



Plug-in Reference

# **VST RACK** PRO

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# About VST Rack Pro

- VST Rack is a plug-in host software for Mac and Windows that enables audio processing using a variety of VST plug-in effects, such as reverbs, maximizers, compressors, and limiters. It uses Steinberg's audio engine, which has a good reputation for its use in Nuendo and Cubase, to provide sound engineers with a high-quality and stable plug-in environment.
- VST Rack Pro is bundled with many classic Yamaha and Steinberg VST plug-ins that have been used in Yamaha digital mixers and Steinberg digital audio workstations. This provides you access to high-quality effects without any need to purchase new VST plug-ins.

## Manuals

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

## NOTICE

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

# List of Plug-ins

## Steinberg



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

Delay	MonoDelay
	PingPongDelay
	StereoDelay
Distortion	AmpSimulator
	Quadrafuzz v2
	SoftClipper
	VST Amp Rack
Dynamics	DeEsser
	EnvelopeShaper
	Maximizer
	MultibandCompressor
	MultibandEnvelopeShaper
	Tube Compressor
	VintageCompressor
EQ	GEQ-30
	StudioEQ
Modulation	Cloner
	Rotary
	StudioChorus
Reverb	REvelation
	REVerence
	Roomworks
	Roomworks SE
Tools	TestGenerator
Analyzer	SuperVision

## Yamaha



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

Mastering	[VCM] Vintage Open Deck
Dynamics	[VCM] Buss Comp 369
	[VCM] Compressor 260
	[VCM] Compressor 276
EQ	[VCM] EQ-1A
	[VCM] Equalizer 601
	Dynamic EQ
Reverb	REV-X

# Delay Plug-ins

## MonoDelay

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



### LO FILTER

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

### HI FILTER

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

### DELAY

Sets the delay time in milliseconds.

### SYNC

Activates/Deactivates tempo sync.

### FEEDBACK

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

### MIX

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

## PingPongDelay

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



This plug-in works only on stereo racks.



### LO FILTER

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

### HI FILTER

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

### DELAY

Sets the delay time in milliseconds.

### SYNC

Activates/Deactivates tempo sync.

### FEEDBACK

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

### MIX

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

### SPATIAL

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

### START LEFT/START RIGHT

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



## StereoDelay

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



This plug-in works only on stereo racks.



### FEEDBACK

Sets the number of repeats for each delay.

### DELAY

Sets the delay time in milliseconds.

### SYNC

Activates/Deactivates tempo sync for the corresponding delay.

### MIX

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

### LO FILTER

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

### PAN

Sets the stereo position.

### HI FILTER

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

# Distortion Plug-ins

## AmpSimulator

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



### Select Amplifier Model

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

### DRIVE

Controls the amount of amp overdrive.

### BASS

Tone control for the low frequencies.

### MID

Tone control for the mid frequencies.

### TREBLE

Tone control for the high frequencies.

### PRESENCE

Boosts or dampens the higher frequencies.

### VOLUME

Controls the overall output level.

### Select Cabinet Model

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

### DAMPING LOW/HIGH

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

## Quadrafuzz v2

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



### Frequency Band Editor

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

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

### Global Settings

#### SB

Switches between multi band and single band mode.

#### SCENES

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

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



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

## MIX

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

## OUT (-24 to 24 dB)

Sets the output level.

## Band Settings

### Mute

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

### Bypass Frequency Band

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

### Solo Frequency Band

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

### [IN/OUT] Meters

Display the input and output level.

## GATE

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

## TAPE

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

### DRIVE

Controls the amount of tape saturation.

### TAPE MODE DUAL

Simulates the use of two machines.

## TUBE

This band mode simulates the saturation effects using analog tubes.

### DRIVE

Controls the amount of tube saturation.

## **Tube(s)**

Determines the number of tubes that are simulated.

## **DIST**

This band mode adds distortion to your rack.

### **DRIVE**

Controls the amount of distortion.

### **FBK**

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

## **AMP**

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

### **DRIVE**

Controls the amount of amp overdrive.

### **Amp Types**

**You can select the following types of guitar amps**

- Amp Clean
- Amp Crunch
- Amp Lead

## **DEC**

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

### **DECIMATOR**

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

### **MODE**

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

### **S&H**

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

## **Delay**

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

### **TIME**

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

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

**SYNC**

Activates/Deactivates tempo sync for the corresponding delay.

**DUCK**

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

**MIX**

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

**FBK**

Sets the number of repeats for each delay.

**MODE**

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



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

**Slider****WIDTH**

Sets the stereo width for the corresponding band.

**OUT**

Sets the output gain for the corresponding band.

**PAN**

Sets the stereo position for the corresponding band.

**MIX**

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

## SoftClipper

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



### INPUT (-12 to 24 dB)

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

### MIX

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

### OUTPUT

Sets the output level.

### SECOND

Controls the second harmonic.

### THIRD

Controls the third harmonic.

## VST Amp Rack

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



At the top of the plug-in panel, there are seven buttons, arranged according to the position of the corresponding elements in the signal chain. Each button opens a different page in the display section of the plug-in panel ([Pre-Effects], [Amplifiers], [Cabinets], [Post-Effects], [Microphone], [Configuration], and [Master]).

