

VST RACK

VST Rack Plug-in Reference

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About VST Rack Pro/Elements

- VST Rack Pro/Elements is a plug-in host software product for Mac and Windows PCs that processes audio using a diverse range of VST plug-in effects such as delays, maximizers, reverbs and more. This software uses the widely recognized Steinberg audio engine featured in Nuendo and Cubase, offering sound engineers a stable VST plug-in environment with high audio quality.
- VST Rack Pro is bundled with classic Yamaha and Steinberg VST plug-ins that have been used in Yamaha digital mixers and Steinberg digital audio workstations. VST Rack Elements is bundled with many classic Steinberg VST plug-ins. This provides you access to high-quality effects without any need to purchase new VST plug-ins.

Manuals

- **VST Rack Installation Guide (PDF)**
Describes how to install VST Rack.
- **VST Rack User's Guide (PDF/HTML)**
Describes all of the items that are required to configure and operate VST Rack.
- **VST Rack Plug-in Reference (this manual)**
Describes the plug-in parameters in detail.

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- This manual provides the latest specifications as of the time of its publication. The latest version is available for download on the Yamaha website.

List of Plug-ins

Steinberg / Yamaha



Steinberg plug-ins can only be used in VST Rack.

		VST Rack Pro	VST Rack Elements
Analyzer	SuperVision	✓	
Delay	MonoDelay	✓	✓
	PingPongDelay	✓	
	StereoDelay	✓	✓
Distortion	AmpSimulator	✓	
	Magneto II	✓	
	Quadrafuzz v2	✓	
	SoftClipper	✓	
	VST Amp Rack	✓	
Dynamics	DeEsser	✓	
	EnvelopeShaper	✓	
	Maximizer	✓	✓
	MultibandCompressor	✓	
	MultibandEnvelopeShaper	✓	
	Tube Compressor	✓	
	VintageCompressor	✓	
EQ	GEQ-30	✓	✓
	StudioEQ	✓	
Filter	Morph Filter	✓	
Modulation	Cloner	✓	
	FX Modulator	✓	
	Rotary	✓	
	StudioChorus	✓	
Pitch Shift	VoiceDesigner	✓	
Reverb	REVelation	✓	
	REVerence	✓	
	Roomworks	✓	
	Roomworks SE	✓	✓

Spatial + Panner	StereoEnhance	✓	
Tools	TestGenerator	✓	✓

Yamaha



Yamaha plug-ins can be used in host applications other than VST Rack, such as Cubase.

		VST Rack Pro	VST Rack Elements
Delay	Analog Delay	✓	
Dynamics	Buss Comp 369	✓	
	Compressor 260	✓	
	Compressor 276	✓	
	MBC4	✓	
EQ	Dynamic EQ	✓	
	EQ-1A	✓	
	Equalizer	✓	
Filter	FBS 7	✓	
Mastering	Vintage Open Deck	✓	
Reverb	REV-X	✓	✓

Analyzer Plug-ins

SuperVision

SuperVision is a professional tool suite for monitoring and analyzing your audio. The plug-in comes with several different modules for level, spectral, phase, or waveform analysis. Up to nine module slots allow you to create custom layouts for better overview.

SuperVision features two different processing modes: Maximum Audio Performance and Sample-Accurate Display. You can choose which mode to use for each module independently.



Toolbar

Pause Measurement



Pauses/Continues the measurement for the selected module. Alt/Opt -click this button to pause/continue all modules at the same time.

NOTE

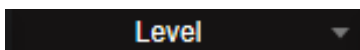
- You can also pause/continue the measurement for the selected module by rightclicking it.
- If a module is paused, you can still adjust the graphical display that holds the last measured values.
- For all modules that show a playback cursor, you can click on the paused display to set the project cursor to this position. This is not possible during recording.

Hold Current Values on Stop



If this button is activated, the last measured values remain on the display when playback is stopped.

Module selector



Allows you to select a module for the selected slot.

Open Module Settings



Opens the Module Settings window. It provides the settings for the selected module.

Reset Module Values



Resets the measured values of the selected module. Alt/Opt -click to reset all modules at the same time.

NOTE

You can also reset the measured values of a module by Ctrl/Cmd -clicking it.

Reset Module Values on Start



If this button is activated, all values are automatically reset when you start playback.

Split Horizontally



Splits the selected module slot horizontally.

NOTE

This button is not available if a module is maximized.

Split Vertically



Splits the selected module slot vertically.

NOTE

This button is not available if a module is maximized.

Module Slot Controls

Each module slot shows the following controls in the top right corner if you move the mouse over it:

Remove module slot



Removes the module slot from the current plug-in layout.

Split horizontally




Splits the module slot horizontally.

Split vertically



Splits the module slot vertically.

You can maximize a module by double-clicking it. To downsize it, double-click it again or click the standard view button .

If your layout shows more than one module, you can change the focus by clicking a module or pressing Tab.


In some modules, for example, Level, Loudness, or Time, you can press Ctrl/Cmd - S to copy parameter values as

text from the selected module to the clipboard for further use in other applications.

You can show the current frames per second (fps) for all modules by pressing Alt/Opt - F.

Module Settings Window

In the Module Settings window, you can make individual settings for the selected module.

- To open the Module Settings window, click Open Module Settings  on the plug-in toolbar. The settings on the Module Settings window toolbar are available for all modules:

Reset Settings



Resets all parameter settings to the default values of the selected module.

Maximum Audio Performance/Sample-Accurate Display



Sets the processing mode for the selected module.

If this button is activated, Maximum Audio Performance is selected. In this mode, the plug-in has no impact on the audio performance, but the analysis might not be sample-accurate.

If this button is deactivated, Sample-Accurate Display is selected. In this mode, no audio sample gets lost for analysis, but the audio performance might be slightly reduced.

NOTE

Sample-Accurate Display is not available for all modules.

Enable Warnings



If this button is activated, a red border around the affected module indicates that the displayed analysis result might not be completely sample-accurate.

NOTE

This setting is only available in Maximum Audio Performance mode.

Force Horizontal Display



If this button is activated, the module is always displayed horizontally when you resize it.

NOTE

This setting is not available for all modules.

Force Vertical Display



If this button is activated, the module is always displayed vertically when you resize it.

NOTE

This setting is not available for all modules.

For the specific settings of a module, see the corresponding module description.

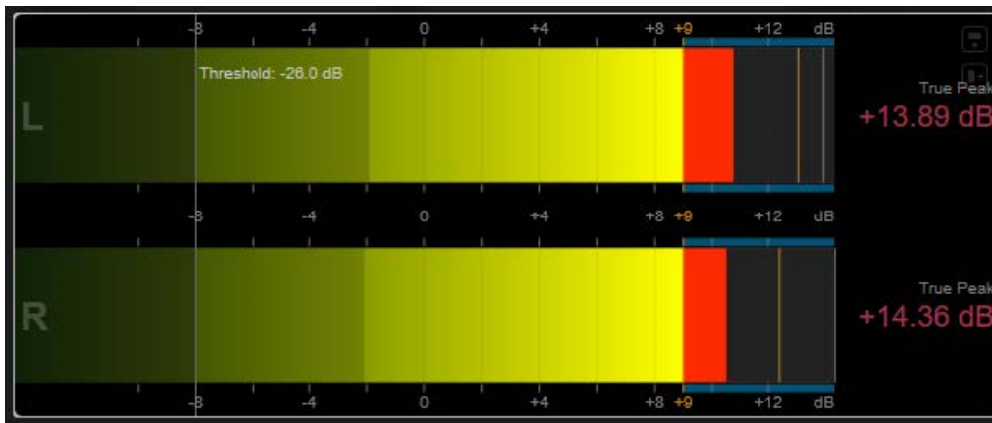
Signal Modules

The modules in this category allow you to visualize the level of the audio signal.

The following modules and module-specific settings are available:

Level

This module shows the level of your audio. It provides a multi-channel level meter and a maximum level value display.



The following module-specific settings are available in the **Module Settings** window:

Scale

Allows you to select a scale according to different broadcast standards (Internal, Digital, DIN, EBU, British, Nordic, K-20, K-14, K-12, +3 dB Digital, +6 dB Digital, or +12 dB Digital).

Peak Hold

Specifies for how long the peak levels are held on the display.

Peak Fallback

Sets the speed of release for the level meters and peak indicators.

NOTE

- Alternatively, you can move the mouse pointer over the selected module, hold Ctrl/Cmd , and use the mouse wheel to adjust this parameter.
- If this control is turned all the way to the left, the peak indicators are disabled.

Threshold

Sets a threshold level below which the display is masked.

NOTE

Alternatively, you can move the mouse pointer over the selected module and use the mouse wheel to adjust this parameter.

Offset

Sets the offset between measured and displayed value in dB.
This parameter is only available for the DIN, EBU, British, and Nordic scale.

Clipping

Sets the clipping value for the Internal scale.

Minimum

Sets the minimum value for the Internal scale.

Maximum

Sets the maximum value for the Internal scale.

Color

Sets the color of the meters. You can choose between the Scale color and the Track color.

RMS AES17

Shows the level in accordance with AES17 (RMS +3 dB).

RMS Resolution

Sets the RMS resolution for the level display in milliseconds.

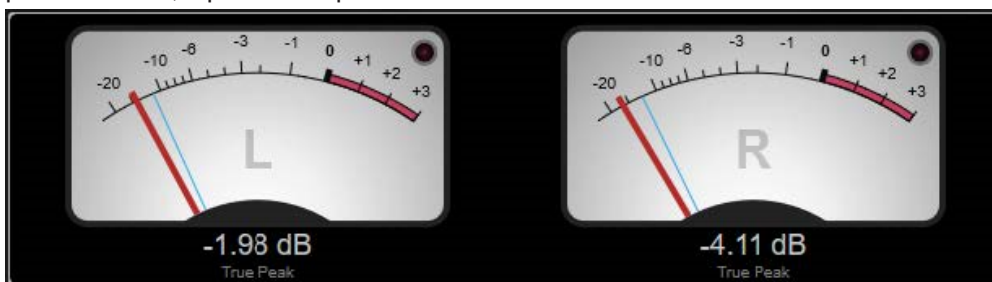
Max. Value

Sets the measurement mode for the maximum level value display. The following modes are available:

- True Peak shows estimated real peak value for each channel.
- Peak Max. shows maximum sample value for each channel.
- RMS Max. shows the maximum RMS value for each channel.
- RMS Max. + True Peak shows the highest maximum RMS and the highest estimated real peak value of all channels.
- RMS Max. + Peak Max. shows the highest maximum RMS and the highest maximum sample value of all channels.

VU

This module shows the level of your audio on a classic VU meter. In addition to the VU meter needle and the LED peak indicator, it provides a peak level indicator needle and a numeric maximum level value display.



The following module-specific settings are available in the **Module Settings** window:

Scale

Allows you to select a scale according to different broadcast standards (Internal, Digital, DIN, EBU, British, Nordic, K-20, K-14, K-12, +3 dB Digital, +6 dB Digital, +12 dB Digital, VU dB, and VU dBFS).

Peak Hold

Specifies for how long the peak levels are held on the display.

Peak Fallback

Sets the speed of release for the level meters and peak indicators.

NOTE

- Alternatively, you can move the mouse pointer over the selected module, hold Ctrl/Cmd, and use the mouse wheel to adjust this parameter.
- If this control is turned all the way to the left, the peak indicators are disabled.

Meter Mode

Sets the behavior of the needle.

- VU mode imitates the physical behavior of an analog VU meter that shows the current peak value.
- Peak mode shows the current peak value.
- RMS mode shows the current RMS value.

Offset

Sets the offset between measured and displayed value in dB.

This parameter is only available for the DIN, EBU, British, and Nordic scale.

Clipping

Sets the clipping value for the Internal scale.

Minimum

Sets the minimum value for the Internal scale.

Maximum

Sets the maximum value for the Internal scale.

Color

Sets the color of the meters. You can choose between the Track color and a Dark or Light scheme.

RMS AES17

Shows the level in accordance with AES17 (RMS + 3 dB).

RMS Resolution

Sets the RMS resolution for the level display in milliseconds.

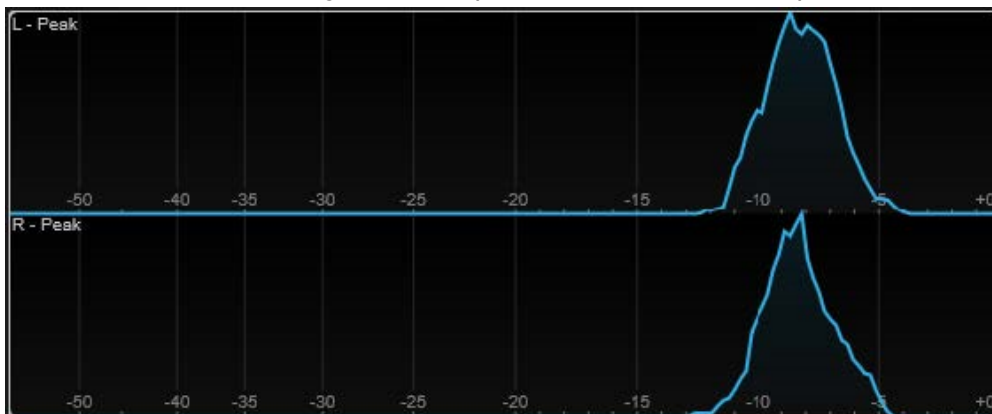
Max. Value

Sets the measurement mode for the maximum level value display. The following modes are available:

- True Peak shows the estimated real peak value for each channel.
- Peak Max. shows the maximum sample value for each channel.
- RMS Max. shows the maximum RMS value for each channel.
- RMS Max. + True Peak shows the highest maximum RMS and the highest estimated real peak value of all channels.
- RMS Max. + Peak Max. shows the highest maximum RMS and the highest maximum sample value of all channels.

Level Histogram

This module shows a histogram for the peak or RMS value of the input level.



The following module-specific settings are available in the **Module Settings** window:

Scale

Allows you to select a scale according to different broadcast standards (Internal, Digital, DIN, EBU, British, Nordic, K-20, K-14, K-12, +3 dB Digital, +6 dB Digital, or +12 dB Digital).

Meter Mode

Sets the displayed level value.

- Peak mode shows a histogram of the peak value.
- RMS mode shows a histogram of the RMS value.

Peak Fallback

Sets the speed of release for the level meters and peak indicators.

NOTE

- If you change this parameter during playback, you must click Reset Module Values to update the display.
- Alternatively, you can move the mouse pointer over the selected module, hold Ctrl/Cmd, and use the mouse wheel to adjust this parameter.
- If this control is turned all the way to the left, the peak indicators are disabled.

Offset

Sets the offset between measured and displayed value in dB.
This parameter is only available for the DIN, EBU, British, and Nordic scale.

Clipping

Sets the clipping value for the Internal scale.

Minimum

Sets the minimum value for the Internal scale.

Maximum

Sets the maximum value for the Internal scale.

RMS AES17

Shows the level in accordance with AES17 (RMS +3 dB).

RMS Resolution

Sets the RMS resolution for the level display in milliseconds.

Smooth

Smooths the display of the level curve.

NOTE

Alternatively, you can move the mouse pointer over the selected module and use the mouse wheel to adjust this parameter.

Measurement Modules

The modules in this category allow you to measure the loudness and intelligibility of the audio signal. The following modules and module-specific settings are available:

Loudness

This module shows the loudness of your audio in LU (Loudness Units) or LUFS (Loudness Units referenced to Full Scale) according to EBU R 128.



TP (True Peak)

Shows the maximum true peak level in dB.

I (Integrated)

Shows the integrated loudness value. This is the average loudness value that is measured over the whole audio range in LU or LUFS.

S (Short-Term)

Shows the short-term loudness value that is measured every second on an audio block of 3 seconds in LU or LUFS. This gives information about the loudest audio passages.

M Max.(Momentary Max.)

Shows the maximum value of all momentary loudness values that are measured every 100 ms in an audio range of 400 ms in LU or LUFS.

R (Range)

Shows the loudness range (LRA) that is measured over the whole audio range in LU. The loudness range reports the ratio between the loudest and the quietest non-silent sections. The audio is divided into small blocks. There is one audio block every second and each block lasts 3 seconds, so that the analyzed blocks overlap. The top 10 % of the quiet blocks and the top 5 % of the loud blocks are excluded from the final analysis. The calculated loudness range is the ratio between the loudest and quietest remaining audio blocks. This measurement helps you to decide how much compression or expansion must be applied to the audio.

An asterisk (*) after a loudness range value indicates that less than 1 minute of audio was analyzed.

Min. shows the minimum loudness range value in LU. Max. shows the maximum loudness range value in LU.

***NOTE**

EBU R 128 does not recommend loudness range measurement for audio shorter than 1 minute due to too few data points.

PLR (Peak-to-Loudness)

Shows the peak-to-loudness ratio (PLR), also referred to as the crest factor, which is the difference between the maximum true peak level value and the integrated loudness value.

Time

Shows the overall duration of the loudness measurement.

The following module-specific settings are available in the Module Settings window:

Unit

Allows you to switch the meter scale between LUFS (absolute values) and LU (relative values).

Scale

Allows you to set the meter to either EBU +9 scale (linear), EBU +18 scale (linear), or +23 scale (logarithmic).

Ref.Integrated

Sets a reference value for the integrated loudness. If higher values are detected, the loudness meter indicates clipping.

Tol.Integrated

Sets a tolerance value for the integrated loudness.

Ref.True Peak

Sets a reference value for the true peak level. If higher values are detected, the loudness meter indicates clipping.

Tol.True Peak

Sets a tolerance value for the true peak level.

Ref.Short-Term

Sets a reference value for the short-term loudness. If higher values are detected, the loudness meter indicates clipping.

Tol.Short-Term

Sets a tolerance value for the short-term loudness.

Ref.Momentary

Sets a reference value for the maximum momentary loudness. If higher values are detected, the loudness meter indicates clipping.

Tol.Momentary

Sets a tolerance value for the maximum momentary loudness.

Ref.Range

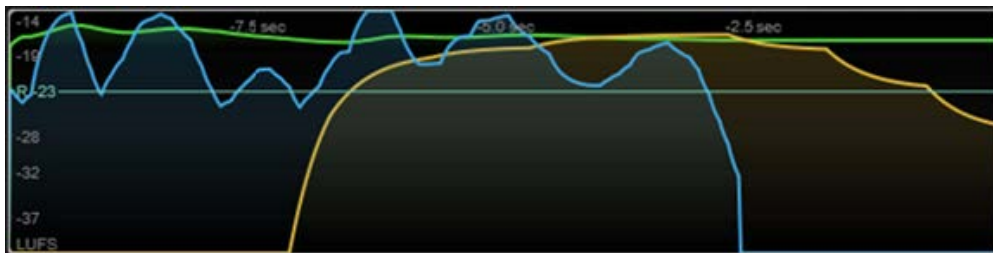
Sets a reference value for the loudness range. If higher values are detected, the loudness meter indicates clipping.

Tol.Range

Sets a tolerance value for the loudness range.

Loudness Curve

This module shows loudness values as a time curve.



The following module-specific settings are available in the **Module Settings** window:

Duration

Sets the duration of the audio stream that is displayed.

NOTE

Alternatively, you can move the mouse pointer over the selected module, hold Ctrl/Command, and use the mouse wheel to adjust this parameter.

Unit

Allows you to switch the meter scale between LUFS (absolute values) and LU (relative values).

Scale

Allows you to set the meter to either EBU +9 scale (linear), EBU +18 scale (linear), or +23 scale (logarithmic).

Smooth

Smooths the display of the loudness curve.

Momentary

Shows/Hides the maximum momentary loudness curve.

Short-Term

Shows/Hides the short-term loudness curve.

Integrated

Shows/Hides the integrated loudness curve.

Range

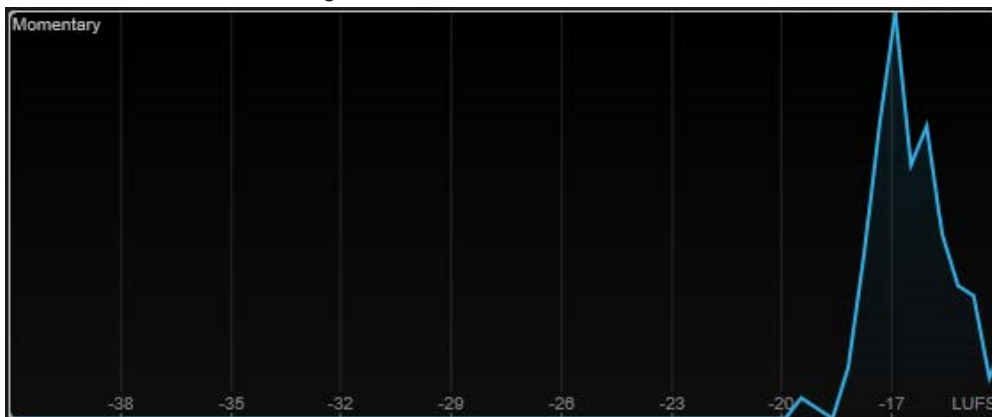
Shows/Hides a gray area around the integrated loudness curve that represents the loudness range.

Ref.Integrated

Sets a reference value for the integrated loudness.

Loudness Histogram

This module shows a histogram for the loudness or the loudness ratio value.



The following module-specific settings are available in the **Module Settings** window:

Unit

Allows you to switch the meter scale between LUFS (absolute values) and LU (relative values).

Scale

Allows you to set the meter to either EBU +9 scale (linear), EBU +18 scale (linear), or +23 scale (logarithmic).

Meter Mode

Sets the displayed loudness or loudness ratio value.

- Momentary mode shows a histogram of the maximum value of all momentary loudness values that are measured every 100 ms in an audio range of 400 ms.
- Short-Term shows a histogram of the short-term loudness value that is measured every second on an audio block of 3 seconds.
- Integrated shows a histogram of the integrated loudness value.
- PLR shows a histogram of the peak-to-loudness ratio, also referred to as the crest factor, which is the difference between the maximum true peak level value and integrated loudness value.
- PSR shows a histogram of the peak-to-short-term-loudness ratio according to AES Convention e-Brief 373.

Smooth

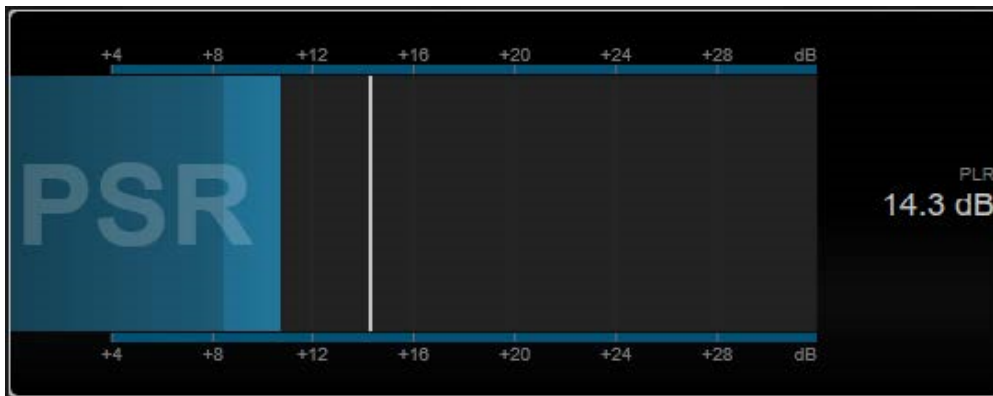
Smooths the display of the loudness curve.

NOTE

Alternatively, you can move the mouse pointer over the selected module and use the mouse wheel to adjust this parameter.

Loudness Ratio

This module shows the peak-to-loudness ratio (PLR) and the peak-to-short-term-loudness ratio (PSR) values according to AES specifications.



PSR

Shows the peak-to-short-term-loudness ratio according to AES Convention e-Brief 373. The darker area of the meter indicates the minimum PSR value.

PLR

Shows the peak-to-loudness ratio, also referred to as the crest factor, which is the difference between the maximum true peak level value and integrated loudness value. The current PLR value is displayed numerically and also indicated by a thin bar on the meter.

The following module-specific setting is available in the Module Settings window:

Ref.Level

Sets the reference level below which the PSR display turns red.

Time Smooth

Smooths the temporal display of the PSR value.

