Plug-in Reference

The Steinberg Documentation Team: Cristina Bachmann, Martina Becker, Heiko Bischoff, Lillie Harris, Christina Kaboth, Insa Mingers, Matthias Obrecht, Sabine Pfeifer

Translation: Ability InterBusiness Solutions (AIBS), Moon Chen, Jérémie Dal Santo, Rosa Freitag, GiEmme Solutions, Josep Llodra Grimalt, Vadim Kupriianov, Roland Münchow, Boris Rogowski, Sergey Tamarovsky

This document provides improved access for people who are blind or have low vision. Please note that due to the complexity and number of images in this document, it is not possible to include text descriptions of images.

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Table of Contents

Included VST Audio Effect 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.

AutoPan

This auto-pan effect provides several parameters to modulate the left/right stereo position. You can use presets or create individual curves for the modulation waveform. **AutoPan** also allows for chopping effects by linking the modulation of left and right channel.

Waveform display

Shows the shape of the modulation waveform and allows you to manually adjust it. To draw an individual curve, click a node and move the mouse. To draw a straight line, **Shift**-click a node and move the mouse.

Waveform preset buttons

Allow you to select presets for the modulation waveform.

- **Sine** creates a smooth sweep.
- **Triangle** creates a ramp, that is, a linear movement from full right to full left and back.
- **Square** creates an instant jump to full right, then to full left, and then back to center.
- **Random One Shot** creates a random curve. Click this button again to create a new random curve.
- **Random Continuous** automatically creates a new random curve after each period.

Phase

Sets the offset for the starting point of the curve. If multiple **AutoPan** plug-ins are used on different tracks, for example, different offset settings for each track allow for a more organic overall sound.

Factor

If **Sync** is activated, this parameter multiplies the sync rate by the selected factor. This allows you to create very slow movements in panorama.

Rate

Sets the auto-pan speed and shows the movement within the panorama. If **Sync** is deactivated, the speed is set in Hertz. If **Sync** is activated, you can set the speed in tempo values.

Sync

Activates/Deactivates tempo sync.

Link

If this button is activated, the left and right channel are modulated simultaneously. This results in a chopping effect instead of auto-panning.

In this mode, **Width** sets the intensity of the volume modulation.

Width

Sets the amount of deflection to the left and right side of the stereo panorama. If **Link** is activated, this parameter sets the intensity of the volume modulation.

Smooth

Allows you to smooth the transition between individual steps of the panorama curve.

Brickwall Limiter

Brickwall Limiter ensures that the output level never exceeds a set limit.

Due to its fast attack time, **Brickwall Limiter** can reduce even short audio level peaks without creating audible artifacts. However, this plug-in creates a latency of 1 ms. **Brickwall Limiter** features separate meters for input, output, and the amount of limiting. Position this plug-in at the end of the signal chain, before dithering.

Threshold (-20 to 0 dB)

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

Release (3 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 the **Auto** button is activated, the plug-in automatically finds the best release setting for the audio material.

Link Channels

If this option is activated, the plug-in uses the input from the channel with the highest level. If the option is deactivated, each channel is analyzed separately.

Detect Intersample Clipping

If this option is activated, the plug-in uses oversampling to detect and limit signal levels between two samples to prevent distortion when converting digital signals into analog signals.

NOTE

Brickwall Limiter is designed for the reduction of occasional peaks in the signal. If the **Gain Reduction** meter indicates constant limiting, try raising the threshold or lowering the overall level of the input signal.

Chorus

This plug-in is a single-stage chorus effect. It doubles the audio that is sent into it with a slightly detuned version.

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

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

Compressor

Compressor reduces the dynamic range of the audio, making softer sounds louder or louder sounds softer, or both.

Compressor features a separate display that graphically illustrates the compressor curve that is shaped according to the **Threshold** and **Ratio** parameter settings. **Compressor** also features a **Gain Reduction** meter that shows the amount of gain reduction in dB, **Soft knee**/**Hard knee** compression modes, and a program-dependent auto feature for the **Release** parameter.

Threshold (-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.

Soft Knee

If this button is deactivated, signals above the threshold are compressed instantly according to the set ratio (hard knee). If **Soft Knee** is activated, the onset of compression is more gradual, producing a less drastic result.

High Ratio

Sets the ratio to a fixed value of 20:1.

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

Compensates for output gain loss caused by compression. If **Auto Make-Up Gain** is activated, the output is automatically adjusted for gain loss.

Dry Mix

Mixes the dry input signal to the compressed signal.

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.

Hold (0 to 5000 ms)

Sets the time the applied compression affects the signal after exceeding the threshold. Short hold times are useful for DJ-style ducking, while longer hold times are required for music ducking, for example, when working on a documentary film.

Release (10 to 1000 ms or Auto mode)

Sets the time after which the gain returns to its 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.

Analysis (Pure Peak to Pure RMS)

Determines whether the input signal is analyzed according to peak or RMS values, or a mixture of both. A value of 0 is pure peak and 100 pure RMS. **RMS** mode operates using the average power of the audio signal as a basis, whereas **Peak** mode operates more on peak levels. As a general guideline, **RMS** mode works better on material with few transients such as vocals, and **Peak** mode works better for percussive material with a lot of transient peaks.

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.

Distortion

Distortion adds crunch to your tracks.

Boost

Increases the distortion amount.

Oversampling

Activates/Deactivates oversampling. Oversampling results in less artifacts for higher distortion.

NOTE

If this parameter is activated, the effect requires more processing power.

Mix

Sets the level balance between dry and wet signal.

Tone

Changes the tonal characteristic of the output signal.

Feedback

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

Spatial

Changes the distortion characteristics of the left and right channels, thus creating a stereo effect.

Output

Sets the output level.

DJ-EQ

DJ-EQ is an easy-to-use 3-band parametric equalizer that resembles the EQs found on typical DJ mixers. This plug-in is designed for quick sound fixes.

Graphical display

Allows you to set the amount of boost or attenuation for the low, mid, or high band by dragging.

- To set the low, mid, and high frequency gain, click and drag the corresponding band handle.
- To fine-adjust the gain, press **Shift** and drag.

- To set a parameter to zero, press **Ctrl/Cmd** and click it.

Low Frequency Gain/Mid Frequency Gain/High Frequency Gain

Set the amount of boost or attenuation for the low, mid, and high band.

Cut Low Frequencies/Cut Mid Frequencies/Cut High Frequencies

Cut the low, mid, and high band.

Output meter

Shows the level of the output signal.

DualFilter

DualFilter filters out specific frequencies while allowing others to pass through.

Position

Sets the filter cutoff frequency. If you set this to a negative value, **DualFilter** acts as a low-pass filter. Positive values cause **DualFilter** to act as a high-pass filter.

Resonance

Sets the sound characteristic of the filter. With higher values, a ringing sound is heard.

EnvelopeShaper

EnvelopeShaper can be used to attenuate or boost the gain of the attack and release phase of audio material.

You can use the controls 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.

Flanger

This is a classic flanger effect with added stereo enhancement.

Delay

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

Feedback

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

Mode

Allows you to toggle between **LFO** and **Manual** mode.

In **LFO** mode, you can define the sweep rate or sync it to the project tempo. In **Manual** mode, you can change the sweep manually.

Rate

If **Tempo Sync** is activated, **Rate** allows you to specify the base note value for synchronizing the flanger 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.

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.

Lo Range/Hi Range

Set the frequency boundaries for the flanger sweep.