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

### Pre-Effects/Post-Effects

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

#### Wah Wah

**Pedal** – Controls the filter frequency sweep.

#### Volume

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

#### Compressor

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

#### Limiter

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

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

#### Maximizer

**Amount** – Sets the loudness of the signal.

#### Chorus

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

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



### Phaser

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

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

### Flanger

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

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

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

### Tremolo

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

**Depth** – Governs the depth of the amplitude modulation.

### Octaver

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

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

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

### Delay

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

**Feedback** – Sets the number of repeats for the delay.

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

### Tape Delay

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

**Feedback** – Sets the number of repeats for the delay.

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

### Tape Ducking Delay

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

**Feedback** – Sets the number of repeats for the delay.

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

### Overdrive

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

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

**Level** – Adjusts the output level.

## Fuzz

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

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

**Level** – Adjusts the output level.

## Gate

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

**Release** – Sets the time after which the gate closes.

## Equalizer

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

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

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

## Reverb

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

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

## Sync Mode

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

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



## Using Effects

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



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

## Amplifiers

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

### Plexi

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

### Plexi Lead

British rock tone of the 70s and 80s.

### Diamond

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

### Blackface

Classic American clean tone.

### Tweed

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

### Deluxe

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

### British Custom

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

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

### Selecting and Deactivating Amplifiers

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

## Cabinets

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

### Selecting and Deactivating Cabinets

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

## Microphones

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

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

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

between the two microphones.

## Placing the Microphone

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

## Configuration

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

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



In stereo mode, the effect requires more processing power.

## Master

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

### Input/Output Level Meters

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

### Using the Master Controls

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

## View Settings

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

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

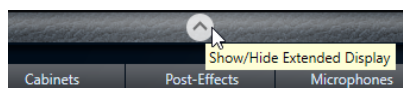
In the compact view, the display section is hidden from view. You can change the amp settings and switch amps or cabinets using the mouse wheel.

### Using the Smart Controls

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

## Switching between Default and Compact View

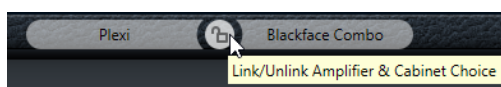
- To toggle between the different views, click the down/up arrow button ([Show/Hide Extended Display]) at the top center of the plug-in frame.



## Changing the Amplifier and Cabinet Selection in the Compact View

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

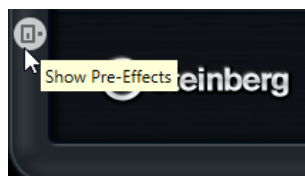
- To select a different amplifier or cabinet, click the name and select a different model from the pop-up menu.
- To lock the amplifier and cabinet combination, activate the [Link/Unlink Amplifier & Cabinet Choice] button. If you now select another amp model, the cabinet selection follows. However, if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



## Previewing Effect Settings

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

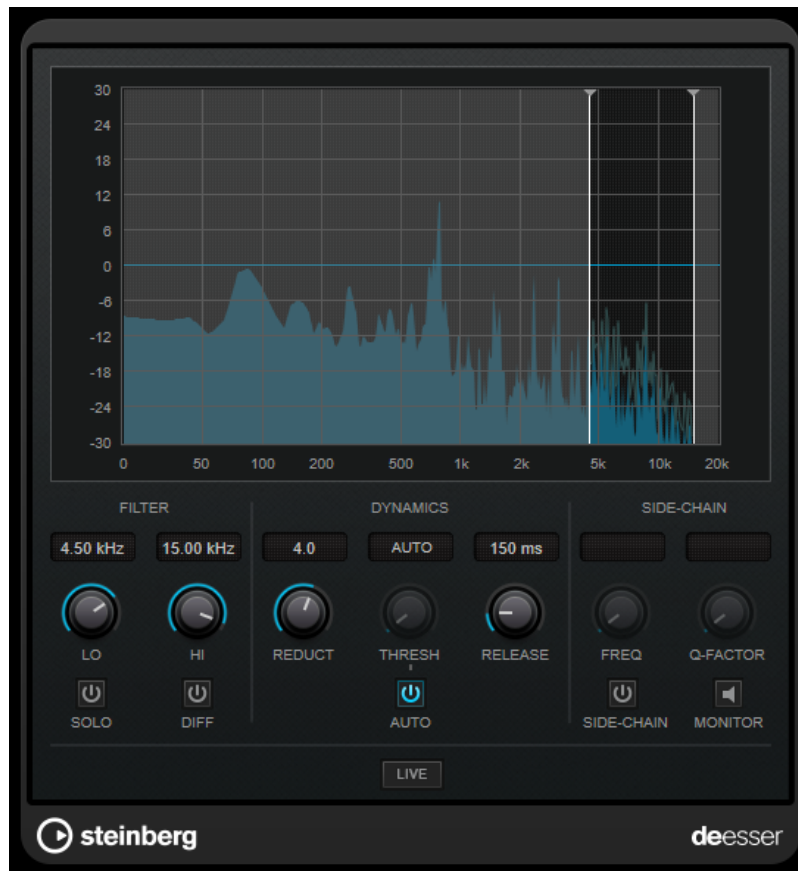
- Click and hold the [Show Pre-Effects] button at the bottom left or [Show Post-Effects] button at the bottom right of the plug-in frame.