NOTE

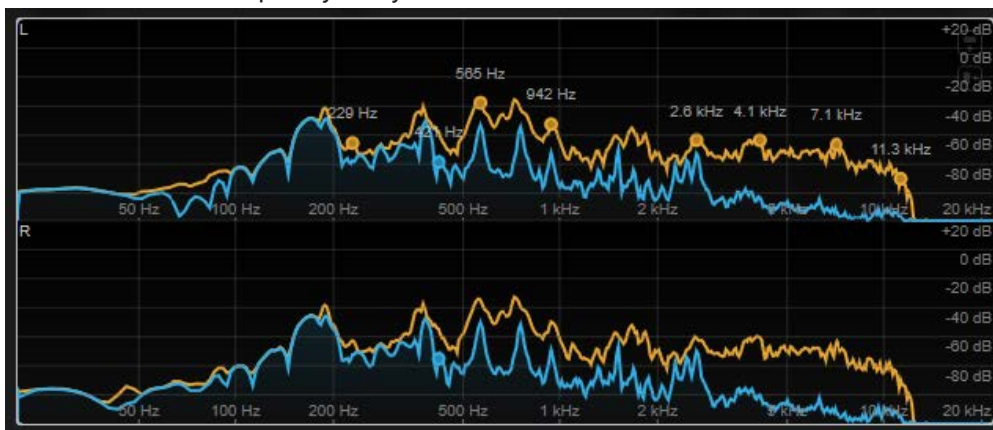
Alternatively, you can move the mouse pointer over the selected module, hold Alt , and use the mouse wheel to adjust this parameter.

Spectral Domain Modules

The modules in this category allow you to visualize spectral information about the audio signal. The following modules and module-specific settings are available:

Spectrum Curve

This module uses FFT (Fast Fourier Transform) techniques to show a frequency graph, providing a precise and detailed real-time frequency analysis.



The display shows the frequency spectrum as a linear graph. If you move the mouse pointer over the display, a peak curve is shown in orange. Move the mouse pointer over the curves to show the local maximum values in Hz. Press Ctrl/Cmd to show the maximum values in dB or press Shift to show their pitch.

The following module-specific settings are available in the Module Settings window:

Time Smooth

Smooths the temporal display.

NOTE

Alternatively, you can move the mouse pointer over the selected module, hold Alt , and use the mouse wheel to adjust this parameter.

Peak Fallback

Sets the speed of release for the spectrum curve and the peak curve.

NOTE

- Alternatively, you can move the mouse pointer over the selected module, hold Ctrl/Cmd , and use the mouse wheel to adjust this parameter.
- If this control is turned all the way to the left, the peak curve is disabled.

Freq.Smooth

Smooths the frequency display of the spectrum curve.

NOTE

Alternatively, you can move the mouse pointer over the selected module and use the mouse wheel to adjust this parameter.

FFT Window

Sets the block size of the window that is used for the analysis. If Multi is selected, 3 different block sizes are used at the same time.

Minimum

Sets the minimum value of the scale.

Maximum

Sets the maximum value of the scale.

Slope

Adds a slope to the frequency spectrum.

Masking

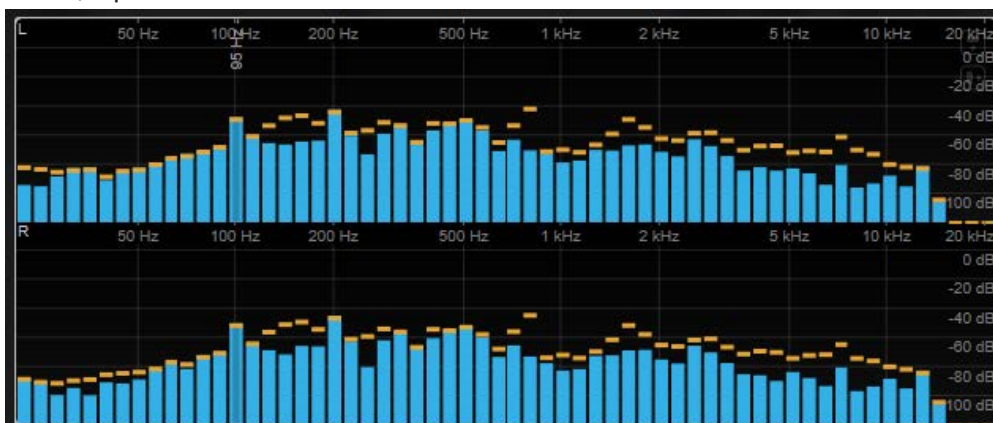
If this button is activated, frequency ranges that are affected by a side-chain signal are displayed.

NOTE

VST Rack does not support side-chaining. Consequently, this plug-in does not accept side-chain input.

Spectrum Bar

This module shows a graphical representation of the frequency spectrum, analyzed into separate frequency bands, represented as vertical bars.



Move the mouse pointer over a bar to show the frequency range in Hz. Press Ctrl/Cmd to show the current value in dB or press Shift to show its pitch range.

The following module-specific settings are available in the **Module Settings** window:

Time Smooth

Smooths the temporal display.

NOTE

Alternatively, you can move the mouse pointer over the selected module, hold Alt, and use the mouse wheel to adjust this parameter.

Peak Fallback

Sets the speed of release for the level meters and peak indicators.

NOTE

- Alternatively, you can move the mouse pointer over the selected module, hold Ctrl/Cmd, and use the mouse wheel to adjust this parameter.
- If this control is turned all the way to the left, the peak indicators are disabled.

Threshold

Sets a threshold level below which the display is masked.

NOTE

Alternatively, you can move the mouse pointer over the selected module and use the mouse wheel to adjust this parameter.

Bands/Oct.

Sets the number of bands per octave.

Minimum

Sets the minimum value of the scale.

Maximum

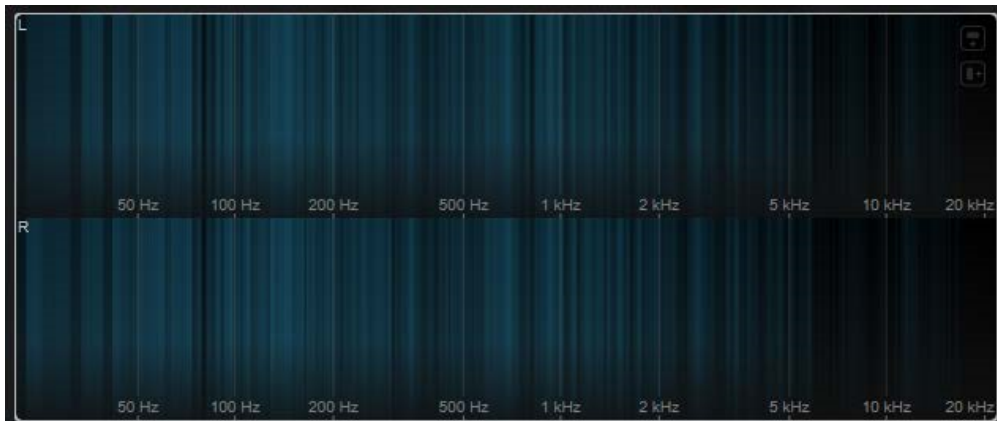
Sets the maximum value of the scale.

Slope

Adds a slope to the frequency spectrum.

Spectrum Intensity

This module represents the frequency magnitude of the audio. The more intensely a bar is colored, the higher the magnitude at this frequency.



The following module-specific settings are available in the **Module Settings** window:

Time Smooth

Smooths the temporal display.

NOTE

Alternatively, you can move the mouse pointer over the selected module, hold Alt , and use the mouse wheel to adjust this parameter.

FFT Window

Sets the block size of the window that is used for the analysis. If Multi is selected, 3 different block sizes are used at the same time.

Color

Allows you to choose a color scheme.

Minimum

Sets the minimum value of the scale.

Maximum

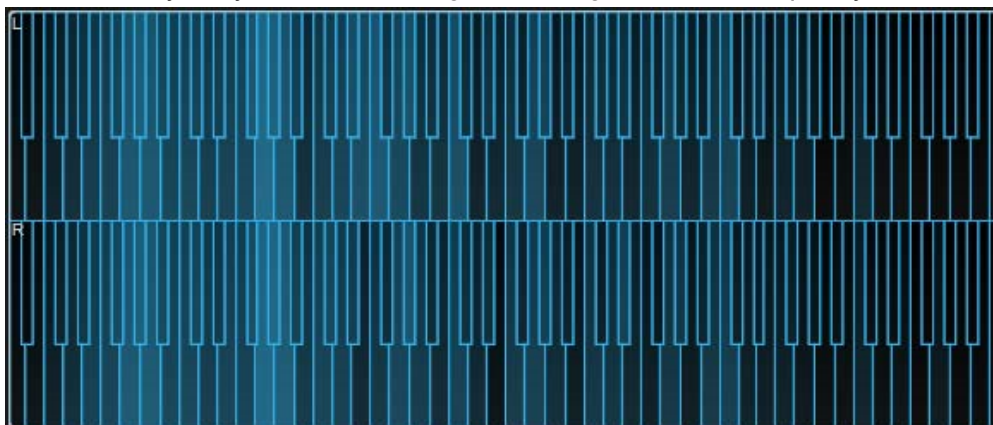
Sets the maximum value of the scale.

Slope

Adds a slope to the frequency spectrum.

Spectrum Keyboard

This module represents the frequency magnitude of the audio mapped to the keys of a piano keyboard. The more intensely a key is colored, the higher the magnitude at its frequency.



The following module-specific settings are available in the **Module Settings** window:

Time Smooth

Smooths the temporal display.

NOTE

Alternatively, you can move the mouse pointer over the selected module, hold Alt , and use the mouse wheel to adjust this parameter.

Color

Allows you to choose a color scheme.

Minimum

Sets the minimum value of the scale.

Maximum

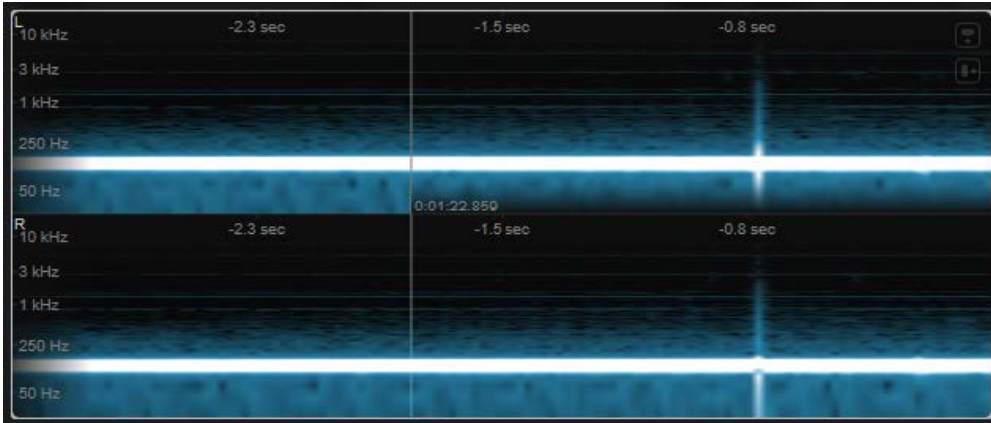
Sets the maximum value of the scale.

Slope

Adds a slope to the frequency spectrum.

Spectrogram

This module shows the last seconds of the audio stream. This allows you to detect disturbances in the spectrogram and to monitor the noise level and frequencies, for example.



NOTE

This module runs in Maximum Audio Performance mode.

The following module-specific settings are available in the **Module Settings** window:

FFT Window

Sets the block size of the window that is used for the analysis. This allows you to adjust the trade-off between temporal resolution and frequency resolution. If you specify a higher value, more frequencies are analyzed but they are located less accurately in the time domain.

Duration

Sets the duration of the audio stream that is displayed.

NOTE

Alternatively, you can move the mouse pointer over the selected module, hold Ctrl/Command, and use the mouse wheel to adjust this parameter.

Color

Allows you to choose a color scheme.

Minimum

Sets the minimum value of the scale.

Sets the maximum value of the scale.

NOTE

Alternatively, you can move the mouse pointer over the selected module and use the mouse wheel to adjust the Minimum and Maximum parameters simultaneously.

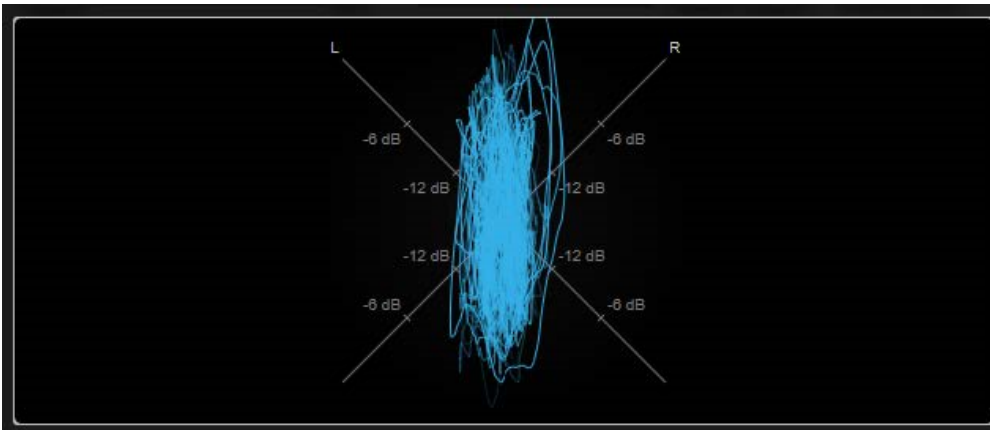
Phase Modules

The modules in this category allow you to visualize the phase or directional relationship between the channels of the audio signal.

The following modules and module-specific settings are available:

Phasescope

This module uses a vectorscope display to show the phase and amplitude relationship between the left and right stereo channels. This provides you with directional information about a stereo audio signal.



Hold Shift and move the mouse pointer over the display to measure the angle.

NOTE

In addition to the signal of the track, the display can show the signal of a side-chain input. For this to work, you must select a Main & Side-Chain view from the channel selector. The side-chain signal is then shown in white.

The following module-specific settings are available in the **Module Settings** window:

Zoom

Allows you to zoom on the graphical display.

NOTE

Alternatively, you can move the mouse pointer over the selected module and use the mouse wheel to adjust this parameter.

Auto Zoom

If this button is activated, the zoom factor is adapted automatically.

Mode

Sets the display mode. The following modes are available: Lines, Dots, Envelope.

Peak Fallback

Sets the speed of release for the peak envelope in Envelope mode.

NOTE

If this control is turned all the way to the left, the peak envelope is disabled.

Scale

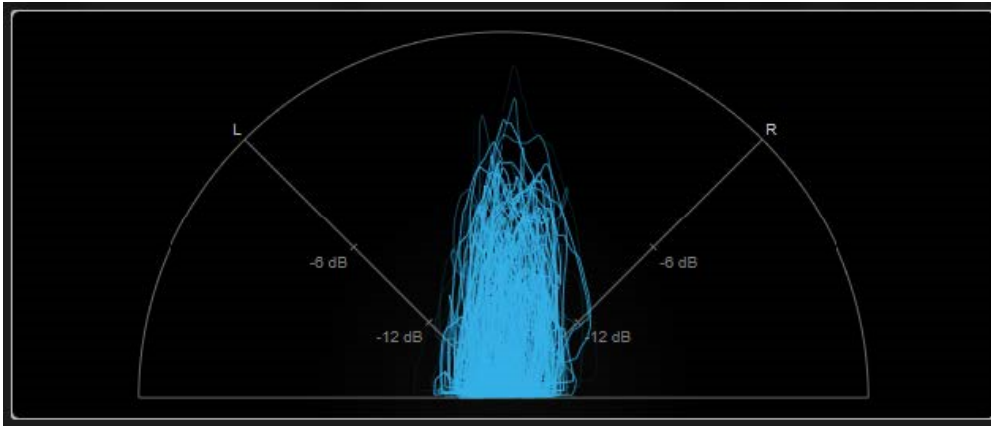
Activates/Deactivates the axis labeling.

NOTE

This option is only available if Auto Zoom is deactivated.

Panorama

This module uses a polar coordinate display to show the phase and amplitude relationship between the left and right stereo channels. This provides you with directional information about a stereo audio signal.

**NOTE**

In addition to the signal of the track, the display can show the signal of a side-chain input. For this to work, you must select a Main & Side-Chain view from the channel selector. The side-chain signal is then shown in white.

The following module-specific settings are available in the **Module Settings** window:

Zoom

Allows you to zoom on the graphical display.

NOTE

Alternatively, you can move the mouse pointer over the selected module and use the mouse wheel to adjust this parameter.

Auto Zoom

If this button is activated, the zoom factor is adapted automatically.

Mode

Sets the display mode. The following modes are available: Lines, Dots, Envelope.

Peak Fallback

Sets the speed of release for the peak envelope in Envelope mode.

NOTE

If this control is turned all the way to the left, the peak envelope is disabled.

Scale

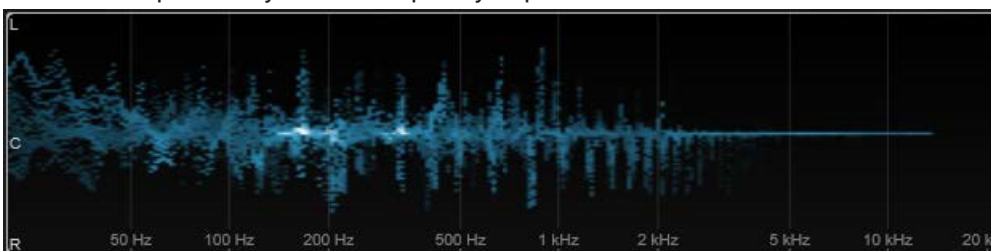
Activates/Deactivates the axis labeling.

NOTE

This option is only available if Auto Zoom is deactivated.

Multipanorama

This module provides you with frequency-dependent directional information about a stereo audio signal.



The following module-specific settings are available in the **Module Settings** window:

Time Smooth

Sets the time for which an energy impulse is displayed.

Bands/Oct.

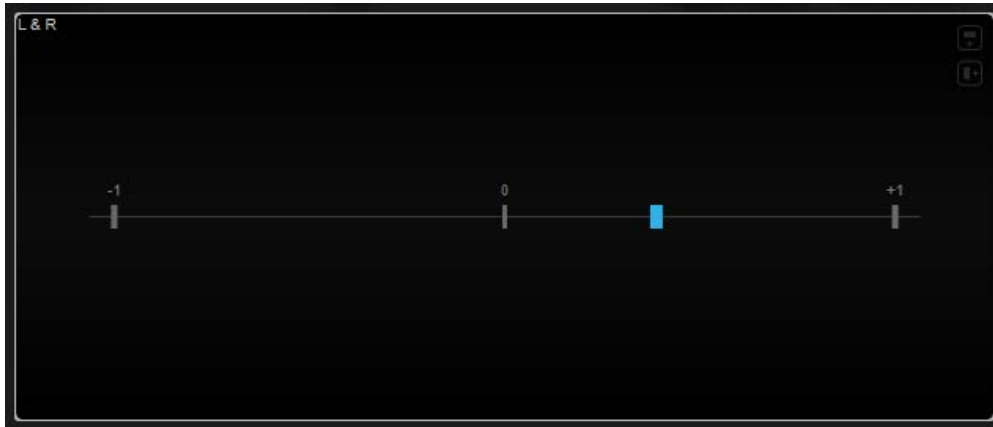
Sets the number of bands per octave.

Color

Allows you to choose a color scheme.

Correlation

This module visualizes the phase-correlation between the left and right channel. This allows you, for example, to check the mono compatibility of a stereo recording.



The following module-specific setting is available in the **Module Settings** window:

Time Smooth

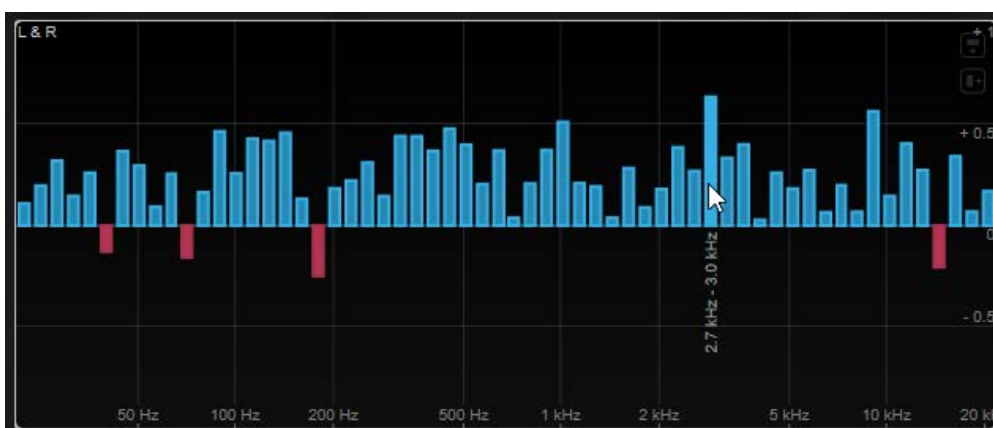
Smooths the temporal display of the correlation.

NOTE

Alternatively, you can move the mouse pointer over the selected module, hold Alt, and use the mouse wheel to adjust this parameter

Multicorrelation

This module visualizes the phase-correlation between the left and right channel for different frequency bands.



Move the mouse pointer over a bar to display its frequency range in Hz. Hold Ctrl/Cmd to display its current value. Hold Shift to display its pitch range.

The following module-specific settings are available in the **Module Settings** window:

Time Smooth

Smooths the temporal display of the correlation.

NOTE

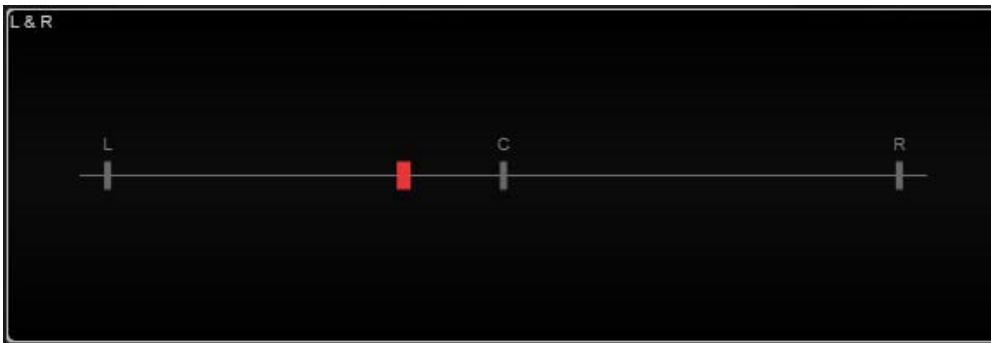
Alternatively, you can move the mouse pointer over the selected module, hold Alt, and use the mouse wheel to adjust this parameter.

Bands/Oct.

Sets the number of bands per octave.

Balance

This module visualizes the balance between the left and right channel.



The following module-specific setting is available in the **Module Settings** window:

Time Smooth

Smooths the temporal display of the correlation.

NOTE

Alternatively, you can move the mouse pointer over the selected module, hold **Alt**, and use the mouse wheel to adjust this parameter.

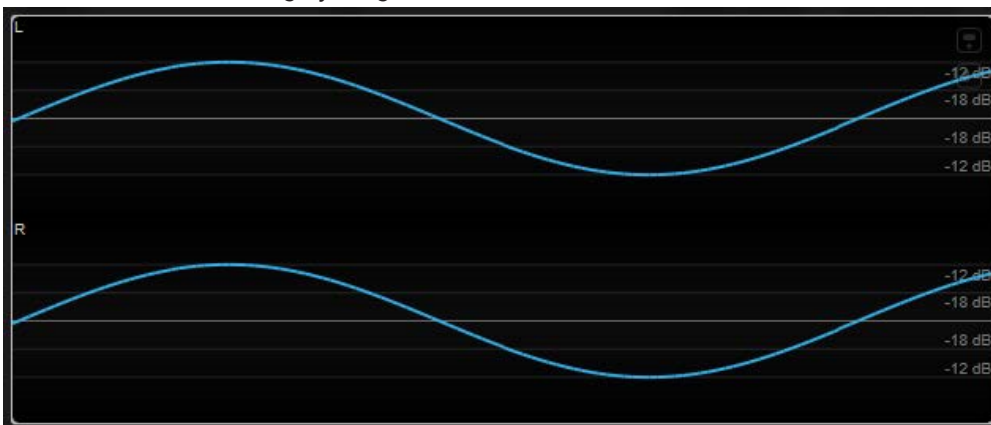
Waveform Modules

The modules in this category allow you to visualize the waveform of the audio signal.

The following modules and module-specific settings are available:

Oscilloscope

This module shows a highly magnified view of the waveform.