Waveform Shape

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

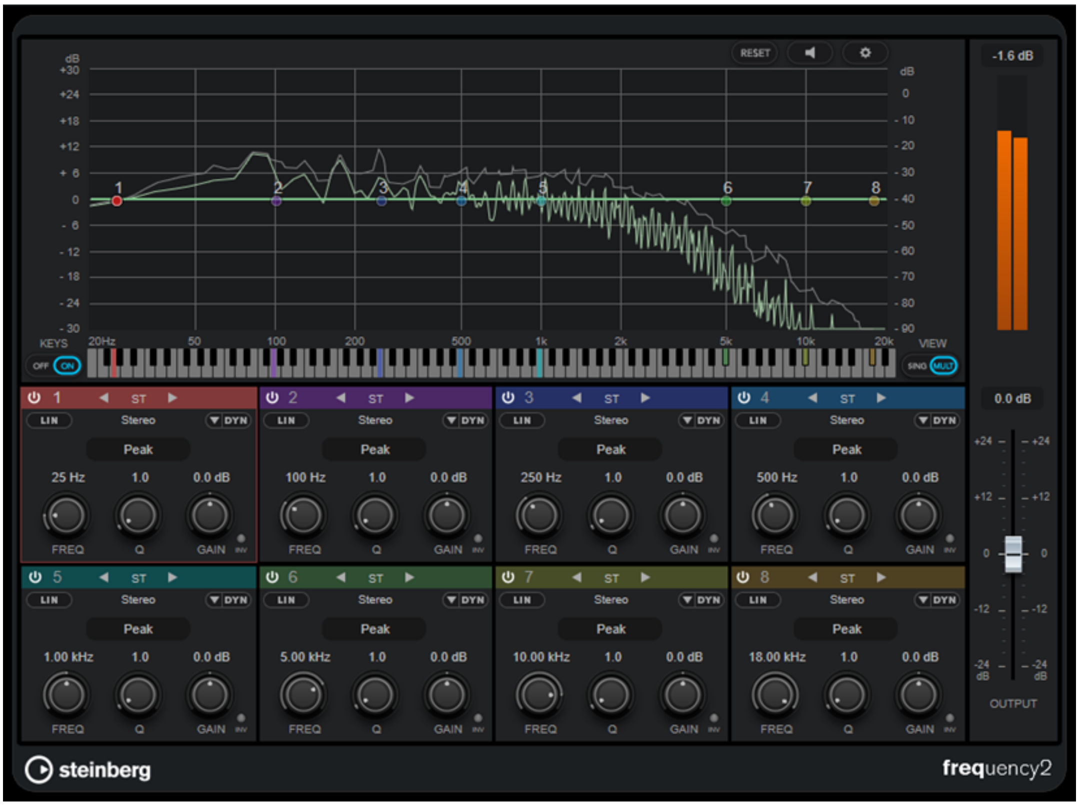
Lo Filter/Hi Filter

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

Frequency 2

Frequency 2 is a high-quality equalizer with 8 fully parametric bands. The bands can act as shelving filter, as peak or notch filter (band-pass), or as cut filter (low-pass/high-pass).

The plug-in supports separate internal or external side-chaining for each band. For **Low Shelf**, **High Shelf**, and **Peak** filters, dynamic filtering lets you determine when and how the EQ is applied, depending on the dynamics of the audio material.



Main Section

Reset

Alt/Opt-click this button to reset all parameter values.

Auto Listen for Filters

If this option is activated and you edit a parameter of a band, the corresponding frequency range is isolated. This helps you to focus on a particular frequency range and allows you to locate unwanted frequencies in your audio.

Global Settings

Opens the settings dialog for the spectrum display.

Keys

Shows/Hides the keyboard below the graphical editor.

On the keyboard, color indicators reflect the center frequencies of all active equalizer bands. You can adjust the frequency of a band by dragging its color indicator. If you drag the color indicator of a band to a key, the band is set to its exact frequency.

View

Toggles between single-band and multi-band view. Single-band view provides additional parameters for each band.

NOTE

To toggle between single-band and multi-band view, you can also double-click on the top of a band section.

Output

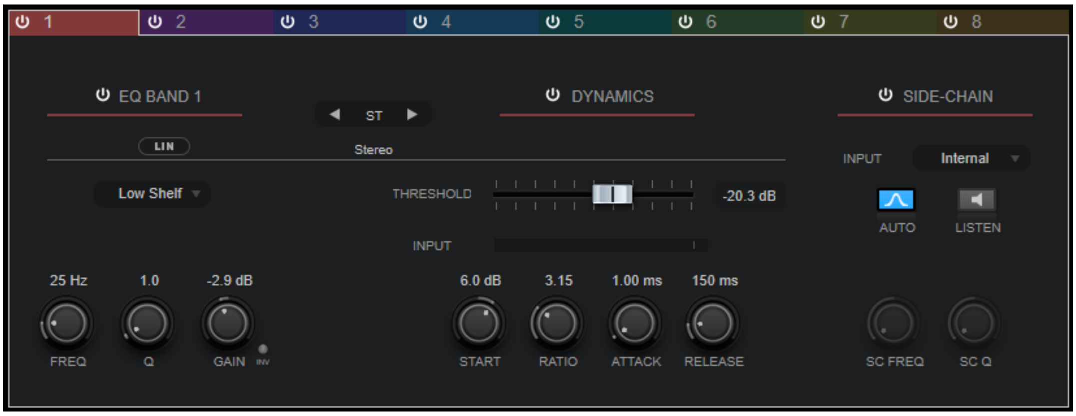
Adjusts the overall output level.

Output meter

Shows the level of the overall output signal.

Band Settings

Multi-band view



Single-band view

Activate/Deactivate Band

Activates/Deactivates the corresponding band.

NOTE

- To activate/deactivate a band, you can also double-click the corresponding handle in the graphical editor.
- You can modify the parameters of deactivated bands.

Switch Processing buttons

Allow you to switch between left/right, stereo, and mid/side processing. In **Left/Right** or **Mid/Side** processing mode, you can make different settings for the two channels.

IMPORTANT

When using **Mid/Side** processing mode, we recommend that you activate **Linear Phase Processing** in order to avoid unwanted sound colorization.

NOTE

This setting is only available for stereo tracks.

Linear Phase Processing

Activates/Deactivates linear phase mode for the corresponding band.

Linear phase mode avoids unwanted frequency-dependent phase shifts of the audio signal that might occur with standard minimum phase equalizing.

Activating this option deactivates dynamic filtering for the corresponding band.

NOTE

- Linear phase mode leads to an increase in latency.
- In rare cases, for example, when using low-cut filtering with a high slope for bass signals, you may hear an unwanted pre-ringing effect.

Filter type

You can choose between the filter types **Low Shelf**, **Peak**, **High Shelf**, and **Notch**. For band 1 and 8, you can also select the types **Cut 6**, **Cut 12**, **Cut 24**, **Cut 48**, and **Cut 96**.

- **Low Shelf** boosts or attenuates frequencies below the cutoff frequency by the specified amount.
- **Peak** boosts or attenuates frequencies at the set frequency value with a bell-shaped filter.
- **High Shelf** boosts or attenuates frequencies above the cutoff frequency by the specified amount.
- **Notch** boosts or attenuates frequencies at the set frequency value with a very narrow filter.
- **Cut** attenuates frequencies below (band 1) or above (band 8) the set frequency. You can choose between different slopes: 6 dB, 12 dB, 24 dB, 48 dB, or 96 dB per octave.

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.

NOTE

- You can adjust the **Freq** parameter of a band in the graphical editor by -clicking the corresponding handle and moving the mouse left and right.
- Ensure that you enter a space between the note and the cent offset. Only in this case, the cent offsets are taken into account.

If the band is active, the frequency value is reflected as a highlighted key on the keyboard below the graphical editor.

Q

For **Peak** and **Notch** filters, this parameter controls the width of the band. For **Low Shelf** and **High Shelf** filters, it adds a drop or a boost, depending on the gain setting of the band.

NOTE

- You can adjust the **Q** parameter of a band in the graphical editor by -clicking the corresponding handle 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. If **Dynamic Filtering** is activated, this is also the target gain value.

NOTE

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

Invert Gain

Inverts the value of the gain parameter. Positive gain values become negative and vice versa.

Show Dynamics Parameters

Shows/Hides parameters for dynamic filtering in multi-band view.

NOTE

This setting is only available for the filter types **Low Shelf**, **Peak**, **High Shelf**, and **Notch**.

Activate/Deactivate Dynamic Filtering

Activates/Deactivates dynamic filtering for the corresponding band. If this option is activated, the filtering of the band depends on the dynamics of the input signal.

Activating this setting deactivates linear phase mode for the corresponding band.

NOTE

This setting is only available for the filter types **Low Shelf**, **Peak**, **High Shelf**, and **Notch**.

Threshold

Determines the threshold level. Only signal levels above the threshold are dynamically filtered.

Input meter

Shows the level of the input signal.

NOTE

In multi-band view, the input level meter is combined with the threshold handle.

Start

Allows you to adjust the starting point for the gain. Dynamic filtering takes place from this point to the set **Gain** of the EQ band.

NOTE

This setting is only available in single-band view.

Ratio

The higher the level of the input signal is above the threshold, the more filtering occurs. Low ratio values mean that the filter starts to boost or attenuate smoothly above the threshold. High ratio values mean that the target gain is reached almost immediately.

Attack

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

Release

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

Side-Chain

Activates/Deactivates internal side-chaining for the corresponding band. This allows you to shape the input signal according to the side-chain filter parameters.

NOTE

- This setting is only available in single-band view.
- Side-chaining is only available if dynamic filtering is activated.
- If side-chaining is activated, this is indicated by showing **SC** on the top of a band section.

Input

Sets the side-chain input for the corresponding band. If **Internal** is selected, the input signal of the plug-in is used as side-chain source. If **Side-Chain 1** to **Side-Chain 8** is selected, the signal of tracks that are routed to the corresponding plug-in side-chain input is used.

NOTE

This setting is only available in single-band view.

Side-Chain Filter Auto

Activates/Deactivates automatic filtering of the side-chain signal. If this parameter is activated, the **SC Freq** and **SC Q** parameters are deactivated. Instead, the **Freq** and **Q** values of the corresponding band are used.

