toward the center line (cut when triggered).
    2. **Negative Gain**: Using the Down trigger mode will move the filter from the center line toward the static negative gain value.

Dynamic EQ HUD Controls

Click on a node to view the HUD panel for a given band.

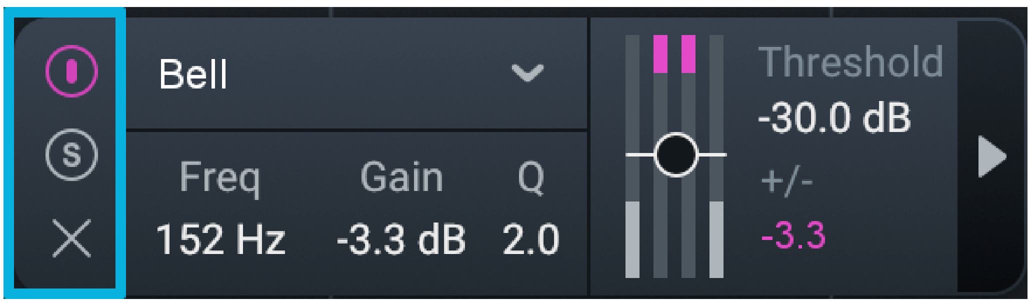


The HUD panel includes the following sections:

1. **General Controls**: Enable/disable, Solo, and Remove.
2. **Filter Controls**: Filter Shape, Frequency, Gain, and Q.
3. **Threshold Controls**: Threshold, Threshold Input meter, and gain change readout.
4. **Advanced Controls**: Auto Scale, Attack, Release, and Static Offset Gain.

General Controls

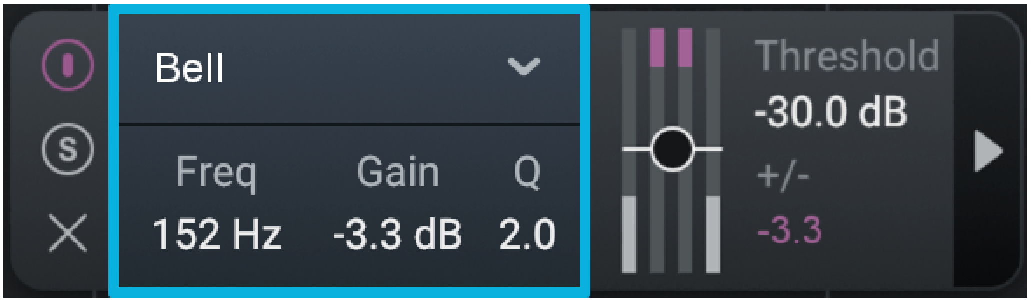
You can enable/disable, solo, or remove a band with the buttons along the left edge of the HUD.



1. **POWER**: Click the **power** button to enable or disable processing for the selected band.
2. **SOLO**: Click the **S** button to isolate playback of the selected band.
3. **REMOVE**: Click the **X** button to remove the associated band.

Filter Controls

You can adjust filter shape, frequency, gain, and Q/bandwidth in this section of the HUD.



1. **Filter Shape** Selects the filter shape for the associated band. Filter shape types include:

    1. **Baxandall**: Inspired by the Baxandall EQ, with the addition of freely adjustable frequency. Choose between **Bass** for a gentle low shelf filter or **Treble** for a gentle high shelf filter. **When to use**: Transparent way of addressing extreme lows and extreme highs for a more natural, gentle sounding effect.
    2. **Band Shelf**: Bell filter with wide, flat top. **When to use**: To change the relation between the harmonics in your audio. Useful for boosting or attenuating a block of frequencies.
    3. **Peak Bell**: Smoothly boosts or cuts an adjustable region around a specific frequency. Looks like a bell, come on what do you want from me. **When to use**: Changes the overall color or texture of the sound with larger gain adjustments (boost or cut). This will be a more noticeable change than Proportional Q.
    4. **Proportional Q**: Unique filter that varies shape in proportion to the amount of boost or cut applied. As you increase or decrease the gain, the change is proportional to bandwidth. **When to use**: Tight, precise corrective cuts-the bigger gain adjustment, the tighter the cut will become.

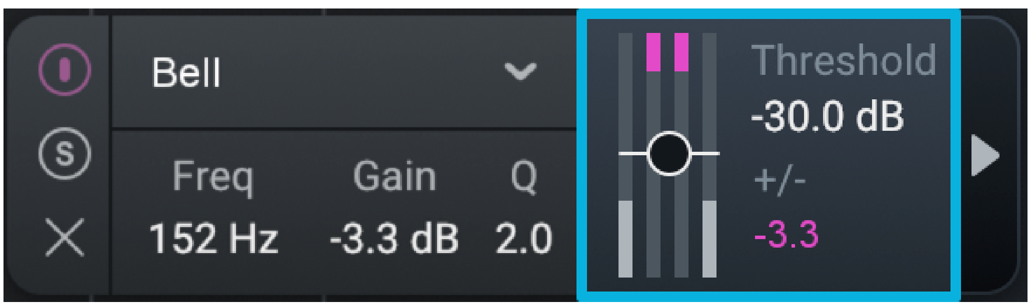
2. **Freq**: Determines the center or cutoff frequency of the selected band. Units: Hz (Hertz); Range: 20 Hz to 20 kHz.
3. **Gain**: Determines the amount of gain applied to the selected filter. Units: dB (decibels); Range: -30 dB to +15 dB.
4. **Q**: Determines the width or slope of parameteric/bell filters. Units: cF/Bandwidth.

★ **TIP: HUD VALUE ADJUSTMENTS**

1. Click and drag on a HUD text readout to scrub the value.
2. Double-click on a text readout in the HUD, type a value in the inline edit field, press enter or click outside of the field to change the value.

Threshold Controls

You can adjust the Threshold and view related meters in this section of the HUD.



1. **Threshold Input meter**: Vertical meter bars displayed behind the Threshold slider.

    1. **Input Level Meter**: Displays the input signal of the selected filter in the outer vertical meter bars.
    2. **Gain Reduction/Addition**: Displays the amount of level adjustment applied to the selected filter in the inner vertical meter bars.

2. **Threshold (dB)**: Determines the signal level at which dynamic gain adjustments will be triggered for the selected filter. **Adjustment methods**:

    1. Click and drag the slider up and down.
    2. Click on the slider handle then use the Up or Down arrow keys.
    3. Double-click on the value readout, type the value into the

inline edit field, and press enter to save it.

3. **+/-**: Displays the gain reduction/addition applied to the filter as a text based readout.

Advanced Controls

Click the arrow on the right side of the HUD to expand the Advanced panel.

The Advanced Panel includes the following controls:

1. **Auto Scale**: When enabled, the attack and release values will be scaled automatically depending on the frequency of the associated band.
2. **Attack**: Adjusts how long it takes the dynamic trigger to react to a signal crosses the Threshold value. You can adjust the Attack value by clicking and dragging on the value readout or by double-clicking on the value readout and typing a value into the inline edit field.
3. **Release**: Adjusts how long it takes for the dynamic trigger to return the filter to its static settings when the input signal falls below the Threshold value. You can adjust Release by clicking and dragging on the value readout or by double-clicking on the value readout and typing a value into the inline edit field.
4. **Offset**: Sets a static gain offset for the associated band.

Equalizer

Table of Contents

Overview

Ozone’s versatile Equalizer allows you to add warmth and character with analog-matched filters, or precisely boost and cut frequencies with digital linear-phase filters.

The module interface includes the following sections:

1. **Module Header**
2. **Spectrum and EQ Curve**
3. **Controls**

Module Header

The module header includes the following controls:

1. **Delta**: Monitors the difference in the signal before and after the Equalizer module.
2. **View Selector**: See the **Views** section below for detailed descriptions of the different views.
3. **Global Filter Mode**: Selects the type of filter algorithm used to process all bands in the EQ module.

    1. **Analog**: Selects minimum-phase IIR (infinite impulse response) filters.
    2. **Digital**: Selects linear-phase FIR filter types for processing in the EQ module. FIR filters retain the phase of the original signal, but are more expensive in regard to CPU usage.

> **DIGITAL ONLY FEATURES**
>
> **Phase** and **Surgical Filter Shapes** are only available when Digital is selected.

4. **Channel Processing Mode**: Selects the Channel Processing Mode used in the Equalizer.

    1. The Equalizer supports **Stereo**, **Mid/Side**, **Left/Right**, and **Transient/Sustain** modes.
    2. See the **General Controls** chapter for more information.

5. **Amount**: Allows you to quickly scale the Gain of all bands in the Equalizer. The Amount slider ranges from 0% to 200% and defaults to 100%. Amount does not apply Gain scaling when set to 100%.

    1. **Minimum value: 0%** (far left): Scales all band Gain controls to 0, equivalent to bypassing all bands.
    2. **Maximum value: 200%** (far right): Scales all band Gain controls to 2 times their current value.

    > **NOTES**
    >
    > 1. Amount will not scale filters (with adjustable Gain) past their maximum positive or negative gain values (+15 dB / -30 dB).
    > 2. Amount is always applied to both EQ channels in Mid/Side or Left/Right channel processing mode.

6. **Reset**: Returns all controls in the module to their factory default values.

Views

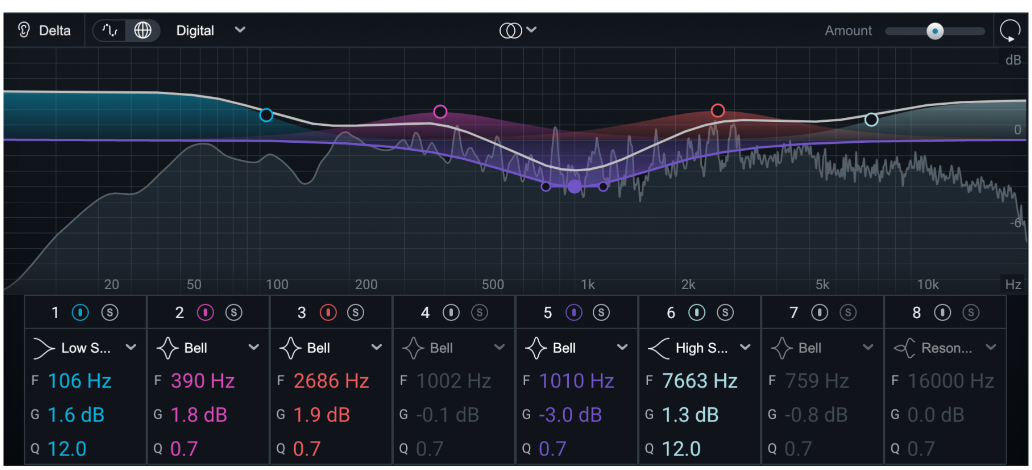
You can switch between **HUD View** and **All Bands View** using the view selection buttons.

HUD View

This is the default view in the Equalizer module. In this view, a HUD panel will appear when you select a node.

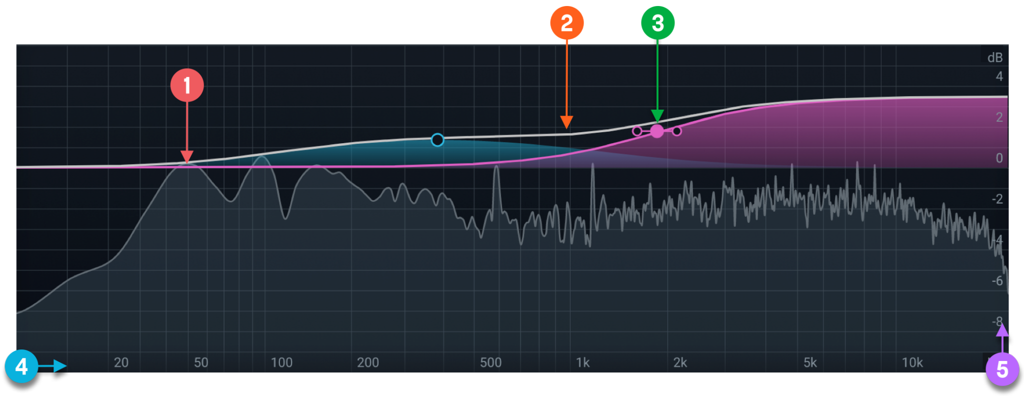
All Bands View

When selected, band controls are displayed in a table format below the spectrum. The HUD panel is not displayed when nodes are selected in All Bands View.



Spectrum and EQ Curve

The main spectrum view of the Equalizer includes the following features:



1. **Spectrum Analyzer**: Displays the magnitude (amplitude, in decibels) of a signal across the frequency spectrum in real-time. The spectrum analyzer in the Equalizer module displays the output signal of Ozone.
2. **Composite curve**: The combined filter response of all enabled bands. Represented by a thick white line drawn across the spectrum. Each band contributes to the overall shape of this curve.

> **AMOUNT CONTROL AND THE COMPOSITE CURVE**
>
> The Amount control in the **module header** will modify the shape of the white composite curve to reflect gain scaling applied to the filters. The composite curve without Amount scaling applied is shown in grey when the Amount control is adjusted.

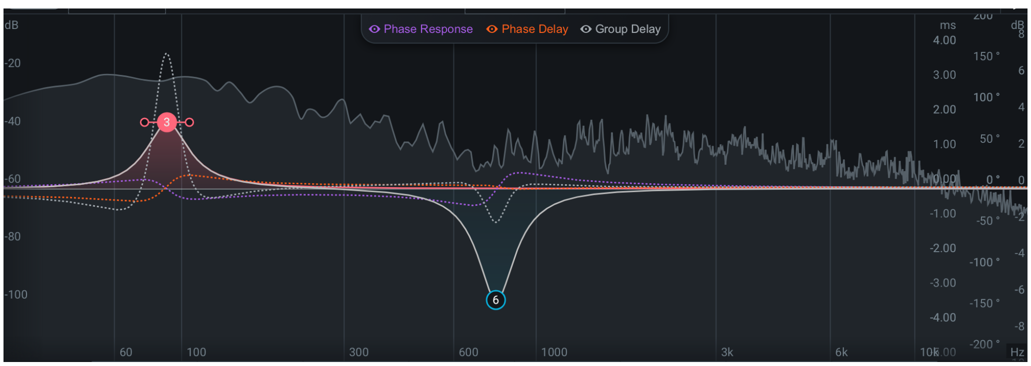
3. **Filter response curve**: The filter response of the currently selected node. Represented by a highlighted line and filled area under the band which appears when you select a node. If there are no nodes selected, the filter response curve is hidden and you will only see the composite curve.
4. **Spectrum Frequency Scale**: (Hz) Horizontal scale displayed along the **bottom** of the spectrum view. **Range**: 1 Hz to 22,050 Hz.
5. **Equalizer Gain Scale**: (dB) The filter gain scale is located on the **right** side of the spectrum view. **Range**: +15 dB to -30 dB.

SPECTRUM SCALES

1. The scales for the filter gain and the spectrum magnitude are different, by design. If they were made to match, you wouldn't see enough of the spectrum for it to be useful.
2. See the **Options** chapter for more information about spectrum scale options.

Extra Curves

When the **Show Extra Curves** option is enabled in the EQ options tab, three additional rulers appear on the right-hand side of the spectrum analyzer: **Phase Response** (degrees), **Phase Delay** (ms), **Group Delay** (ms).



You can hide and show each curve individually by clicking on the buttons that appear in the legend along the top of the spectrum view.

1. **Phase Delay**: A calculation of phase response represented in time (ms).
2. **Phase Response**: A calculation of phase response represented in degrees. This curve is most useful when using analog or minimum phase equalization.
3. **Group Delay**: A calculation of the delay of amplitude envelopes in time (ms). This curve is most useful when working with transients.

Alt-Solo

You can use the alt/option key when clicking on a node or anywhere in the spectrum to momentarily solo a specific frequency region. When you release the mouse click, alt-solo will be disabled.

1. **Band Solo**: Hold the **alt** or **option** key and click on an EQ node. This is equivalent to pressing the Solo button in the HUD.
2. **Alt-Solo in Spectrum**: Hold the **alt** or **option** key and click anywhere on the frequency spectrum to solo frequencies surrounding the location of the cursor. You can adjust the bandwidth of the alt-solo filter in the Options window.

Working with EQ Nodes

The following sections describe how to add, adjust and remove EQ nodes.

1. **Adding Nodes**
2. **Adjusting Nodes**
3. **Removing Nodes**

Adding Nodes

1. Hover your cursor over the composite curve, click the **+** button that appears to add a new band to that frequency position.
2. Press `command` + `return` (Mac) or `ctrl` + `return` (Windows) to add a new node to the center of the frequency spectrum.
3. Double-click anywhere in the spectrum to add a new node at the frequency location of your cursor.

DEFAULT FILTER SHAPE ASSIGNMENT BASED ON FREQUENCY

Default filter shapes are assigned to nodes depending on their initial frequency value.

1. From 20 Hz to 100 Hz: **Baxandall Bass**. If a Baxandall Bass filter already exists in the curve, the default filter shape will be Proportional Q.
2. From 100 Hz to 8 kHz: **Proportional Q**.
3. From 8 kHz to 20 kHz: **Baxandall Treble**. If a Baxandall Treble filter already exists in the curve, the default filter shape will be Proportional Q.

Adjusting Nodes

The following methods can be used to move/adjust filter nodes:

1. Click and drag a node up and down to adjust gain. Click and drag a node left and right to adjust frequency.
2. Hold the `shift` key while clicking and dragging a node **to lock the movement** to the horizontal axis (for frequency) or vertical axis (for gain).
3. Click and drag the handles that appear on the left/right side of a selected node to adjust the Q/Slope value.
4. Press the **up**/**down** arrow keys to adjust the gain of a selected node. Press the **left**/**right** arrow keys to adjust the frequency of a selected node.
5. Hold `shift` while using the arrow keys **to make coarse value adjustments**.
6. Hold `command` (Mac) or `ctrl` (Windows) while using the arrow keys **to make fine value adjustments**.
7. Double-click on a node to reset all band parameters to their default values.