# Dynamics Plug-ins

## DeEsser

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



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

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

## Display

Shows the spectrum of the input signal.

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

## FILTER

### LO/HI

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



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

## SOLO

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

## DIFF

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

## DYNAMICS

### REDUCT

Controls the intensity of the de-essing effect.

### THRESH (-50 to 0 dB)

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

### RELEASE (1 to 1000 ms)

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

### AUTO

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

## SIDE-CHAIN

### FREQ (25 Hz to 20 kHz)

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



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

### SIDE-CHAIN

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

### Q-FACTOR

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

### MONITOR

Allows you to monitor the filtered signal.

### LIVE

If this button is activated, the look-ahead feature of the effect is deactivated. Look-ahead

produces more accurate processing, but adds a specific amount of latency as a trade-off. If [Live] mode is activated, there is no latency, which is better for live processing.

## **Positioning the DeEsser in the Signal Chain**

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



## EnvelopeShaper

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

You can use the knobs or drag the breakpoints in the graphical display to change parameter values. Be careful with levels when boosting the gain and if needed reduce the output level to avoid clipping.



### **ATTACK (-20 to 20 dB)**

Sets the gain of the attack phase of the signal.

### **LENGTH (5 to 200 ms)**

Sets the length of the attack phase.

### **RELEASE**

Sets the gain of the release phase of the signal.

### **OUTPUT**

Sets the output level.

## Maximizer

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



### CLASSIC

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

### MODERN

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

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

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

### OPTIMIZE

Determines the loudness of the signal.

### MIX

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

### OUTPUT

Sets the maximum output level.

### SOFT CLIP

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

## MultibandCompressor

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



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

## Frequency Band Editor

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


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

### LIVE

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

processing.

### Bypass Frequency Band

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

### Solo Frequency Band

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

### OUTPUT (-24 to 24 dB)

Sets the output level.

## [COMPRESSOR] Section

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

### THRESH (-60 to 0 dB)

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

### RATIO

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

### ATTACK (0.1 to 100 ms)

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

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

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

## SC (Side-chain Section)

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

### Important

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



### SIDE-CH

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

**FREQ**

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

**Q-FACTOR**

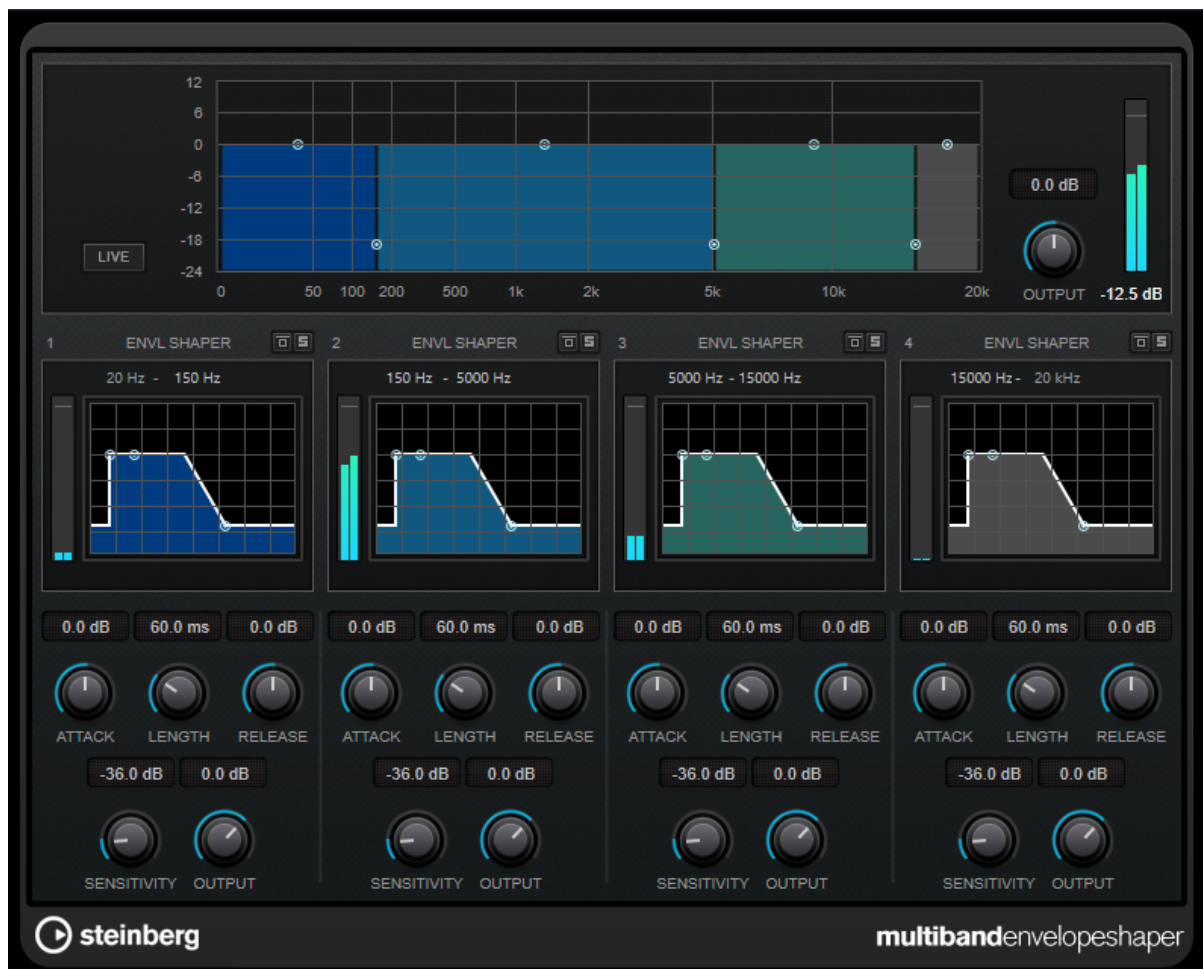
If [Side-Chain] is activated, this sets the resonance or width of the filter.