NOTE

This setting is only available in single-band view.

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.

NOTE

This setting is only available in single-band view.

SC Freq

Sets the frequency of the side-chain filter for 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.

NOTE

- Ensure that you enter a space between the note and the cent offset. Only in this case, the cent offsets are taken into account.
- This setting is only available in single-band view.

SC Q

Sets the resonance or width of the side-chain filter for the corresponding band.

NOTE

This setting is only available in single-band view.

Global Settings

- To open the **Global Settings**, click **Global Settings** above the spectrum display.

Spectrum Display

Show Spectrum

Activates/Deactivates the spectrum display.

Peak Hold

Holds the peak values of the spectrum display for a short time.

Smooth

Determines the reaction time of the spectrum display. Lower values result in faster reaction times and smoother transitions.

Bar Graph

If this option is activated, the frequency spectrum is analyzed into 60 separate bands that are displayed as vertical bars.

Two Channels

If this option is activated, the spectrums of the left and right channels are displayed separately.

Slope

Tilts the spectrum display around the 1 kHz pivot.

EQ Curve

Show Curve

Shows/Hides the EQ curve in the spectrum display.

Filled

If this option is activated, the EQ curve is filled. **Amount** allows you to specify the degree of coverage between 10 and 80 %.

Gate

Gating, or noise gating, silences audio signals below a set threshold. As soon as the signal level exceeds the threshold, the gate opens to let the signal through.

Attack (0.1 to 1000 ms)

Sets the time after which the gate opens when it is triggered.

NOTE

Deactivate the **Live** button to make sure that the gate is already open when a signal above the threshold is played back.

Hold (0 to 2000 ms)

Determines how long the gate remains open after the signal drops below the threshold level.

Release (10 to 1000 ms or Auto mode)

Sets the time after which the gate closes after the set **Hold** time. If **Auto Release** is activated, the plug-in automatically finds the best release setting for the audio material.

Threshold

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

State LED

Indicates whether the gate is open (LED lights up in green), closed (LED lights up in red), or in an intermediate state (LED lights up in yellow).

Analysis (Pure Peak to Pure RMS)

Determines whether the input signal is analyzed according to peak or RMS values, or a mixture of both. A value of 0 is pure peak and 100 pure RMS. **RMS** mode operates using the average power of the audio signal as a basis, whereas **Peak** mode operates more on peak levels. As a general guideline, **RMS** mode works better on material with few transients such as vocals, and **Peak** mode works better for percussive material with a lot of transient peaks.

Range

Adjusts the attenuation of the gate when it is shut. If **Range** is set to minus infinite -oo, the gate is completely shut. The higher the value, the higher the level of the signal that passes through the shut gate.

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.

Side-Chain Section

Side-Chain

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

Monitor

Allows you to monitor the filtered signal.

Center

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.

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.

Limiter

Limiter is designed to ensure that the output level never exceeds a set output level, to avoid clipping in following devices.

Limiter can adjust and optimize the **Release** parameter automatically according to the audio material, or it can be set manually. **Limiter** features separate meters for the input, output and the amount of limiting (middle meters).

Input (-24 to 24 dB)

Sets the input gain.

Release (0.1 to 1000 ms or Auto mode)

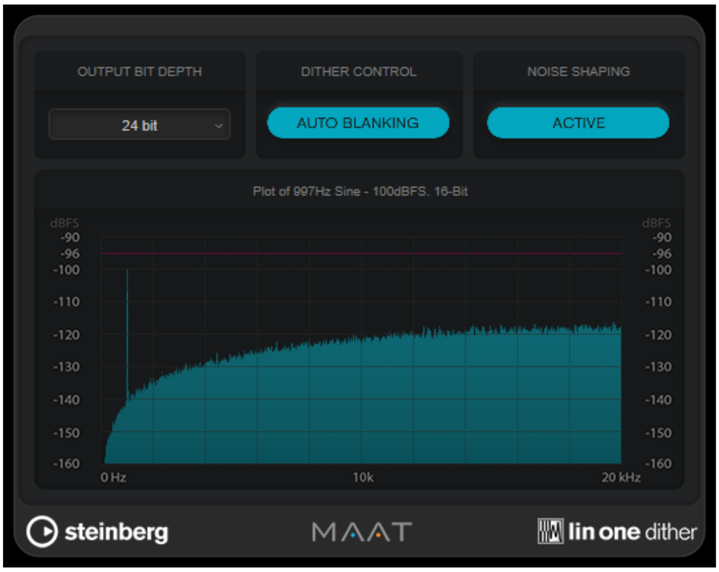
Sets the time after which the gain returns to its original level. If **Auto Release** is activated, the plug-in automatically finds the best release setting for the audio material.

Output

Sets the maximum output level.

Lin One Dither

Lin One Dither is a dithering plug-in that uses advanced algorithms and offers additional noise shaping to increase the apparent signal-to-noise ratio by altering the spectrum of the low-level audio signal.



NOTE

We recommend applying dithering post-fader on output busses.

Output Bit Depth

Sets the bit depth of the output signal.

NOTE

Dithering changes the bit depth, but not the sample size. For example, when dithering 24 bit to 16 bit, the file is still 24 bit in size, although only 16 bits of information have significance. When processing a 16-bit file, specify the **Output Bit Depth** value accordingly to avoid the creation of unnecessarily large files.

Dither Control

If **Auto Blanking** is activated, the dither noise is gated during silent passages.

Noise Shaping

Activates/Deactivates noise shaping. Noise shaping increases the apparent signal-to-noise ratio.

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 contemporary styles of music.

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

- **Release** sets the overall release time.
- **Recover** allows for a faster signal recovering at the beginning of the release phase.

Optimize

Determines the loudness of the signal.

Mix

Sets the level balance between dry and 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.

MonoDelay

This is a mono delay effect. The delay line uses 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 control 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 control 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.

MonoToStereo

MonoToStereo turns a mono signal into a pseudo-stereo signal. The plug-in can be used on mono audio or on stereo audio with equal channels.

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

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.

Color

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

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 **Morph Factor** 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.

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. This allows you to set the plug-in into ducking mode per 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 Band

Bypasses a frequency band.

Solo Band

Solos the corresponding frequency band.

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 controls. The threshold is marked by the first breakpoint where the line deviates from the straight diagonal.

Threshold (-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 its 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.

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-Chain

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

Frequency

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.

Octaver

This plug-in can generate two additional voices that track the pitch of the input signal one octave and two octaves below the original pitch. **Octaver** is best used with monophonic signals.

Direct

Sets the level balance between the dry signal and the wet signal. A value of 0 means that 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 generated signal one octave below the original pitch. A setting of 0 means that the voice is muted.

Octave 2

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

Phaser

Phaser produces the well-known swooshing phasing effect with additional stereo enhancement.

Feedback

Determines the character of the phaser effect. Higher settings produce a more pronounced effect.

Width

Sets the intensity of the modulation effect between higher and lower frequencies.

Mode

Allows you to toggle between **LFO** and **Manual** mode.

In **LFO** mode, you can define the sweep rate or sync it to the project tempo. In **Manual** mode, you can change the sweep manually.

Rate

If **Tempo Sync** is activated, **Rate** allows you to specify the base note value for tempo-syncing the phaser sweep (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.

Spatial

If you are using multi-channel audio, the **Spatial** parameter creates a 3-dimensional impression by delaying modulation in each channel.

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/Hi Filter

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

PingPongDelay

This is a stereo delay effect that alternates each delay repeat between the left and right channels. The delay line uses 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 control 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 control 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

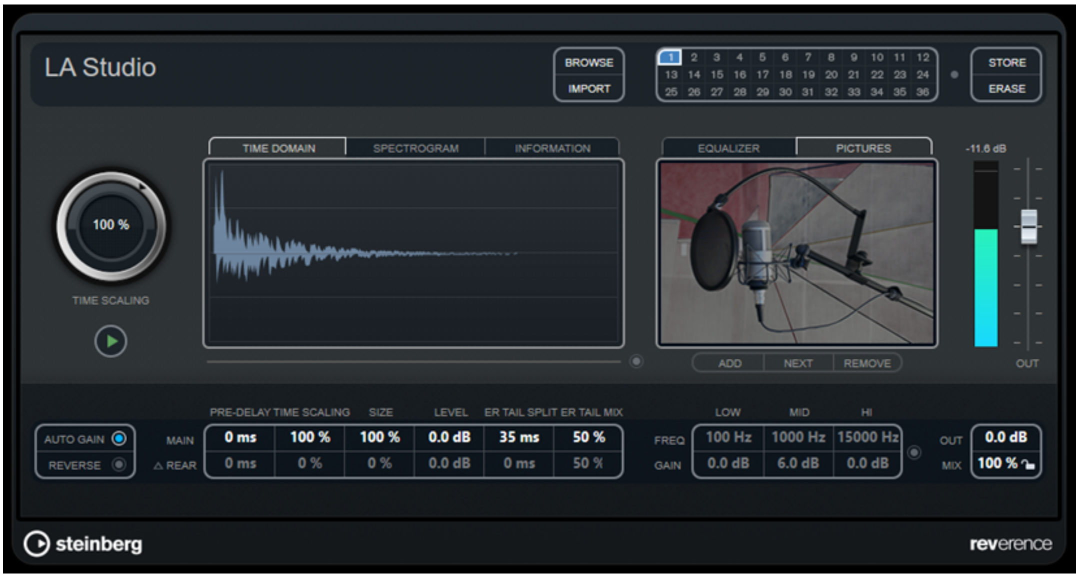
Sets the stereo width for the left/right repeats. Turn clockwise for a more pronounced stereo ping-pong effect.