**NOTE**

In addition to the signal of the track, the display can show the signal of a side-chain input. For this to work, you must select a Main & Side-Chain view from the channel selector. The side-chain signal is then shown in white.

The following module-specific settings are available in the **Module Settings** window:

Zoom

Allows you to zoom on the graphical display by adjusting the amplitude.

NOTE

Alternatively, you can move the mouse pointer over the selected module, hold **Alt/Opt**, and use the mouse wheel to adjust this parameter.

Frequency

Allows you to zoom on the graphical display by adjusting the frequency.

NOTE

Alternatively, you can move the mouse pointer over the selected module, hold Ctrl/Cmd , and use the mouse wheel to adjust this parameter.

Trigger

Sets the channel that is used to synchronize the audio signal.

NOTE

Alternatively, you can click the waveform of the corresponding channel.

Scale

Activates/Deactivates the axis labeling.

NOTE

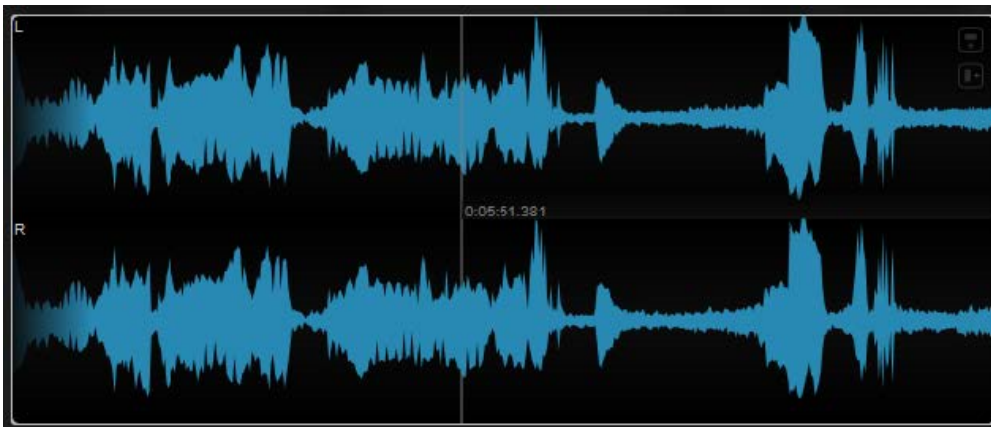
This option is only available if Auto Zoom is deactivated.

Phase

Allows you to shift the zero-crossing position.

Wavescope

This module displays the real-time waveform of the audio signal.



Move the mouse pointer over a waveform position to display the corresponding project time.

NOTE

In addition to the signal of the track, the display can show the signal of a side-chain input. For this to work, you must select a Main & Side-Chain view from the channel selector. The side-chain signal is then shown in white.

The following module-specific settings are available in the **Module Settings** window:

Zoom

Allows you to zoom on the graphical display.

NOTE

Alternatively, you can move the mouse pointer over the selected module and use the mouse wheel to adjust this parameter.

Duration

Sets the duration of the audio stream that is displayed.

NOTE

Alternatively, you can move the mouse pointer over the selected module, hold Ctrl/Cmd , and use the mouse wheel to adjust this parameter.

Tempo Sync

If this button is activated, you can set Duration in beats.

NOTE

The equivalent duration is limited to a minimum of 0.5 s and a maximum of 30 s.

Scale

Activates/Deactivates the axis labeling.

NOTE

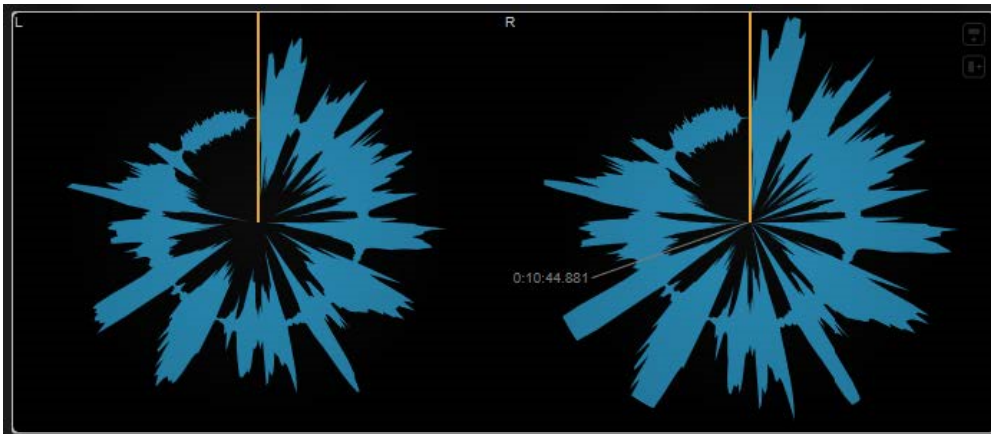
This option is only available if Auto Zoom is deactivated.

Station.Cursor

If this button is activated, the waveform moves continuously under the stationary cursor. If this button is deactivated, the waveform is refreshed when the cursor moves over it.

Wavecircle

This module displays the real-time waveform of the audio signal as a circle.



Move the mouse pointer over a waveform position to display the corresponding project time.

NOTE

In addition to the signal of the track, the display can show the signal of a side-chain input. For this to work, you must select a Main & Side-Chain view from the channel selector. The side-chain signal is then shown in white.

The following module-specific settings are available in the **Module Settings** window:

Zoom

Allows you to zoom on the graphical display.

NOTE

Alternatively, you can move the mouse pointer over the selected module and use the mouse wheel to adjust this parameter.

Duration

Sets the duration of the audio stream that is displayed.

NOTE

Alternatively, you can move the mouse pointer over the selected module, hold Ctrl/Command, and use the mouse wheel to adjust this parameter.

Tempo Sync

If this button is activated, you can set Duration in beats.

NOTE

The equivalent duration is limited to a minimum of 0.5 s and a maximum of 30 s.

Reverse

Changes the rotation direction.

Station.Cursor

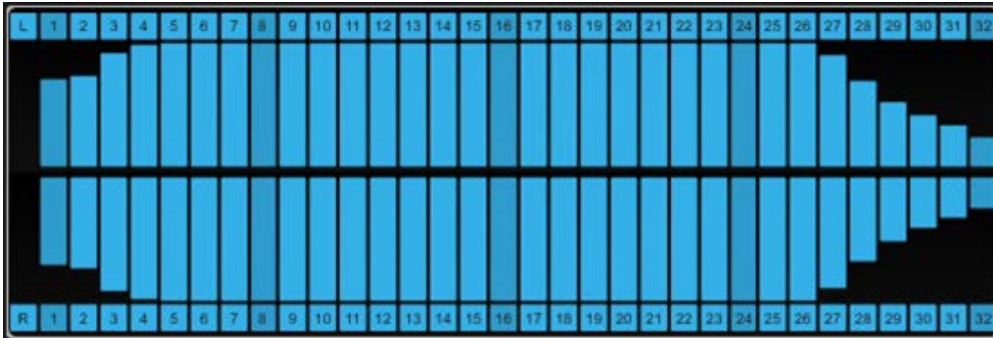
If this button is activated, the waveform moves continuously under the stationary cursor. If this button is deactivated, the waveform is refreshed when the cursor moves over it.

Other Modules

This category provides a bit meter.

Bits

This module shows which bits are currently used by the audio signal. The shorter a bar, the less a bit is being used. A colored square above or below a bar indicates if a bit has been used at all since the start of the measurement.



The following module-specific setting is available in the **Module Settings** window:

Time Smooth

Smooths the temporal display, which facilitates monitoring which of the bits were used last. With Time Smooth deactivated, the bars are either fully displayed or fully hidden, which simply indicates if a bit is used or not.

NOTE

Alternatively, you can move the mouse pointer over the selected module and use the mouse wheel to adjust this parameter.

Delay Plug-ins

MonoDelay

This is a mono delay effect. The delay line uses either tempo-based or freely specified delay time settings.



LO FILTER

Affects the feedback loop of the effect signal and allows you to roll off low frequencies. The button below the knob activates/deactivates the filter.

HI FILTER

Affects the feedback loop of the effect signal and allows you to roll off high frequencies. The button below the knob activates/deactivates the filter.

DELAY

Sets the delay time in milliseconds.

SYNC

Activates/Deactivates tempo sync.

FEEDBACK

Sets the amount of the signal that is sent back into the delay input. The higher this value, the higher the number of repeats.

MIX

Sets the level balance between the dry signal and the wet signal. If the effect is used as a send effect, set this parameter to the maximum value, as you can control the dry/effect balance with the send level.

PingPongDelay

This is a stereo delay effect that alternates each delay repeat between the left and right channels. The delay line uses either tempo-based or freely specified delay time settings.

NOTE

This plug-in works only on stereo racks.



LO FILTER

Affects the feedback loop of the effect signal and allows you to roll off low frequencies. The button below the knob activates/deactivates the filter.

HI FILTER

Affects the feedback loop of the effect signal and allows you to roll off high frequencies. The button below the knob activates/deactivates the filter.

DELAY

Sets the delay time in milliseconds.

SYNC

Activates/Deactivates tempo sync.

FEEDBACK

Sets the amount of the signal that is sent back into the delay input. The higher this value, the higher the number of repeats.

MIX

Sets the level balance between the dry signal and the wet signal. If the effect is used as a send effect, set this parameter to the maximum value, as you can control the dry/effect balance with the send level.

SPATIAL

Sets the stereo width for the left/right repeats. Turn clockwise for a more pronounced stereo ping-pong effect.

START LEFT/START RIGHT

Determines whether the delay repeat starts on the left or the right channel.

StereoDelay

This is a stereo delay effect that includes two independent delay effects. It can either be tempo-based or use freely specified delay time settings.

NOTE

This plug-in works only on stereo racks.



FEEDBACK

Sets the number of repeats for each delay.

DELAY

Sets the delay time in milliseconds.

SYNC

Activates/Deactivates tempo sync for the corresponding delay.

MIX

Sets the level balance between the dry signal and the wet signal. If the effect is used as a send effect, set this parameter to the maximum value, as you can control the dry/effect balance with the send level.

LO FILTER

Affects the feedback loop of the effect signal and allows you to roll off low frequencies. The button below the knob activates/deactivates the filter.

PAN

Sets the stereo position.

HI FILTER

Affects the feedback loop of the effect signal and allows you to roll off high frequencies. The button below the knob activates/deactivates the filter.

Distortion Plug-ins

AmpSimulator

AmpSimulator is a distortion effect that emulates the sound of various types of guitar amp and speaker cabinet combinations. A wide selection of amp and cabinet models is available.



Select Amplifier Model

This pop-up menu allows you to select an amplifier model. You can bypass this section by selecting [No Amp].

DRIVE

Controls the amount of amp overdrive.

BASS

Tone control for the low frequencies.

MID

Tone control for the mid frequencies.

TREBLE

Tone control for the high frequencies.

PRESENCE

Boosts or dampens the higher frequencies.

VOLUME

Controls the overall output level.

Select Cabinet Model

This pop-up menu allows you to select a speaker cabinet model. You can bypass this section by selecting [No Speaker].

DAMPING LOW/HIGH

These tone controls allow you to shape the sound of the selected speaker cabinet.

Quadrafuzz v2

Quadrafuzz v2 is a multi-band distortion and multi-effect plug-in for processing drums and loops, but also for treatment of vocals. You can distort up to 4 bands. 5 different distortion modes with several sub-modes are available.



Frequency Band Editor

The frequency band editor in the upper half of the panel is where you set the width of the frequency bands as well as the output level. The vertical value scale to the left shows the gain level of each frequency band. The horizontal scale shows the available frequency range.

- To define the frequency range of the different frequency bands, use the handles at the sides of each frequency band.
- To attenuate or boost the output level of each frequency band by ± 15 dB, use the handles on top of each frequency band.

Global Settings

SB

Switches between multi band and single band mode.

SCENES

You can save up to 8 different settings. If the default setting of a scene is active, the selected scene button lights up yellow.

If you change the default settings, the button lights up green, indicating that this scene has customized settings.



To copy the settings of a scene to another scene, select the scene that you want to copy, click [Copy], and click one of the numbered buttons.

You can automate the selection of scenes.

MIX

Sets the level balance between the dry signal and the wet signal.

OUT (-24 to 24 dB)

Sets the output level.

Band Settings

Mute

To mute a frequency band, activate the [M] button in each section.

Bypass Frequency Band

To bypass a frequency band, activate the [Band] button in each section.

Solo Frequency Band

To solo a frequency band, activate the [S] button on each section. Only one frequency band can be soloed at a time.

[IN/OUT] Meters

Display the input and output level.

GATE

Determines the level at which the gate is activated. Signal levels above the set threshold open the gate and signal levels below the set threshold close the gate.

TAPE

This band mode simulates the saturation and compression of recording on analog tape machines.

DRIVE

Controls the amount of tape saturation.

TAPE MODE DUAL

Simulates the use of two machines.

TUBE

This band mode simulates the saturation effects using analog tubes.

DRIVE

Controls the amount of tube saturation.

Tube(s)

Determines the number of tubes that are simulated.

DIST

This band mode adds distortion to your rack.

DRIVE

Controls the amount of distortion.

FBK

Feeds part of the output signal back to the effect input. Higher settings increase the distortion effect.

AMP

This band mode simulates the sound of various types of guitar amps.

DRIVE

Controls the amount of amp overdrive.

Amp Types

You can select the following types of guitar amps

- Amp Clean
- Amp Crunch
- Amp Lead

DEC

This band mode allows you to decimate and truncate the input audio signal to create a noisy, distorted sound.

DECIMATOR

Controls the resulting bit-resolution. The lower the resolution, the higher the distortion effect.

MODE

Allows you to select one of the four operating modes. In each mode, the effect sounds differently. Modes [I] and [III] are nastier and noisier, while modes [II] and [IV] are more subtle.

S&H

Sets the amount by which the audio samples are decimated. At the highest setting, nearly all of the information describing the original audio signal is eliminated, turning the signal into unrecognizable noise.

Delay

To open the [Delay] section, click the [Delay] button.

TIME

If tempo sync is activated, this is where you specify the base note value for tempo-syncing the effect (1/1 to 1/32, straight, triplet, or dotted).

If tempo sync is deactivated, the delay time can be set freely with the [Time] knob.

SYNC

Activates/Deactivates tempo sync for the corresponding delay.

DUCK

Determines how much the delay signal ducks when an audio signal is present.

MIX

Sets the level balance between the dry signal and the wet signal.

FBK

Sets the number of repeats for each delay.

MODE

If this option is activated, the delay signal is routed back into the distortion unit to create a feedback with distortion.

NOTE

High [FBK] values and low [DUCK] values in combination with activated [MODE] can lead to unwanted noise.

Slider**WIDTH**

Sets the stereo width for the corresponding band.

OUT

Sets the output gain for the corresponding band.

PAN

Sets the stereo position for the corresponding band.

MIX

Sets the level balance between the dry signal and the wet signal.

SoftClipper

SoftClipper adds soft overdrive, with independent control over the second and third harmonic.



INPUT (-12 to 24 dB)

Sets the pre-gain. Use high values if you want an overdriven sound just on the verge of distortion.

MIX

Sets the level balance between the dry signal and the wet signal.

OUTPUT

Sets the output level.

SECOND

Controls the second harmonic.

THIRD

Controls the third harmonic.

VST Amp Rack

VST Amp Rack is a powerful guitar amp simulator. It offers a choice of amplifiers and speaker cabinets that can be combined with stomp box effects.



At the top of the plug-in panel, there are seven buttons, arranged according to the position of the corresponding elements in the signal chain. Each button opens a different page in the display section of the plug-in panel ([Pre-Effects], [Amplifiers], [Cabinets], [Post-Effects], [Microphone], [Configuration], and [Master]). Below the display section, the selected amplifier is shown. The color and texture of the area below the amplifier indicate the selected cabinet.

Pre-Effects/Post-Effects

On the [Pre-Effects] and [Post-Effects] pages, you can select up to six common guitar effects. On both pages, the same effects are available, the only difference being the position in the signal chain (before and after the amplifier). On each page, every effect can be used once. Each effect features an [On/Off] button known from stompbox effects, as well as individual parameters.

Wah Wah

Pedal – Controls the filter frequency sweep.

Volume

Pedal – Controls the level of the signal passing through the effect.

Compressor

Intensity – Sets the amount by which an input signal is being compressed.

Limiter

Threshold – Sets the maximum output level. Signal levels above the set threshold are cut off.

Release – Sets the time after which the gain returns to the original level.

Maximizer

Amount – Sets the loudness of the signal.

Chorus

Rate – Allows you to set the sweep rate. This parameter can be synchronized to the project tempo.

Width – Sets the depth of the chorus effect. Higher settings produce a more pronounced effect.

Phaser

Rate – Allows you to set the sweep rate. This parameter can be synchronized to the project tempo.

Width – Determines the width of the modulation effect between higher and lower frequencies.

Flanger

Rate – Allows you to set the sweep rate. This parameter can be synchronized to the project tempo.

Feedback – Determines the character of the flanger effect. Higher settings produce a more metallic sounding sweep.

Mix – Sets the level balance between the dry signal and the wet signal.

Tremolo

Rate – Allows you to set the modulation speed. This parameter can be synchronized to the project tempo.

Depth – Governs the depth of the amplitude modulation.

Octaver

Direct – Adjusts the mix of the original signal and the generated voices. A value of 0 means only the generated and transposed signal is heard. By raising this value, more of the original signal is heard.

Octave 1 – Adjusts the level of the signal that is generated one octave below the original pitch. A setting of 0 means that the voice is muted.

Octave 2 – Adjusts the level of the signal that is generated two octaves below the original pitch. A setting of 0 means that the voice is muted.

Delay

Delay – Sets the delay time in milliseconds. This parameter can be synchronized to the project tempo.

Feedback – Sets the number of repeats for the delay.

Mix – Sets the level balance between the dry signal and the wet signal.

Tape Delay

Delay – Tape Delay creates a delay effect known from tape machines. Sets the delay time in milliseconds. This parameter can be synchronized to the project tempo.

Feedback – Sets the number of repeats for the delay.

Mix – Sets the level balance between the dry signal and the wet signal.

Tape Ducking Delay

Delay – Tape Ducking Delay creates a delay effect known from tape machines with a ducking parameter. The Delay parameter sets the delay time in milliseconds. This parameter can be synchronized to the project tempo.

Feedback – Sets the number of repeats for the delay.

Duck – Works like an automatic mix parameter. If the level of the input signal is high, the portion of the effect signal is lowered, or ducked (low internal mix value). If the level of the input signal is low, the portion of the effect signal is raised (high internal mix value). This way the delayed signal stays rather dry during loud or intensely played passages.

Overdrive

Drive – Overdrive creates a tube-like overdrive effect. The higher this value, the more harmonics are added to the output signal of this effect.

Tone – Works as a filter effect on the added harmonics.

Level – Adjusts the output level.

Fuzz

Boost – Fuzz creates a rather harsh distortion effect. The higher this value, the more distortion is created.

Tone – Works as a filter effect on the added harmonics.

Level – Adjusts the output level.

Gate

Threshold – Determines the level at which the gate is activated. Signal levels above the set threshold open the gate and signal levels below the set threshold close the gate.

Release – Sets the time after which the gate closes.

Equalizer

Low – Changes the level of the low-frequency portion of the incoming signal.

Middle – Changes the level of the mid-frequency portion of the incoming signal.

High – Changes the level of the high-frequency portion of the incoming signal.

Reverb

Type – A convolution-based reverb effect. This parameter allows you to switch between different reverb types ([Studio], [Hall], [Plate], and [Room]).

Mix – Sets the level balance between the dry signal and the wet signal.

Sync Mode

Some parameters can be synchronized to the tempo of VST Rack.

The names of these parameters are bordered. Click a knob to activate or deactivate tempo sync. An LED at the top right of the knob indicates that sync mode is active. You can then select a base note value for tempo syncing from the pop-up menu above the control.



Using Effects

- To insert a new effect, click the [+] button that appears if you point the mouse at an empty plug-in slot or at one of the arrows before or after a used effect slot.
- To remove an effect from an effect slot, click the effect name and select [No Effect] from the pop-up menu.
- To change the order of the effects in the chain, click on an effect and drag it onto the line at the destination location.
- To activate or deactivate an effect, click the pedal-like button below the effect name. If an effect is active, the LED next to the button is lit.

NOTE

- Pre-effects and post-effects can be mono or stereo, depending on the rack configuration.

Amplifiers

The amps available on the [Amplifiers] page are modeled on real-life amplifiers. Each amp features settings typical for guitar recording, such as gain, equalizers, and master volume. The sound-related parameters Bass, Middle, Treble, and Presence have a significant impact on the overall character and sound of the corresponding amp.

Plexi

Classic British rock tone; extremely transparent sound, very responsive.

Plexi Lead

British rock tone of the 70s and 80s.

Diamond

The cutting edge hard rock and metal sounds of the 90s.

Blackface

Classic American clean tone.

Tweed

Clean and crunchy tones; originally developed as a bass amp.

Deluxe

American crunch sound coming from a rather small amp with a big tone.

British Custom

Produces the sparkling clean or harmonically distorted rhythm sounds of the 60s.

The different amps keep their settings if you switch models. If you want to use the same settings after reloading the plug-in, you need to set up a preset.

Selecting and Deactivating Amplifiers

To switch amps on the [Amplifiers] page, click the model that you want to use. Select [No Amplifier] if you only want to use the cabinets and effects.

Cabinets

The cabinets available on the [Cabinets] page simulate real-life combo boxes or speakers. For each amp, a corresponding cabinet type is available, but you can also combine different amps and cabinets.

Selecting and Deactivating Cabinets

- To switch cabinets on the [Cabinets] page, click the model that you want to use. Select [No Cabinet] if you only want to use the amps and effects.
- If you select [Link Amplifier & Cabinet Choice], the plug-in automatically selects the cabinet corresponding to the selected amp model.

Microphones

On the [Microphones] page, you can choose between different microphone positions. These positions result from two different angles (center and edge) and three different distances from the speaker, as well as an additional center position at an even greater distance from the speaker.

You can choose between two microphone types: a large-diaphragm condenser microphone and a dynamic microphone. You can crossfade between the characteristics of the two microphones.

- To select one of the microphone types or blend between the two types, turn the [Mix] control between the two microphones.

Placing the Microphone

- To select a microphone position, click the corresponding ball in the graphic. The selected position is marked in red.

Configuration

On the [Configuration] page, you can specify whether you want to use VST Amp Rack in stereo or in mono mode.

- To process the pre-effects, the amplifier, and the cabinets in full stereo mode, make sure that the plug-in is inserted on a stereo rack, and activate the [Stereo] button.

NOTE

In stereo mode, the effect requires more processing power.

Master

Use the [Master] page to fine-tune the sound.

Input/Output Level Meters

The input and output level meters on the left and the right of the Master section show the signal level of your audio. The rectangle on the input meter indicates the optimum incoming level range. In compact view, the input and output levels are indicated by two LEDs at the top left and right.

Using the Master Controls

- To activate/deactivate the equalizer, click the pedal-like [On/Off] button. If the equalizer is active, the LED next to the button is lit.
- To activate/deactivate an equalizer band, click the corresponding [Gain] knob. If a band is active, the LED to the left of the [Gain] knob is lit.
- To tune your guitar strings, click the pedal-like [On/Off] button to activate the Tuner and play a string. If the correct pitch is displayed and the row of LEDs below the digital display is green, the string is tuned correctly.
If the pitch is too low, red LEDs are lit on the left. If the pitch is too high, red LEDs are lit on the right. The more LEDs are lit, the lower/higher is the pitch.
- To mute the output signal of the plug-in, click the pedal-like [Master] button. If the output is muted, the LED is not lit. Use this to tune your guitar in silence, for example.
- To change the volume of the output signal, use the [Level] control on the [Master] page.

View Settings

You can choose between 2 views: the default view and a compact view, which takes up less screen space.

In the default view, you can use the buttons at the top of the plug-in panel to open the corresponding page in the display section above the amp controls. You can horizontally resize the plug-in panel by clicking and dragging the edges or corners.

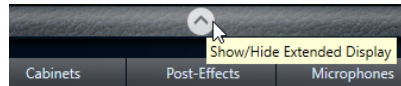
In the compact view, the display section is hidden from view. You can change the amp settings and switch amps or cabinets using the mouse wheel.

Using the Smart Controls

Smart controls become visible on the plug-in frame when you move the mouse pointer over the plug-in panel.