**MONITOR**

Allows you to monitor the filtered signal.

## MultibandEnvelopeShaper

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



### Frequency Band Editor


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

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

#### LIVE

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

#### Bypass Frequency Band

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

#### Solo Frequency Band

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

### **OUTPUT (-24 to 24 dB)**

Sets the output level.

## **Shaper Section**

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

### **ATTACK (-20 to 20 dB)**

Sets the gain of the attack phase of the signal.

### **LENGTH (5 to 200 ms)**

Sets the length of the attack phase.

### **RELEASE**

Sets the gain of the release phase of the signal.

### **SENSITIVITY (-40 to -10 dB)**

Sets the sensitivity of the detection.

### **OUTPUT**

Sets the output level.

## Tube Compressor

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

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



### VU Meter

Shows the amount of gain reduction.

### [IN/OUT] Meters

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

### INPUT

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

### DRIVE (1.0 to 6.0 dB)

Controls the amount of tube saturation.

### OUTPUT (-12 to 12 dB)

Sets the output gain.

### CHARACTER

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

### ATTACK (0.1 to 100 ms)

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

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

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



## **MIX**

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

## **RATIO**

Toggles between a low and a high ratio value.

## **SC(Side-Chain)**

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

## **Side-Chain Section**

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

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

### **Center (50 to 20000 Hz)**

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

### **Q-FACTOR**

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

### **MONITOR**

Allows you to monitor the filtered signal.

## VintageCompressor

VintageCompressor is modeled after vintage type compressors.

This compressor features separate controls for [Input] and [Output] gain, [Attack], and [Release]. In addition, there is a [Punch] mode which preserves the attack phase of the signal and a program-dependent [Auto Release] function.



### VU Meter

Shows the amount of gain reduction.

### [IN/OUT] Meters

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

### INPUT

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

### ATTACK (0.1 to 100 ms)

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

### [PUNCH] Button

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

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

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

### MIX

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

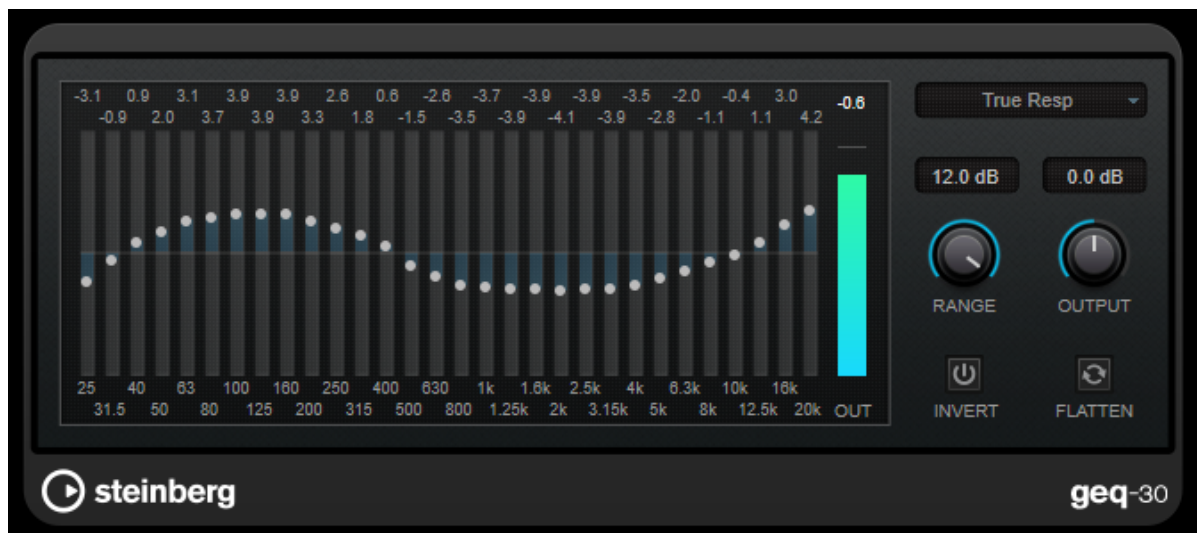
### OUTPUT (-48 to 24 dB)

Sets the output gain.