Start Left/Start Right

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

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 **Filters** section (by clicking the **Set Up Window Layout** button at the top right of the browser window).

Import

Click this button to load your own impulse response files from disk. 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.

RELATED LINKS
Reverb Settings on page 35
EQ Settings on page 37
Pictures Section on page 38
Custom Impulse Responses on page 39
Relocating Content on page 41

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.

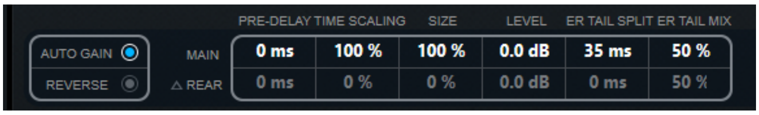
   NOTE

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

RELATED LINKS
Importing Impulse Responses on page 39

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.

Rear

NOTE

This parameter is not available in Dorico.

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

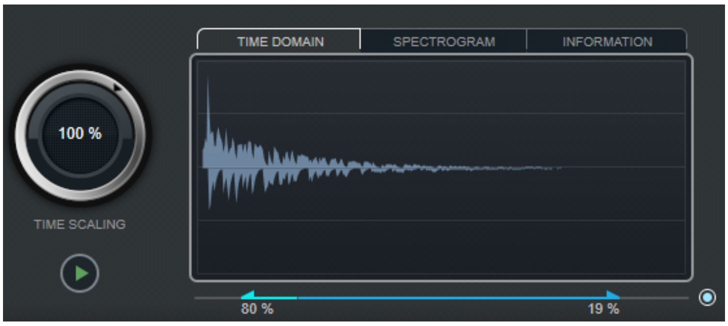
Sets a split point between the early reflections and the tail, allowing you to determine where the reverb tail begins. A value of 60 means that the split point is set to 60 ms.

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

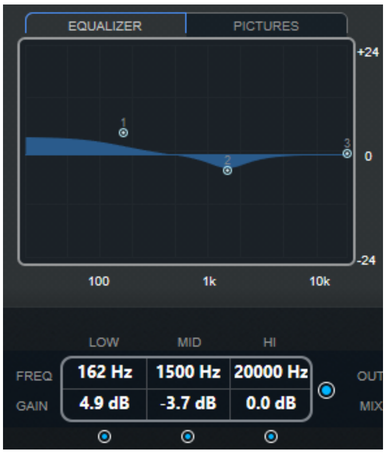
This slider allows you to trim the start and end of the impulse response. Drag the front handle to trim the start of the impulse response, and the end handle to trim the reverb tail.

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)

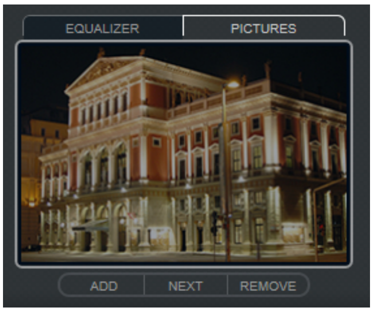
Sets the frequency of the high band.

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 your hard disk.

Output Settings

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

Output activity 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 your own impulse responses 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.

In Dorico, **REVerence** always uses a stereo channel width. When importing impulse response files with more than 2 channels, the plug-in only reads as many channels as needed. When importing mono impulse response files, **REVerence** uses the mono signal for both stereo channels.

RELATED LINKS

True Stereo on page 40

Importing Impulse Responses

REVerence allows you to import your own impulse response files. Before importing these impulse response files, you can preview their effect.

PREREQUISITE

To preview the effect of impulse response files during the import process, do one of the following:

- If you use **REVerence** as an insert effect, play back the event to which you want to add the effect in a loop.
- If you use **REVerence** in the **Direct Offline Processing** window, activate **Audition** and **Audition Loop**.

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 (for example, the **Audio Connections** window), see below.
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.

Reading Order of Input Channels

REVerence reads input channels in the following order.

Number of input channels	Channel order in REVerence
1	L
2	L/R
3	L/R/C (C is ignored)
4	L/R/LS/RS (LS/RS are ignored)
4	LL/LR/RL/RR (True Stereo)
5	L/R/C/LS/RS (C/LS/RS are ignored)
6	L/R/C/LFE/LS/RS (C/LFE/LS/RS are ignored)

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	The signal from this source…	…was recorded with this microphone
LL	left source	left microphone
LR	left source	right microphone
RL	right source	left microphone

Channel	The signal from this source…	…was recorded with this microphone
RR	right source	right microphone

REVerence automatically works in true-stereo mode if you load a 4-channel impulse response.

Therefore, if you are working with surround files, that is, 4-channel impulse responses recorded with a Quadro configuration (L/R, LS/RS), these files would be processed in true-stereo mode, too.

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, **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 you 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.

   If you are unsure where the presets need to be stored, you can find the paths in the **MediaBay** (see the **Operation Manual**).
3. Open **REVerence** on the second computer and try to load the preset or program that you want to work with.

   The **Locate 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 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. When using **RoomWorks SE** inserted in an FX channel, you most likely want to set this to 100 %.

Rotary

This modulation effect simulates the classic 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)

This pop-up menu allows you to select the MIDI controller that controls the rotary speed. If you do not want to use MIDI realtime control, set this to **Automation**. If you select **PitchBend** as MIDI controller, the speed changes with an up or down flick of the bender. If other MIDI controllers are used, the speed changes at MIDI value 64.

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 control 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** control and/or with a MIDI controller that you can select on the **Speed Mod Control (MIDI)** pop-up menu.

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 dry and wet signal.

StereoEnhancer

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

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 possible unwanted coloring of the sound which sometimes can occur when enhancing the stereo image.

Color

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

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.

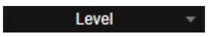
NOTE

- You can also pause/continue the measurement for the selected module by right-clicking 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 in 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 in the display when playback is stopped.

Module selector

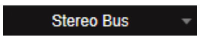


Allows you to select a module for the selected slot.

Open Module Settings

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

Channel selector



Allows you to select which channels are displayed. The available channel configurations depend on the channel configuration of the track and on the selected module.
Mixdown allows you to display the average value of all channels within the track.

NOTE

- The channel selector is only available for configurations with two or more channels.

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**.

RELATED LINKS
Module Settings Window on page 48
Signal Modules on page 49
Spectral Domain Modules on page 60
Phase Modules on page 56
Waveform Modules on page 65

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.

RELATED LINKS
Signal Modules on page 49
Spectral Domain Modules on page 60
Phase Modules on page 56
Waveform Modules on page 65

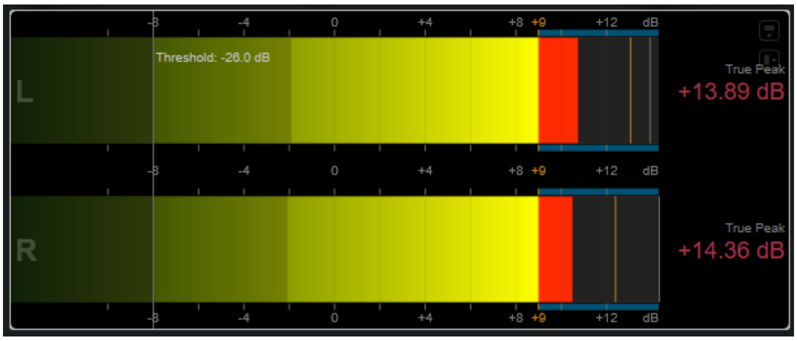
Signal Modules

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 in 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 , 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.

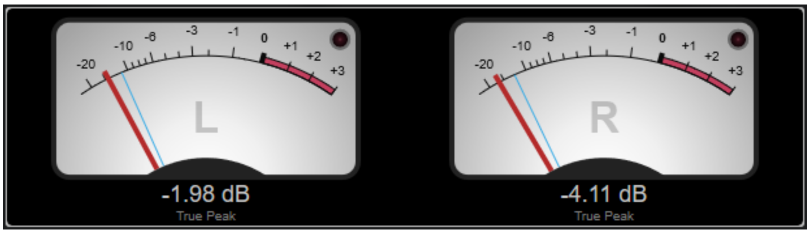
Max. Value

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 in 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 , 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.