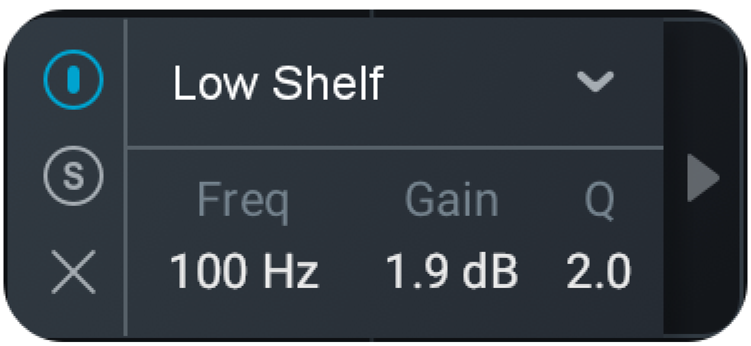
Removing Nodes

The following methods can be used to remove filter nodes:

1. **In the HUD view**: Select a node and click the **X** button in the HUD to remove it.
2. **In All Bands View**: Click the power button associated with the band you want to remove in the controls section.
3. Click & drag to select multiple nodes. Press `delete` or `backspace` key to **remove all selected nodes**.
4. Hold `Shift` and click on individual nodes to select multiple nodes. Use the `delete` or `backspace` key to **remove all selected nodes**.

Controls

The following sections detail the EQ filter controls in the context of the **HUD View**. Any **All Bands View** differences or exceptions are noted in the control descriptions below.

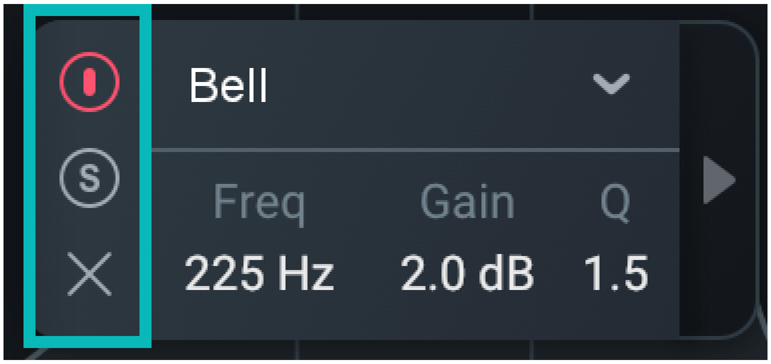


The HUD panel is divided into the folowing sections:

1. **General Controls**: Enable/disable, Solo, and Remove.
2. **Filter Controls**: Filter Shape, Frequency, Gain, and Q/Slope.
3. **Advanced Controls**: Phase. *Digital Mode only.*

General Controls

You can enable/disable, solo, or remove a band with the buttons along the left edge of the HUD.



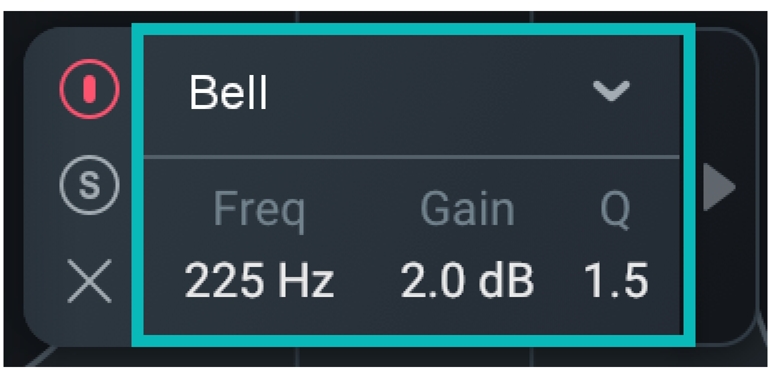
1. **Power**: Click the **power** button to enable or disable processing for the selected band.

   > **POWER BUTTON BEHAVIOR IN ALL BANDS VIEW**
   >
   > When All Bands View is selected, clicking on a band power button will **remove** the associated band, rather than disable it.

2. **Solo**: Click the **S** button to isolate playback of the selected band.
3. **Remove**: Click the **X** button to remove the associated band.

Filter Controls

You can adjust filter shape, frequency, gain, and Q/bandwidth in this section of the HUD.



Filter Shape

Selects the filter shape for the associated band. The filter shapes are organized into sub-menus based on filter type and usage.

1. **BELL menu**: These filters can be used to boost or cut a specific center frequency level (*Peak type filters*). The following filter shapes are included in this menu:

    1. **Bell**: Smoothly boosts or cuts an adjustable region around a specific frequency.
    2. **Proportional Q**: Unique filter that varies shape in proportion to the amount of boost or cut applied. As a cut or boost is increased further away from center of the EQ curve, the shape tightens for more precision.
    3. **Band Shelf**: Bell filter with wide, flat top. Useful for boosting or attenuating a block of frequencies.

2. **LOW SHELF and HIGH SHELF menus**: These filters can be used to boost or cut the frequency content above or below a specified frequency (*Shelf type filters*). The following filter shapes are included in the sub-menu:

    1. **Analog**: Efficient shelf filter for simple boosts and cuts.
    2. **Baxandall**: Gentle shelf filters. Modeled after the Baxandall EQ, with the addition of adjustable frequency.
    3. **Vintage**: Inspired by the renowned Pultec analog equalizer. Exhibits a complementary frequency dip, creating a complex slope with one node.
    4. **Resonant**: Exhibits a complimentary resonance at both ends of the filter slope creating a complex shape with one node.

3. **LOWPASS and HIGHPASS menus**: These filters can be used to attenuate frequency content that is below (for highpass) or above (for lowpass) a specified cutoff frequency (*Pass type filters*). The following filter shapes are included in this sub-menu:

    1. **Flat**: Butterworth filter; optimized for maximum flatness without ripple or resonance in the passband or stopband (stability).
    2. **Resonant**: Filter equipped with a resonance control to emphasize the cutoff frequency with positive gain. Boosts

content at the cutoff frequency to add character and emphasize the lowest or highest part of the signal.
    3. **Brickwall**: Elliptic filters. Optimized for steepness with minimal ripple in the passband and stopband.
4. **SURGICAL**:*(**Digital** mode only)* Precise digital filter shape available in all filter shape sub-menus *when Digital Filter Mode is selected in the module header.*

Frequency

Determines the center or cutoff frequency of the selected band. Units: Hz (Hertz); Range: 20 Hz to 20 kHz.

Gain

Determines the amount of gain applied to the selected filter. Units: dB (decibels); Range: -30 dB to +15 dB.

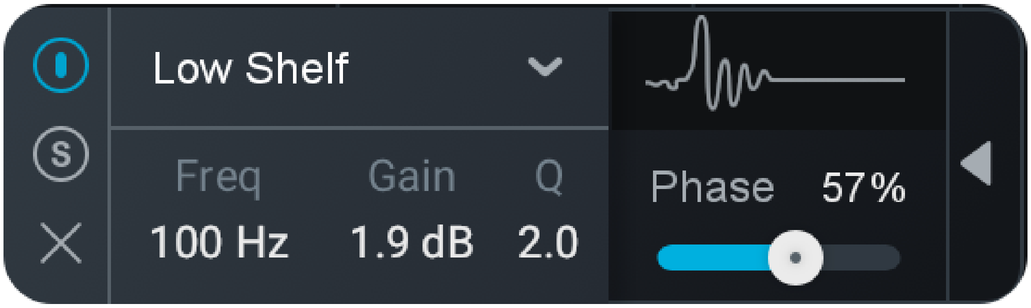
Q/Slope

1. **Q**: Determines the bandwidth (cF) of the selected filter.
2. **Slope**: Determines the **Slope** of the filter below or above the set cutoff frequency. Slope is measured in decibels per octave (dB/oct).

> **FIXED Q/SLOPE**
>
> The **Baxandall Bass**, **Baxandall Treble**, **HP Brickwall** and **LP Brickwall** filter shapes do not have adjustable Q/Slope values.

Advanced Controls

You can access the Advanced panel by clicking the arrow button on the right hand side of the HUD. If the **Global Filter Mode** is set to **Analog**, the Advanced panel button will be **disabled**. You can only access the Advanced panel of the HUD when the **Global Filter Mode** is set to **Digital**.



Phase

Adjusts the phase response of the currently selected filter.

1. **0% Phase**: the selected band will have a Linear phase response.
2. **100% Phase**: the selected band will have a Minimum phase response.

PHASE CONTROL AVAILABILITY

1. Phase is only adjustable when Digital Global Filter Mode is selected.
2. Phase is not included in the controls table of the **All Bands View**. You can only access Phase from the Advanced panel of the HUD.

Exciter

Table of Contents

Overview

Ozone’s Exciter module offers up to four bands of configurable saturation with numerous modes giving you the ability to customize how saturation is introduced into your music.

1. **Module Header**
2. **Views**
3. **Controls**

Module Header

The Exciter module header includes the following controls:

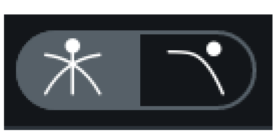
1. **Delta**: Monitors the difference in the signal before and after the Exciter module.
2. **View Selector**: See the **Views** section below for detailed descriptions of the different meter views.
3. **Channel Processing Modes:** Selects the channel processing mode used by the Exciter.

    1. The Exciter supports **Stereo**, **Mid/Side**, and **Transient/Sustain** modes.
    2. See the **General Controls** chapter for more information.

4. **Learn**: Enables automatic crossover point placement. When active, crossover points will be moved to minima detected in the frequency spectrum of your track. When crossovers have been set to their ideal frequency values, Learn will automatically disable itself. **Note**: Learn is not available in single band processing mode.
5. **Reset**: Returns all module controls to their default values.

Views

You can toggle between the different Exciter views using the view selector buttons in the module header area.

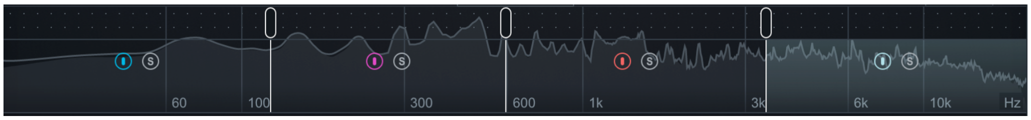


The Exciter module includes the following views:

1. **Crossover Spectrum**
2. **Post Filter**

Crossover Spectrum

All multiband modules in Ozone support up to four processing bands. All multiband capable modules are single band by default. You can create new processing bands and manage multiband crossovers in the Crossover Spectrum view. **Note**: crossover cutoff frequencies are not shared or linked across multiband modules in the main Ozone plug-in.

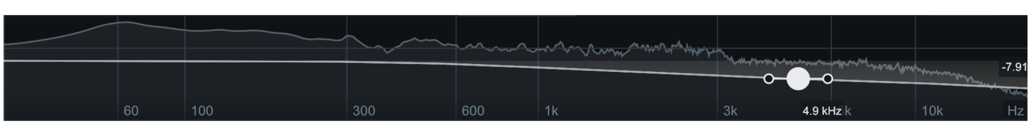


Adjusting Crossovers

1. **Add band**: Hover your cursor over the crossover spectrum view and click the **+** button to add a new crossover point.
2. **Enable/disable band**: Click the **power** button to enable/disable processing in the associated band.
3. **Solo band**: Click the **S** button to isolate playback of the associated band.
4. **Remove band**: Hover over a band in the crossover view and click the **x** that appears to remove the band.
5. **Adjust crossover cutoffs**: Click and drag a handle to adjust the crossover frequency. You can also double-click on a crossover handle and enter the value manually in the inline edit field that appears.

Post Filter

Displays harmonic highlights that represent the saturation being applied to your signal across the frequency spectrum. You can adjust a high-shelf filter that affects the wet (processed) output of the module to tame any high frequency content that may have been introduced by the module's processing.



The affected frequencies on the spectrum are indicated by a bright white fill. Non-affected frequencies on the spectrum will not display a fill color. The histogram display is calculated after the Post Filter is applied. Adjusting the Post Filter will update the histogram display to show the saturation occurring at the output of the Exciter module.

Controls

The Exciter module includes the following controls:

Oversampling

Increases the sampling rate of the applied distortion to reduce aliasing. **This option applies to all bands.**

> **OVERSAMPLING AND CPU**
>
> Oversampling utilizes more processing power to increase the quality level of the Exciter module.

Link Bands

When enabled, adjustments made to a control will be linked and applied to all other bands by the same amount.

Modes

Select one of the egg cider’s different modes to find the sonic characteristic that fits your in your mix. You can choose between the following Mode options:

1. **Analog**: Emulates the sound of transistor type odd harmonics giving a driven grit to your audio.
2. **Retro**: Based on characteristics of transistors, with a slowly decaying row of odd harmonics.
3. **Tape**: Offers a brighter sounding saturation, due to the odd harmonics found when saturating analog tape.
4. **Tube**: Characterized by its clear “tonal” excitation with an emphasis on dynamic or transient attacks.
5. **Warm**: Generates only even harmonics that decay quickly.
6. **Triode**: Modeled after a tube circuit for realistic analog warmth. It uses one half of a tube circuit for a subtler overdrive than the Dual Triode mode.
7. **Dual Triode**: Models a full circuit using a vacuum tube, introducing more pronounced overdrive with a warmer tone.

Amount

Controls the amount of the harmonic excitation for the associated band.

Mix

Controls the amount of processed signal that is blended back into the unprocessed signal.

Imager

Table of Contents

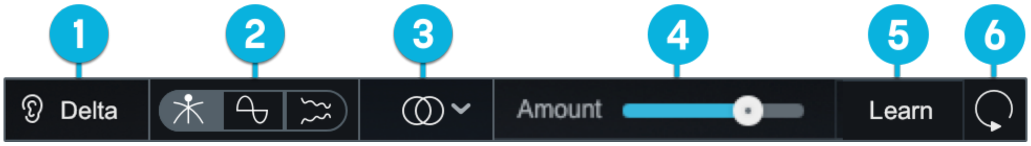
Overview

The Imager allows you to adjust the stereo width of your mix using multiband stereo imaging. The Imager module interface is divided into four main sections:

1. **Module Header**
2. **Views**
3. **Controls**
4. **Vectorscope**

Module Header

The Imager module header includes the following controls:



1. **Delta**: Monitors the difference in the signal before and after the Imager module.
2. **View Selector**: See the **Views** section below for detailed descriptions of the different meter views.
3. **Channel Processing Mode**: Selects the Channel Processing Mode used in the Imager module.

    1. The Imager module supports **Stereo**, or **Transient/Sustain** mode.
    2. See the **General Controls** chapter for more information.

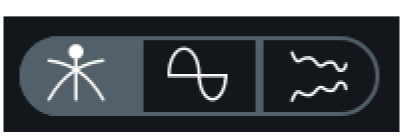
4. **Amount**: Allows you to quickly scale the Width of all bands in the Imager. The Amount slider ranges from 0% to 100% and defaults to 100%.

    1. **Minimum value: 0%** (far left): Scales all band Width controls to 0, equivalent to bypassing all bands.
    2. **Maximum value: 100%** (far right): Scales all band Width controls to their current value.

5. **Learn**: Enables automatic crossover point placement. When active, crossover points will be moved to minima detected in the frequency spectrum of your track. When crossovers have been set to their ideal frequency values, Learn will automatically disable itself. **Note**: Learn is not available in single band processing mode.
6. **Reset**: Returns all module controls to their default values.

Views

You can toggle between the different Imager views using the view selector buttons in the module header area.



The Imager module includes the following views:

1. **Crossover Spectrum**
2. **Correlation Trace**
3. **Stereo Width Spectrum**

Crossover Spectrum

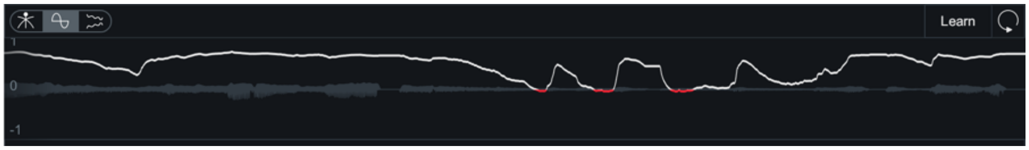
All multiband modules in Ozone support up to four processing bands. You can create new processing bands and manage multiband crossovers in the Crossover Spectrum view. **Note**: crossover cutoff frequencies are not shared or linked across multiband modules in the main Ozone plug-in.

Adjusting Crossovers

1. **Add band**: Hover your cursor over the crossover spectrum view and click the **+** button to add a new crossover point.
2. **Enable/disable band**: Click the **power** button to enable/disable processing in the associated band.
3. **Solo band**: Click the **S** button to isolate playback of the associated band.
4. **Remove band**: Hover over a band in the crossover view and click the **x** that appears to remove the band.
5. **Adjust crossover cutoffs**: Click and drag a handle to adjust the crossover frequency. You can also double-click on a crossover handle and enter the value manually in the inline edit field that appears.

Correlation Trace

Draws stereo correlation values over time. Negative correlation (out of phase) values are drawn in red; positive correlation (in phase) values are drawn in white.



The correlation trace displays information based on the output signal of Ozone, so adjustments to the Imager controls are reflected in this meter.

Stereo Width Spectrum

Displays a hybrid mirrored spectrum analyzer of the signal's stereo width.

When the signal is fully mono, only a straight line will be drawn in the center of this view.

Controls

The Imager module includes the following controls:

1. **Width**
2. **Link Bands**
3. **Stereoize**
4. **Recover Sides**

Width

Adjusts the amount of gain applied to side channel content. Positive values will increase perceived stereo width, and negative values will decrease the perceived stereo width of a band. A setting of -100 will make the output of the associated band effectively mono.

> **SCALED IMAGER SLIDERS**
>
> These sliders can be scaled by the Amount control in the header. When scaled to less than 100%, the fill of the slider will be reduced to reflect this.

> **NOTE**
>
> If the input signal to the Imager is either: mono (i.e. single channel) or *effectively* mono (i.e. a stereo file with the same content on both channels), you will need to enable the Stereoize feature for Width adjustments to have an effect on the signal.

Link Bands

Enable to link Width controls adjustments across all bands. When enabled and a Width control is adjusted for one band, all other Width controls will be adjusted by the same amount.

Stereoize

Stereoize processing is applied to all processing bands equally. Stereoize is only available when the **Width** of at least one band is being increased. The following controls allow you to enable the effect and adjust the associated settings.

1. **Stereoize power button**: Enables/disables stereoize processing.
2. **Stereoize Amount**: Adjust the slider to add natural-sounding stereo width to narrow recordings and to control the character of the stereo effect in conjunction with the width sliders.

   > **STEREOIZE IS MONO COMPATIBLE**
   >
   > The Stereoize effect is completely mono compatible. Even if you add width to audio, it can still be played back in mono without producing unpleasant artifacts.