Switching between Default and Compact View

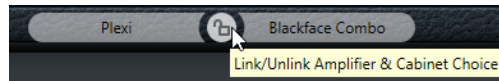
- To toggle between the different views, click the down/up arrow button ([Show/Hide Extended Display]) at the top center of the plug-in frame.



Changing the Amplifier and Cabinet Selection in the Compact View

In the compact view, a smart control on the lower border of the plug-in frame allows you to select different amplifier and cabinet models.

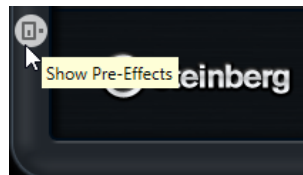
- To select a different amplifier or cabinet, click the name and select a different model from the pop-up menu.
- To lock the amplifier and cabinet combination, activate the [Link/Unlink Amplifier & Cabinet Choice] button. If you now select another amp model, the cabinet selection follows. However, if you select a different cabinet model, the lock is deactivated.



Previewing Effect Settings

In both views, you can show a preview of the pre- and post-effects that you selected on the corresponding pages:

- Click and hold the [Show Pre-Effects] button at the bottom left or [Show Post-Effects] button at the bottom right of the plug-in frame.



Magneto II

Magneto II simulates the saturation and compression of recording on analog tape machines.



Saturation

Determines the amount of saturation and the generation of overtones. This leads to a small increase in input gain.

Saturation On/Off

Activates/Deactivates the saturation effect.

Dual Mode

Simulates the use of two machines.

Frequency Range Low/High

These parameters set the frequency range of the spectrum band to which the tape effect is applied.

For example, to avoid the saturation of lower frequencies, set the Low value to 200 Hz or 300 Hz. To avoid the saturation of very high frequencies, set the High parameter to values below 10 kHz.

Solo

Allows you to hear only the set frequency range including the tape simulation effect. This helps you to determine the appropriate frequency range.

HF-Adjust

Sets the amount of high frequency content of the saturated signal.

HF-Adjust On/Off

Activates/Deactivates the HF-Adjust filter.

Dynamics Plug-ins

DeEsser

DeEsser is a special type of compressor that reduces excessive sibilance, primarily for vocal sounds.



You can use it, for example, when close proximity microphone placement and equalizing lead to situations where the overall sound is just right, but where unwanted sibilants occur.

When recording a voice, the position of DeEsser in the signal chain is usually after the microphone pre-amp and before a compressor/limiter. This keeps the compressor/limiter from unnecessarily limiting the overall signal dynamics.

Display

Shows the spectrum of the input signal.

- To adjust the frequency band, drag the border lines or click in the middle of the band and drag.
- To change the width of the frequency band, hold [Shift] and drag to the left or right.

FILTER

LO/HI

Sets the left and right border of the frequency band. You can set the frequency either in Hz or as a note value. If you enter a note value, the frequency is automatically displayed in Hz accordingly. For example, a note value of A3 sets the frequency to 440 Hz. When you enter a note value, you can also enter a cent offset. For example, enter "A5 -23" or "C4 +49." When you enter a note value, you can also enter a cent offset. For example, enter "A5 -23" or "C4 +49."

NOTE

Make sure that you enter a space between the note and the cent offset. Only in this case, the cent offsets are taken into account.

SOLO

Solos the frequency band. This helps you to find the appropriate position and width of that band.

DIFF

Plays back what DeEsser removed from the signal. This helps you to adjust the frequency band, threshold, and reduction parameters, so that only sharp s-sounds are removed, for example.

DYNAMICS

REDUCT

Controls the intensity of the de-essing effect.

THRESH (-50 to 0 dB)

If the [Auto] option is deactivated, you can use this control to set a threshold for the incoming signal level above which the plug-in starts to reduce the sibilants.

RELEASE (1 to 1000 ms)

Sets the time after which the de-essing effect returns to zero when the signal drops below the threshold.

AUTO

Automatically and continually sets an optimum threshold setting independent of the input signal. The [Auto] option does not work for low-level signals (< -30 db peak level). To reduce the sibilants in such a file, set the threshold manually.

SIDE-CHAIN

FREQ (25 Hz to 20 kHz)

If [Side-Chain] is activated, this sets the frequency of the filter. You can set the frequency either in Hz or as a note value. If you enter a note value, the frequency is automatically displayed in Hz accordingly. For example, a note value of A3 sets the frequency to 440 Hz. When you enter a note value, you can also enter a cent offset. For example, enter "A5 -23" or "C4 +49." When you enter a note value, you can also enter a cent offset. For example, enter "A5 -23" or "C4 +49."

NOTE

Make sure that you enter a space between the note and the cent offset. Only in this case, the cent offsets are taken into account.

SIDE-CHAIN

Activates the internal side-chain filter. You can now shape the input signal according to the filter parameters. Internal side-chaining can be useful for tailoring how the gate operates.

Q-FACTOR

If [Side-Chain] is activated, this sets the resonance or width of the filter.

MONITOR

Allows you to monitor the filtered signal.

LIVE

If this button is activated, the look-ahead feature of the effect is deactivated. Look-ahead produces more accurate processing, but adds a specific amount of latency as a trade-off. If [Live] mode is activated, there is no latency, which is better for live processing.

Positioning the DeEsser in the Signal Chain

When recording a voice, the position of DeEsser in the signal chain is usually after the microphone pre-amp and before a compressor/limiter. This keeps the compressor/limiter from unnecessarily limiting the overall signal dynamics.

EnvelopeShaper

EnvelopeShaper can be used to attenuate or boost the gain of the attack and release phase of audio material. You can use the knobs or drag the breakpoints in the graphical display to change parameter values. Be careful with levels when boosting the gain and if needed reduce the output level to avoid clipping.



ATTACK (-20 to 20 dB)

Sets the gain of the attack phase of the signal.

LENGTH (5 to 200 ms)

Sets the length of the attack phase.

RELEASE

Sets the gain of the release phase of the signal.

OUTPUT

Sets the output level.

Maximizer

Maximizer raises the loudness of audio material without the risk of clipping. The plug-in provides two modes, [Classic] and [Modern], that offer different algorithms and parameters.



CLASSIC

[Classic] mode provides the classic algorithms from previous versions of this plug-in. This mode is suited for all styles of music.

MODERN

In [Modern] mode, the algorithm allows for more loudness than in [Classic] mode. This mode is particularly suited for music that requires loud volume.

[Modern] mode also provides additional settings to control the release phase:

- [Release] sets the overall release time.
- [Recover] allows for a faster signal recovery at the beginning of the release phase.

OPTIMIZE

Determines the loudness of the signal.

MIX

Sets the level balance between the dry signal and the wet signal.

OUTPUT

Sets the maximum output level.

SOFT CLIP

If this button is activated, Maximizer starts limiting or clipping the signal softly. At the same time, harmonics are generated, adding a warm, tube-like characteristic to the audio material.

MultibandCompressor

MultibandCompressor allows a signal to be split into four frequency bands. You can specify the level, bandwidth, and compressor characteristics for each band.



NOTE

To compensate for output gain loss that is caused by compression, MultibandCompressor uses an automatic make-up gain. If side-chaining is activated for a frequency band in the side-chain section, the automatic make-up gain is deactivated for this band.

Frequency Band Editor


The frequency band editor in the upper half of the panel is where you set the width of the frequency bands as well as their level after compression. The vertical value scale to the left shows the gain level of each frequency band. The horizontal scale shows the available frequency range.

- To define the frequency range of the different frequency bands, use the handles at the sides of each frequency band.
- To attenuate or boost the gain of the frequency bands by ± 15 dB after compression, use the handles at the top of each frequency band.

LIVE

If this button is activated, the look-ahead feature of the effect is deactivated. Look-ahead produces more accurate processing, but adds a specific amount of latency as a trade-off. If [Live] mode is activated, there is no latency, which is better for live processing.

Bypass Frequency Band

To bypass a frequency band, activate the [Band] button () in each section.

Solo Frequency Band

To solo a frequency band, activate the [S] button on each section. Only one frequency band can be soloed at a time.

OUTPUT (-24 to 24 dB)

Sets the output level.

[COMPRESSOR] Section

You can specify the [Threshold] and [Ratio] by moving breakpoints or using the corresponding knobs. The threshold is marked by the first breakpoint where the line deviates from the straight diagonal.

THRESH (-60 to 0 dB)

Determines the level where the compressor kicks in. Only signal levels above the set threshold are processed.

RATIO

Sets the amount of gain reduction applied to signals above the set threshold. A ratio of 3:1 means that for every 3 dB the input level increases, the output level increases by 1 dB.

ATTACK (0.1 to 100 ms)

Determines how fast the compressor responds to signals above the set threshold. If the attack time is long, more of the early part of the signal passes through unprocessed.

RELEASE (10 to 1000 ms or [AUTO] mode)

Sets the time after which the gain returns to the original level when the signal drops below the threshold. If [Auto Release] is activated, the plug-in automatically finds the best release setting for the audio material.

SC (Side-chain Section)

To open the side-chain section, click the [SC] button at the bottom left of the plug-in window.

Important

To be able to use the side-chain function for the bands, global side-chain must be activated for the plug-in.



SIDE-CH

Activates the internal side-chain filter. The side-chain signal can then be shaped according to the filter parameters.

FREQ

If [Side-Chain] is activated, this sets the frequency of the side-chain filter.

Q-FACTOR

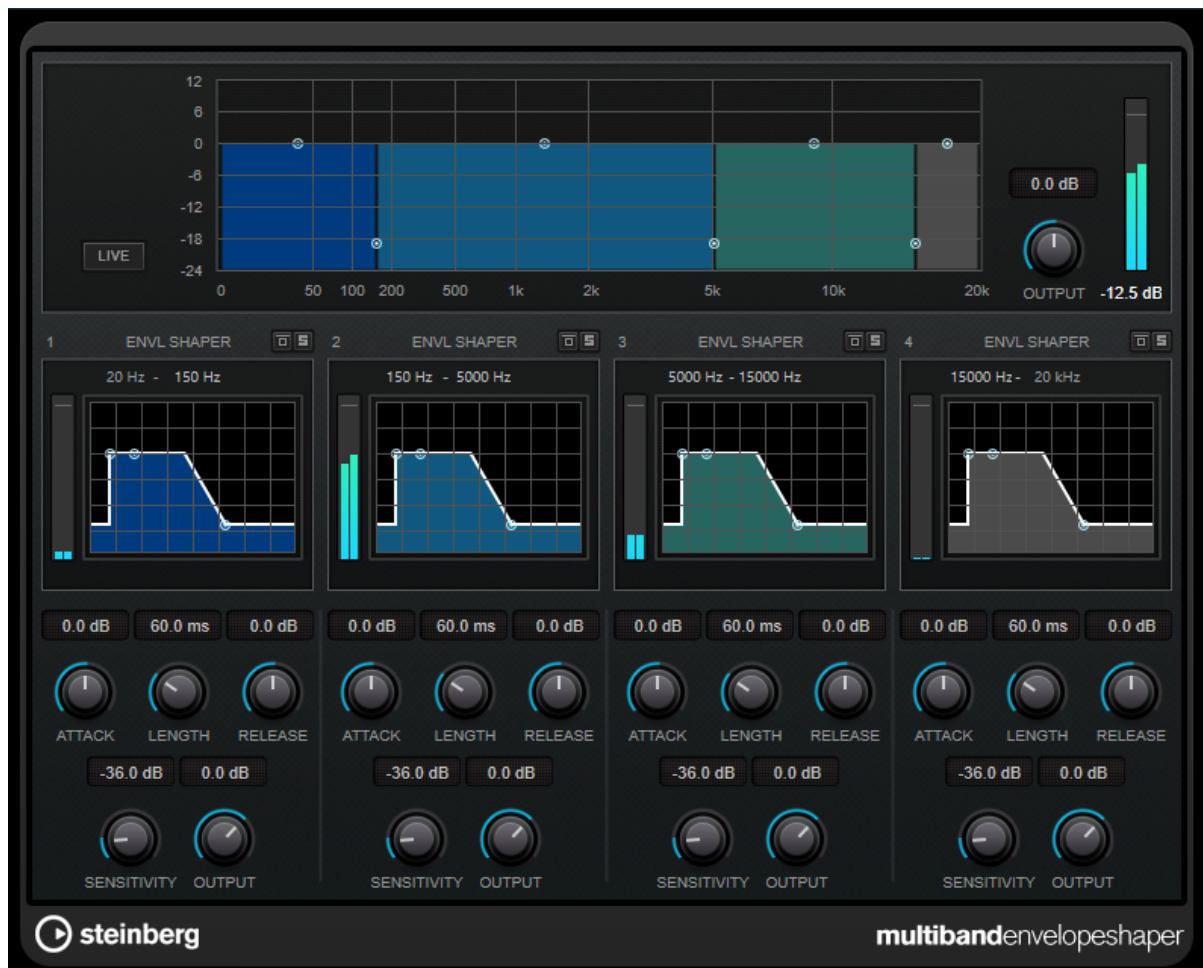
If [Side-Chain] is activated, this sets the resonance or width of the filter.

MONITOR

Allows you to monitor the filtered signal.

MultibandEnvelopeShaper

MultibandEnvelopeShaper allows a signal to be split into four frequency bands. You can attenuate or boost the gain of the attack and release phase of audio material for each band.



Frequency Band Editor


The frequency band editor in the upper half of the panel is where you set the width of the frequency bands as well as their level. The vertical value scale to the left shows the gain level of each frequency band. The horizontal scale shows the available frequency range.

- To define the frequency range of the different frequency bands, use the handles at the sides of each frequency band.
- To attenuate or boost the gain of the frequency band, use the handles at the top of each frequency band.

LIVE

If this button is activated, the look-ahead feature of the effect is deactivated. Look-ahead produces more accurate processing, but adds a specific amount of latency as a trade-off. If [Live] mode is activated, there is no latency, which is better for live processing.

Bypass Frequency Band

To bypass a frequency band, activate the [Band] button () in each section.

Solo Frequency Band

To solo a frequency band, activate the [S] button on each section. Only one frequency band can be soloed at a time.

OUTPUT (-24 to 24 dB)

Sets the output level.

Shaper Section

You can specify the [Attack], [Length], and [Release] by moving breakpoints or using the corresponding knobs. Be careful with levels when boosting the gain. You can reduce the output level to avoid clipping.

ATTACK (-20 to 20 dB)

Sets the gain of the attack phase of the signal.

LENGTH (5 to 200 ms)

Sets the length of the attack phase.

RELEASE

Sets the gain of the release phase of the signal.

SENSITIVITY (-40 to -10 dB)

Sets the sensitivity of the detection.

OUTPUT

Sets the output level.

Tube Compressor

This versatile compressor with integrated tube-simulation allows you to achieve smooth and warm compression effects.

The VU meter shows the amount of gain reduction. Tube Compressor features an internal side-chain section that lets you filter the trigger signal.



VU Meter

Shows the amount of gain reduction.

[IN/OUT] Meters

Shows the highest peaks of all available input and output channels.

INPUT

Determines the compression amount. The higher the input gain, the more compression is applied.

DRIVE (1.0 to 6.0 dB)

Controls the amount of tube saturation.

OUTPUT (-12 to 12 dB)

Sets the output gain.

CHARACTER

Keeps the bass tight and preserves its attacks by decreasing the tube saturation for lower frequencies, and adds brilliance by adding harmonics for higher frequencies.

ATTACK (0.1 to 100 ms)

Determines how fast the compressor responds. If the attack time is long, more of the early part of the signal passes through unprocessed.

RELEASE (10 to 1000 ms or [Auto] mode)

Sets the time after which the gain returns to its original level. If [Auto Release] is activated, the plug-in automatically finds the best release setting for the audio material.

MIX

Adjusts the mix between dry signal and wet signal, preserving the transients of the input signal.

RATIO

Toggles between a low and a high ratio value.

SC(Side-Chain)

Activates the internal side-chain filter. The input signal can then be shaped according to the filter parameters. Internal side-chaining can be useful for tailoring how the gate operates.

Side-Chain Section

FILTER TYPE (Low-Pass/Band-Pass/High-Pass)

If [Side-Chain] is activated, these buttons allow you to set the filter type to low-pass, band-pass, or high-pass.

Center (50 to 20000 Hz)

If [Side-Chain] is activated, this sets the center frequency of the filter.

Q-FACTOR

If [Side-Chain] is activated, this sets the resonance or width of the filter.

MONITOR

Allows you to monitor the filtered signal.

VintageCompressor

VintageCompressor is modeled after vintage type compressors.

This compressor features separate controls for [Input] and [Output] gain, [Attack], and [Release].

In addition, there is a [Punch] mode which preserves the attack phase of the signal and a program-dependent [Auto Release] function.



VU Meter

Shows the amount of gain reduction.

[IN/OUT] Meters

Shows the highest peaks of all available input and output channels.

INPUT

Determines the compression amount. The higher the input gain, the more compression is applied.

ATTACK (0.1 to 100 ms)

Determines how fast the compressor responds. If the attack time is long, more of the early part of the signal passes through unprocessed.

[PUNCH] Button

If this is activated, the early attack phase of the signal is preserved, retaining the original punch in the audio material, even with short Attack settings.

RELEASE (10 to 1000 ms or [AUTO] mode)

Sets the time after which the gain returns to its original level. If the [Auto] button is activated, the plug-in automatically finds the best release setting for the audio material.

MIX

Adjusts the mix between dry signal and wet signal, preserving the transients of the input signal.

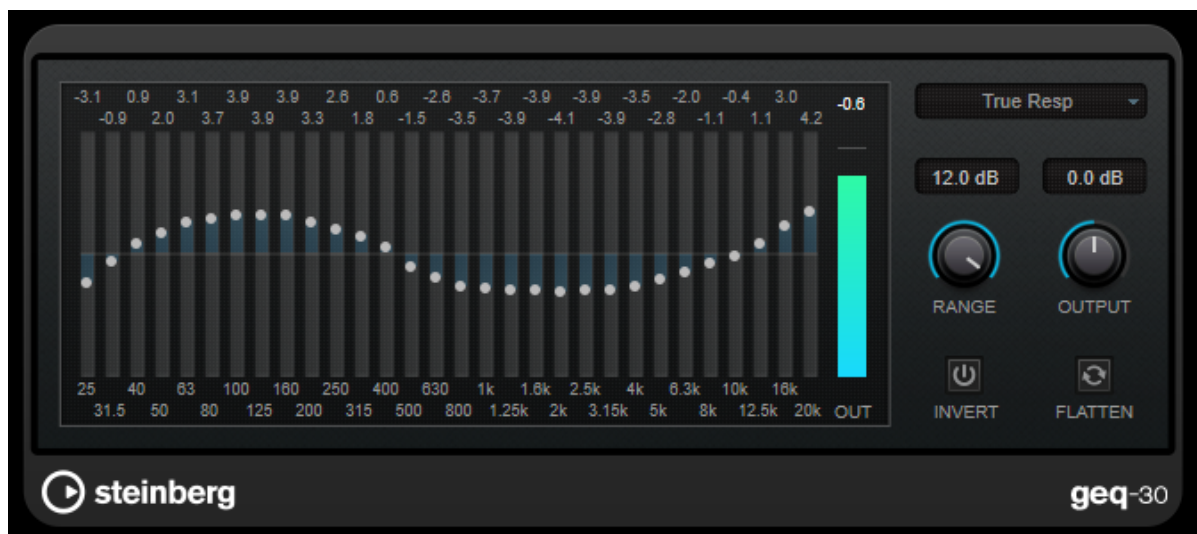
OUTPUT (-48 to 24 dB)

Sets the output gain.

EQ Plug-ins

GEQ-30

These are graphic equalizers. GEQ-30 has 30 available frequency bands.



Each band can be attenuated or boosted by up to 12 dB, allowing for fine control of the frequency response. In addition, there are several preset modes available that can add color to the sound of GEQ-30. You can draw response curves in the main display by clicking and dragging with the mouse. You have to click one of the sliders before you drag across the display. At the bottom of the window, the individual frequency bands are shown in Hz. At the top of the display, the amount of attenuation/boost is shown in dB.

RANGE

Allows you to adjust how much a set curve cuts or boosts the signal.

OUTPUT

Sets the overall gain of the equalizer.

INVERT

Inverts the current response curve.

FLATTEN

Resets all the frequency bands to 0 dB.

EQ Modes

The Mode pop-up menu in the upper right corner allows you to select an EQ mode, which add color or character to the equalized output in various ways.

True Response

Applies serial filters with an accurate frequency response.

Digital Standard

In this mode, the resonance of the last band depends on the sample rate.

Classic

Applies a classic parallel filter structure where the response does not follow the set gain values accurately.

VariableQ

Applies parallel filters where the resonance depends on the amount of gain.

ConstQ asym

Applies parallel filters where the resonance is raised when boosting the gain and vice versa.

ConstQ sym

Applies parallel filters where the resonance of the first and last bands depends on the sample rate.

Resonant

Applies serial filters where a gain increase of one band lowers the gain in adjacent bands.

StudioEQ

Studio EQ is a high-quality 4-band parametric stereo equalizer. All four bands can act as fully parametric peak filters. In addition, the low and high bands can act as either shelving filters (three types) or as cut filters (low-pass/high-pass).



Main Layout

RESET

Click this button while holding [Alt/option] to reset all parameter values.

SPECTRUM

Shows the spectrum before and after filtering.

OUTPUT

Adjusts the overall output level.

AUTO GAIN

If this button is activated, the gain is automatically adjusted, keeping the output level nearly constant regardless of the EQ settings.

Band Settings



Activate/Deactivate Band

Activates/Deactivates the corresponding band.

NOTE

If a band is deactivated, you can still modify its parameters.

FREQ

Sets the frequency of the corresponding band. You can set the frequency either in Hz or as a note value. If you enter a note value, the frequency is automatically changed to Hz. For example, a note value of A3 sets the frequency to 440 Hz. When you enter a note value, you can also enter a cent offset. For example, enter "A5 -23" or "C4 +49." When you enter a note value, you can also enter a cent offset. For example, enter "A5 -23" or "C4 +49."

NOTE

- You can adjust the [Freq] parameter of a band in the graphical editor by clicking the corresponding handle while holding [Alt/option] and moving the mouse left and right.
- Make sure that you enter a space between the note and the cent offset. Only in this case, the cent offsets are taken into account.

INV

Inverts the gain value of the filter. Use this button to filter out unwanted noise. When looking for the frequency to omit, it sometimes helps to boost it in the first place (set the filter to positive gain). After you have found the frequency of the noise, you can use the [Inv] button to cancel it out.

Q

For [Peak] filters, this parameter controls the width of the band. For [Shelf] filters, it adds a drop or a boost, depending on the gain setting of the band. For [Cut] filters, it adds a resonance.

NOTE

You can adjust the [Q] parameter of a band in the graphical editor by clicking the corresponding handle while holding [Shift] and moving the mouse up and down. Alternatively, you can point on the handle and turn the mouse wheel.

GAIN

Sets the amount of attenuation/boost for the corresponding band.

NOTE

- You can adjust the [Gain] parameter of a band in the graphical editor by clicking the corresponding handle while holding [Ctrl]/[command] and moving the mouse up and down.
- This parameter is not available for [Cut] filters.

Filter type

For the low and high band, you can choose between three types of shelving filters, a peak filter (band-pass), and a cut filter (low-pass/high-pass). If [Cut] mode is selected, the [Gain] parameter is fixed.

- [Shelf I] adds resonance in the opposite gain direction slightly above the set frequency.
- [Shelf II] adds resonance in the gain direction at the set frequency.
- [Shelf III] is a combination of [Shelf I] and [Shelf II].

Filter

MorphFilter

MorphFilter lets you mix low-pass, high-pass, band-pass, and band-reduction filter effects, allowing for creative morphings or mixtures between two filters.



Filter A buttons

Allow you to select the characteristic of the first filter.

- **Low Pass**

Eliminates high-frequency signal components. Filter slopes of 6, 12, 18, and 24 dB per decade are available.

- **Band Pass**

Allows signals falling within a certain frequency range to pass through. Filter slopes of 12 and 24 dB per decade are available.

Filter B buttons

Allow you to select the characteristic of the second filter.

- **High Pass**

Eliminates low-frequency signal components. Filter slopes of 6, 12, 18, and 24 dB per decade are available.

- **Band Rejection**

Lets all frequencies pass, except those in the stop band. Filter slopes of 12 and 24 dB per decade are available.

Resonance Factor

Sets the resonance value of both filters simultaneously.

Frequency

Sets the cutoff frequency of both filters simultaneously.