# EQ Plug-ins

## GEQ-30

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



Each band can be attenuated or boosted by up to 12 dB, allowing for fine control of the frequency response. In addition, there are several preset modes available that can add color to the sound of GEQ-30.

You can draw response curves in the main display by clicking and dragging with the mouse. You have to click one of the sliders before you drag across the display.

At the bottom of the window, the individual frequency bands are shown in Hz. At the top of the display, the amount of attenuation/boost is shown in dB.

### RANGE

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

### OUTPUT

Sets the overall gain of the equalizer.

### INVERT

Inverts the current response curve.

### FLATTEN

Resets all the frequency bands to 0 dB.

## EQ Modes

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

### True Response

Applies serial filters with an accurate frequency response.

### Digital Standard

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

### Classic

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

accurately.

**VariableQ**

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

**ConstQ asym**

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

**ConstQ sym**

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

**Resonant**

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

## StudioEQ

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



### Main Layout

#### RESET

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

#### SPECTRUM

Shows the spectrum before and after filtering.

#### OUTPUT

Adjusts the overall output level.

#### AUTO GAIN

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

### Band Settings



#### Activate/Deactivate Band

Activates/Deactivates the corresponding band.



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

## FREQ

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



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

## INV

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

## Q

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



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

## GAIN

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



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

## Filter type

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

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

# Modulation Plug-ins

## Cloner

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



### Graphical Display

#### VOICES

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

#### [DETUNE] Slider (1 to 4)

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

#### [DELAY] Slider (1 to 4)

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

#### DETUNE

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

#### NATURAL

Changes the pitch algorithm that is used for detune.

#### HUMANIZE (Detune)

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

#### STATIC (Detune)

Activate this button to use a static amount of detune.

#### MIX

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

**SPATIAL**

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

**OUTPUT (-12 to 12 dB)**

Sets the output gain.

**DELAY**

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

**HUMANIZE (Delay)**

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

**STATIC (Delay)**

Activate this button to use a static delay amount.



## Rotary

This modulation effect simulates the effect of a rotating speaker.



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

### Speed settings

#### SPEED MOD Control (MIDI)

#### Speed Selector (stop/slow/fast)

Allows you to control the speed of the rotary speaker.

#### SPEED MOD

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

#### Set Speed Change Mode

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

### Additional settings

#### OVERDRIVE

Applies a soft overdrive or distortion.

#### CROSSOVER

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

### Horn

#### SLOW

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

#### FAST

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

**ACCEL.**

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

**AMP MOD**

Controls the high rotor amplitude modulation.

**FREQ MOD**

Controls the high rotor frequency modulation.

**Bass****SLOW**

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

**FAST**

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

**ACCEL.**

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

**AMP MOD**

Adjusts the modulation depth of the amplitude.

**LEVEL**

Adjusts the overall bass level.

**Mics****PHASE**

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

**ANGLE**

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

**DISTANCE**

Sets the simulated microphone distance from the speaker in inches.

**Final Settings****OUTPUT**

Sets the output level.

**MIX**

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

## StudioChorus

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



### DELAY

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

### WIDTH

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

### SPATIAL

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

### MIX

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

### RATE

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

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

### SYNC

Activates/Deactivates tempo sync.

### Waveform Shape

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

---

**LO FILTER/HI FILTER**

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

# Reverb Plug-ins

## REVELation

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



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

### PRE DELAY

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

### EARLY REFLECTION

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

### ER/TAIL

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

### SIZE

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

### LOW CUT

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

**HIGH CUT**

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

**DELAY**

Delays the onset of the reverb tail.

**ROOM SIZE**

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

**MAIN TIME**

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

**HIGH TIME**

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

**LOW TIME**

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

**HIGH FREQ**

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

**LOW FREQ**

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

**SHAPE**

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

**DENSITY**

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

**HIGH CUT**

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

**WIDTH**

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

## **MIX**

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

### **Lock Mix Value**

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

## **Modulation**

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

### **Modulation Rate**

Specifies the frequency of the pitch modulation.