3. **Stereoize Modes:** There are two different stereoize processing modes available in the Imager:

   1. **Mode I**: Haas Effect-based decorrelation processing. This mode creates a delayed copy of the mid channel signal and injects it into the side channel.
   2. **Mode II**: A newly developed alternative to the classic stereoize mode. This new mode has a slightly different tonal quality from the original and helps to preserve transients at higher settings.

Recover Sides

Recover Sides processing is applied to all processing bands equally. Recover Sides is only available when the **Width** of at least one band is being decreased. The following controls allow you to enable the effect and adjust the associated settings.

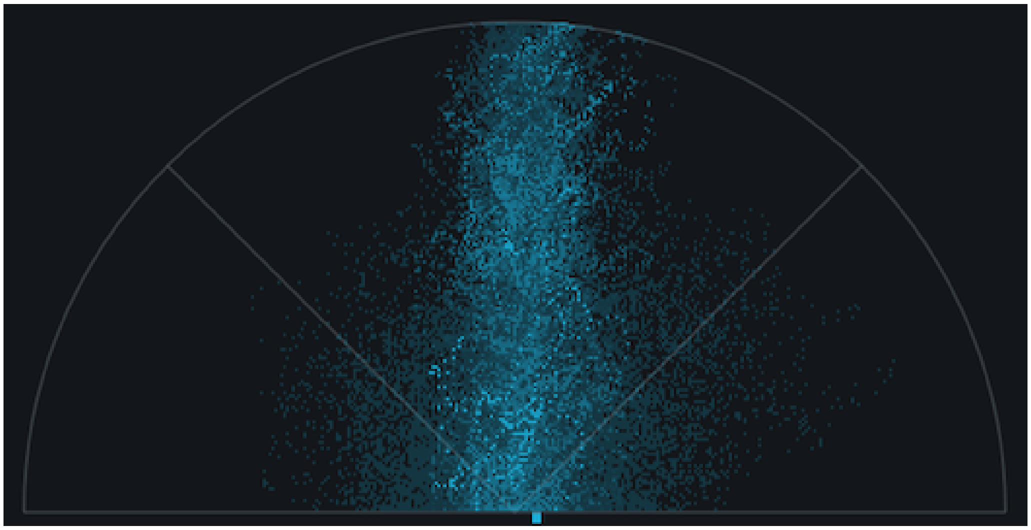
1. **Recover Sides power button**: Enables/disables recover sides processing.
2. **Recover Sides gain**: Applies a gain offset to the recovered side signal.
3. **Recover Sides solo**: Monitors only the side channel information that is being recovered and added to the middle of the stereo image.

Vectorscope

The vectorscope meters provide a view of the stereo image of the signal after all other processing is applied, regardless of the Imager's position in the signal chain.

Polar Sample

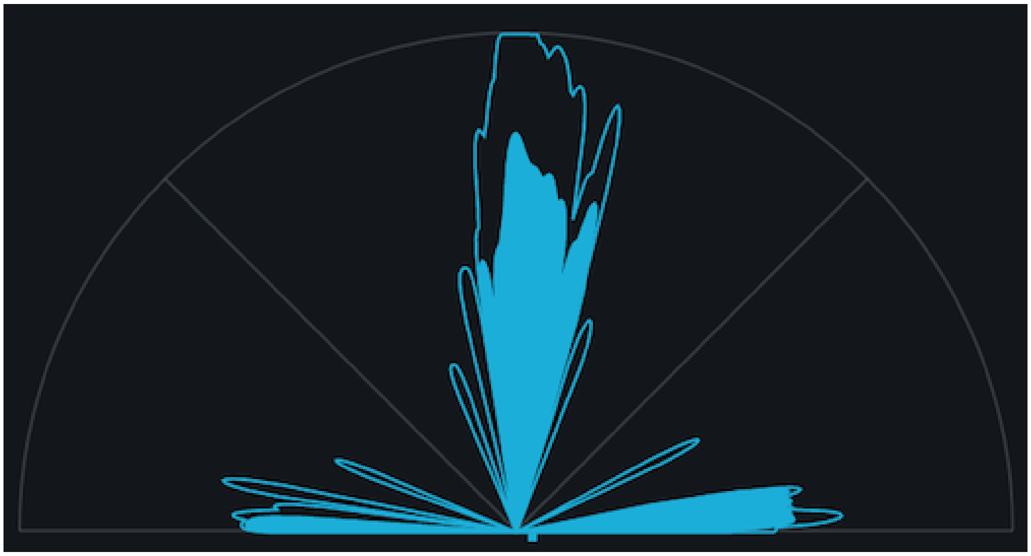
Plots dots per sample on a polar coordinate display to highlight the stereo image of the incoming signal.



Patterns that appear within the 45-degree safe lines represent **in-phase** audio. Patterns outside these lines represent **out-of-phase** audio. The history of the Polar Sample Vectorscope fades out slowly. You can reset the display by clicking on the meter.

Polar Level

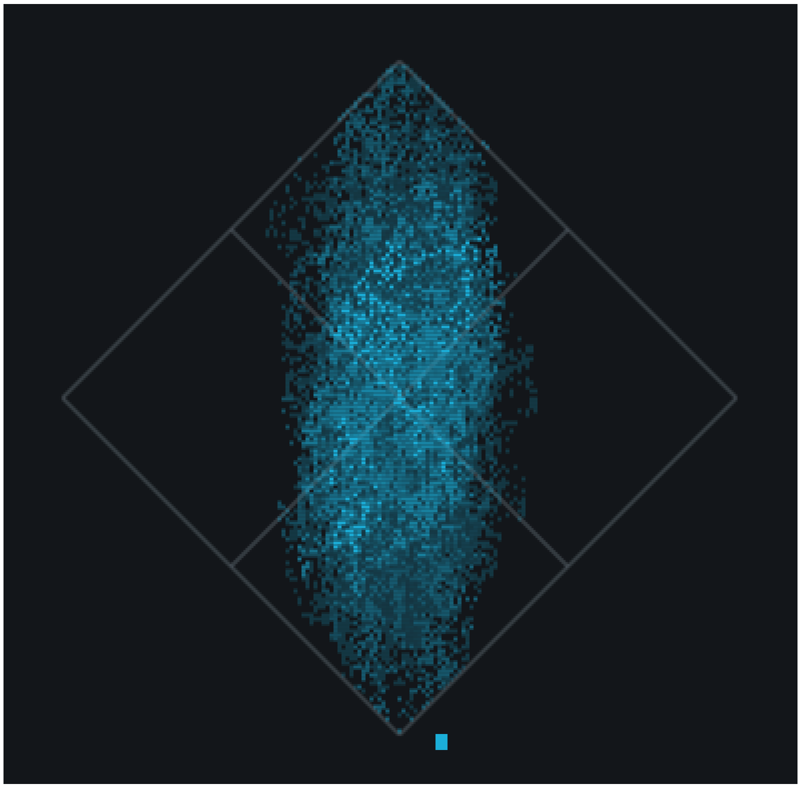
Plots rays on a polar coordinate display that represent sample averages and displays the stereo energy of a recording.



The length of the rays represents *amplitude*. The angle of the rays represents their position in the *stereo image*. Rays within the 45-degree safe lines represent **in-phase** audio. Rays beyond the safe lines represents **out-of-phase** audio.

Lissajous

Plots per-sample dots on a traditional oscilloscope display similar to the Polar Sample vectorscope.



Typically, stereo recordings produce a random pattern on a Lissajous Vectorscope that is taller than it is wide. **Vertical patterns** mean left and right channels are similar (approaching mono). **Horizontal patterns** mean the two channels are very different, which could result in mono compatibility problems.

> **CLIPPING IN THE VECTORSCOPE**
>
> Clipped samples are drawn in **red** along the edges of the vectorscope meter frame. Click on the meter to reset the clipped samples display.

Correlation Meter Bar

The vertical bar meter to the right of the vectorscope indicates the degree of similarity (or correlation) between the left and right channels.

TIPS

1. If the left and right channels are exactly the same, the correlation value would be +1 and the meter tick would be drawn at the top.
2. If the left and right channels are exactly out-of-phase the correlation would be -1 and the meter tick would be drawn all the way at the bottom.
3. In general, most recordings have phase correlations in the 0 to +1 range. A brief readout towards the bottom half of the meter is not necessarily a problem but could represent a possible mono compatibility issue.
4. As you increase stereo widening, the phase correlation will tend to draw more towards the bottom half of the meter, as the left and right channels will become “wider” and less similar.

Stereo Balance Meter

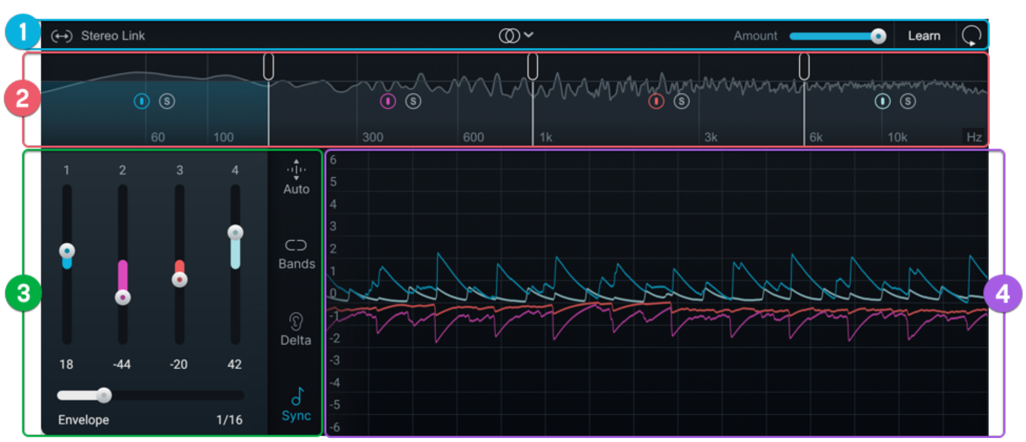
The horizontal bar meter directly below the vectorscope meter indicates the stereo balance of the signal. When the meter tick is drawn all the way to the left, it indicates there is only information on the left channel. When the meter tick is drawn all the way to the right, it indicates that there is only information on the right channel.

Impact

Table of Contents

Overview

The Impact module directly controls the microdynamics of a signal, enabling dynamic range expansion and compression with a single, intuitive control.



1. **Module Header**
2. **Crossover Spectrum**
3. **Controls**
4. **Gain Traces**

Module Header

The Impact module header includes the following controls:

1. **Stereo Link**: When enabled, Impact will apply the same adjustment to the left and right channels based on the microdynamics of the mid signal. In the default, unlinked state, Impact will adjust the microdynamics of mix elements in their specific location in the stereo field.
2. **Channel Processing Modes:** Selects the channel processing mode used by the Impact module.

    1. Impact supports **Stereo** and **Mid/Side** mode.
    2. See the **General Controls** chapter for more information.

3. **Amount**: Allows you to quickly scale the Impact of all bands. The Amount slider ranges from 0% to 100% and defaults to 100%.

    1. **Minimum value: 0%** (far left): Scales all band Impact controls to 0, equivalent to bypassing all bands.
    2. **Maximum value: 100%** (far right): Scales all band Impact controls to their current value.

4. **Learn**: Enables automatic crossover point placement. When active, crossover points will be moved to minima detected in the frequency spectrum of your track. When crossovers have been set to their ideal frequency values, Learn will automatically disable itself
5. **Reset**: Returns all module controls to their default values.

Crossover Spectrum

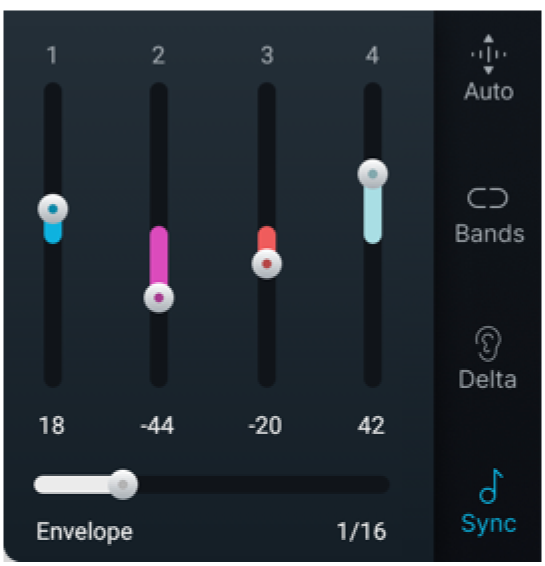
All multiband modules in Ozone support up to four processing bands. You can create new processing bands and manage multiband crossovers in the Crossover Spectrum view. **Note**: crossover cutoff frequencies are not shared or linked across multiband modules in the main Ozone plug-in.

Adjusting Crossovers

1. **Add band**: Hover your cursor over the crossover spectrum view and click the **+** button to add a new crossover point.
2. **Enable/disable band**: Click the **power** button to enable/disable processing in the associated band.
3. **Solo band**: Click the **S** button to isolate playback of the associated band.
4. **Remove band**: Hover over a band in the crossover view and click the **x** that appears to remove the band.
5. **Adjust crossover cutoffs**: Click and drag a handle to adjust the crossover frequency. You can also double-click on a crossover handle and enter the value manually in the inline edit field that appears.

Controls

The Impact module includes the following controls:



Impact

Adjusts the amount of expansion or compression applied to the microdynamics of the signal. Positive values will expand microdynamics for a more open, punchy sound, and negative values will compress microdynamics for a more dense, glued sound.

> **SCALED IMPACT SLIDERS**
>
> These sliders can be scaled by the Amount control in the header. When scaled to less than 100%, the fill of the slider will be reduced to reflect this.

Envelope

Adjusts the amount of time for expansion or compression to return to baseline after a microdynamic event. This control can be set in milliseconds or beat divisions synced to the host tempo when Sync is enabled.

Auto-gain

When enabled, make-up gain is automatically calculated and applied to the output signal to compensate for level differences introduced by Impact processing. The automatic gain control calculates the RMS levels of the input and output signals independently for each Impact crossover band. Gain is automatically applied to the output signal based on the RMS level difference between the input and output signals. When disabled, the Impact gain traces will return to 0dB. When enabled, the “center of mass” of the Impact gain traces will remain at 0dB.

Link Bands

Enable to link Impact amount adjustments across all bands. When enabled and an Impact control is adjusted for one band, all other Impact controls will be adjusted by the same amount.

Delta

Monitors the difference in the signal before and after the Impact module.

Sync

Enable to set Envelope times with beat divisions synced to the host tempo.

Gain Traces

Displays a scrolling trace showing the microdynamic adjustment being applied to all active bands. When adjusting the Impact amount of a particular band, the other band's gain traces will be dimmed to more easily view the changes you are making.

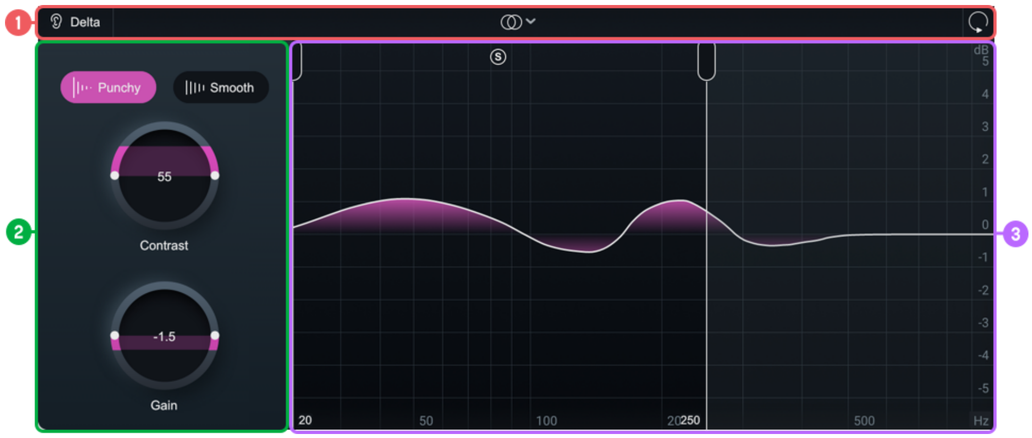
Low End Focus

Table of Contents

Overview

Low End Focus is designed to intelligently reduce muddiness, increase low end impact, and address other common low end issues that may not be easily improved with traditional tools such as EQ or Dynamics.

The module interface is comprised of three main sections:



1. **Module Header**
2. **Controls**
3. **Difference Meter and Action Region**

Module Header

The module header includes the following controls:

1. **Delta**: Monitors the difference in the signal before and after the Low End Focus module.
2. **Channel Processing Mode**: Selects the Channel Processing Mode used in the Low End Focus module.

    1. The Low End Focus module supports **Stereo**, **Mid/Side**, or **Transient/Sustain** mode.
    2. See the **General Controls** chapter for more information.

3. **Reset**: Returns all module controls to their default values

---

Controls

Low End Focus includes the following controls:

Modes

Determines the response time of Low End Focus processing. These modes influence how processing is applied to sustained and transient content within the action region. You can choose between the following modes:

1. **Punchy**: Uses faster response times to emphasize transient content.
2. **Smooth**: Uses slower response times to enhance sustained content.

Contrast

Adjusts the spectral contrast between low and high level signals within the action region.

1. **Positive values**: Increases the difference between low and high level content. Attenuates ‘out-of-focus’ low level content, bringing out transients and leading to a punchier sound.
2. **Negative values**: Decreases the difference between low and high level content. Blurs the low end in a similar way to saturation. Transients will lose focus similar to the smoothing effect of some analog compression.

Gain

Sets the amount of makeup gain applied to the action region.

Difference Meter and Action Region

Displays the tonal changes that the Low End Focus processing is applying. This is similar to an EQ where frequency is measured from left to right and gain is up and down. Different from an EQ, Low End Focus will adapt its spectral contour over time.

This area also includes the action region cutoff handles that will focus the processing to a particular frequency range. Low End Focus allows for an action region range of 20 Hz to 300 Hz.

1. **Action Region Cutoff Adjustments**: Drag a handle to adjust a single cutoff value, click between the two handles to move them together, or single-click on the value readout and manually enter a cutoff value.
2. **Solo**: Isolates playback of the input signal (before processing) within the action region cutoffs.