Graphical display

Visualizes the settings for all parameters. The handle allows you to adjust the MorphFactor and the Frequency parameters simultaneously.

Output meter

Shows the level of the output signal.

Morph Factor

Allows you to mix the output between both filters.

Modulation Plug-ins

Cloner

Cloner adds up to 4 detuned and delayed voices to the signal for rich modulation and chorus effects.



Graphical Display

VOICES

Sets the number of voices. Each voice has a [Detune] and a [Delay] slider.

[DETUNE] Slider (1 to 4)

Sets the relative amount of detune for each voice. Positive and negative values can be set. If set to zero, no detune takes place for that voice.

[DELAY] Slider (1 to 4)

Sets the relative delay amount for each voice. If set to zero, no delay takes place for that voice.

DETUNE

Sets the overall amount of detune for all voices. If set to zero, no detune takes place regardless of the [Detune] slider settings.

NATURAL

Changes the pitch algorithm that is used for detune.

HUMANIZE (Detune)

Sets the amount of detune variation if [Static Detune] is deactivated. With [Humanize], the detune is continuously modulated for a more natural effect.

STATIC (Detune)

Activate this button to use a static amount of detune.

MIX

Sets the level balance between the dry signal and the wet signal. If the effect is used as a send effect, set this parameter to the maximum value, as you can control the dry/effect balance with the send level.

SPATIAL

Spreads the voices across the stereo spectrum. Turn the control clockwise for a deeper stereo effect.

OUTPUT (-12 to 12 dB)

Sets the output gain.

DELAY

Governs the overall depth of the delay for all voices. If set to zero, no delay takes place regardless of the [Delay] slider settings.

HUMANIZE (Delay)

Controls the amount of delay variation if [Static Delay] is deactivated. With [Humanize], the delay is continuously modulated for a more natural effect.

STATIC (Delay)

Activate this button to use a static delay amount.

FX Modulator

This multi-effect modulation plug-in combines several modulation effects that allow for extensive sound shaping, from classic ducking effects to exciting rhythmic patterns. You can create custom-shaped LFOs and modulate up to 6 integrated effect modules at once. The LFO can be triggered via MIDI or separate side-chain inputs, where the envelope of the side-chain signal is added to the LFO.

NOTE

VST Rack does not support side-chaining. Consequently, this plug-in does not accept side-chain input.



Curve Section

This section contains a curve editor that allows you to define the modulation curve. You can apply predefined curve shapes, or create and store custom curve shapes.

Factory

The factory bank allows you to apply predefined curve shapes to the whole curve or a selection range. You cannot change the assignment of the factory slots. However, you can adjust factory curves in the editor after applying it.

Bank 1-3

User banks allow you to apply custom curve shapes to the whole curve or a selection range.

- To add the curve shape that is displayed in the editor to a user bank, click an empty slot.
- To remove the curve shape of a slot, click x and click again to confirm. To cancel, click elsewhere. Load opens a browser window that shows the available bank presets.
- To load a bank preset to the selected bank, double-click it.

- To filter the list of bank presets, click Set Up Window Layout at the top right of the browser window, activate Filters, and select your filter settings.
Save allows you to save the curve shapes in the selected bank as a bank preset.

Lock Banks 1-3 When Loading Presets

Locks/Unlocks Bank 1, Bank 2, and Bank 3 while browsing through presets.

Curve editor

Shows the shape of the modulation curve and allows you to manually adjust it. The editor allows the following editing techniques:

- To add a node, double-click the curve.
- To move a node, drag it. If you move a node close to a horizontal or vertical grid line, it automatically snaps to the grid. To suspend this behavior, hold Shift while dragging.
- To delete a node or shape handle, double-click it.
- To move, delete, or flip multiple nodes or shape handles, drag a selection rectangle around them and edit them.
- To modify the shape of the curve, drag the curve between two nodes.
- To create an S-shaped curve, hold Shift and drag the shape handle left of a node.
- To compress or expand the curve, hold Alt and drag a node of a selection up or down.

Threshold

Sets the threshold value for 1 Cycle trigger mode. The threshold value is shown as a horizontal line within the display. This control is only available if Side-Chain is activated and 1 Cycle mode is selected in the Trigger section.

Duplicate Curve

Duplicates the shape of the curve.

Undo/Redo

Undoes/Redoes actions in the curve editor.

NOTE

Each modulation curve has its own undo/redo history.

Shift Curve to the Left

Moves the curve to the left.

Shift Curve to the Right

Moves the curve to the right.

Flip Vertically

Flips the shape of the whole curve or a selection range vertically.

Flip Horizontally

Flips the shape of the whole curve or a selection range horizontally.

Select All Curve Points

Selects all curve nodes and handles.

Reset Curve

Resets the curve to default.

Create Random Curve

Applies a random shape to the whole curve or a selection range.

Show Curve Editing Instructions

Shows/Hides a list of techniques for editing the modulation curve in the editor.

Main Section

This section contains the general modulation parameters and settings.

Time

Sets the time base for the modulation curve individually for each modulated effect parameter. The default value of 1/1 means that 1 cycle of modulation has a length of 1 bar. The tempo of the song is automatically taken into account.

If Beats is activated, Time is displayed in beats instead of milliseconds.

If Phase Sync is activated, the modulation curve is synchronized to the song position in the project, making the modulation highly reproducible. If Phase Sync is deactivated, Time allows a free modulation, making effects like chorus and flanger sound unpredictable.

Smooth

Smooths sharp edges and steep slopes of the modulation curve individually for each modulated effect parameter. This can avoid artifacts that are caused by abrupt value changes.

Mix

Sets the level balance between the dry and the wet signal.

Lock Mix Value When Loading Presets

Locks/Unlocks the balance between dry and wet signal when loading presets.

Output

Sets the output level.

Effects Section

This section allows you to add, edit, and remove effects. You can create effect chains by adding up to 6 different effect modules from 14 effects available overall. You can modify the order of the effects in the chain by dragging the modules.

For each effect, you can see which is its target parameter that is affected by the modulation curve. Some effects have multiple target parameters. A filter bank allows you to restrict the modulation to a defined frequency range.

Effects

Shows/Hides the effect parameters and filter bank settings.

Target

Shows the parameter that is modulated by the modulation curve. For effects with multiple target parameters, click a parameter to show and edit its specific modulation curve.

Filter Bank

Activates/Deactivates the filter bank for the selected module. If activated, the effect is restricted to the range defined on the frequency display. Frequencies below and above this range are bypassed.

Spectrum

Activates/Deactivates the spectrum display for the effect signal on the frequency display.

NOTE

Activating this option increases the CPU load.

Solo

Allows you to listen to the set frequency range exclusively. Frequencies below and above this range are muted.

Frequency display

Shows the spectrum of the effect signal and allows you to define the frequency range of the filter bank. To adjust the range, drag the frequency handles or drag the area between the handles.

Trigger Section

This section allows you to determine how the modulation curves are applied when triggered via MIDI or side-chaining.

The following settings are available for MIDI triggering:

Trigger

Shows/Hides the settings for MIDI and side-chain triggering.

MIDI

Activates/Deactivates triggering via MIDI.

Mode

Sets the mode for MIDI triggering.

- If Hold is selected, the modulation curve is applied as long as a MIDI note is played.
- If 1 Cycle is selected and a MIDI note is played, the modulation curve is only applied once, regardless of the note length. You must play a note again to apply the curve again.

NOTE

This mode is useful to create MIDI-triggered ducking effects with the Volume or Compressor effect modules.

The following settings are available for side-chain triggering:

Side-Chain

Activates/Deactivates triggering via side-chain input.

Mode

Sets the mode for side-chain triggering.

- If Continuous is selected, the envelope of the side-chain signal is added to the modulation curve.
- If 1 Cycle is selected and the envelope of the side-chain signal exceeds the threshold, the modulation curve is applied once.

NOTE

- This mode is useful to create envelope-triggered ducking effects with the Volume or Compressor effect modules.
- If this mode is selected, the envelope of the side-chain signal and the threshold are shown in the curve editor, and you can set the Threshold there.

Input

Sets the side-chain input. If Internal is selected, the input signal of the plug-in is used as the side-chain source. If Side-Chain 1 to Side-Chain 6 is selected, the signal of tracks that are routed to the corresponding plug-in side-chain input is used.

Side-Chain Filter Listen

Allows you to solo the side-chain filter. This way, you can quickly check the part of the signal that is filtered out using the current settings.

Frequency

Sets the frequency of the side-chain filter.

You can set the frequency either in Hz or as a note value. If you enter a note value, the frequency is automatically changed to Hz. For example, a note value of A3 sets the frequency to 440 Hz. When you enter a note value, you can also enter a cent offset. For example, enter A5 -23 or C4 +49.

NOTE

- Ensure that you enter a space between the note and the cent offset. Only then are cent offsets taken into account.

Q

Sets the resonance or width of the side-chain filter.

Attack

Sets the attack time for the side-chain envelope signal.

Release

Sets the release time for the side-chain envelope signal.

Gain

Attenuates or boosts the side-chain envelope signal.

Effect Modules

Modules allow you to create an effect chain. Each effect can be used only once in the module chain. You can drag modules in the module chain to rearrange them and change the processing order.

General Effect Settings

The following settings are available for each module:

Bypass



Bypasses the module. This allows you to compare the sound of the unprocessed signal to that of the processed signal.

Solo



Solos the module. Only one module can be soloed at a time.

Remove

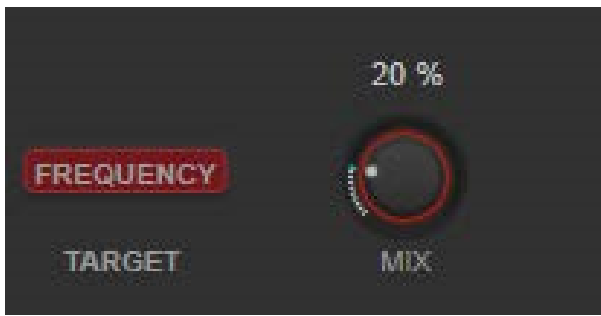


Removes the module from the module chain.

The following effect modules are available:

Chorus

This is a single-stage chorus effect that doubles the input signal with a slightly detuned version.

**Target**

Shows the parameter that is modulated. Frequency changes the modulation.

Mix

Sets the level balance between the dry and the wet signal.

Flanger

This is a classic flanger effect.

**Target**

Shows the parameter that is modulated. Frequency changes the modulation.

Feedback

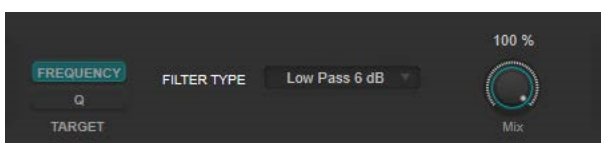
Determines the character of the flanger effect. Higher settings produce a more metallic-sounding sweep.

Mix

Sets the level balance between the dry and the wet signal.

Filter

This is a filter modulation effect. The frequency and resonance of the filter can be modulated by an LFO.

**Target**

Selects a parameter for editing its modulation curve. Frequency changes the filter frequency. Q changes the resonance of the filter.

Filter Type

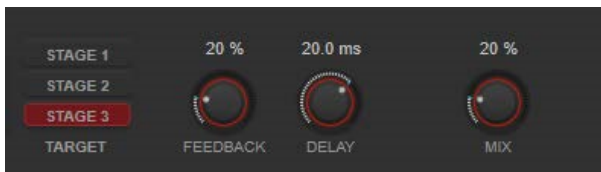
Sets the filter type. Low-pass, high-pass, band-pass, and notch filters are available.

Mix

Sets the level balance between the dry and the wet signal.

MultiMod

This is a three-stage chorus-flanger combined effect.



Target

Selects a stage for editing its modulation curve.

Feedback

Determines the character of the flanger effect. Higher settings produce a more metallic-sounding sweep.

Delay

Affects the frequency range of the modulation sweep by adjusting the initial delay time.

Mix

Sets the level balance between the dry and the wet signal.

Width

This effect expands or reduces the stereo width.

NOTE

This effect module only works if the plug-in is applied on a stereo track.



Target

Shows the parameter that is modulated. Mix changes the level balance between the dry signal and the wet signal.

Delay

Increases the amount of differences between the left and right channels to further increase the stereo effect.

Color

Generates additional differences between the channels to increase the stereo enhancement.

Mono In

Defines the input signal as dual-mono audio.

NOTE

For the stereo effect to work, you must activate this option if your input signal is a dual-mono audio file.

Listen Mono Out

Sets the output to mono. This allows you to check for unwanted coloring of the sound which can occur when creating an artificial stereo image.

Pan

This is a panning effect.

NOTE

This effect module only works if the plug-in is applied on a stereo track.



Target

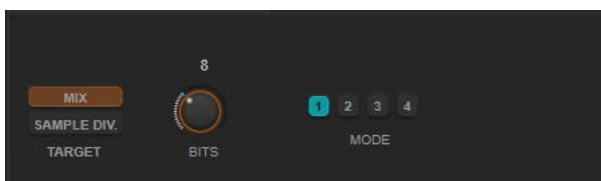
Shows the parameter that is modulated. Pan changes the level balance between the left and right channel.

Pan Law

Sets the stereo pan law. 6 dB, 4.5 dB, 3 dB, and 0 dB determine the attenuation of the signal at the center position. Equal Power means that the power of the signal remains the same regardless of the pan setting.

Bit Crusher

This effect uses bit reduction to decimate and truncate the input audio signal to get a noisy, distorted sound.



Target

Selects a parameter for editing its modulation curve. Mix changes the level balance between the dry signal and the wet signal. Sample Div. changes the amount by which the audio samples are decimated. At the highest setting, nearly all of the information describing the original audio signal is eliminated, turning the signal into unrecognizable noise.

Bits (0 to 24 bits)

Defines the bit resolution. A setting of 24 gives the highest audio quality, while a setting of 1 creates mostly noise.

Mode

Allows you to select one of the four operating modes. In each mode, the effect sounds differently. Modes 1 and 3 are nastier and noisier, while modes 2 and 4 are more subtle.

Overdrive

This effect creates a tube-like overdrive effect.



Target

Shows the parameter that is modulated. Drive adds harmonics to the output signal.

Level

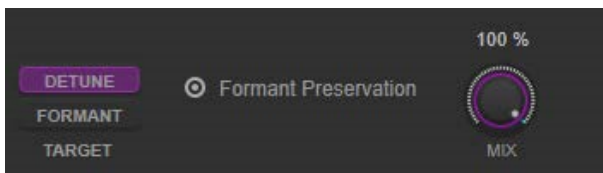
Adjusts the output level.

Mix

Sets the level balance between the dry and the wet signal.

Pitch Shifter

This is a pitch-shifting effect.



Target

Selects a parameter for editing its modulation curve. Detune changes the pitch of the input signal in semitones. Formant changes the natural timbre of the input signal.

Formant Preservation

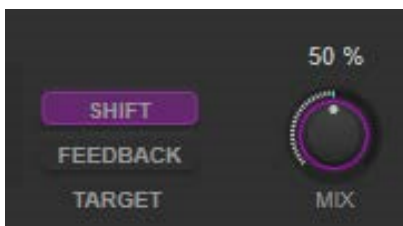
Keeps the formants when changing the pitch with the Detune control.

Mix

Sets the level balance between the dry and the wet signal.

Frequency Shifter

This effect shifts each frequency of the input signal by a fixed amount, which alters the harmonic relations. Adding feedback produces a sound similar to a phaser.



Target

Selects a parameter for editing its modulation curve. Shift changes the amount of frequency shift. Feedback changes the amount of the signal that is sent from the output of the effect back to its input.

Mix

Sets the level balance between the dry and the wet signal.

Compressor



Target

Shows the parameter that is modulated. Trig. Level changes the envelope. Compared to a standard compressor effect which generates the envelope from the input signal, this module uses the modulation curve as the envelope. When the envelope signal exceeds the threshold, compression is applied.

Threshold

Determines the level where the compressor kicks in.

Ratio

Sets the amount of gain reduction applied to signals above the set threshold. A ratio of 3:1 means that for every 3 dB the input level increases, the output level increases by 1 dB.

Attack (0.1 to 100 ms)

Determines how fast the compressor responds to signals above the set threshold.

If the attack time is long, more of the early part of the signal passes through unprocessed.

Release (10 to 1000 ms)

Sets the time after which the gain returns to its original level when the signal drops below the threshold.

Make-Up (0 to 24 dB or Auto mode)

Compensates for output gain loss caused by compression.

Volume



Target

Shows the parameter that is modulated. Level changes the output level.

Mix

Sets the level balance between the dry and the wet signal.

Time Shifter



Target

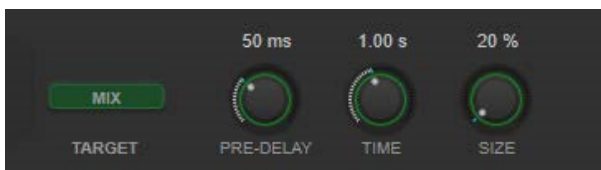
Shows the parameter that is modulated. Delay changes the delay time.

Mix

Sets the level balance between the dry and the wet signal.

Reverb

This is a versatile reverb for realistic room ambience and reverb effects.



Target

Shows the parameter that is modulated. Mix changes the level balance between the

dry signal and the wet signal.

Pre-Delay

Determines how much time passes before the reverb is applied. This allows you to simulate larger rooms by increasing the time it takes for the first reflections to reach the listener.

Time

Allows you to set the reverb time in seconds.

Size

Alters the delay times of the early reflections to simulate larger or smaller spaces.

Rotary

This modulation effect simulates the effect of a rotating speaker.



A rotary speaker cabinet features speakers rotating at variable speeds to produce a swirling chorus effect, commonly used with organs.

Speed settings

SPEED MOD Control (MIDI)

Speed Selector (stop/slow/fast)

Allows you to control the speed of the rotary speaker.

SPEED MOD

If the [Set Speed Change Mode] setting is set to the right, this knob allows you to modulate the rotary speed.

Set Speed Change Mode

If this is set to the left, the speed selector settings are taken into account. If this is set to the right, you can modulate the speed with the [Speed Mod] knob.

Additional settings

OVERDRIVE

Applies a soft overdrive or distortion.

CROSSOVER

Sets the crossover frequency (200 to 3000 Hz) between the low and high frequency loudspeakers.

Horn

SLOW

Allows for a fine adjustment of the high rotor [slow] speed.

FAST

Allows for a fine adjustment of the high rotor [fast] speed.

ACCEL.

Allows for a fine adjustment of the high rotor acceleration time.

AMP MOD

Controls the high rotor amplitude modulation.

FREQ MOD

Controls the high rotor frequency modulation.

Bass

SLOW

Allows for a fine adjustment of the low rotor [slow] speed.

FAST

Allows for a fine adjustment of the low rotor [fast] speed.

ACCEL.

Allows for a fine adjustment of the low rotor acceleration time.

AMP MOD

Adjusts the modulation depth of the amplitude.

LEVEL

Adjusts the overall bass level.

Mics

PHASE

Adjusts the phasing amount in the sound of the high rotor.

ANGLE

Sets the simulated microphone angle. A value of 0 ° corresponds to a mono miking setup with a single microphone in front of the speaker cabinet, 180 ° corresponds to a stereo miking setup with a microphone on each side of the cabinet.

DISTANCE

Sets the simulated microphone distance from the speaker in inches.

Final Settings

OUTPUT

Sets the output level.

MIX

Sets the level balance between the dry signal and the wet signal.

StudioChorus

StudioChorus is a two-stage chorus effect that adds short delays to the signal and modulates the pitch of the delayed signals to produce a doubling effect. The two separate stages of chorus modulation are independent and are processed serially (cascaded).



DELAY

Affects the frequency range of the modulation sweep by adjusting the initial delay time.

WIDTH

Sets the depth of the chorus effect. Higher settings produce a more pronounced effect.

SPATIAL

Sets the stereo width of the effect. Turn clockwise for a wider stereo effect.

MIX

Sets the level balance between the dry signal and the wet signal. If the effect is used as a send effect, set this parameter to the maximum value, as you can control the dry/effect balance with the send level.

RATE

If [Tempo Sync] is activated, [Rate] allows you to specify the base note value for synchronizing the modulation sweep to the tempo of the host application (1/1 to 1/32, straight, triplet, or dotted).

If [Tempo Sync] is deactivated, the sweep rate can be set freely with the [Rate] dial.

SYNC

Activates/Deactivates tempo sync.

Waveform Shape

Allows you to select the modulation waveform, altering the character of the chorus sweep. A sine and a triangle waveform are available.

LO FILTER/HI FILTER

Allows you to roll off low and high frequencies of the effect signal.

Pitch Shift

VoiceDesigner

VoiceDesigner is a versatile sound design plug-in that provides extreme pitch-shifting and morphing effects, using an external side-chain signal or the integrated sound generator. You can also use the plug-in to create robotic voices with just one click. The mixing section allows you to set different levels and frequency ranges for the dry, wet, side-chain, and generator signal.



Robot

Activates/Deactivates the robotic voice effect. The Whisper parameter allows you to soften the sound of the robotic voice.

Morph

Activates/Deactivates the morphing effect. The input signal is processed using characteristics of a side-chain signal or of the integrated sound generator. The Mode selector allows you to toggle between the two morphing modes A and B.

FX

Activates/Deactivates the Delay and the Feedback sound effects.

Delay

Adds a delay to the signal.

Feedback

Adds feedback to the signal.

Transition

Allows you to morph the input signal into the side-chain or generator signal. The slider sets the morphing amount. This parameter is only available in morphing mode A.

Response

Sets the response time of the morphing algorithm. Fast response settings preserve fast changing transients of the input signals, such as consonants in speech signals. Slower response settings lead to blurry, pad-like sounds. This parameter is only available in morphing mode B.

Swap

Swaps source and target for the morphing effect. This parameter is only available in morphing mode B.

Resolution

Sets the resolution of the morphing signal. Lower values result in a more rhythmical sound. Higher values preserve the intelligibility of speech signals.

Generator shape selector

Lets you choose the characteristic of the internal sound generator. White Noise and Pink Noise, and Square and Sawtooth waveforms are available.

Frequency

Sets the frequency for the Square and Sawtooth waveforms of the internal sound generator.

Detune

Changes the pitch of the input signal.

Formant

Changes the natural timbre of the input signal.

Preserve

Keeps the formants when changing the pitch with the Detune control.

Spatial

Adds an ambience effect by using slightly different settings on all channels.

Dry

Sets the level for the dry input signal. The slider below allows you to set a low-cut and high-cut filter for the input signal.

Generator/Side-Chain

If external side-chain is activated, this control sets the level of the side-chain input. If external side-chain is deactivated, it sets the level for the internal sound generator.

The slider below allows you to set a low-cut and high-cut filter for the generator or side-chain signal.

Wet

Sets the level for the effect signal. The slider below allows you to set a low-cut and high-cut filter for the effect signal.

Output

Sets the output level.

Output meter

Shows the level of the output signal.

Reverb Plug-ins

REvelation

REvelation produces a high-quality algorithmic reverb with early reflections and reverb tail.



The early reflections are responsible for the spatial impression in the first milliseconds of the reverb. For emulating different rooms, you can choose between different early reflections patterns and adjust their size. The reverb tail, or late reverberation, offers parameters for controlling the room size and the reverb time. You can adjust the reverb time individually in 3 frequency bands.

PRE DELAY

Determines how much time passes before the reverb is applied. This allows you to simulate larger rooms by increasing the time it takes for the first reflections to reach the listener.

EARLY REFLECTION

Here, you select an early reflections pattern. The early reflections pattern contains the most important delays that deliver the key information for the spatial impression of the room.

ER/TAIL

Sets the level balance between the early reflections and the reverb tail. At a setting of 50 %, early reflections and tail have the same volume. Settings below 50 % raise the early reflections and lower the tail, as a result the sound source moves towards the front of the room. Settings above 50 % raise the tail and lower the early reflections, as a result the sound source moves towards the back of the room.

SIZE

Adjusts the length of the early reflections pattern. At a setting of 100 %, the pattern is applied with its original length and the room sounds the most natural. At settings below 100 %, the early reflections pattern is compressed and the room is perceived smaller.

LOW CUT

Attenuates the low frequencies of the early reflections. The higher this value, the less low frequencies are present in the early reflections.

HIGH CUT

Attenuates the high frequencies of the early reflections. The lower this value, the less high frequencies the early reflections will have.

DELAY

Delays the onset of the reverb tail.

ROOM SIZE

Controls the dimensions of the simulated room. At a setting of 100 %, the dimensions correspond to a cathedral or a large concert hall. At a setting of 50 %, the dimensions correspond to a medium-sized room or studio. Settings below 50 % simulate the dimensions of small rooms or a booth.