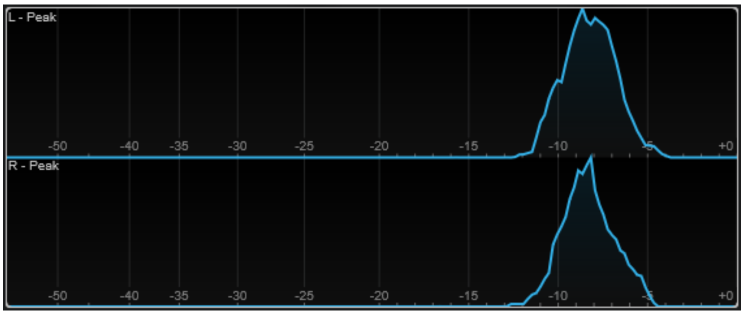
Max. Value

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 , 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

Smoothes 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.

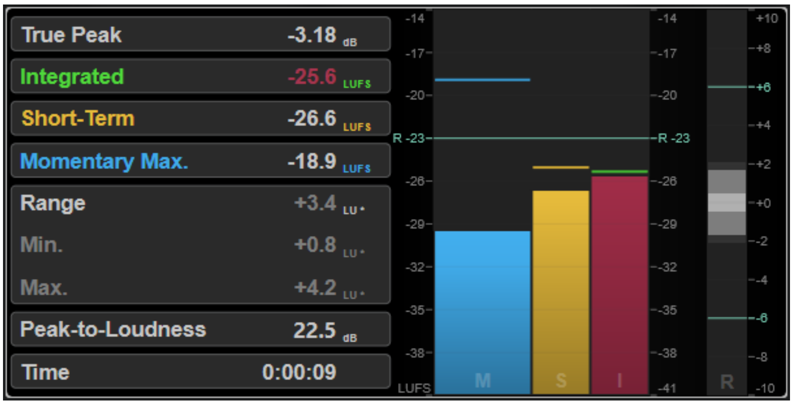
Measurement Modules

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 switch the meter between the EBU +9 scale and the EBU +18 scale.

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

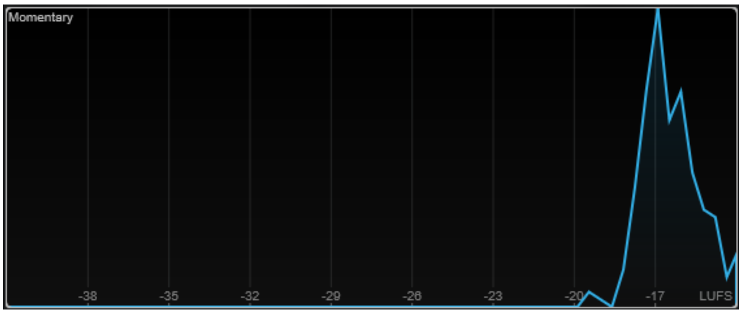
Sets a reference value for the loudness range. If higher values are detected, the loudness meter indicates clipping.

Tol. Range

Sets a tolerance value for the loudness range.

Loudness 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 switch the meter between the EBU +9 scale and the EBU +18 scale.

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

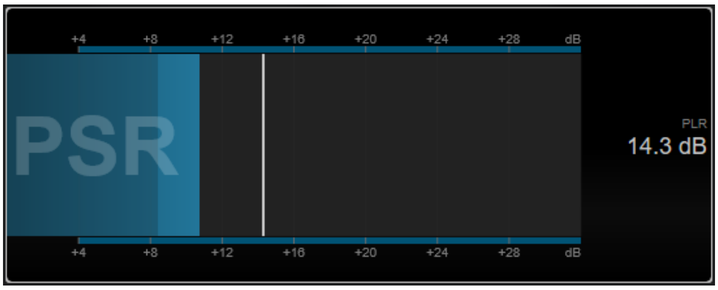
Smoothes the display of the loudness curve.

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

Smoothes the temporal display of the PSR value.

NOTE

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

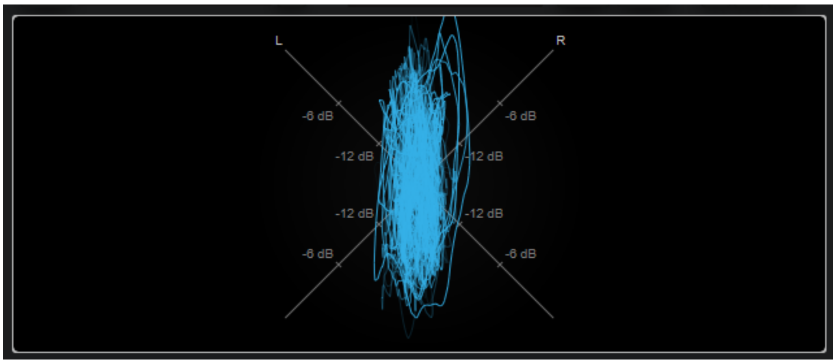
Phase Modules

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.

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

Zoom

Allows you to zoom in 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

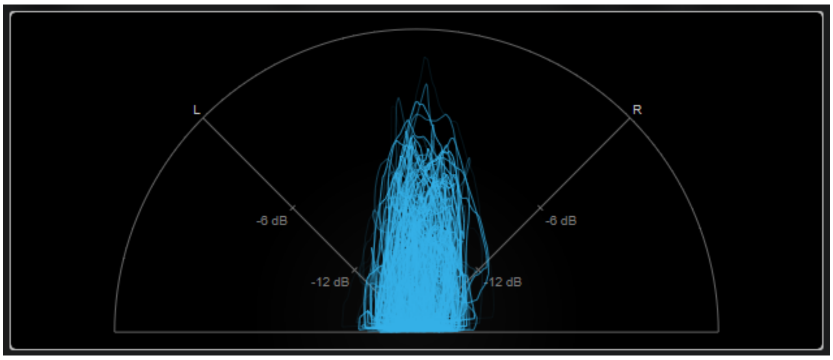
Activates/Deactivates the axis labeling.

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.



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

Zoom

Allows you to zoom in 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

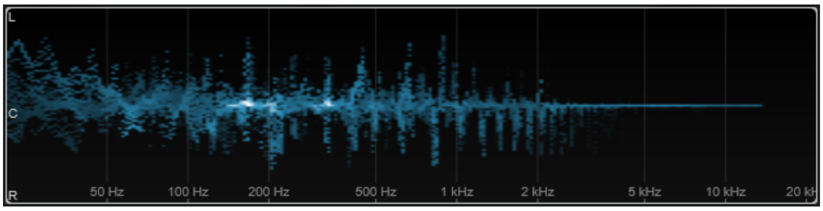
Activates/Deactivates the axis labeling.

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.

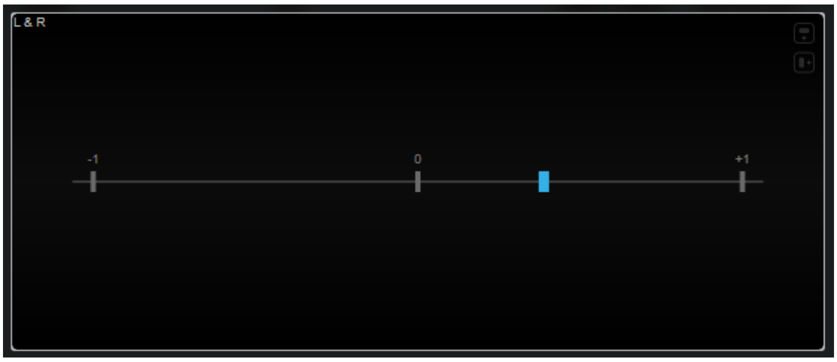
Sets the number of bands per octave.

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

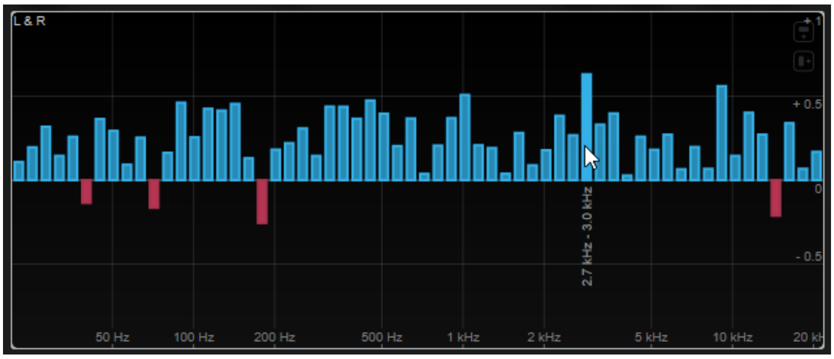
Smoothes the temporal display of the correlation.