### **Modulation Depth**

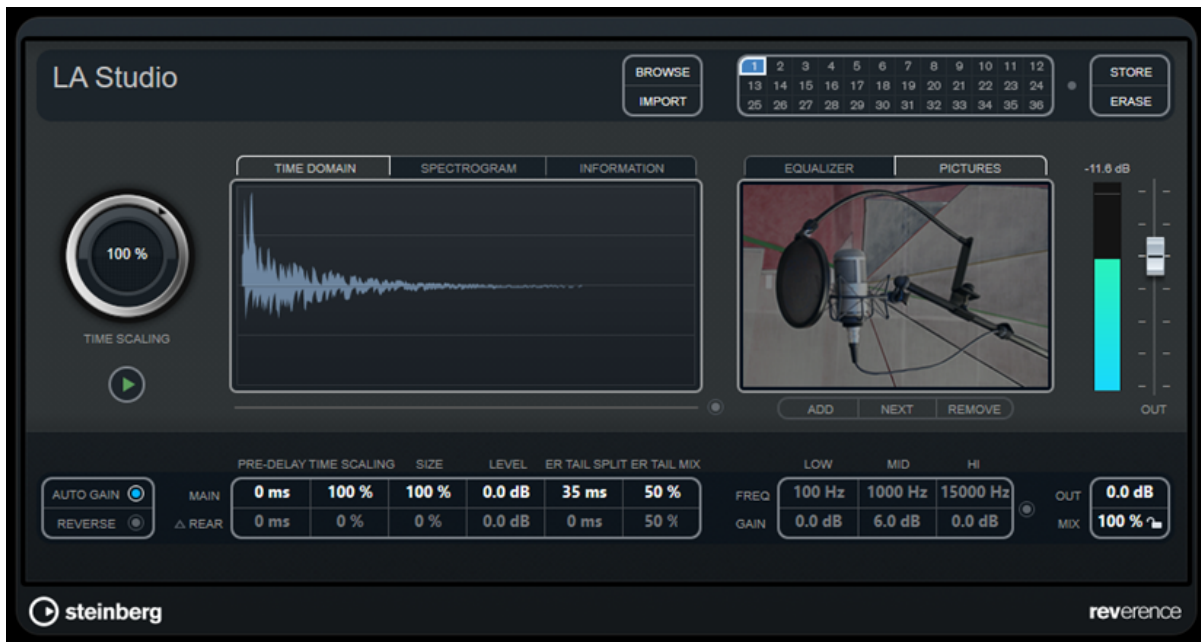
Sets the intensity of the pitch modulation.

### **Modulation Activate**

Activates/Deactivates the chorusing effect.

## REVerence

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



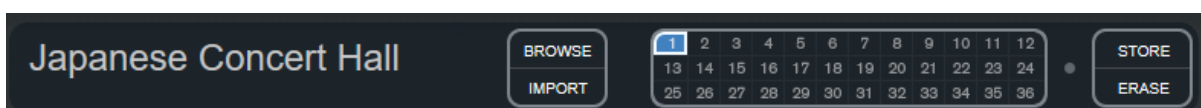
This is done by processing the audio signal according to an impulse response – a recording of an impulse in a room or another location that recreates the characteristics of the room. As a result, the processed audio sounds as if it were played in the same location. Included with the plug-in are top quality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



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

## Program Matrix

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



### Program Name

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

### BROWSE

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

### IMPORT

Click this button to load impulse response files from an external source. The files should have a



maximum length of 10 seconds. Longer files are automatically cut.

### Program Slots

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

### Smooth Parameter Changes

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

### STORE

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

### ERASE

Removes the selected program from the matrix.

## Programs vs. Presets

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

### Preset



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



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

### Program



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

## Presets

Presets are useful in the following situations:

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

## Programs

Programs offer the following advantages:

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

## Setting up Programs

### Procedure

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



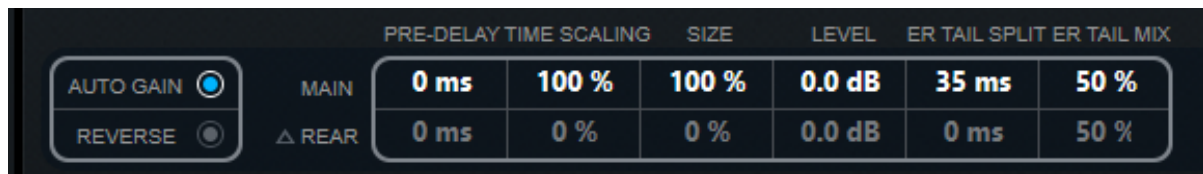
If you want to use your set of programs in other projects, save your settings as a plug-in preset.

### Related Links

[Importing Impulse Responses](#)

## Reverb Settings

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



### MAIN

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

### REAR

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

### AUTO GAIN

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

### REVERSE

Reverses the impulse response.

### PRE-DELAY

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

### TIME SCALING

Controls the reverb time.

### SIZE

Determines the size of the simulated room.

### LEVEL

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

### ER TAIL SPLIT

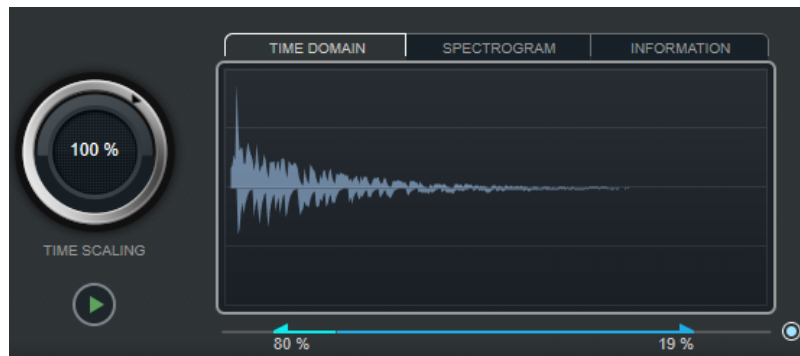
Sets a split point between the early reflections and the tail, allowing you to determine where the reverb tail begins. A value of 60 means that the split point is set to 60 ms.