MAIN TIME

Controls the overall reverb time of the tail. The higher this value, the longer the reverb tail will decay. At a setting of 100 %, the reverb time is infinitely long. The [Main Time] parameter also represents the mid band of the reverb tail.

HIGH TIME

Controls the reverb time for the high frequencies of the reverb tail. With positive values, the decay time of the high frequencies is longer. With negative values, it is shorter. Frequencies are affected depending on the [High Freq] parameter.

LOW TIME

Controls the reverb time for the low frequencies of the reverb tail. For positive values, low frequencies decay longer and vice versa. Frequencies will be affected depending on the [Low Freq] parameter.

HIGH FREQ

Sets the cross-over frequency between the mid and the high band of the reverb tail. You can offset the reverb time for frequencies above this value from the main reverb time with the [High Time] parameter.

LOW FREQ

Sets the cross-over frequency between the low and the mid band of the reverb tail. The reverb time for frequencies below this value can be offset from the main reverb time with the [Low Time] parameter.

SHAPE

Controls the attack of the reverb tail. At a setting of 0 %, the attack is more immediate, which is a good setting for drums. The higher this value, the less immediate the attack.

DENSITY

Adjusts the echo density of the reverb tail. At a setting of 100 %, single reflections from walls cannot be heard. The lower this value, the more single reflections can be heard.

HIGH CUT

Attenuates the high frequencies of the reverb tail. The lower this value, the less high frequencies the reverb tail will have.

WIDTH

Controls the width of the stereo image. At a setting of 0 %, the output of the reverb is mono, at 100 % it is stereo.

MIX

Sets the level balance between the dry signal and the wet signal. If the effect is used as a send effect, set this parameter to the maximum value, as you can control the dry/effect balance with the send level.

Lock Mix Value

Activate this button (padlock symbol) next to the [Mix] parameter to lock the dry/wet balance while browsing

through the available presets.

Modulation

Modulation allows you to enrich the reverb tail through subtle pitch modulations.

Modulation Rate

Specifies the frequency of the pitch modulation.

Modulation Depth

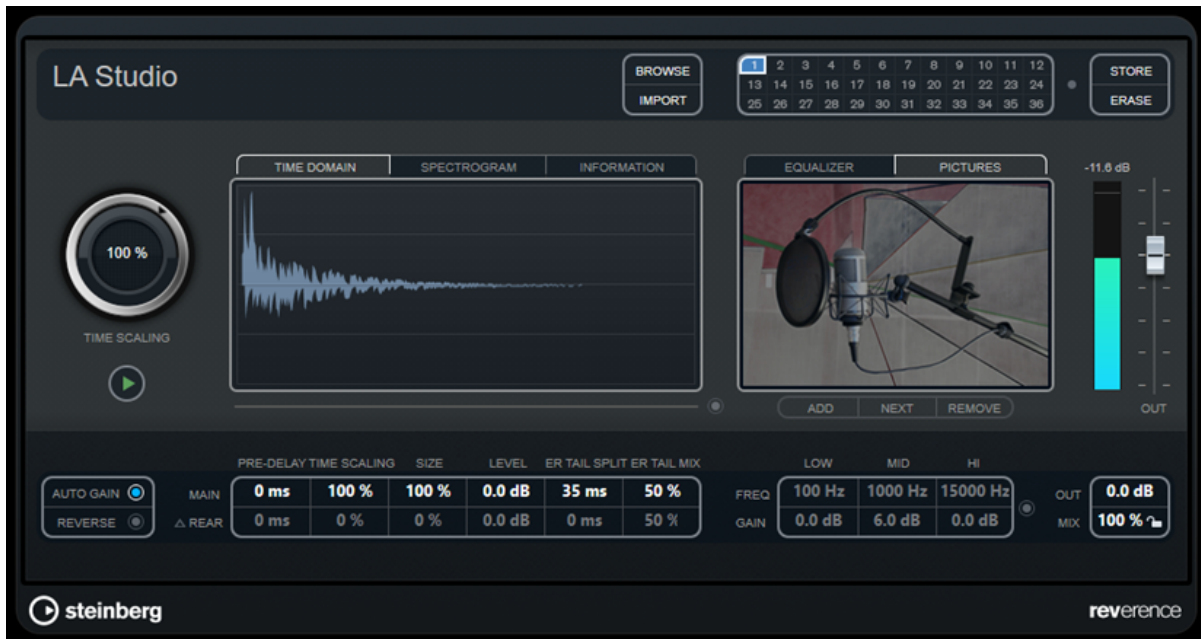
Sets the intensity of the pitch modulation.

Modulation Activate

Activates/Deactivates the chorusing effect.

REVerence

REVerence is a convolution tool that allows you to apply room characteristics (reverb) to the audio.



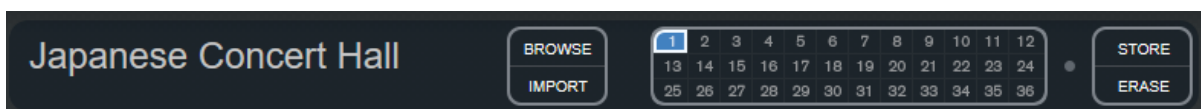
This is done by processing the audio signal according to an impulse response – a recording of an impulse in a room or another location that recreates the characteristics of the room. As a result, the processed audio sounds as if it were played in the same location. Included with the plug-in are top quality samples of real spaces to create reverberation.

NOTE

REVerence can be very demanding in terms of RAM. This is because the impulse responses that you load into the program slots are preloaded into RAM to guarantee artifact-free switching between programs. Therefore, you should always load only those programs that you need for a given task.

Program Matrix

A program is the combination of an impulse response and its settings. These include reverb settings, EQ settings, pictures, and output settings. The program matrix allows you to load programs or to view the name of the impulse response.



Program Name

In the upper left corner of the plug-in panel, either the name of the loaded impulse response file or the name of the program is shown. After loading an impulse response, its number of channels and the length in seconds are displayed for a few seconds.

BROWSE

This button opens a browser window showing the available programs. If you select a program in the browser, it is loaded into the active program slot. To filter the list of impulse responses by room type or the number of channels, for example, activate the [Filter] section (by clicking the [Set Up Window Layout] button at the top right of the browser window).

IMPORT

Click this button to load impulse response files from an external source. The files should have a maximum length of 10 seconds. Longer files are automatically cut.

Program Slots

Use these slots to load all the impulse responses that you want to work with in a session. The selected program slot is indicated by a white frame. Used slots are shown in blue. A red program slot indicates that the impulse response file is missing. Double-clicking an empty program slot opens a browser window, showing the available programs. Clicking a used program slot recalls the corresponding program and loads it into REVerence. If you move the mouse over a used slot, the corresponding program name is displayed below the active program name.

Smooth Parameter Changes

This button is located between the program slots and the [Store]/[Erase] buttons. If it is activated, a crossfade is performed when switching programs. Leave this button deactivated while looking for a suitable program or an appropriate setting for an impulse response. Once you have set up the program matrix to your liking, activate the button to avoid hearing artifacts when switching between programs.

STORE

Stores the active impulse response and its settings as a program.

ERASE

Removes the selected program from the matrix.

Programs vs. Presets

You can save your REVerence settings as VST plug-in presets or programs. Both presets and programs use the file extension .vstpreset and appear in the same category in the MediaBay, but they are represented by different icons.

Preset



A REVerence preset contains all settings and parameters for the plug-in, that is, a link to all loaded impulse responses along with their parameter settings and positions in the program matrix. Presets are loaded via the presets pop-up menu at the top of the plug-in panel.

NOTE

Manually imported impulse responses themselves are not part of the program or preset. If you want to move your project to a different computer, you have to move the impulse responses as well.

Program



A REVerence program only contains the settings related to a single impulse response. Programs are loaded and managed via the program matrix.

Presets

Presets are useful in the following situations:

- To save a complete setup with different impulse responses for later use (for example, different setups for explosion sounds that can be reused for other scenes or movies).
- When you want to save different parameter sets for the same impulse response so that you can later choose the set that best suits your needs.

Programs

Programs offer the following advantages:

- Up to 36 programs can be loaded into the program matrix for instant recall.
- A program provides a quick and easy way to save and recall the settings for a single impulse response, allowing for short loading times.
- When automating a project and loading a REVerence program, only one automation event is written. If you load a plug-in preset instead (which contains a lot more settings than a program), a lot of unnecessary automation data (for the settings that you did not use) is written.

Setting up Programs

Procedure

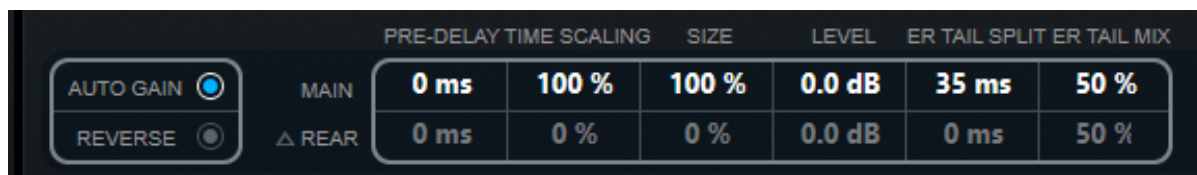
1. In the program matrix, click on a program slot to select it.
A blinking white frame indicates that this program slot is selected.
2. Click the [Browse] button or click the empty slot again to load one of the included programs. You can also import a new impulse response file.
3. In the browser, select the program containing the impulse response that you want to use and click [OK].
The name of the loaded impulse response is shown in the upper left corner of the REVerence panel.
4. Set up the REVerence parameters as and click the [Store] button to save the impulse response with the current settings as a new program.
5. Set up as many programs as you need by following the steps above.

NOTE

If you want to use your set of programs in other projects, save your settings as a plug-in preset.

Reverb Settings

The reverb settings allow you to change the characteristics of the room.



MAIN

All values shown in the top row apply to all speakers or to the front channels if you are working with surround tracks.

REAR

If you are working with surround tracks up to 5.1, you can use this row to set up an offset for the rear channels.

AUTO GAIN

If this button is activated, the impulse response is automatically normalized.

REVERSE

Reverses the impulse response.

PRE-DELAY

Determines how much time passes before the reverb is applied. This allows you to simulate larger rooms by increasing the time it takes for the first reflections to reach the listener.

TIME SCALING

Controls the reverb time.

SIZE

Determines the size of the simulated room.

LEVEL

A level control for the impulse response. This governs the volume of the reverb.

ER TAIL SPLIT

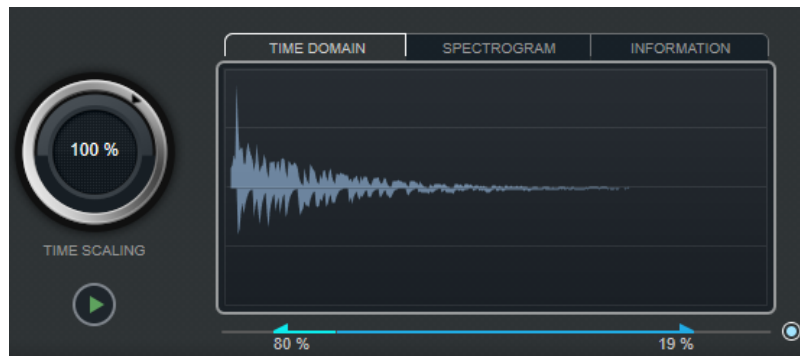
Sets a split point between the early reflections and the tail, allowing you to determine where the reverb tail begins. A value of 60 means that the split point is set to 60 ms.

ER TAIL MIX

Allows you to set up the relation of early reflections and tail. Values above 50 attenuate the early reflections and values below 50 attenuate the tail.

Impulse Response Display

The display section allows you to view the impulse response details and to change the length of the response.



TIME SCALING

This wheel lets you adjust the reverb time.

Play



When clicking this button to apply the loaded impulse response, a short click is played. This provides a neutral test sound that makes it easier for you to know how different settings influence the reverb characteristics.

TIME DOMAIN

This display shows the waveform of the impulse response.

SPECTROGRAM

This display shows the analyzed spectrum of the impulse response. Time is displayed along the horizontal axis, frequency along the vertical axis, and volume is represented by the color.

INFORMATION

This display shows additional information, such as the name of the program and the loaded impulse response, the number of channels, the length, and Broadcast Wave File information.

Activate Impulse Trimming

Use this button at the bottom right of the impulse display section to activate impulse trimming. The Trim slider is shown below the Impulse display.

Trim

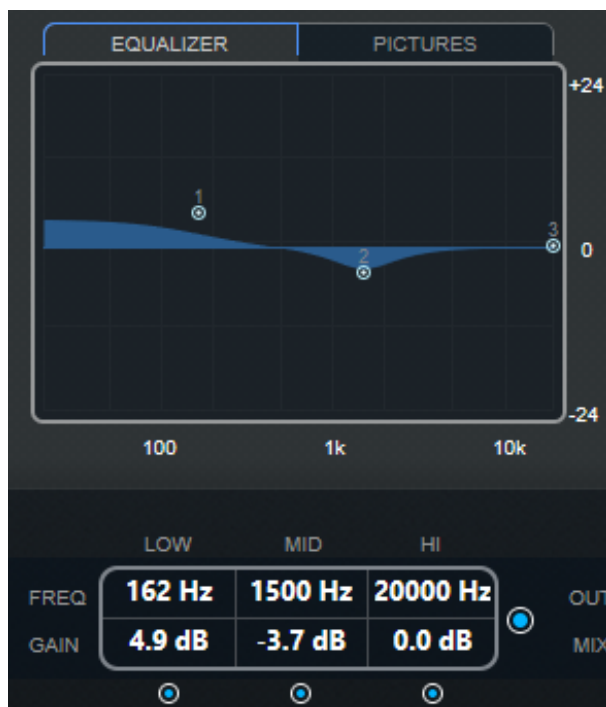
This slider allows you to trim the start and end of the impulse response. Drag the front handle to trim the start of the impulse response, and the end handle to trim the reverb tail.

NOTE

The impulse response is cut without any fading.

EQ Settings

In the Equalizer section, you can tune the sound of the reverb.



EQ Curve

Shows the EQ curve. You can use the EQ parameters below the display to change the EQ curve, or modify the curve manually by dragging the curve points.

Activate EQ

This button to the right of the EQ parameters activates the EQ for the effect plug-in.

Low Shelf On

Activates the low shelf filter that boosts or attenuates frequencies below the cutoff frequency by the specified amount.

LOW FREQ (20 to 500)

Sets the frequency of the low band.

LOW GAIN (-24 to +24)

Sets the amount of attenuation/boost for the low band.

Mid Peak On

Activates the mid peak filter that creates a peak or notch in the frequency response.

MID FREQ (100 to 10000)

Sets the center frequency of the mid band.

MID GAIN (-12 to +12)

Sets the amount of attenuation/boost for the mid band.

Hi Shelf On

Activates the high shelf filter that boosts or attenuates frequencies above the cutoff frequency by the specified amount.

HI FREQ (5000 to 20000)

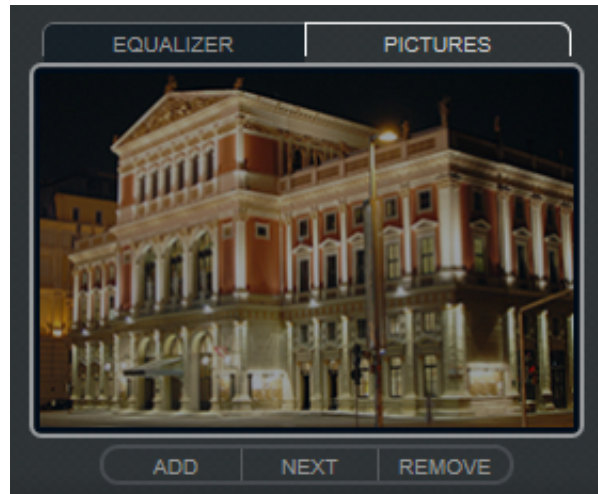
Sets the frequency of the high band.

HI GAIN (-24 to +24)

Sets the amount of attenuation/boost for the high band.

Pictures Section

In the [Pictures] section, you can load graphics files to illustrate the setting, for example, the recording location or microphone arrangement of the loaded impulse response. Up to five pictures can be loaded.



NOTE

Pictures are only referenced by the plug-in and are not copied to the project folder.

ADD

Opens a file dialog where you can navigate to the graphics file that you want to import. JPG, GIF, and PNG file formats are supported.

NEXT

If several pictures are loaded, you can click this button to display the next image.

REMOVE

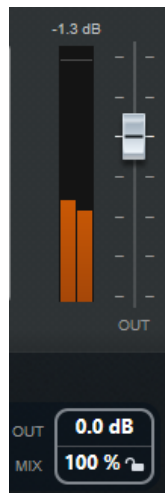
Deletes the active picture.

NOTE

This does not remove the graphics file from the reference source.

Output Settings

In the output section, you can control the overall level and determine the dry/wet mix.



Output Meter

Indicates the overall level of the impulse response and its settings.

OUT

Adjusts the overall output level.

MIX

Sets the level balance between the dry and the wet signal.

Lock Mix Value

Activate this button (padlock symbol) next to the [Mix] parameter to lock the dry/wet balance while browsing through the available presets and programs.

Custom Impulse Responses

In addition to working with the impulse responses included with REVerence, you can import impulse responses from external sources and save them as programs or presets. WAVE and AIFF files with a mono, stereo, true-stereo, or multi-channel (up to 5.0) configuration are supported. If a multi-channel file contains an LFE channel, this channel is ignored.

REVerence uses the same channel width as the rack it is inserted on. When importing impulse response files with more channels than the corresponding rack, the plug-in only reads as many channels as needed. If the impulse response file contains fewer channels than the rack, REVerence generates the missing channels (for example, the center channel as a sum of the left and right channels). If the rear channels are missing (when importing a stereo response file onto a 4.0 channel rack, for example), the left and right channels are also used for the rear channels. In this case you can use the Rear offset parameter to create more spatiality.

Importing Impulse Responses

REVerence allows you to import impulse response files from external sources. Before importing these impulse response files, you can preview their effect.

Procedure

1. In the program matrix, click [Import].
2. In the file dialog that opens, navigate to the location of your impulse response files.
3. Optional: Select an impulse response file to preview it.
4. Select the file that you want to import and click [Open].
The file is loaded into REVerence. The channels from an interleaved file are imported in the same order as in other areas of the program.
5. Make the appropriate settings and add a picture, if available.

Pictures residing in the same folder as the impulse response file or in the parent folder are automatically found and displayed.

6. Click the [Store] button to save the impulse response and its settings as a program. That way you can recall the setup at any time.

Result

The program slot turns blue, indicating that a program is loaded.

NOTE

When saving a program, the impulse response file itself is only referenced. It still resides in the same location as before and is not modified in any way.

After Completing This Task

Repeat these steps for any impulse response files that you want to work with.

True Stereo

Impulse responses recorded as true-stereo files allow you to create a very realistic impression of the corresponding room.

REVerence can only process true-stereo impulse response files with the following channel configuration (in exactly that order): LL, LR, RL, RR.

The channels are defined as follows:

Channel	Signal from this source	Recorded with this microphone
LL	Left source	Left microphone
LR	Left source	Right microphone
RL	Right source	Left microphone
RR	Right source	Right microphone

REVerence automatically works in true-stereo mode if the plug-in is inserted on a stereo rack and you load a 4-channel impulse response.

So how can you prevent REVerence from unintentionally processing surround files in true-stereo mode? The answer is a [Recording Method] attribute that can be written to the iXML chunk of the corresponding impulse response file. Whenever you load an impulse response with a 4-channel configuration on a stereo rack, REVerence searches the iXML chunk of the file. If the plug-in finds the [Recording Method] attribute, the following happens:

- If the attribute is set to [TrueStereo], the plug-in works in true-stereo mode.
- If the attribute is set to [A/B] or [Quadro], the plug-in works in normal stereo mode and processes only the L/R channels of the surround file.

Relocating Content

Once you have imported your own impulse responses to REVerence, you can comfortably work with them on your computer. But what if you need to transfer your content to another computer, for example, because you work sometimes with a PC and sometimes with a notebook, or you need to hand over a project to a colleague in the studio?

The factory content is not a problem because it is also present on the other computer. For these impulse responses, you just need to transfer your REVerence programs and presets to be able to access your setups.

User content is a different matter, though. If you have transferred your audio files to an external drive or a

different hard disk location on the other computer, REVerence cannot access the impulse responses any more because the old file paths have become invalid.

Procedure

1. Transfer your audio files to a location that you can access from the second computer (for example, an external hard disk).
If you keep the files in the same folder structure as on the first computer, REVerence automatically finds all files contained in this structure.
2. Transfer any REVerence presets or programs that you need to the second computer.
3. Open REVerence on the second computer and try to load the preset or program that you want to work with.
The [Open Impulse Response] dialog opens.
4. Navigate to the folder that contains your impulse responses.
5. Click [Open].

Result

REVerence is now able to access all the impulse responses stored in this location.

Important

The new path to these audio files has not been saved yet. To make the files permanently available without having to use the locate dialog, you need to save your programs or presets under a different name.

Roomworks

RoomWorks is a highly adjustable reverb plug-in for creating realistic room ambience and reverb effects in stereo and surround formats. The CPU usage is adjustable to fit the needs of any system. From short room reflections to cavern-sized reverb, this plug-in delivers high quality reverberation.



Input Filters

LOW FREQ

Determines the frequency at which the low-shelving filter takes effect. Both the high and low settings filter the input signal prior to reverb processing.

HIGH FREQ

Determines the frequency at which the high-shelving filter takes effect. Both the high and low settings filter the input signal prior to reverb processing.

LOW GAIN

Sets the amount of boost or attenuation for the low-shelving filter.

HIGH GAIN

Sets the amount of boost or attenuation for the high-shelving filter.

Reverb Character

PRE-DELAY

Determines how much time passes before the reverb is applied. This allows you to simulate larger rooms by increasing the time it takes for the first reflections to reach the listener.

SIZE

Alters the delay times of the early reflections to simulate larger or smaller spaces.

REVERB TIME

Allows you to set the reverb time in seconds.

DIFFUSION

Affects the character of the reverb tail. Higher values lead to more diffusion and a smoother sound, while lower values lead to a clearer sound.

WIDTH

Controls the width of the stereo image. At a setting of 0 %, the output of the reverb is mono, at 100 % it is stereo.

VARIATION

Clicking this button generates a new version of the same reverb program using altered reflection patterns. This is helpful if some sounds are causing odd ringing or undesirable results. Creating a new variation often solves these issues. There are 1000 possible variations.

HOLD

Activating this button freezes the reverb buffer in an infinite loop. You can create some interesting pad sounds using this feature.

Damping

LOW FREQ

Determines the frequency below which low-frequency damping occurs.

HIGH FREQ

Determines the frequency above which high-frequency damping occurs.

LOW LEVEL

Affects the decay time of the low frequencies. Normal room reverb decays quicker in the high- and low-frequency range than in the mid-range. Lowering the level percentage causes low frequencies to decay quicker. Values above 100 % cause low frequencies to decay more slowly than the mid-range frequencies.

HIGH LEVEL

Affects the decay time of the high frequencies. Normal room reverb decays quicker in the high- and low-frequency range than in the mid-range. Lowering the level percentage causes high frequencies to decay quicker. Values above 100 % cause high frequencies to decay more slowly than the mid-range frequencies.

Envelope

AMOUNT

Determines how much the envelope attack and release controls affect the reverb itself. Lower values have a more subtle effect while higher values lead to a more drastic sound.

ATTACK

The envelope settings in RoomWorks control how the reverb follows the dynamics of the input signal in a fashion similar to a noise gate or downward expander. Attack determines how long it takes for the reverb to reach full volume after a signal peak (in milliseconds). This is similar to a pre-delay, but the reverb is ramping up instead of starting all at once.

RELEASE

Determines how long after a signal peak the reverb can be heard before being cut off, similar to a release time of a gate.

OUTPUT

MIX

Sets the level balance between the dry signal and the wet signal. If RoomWorks is used as an insert effect for an FX channel, you most likely want to set this to 100% or use the [Wet only] button.

WET ONLY

This button deactivates the [Mix] parameter, setting the effect to 100% wet or affected signal.

EFFICIENCY

Determines how much processing power is used for RoomWorks. The lower the value, the more CPU resources are used, and the higher the quality of the reverb. Impressive effects can be created with very high [Efficiency] settings (above 90 %).