NOTE

Alternatively, you can move the mouse pointer over the selected module, hold , 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

Smoothes the temporal display of the correlation.

NOTE

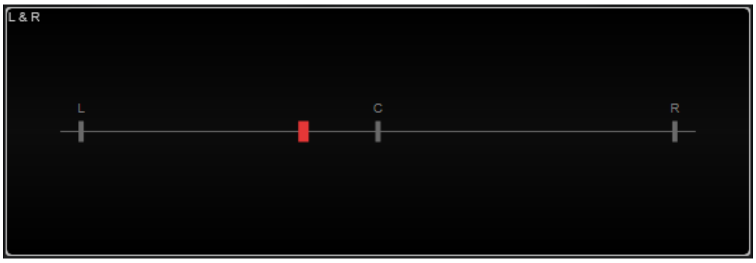
Alternatively, you can move the mouse pointer over the selected module, hold , and use the mouse wheel to adjust this parameter

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

Smoothes the temporal display of the correlation.

NOTE

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

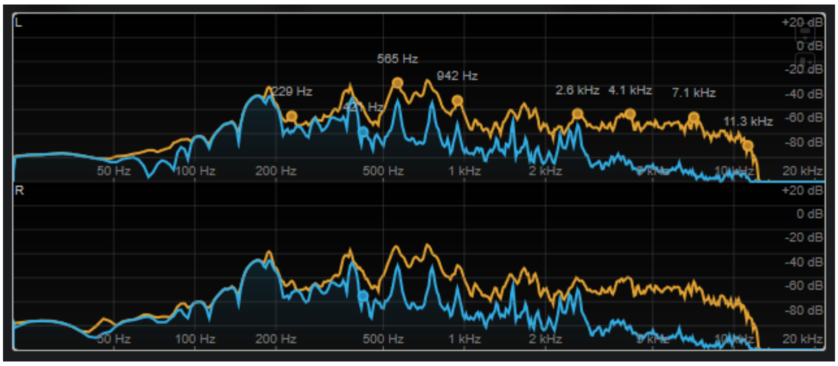
Spectral Domain Modules

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

Smoothes the temporal display.

NOTE

Alternatively, you can move the mouse pointer over the selected module, hold , 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

Smoothes 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

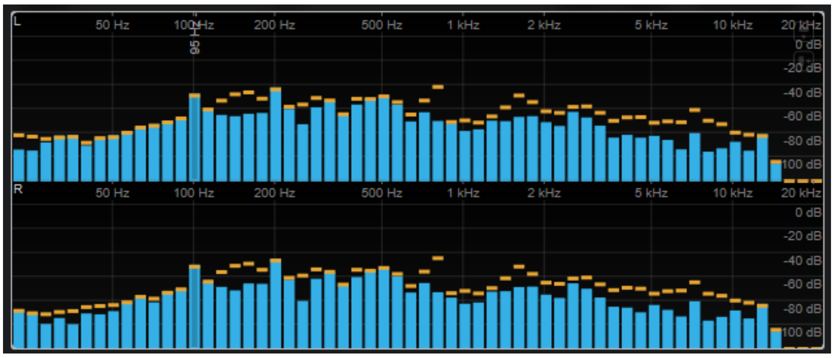
Sets the maximum value of the scale.

Slope

Adds a slope to the frequency spectrum.

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

Smoothes the temporal display.

NOTE

Alternatively, you can move the mouse pointer over the selected module, hold , 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 , 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

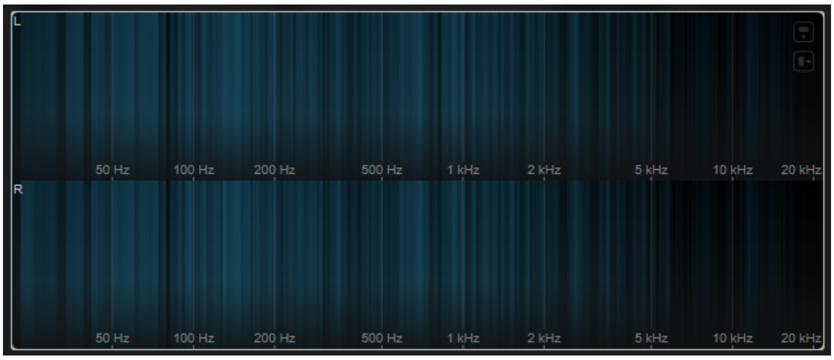
Sets the maximum value of the scale.

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

Smoothes the temporal display.

NOTE

Alternatively, you can move the mouse pointer over the selected module, hold , 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

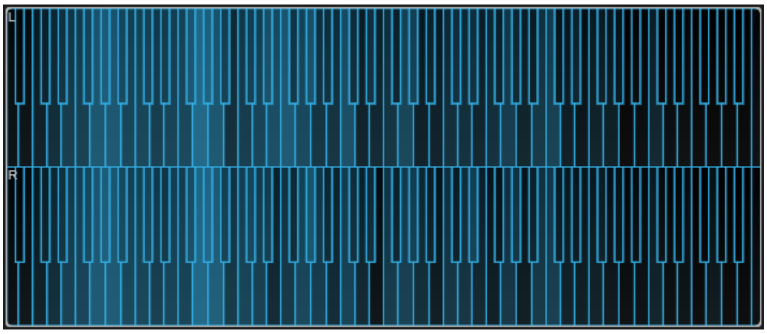
Sets the maximum value of the scale.

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

Smoothes the temporal display.

NOTE

Alternatively, you can move the mouse pointer over the selected module, hold , 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

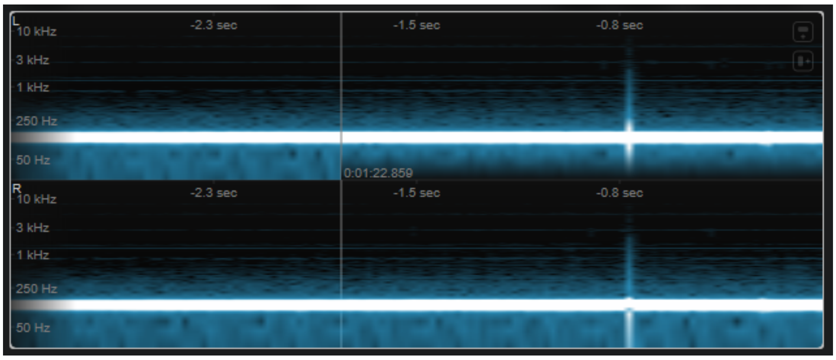
Sets the maximum value of the scale.

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 , 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.

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.

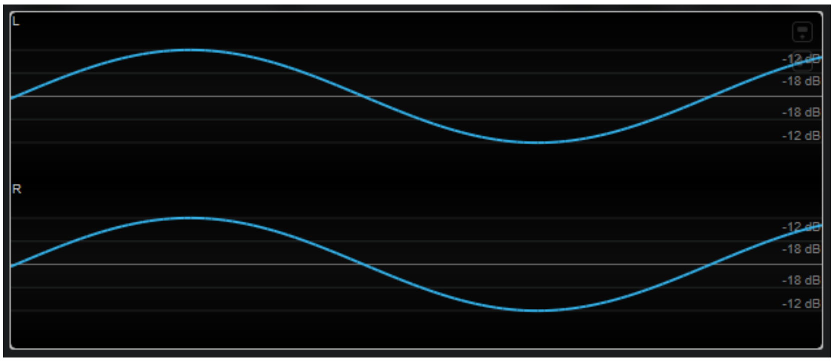
Waveform Modules

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.



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

Zoom

Allows you to zoom in 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 in 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

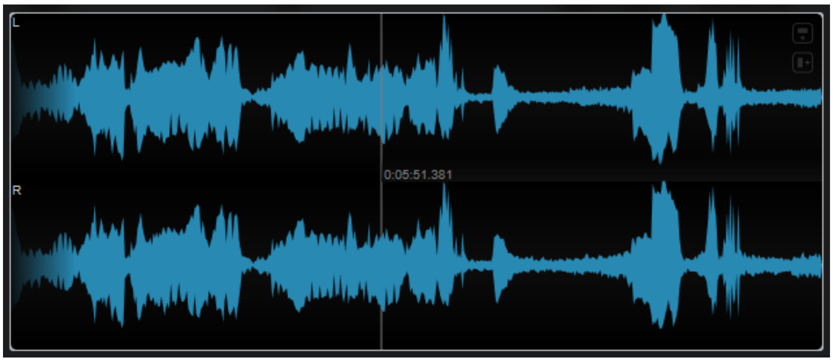
This option is only available if **Auto Zoom** is deactivated.

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.