### ER TAIL MIX

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

## Impulse Response Display

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



## TIME SCALING

This wheel lets you adjust the reverb time.

## Play



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

## TIME DOMAIN

This display shows the waveform of the impulse response.

## SPECTROGRAM

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

## INFORMATION

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

## Activate Impulse Trimming

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

## Trim

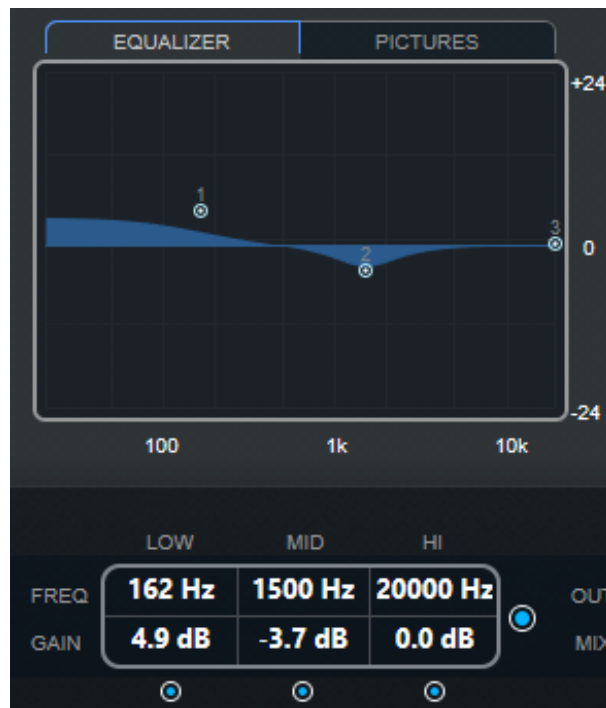
This slider allows you to trim the start and end of the impulse response. Drag the front handle to trim the start of the impulse response, and the end handle to trim the reverb tail.



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)

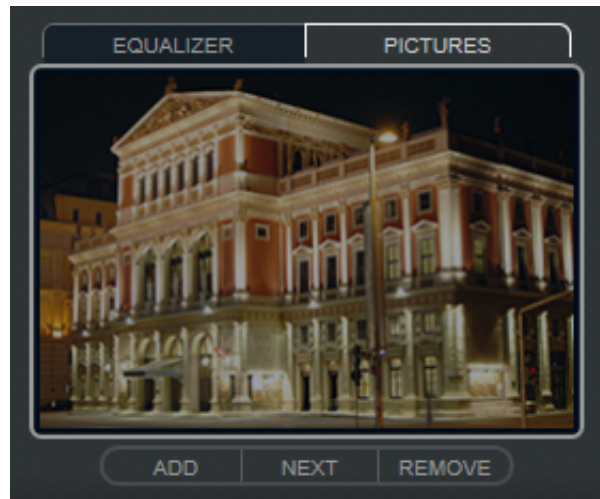
Sets the frequency of the high band.

## 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.



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

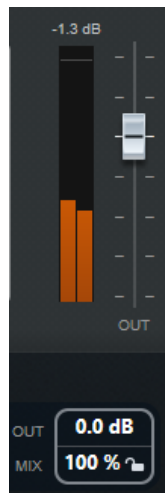
Deletes the active picture.



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

## Output Settings

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



### Output Meter

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

### OUT

Adjusts the overall output level.

### MIX

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

### Lock Mix Value

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

## Custom Impulse Responses

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

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

## Importing Impulse Responses

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

### Procedure

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

5. Make the appropriate settings and add a picture, if available. Pictures residing in the same folder as the impulse response file or in the parent folder are automatically found and displayed.
6. Click the [Store] button to save the impulse response and its settings as a program. That way you can recall the setup at any time.

## Result

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



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

## After Completing This Task

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

## True Stereo

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

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

The channels are defined as follows:

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

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

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

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

## Relocating Content

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

The factory content is not a problem because it is also present on the other computer. For these



impulse responses, you just need to transfer your REVerence programs and presets to be able to access your setups.

User content is a different matter, though. If you have transferred your audio files to an external drive or a different hard disk location on the other computer, REVerence cannot access the impulse responses any more because the old file paths have become invalid.

### **Procedure**

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

### **Result**

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

### **Important**

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

## Roomworks

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



## Input Filters

### LOW FREQ

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

### HIGH FREQ

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

### LOW GAIN

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

### HIGH GAIN

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

## Reverb Character

### PRE-DELAY

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

### SIZE

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

### REVERB TIME

Allows you to set the reverb time in seconds.

### DIFFUSION

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

### WIDTH

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

## VARIATION

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

## HOLD

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

## Damping

### LOW FREQ

Determines the frequency below which low-frequency damping occurs.

### HIGH FREQ

Determines the frequency above which high-frequency damping occurs.