Master Rebalance

Table of Contents

Overview

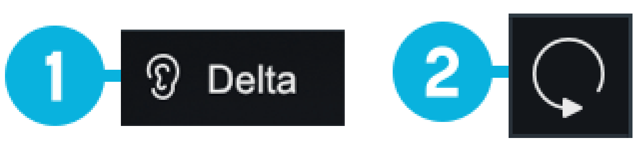
Master Rebalance leverages a machine learning algorithm trained to identify different instrument types in a track. You can use Master Rebalance to make real-time gain adjustments to a selected focus element in your master without needing to adjust stems or individual tracks from the original mix.

The Master Rebalance module is divided into the following sections:

1. **Module Header**
2. **Spectrum View**
3. **Controls**

Module Header

The module header includes a Reset control:



1. **Delta**: Monitors the difference in the signal before and after the Master Rebalance module.
2. **Reset**: Returns all module controls to their default values.

Spectrum View

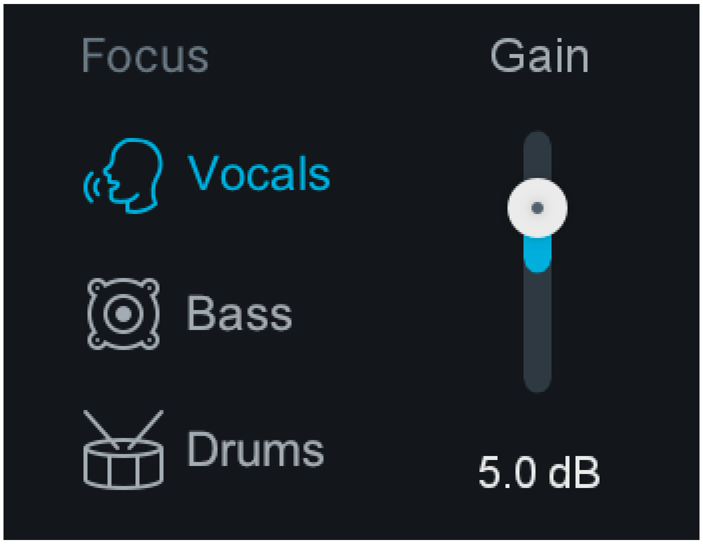
The Master Rebalance spectrum view differs from the spectrum metering included in other Ozone modules. The following image outlines the key components of the Master Rebalance spectrum view:

1. **Focus spectrum**: The blue ‘focus’ spectrum analyzer displays only the signal content identified as belonging to the selected **Focus**

element. Any gain adjustments made to the Focus will be reflected in the blue Focus spectrum analyzer.
2. **Residual spectrum**: The gray residual spectrum analyzer displays any signal content that is **not** identified as belonging to the selected **Focus** element. **For example**: if Vocals is selected as the Focus element, the gray residual spectrum will display any signal content that is not classified as Vocals.

Controls

The following controls are included in the Master Rebalance module:



Focus

Selects the focus element (instrument type) that will be detected and adjusted. The following focus options are available: **Vocals**, **Bass**, and **Drums**.

> **FOCUS SELECTION**
>
> Only one Focus mode can be selected at a time.

Gain

Adjusts the level of the selected focus element.

Match EQ

Table of Contents

Overview

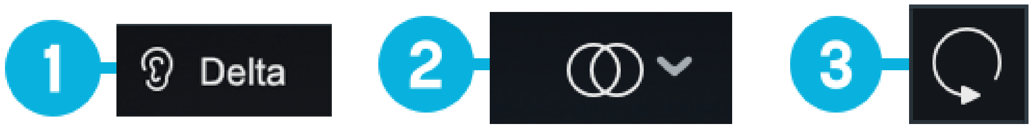
The Match EQ module is a digital linear-phase EQ with the ability to use over 8,000 bands of frequencies for very precise matching.

The module interface includes the following sections:

1. **Match EQ Module Header**
2. **Spectrum and Matched EQ Curve**
3. **Reference Spectrum Snapshot**
4. **Apply To Spectrum Snapshot**
5. **Fine Tune**

Match EQ Module Header

The module header includes the following controls:



1. **Delta** Monitors the difference in the signal before and after the Match EQ module.
2. **Channel Processing Modes:** Determines the channel processing mode for the module.

    1. Match EQ supports: **Stereo**, **Mid/Side**, **Left/Right**, and **Transient/Sustain** modes.
    2. See the **General Controls** chapter for more information.

3. **Reset**: Returns all module controls to their default values.

> **NOTE**
>
> The **Reference** and **Apply To** curves are not cleared by pressing Reset.

Spectrum and Matched EQ Curve

The spectrum view includes the following features:

1. **Spectrum Analyzer (grey)**: Displays the magnitude (amplitude, in decibels) of a signal across the frequency spectrum in real-time. The spectrum analyzer in the Match EQ module displays the output signal of Ozone.
2. **Matched curve (white)**: The filter response of the matched curve. Represented by a thick white line drawn across the spectrum. This is only displayed after a Reference and Apply To snapshot have been captured.
3. **Reference Spectrum Snapshot (goldenrod)**: Display of the captured Reference snapshot.
4. **Apply To Spectrum Snapshot (blue)**: Display of the captured Apply To snapshot.
5. **Matched Curve Cutoff handles**: Sets low and high frequency

cutoffs for the matched curve processing.

1. Drag the cutoff handles to set lower and upper range boundaries for the matched curve processing. As you drag a cutoff toward the middle of the spectrum, the matched curve will flatten out to indicate no processing is being applied.
2. Click on the value readouts for the cutoff handles to manually enter a frequency value in the inline edit field that appears.
3. These can be useful when you want to concentrate on matching a particular region of the frequency spectrum. For example: apply the matched curve to the low end without affecting the high end.

6. **Spectrum Scales**:

    1. **Spectrum Magnitude Scale**: (dB) Vertical scale displayed on the **left** side of the spectrum view.
    2. **Spectrum Frequency Scale**: (Hz) Horizontal scale displayed along the **bottom** of the spectrum view.
    3. **Matched Curve Gain Scale**: (dB) The gain scale for the matched filter curve is located on the **right** side of the spectrum view.

Reference Spectrum Snapshot

Capture a spectrum snapshot of a reference or source track that you want to match your track to.

1. **Capture**: Click and playback the track you want to use as a reference curve to begin recording a snapshot.

    > ★ **CAPTURE REFERENCE PANEL PLAYBACK**
    >
    > If you want to use a track loaded in the **Reference** panel, enable Reference playback in the I/O panel and capture the Reference track directly from the Reference panel.

2. **Stop**: Click to stop capturing and freeze the Reference spectrum snapshot.
3. **Clear**: Clears the current Reference spectrum snapshot.

SNAPSHOTS AND PRESETS

1. **Reference** Spectrum snapshots **are** saved with Ozone presets.
2. **Apply To** spectrum snapshots **are not** saved with with Ozone presets.
3. Loading a preset with a saved Reference snapshot will overwrite any currently captured Reference snapshot with the snapshot saved with the preset.
4. Loading presets will not affect the Apply to spectrum snapshot.

Apply To Spectrum Snapshot

Capture a spectrum snapshot of the track you are working on. The captured curve will be shown in **blue** in the snapshot view and spectrum view.

1. **Capture**: Click Capture and playback your current track to begin capturing a snapshot.
2. **Stop**: Click to stop capturing and freeze the Apply To spectrum snapshot.
3. **Clear**: Clears the current Apply To spectrum snapshot.

Fine Tune

The Fine Tune controls can be used to adjust the shape and intensity of the matched curve.

1. **Smoothing**: Determines the amount of precision to apply to the matched curve. Higher smoothing is less precise. Lower smoothing is more precise.
2. **Amount**: Determines the amount of processing (intensity) to use when matching the Reference curve to the Apply To curve.

★ **RECOMMENDATIONS**

1. A Matched Curve amount of 100% and a Smoothing amount of 0% might be technically the closest match to your “Reference” mix, but in reality it’s probably not the most effective combination of the settings. Those settings will try to capture every peak, valley, and level, which can result in extreme, unnatural EQs.
2. We suggest working with the Matched Curve amount under 50%. If your Matching EQ curve has narrow peaks and valleys, increase the Smoothing parameter to smooth them out. Your goal is to capture the overall tonal shape of the Reference as opposed to an exact match.

Maximizer

Table of Contents

Overview

Ozone's acclaimed IRC (Intelligent Release Control) technology lets you boost the overall level of your mixes without sacrificing dynamics and clarity. Upward Compress and Soft Clip are also available as methods to increase the loudness and level of your masters. The Maximizer applies to the entire bandwidth of the mix; it is not a multiband effect.

The module interface includes the following main sections:

1. **Module Header**
2. **Main Controls**
3. **Secondary Controls**
4. **Metering**

Module Header

The module header area includes the following controls:

1. **Delta**: Monitors the change created by Maximizer. This includes the peaks attenuated by the limiter and the boost applied by Upward Compress and Soft Clip. This does not monitor gain changes from the **Gain** or **Output Level** controls.
2. **View Selector**: See the **Views** section below for detailed descriptions of the different meter views.
3. **Learn Input Gain**: When enabled, the Maximizer will listen to your track for five seconds and set the **Gain** such that the output loudness matches the **Target LUFS** value. Learn Input Gain will automatically turn off when learning is complete.

   NOT RECOMMENDED FOR LOUDNESS COMPLIANCE

   Learn Input Gain should **not** be used to meet loudness compliance standards.
4. **Target LUFS**: Sets the loudness target (in LUFS) that the Maximizer will use when calculating the optimal Gain for your track.
5. **Reset**: Returns all module controls to their default values.

Main Controls

The Maximizer includes the following main controls:

1. **Mode**
2. **Gain**
3. **Gain and Output Level Link**
4. **Output Level**
5. **True Peak**
6. **Character**

Mode

The Maximizer includes the following Intelligent Release Control (IRC) modes:

1. **IRC Low Latency**: Provides the intelligent digital loudness maximization of IRC 1 with lower latency. IRC Low Latency is the **lowest latency** and **least CPU intensive** IRC mode.
2. **IRC 1**: Provides intelligent digital loudness maximization of the signal. It does this by analyzing the source material and applying limiting in a psychoacoustically pleasing manner, reacting quickly to transients (to prevent pumping) and reacting more slowly to steady bass tones (to prevent distortion).
3. **IRC 2**: Similar to IRC 1, but optimized to preserve transients even more, so they sound sharper and clearer in the output signal, even when aggressive limiting is taking place.
4. **IRC 3**: Allows for the most aggressive limiting by using an advanced psychoacoustic model to intelligently determine the speed of limiting that can be applied to the incoming signal, before producing distortion that is detectable to the human ear. **Very CPU-intensive**, and produces **high latency, especially at higher sampling rates.** You may find that at sampling rates greater than 48kHz you are unable to use IRC 3 mode in real-time.

    1. **IRC 3 Character Styles**: The following character styles will help you manage the limiter’s sound by constraining its release behavior:

        1. **Clipping**: The most aggressive style setting of IRC 3 and will slightly colorize your mix with distortion or achieve the highest degree of loudness with the greatest risk of clipping.
        2. **Crisp**: Agressively constrains the limiter’s release behavior and will favor distortion over any pumping.
        3. **Balanced**: Constrains the release behavior of the limiter in a generally transparent way and should be suitable for most material.
        4. **Pumping**: The least aggressive style setting for IRC 3 and does not constrain the limiter’s release behavior. It can tend toward a slower release behavior and may

result in pumping. This is the “legacy” setting and is the behavior used in Ozone version 5.01 and earlier.

5. **IRC 4**: This mode builds upon our existing IRC technology by shaping the spectrum to further reduce pumping and distortion. As the signal goes farther over 0 dBFS, the IRC 4 algorithm limits frequency bands that contribute most to these peaks. This reduces intermodulation between different signal components and uses dozens of psychoacoustically spaced bands in order to react to any type of audio. When no limiting is necessary, the spectrum will be unaltered. **Most CPU-intensive**, and produces **high latency**, especially at higher sampling rates**.

    1. **IRC 4 Character Styles**: The following character styles will help you manage the limiter’s sound by constraining its release behavior:

        1. **Classic**: Provides general enhancement of the overall mix with a sound more reminiscent of Ozone’s earlier limiting algorithms which are still being used by professionals today.
        2. **Modern**: Provides general enhancement and life to your mix but with greater detail and clarity than the Classic style.
        3. **Transient**: Optimized for maximum preservation of all transients resulting in a highly detailed overall sound that may benefit some mixes needing added clarity.

Gain

Adjust the Gain of the Maximizer to push the signal into 0 dBFS, where limiting will begin to take place.

> ★ **GAIN VS. THRESHOLD**
>
> Previous versions of Ozone Maximizer presented this parameter as Threshold. The new Gain control effectively works the same as the Threshold control, except that now you turn Gain up instead of turning Threshold down.

Gain and Output Level Link

Enable to link the **Gain** and **Output Level** controls. Adjusting either control in linked mode will adjust the other control by the inverse amount. This can be useful for hearing the changes from the limiter without the “louder is better” bias.

Output Level

Adjust to set the maximum output level of the Maximizer.

> ★ **OUTPUT LEVEL VS. CEILING**
>
> Previous versions of Ozone Maximizer presented this parameter as Ceiling. The new Output level control is doing the same thing, it has just been re-named to be more intuitive.

True Peak

When enabled, the limiter will account for the levels of each digital sample and the levels of the analog signal that will eventually be produced by D/A conversion. This is sometimes necessary, since an analog signal’s peak level can exceed the corresponding digital signal’s peak level by more than 3 dB.

> **TRUE PEAK LIMITING & CPU USAGE**
>
> This option will result in a small increase in CPU usage, but if your mixes are running very hot you may want to enable it to ensure that absolutely no distortion is introduced when your audio is finally run through a D/A converter.

Character

Adjust to customize the overall response time (attack and release times) of the maximizer processing. The attack and release times used are dependent on the selected **Mode**, and allows a continuous range from Clipping (0.0) to Very Slow (10.0) in each mode. There is also a description of each character range.

Secondary Controls

The Maximizer includes the following secondary controls:

1. **Upward Compress**
2. **Soft Clip**
3. **Transient Emphasis**
4. **Stereo Independence**

Upward Compress

Upward compression is a highly transparent form of compression. Rather than reducing the gain of peaks, it boosts the gain of quiet sections. Our Upward Compress processing is parallel compression, but with a very intentional design that gives it particularly useful properties:

1. The main control is in decibels. This is the maximum amount of gain that will be applied to quiet sections.
2. The processing is level-matched across its range so that peaks at 0 dBFS will remain at roughly the same level at any Upward Compress setting.

Upward Compress is applied before the IRC Maximizer.

Soft Clip

Enable Soft Clip processing before the IRC Maximizer by clicking the Soft Clip power button. Adjusting the Amount control provides a high fidelity loudness boost by controlling the wet/dry of the Soft Clip processing. At 100% the signal will fully clip at 0 dBFS.

SOFT CLIP OVERSAMPLING

The Soft Clip processing is oversampled 4x to prevent aliasing distortion.

Soft Clip has three modes that provide a variety of loudness boosts and saturate the edges of the signal differently:

1. **Light**: begins saturating the signal at -3 dBFS.
2. **Moderate**: begins saturating the signal at -9 dBFS.
3. **Heavy**: begins saturating the signal at -30 dBFS.

Transient Emphasis

Enable Transient Emphasis adjustment by clicking the Transient Emphasis power button. Adjusting the **Amount** control allows you to fine-tune the shaping of transients before limiting takes place. This can be useful for preserving sharper sounds, like drums, while still optimizing loudness.

> ★ **TRANSIENT EMPHASIS SETTINGS**
>
> Using higher amount values for Transient Emphasis will result in more pronounced transients after the limiting process.

Stereo Independence

The Stereo Independence controls represent the next iteration of the Stereo Unlink control in previous versions of Ozone. By default, the Stereo Independence controls (Transient and Sustain) will be linked and set to 0% - mimicking the default settings of the previous Stereo Unlink control.

1. **Transient Slider:** Adjusts how the limiter responds to transient material across channels.
2. **Sustain Slider:** Adjusts how the limiter responds to sustained material across channels.
3. **Link:** Links the Transient and Sustain sliders.

EXAMPLES

1. **Both Sliders set to 100%**: It is possible to achieve a louder output from the Maximizer, but this can result in a narrow stereo image. To alleviate the narrowing effect of the *Stereo Unlink control*, we split this feature into two sliders.
2. **Sliders independently set to non-zero values**: Applies limiting to transient and sustained material separately, based on a level envelope generated from a ratio of the individual channel levels and the entire stereo image.

Views

You can toggle between the different views using the view selector buttons in the module header area.

The Maximizer module includes the following views:

1. **Spectrum Analyzer**
2. **Waveform and Gain Trace**
3. **Gain Reduction Meter**

Spectrum Analyzer

Displays the magnitude (amplitude, in decibels) of a signal across the frequency spectrum in real-time. The spectrum analyzer displays the output signal of Ozone.

Gain Trace

Displays a scrolling waveform with a superimposed trace reflecting the amount of gain applied over time.

1. The trace the comes down from the top shows the attenuation from the limiter.
2. The trace that comes up from the bottom shows the boost from Upward Compress.

Gain Reduction Meter

1. Displays the gain reduction applied by the limiter in the left and right channels.
2. The text readout at the bottom of the meter displays the current gain reduction amount that is being applied by the limiter.

Spectral Shaper

Table of Contents

Overview

The Spectral Shaper can be used to apply high-resolution attenuation to problematic frequencies across the frequency spectrum with configurable time constants, timbre adjustment, and a variable full spectrum action region.

The Spectral Shaper interface includes the following sections:

1. **Module Header**
2. **Controls**
3. **Views**

Module Header

The module header includes the following controls:

1. **Delta** Monitors the difference in the signal before and after the Spectral Shaper module.
2. **View Selector**: See the **Views** section below for detailed descriptions of the meter views.
3. **Channel Processing Modes:** Selects the channel processing mode used by the Spectral Shaper.

    1. The Spectral Shaper supports **Stereo**, **Mid/Side**, and **Transient/Sustain**.
    2. See the **General Controls** chapter for more information.

4. **Learn**: Enables automatic action region cutoff point placement. When cutoffs have been set to their ideal positions, Learn will be automatically disabled.
5. **Reset**: Returns all module controls to their default values.

Controls

You can adjust the following controls in the Spectral Shaper:

Mode

Sets the intensity of reduction applied by the Spectral Shaper. Options include: Low, Medium, High.

Amount

Adjusts how much spectral gain reduction is applied to the action region.