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

Zoom

Allows you to zoom in 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 , 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

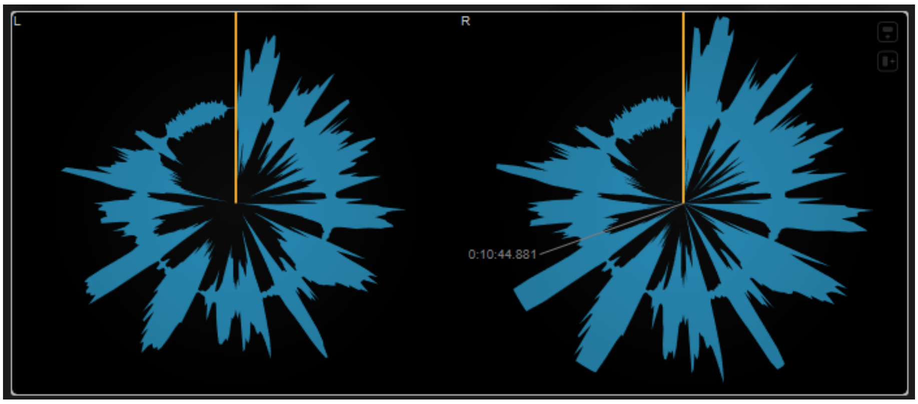
This option is only available if **Auto Zoom** is deactivated.

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.

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

Zoom

Allows you to zoom in 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 , 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.

ToneBooster

ToneBooster is a filter that allows you to raise the gain in a selected frequency range. It is particularly useful if it is inserted before **AmpSimulator** in the plug-in chain, greatly enhancing the tonal varieties available.

Gain

Adjusts the gain of the selected frequency range by up to 24 dB.

Tone

Sets the center filter frequency.

Width

Sets the resonance of the filter.

Mode selector

Sets the basic operational mode of the filter: **Peak** or **Band** Mode.

RELATED LINKS
AmpSimulator on page 4

Tremolo

Tremolo produces amplitude modulation.

Rate

If **Tempo Sync** is activated, **Rate** allows you to 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 modulation speed can be set freely with the **Rate** dial.

Sync

Activates/Deactivates tempo sync.

Depth

Governs the depth of the amplitude modulation.

Spatial

Adds a stereo effect to the modulation.

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

Show 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 creating 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 initial 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.

Side-Chain

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

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.

Vibrato

Vibrato creates pitch modulation.

Depth

Sets the intensity of the pitch modulation.

Rate

If **Tempo Sync** is activated, **Rate** allows you to 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 modulation speed can be set freely with the **Rate** dial.

Sync

Activates/Deactivates tempo sync.

Spatial

Adds a stereo effect to the modulation.

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

Show 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 initial part of the signal passes through unprocessed.

Punch

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 **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.

Output (-48 to 24 dB)

Sets the output gain.

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 six buttons, arranged according to the position of the corresponding elements in the signal chain. These buttons open different pages in the display section of the plug-in panel: **Pre-Effects**, **Amplifiers**, **Cabinets**, **Post-Effects**, **Microphones**, **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/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 – Determines 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 – Determines the loudness of the signal.

Chorus

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

Width – Determines 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. 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.

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 trigger the gate to open, and signal levels below the set threshold close the gate.

Release – Sets the time after which the gate closes.

Equalizer

Low – Sets the level of the low frequency portion of the incoming signal.

Middle – Sets the level of the mid frequency portion of the incoming signal.

High – Sets 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.

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 **None** from the pop-up menu.
- To change the order of the effects in the chain, click on an effect and drag it to another position.
- 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 track configuration.
- Using quick controls you can set up an external MIDI device, such as a foot controller, to control the **VST Amp Rack** effects. For more information about quick controls, see the **Operation Manual**.

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.

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.

All amps provide the following sound-related parameters which have a significant impact on their overall character and sound:

Gain
: Sets the amount of boost for the amp.

Bass
: Allows you to raise or lower the low frequency part of the signal.

Middle
: Allows you to raise or lower the mid frequency part of the signal.

Treble
: Allows you to raise or lower the high frequency part of the signal.

Presence
: Allows you to raise or lower the upper mid frequency part of the signal.

Master
: Sets the output level for the amp.

The different amps keep their settings if you switch models. However, 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 track, and activate the **Stereo** button.
- To use the effect in mono-mode, make sure that the plug-in is inserted on a mono track, and activate the **Mono** 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** control. If a band is active, the LED to the left of the **Gain** control 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.

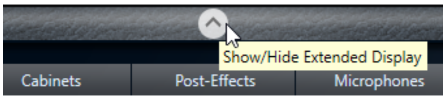
In the compact view, the display section is hidden from view. You can change the amp settings and switch amps or cabinets using the mouse wheel.

Using the Smart Controls

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

Switching between Default and Compact View

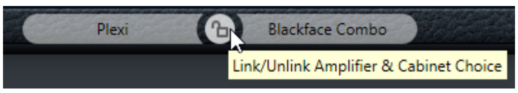
- To toggle between the different views, click the down/up arrow button (Show/Hide Extended Display) at the top center of the plug-in frame.



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.

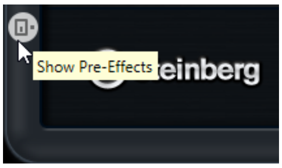
- To select a different amplifier or cabinet, click the name and select a different model from the pop-up menu.
- To lock the amplifier and cabinet combination, activate the **Link/Unlink Amplifier & Cabinet Choice** button. If you now select another amp model, the cabinet selection follows. However, 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** or **Show Post-Effects** button at the bottom left or right of the plug-in frame.



VST Bass Amp

VST Bass Amp is a bass 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, the following buttons open different pages in the display section of the plug-in panel: **Pre-Effects**, **Amplifiers**, **Cabinets**, **Post-Effects**, **Microphones**, **Configuration**, and **Master**.

These buttons are arranged according to the position of the corresponding elements in the signal chain.

Below the display section, the selected bass amplifier is displayed. The color and texture of the area below the bass amp indicate the selected cabinet.

Pre/Post-Effects

On the **Pre-Effects** and **Post-Effects** pages, you can select up to six common bass effects. On both pages, the same effects are available, the only difference being the position in the signal chain (before or after the bass 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.

Envelope Filter

Range – Determines the frequency range.

Q-Factor – Sets the intensity of the envelope filter effect.

Sensitivity – Determines how sensitive the effect reacts to the instrument level.

Attack – Determines how quickly an effect reacts to the input signal.

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

Type – Sets the filter type.

Release – Determines how quickly the effect fades after the input signal stops.

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.

Compressor MB

Lo Intensity – Sets the compressor effect in the low frequency band. Activate/deactivate **Auto Makeup Mode** by clicking the LED at the top right of the control.

Hi Intensity – Sets the compressor effect in the high frequency band. Activate/deactivate **Auto Makeup Mode** by clicking the LED at the top right of the control.

Crossover – Determines the crossover frequency between the low frequency band and the high frequency band.

Output – Sets the output level.

Limiter

Threshold – Determines 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 – Determines the loudness of the signal.

Chorus

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

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

Tone – Allows you to attenuate low frequencies.

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

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.

Tone – Allows you to attenuate the low frequencies.

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

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.

Tone – Allows you to attenuate the low frequencies.

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

DI Driver

Level – Sets the output level.

Blend – Blends between normal and tube emulation circuitry. With **Blend** at 0, **Drive** and **Presence** are not active.

Bass – Boosts or attenuates low frequencies.

Treble – Boosts or attenuates high frequencies.

Presence – Boosts or attenuates upper harmonics and attacks.

Drive – Sets gain and overdrive.

Enhancer

Enhance – Simulates the classic enhancer effect.

Tone – Allows you to attenuate low frequencies.

Octaver

Direct – Adjusts the level of the original signal. 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.

Tone – Changes the sound character of the generated signal.

Delay

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

Feedback – The higher this setting, the more delay repeats are created.

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

Tape Ducking Delay

Delay – The **Delay** parameter sets the delay time in milliseconds.

Feedback – The higher this setting, the more delay repeats are created.

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.

Tone – Allows you to attenuate the low frequencies.

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

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.

Magneto II

Drive – Controls the amount of tape saturation.

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

HF-Adjust – Sets the amount of high frequency content of the saturated signal.

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 – Sets the level of the low frequency portion of the incoming signal.

Middle – Sets the level of the mid frequency portion of the incoming signal.

High – Sets the level of the high frequency portion of the incoming signal.

Graphical EQ

Display – Consists of 8 sliders that set the level of each frequency band. Allows you to draw response curves by clicking and dragging with the mouse.

Reset Sliders – At the lower right of the Display. Flattens all values to 0 dB.

Output Slider – Allows you to control the frequency response.

Reverb

Type – A convolution-based reverb effect. This parameter allows you to switch between the reverb types **Studio**, **Hall**, **Plate**, and **Room**.