EXPORT

Determines if during audio export RoomWorks uses the maximum CPU power for the highest quality reverb. During export, you may want to keep a higher efficiency setting to achieve a specific effect. If you want the highest quality reverb during export, make sure this button is activated.

Output Meter

Shows the level of the output signal.

Roomworks SE

RoomWorks SE is a smaller version of the RoomWorks plug-in. RoomWorks SE delivers high quality reverberation, but has fewer parameters and is less CPU demanding than the full version.



PRE-DELAY

Determines how much time passes before the reverb is applied. This allows you to simulate larger rooms by increasing the time it takes for the first reflections to reach the listener.

REVERB TIME

Allows you to set the reverb time in seconds.

DIFFUSION

Affects the character of the reverb tail. Higher values lead to more diffusion and a smoother sound, while lower values lead to a clearer sound.

LOW LEVEL

Affects the decay time of the low frequencies. Normal room reverb decays quicker in the high- and low-frequency range than in the mid-range. Lowering the level percentage causes low frequencies to decay quicker. Values above 100 % cause low frequencies to decay more slowly than the mid-range frequencies.

HIGH LEVEL

Affects the decay time of the high frequencies. Normal room reverb decays quicker in the high- and low-frequency range than in the mid-range. Lowering the level percentage causes high frequencies to decay quicker. Values above 100 % cause high frequencies to decay more slowly than the mid-range frequencies.

MIX

Sets the level balance between the dry signal and the wet signal.

Spatial + Panner

StereoEnhancer

StereoEnhancer expands the stereo width of stereo audio material. It cannot be used with mono audio.

NOTE

This plug-in works only on stereo tracks.



Delay

Increases the amount of differences between the left and right channels to further increase the stereo effect.

Width

Controls the width or depth of the stereo enhancement. Turn clockwise to increase the enhancement.

Mono

Switches the output to mono, to check for unwanted coloring of the sound, which can occur when enhancing the stereo image.

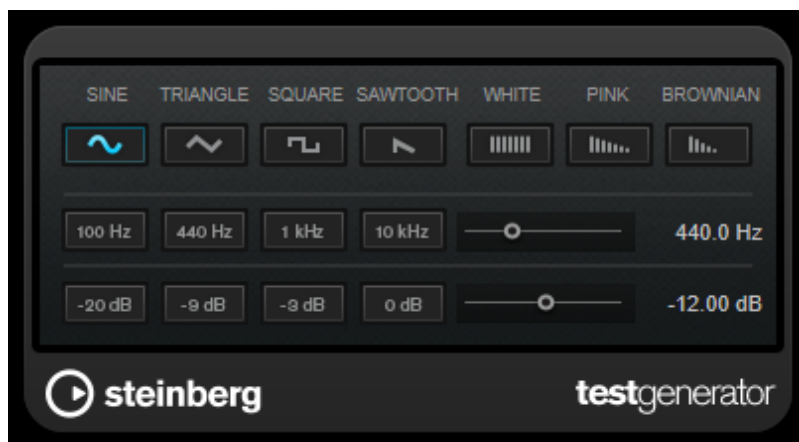
Color

Generates additional differences between the channels to increase the stereo enhancement.

Tools Plug-ins

Test Generator

This utility plug-in allows you to generate an audio signal.



The resulting file can then be used for a number of purposes:

- Testing the specifications of audio equipment
- Measurements of various kinds, such as calibrating tape recorders
- Testing signal processing methods
- Educational purposes

The TestGenerator is based on a waveform generator that can generate a number of basic waveforms such as sine and saw as well as various types of noise. Furthermore, you can set the frequency and amplitude of the generated signal.

Waveform and Noise Section

Allows you to set the basis for the signal generated by the waveform generator. You can choose between four basic waveforms (sine, triangle, square, and sawtooth) and three types of noise (white, pink, and brownian).

Frequency Section

Allows you to set the frequency of the generated signal. You can set the frequency either in Hz or as a note value. If you enter a note value, the frequency is automatically changed to Hz. For example, a note value of A3 sets the frequency to 440 Hz. When you enter a note value, you can also enter a cent offset. For example, enter "A5 -23" or "C4 +49."

NOTE

Make sure that you enter a space between the note value and the cent offset. Only in this case, the cent offsets are taken into account.

Gain Section

Allows you to set the amplitude of the signal. The higher the value, the stronger the signal. You can select one of the preset values, or use the slider to set a value between OFF and 0 dB.

Delay Plug-ins

AnalogDelay

This delay effect is based on the Yamaha E1010 analog delay released in the 1970s, with a few refinements to bring it up to date for modern applications. It delivers the same deep, organic echo sound that was a characteristic of the BBD analog delay devices used in the original E1010. Modulation can be added for a thick chorus sound. And, taking versatility to the next level, it is possible to select BBD sounds that were not available in the original E1010 so characters ranging from clear and precise to warm are available for use.



① DRIVE METER

This meter indicates the amount of drive by input level. Because Analog Delay features BBD and dynamics circuits, higher input level would bring the delayed sound up to maximum and increase the drive amount.

② DELAY TIME INDICATOR

Indicates the current delay time and converted BPM value. The tap tempo function can be used, and pressing the Sync button changes the delay time and BPM display to match the tap tempo.

③ BBD TYPE SWITCHES

Adjust the characteristics of the delay sound. The characteristics become stronger in the order of A to E.

A: Clear sound as heard from a vintage digital delay

B to D: Natural sound, equivalent to the E1010

E: Warm sound similar to an instrumental stomp box

④ TIME RANGE SWITCHES

Specify the delay time range controlled via the DELAY knob.

⑤ INPUT KNOB

Adjusts the input gain. As the input increases, the delay's drive amount also increases. If the input level is lower, a cleaner sound is produced. If the input level is higher, a deeper delay sound is produced.

⑥ BASS KNOB

Adjusts the low frequency range level at the input stage.

⑦ TREBLE KNOB

Adjusts the high frequency range level at the input stage.

⑧ DELAY KNOB

Adjusts the delay time.

⑨ **FEEDBACK KNOB**

Adjusts the feedback amount of the delay signal.

⑩ **FREQUENCY KNOB**

Adjusts the frequency of modulation.

⑪ **DEPTH KNOB**

Adjusts the depth of modulation.

⑫ **MIX KNOB**

Adjusts the mix balance between dry and delay sounds.

Dynamics Plug-ins

Buss Comp 369

Buss Comp 369 emulates the bus compressor that has been used as the standard in recording studios and by broadcasters since the 1980s. In contrast to Compressor 276, which excels at creating aggressive compression sounds, it features smooth, natural compression that does not impair the expressiveness of the source. It has both a built-in compressor and limiter, which can be used separately or in combination depending on the application. In addition, the rich harmonics of the input/output circuit audio transformer and discrete class A amplifier add depth and cohesiveness to the sound quality. This plug-in operates in mono or stereo mode, depending on the channel configuration of the rack.

In stereo mode

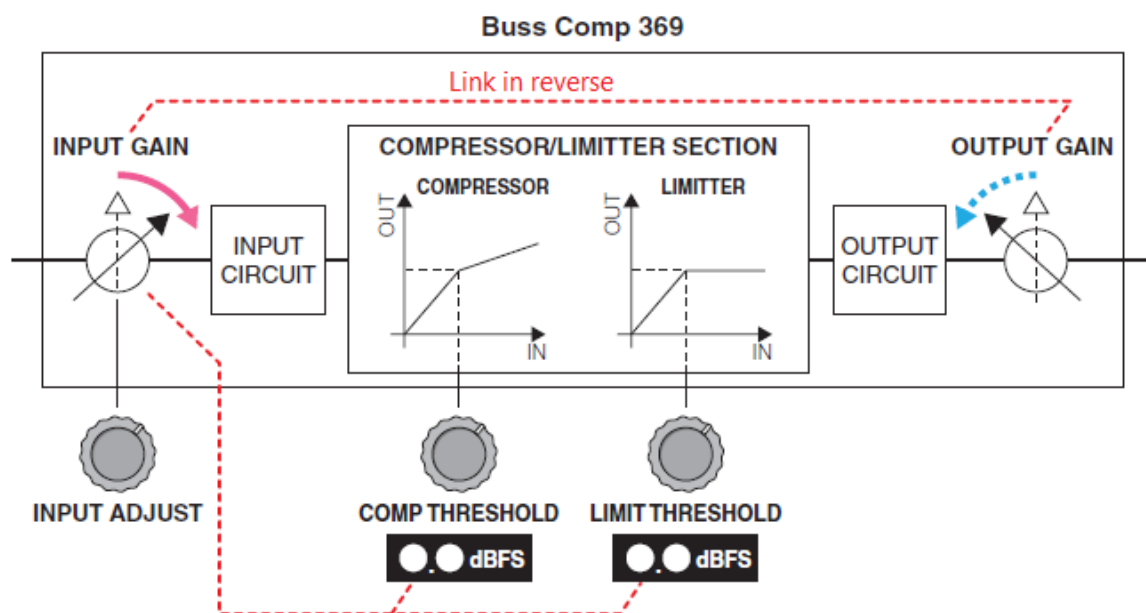


In mono mode



[INPUT ADJUST] Knob

Adjusts the input gain. However, the output gain changes in the opposite direction so that the passing volume (volume when [COMP IN] and [LIMIT IN] are off) does not change. For example, when [INPUT ADJUST] is +5 dB, the input gain is +5 dB and the output gain is -5 dB.

How INPUT ADJUST Works

Since the threshold changes substantially, the displayed value changes in tandem. However, the internal state of the COMPRESSOR/LIMITER SECTION does not change.

[LINK] Switch (in stereo mode)

Activates/Deactivates the stereo link.

COMP [IN] Switch

Activates/Deactivates the compressor.

COMP [THRESHOLD] Knob

Sets the threshold for the compressor. However, the value changes in conjunction with [INPUT ADJUST].

COMP [RECOVERY] Knob

Sets the release time of the compressor. The release time of a1 (auto 1) and a2 (auto 2) changes automatically.

a1: Changes automatically between 100 ms and 2 sec.

a2: Changes automatically between 50 ms and 5 sec.

COMP [GAIN] Knob

Sets the make-up gain of the compressor. It works in front of the limiter.

COMP [RATIO] Knob

Adjust the ratio of the compressor.

[METER] Select Switch (in stereo mode)

Selects the signal source for the meters. You can select from three options: IN (input level), GR (gain reduction), and OUT (output level). When you switch the signal source, the meter design (VU/GR/VU) also changes.

[VU] Select Switch (in mono mode)

Selects the signal source for the VU meter. You can select either input level or output level.

LIMIT [IN] Switch

Activates/Deactivates the limiter.

LIMIT [ATTACK] Switch

Sets the attack time of the limiter.

FAST: 2 ms

SLOW: 4 ms

LIMIT [THRESHOLD] Knob

Sets the threshold for the limiter. However, the value changes in conjunction with [INPUT ADJUST].

LIMIT [RECOVERY] Knob

Sets the release time of the limiter. The release time of a1 (auto 1) and a2 (auto 2) changes automatically.

a1: Changes automatically between 100 ms and 2 sec.

a2: Changes automatically between 50 ms and 5 sec.

Compressor 260

Compressor 260 emulates the characteristics of a compressor and limiter that were widely used in the mid-1970s.

This plug-in operates in mono or stereo mode, depending on the channel configuration of the rack.



[THRESHOLD] Knob

Sets the level at which compression is applied. Only signal levels that are above the set threshold are processed.

[KNEE] Switch

Sets how fast compression is applied. When set to [SOFT], the compression ratio increases slowly as the level increases, and when set to [HARD] it is compressed immediately.

[ATTACK] Knob

Sets the time it takes for Compressor 260 to respond to a signal level that is above the threshold. The longer the [ATTACK] time, the greater the amount of signal (attack) that passes unprocessed.

[RELEASE] Knob

Sets the time it takes for the gain to return to the original level when the signal drops below the threshold.

[RATIO] Knob

Adjust the ratio. Turn the control to the right to increase the compression ratio. If you turn it completely to the right, the plug-in will operate as a limiter.

[OUTPUT] Knob

Adjusts the output gain.

[GR] Meter

Displays the amount of gain reduction applied by the plug-in.

[OUT] Meter

Displays the output level of the effect.

Compressor 276

Compressor 276 emulates the characteristics of an analog compressor that is a staple in recording studios. You can obtain thick, strong sounds that are suitable for drums and bass parts.

This plug-in operates in mono or stereo mode, depending on the channel configuration of the rack.



[INPUT] Knob

Adjusts the input level.

[RATIO] Knob

Adjust the ratio of the compressor.

[ATTACK] Knob

Adjusts the attack time of the compressor.

[OUTPUT] Knob

Adjusts the output gain.

[AUTO MAKEUP] Switch

When activated, it automatically compensates for the drop in output gain when the compressor is applied.

[INTERNAL SC HPF] Switch

When activated, less compression is applied to the low frequencies so that the low frequencies are emphasized.

[RELEASE] Knob

Adjusts the release time of the compressor.

[VU] Meter

Displays the signal that is selected with the [GR]/[-10]/[-20]/[OFF] switch.

[GR]/[-10]/[-20]/[OFF] Switch

Selects the signal that is monitored by the VU meter.

GR: Displays the amount of gain reduction.

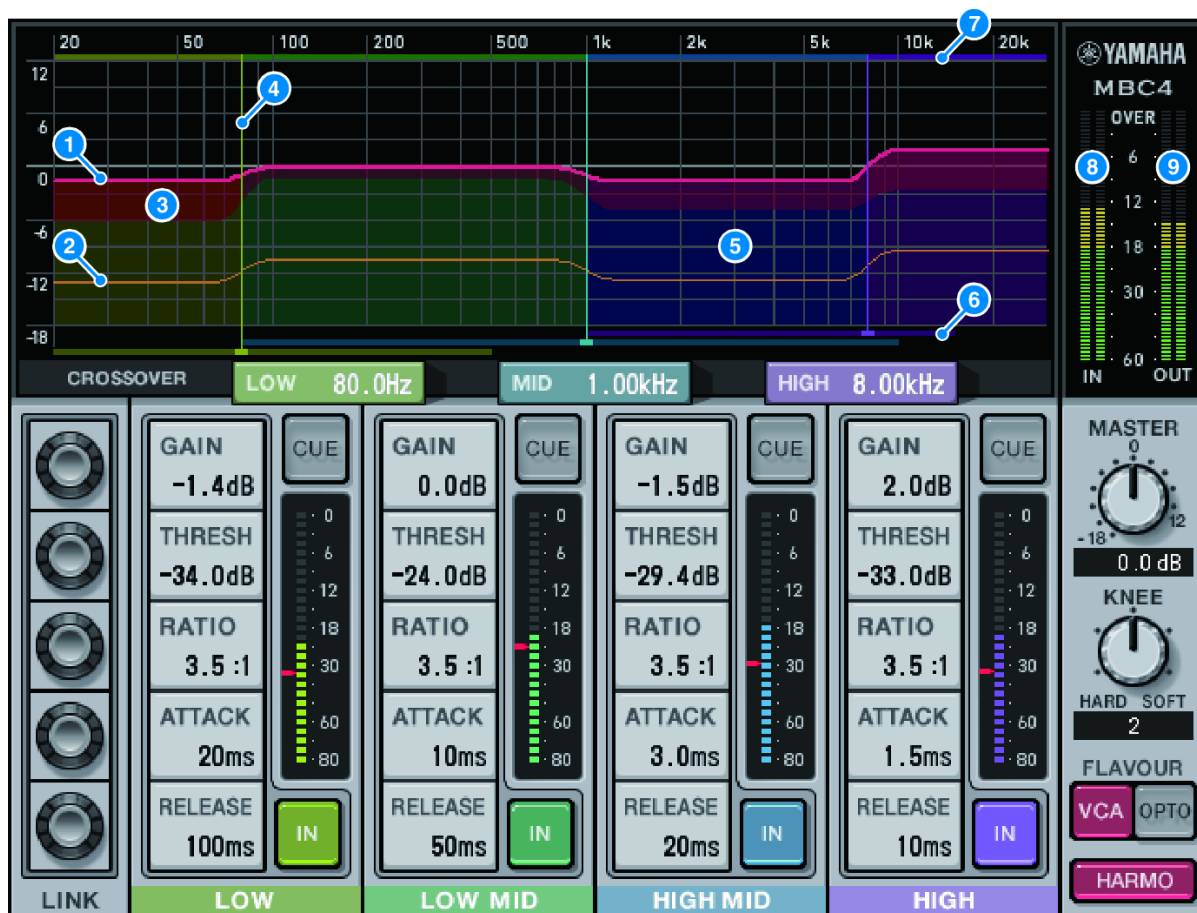
-10: Displays the signal with 0VU as -10 dBu.

-20: Displays the signal with 0VU as -20 dBu.

OFF: Deactivates the VU meter.

MBC4

This high-quality four-band compressor utilizes VCM technology and features a GUI that offers outstanding operability and visibility. All advantages of the musical behavior of analog compressors have been built into the gain reduction circuitry of the MBC4, allowing smooth control of dynamics while retaining the overall image of the original sound. The sound image can also be visually manipulated via a graphical display.



① GAIN LINE

This red horizontal line indicates the GAIN parameter values for the corresponding band.

② MAXIMUM GR GUIDELINE

This orange horizontal line indicates a guideline for the maximum gain reduction. This amount is approximate, and does not necessarily show a precise maximum gain reduction amount.

③ GR ZONE

This red area changes according to the gain reduction amount for the corresponding band.

④ CROSSOVER LINE

This vertical line indicates the crossover point for the corresponding band.

⑤ CROSSOVER ZONE

This band-color area changes according to the CROSSOVER parameter value for the corresponding band.

⑥ CROSSOVER RANGE

This band-color horizontal bar graph indicates a variable range of the CROSSOVER parameter value for the corresponding band.

⑦ CROSSOVER BAND WIDTH

This band-color horizontal bar graph changes according to the CROSSOVER parameter value for the

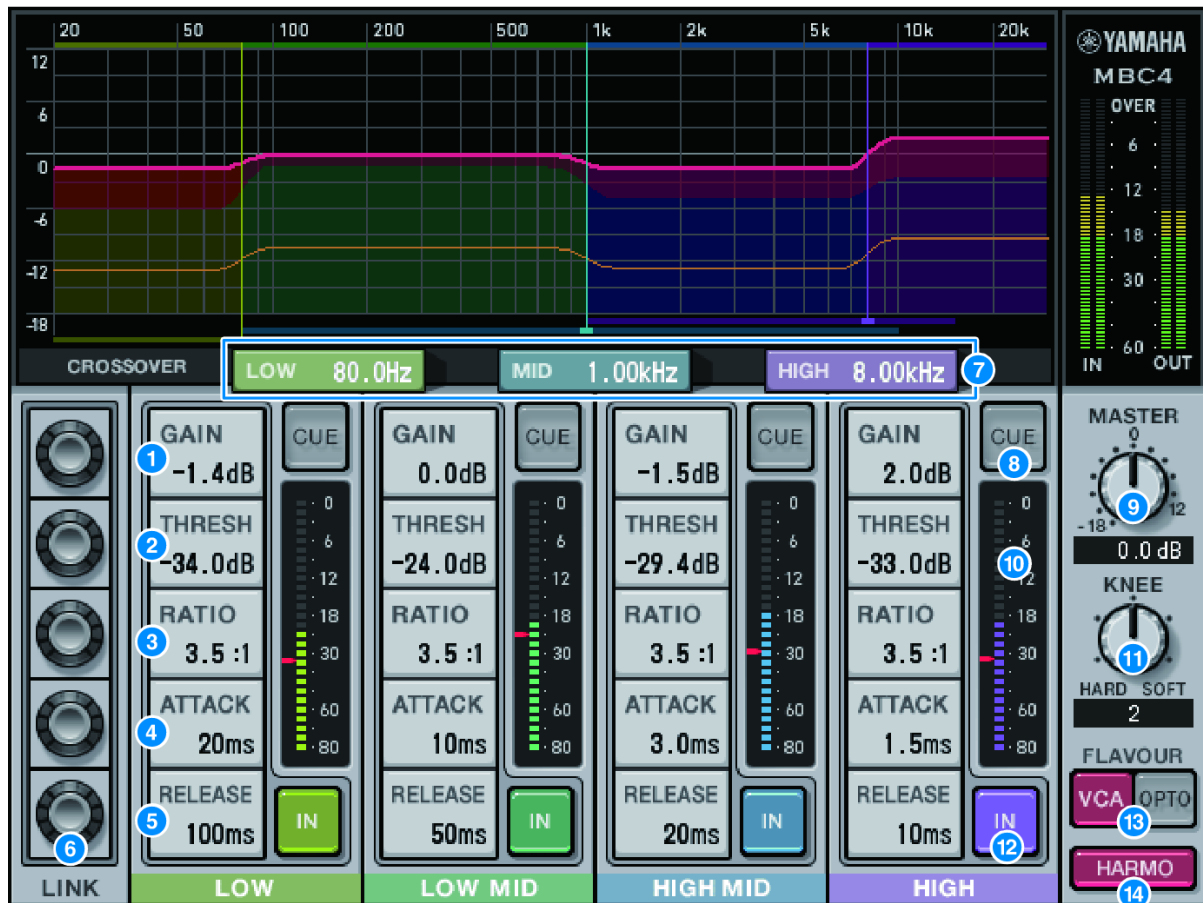
corresponding band.

⑧ INPUT METER

Shows the level of the input signal. One meter is shown in the case of DUAL, and two meters in the case of STEREO.

⑨ OUTPUT METER

Shows the level of the output signal. One meter is shown in the case of DUAL, and two meters in the case of STEREO.



⑩ GAIN

Adjusts the output gain for the corresponding band.

⑪ THRESHOLD

Adjusts the level at which the compressor begins to apply for the corresponding band. The value will be also indicated by a red marker on the INPUT meter.

⑫ RATIO

Adjusts the compression ratio for the corresponding band.

⑬ ATTACK

Adjusts the attack time at which the compressor begins to apply for the corresponding band.

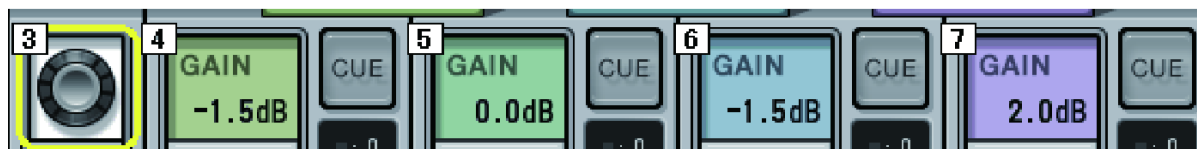
⑭ RELEASE

Adjusts the release time of the compressor for the corresponding band.

⑮ LINK KNOB

If you select the parameter's LINK knob, the display color will change as shown below. The corresponding parameters for four bands will link together, and the values will change collectively. Note that if the parameter

value of one of the bands reaches the maximum or minimum, the link operation will stop. Parameters that can be linked are displayed in the corresponding band color.



⑩ CROSSOVER

Adjusts the crossover frequencies used to divide the signal into bands.

⑪ CUE BUTTON

Press this button to cue-monitor the output of the corresponding band. If the IN button is off, a signal without compressor effects will be monitored. You can turn on the CUE buttons for multiple bands.

⑫ MASTER GAIN KNOB

Adjusts the gain of the final output signal.

⑬ INPUT METER

Displays the level of the audio signal input to each band.

⑭ KNEE KNOB

Adjusts the knee of the compressor.

⑮ IN BUTTON

Switches the compressor on/off for the corresponding band. Even if this button is off, you will be able to adjust the corresponding band parameters.

⑯ FLAVOUR BUTTON

Switches the type of the compressor between VCA (accurate effects) and OPTO (natural effects).

⑰ HARMONICS BUTTON

Switches on/off musical harmonics that emulate analog circuitry.

EQ Plug-ins

Dynamic EQ

Dynamic EQ is a newly developed equalizer, and is not an emulation of a particular model. Since a filter that extracts the same band as the EQ is set on the side-chain, the EQ gain changes dynamically only when the frequency band with an input signal becomes large or small. You can apply EQ to a specific band, like a compressor or an expander. For example, if you use it as a de-esser for vocals, the EQ will be applied to that band only when the sibilants and high-frequency noises are at a jarring level. The result is a natural sound that does not compromise the original sound quality. It is also equipped with two full-band Dynamic EQs that can be used in various ways.

This plug-in operates in mono or stereo mode, depending on the channel configuration of the rack.



[BAND ON/OFF] Button




Activates/Deactivates the frequency band.