Tone

Adjusts the spectral tilt of the Spectral Shaper processing. Positive values tilt towards a brighter overall spectral character, negative values tilt towards a darker overall spectral character.

Attack

Sets the amount of time it takes for the Spectral Shaper to apply gain reduction.

Release

Sets the amount of time it takes for the Spectral Shaper to stop applying gain reduction.

Views

You can toggle between the different views using the view selector buttons.

The Spectral Shaper module includes the following views:

1. **Difference Meter and Action Region**
2. **Gain Reduction Trace**

Difference Meter and Action Region

Allows you to adjust and audition the Spectral Shaper Action Region range. This view offers Action Region controls superimposed over a difference meter. This displays the spectral gain reduction processing that Spectral Shaper is applying. This is similar to an EQ where frequency is measured from left to right and gain is up and down.

The Action Region includes the following controls:

1. **Action Region Range Cutoff Handles**: Adjusts the low and high cutoff frequency values for the action region.

    1. Click and drag a handle left or right to adjust the cutoff frequency.
    2. Click in between both handles and drag left or right to move both handles together.
    3. Single-click on the text readout next to the bottom of a handle, type in a new frequency value and press enter to set the handle to that value.

2. **Action Region Solo**: Solos the input signal to the Action Region *before* processing.

Gain Reduction Trace

Displays a scrolling waveform with a superimposed trace reflecting the amount of gain reduction applied over time.

Stabilizer

Table of Contents

Overview

Stabilizer is an adaptive mastering equalizer that reacts to the incoming audio and applies frequency corrections to tonally balance a signal or tame resonances.

The Stabilizer interface includes the following sections:

1. **Module Header**
2. **Controls**
3. **Difference Meter**

Module Header

The module header includes the following controls:

1. **Delta**: Monitors the difference in the signal before and after the Stabilizer module. This can be particularly useful to hear what is being removed in Cut mode.
2. **Channel Processing Mode**: Selects the Channel Processing Mode used in the Stabilizer module.

    1. The Stabilizer module supports **Stereo**, **Mid/Side**, or **Transient/Sustain** mode.
    2. See the **General Controls** chapter for more information.

3. **Reset**: Returns all module controls to their default values.

Controls

The Stabilizer module includes the following controls:

Target

Selects the target tonal balance for Stabilizer processing. These target profiles are visible in the Master Assistant Overview. The “All-purpose” tonal balance target is based on the “Bass Heavy” target curve in Tonal Balance Control.

When a reference file target is selected in Master Assistant, an “Assistant” target will be available in Stabilizer. To recall this target later with other instances of Stabilizer, you can save it as a module preset.

Mode

Determines the type of EQ adjustments Stabilizer will make to match the selected Target. You can choose between two modes:

1. **Shape**: Stabilizer applies EQ boosts and cuts as it corrects your audio to the Target tonal balance. The processing will attempt to remain loudness neutral by applying an equal amount of boost and cut.
2. **Cut**: Stabilizer only applies EQ cuts as it tames resonant frequencies that exceed the upper bounds of the Target tonal

balance.

Amount

Scales the gain of the Stabilizer’s adaptive tonal corrections. At 100, the maximum gain that Stabilizer can possibly boost is 9 dB.

Speed

Controls how quickly the Stabilizer tonal corrections will react to the incoming audio signal. Higher speeds can provide a more precisely-timed correction but may introduce artifacts.

Smoothing

Controls the contour of the tonal corrections applied by Stabilizer in Shape Mode. At 100 smoothing, the EQ correction will look like 3 - 4 filters. At 0 smoothing, the EQ correction will look like many filters working across the frequency spectrum. Lower Smoothing values will provide a more precise frequency correction but may introduce artifacts.

Sensitivity

Controls how often the Stabilizer will detect resonances. At 0 Sensitivity, Stabilizer will permit all but the most excessive resonances without attempting to correct them. At 100 Sensitivity, Stabilizer will suppress any tonal deviation from the selected Target.

Tame Transients

Enables Stabilizer to react instantly to tonally correct any transient material.

Low Mid High Amounts

To control how much processing Stabilizer applies across the spectrum, you can scale the amount of tonal correction in three frequency areas:

1. **Low**: Below 100 Hz. The processing will attempt to remain

loudness neutral by applying an equal amount of boost and cut.
2. **Mid**: Between 100 Hz and 5.6 kHz.
3. **High**: Above 5.6 kHz.

Difference Meter

Displays the tonal changes that the Stabilizer processing is applying. This is similar to an EQ where frequency is measured from left to right and gain is up and down. Different from an EQ, Stabilizer will adapt its tonal corrections over time.

Vintage Compressor

Table of Contents

Overview

The Vintage Compressor is an emulation of a feedback compressor with a detection filter in the feedback loop. The Vintage Compressor includes a versatile detection filter in the feedback loop that filters the signal passed into the level detector to change which components of the signal are triggering the compressor. Some compressors use a highpass filter or a high shelf boost to reduce pumping. Others have a wide boost in the high frequencies to let high frequency content drive the compressor. The Vintage Compressor in Ozone includes all of these filters. We combined some of the best elements of vintage analog compressors to create this feedback compressor algorithm.

The Vintage Compressor module interface is divided into the following sections:

1. **Module Header**
2. **Views**
3. **Controls**

Module Header

The Vintage Compressor module header area contains the following controls:

1. **Delta**: Monitors the difference in the signal before and after the Vintage Compressor module.
2. **View Selector**: See the **Views** section below for detailed descriptions of the different meter views.
3. **Channel Processing Modes:** Selects the channel processing mode used by the Vintage Compressor.

    1. The Vintage Compressor supports **Stereo** & **Mid/Side**.
    2. See the **General Controls** chapter for more information.

4. **Reset**: Returns all module controls to their default values.

Views

You can toggle between the different views using the view selector buttons in the module header area.

The Vintage Compressor module includes the following views:

1. **Detection Filter**
2. **Gain Reduction Trace**

Detection Filter

This view includes a spectrum analyzer and detection input filter controls. Use the filter nodes to adjust the frequency response of the detection circuit used by the Vintage Compressor, so that it is more or less sensitive to specific frequencies.

Detection Filter Solo

Click the **(S)** button in the upper left corner of this view to listen to the filtered input to the detection circuit. This is useful for quickly listening to the input signal being used to trigger the Vintage Compressor processing.

Gain Reduction Trace

Displays a scrolling waveform with a superimposed trace reflecting the amount of gain reduction applied over time.

Controls

The following controls are available in the Vintage Compressor:

1. **Mode**
2. **Threshold**
3. **Ratio**
4. **Attack and Release**
5. **Gain**
6. **Auto Gain**

Mode

Adjusts the overall character of the Vintage Compressor. Mode options include:

1. **Sharp**: Provides crisp dynamics and a greater emphasis on transients while maintaining the body of the signal.
2. **Balanced**: Gives a signal-dependent balance between dynamics preservation and overall enhancement to the body of the signal.
3. **Smooth**: Smoothes out transient and dynamic material while enhancing and bringing out the rest of the signal, resulting in a thicker and fuller sound.

Threshold

Adjust the threshold to set the point where the dynamics processing takes place. Since some modes have a soft knee, mild compression may occur below this point.

Threshold Meter

Displays the input level to the module alongside the gain reduction applied by the module's processing.

1. The two meters on the far left and right of the threshold meter display the input audio level.
2. The two meters in between the input meters display the gain reduction applied.

Ratio

Sets the amount of attenuation applied to the signal once it has passed the threshold. Higher ratios will result in more compression.

Attack and Release

Adjust the attack and release controls to set how quickly the Vintage Compressor reacts to audio that crosses the threshold.

1. **Attack:** Determines how quickly the Vintage Compressor reacts when the threshold is reached.
2. **Release:** Determines the amount of time before the Vintage Compressor returns the level to normal once the signal is no longer above the threshold. This control sets the release time for transients; for sustained compression, the release time will be significantly longer.

Gain

Adjusts the output gain of the Vintage Compressor module to makeup for decreases in volume.

Auto Gain

When enabled, Auto Gain calculates the RMS levels of both the input and output signals of the Vintage Compressor in order to apply the appropriate gain to the output signal.

Auto Gain acts as a smart “make-up gain” control that adapts over time and adjusts your levels to a comparable level of your unprocessed audio.

Vintage EQ

Table of Contents

Overview

The Vintage EQ module can be used to add the emulated frequency response of the Pultec EQP-1A and Pultec MEQ-5. The module interface is divided into the following sections:

1. **Module Header**
2. **Spectrum and EQ Curve**
3. **Controls**

Module Header

The module header area includes the following controls:

1. **Delta**: Monitors the difference in the signal before and after the Vintage EQ module.
2. **Channel Processing Modes:** Selects the channel processing mode used by the Vintage EQ.
    1. The Vintage EQ supports **Stereo**, **Mid/Side**, **Left/Right**, and

Transient/Sustain channel processing modes.
   2. See the **General Controls** chapter for more information.
3. **Reset**: Returns all module controls to their default values.

Spectrum and EQ Curve

The Vintage EQ module includes a spectrum analyzer and a composite EQ curve.

1. **Spectrum Analyzer**: Displays the magnitude (amplitude, in decibels) of a signal across the frequency spectrum in real-time. The spectrum analyzer in the Equalizer module displays the output signal of Ozone.
2. **Composite curve**: The combined filter response of all bands. Represented by a thick white line drawn across the spectrum. Each band contributes to the overall shape of this curve.

Controls

The Vintage EQ control panel includes two sections of controls. The controls along the top are modeled after the **Pultec EQP-1A Equalizer** and the controls along the bottom are modeled after the **Pultec MEQ-5 Equalizer**.

Pultec EQP-1A Controls

The top row of controls includes filters modeled after the Pultec EQP-1A.

A unique feature of the Pultec EQP-1A is the ability to simultaneously adjust the boost and cut parameters for the low frequency band. While the original manual warns “Do not attempt to boost and attenuate simultaneously," these controls interact in a way that many engineers have found to be desirable. Rather than canceling each other out, combining these controls results in a boost in the low frequencies, followed by a dip at slightly higher frequencies. This unique shape is also available in the Ozone **Equalizer** module with the "Vintage Low Shelf" filter shape.

Low Boost/Cut Controls

1. **Low Frequency (Hz)**: Sets the frequency of the low shelf filter. Options include: 20 Hz, 30 Hz, 45 Hz, 60 Hz, and 100 Hz.

   > **45 HZ OPTION**
   >
   > The original hardware unit did not include a 45 Hz option, but we've added this frequency because we found it to be useful for mastering.

2. **Low Boost (dB)**: Adjusts the amount of positive gain (boost) applied to the filter.
3. **Low Cut (dB)**: Adjusts the amount of negative gain (cut) applied to the filter.

High Boost Controls

1. **High Boost Frequency (Hz)**: Sets the center frequency of the high boost peaking filter. Options include: 3 kHz, 4 kHz, 5 kHz, 8 kHz, 10 kHz, 12 kHz, and 16 kHz.
2. **High Boost Amount (dB)**: Adjusts the amount of positive gain (boost) applied to the filter.
3. **High Boost Q**: Adjusts the bandwidth of the high frequency boost filter. The boost amount and bandwidth interact (i.e. the Q changes as you boost), as they do in the EQP-1A.

High Cut Controls

1. **High Cut Frequency (Hz)**: Sets the frequency of the high shelf filter. Options include: 5 kHz, 10 kHz, and 20kHz.
2. **High Cut (dB)**: Adjusts the amount of negative gain (cut) applied to the filter.

Pultec MEQ-5 Equalizer Controls

The bottom row of controls includes filters modeled after the Pultec MEQ-5.

All of the mid frequency controls are peaking filters. As in the original MEQ-5, the filter bandwidths are affected by the boost/cut amount.

Low-Mid Controls

1. **Low-Mid Frequency (Hz)**: Sets the center frequency of the low-mid peaking filter. Options include: 200 Hz, 300 Hz, 500 Hz, 700 Hz, and 1000 Hz.
2. **LM Boost (dB)**: Adjusts the amount of positive gain (boost) applied to the filter.

Mid Controls

1. **Mid Cut Frequency (Hz)**: Sets the center frequency of the Mid peaking filter. Options include: 200 Hz, 300 Hz, 500 Hz, 700 Hz, 1 kHz, 1.5 kHz, 2 kHz, 3 kHz, 4 kHz, 5 kHz, and 7 kHz.
2. **Mid Cut (dB)**: Adjusts the amount of negative gain (cut) applied to the filter.

High-Mid Controls

1. **High-Mid Frequency (Hz)**: Sets the center frequency of the high-mid filter. Options include 1.5 kHz, 2 kHz, 3 kHz, 4 kHz, and 5 kHz.
2. **HM Boost (dB)**: Sets the amount of positive gain (boost) applied to the filter.

Vintage Limiter

Table of Contents

Overview

The Vintage Limiter, modeled after the Fairchild 670, allows you to create a louder and fuller master by limiting the dynamic range and boosting the overall level of your mix.

The Vintage Limiter module interface is divided into the following sections:

1. **Module Header**
2. **Views**
3. **Controls**

Module Header

The module header includes the following controls:

1. **Delta**: Monitors the difference in the signal before and after the Vintage Limiter module. This does not monitor gain changes from

the **Threshold** or **Ceiling** controls.
2. **View Selector**: See the **Views** section below for detailed descriptions of the different views.
3. **Reset**: Returns all controls in the module to their factory default values.

Views

You can toggle between the different views using the view selector buttons in the module header area.

The Vintage Limiter module includes the following views:

1. **Spectrum Analyzer**
2. **Gain Reduction Trace**

Spectrum Analyzer

Displays the magnitude (amplitude, in decibels) of a signal across the frequency spectrum in real-time. The spectrum analyzer displays the output signal of Ozone.

Gain Reduction Trace

Displays a scrolling waveform with a superimposed trace reflecting the amount of gain reduction applied over time.

Controls

The Vintage Limiter includes the following controls:

Mode

There are three different modes that affect the overall character of the Vintage Limiter.

1. **Analog**: With a fast attack and variable release time, this mode provides a tight bass response with a “thick” limiting quality. Brings out the low-end transients while still providing the smoothness that is characteristic of analog circuitry.
2. **Tube**: A more balanced limiter with variable attack and release times. Provides smooth feedback limiting with a wider range of sonic characteristics that vary depending on your incoming signal. Despite its non-linearity, it still allows for modern precision in preventing any clipping or peaks.
3. **Modern**: Blends thicker vintage characteristics and wide range of non-linearity with modern IRC limiting, variable release times, and transient reproduction.

Threshold

Determines the level at which limiting will start.

Threshold Input Meter

1. The two outer meters display input level to the limiter.
2. The two inner meters display gain reduction applied by the limiter.
3. The text readout at the bottom of the meter displays the current gain reduction amount that is being applied by the limiter.

Ceiling

Adjust to see the maximum output level of the Vintage Limiter.

RECOMMENDATION

It is generally recommended to use a setting of -0.3 dB when dithering, or a more dramatic setting (-0.6 dB to -0.8 dB) when converting to MP3 or AAC formats in order to prevent clipping during conversion. See the **Codec Preview** section for more details.

Link

Enable to link the **Threshold** and **Ceiling** controls. Adjusting either control in linked mode will adjust the other control by the same amount and vice versa.

True Peak

Enables the limiter to take into account not only the levels of each digital sample but also the levels of the analog signal that will eventually be produced by D/A conversion. This is sometimes necessary, since an analog signal’s peak level can exceed its corresponding digital signal’s peak level by more than 3 dB.

TRUE PEAK LIMITING & CPU USAGE

Enabling True Peak Limiting will result in a small increase in CPU usage. If your mixes are running hot, you may want to enable it to ensure that no distortion is introduced when your audio is finally run through a D/A converter.

Character

Adjusts the attack and release times of the Vintage Limiter. The attack and release times used are dependent on the selected **Mode**, and allows a continuous range from Fast (0.0) to Slow (10.0) in each mode.

Vintage Tape

Table of Contents

Overview

The Vintage Tape module is inspired by a well-maintained Studer A810 two-track tape deck, a clean and accurate machine perfect for mastering. Add the frequency response (magnitude plus phase) and saturation characteristics of magnetic tape without the crosstalk, hiss, wow, and flutter that could ruin a master.

The Vintage Tape module interface is divided into the following sections:

1. **Module Header**
2. **Spectrum Analyzer**
3. **Controls**

Module Header

The module header includes a Reset control:

1. **Delta**: Monitors the difference in the signal before and after the Vintage Tape module.
2. **Reset**: Returns all module controls to their default values.

> **DELTA AND PHASE SHIFT**
>
> The Vintage Tape processing shifts the phase of the audio signal. This phase shift is audible in the Delta signal, so be aware that the audio is not changing as much as the Delta signal may lead you to believe.

Spectrum Analyzer

Displays the magnitude (amplitude, in decibels) of the signal across the frequency spectrum in real-time. The spectrum analyzer displays the output signal of Ozone.

Controls

The Vintage Tape module includes the following controls:

Speed

Sets the tape speed in inches per second (IPS). You can choose between the following tape speeds in the Vintage Tape: **7.5**, **15**, or **30** IPS. In a hardware tape machine, this setting determines the rate at which the magnetic tape physically moves past the tape head during playback or recording. Faster speeds can improve high frequency response and overall quality. Slower speeds can cause a uniform decrease in the linear frequency response of tape, a shift in background noise toward lower frequencies, and increased background noise.

Input drive

Adjusts the gain of the input signal before tape emulation. The Vintage Tape module internally compensates for gain changes as you make adjustments, in order to avoid large or unexpected jumps in output level.

Bias

Adjusts the shape of the distortion curve. Negative bias values will boost high frequency content, resulting in more high frequency distortion than postive bias values. Positive bias values can begin to limit the dynamic range of the signal, lending a different type of saturation to the output.

Harmonics

Adjusts the amount of even-order harmonic distortion added to the output signal. Increasing even-order harmonic distortion can help to emulate the character of AC bias design inaccuracies or the distortion from machine electronics.

Low Emphasis

Adjusts the gain and shape of the low end “head bump” (resonant peak) of the tape.

1. **Lower Values**: Removes the resonant peak, flattening out the frequency response while maintaining the low-end rolloff. This can help achieve the warmth of analog without adding mud or too much bass.
2. **Higher Values**: Boosts the resonant peak up to 10dB, adding low end punch and emphasis.