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

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 **None** from the pop-up menu.
- To change the order of the effects in the chain, click on an effect and drag it to another position.
- 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 track configuration.
- Using quick controls you can set up an external MIDI device, such as a foot controller, to control the **VST Bass Amp** effects. For more information about quick controls, see the **Operation Manual**.

Amplifiers

The amps available on the **Amplifiers** page are modeled on real-life amplifiers. Each amp features settings typical for bass recording, such as gain, equalizers, and master volume. Shape 1 and Shape 2 offer predefined tone shaping.

ValveAmp300

A famous tube amplifier from the 70s, useful for rock playing styles.

Greyhound

An amplifier, well known for its typical growl, useful for several playing styles.

GreenT

A classic amplifier from the 80s, useful for funk and rock playing styles.

Paradise
An amplifier from the 90s, with a hifi-like clear tone, that makes it applicable for several styles.

Tweed
A classic vintage amplifier from the 50s, with a characterful and bright tone. Originally created for bassists, also used by many guitar players.

iTech
A modern amplifier, with a universal sound.

All amps provide the following sound-related parameters, which have a significant impact on their overall character and sound:

Gain
Sets the amount of boost for the amp.

Bass
Allows you to raise or lower the low frequency part of the signal.

Shape 1
Adds a predefined tone shaping to the low-mid frequency part of the signal.

Lo Mid
Allows you to raise or lower the lower mid frequency part of the signal. The corresponding **Freq** control sets the frequency.

Hi Mid
Allows you to raise or lower the upper mid frequency part of the signal. The corresponding **Freq** control sets the frequency.

Shape 2
Adds a predefined tone shaping to the mid-high frequency part of the signal.

Treble
Allows you to raise or lower the high frequency part of the signal.

Master
Sets the output level for the amp.

The different amps keep their settings if you switch models, but amp settings are lost when closing **VST Bass AMP**. 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.

NOTE

To scroll through amplifiers, use the mouse wheel when hovering over the amplifier panel.

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.

The following cabinets are available:

4x10"

10" speakers provide a punchy clear sound that is suitable for “Slap” bass and regular playing styles.

10" speakers have a cleaner sound and more punch than 15" speakers.

8x10"

Compared to 4x10", double the amount of speakers.

4x12"

12" speakers provide a mellow and full sound, making them a good choice between 10" and 15" speakers.

1x15"

15" speakers provide more low frequencies compared to the other cabinets. They are suitable for rock and vintage oriented styles.

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 types.

57

Dynamic microphone with cardioid pickup pattern.

121

Ribbon microphone with figure-8 pattern.

409

Dynamic microphone with supercardioid pickup pattern.

421

Dynamic microphone with cardioid polar pattern.

545

Dynamic microphone with cardioid pattern that minimizes feedback.

5

Dynamic microphone with cardioid pickup pattern.

30

Reference and measurement microphone with omni directional polar pattern.

87

Condenser microphone with omni directional pattern.

You can choose between different microphone positions. These positions result from two different angles (on axis and off axis) and three different distances from the cabinet.

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.

- To select a microphone position, click the corresponding ball in front of the cabinet. The selected position is marked in red.
- To determine the ratio between **line** and **mic**, turn the **Mix** control on the left of the cabinet.

NOTE

To scroll through microphones, use the mouse wheel when hovering over a microphone.

Configuration

On the **Configuration** page, you can specify whether you want to use **VST Bass Amp** 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 track, and activate the **Stereo** button.
- To use the effect in mono-mode, make sure that the plug-in is inserted on a mono track, and activate the **Mono** button.

NOTE

In stereo mode, the effect requires more processing power. Use mono configuration on a stereo track to save 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 all other views, 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** control. If a band is active, the LED to the left of the **Gain** control is lit.
- To tune your guitar strings, click the pedal-like **On/Off** button to activate **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.
- NOTE

  Master EQ functions only when a cabinet is selected.

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.

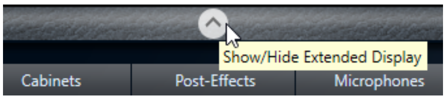
In the compact view, the display section is hidden from view. You can change the amp settings and switch amps or cabinets using the mouse wheel.

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

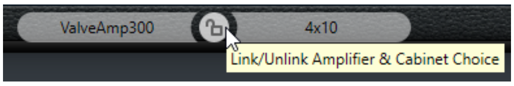
- To toggle between the different views, click the down/up arrow button (**Show/Hide Extended Display**) at the top center of the plug-in frame.



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.

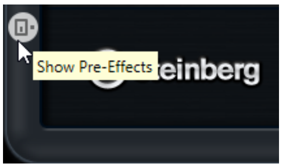
- To select a different amplifier or cabinet, click the name and select a different model from the pop-up menu.
- To lock the amplifier and cabinet combination, activate the **Link/Unlink Amplifier & Cabinet Choice** button. If you now select another amp model, the cabinet selection follows. However, 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** or **Show Post-Effects** button at the bottom left or right of the plug-in frame.



VSTDynamics

VSTDynamics is an advanced dynamics processor. It combines three separate effects: **Gate**, **Compressor**, and **Limiter**, covering a variety of dynamic processing functions.

The window is divided into three sections containing controls and meters for each effect. Activate the individual effects using the **Gate**, **Compressor**, and **Limiter** buttons. You can select between three different routing options using the **Module Configurator** button.

Gate

Gating, or noise gating, is a method of dynamic processing that silences audio signals below a set threshold. As soon as the signal level exceeds the threshold, the gate opens to let the signal through. The gate trigger input can also be filtered using an internal side-chain signal.

The following parameters are available:

Input meter

Shows the level of the input signal.

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.

Threshold

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

State LED

Indicates whether the gate is open (LED lights up in green), closed (LED lights up in red), or in an intermediate state (LED lights up in yellow).

Release (10 to 1000 ms or Auto mode)

Sets the time after which the gate closes after the set **Hold** time. If **Auto Release** is activated, the plug-in automatically finds the best release setting for the audio material.

Hold (0 to 2000 ms)

Determines how long the gate remains open after the signal drops below the threshold level.

Range

Adjusts the attenuation of the gate when it is shut. If **Range** is set to minus infinite -∞, the gate is completely shut. The higher the value, the higher the level of the signal that passes through the shut gate.

Side-Chain

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

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

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.

Compressor

Compressor reduces the dynamic range of the audio, making softer sounds louder or louder sounds softer, or both. It features a separate display that graphically illustrates the compressor curve shaped according to your settings.

Input meter

Shows the level of the input signal.

Graphical display

Visualizes the settings for **Threshold** and **Ratio** and allows you to adjust them by dragging the handles.

Gain Reduction meter

Shows the amount of gain reduction.

Threshold (-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.

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

Compensates for output gain loss caused by compression. If **Auto Make-Up Gain** is activated, the output is automatically adjusted for gain loss.

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 (attack) passes through unprocessed.

Release (10 to 1000 ms or Auto mode)

Sets the time after which the gain returns to its 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.

Limiter

A limiter ensures that the output level never exceeds a set threshold to avoid clipping in effects following in the chain. Conventional limiters usually require a very accurate setup of the attack and release parameters to prevent the output level from going beyond the set threshold level. **Limiter** adjusts and optimizes these parameters automatically according to the audio material.

Input meter

Shows the level of the input signal.

Gain Reduction meter

Shows the amount of gain reduction.

Soft Clip

If this button is activated, the signal is limited when the signal level exceeds -6 dB. At the same time, harmonics are generated, adding a warm, tube-like characteristic to the audio material.

Output

Sets the maximum output level.

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.

Output section

Output meter

Shows the level of the output signal.

Module Configurator

Changes the signal flow through the three effects. Changing the order of the effects can produce different results, and the available routing configurations allow you to quickly compare what works best for a given situation. Click **Module Configurator** to toggle between the following routing configurations:

- G-C-L (Gate-Compressor-Limiter)
- C-L-G (Compressor-Limiter-Gate)
- C-G-L (Compressor-Gate-Limiter)

WahWah

WahWah is a variable slope band-pass filter that can be auto-controlled via MIDI modeling the well-known analog pedal effect.

You can independently specify the frequency, width, and gain for the **Low** and **High** Pedal positions. The crossover point between the Lo and Hi Pedal positions lies at 50.

WahWah Parameters

Pedal

Controls the filter frequency sweep.

Pedal Control (MIDI)

Allows you to choose the MIDI controller that controls the plug-in. Set this to **Automation** if you do not want to use MIDI realtime control.

Freq Low/Freq High

Set the frequency of the filter for the Lo and Hi pedal positions.

Width Low/Width High

Set the width (resonance) of the filter for the Lo and Hi pedal positions.

Gain Low/Gain High

Set the gain of the filter for the Lo and Hi pedal positions.

Filter Slope selector

Allows you to choose between two filter slope values: 6 dB or 12 dB.

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