SIDECHAIN [LISTEN] Button

When activated, the side-chain signal linked to the dynamics is output. At this time, the graph displays the characteristics of the side-chain filter.

[FILTER TYPE] Buttons

Switches between the main bus equalizer and side-chain filter types. The main EQ and side-chain filter work together as shown below.

FILTER TYPE			
	(Low Shelf)	(Bell)	(Hi Shelf)
Main EQ	Low Shelf	Bell	Hi Shelf

FILTER TYPE	 (Low Shelf)	 (Bell)	 (Hi Shelf)
Side-Chain Filter	LPF	BPF	HPF

[FREQUENCY] Knob

Sets the frequency to be operated by the equalizer and side-chain filter.

[Q] Knob

Sets the Q (sharpness) of the equalizer and side-chain filter.

The more you turn the knob to the right, the band covered by the equalizer and side-chain filter becomes wider.

[THRESHOLD] Knob

Sets the threshold value at which the processing effect starts to be applied.

[RATIO] Knob

Sets the ratio of boost/cut for the input signal.

Turn the knob to the right to boost, and to the left to cut. The maximum effect is achieved when the knob is turned completely in either direction.

[ATTACK/RELEASE] Buttons

Selects from three types of attack time/release time when compression or boost is applied.

FAST has a faster attack and faster release, SLOW has a faster attack and slower release, and AUTO automatically adjusts the attack/release according to the frequency band.

[MODE] Button

Sets whether the side-chain signal works when it exceeds the threshold value (ABOVE) or when it falls below it (BELOW).

EQ GAIN Meter

Displays the dynamically changing EQ gain.

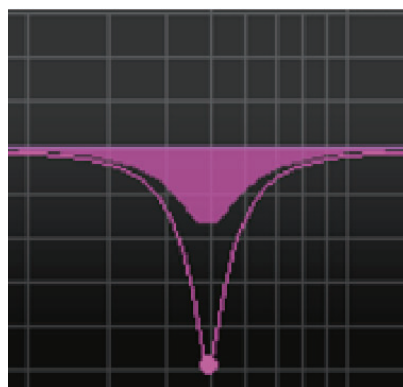
THRESHOLD Meter

Displays the level of the side-chain signal relative to the threshold level.

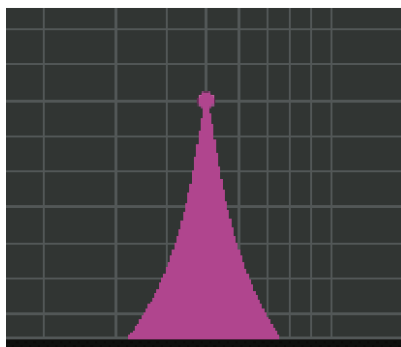
Graphical Display

Displays the characteristics of the equalizer.

Usually, you will see a reference EQ graph that displays frequency and effectiveness, and a dynamic EQ graph that displays the characteristics of dynamically changing EQs.



If SIDECHAIN [LISTEN] is activated, the characteristics of the side-chain filter are displayed.



EQ-1A

EQ-1A emulates a vintage EQ that is said to be representative of passive EQs. It has a unique operation style in which the low and high frequency bands are adjusted with separate boost and attenuation (cut) controls. The frequency response, which is completely different from other commonly used EQs, has a character that is unique to this model. In addition, the texture produced by the input/output circuits and vacuum tubes is very musical, for a well-balanced sound.

This plug-in operates in mono or stereo mode, depending on the channel configuration of the rack.



[IN] Switch

Activates/Deactivates the processor.

When deactivated, the filter section is bypassed, but the signal still passes through the input/output transformer and amplifier circuitry.

[LOW FREQUENCY] Knob

Adjusts the frequency of the low frequency filter.

(LOW) [BOOST] Knob

Adjusts the amount of boost in the frequency band set with the [LOW FREQUENCY] knob.

(LOW) [ATTEN] Knob

Adjusts the amount of attenuation in the frequency band set with the [LOW FREQUENCY] knob.

[BAND WIDTH] Knob

Sets the bandwidth to operate with the high frequency filter.

The more you turn it to the right (Broad) side, the wider it becomes and at the same time the peak level drops. It affects only the characteristics on the boost side.

[HIGH FREQUENCY] Knob

Adjusts the frequency of the high frequency filter. It affects only the characteristics on the boost side.

(HIGH) [BOOST] Knob

Adjusts the amount of boost in the frequency band set with the [HIGH FREQUENCY] knob.

(HIGH) [ATTEN] Knob

Adjusts the amount of attenuation in the frequency band set with the [ATTEN SEL] knob.

(HIGH) [ATTEN SEL] Knob

Switches the frequency band that is attenuated with the [ATTEN] knob.

Equalizer 601

Equalizer 601 emulates the characteristics of an analog equalizer from the 1970s. You can get a sense of drive by reproducing the distortion that is unique to analog circuits.

This plug-in operates in mono or stereo mode, depending on the channel configuration of the rack.



Frequency Response Curve Graph

The frequency response curve for all bands and the frequency response curve for each band are displayed. You can drag your mouse on the graph to change the frequency and gain of the filter.

[+]/[-] Buttons

Enlarge or reduces the vertical axis of the graph display.

[TYPE] Switch

Selects one of the two equalizer types with different effects.

DRIVE emulates changes in the frequency response of analog circuits, with a drive-like sound that emphasizes analog characteristics by adding distortion.

CLEAN emulates changes in the frequency response of analog circuits with clear, distortion-free sound that digital is good at.

[INPUT] Knob

Adjusts the input level.

Level Meter

Displays the output level of the effect.

[OUTPUT] Knob

Adjusts the output gain.

[FLAT] Button

Resets the gain of all bands to 0 dB.

[Q/TYPER] Knob

Controls the shape of the frequency response curve of the filter in each band.

For the four bands in the MID frequencies, you can set the sharpness (Q) of the frequency response curve. For the LO band and the HI band, you can select the filter type from four options.

[F] (Frequency) Knob

Center frequency of the filter.

[G] (Gain) Knob

Gain value of the filter.

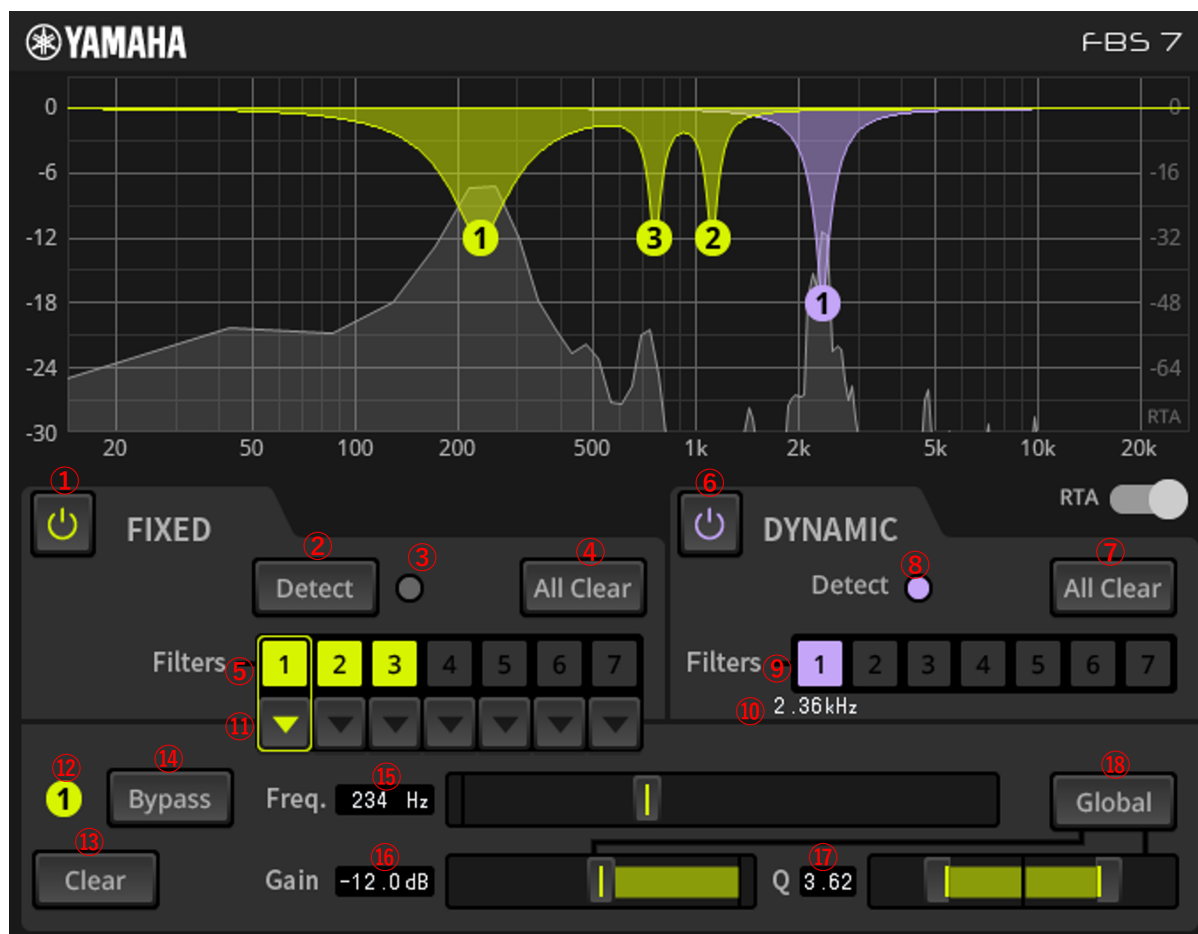
[ON] Button

Activates/Deactivates the filter. The filters for bands that are deactivated are disabled.

Filter

FBS 7

FBS 7 is a notch filter-type feedback canceller with an automatic detection function. This plug-in provides two modes: FIXED and DYNAMIC. Both modes have a seven-band filter and automatic feedback detection function. FIXED mode is used to detect feedback points and create a margin against feedback before the actual performance. DYNAMIC mode is used to immediately respond to feedback that occurs during a performance. These two modes can be used in combination to effectively suppress feedback. The RTA display and filter graph can also be used to easily check feedback conditions.



① Fixed On Button

Toggles between enabling and disabling the measurement results for Fixed mode.

② Fixed Detect Button

Toggles between running and stopping a measurement. Measurements are not stopped automatically. Dynamic filters are disabled while measurements are made. Filters cannot be edited while measurements are being made. For more information about how to make measurements, refer to “Using the Fixed Method to Make FBS Measurements.”

③ Fixed Detect Display

Comes on when feedback is detected in Fixed mode.

④ Fixed All Clear Button

Clears all filter settings in Fixed mode.

⑤ Fixed Filter State Display

Indicates the status of each filter in Fixed mode.

Free (gray): No applicable filter is in use.

Bypass (dark yellow): An applicable filter is in use and set to Bypass.

Active (yellow): An applicable filter is in use and currently enabled.

⑥ Dynamic On Button

Toggles between enabling and disabling the Dynamic function. When turned on, FBS starts operating if feedback is not currently detected in Fixed mode.

⑦ Dynamic Detect Display

Comes on when feedback is detected in Dynamic mode.

⑧ Dynamic All Clear

Clears all filter settings in Dynamic mode.

⑨ Dynamic Filter State Display

Indicates the status of each filter in Dynamic mode.

Free (gray): No applicable filter is in use.

On (purple): An applicable filter is in use.

⑩ Dynamic Filter Frequency Display

Indicates the detected frequency, when the mouse is placed over a filter number.

⑪ Filter Sel Button

Selects a filter.

⑫ Selected Filter Number Display

Indicates the number of the selected filter.

⑬ Clear Button

Clears the settings for the selected filter.

⑭ Bypass Button

Toggles between enabling and disabling Bypass for the selected filter.

⑮ Frequency Slider

Sets the center frequency for the selected filter.

⑯ Gain Slider

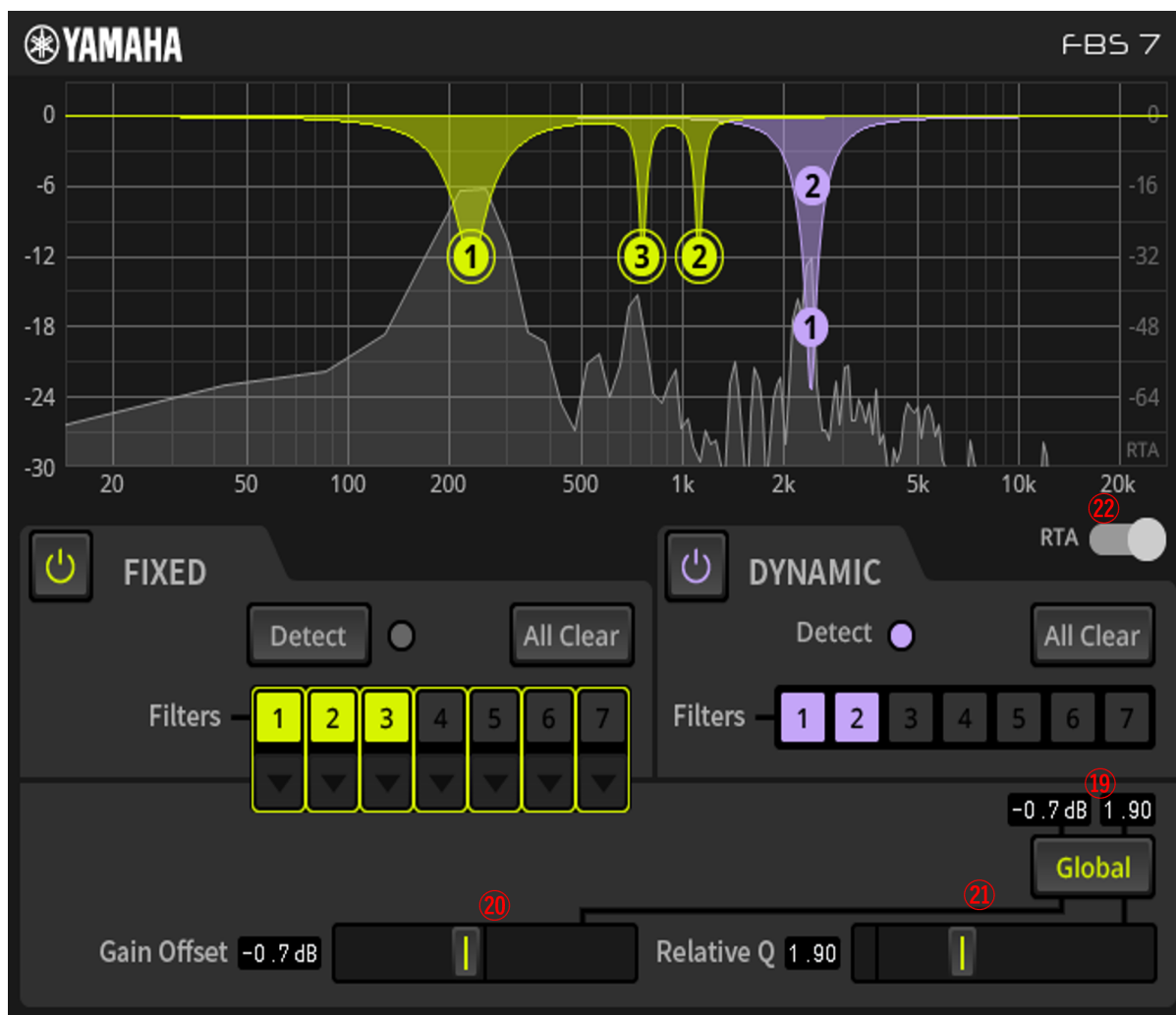
Sets the gain for the selected filter.

⑰ Q Slider

Sets the Q for the selected filter.

⑱ Global Button

Toggles between displaying the Global Settings screen and displaying the Edit screen for each filter.



19 Global Value Display

Displays parameter values when Global Q and Global Gain are anything other than 1.0 or 0.0. These values cannot be edited.

20 Global Relative Q Slider

Sets the Global Q to be applied to all Fixed filters. The Q value that is actually applied to each filter is the Q set for each filter multiplied by the Global Q. (If Global Q is 1.5, the actual value of Q for a filter with Q set to 20.0 is 30.0).

21 Global Gain Offset Slider

Sets the Global Gain to be applied to all Fixed filters. The Gain value that is actually applied to each filter is the Gain set for each filter added to the Global Gain.

22 Graph Button

Toggles between displaying and hiding the frequency response for the output signal.

Mastering Plug-ins

Vintage Open Deck

Vintage Open Deck emulates the analog circuitry and tape characteristics of a legendary open reel tape recorder. This plug-in consists of a recording deck and a reproduction deck, and there are four variations available for each.

Vintage Open Deck is a stereo effect. When it is added to a mono rack, only the left channel of the plug-in is used.



Selecting a Deck

The pop-up menus at the top of the plug-in panel allow you to specify the type of recording tape deck or reproduction tape deck to use. You can use the switch between the pop-up menus to select the source of the signal to monitor with the deck's peak and VU meters.

Recording Deck Controls

The recording deck controls are on the left side of the plug-in.

RECORD

Adjusts the input level of the recording deck. Increasing the input level will increase the tape compression, narrowing the dynamic range and distorting the sound.

AUTO MAKEUP

When this is on, the [REPRODUCE] control settings are set to match the [RECORD] controls. This ensures that the overall output level is the same when you increase the input level of the recording deck. Therefore, you can control the distortion level without changing the output level.

ADJUST-HIGH

Adjusts the high-frequency gain of the recording deck.

BIAS

Adjusts the bias of the recording deck. This allows you to control the distortion level. Turning this control to the [LESS] side raises the signal level and boosts the high frequencies. Turning it to the [OVER] side narrows the dynamic range and causes high-frequency saturation.

Reproduction Deck Controls

The reproduction deck controls are on the right side of the plug-in panel.

REPRODUCE

Adjusts the output level of the reproduction deck.

ADJUST-HIGH

Adjusts the high-frequency gain of the reproduction deck.

ADJUST-LOW

Adjusts the low-frequency gain of the reproduction deck.

Deck Meters

Depending on the setting of the [RECORD/REPRO] switch, the meters in the center of the plug-in panel show the signal input to the recording deck or the signal output from the reproduction deck. If this plug-in is added to a mono rack, only the top meter is used.

Adjusting the VU Meter Level

You can adjust the level assigned to 0VU on the meter. This is useful for more accurately monitoring very low level signals. Use the following procedure.

- . Turn on the [VU ADJUST] button at the bottom left of the plug-in panel.
- . In the meter section, use the [VU ADJUST] control to set the 0VU level.

Tape Parameters

The tape parameter controls are at the bottom of the plug-in.

SPEED

Allows you to set the tape speed. This parameter changes the sound characteristics of the tape head saturation. 30 ips (inches per second) provides better high-frequency response compared to 15 ips. 15 ips provides a rich low-frequency response.

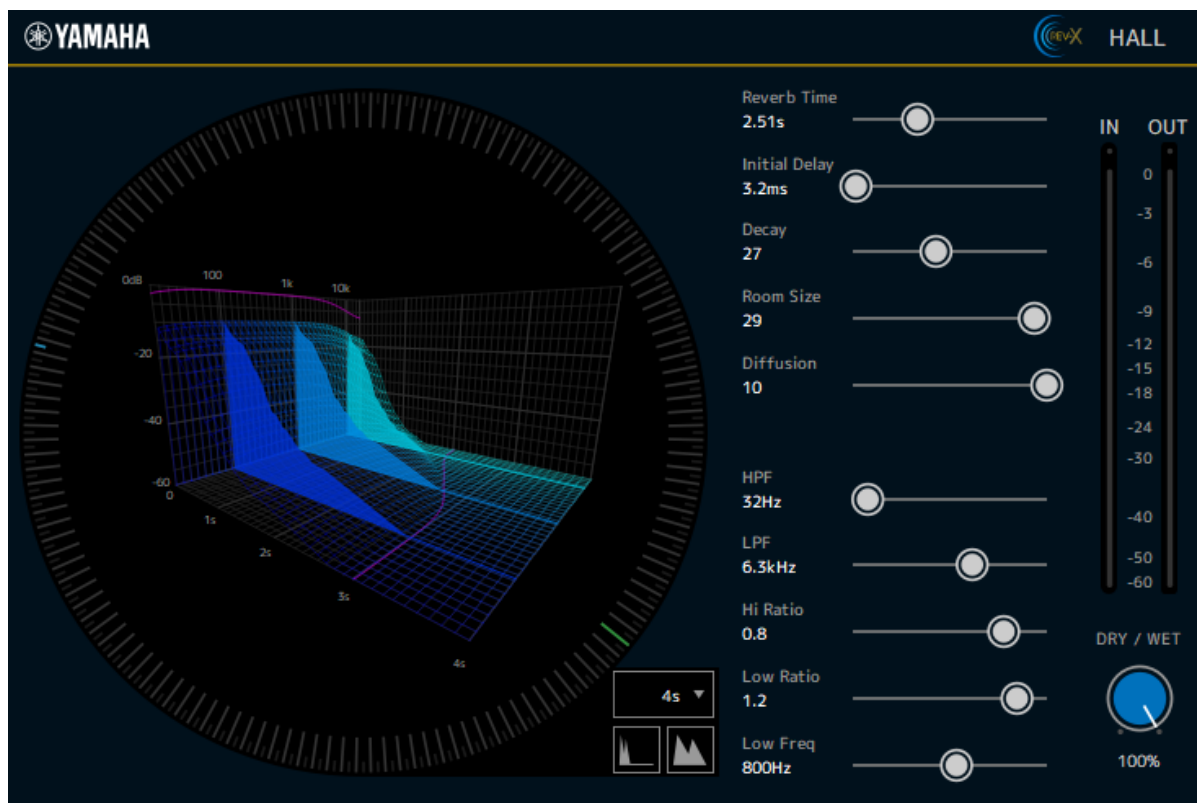
TAPE KIND

This allows you to change the sound characteristics of the tape saturation.

Reverb Plug-ins

REV-X

REV-X is a reverb algorithm with features such as sound quality with high-density and rich reverberation, smooth attenuation, and breadth and depth that make the best use of the original sound. You can select from three types of reverb according to the sound field and purpose: REV-X HALL, REV-X ROOM, and REV-X PLATE.



EFFECT TYPE

Selects the effect type.

Reverb Time

This is the time required for the reverberation to decay and disappear. The higher the value, the longer the reverberation will last.

Initial Delay

This is the delay from the input of the original sound to the start of reverberation. The higher the value, the later the occurrence of reverberation.

Decay

This is the envelope shape of the reverberation. The characteristics of the reverberation will change depending on this value.

Room Size

This is the size of the room. The higher the value, the larger the simulated space. This value is linked with [Reverb Time]. Changing this value will also change [Reverb Time].

Diffusion

This is the density and spread of reverberation. The higher the value, the higher the density and the stronger the sense of spaciousness.

HPF

This is a filter that cuts the low-frequency components of reverberation. Components that are below the frequency specified by this value are cut. This filter has no effect on the original sound.

LPF

This is a filter that cuts the high-frequency components of reverberation. Components that are above the frequency specified by this value are cut. This filter has no effect on the original sound.

Hi Ratio

This is the length of the high-frequency reverberation. The high-frequency reverberation time is specified as a ratio with [Reverb Time].

Low Ratio

This is the length of the low-frequency reverberation. The low-frequency reverberation time is specified as a ratio with [Reverb Time].

Low Freq

This is the reference frequency for [Low Ratio]. Frequency bands below this value are affected by [Low Ratio].

Filter Frequency Response Curve

The curve changes depending on the HPF and LPF values.

Reverberation Image

This is an image of the reverberation of the high band (10 kHz), mid band (1 kHz), and low band (100 Hz). The image changes depending on the value of each parameter. The vertical axis represents the level, the horizontal axis represents the reverberation time, and the shape represents the envelope.

Reverberation Time Curve

This curve displays the reverberation time of the high band (10 kHz), mid band (1 kHz), and low band (100 Hz). The curve changes depending on the values for [Reverb Time], [Hi Ratio], and [Low Ratio].

Zoom

Displays the length of time allocated to the time axis (horizontal axis of the graph) in seconds.

Zoom Out Button

Click this button to increase the number of seconds allocated to the time axis (horizontal axis of the graph). As a result, the width of the graph is reduced.

Zoom In Button

Click this button to decrease the number of seconds allocated to the time axis (horizontal axis of the graph). As a result, the width of the graph is increased.

MIX Slider

Adjusts the balance between the original sound and the effect. If this value is 0%, only the original sound is output. If it is 100%, only the effect is output.