High Emphasis

Compensates for high-frequency losses to give your audio extra energy without being too harsh or bright. The default setting of 4.0 gives you an authentic, close-to-flat high frequency response.

1. **Lower Values**: Adds gentle, roll-off at high-frequencies.
2. **Higher Values**: Adds a high-end shimmer.

Codec Preview

Table of Contents

Overview

Lossy audio formats, such as MP3 or AAC, use psychoacoustic algorithms to identify and remove less audible portions of an audio file in order to reduce the overall file size. This process can introduce artifacts ranging from subtle to obvious artifacts in the compressed file. Codec Preview allows you audition lossy compression formats and compensate for any undesirable artifacts they introduce *before* exporting your master.

Working with Codec Preview

You can access the Codec Preview panel by clicking on the **Codec** button in the I/O panel in the Ozone mothership plug-in.

You can **enable** or **disable** Codec Preview by clicking on the **power** button to the left of the **Codec** button in the I/O panel.

USING CODEC PREVIEW IN A DAW OR NLE

When working with Codec Preview within a DAW or NLE, you should insert Ozone *after* all other inserts on the master bus to ensure you are monitoring Codec Preview on the signal that is closest to the uncompressed export.

Controls

The Codec Preview panel allows you to adjust codec properties (format and bit rate) and listen to the artifacts introduced by the selected codec in isolation.

Format

Selects the codec format to preview, options include: **AAC** and **MP3**. Codec Preview makes use of the following encoders:

1. **AAC**: Fraunhofer AAC codec.
2. **MP3**: **LAME mp3 encoder**.

APPLE SILICON NOTE: AAC OPTION IS ROSETTA ONLY

The AAC format option is **only available** when Ozone is opened using **Rosetta** on Apple silicon-based Macs. MP3 is available in native and Rosetta.

Bit Rate (Constant)

Sets the constant bit rate value used to preview the selected codec format. Bit rate is measured in units of kbps (or kbit/s, kilobits per second) and represents the number of bits processed (encoded) within a given period of time. Constant bit rate (CBR) encoding indicates that the bit rate is the same for the entire duration of the file. You can choose from the following constant bit rate values:

1. **96 kbps**
2. **112 kbps**
3. **128 kbps**
4. **160 kbps**
5. **192 kbps**
6. **224 kbps**
7. **256 kbps**: *Maximum supported bit rate value for previewing **mono** files.*
8. **320 kbps**

Solo Artifacts

Isolates playback of the signal content that is removed or modified by the currently selected lossy compression settings. This signal is the *difference* between the output of Ozone *before* and *after* Codec Preview is applied.

Related information

Sample rates and performance

AAC and MP3 file formats don’t support sample rates greater than 48kHz. When using Codec Preview in a session with a sample rate greater than 48 kHz, high-quality real-time resampling will be applied in order to accurately represent AAC or MP3 compression.

If you encounter performance issues when working with Codec Preview in high sample rate sessions, we recommend increasing the session buffer size to avoid discontinuities or performance issues. Additionally, real-time resampling is mathematically intensive and may incur significant latency.

Headroom and clipping

Any lossy encoder introduces an approximation error, a noise which can increase peak levels and cause clipping in an audio signal, even if the uncompressed source audio file appears to peak under 0 dB. When mastering for compressed audio formats like AAC and MP3, it’s a good idea to leave between -1 dB and -1.5 dB of headroom to prevent clipping due to file compression.

Codec Preview can help you tailor your processing in order to avoid clipping that might occur as an artifact of lossy codecs. When Codec Preview is enabled, you can use the clip indicators above the Ozone output meters to help inform you of clipping caused by a codec before exporting.

Dither

Table of Contents

Overview

Ozone includes a comprehensive set of dithering tools to help you prepare studio-quality audio for different delivery formats. The Dither panel includes iZotope’s MBIT+ dither algorithm along with a unique set of meters to offer a complete view of the conversion process.

LEARN MORE ABOUT DITHER

Check out our related **educational resources and guides**.

Working with Dither

You can open the Dither panel by clicking on the **Dither** button in the I/O panel. You can enable/disable Dither processing using the power button to the left of the Dither button.

Dither is included in the Ozone mothership plug-in and the Ozone Maximizer component plug-in.

USING DITHER IN A DAW

1. Dither should be applied **after** all other processing.
2. Add Ozone to the last insert slot of the track you are exporting if you plan to add Dither.
3. Use a post-fader insert to ensure Dither is applied after any output gain adjustments.
4. Disable any dithering options in the export dialog of your DAW to avoid applying dither twice.

Controls

The Dither panel includes the following controls:

Bit Depth

Sets the target bit depth for the exported file. Choices include: 24, 20, 16, 12, or 8.

Auto-Blanking

Mutes dither output (i.e. dither noise) automatically when silence is detected (i.e. 0 bits of audio) in the input signal for at least 0.7 seconds.

Dither Amount

Selects the target number of bits (amplitude) of Dither to add to the signal. The following options are available:

1. **Strong**: Completely eliminates the non-linear distortion issues that may result from using **Off** or **Low** with the tradeoff of a slightly increased noise floor.
2. **Medium**: Recommended setting.
3. **Low**: Can leave some non-linear quantization distortion or dither noise modulation.
4. **Off**: Can leave some non-linear quantization distortion or dither noise modulation.

Harmonic Suppression

Enables modification of truncation rules. Truncation can result in harmonic quantization distortion that adds overtones to the signal and distorts the timbre. Enabling this option will modify truncation rules in order to move harmonic quantization distortion *away* from overtones of audible frequencies. This can help maintain better tonal quality in the resulting signal. Harmonic Suppression doesn’t create any random dithering noise floor.

> **HARMONIC SUPPRESSION AND ARTIFACTS**
>
> Harmonic Suppression can only be enabled when Dither Amount is set to Off.

Limit Peaks

Suppresses peaks in the output signal that can be caused by more extreme dither settings. Dither noise is random in nature and has a very low amplitude, so peaks will not often be an issue. In cases where you need to apply more aggressive dithering, high frequency dither noise can be amplified significantly after noise shaping, resulting in spurious peaks in the dither signal (up to -60 dBFS for 16-bit quantization).

Noise Shaping

Determines the amount of noise shaping applied during the dithering process, ranging from **Off** (no noise shaping) to **Max** (roughly 14 dB of audible noise suppression). Noise shaping pushes the dither noise into less audible frequency ranges, allowing for greater dithering with less perceived noise.

Noise Shaping Curve

Displays the general frequency curve of the selected noise shaping profile.

Bit Meter

Displays information about the digital activity of your program material. The bit meter is not a level meter, it displays which bits are being used. If a bit is used (goes from 1 to 0 or vice versa), the box representing that bit will light up.

1. Real-time bit activity in the left and right channels.
2. Overall (“peak hold”) bit activity in the left and right channels.

Generally, you want to see activity in each of the bits that are relevant to the bit depth you are exporting to. A signal like DC offset will toggle a lower bit once (lighting the outside column), but would never toggle it after that since the bit is being held. If you’re dithering down, you only want to see 16, 12, or 8 bits lit (corresponding to the output bit depth of the dither).

WHY DOESN'T THE TOP BIT LIGHT UP IN THE BIT METER?

When samples are stored as a binary number, negative samples are specially encoded. Since the binary representation of negative numbers is not intuitive, Ozone takes the absolute value of each sample before plotting it on the bit meter. Since the MSB (most significant bit) is only set when a sample is negative, this bit will never light up. We simply included the top bit as placeholder for completeness.

Reset

Clears the state of the Bit Meter. Clicking on the Bit Meter will also reset the peak hold information.

DC Offset

This section allows you to monitor and filter DC Offset from your track before limiting.

Audio signals are normally represented by an alternating current (AC) signal with an average value of zero. DC offset can be caused by things like malfunctions in analog recording equipment or faulty A/D conversion. In an FFT (Fast Fourier Transform) analyzer, DC offset is represented with energy at zero frequency (0 Hz). DC offset can be effectively removed with a high-pass filter set to a very low frequency value.

It is important to distinguish DC offset from waveform asymmetry. Asymmetry skews the waveform shape towards positive or negative levels, but the average current stays at zero. Many audio signals are naturally asymmetric in nature. Unlike DC offset, asymmetry has no particular energy near 0 Hz and can be reduced using phase rotation filters.

DC Offset Meter

Displays the amount of DC Offset present in the signal.

Click on the DC Offset meter to clear and reset the calculation.

Filter

Enables a highpass filter with a cutoff frequency of 1 Hz to filter DC offset before limiting is applied.

> **DC OFFSET FILTER MODULE DEPENDENCY**
>
> In the Ozone mothership plug-in, the DC offset meter and filter are only available if the Maximizer or Vintage Limiter module is in the signal chain. When the Filter is enabled, it is applied to the signal before it enters the Maximizer or Vintage Limiter module (whichever comes first, if both are present in the signal chain).

Referencing

Table of Contents

Overview

Referencing in Ozone helps you compare and contrast audio in your DAW by allowing you to import up to 10 reference tracks, visualize differences in your music with overlaid spectrum metering, and quickly A/B your audio without leaving Ozone.

Click on the **Reference** button in the I/O panel to open the Reference panel. Enable reference track playback using the power button next to the Reference button. The Reference feature is only available in the Ozone mothership plug-in.

Importing References

1. Click on the “+” button in the middle of the empty Reference panel to select reference files.
2. **Supported file formats**: wav, aif/aiff, mp3, AAC, FLAC.
3. The Reference panel supports up to 10 reference tracks at a time.

Reference Track Tabs

Each tab represents an imported reference file. You can manage the reference track tabs using the following methods:

1. **Add Tracks**: Click on the **+** button to add a new reference track. (*You can add up to 10 tracks at a time.*)
2. **Re-order Tabs**: Click and drag track tabs to change their order.
3. **Remove Tracks**:
    1. Right-click on a track tab and select **Remove Track**.
    2. Click on a tab to select it, then click on the **X** button.

Reference Loop Segments and Playback Controls

When tracks are initially loaded into the Reference panel, handles will appear overlaid on the waveform display. The handles represent boundaries of predetermined loop points. The loop boundaries are placed based on similar segments detected in the track. By default, the segments will be named with a letter (A/B/C/D/E).

Managing Reference Loop Segments

You can customize, add, and remove loop segments:

1. **Adjust loop segment length**: Click and drag the Region Selector handles LEFT or RIGHT to adjust the length of a loop segment.
2. **Rename loop segments**: Click on the text above a loop segment to open an inline edit field. Enter the desired loop name and press the `Enter` or `Return` key *or* click anywhere in the interface to save the loop segment name.
3. **Insert loop**: Right-click on the waveform where you would like to add a new loop, select “Insert Loop” from the menu.
4. **Remove loop**: Right-click on a loop segment and select “Remove Loop” from the menu.
5. **Select loop for playback**: Click anywhere within a loop segment and press the `spacebar` to loop reference playback on a specific

segment.

1. **Note**: The Reference power button in the I/O panel needs to be enabled in order to hear reference track playback.

Reference Playback Controls

1. **Select All**: Enable to loop the entire reference track.
2. **Gain**: Adjusts the output gain of reference track playback.

Reference Metering Options

The I/O options pop-up menu above the I/O meters contains metering options for Reference tracks.

1. **Replace Input with Reference**: When enabled, the Ozone input meter will display the reference track level.
2. **Show Reference Spectrum**: When enabled, the Reference track spectrum will be displayed in the module spectrum meters alongside the spectrum meter of your current track.

Preset System

Table of Contents

Overview

Ozone’s presets are designed to give you a quick starting point for mastering your projects. You can load, save, or customize presets with the preset manager. There are two types of preset manager windows in Ozone:

1. **Global Preset Manager**: Click on the preset name display bar in the header to open the global preset manager window.
2. **Module Preset Manager**: Click on the Presets button in a module’s signal chain tile to open the associated module preset manager window.

Factory and Custom Presets

The preset manager windows are divided into two tabs:

1. **iZotope**: lists all factory presets installed with Ozone.
2. **Custom**: lists all custom presets you have saved or modified in Ozone.

Default and Working Settings

There are two common options that are always available at the top of the preset list in the **iZotope** and **Custom** tabs, named **Working Settings** and **Default**.

1. **Working Settings**: Loads your most recent changes that are not otherwise associated with a preset.
2. **Default**: Loads the factory default settings.

Custom Default Preset

You can override the factory `<Default>` preset by selecting any preset as your custom default.

Setting a Custom Default Preset

1. Right-click on the preset that you want to use as your default.
2. Select `Set <SelectedPresetName> as Default` in the right-click context menu.
3. After setting your custom default preset, selecting `<Default>` in the preset manager window will load your custom default settings. New instances of Ozone will also use your custom default settings.

Reverting a Custom Default Preset

You can revert to the factory default preset by:

1. Open the preset manager.
2. Right-click on any preset in the list.
3. Select `Reset Default to factory default` from the right-click context menu.

Custom Preset Names and Comments

You can modify preset file/folder names and preset comments when the **Custom** tab is selected.

1. **Edit custom preset name:** Double-click on a preset name to open an inline edit field. Press return to dismiss the inline edit field and save your changes.
2. **Edit custom preset comment:** The area below the preset list displays descriptive text about the currently selected preset. Single-click the comment text box to open an inline edit field, press return to save changes to the comment.

SPECIAL CHARACTERS IN PRESET NAMES

Some characters such as * or / cannot be used as preset names. If you try to type these characters in the name they will be ignored.

Preset Manager Footer

The following buttons are located in the footer of the Preset Manager window:

1. **Delete**: Deletes the currently selected custom preset or preset folder.
2. **Update:** Saves changes to a modified custom preset. **Note**: Update is only available in the *global* preset manager, you cannot update module presets.
3. **Folder**: Adds a new custom preset folder.
4. **New**: Creates a new preset based on the current settings.
5. **Close**: Dismiss the Preset Manager window.

DIRTY STATE INDICATOR

When you make changes to a preset an asterisk (*) will be shown at the beginning of the preset name to indicate that it has been modified. You can add a new preset to save your settings or update the preset to dismiss the dirty state indicator.

ORGANIZE CUSTOM PRESETS

In the Custom tab, you can click and drag presets or folders over other folders in the list to move them into that folder.

Preset Locations

Factory presets are *installed* to the following locations:

1. **Windows:** `C:\Program Files\iZotope\Ozone\Presets\`
2. **Mac:** `/Library/Application Support/iZotope/Ozone/Presets/`

Custom presets are *saved* to the following default locations:

1. **Windows:** `C:\Users\Username\Documents\iZotope\Ozone\User Presets\`
2. **Mac:** `/Users/Username/Documents/iZotope/Ozone/User Presets/`

If you have installed previous versions of Ozone 10 (prior to version 10.2.0) or any version of Ozone Pro, your custom presets will be stored in the following locations:

1. **Windows:** `C:\Users\Username\Documents\iZotope\Ozone Pro\Presets\`
2. **Mac:** `/Users/Username/Documents/iZotope/Ozone Pro/Presets/`

Options

Table of Contents

Overview

You can adjust global and module-specific settings in the Options window. You can open the Options window by clicking the gear button in the upper right hand corner of the Ozone interface.

You can access different groups of options by clicking on a tab in the upper portion of the Options window. Only tabs that are relevant to the plug-in you are using will appear in the options window. For example, the EQ tab is not included in the Maximizer component plug-in options window. You can reset, save or cancel changes to options using the buttons in the footer area of the options window.

Options Window Footer Controls

The bottom of the options window includes the following global parameters:

1. **?**: Opens the Ozone help documentation in your default web browser.
2. **Reset**: Resets all Options in the **currently selected options tab** to their factory default values.
3. **Version information**: Displays the current Ozone version and build number.
4. **Cancel**: Closes the Options window without saving changes.
5. **Ok**: Closes the Options window and saves changes.

General Options

The General Options tab allows you to adjust settings related to graphics, authorization, updates, usage tracking and undo history.

General: Graphics

1. **Show Tooltips:** Enables the display of a descriptive text box when hovering your cursor over a control.
2. **Dim Controls When Bypassed:** Enables an overlay that dims the appearance of controls in the module control panel when Ozone is bypassed.
3. **Window Opacity**: Adjusts the opacity of the Ozone plug-in window, allowing you to see through the interface momentarily without closing the plug-in window.

General: License

1. **Licenses**: This section allows you to view your current license status, manage your license, and access more information about licensing the product.

> **ADDITIONAL HELP WITH LICENSING AND INSTALLATION**
>
> For additional information about licensing, authorization and installation, see the **Installation and Authorization Help** section of the iZotope Support Portal.

General: Other

1. **Send usage data:** Enables usage data to be sent to iZotope to help improve Ozone in the future.

    1. **?**: Opens the iZotope Analytics Opt-in information page in a web browser.

2. **History Depth:** Determines the number of parameter change events to store in the Undo History list.
3. **Keyboard Support**: Allows you to adjust the level of keyboard shortcut support. If you find that Ozone is overriding keyboard shortcuts that you use in your DAW or NLE, try setting this to None or Minimal.

    1. **None**: Disables keyboard shortcut support for the Ozone plug-in window.
    2. **Minimal**: Allows Ozone to accept Tab, Enter, and Arrow key shortcuts.
    3. **Full**: Enables all Ozone keyboard shortcuts when the Ozone plug-in window is in focus.

Spectrum Options

The Spectrum options tab allows you to manage options related to the Spectrum view in all Ozone modules.

1. **Type**: Selects the type of spectrum display used by spectrum analyzer views in all Ozone modules. The following options are

available:

    1. **Linear**: A continuous line connecting the calculated points of the spectrum.
    2. **1/3 Octave**: Splits the spectrum into bars with a width of 1/3 of an octave. Although the spectrum is split into discrete bands, this option can provide excellent resolution at lower frequencies.
    3. **Critical**: Splits the spectrum into bands that correspond to how we hear, or more specifically how we differentiate between sounds of different frequencies. Each band represents sounds that are considered “similar” in frequency.
    4. **Full Octave**: Splits the spectrum into bars with a width of one full octave.
2. **Fill Spectrum**: Allows you to display the real-time spectrum as a solid graph as opposed to a line graph. This option can be used to differentiate the real-time spectrum from the peak hold spectrum.
3. **Peak Hold Time (ms)**: Sets the length of time the peak hold display will hold before updating. You can choose between specific hold times (in milliseconds) or Infinite, where the peak display is held indefinitely.
4. **Show Peak Hold**: Toggles whether Ozone displays and holds the peaks of the spectrum.
5. **Window Type**: Sets the window type for the spectrum. In most cases the default window type will work well, but you can choose from a variety of window types. Each window type has different amplitude and frequency resolution characteristics.
6. **Window Size**: Controls the tradeoff between time and frequency resolution in the spectrum analyzer. Higher values offer greater frequency resolution.
7. **Average Time**: Averages the spectrum according to this setting. Higher average times can be useful for viewing the overall tonal balance of a mix, while shorter average times provide a more real-

time display. Options include: **Real Time, 1 sec, 3 sec, 5 sec, 10 sec, and Infinite.**

8. **Frequency Scale**: Modifies the distribution of frequencies along the frequency axis of the spectrum analyzer. The default frequency scale is Extended Log.

    1. **Mel**: Displays a frequency scale based on human perception of sound that visually corresponds to how we hear differences in pitch.
    2. **Logarithmic (Flat, Extended) Logarithmic**: Displays non-linear scales that offer detail on the low end and midrange, useful for the vast majority of EQ tasks.

9. **Tilt Slope**: tilts the spectrum analyzer around 1 kHz by the specified amount of dB per octave. The default value of 3 dB/oct will present pink noise as flat. The previous default (before this could be adjusted) was 0 dB/oct.

EQ Options

The EQ tab includes EQ Spectrum and Performance options. These options affect Equalizer 1 *and* Equalizer 2 modules.

EQ: Spectrum

1. **Alt-Solo Filter Q**: Sets the bandwidth (Q) of the Alt-Solo feature in the **Equalizer** module(s).
2. **Show Extra Curves**: Toggles the display of Phase Delay, Phase Response, and Group Delay curves in the EQ spectrum.

EQ: Performance

1. **Soft Saturation**: Select to enable soft saturation to in the EQ module. When enabled, signals that clip as a result of being

boosted by the EQ will saturate with an analog character, rather than harsh digital clipping.
2. **Buffer Size**: Select to adjusts the memory buffer size, measured in samples, when applying equalization to the signal.
3. **Frequency Resolution**: Select to set the minimum resolution, in Hz, the Equalizer can be adjusted by. You can choose from: 3 Hz, 6 Hz, 12 Hz, 24 Hz, and 48 Hz.
4. **Filter Size**: Displays the calculated steepness of the filter setting used in the EQ module.

Dynamics Options

This tab allows you to adjust multiband crossover settings and lookahead time for the Dynamics module.

1. **Crossover Type**: Sets the type of crossover used for multiband processing in the Dynamics module.

    1. **Analog**: Provides a natural analog character.
    2. **Digital**: Provides a more transparent sounding crossover.
    3. **Hybrid**: A perfect reconstruction IIR analog crossover designed to reduce phase distortion and frequency distortion found in other analog crossovers while maintaining precise crossover points and the warm characteristics of analog crossovers.

2. **Crossover buffer size**: Determines the digital audio buffer size that is used when the Digital crossover type is selected. Adjusting this value can affect CPU performance.
3. **Crossover Q**: When using the Digital crossover type, you can manually set the bandwidth of the crossover points in the Dynamics module. A higher Q value results in tighter crossovers, while a lower Q provides a more gradual transition from one band to the next.
4. **Look ahead [ms]**: Determines the look ahead time window for input level detection in the Dynamics module. Ranges from 0ms (instantaneous level detection) to 10 ms (level detection based on)

Imager Options

This tab includes options related to metering, processing, and crossovers in the Imager module.

1. **Prevent Antiphase**: Select to automatically prevent any settings from being applied that would result in phase cancellation of the stereo signal, when summed to mono.
2. **Vectorscope Detection Method**: Determines the type of amplitude detection method used by the Vectorscope.

    1. **Peak**: Uses the peak level of the incoming signal.
    2. **RMS**: Uses the average level of the incoming signal.
    3. **Envelope**: Similar to RMS, uses average level of the incoming signal evened out across all frequencies.

3. **Crossover Type**: Sets the type of crossover used for multiband processing in the Imager module.

    1. **Analog**: Provides a natural analog character.
    2. **Digital**: Provides a more transparent sounding crossover.
    3. **Hybrid**: A perfect reconstruction IIR analog crossover designed to reduce phase distortion and frequency distortion found in other analog crossovers while maintaining precise crossover points and the warm characteristics of analog crossovers.

4. **Crossover buffer size**: Determines the digital audio buffer size that is used when the Digital crossover type is selected. Adjusting this value can affect CPU performance.
5. **Crossover Q**: When using the Digital crossover type, you can manually set the bandwidth of the crossover points in the Imager module. A higher Q value results in tighter crossovers, while a lower Q provides a more gradual transition from one band to the next.

Dynamic EQ Options

This tab includes general display options for the Dynamic EQ module.

1. **Alt-solo filter Q**: Adjust to set the bandwidth (Q) of the Alt-Solo feature in the Dynamic EQ module. Ranges from 0.2 to 12.0.

Exciter Options

This tab includes options related to the multiband crossover in the Exciter.

1. **Crossover Type**: Select from analog, digital or hybrid as the crossover type used for multiband processing in the Exciter module.

    1. **Analog**: Provides a natural analog character.
    2. **Digital**: Provides a more transparent sounding crossover.
    3. **Hybrid**: A perfect reconstruction IIR analog crossover designed to reduce phase distortion and frequency distortion found in other analog crossovers while maintaining precise crossover points and the warm characteristics of analog crossovers.

2. **Crossover buffer size**: Determines the digital audio buffer size that is used when the Digital crossover type is selected. Adjusting this value can affect CPU performance.
3. **Crossover Q**: When using the Digital crossover type, you can manually set the bandwidth of the crossover points in the Exciter module. A higher Q value results in tighter crossovers, while a lower Q provides a more gradual transition from one band to the next.

I/O Options

1. **Enable I/O meters**: Select to enable/disable the meters in Ozone's master I/O section.

2. **Detect true peaks**: Select Detect True Peaks to accurately measure the signal that will result from digital to analog conversion. By default the Input/Output meters will only indicate clipping which occurs within the digital domain.

3. **Enable modern bypass gain match behavior**: Determines the behavior of the Gain Match feature in the I/O auditioning section.

    1. **Behavior when unchecked**: Gain matching is applied to the bypassed output of Ozone and does not affect the I/O gain. The bypassed output level will be adjusted by the amount of gain introduced by Ozone processing. This allows you to A/B your processing without noticeable jumps in level between the bypassed and unbypassed signals. This gain adjustment is only applied to the output signal *when Ozone is bypassed* and Gain Match is enabled in the I/O auditioning section.
    2. **Behavior when checked**: Gain matching is applied to the processed output of Ozone. Ozone's output gain will be automatically adjusted to match the unprocessed input level. The output level sliders will be colored blue when this gain matching is active. In this mode, we recommend turning Gain Match on and off as needed rather than keeping it enabled all the time. This behavior does affect the output level of Ozone.

4. **Meter Type**: Choose from one of the following options to set the meter type displayed in the I/O meters:

    1. **RMS**: RMS (Root Mean Square) is a software-based implementation of an analog-style level meter. Using different integration times, you can model popular VU or PPM meters. The RMS meter displays the average level calculated over a short window of time. The RMS meter readout will typically be lower than an equivalent PPM meter

   (Digital/Analog), since it is averaging peaks into the overall loudness.
   2. **PEAK**: The Peak meter is a fast meter that measures instantaneous maximum sample value or peak analog waveform values, depending on the “detect inter-sample peaks” checkbox. If you are tracking the peaks for possible clipping, the Peak meter is appropriate.
   3. **RMS + PEAK**: This is a combined RMS and Peak meter. This meter displays a lower bright bar representing the average level (RMS) and a higher dimmer bar representing peak level. There is also a moving line above the bar representing the most recent peak level or peak hold.
   4. **K-SYSTEM**: Ozone supports Bob Katz’s K-System metering with simultaneous peak and RMS displays.
   5. **MOMENTARY**: This measurement is a calculation of loudness over the course of 400ms.
   6. **SHORT TERM**: This measurement is a calculation of loudness over the course of 3 seconds.
   7. **INTEGRATED**: This measurement is a calculation of loudness over the course of an indefinite period of time.

5. **Meter Scale**: Choose from one of the following options to set the range and scale of the I/O Meters:

   1. **dB (Linear)**: Decibel scale presented linearly from -60 dB to 0.
   2. **dB (Non-linear)**: Full decibel scale (dBfs) presented non-linearly.
   3. **BS.1771**: Loudness scale recommended by the ITU that spans from -45 LUFS to -14.0 LUFS.
   4. **EBU +9**: Loudness scale recommended as a default by the EBU that spans from -41.0 LUFS to -14.0 LUFS.
   5. **EBU +18**: Loudness scale recommended for material with a wide Loudness Range by the EBU that spans from -59.0 LUFS to -5.0 LUFS.

   > **LUFS**
   >
   > LUFS is Loudness Units Full Scale and 1 LUFS = 1 dB.

6. **Meter Source**: The following meter source options are available: **Stereo** and **Mid/Side**. The I/O meters display **Stereo** information by default. When the I/O meters are in **Mid/Side** mode, the meter in the center represents the Mid channel level information and the

meters on the left and right represents Side channel level information.

> **GAIN SLIDERS CONTROL LEFT/RIGHT GAIN IN MID/SIDE MODE**
>
> The input and output gain sliders always control the left and right input/output gain, regardless of the Meter Source selection.

7. **Peak Hold Time (ms)**: Choose from the following options to set how many consecutive samples of audio must exceed 0 dBFS (full scale) before registering as a peak: **5 ms, 250 ms, 500 ms, 1,000 ms, 5,000 ms, and Infinite.**

8. **Integration time (ms)**: Choose from the following options to specify the integration time for the RMS calculation: **10 ms, 50 ms, 300 ms (VU), 1,475 ms, 2,650 ms, 3,825 ms, and 5,000 ms.** In most RMS meters, the integration time is set to around 300 ms.

9. **Readout**: Selects whether the peak hold section of the meters displays the current peak status (current) or instead displays the highest peak that has occurred in the audio file (max peak).

10. **Show Peak Hold**: Displays peak hold bars in the I/O meters.

Elements

Table of Contents

Overview

Ozone 11 Elements is a great way to get started with Ozone 11 and iZotope's system of assistive mixing and mastering tools. Master Assistant is designed to be your second set of AI-powered ears. It will analyze your track and offer an objective suggestion to help you achieve a professional-sounding master. For a beginner, this will hopefully give you the confidence to share your work with the world.

The Ozone 11 Master Assistant comes with a set of ten genre targets that have been derived from the latest chart-topping hits. When a target is selected, the assistant will match your song's tonal balance and width to the average of those songs. You can also generate and save your own custom targets for Master Assistant based on reference song files on your computer. The new Master Assistant overview page provides powerful, high-level control over all of this matching so that you can dial-in the settings to your taste.

Workflow Steps

1. Add Ozone 11 Elements to the Master channel or Stereo Output of your DAW.
2. Click the **Start Listening Button** in the welcome page to begin the analysis phase.
3. **Analysis**: Master Assistant requires audio input to perform analysis and adjust settings.

    1. Playback your track for **at least 8 seconds** so that Master Assistant has enough time to analyze the input audio.
    2. Playback the **loudest portion** of your track to achieve the best results.
    3. Enable loop playback in your DAW if you are analyzing a selection that is less than 8 seconds long.

4. After analyzing your song, Master Assistant will build a processing chain and switch to the Master Assistant controls view when the processing chain is built.

Master Assistant View

The Master Assistant View provides high-level controls and metering in the following sections:

1. **Header**
2. **Target Library**
3. **Tonal Balance**
4. **Loudness**
5. **Vocal Balance**
6. **Extras**

Header

1. **Master Assistant Tab**: If analysis has already been performed, selecting this tab will show the Master Assistant View.
2. **Module View Tab**: Shows the Upgrade View for Ozone Elements. In Ozone Standard and Advanced, this is where modules can be added and adjusted.
3. **Gain match**: matches Ozone's output level to the input level by automatically compensating for any gain changes introduced by the processing.
4. **Global Bypass**: disables all Ozone processing.
5. **Relearn**: launches the analysis phase and sets up new targeted processing for the newly-measured section.
6. **Settings**: shows the Ozone settings menu.
7. **Help**: opens a web browser with the Help Documentation.

Target Library

You can modify the overall target that the Master Assistant will work toward in the Target Library section. Changing the target selection will adjust all Master Assistant processing to work toward matching your track to the selected target. The factory target curves were generated through an analysis of chart-topping hit songs from a range of different genres. The Cinematic target was generated through an analysis of the scores of top box office films.

Each Target is comprised of sub-targets that are associated with different aspects of Master Assistant processing:

1. **Tonal Balance**: the distribution of energy across the audible frequency spectrum.
2. **Vocal Balance**: the difference in loudness between the vocals and the rest of the music.
3. **Stereo Width**: the ratio of mid to side signal power in each band.
4. **Loudness**: the integrated loudness measured in LUFS.

Targets from Reference files

You can create and manage your own custom reference targets by importing audio files from your computer.

1. **Plus button**: opens a system dialog where you can select audio files on your computer to add to your custom Target Library. Adding or selecting a new reference file target will require a short analysis phase to build a new signal chain.
2. **Trash**: deletes a selected reference file target.

> **NOTE**
>
> Master Assistant will analyze an uploaded reference file and learn the loudest eight seconds. The target will be generated from that loudest section of audio.

> **CUSTOM VOCAL BALANCE**
>
> Master Assistant does not create vocal balance targets from custom references. When a custom reference target is selected, the Master Rebalance module will not be added to the signal chain.

Tonal Balance

The Tonal Balance section has one control:

1. **Equalizer**: scales the gain of all EQ nodes from 0%-200%. This links directly to the EQ global amount control in the upper right of Equalizer 1.

    1. The Equalizer 1 module can also be enabled and disabled with the power button for quick comparisons. When Gain Match is enabled, any gain introduced by processing will be compensated when bypass is engaged.

The Tonal Balance section also features metering that will be familiar to users of our product, Tonal Balance Control. When a Target is selected, you will see this reflected in the blue tunnel. The white line represents your audio's tonal balance. Increasing the Equalizer, Stabilizer, or Clarity amount should result in your track aligning more closely with the Target tonal balance.

Loudness

The Master Assistant measures integrated loudness during the analysis phase and sets the Maximizer's Gain to the level necessary to achieve the target loudness for the detected genre (or the loudness of a custom reference).

The Loudness section has one control and a destination selector:

1. **Maximizer**: adjusts the Gain of the Maximizer +/- 4 dB.

    1. The Maximizer module can also be enabled and disabled with the power button for quick comparisons. When Gain Match is enabled, any gain changes will be compensated during this bypass.

The Master Assistant will optimize your audio's output level for two destinations:

1. **Full Scale**: this is intended as a good output level for most masters. The Maximizer's Output Level will be set just shy of full

scale at -0.1 dB with True Peak limiting disabled.

2. **Streaming**: this is intended for audio that will be played back with lossy encoding and loudness normalization, as is typical of most streaming services. The Output Level is set to -1 dB with True Peak limiting enabled.

This section also features a scrolling waveform and gain trace that will meter how much gain reduction or boost is being applied by the Maximizer. This is useful to dial in exactly how much limiting is taking place when adjusting the Maximizer slider.

Vocal Balance

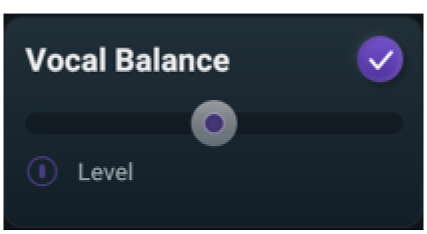
To perform a vocal balance, the Master Assistant will use AI to separate the vocal from the rest of the music during the analysis phase. It measures the integrated loudness of both the vocal and music. Master Assistant will then apply the Master Rebalance module to adjust your vocals towards a target loudness difference between the vocal and music.

The Vocal Balance section has one control:

1. **Level**: adjusts the gain the Master Rebalance module, focused on vocals.

    1. The Master Rebalance module can also be enabled and disabled with the power button for quick comparisons.

If Master Assistant measures the vocal/music balance and determines that it is well balanced for your genre target, it will show a check mark and not add Master Rebalance to the chain.



If Master Assistant does not detect a vocal during analysis, it will not add Master Rebelance and not show a check mark.

Extras

The Extras section has four controls:

1. **Width Match**: scales all of the per-band Imager amounts from 0% to 100%.

    1. The Imager module can also be enabled and disabled with the power button for quick comparisons. When Gain Match is enabled, any gain changes will be compensated during this bypass.

To perform a width match, the Master Assistant will analyze the balance of mid and side information in your audio across four separate bands. It will then use the Imager module to adjust the side channel to match the selected target. Note that Master Assistant may enable Recover Sides at different amounts based on the selected Target. It will not enable Stereoize. If the low frequency band of the analyzed audio is sufficiently narrow, Master Assistant will not attempt to widen the bass to match a selected target.

1. **Stabilizer Amount**: controls the amount of Stabilizer processing. Stabilizer will match its target to the selected genre. If a reference file target is selected, the Master Assistant will generate a custom "Assistant" target for Stabilizer based on the tonal balance of the imported reference file.

    1. The Stabilizer module can also be enabled and disabled with the power button for quick comparisons. When Gain Match is enabled, any gain changes will be compensated during this bypass.

Upgrade View

Ozone 11 Elements is designed to be an introduction to iZotope's assistive mixing and mastering tools. It can easily help you achieve a professional-quality master with your personal taste, but it does not include all of the powerful controls and customization options that are available in Ozone 11 Standard and Advanced. Both of these products are available with free trials, so if you're interested in unlocking the full potential of Ozone 11, click the **Learn More** button.

Glossary

Bias

The process in analog tape recording by which a supersonic signal is introduced to the incoming audio in order to improve the distortion and frequency response specifications of the recording.

Band

Short for “frequency band” which is a range or interval in the frequency spectrum often divided into low, mid or high-frequency bands.

Band shelf

Hybrid filter shape combining attributes of a shelving filter and a peaking filter.

Bandwidth

Describes the range of frequencies being affected by a signal processor. Inversely proportional to Q.

Baxandall

A type of first order shelving equalizer typically found in the tone control of high fidelity home audio components.

Butterworth filter

A filter shape designed to have a flat frequency response in the passband.

Component Plug-in

This term is used to describe the different Ozone plug-ins that include a single Ozone processing module. The following Ozone modules are available as component plug-ins: Dynamic EQ, Dynamics, Equalizer, Exciter, Imager, Impact, Low End Focus, Master Rebalance, Match EQ, Maximizer, Spectral Shaper, Stabilizer, Vintage Compressor, Vintage EQ, Vintage Limiter, and Vintage Tape.

Detection circuit

The component that evaluates signal amplitude to trigger a processor.

Envelope

The varying amplitude of sound over time. This can be broken into four stages: attack, decay, sustain, and release.

Fairchild 670

A feedback-based tube compressor/limiter with a soft knee, unique attack, and release envelopes. An IRC I limiter is applied after the tube processing to transparently prevent clipping to the entire bandwidth of the mix.

FFT (Fast Fourier Transform)

A procedure for the calculation of a signal frequency spectrum. The greater the FFT size, the greater the frequency resolution, i.e., notes and tonal events will be clearer at larger sizes.

Filter

Audio filters are frequency dependent amplifier circuits that boost (amplify), cut (attenuate), or pass ranges of the audible frequency spectrum.

HUD (Heads-Up Display)

A panel through which you can access module controls.

MBIT+ Dither

This is a proprietary iZotope word length reduction technology that reduces quantization distortion with minimal perceived noise. While this might sound like a paradox, MBIT+ is a very smooth, quiet, and almost “analog sounding” technology.

Mothership Plug-in

This term is used to describe to the main Ozone plug-in that includes multiple processing modules, a customizable signal chain, Master Assistant, Referencing, Codec Preview, and Dither.

Q

In an equalizer, it is the center frequency divided by bandwidth.

Resonant Filter

Has a complex nature of adjustment such that at the cut off point you can increase or attenuate the resonance. Typically is accompanied by significant phase shift or ringing.

Resonant Peak (Vintage Tape)

In reference to a tape head bump, this describes the prominence of the amount of gain at the low-end frequency at which the tape head bump is being reproduced.

Saturation

A harmonic type of non-linear distortion.

Sidechain

The signal that feeds the detection circuit in a processor.

Spectrum Analyzer

A meter that measures amplitude across the frequencies which encompass the spectrum of human hearing. The vertical axis represents amplitude while the horizontal axis represents frequency.

Tape Saturation

A wavelength (tape speed divided by frequency) dependent process so the exact sound of the saturation is a complex interplay between tape speed, bias-level, program material and other factors.

Waveform

A visual representation of the envelope of the soundwave. (amplitude representation).

License Information

Table of Contents

Abseil

Anti-Grain Geometry

Version 2.4

ARM_NEON_2_x86_SSE

or otherwise) arising in any way out of the use of this software, even if advised of the possibility of such damage.

---

base64

v0.4.0

Redistribution and use in source and binary forms, with or without modification, are permitted provided that the following conditions are met:

1. Redistributions of source code must retain the above copyright notice, this list of conditions and the following disclaimer.

2. Redistributions in binary form must reproduce the above copyright notice, this list of conditions and the following disclaimer in the documentation and/or other materials provided with the distribution.

THIS SOFTWARE IS PROVIDED BY THE COPYRIGHT HOLDERS AND CONTRIBUTORS “AS IS” AND ANY EXPRESS OR IMPLIED WARRANTIES, INCLUDING, BUT NOT LIMITED TO, THE IMPLIED WARRANTIES OF MERCHANTABILITY AND FITNESS FOR A PARTICULAR PURPOSE ARE DISCLAIMED. IN NO EVENT SHALL THE COPYRIGHT HOLDER OR CONTRIBUTORS BE LIABLE FOR ANY DIRECT, INDIRECT, INCIDENTAL, SPECIAL, EXEMPLARY, OR CONSEQUENTIAL DAMAGES (INCLUDING, BUT NOT LIMITED TO, PROCUREMENT OF SUBSTITUTE GOODS OR SERVICES; LOSS OF USE, DATA, OR PROFITS; OR BUSINESS INTERRUPTION) HOWEVER CAUSED AND ON ANY THEORY OF LIABILITY, WHETHER IN CONTRACT, STRICT LIABILITY, OR TORT (INCLUDING NEGLIGENCE OR OTHERWISE) ARISING IN ANY WAY OUT OF THE USE OF THIS SOFTWARE, EVEN IF ADVISED OF THE POSSIBILITY OF SUCH DAMAGE.

Better Enums

Version 0.11.1

Bravura

redistribute, and sell modified and unmodified copies of the Font Software, subject to the following conditions:

1. Neither the Font Software nor any of its individual components, in Original or Modified Versions, may be sold by itself.

2. Original or Modified Versions of the Font Software may be bundled, redistributed and/or sold with any software, provided that each copy contains the above copyright notice and this license. These can be included either as stand-alone text files, human-readable headers or in the appropriate machine-readable metadata fields within text or binary files as long as those fields can be easily viewed by the user.

3. No Modified Version of the Font Software may use the Reserved Font Name(s) unless explicit written permission is granted by the corresponding Copyright Holder. This restriction only applies to the primary font name as presented to the users.

4. The name(s) of the Copyright Holder(s) or the Author(s) of the Font Software shall not be used to promote, endorse or advertise any Modified Version, except to acknowledge the contribution(s) of the Copyright Holder(s) and the Author(s) or with their explicit written permission.

5. The Font Software, modified or unmodified, in part or in whole, must be distributed entirely under this license, and must not be distributed under any other license. The requirement for fonts to remain under this license does not apply to any document created using the Font Software.

TERMINATION
This license becomes null and void if any of the above conditions are not met.

DISCLAIMER
THE FONT SOFTWARE IS PROVIDED “AS IS”, WITHOUT WARRANTY OF ANY KIND, EXPRESS OR IMPLIED, INCLUDING BUT NOT LIMITED TO ANY WARRANTIES OF MERCHANTABILITY, FITNESS FOR A PARTICULAR PURPOSE AND NONINFRINGEMENT OF COPYRIGHT, PATENT, TRADEMARK, OR OTHER RIGHT. IN NO EVENT SHALL THE COPYRIGHT HOLDER BE LIABLE FOR ANY CLAIM, DAMAGES OR OTHER LIABILITY, INCLUDING ANY GENERAL, SPECIAL, INDIRECT, INCIDENTAL, OR CONSEQUENTIAL DAMAGES, WHETHER IN AN ACTION OF

CONTRACT, TORT OR OTHERWISE, ARISING FROM, OUT OF THE USE OR INABILITY TO USE THE FONT SOFTWARE OR FROM OTHER DEALINGS IN THE FONT SOFTWARE.

C++ Rest SDK

Version 2.10.15

Main Library:

Redistribution and use in source and binary forms, with or without modification, are permitted provided that the following conditions are met:

1. Redistributions of source code must retain the above copyright notice, this list of conditions and the following disclaimer.

2. Redistributions in binary form must reproduce the above copyright notice, this list of conditions and the following disclaimer in the documentation and/or other materials provided with the distribution.

3. Neither the name of the WebSocket++ Project nor the names of its contributors may be used to endorse or promote products derived from this software without specific prior written permission.

THIS SOFTWARE IS PROVIDED BY THE COPYRIGHT HOLDERS AND CONTRIBUTORS “AS IS” AND ANY EXPRESS OR IMPLIED WARRANTIES, INCLUDING, BUT NOT LIMITED TO, THE IMPLIED WARRANTIES OF MERCHANTABILITY AND FITNESS FOR A PARTICULAR PURPOSE ARE DISCLAIMED. IN NO EVENT SHALL PETER THORSON BE LIABLE FOR ANY DIRECT, INDIRECT, INCIDENTAL, SPECIAL, EXEMPLARY, OR CONSEQUENTIAL DAMAGES (INCLUDING, BUT NOT LIMITED TO, PROCUREMENT OF SUBSTITUTE GOODS OR SERVICES; LOSS OF USE, DATA, OR PROFITS; OR BUSINESS INTERRUPTION) HOWEVER CAUSED AND ON ANY THEORY OF LIABILITY, WHETHER IN CONTRACT, STRICT LIABILITY, OR TORT (INCLUDING NEGLIGENCE OR OTHERWISE) ARISING IN ANY WAY OUT OF THE USE OF THIS SOFTWARE, EVEN IF ADVISED OF THE POSSIBILITY OF SUCH DAMAGE.

Bundled Libraries:

****** Base 64 Library (base64/base64.hpp) ******
base64.hpp is a repackaging of the base64.cpp and base64.h files into a single header suitable for use as a header only library. This conversion was done by Peter Thorson (**webmaster@zaphoyd.com**) in 2012. All modifications to the code are redistributed under the same license as the original, which is listed below.

base64.cpp and base64.h

****** SHA1 Library (sha1/sha1.hpp) ******

sha1.hpp is a repackaging of the sha1.cpp and sha1.h files from the shallsha1 library (**http://code.google.com/p/smallsha1/**) into a single header suitable for use as a header only library. This conversion was done by Peter Thorson (**webmaster@zaphoyd.com**) in 2013. All modifications to the code are redistributed under the same license as the original, which is listed below.

****** MD5 Library (common/md5.hpp) ******
md5.hpp is a reformulation of the md5.h and md5.c code from **http://www.opensource.apple.com/source/cups/cups-59/cups/md5.c** to allow it to function as a component of a header only library. This conversion was done by Peter Thorson (**webmaster@zaphoyd.com**) in 2012 for the WebSocket++ project. The changes are released under the same license as the original (listed below)

****** UTF8 Validation logic (utf8_validation.hpp) ******
utf8_validation.hpp is adapted from code originally written by Bjoern Hoehrmann **bjoern@hoehrmann.de**. See **http://bjoern.hoehrmann.de/utf-8/decoder/dfa/** for details.

The original license:

CONNECTION WITH THE SOFTWARE OR THE USE OR OTHER DEALINGS IN THE SOFTWARE.

CpuInfo

Redistribution and use in source and binary forms, with or without modification, are permitted provided that the following conditions are met:

1. Redistributions of source code must retain the above copyright notice, this list of conditions and the following disclaimer.

2. Redistributions in binary form must reproduce the above copyright notice, this list of conditions and the following disclaimer in the documentation and/or other materials provided with the distribution.

THIS SOFTWARE IS PROVIDED BY THE COPYRIGHT HOLDERS AND CONTRIBUTORS “AS IS” AND ANY EXPRESS OR IMPLIED WARRANTIES, INCLUDING, BUT NOT LIMITED TO, THE IMPLIED WARRANTIES OF MERCHANTABILITY AND FITNESS FOR A PARTICULAR PURPOSE ARE DISCLAIMED. IN NO EVENT SHALL THE COPYRIGHT HOLDER OR CONTRIBUTORS BE LIABLE FOR ANY DIRECT, INDIRECT, INCIDENTAL, SPECIAL, EXEMPLARY, OR CONSEQUENTIAL DAMAGES (INCLUDING, BUT NOT LIMITED TO, PROCUREMENT OF SUBSTITUTE GOODS OR SERVICES; LOSS OF USE, DATA, OR PROFITS; OR BUSINESS INTERRUPTION) HOWEVER CAUSED AND ON ANY THEORY OF LIABILITY, WHETHER IN CONTRACT, STRICT LIABILITY, OR TORT (INCLUDING NEGLIGENCE OR OTHERWISE) ARISING IN ANY WAY OUT OF THE USE OF THIS SOFTWARE, EVEN IF ADVISED OF THE POSSIBILITY OF SUCH DAMAGE.

Crashpad

Version 0.8.0

Eigen

Version 3.4.99

Distributed under the **Mozilla Public License v2.0 (MPLv2.0)**.

Full text of the license is available here: **https://www.mozilla.org/en-US/MPL/2.0/**

To receive a copy of the source code for the Eigen library distributed with this product under the under the terms of the MPLv2.0 please contact **dev-support@izotope.com**.

FarmHash

FlatBuffers

FLAC

libFLAC and libFLAC++

Version 1.3.2

FreeType

Version 2.4.6

---

gemmlowp

GLEW

Mesa 3-D graphics library Version: 7.0

---

gsl

CONNECTION WITH THE SOFTWARE OR THE USE OR OTHER DEALINGS IN THE SOFTWARE.

IcoMoon

IcoMoon-Free licensed under: **CC BY 4.0**

Intel® Integrated Performance Primitives (Intel® IPP)

Version 2019.0.5

LIMITATION OF LIABILITY. IN NO EVENT WILL INTEL BE LIABLE FOR ANY DIRECT, INDIRECT, INCIDENTAL, SPECIAL, EXEMPLARY, OR CONSEQUENTIAL DAMAGES (INCLUDING, BUT NOT LIMITED TO, PROCUREMENT OF SUBSTITUTE GOODS OR SERVICES; LOSS OF USE, DATA, OR PROFITS; OR BUSINESS INTERRUPTION) HOWEVER CAUSED AND ON ANY THEORY OF LIABILITY, WHETHER IN CONTRACT, STRICT LIABILITY, OR TORT (INCLUDING NEGLIGENCE OR OTHERWISE) ARISING IN ANY WAY OUT OF THE USE OF THIS SOFTWARE, EVEN IF ADVISED OF THE POSSIBILITY OF SUCH DAMAGE. YOU AGREE TO INDEMNIFY AND HOLD INTEL HARMLESS AGAINST ANY CLAIMS AND EXPENSES RESULTING FROM YOUR USE OR UNAUTHORIZED USE OF THE SOFTWARE.

No support. Intel may make changes to the Software, at any time without notice, and is not obligated to support, update or provide training for the Software.

Termination. Intel may terminate your right to use the Software in the event of your breach of this Agreement and you fail to cure the breach within a reasonable period of time.

Feedback. Should you provide Intel with comments, modifications, corrections, enhancements or other input (“Feedback”) related to the Software Intel will be free to use, disclose, reproduce, license or otherwise distribute or exploit the Feedback in its sole discretion without any obligations or restrictions of any kind, including without limitation, intellectual property rights or licensing obligations.

Compliance with laws. You agree to comply with all relevant laws and regulations governing your use, transfer, import or export (or prohibition thereof) of the Software.

Governing law. All disputes will be governed by the laws of the United States of America and the State of Delaware without reference to conflict of law principles and subject to the exclusive jurisdiction of the state or federal courts sitting in the State of Delaware, and each party agrees that it submits to the personal jurisdiction and venue of those courts and waives any objections. The United Nations Convention on Contracts for the International Sale of Goods (1980) is specifically excluded and will not apply to the Software.

Other names and brands may be claimed as the property of others.

JsonCpp

Version 1.2.1

LAME

libmp3lame

Version 3.99.5

To receive a copy of the source code for the LAME library distributed with this product under the under the terms of the LGPLv2 please contact **dev-support@izotope.com**.

LibXML2

Version 2.7.8

Material Docs Theme

NLohmann JSON

v3.10.4

Netlib numeralgo na10 Aberth’s method

ANY USE OF THE SOFTWARE CONSTITUTES ACCEPTANCE OF THE TERMS OF THE ABOVE STATEMENT.

AUTHOR:

1. DARIO ANDREA
   UNIVERSITY OF PISA, ITALY
   E-MAIL: **bini@dm.unipi.it**

REFERENCE:

1. NUMERICAL COMPUTATION OF POLYNOMIAL ZEROS BY MEANS OF ABERTH'S METHOD NUMERICAL ALGORITHMS, 13 (1996), PP. 179-200

SOFTWARE REVISION DATE:

1. JUNE, 1996

SOFTWARE LANGUAGE:

1. FORTRAN

OGG / Vorbis

libogg and libvorbis

Version 1.3.2

documentation and/ or other materials provided with the distribution.

3. Neither the name of the Xiph.org Foundation nor the names of its contributors may be used to endorse or promote products derived from this software without specific prior written permission.

THIS SOFTWARE IS PROVIDED BY THE COPYRIGHT HOLDERS AND CONTRIBUTORS “AS IS” AND ANY EXPRESS OR IMPLIED WARRANTIES, INCLUDING, BUT NOT LIMITED TO, THE IMPLIED WARRANTIES OF MERCHANTABILITY AND FITNESS FOR A PARTICULAR PURPOSE ARE DISCLAIMED. IN NO EVENT SHALL THE FOUNDATION OR CONTRIBUTORS BE LIABLE FOR ANY DIRECT, INDIRECT, INCIDENTAL, SPECIAL, EXEMPLARY, OR CONSEQUENTIAL DAMAGES (INCLUDING, BUT NOT LIMITED TO, PROCUREMENT OF SUBSTITUTE GOODS OR SERVICES; LOSS OF USE, DATA, OR PROFITS; OR BUSINESS INTERRUPTION) HOWEVER CAUSED AND ON ANY THEORY OF LIABILITY, WHETHER IN CONTRACT, STRICT LIABILITY, OR TORT (INCLUDING NEGLIGENCE OR OTHERWISE) ARISING IN ANY WAY OUT OF THE USE OF THIS SOFTWARE, EVEN IF ADVISED OF THE POSSIBILITY OF SUCH DAMAGE.

---

readerwriterqueue

Copyright (c) 2013-2015, Cameron Desrochers All rights reserved.

Redistribution and use in source and binary forms, with or without modification, are permitted provided that the following conditions are met:

1. Redistributions of source code must retain the above copyright notice, this list of conditions and the following disclaimer.
2. Redistributions in binary form must reproduce the above copyright notice, this list of conditions and the following disclaimer in the documentation and/or other materials provided with the distribution.

THIS SOFTWARE IS PROVIDED BY THE COPYRIGHT HOLDERS AND CONTRIBUTORS “AS IS” AND ANY EXPRESS OR IMPLIED WARRANTIES, INCLUDING, BUT NOT LIMITED TO, THE IMPLIED WARRANTIES OF MERCHANTABILITY AND FITNESS FOR A PARTICULAR PURPOSE ARE DISCLAIMED. IN NO EVENT SHALL THE COPYRIGHT HOLDER OR

Roboto font family

---

ruy

Skia

TagLib

Version 1.9.1

To receive a copy of the source code for the TagLib library distributed with this product under the terms of the MPLv1.1 please contact **dev-support@izotope.com**.

TensorFlow

Version 2.1.0

TinyXML

Tipue Search

---

vectorize

WebView2

---

xsimd

Yoga

Version 1.9.0

MIT License

ZeroMQ

To receive a copy of the source code for the ZeroMQ library distributed with this product under the under the terms of the LGPLv3 please contact **dev-support@izotope.com**.

---

zlib

Jean-loup Gailly **jloup@gzip.org**

Mark Adler **madler@alumni.caltech.edu**