### LOW LEVEL

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

### HIGH LEVEL

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

## Envelope

### AMOUNT

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

### ATTACK

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

### RELEASE

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

## OUTPUT

### MIX

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

### WET ONLY

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

**EFFICIENCY**

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

**EXPORT**

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

**Output Meter**

Shows the level of the output signal.

## Roomworks SE

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



### PRE-DELAY

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

### REVERB TIME

Allows you to set the reverb time in seconds.

### DIFFUSION

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

### LOW LEVEL

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

### HIGH LEVEL

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

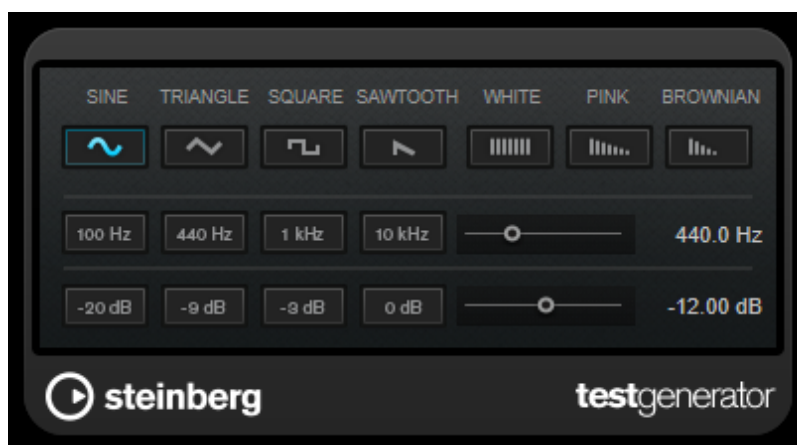
### MIX

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

# Tools Plug-ins

## Test Generator

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



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

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

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

### Waveform and Noise Section

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

### Frequency Section

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



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

### Gain Section

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

# Analyzer Plug-ins

## SuperVision

SuperVision is a professional tool suite for monitoring and analyzing your audio. The plug-in comes with several different modules for level, spectral, phase, or waveform analysis. Its up to 9 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. Click this button while holding [Alt/Opt] to pause/continue all modules at the same time.



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.



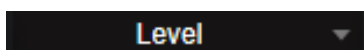
For all modules that show a playback cursor, you can click in the paused display to set the project cursor to this position.

#### Hold Current Values on Stop



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

#### Module Selector



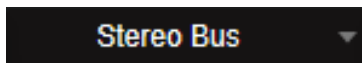
Allows you to select a module for the selected slot.

## 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.



The channel selector is only available for configurations with two or more channels. NOTE: If side-chain is activated, you can select between [Main] or [Side-Chain] channels. Combined [Main & Side-Chain] views are also available for some modules.

## Reset Module Values



Resets the measured values of the selected module. While holding [Alt/Opt], click to reset all modules at the same time.



You can also reset the measured values of a module by clicking them while holding [Ctrl]/[command].

## Reset Module Values on Start



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

## Split Horizontally



Splits the selected module slot horizontally.



This button is not available if a module is maximized.

## Split Vertically

Splits the selected module slot vertically.



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.

In some modules, for example, [Level], [Loudness], or [Time], you can use [Ctrl]/[command] + [S] to copy parameter values as text from the selected module to the clipboard for further use in other applications.

# Mastering Plug-ins

## [VCM] Vintage Open Deck

Vintage Open Deck emulates the analog circuitry and tape characteristics of a legendary open reel tape recorder.

This plug-in consists of a recording deck and a reproduction deck, and there are four variations available for each.

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



### Selecting a Deck

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

### Recording Deck Controls

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

#### RECORD

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

#### AUTO MAKEUP

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

### **ADJUST-HIGH**

Adjusts the high-frequency gain of the recording deck.

### **BIAS**

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

## **Reproduction Deck Controls**

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

### **REPRODUCE**

Adjusts the output level of the reproduction deck.

### **ADJUST-HIGH**

Adjusts the high-frequency gain of the reproduction deck.

### **ADJUST-LOW**

Adjusts the low-frequency gain of the reproduction deck.

## **Deck Meters**

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

## **Adjusting the VU Meter Level**

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

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

## **Tape Parameters**

The tape parameter controls are at the bottom of the plug-in.

### **SPEED**

Allows you to set the tape speed. This parameter changes the sound characteristics of the tape head saturation. 30 ips (inches per second) provides better high-frequency response compared to 15 ips. 15 ips provides a rich low-frequency response.

### **TAPE KIND**

This allows you to change the sound characteristics of the tape saturation.

# Dynamics Plug-ins

## [VCM] Buss Comp 369

Buss Comp 369 emulates the bus compressor that has been used as the standard in recording studios and by broadcasters since the 1980s. In contrast to Compressor 276, which excels at creating aggressive compression sounds, it features smooth, natural compression that does not impair the expressiveness of the source.

It has both a built-in compressor and limiter, which can be used separately or in combination depending on the application. In addition, the rich harmonics of the input/output circuit audio transformer and discrete class A amplifier add depth and cohesiveness to the sound quality.

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

In stereo mode



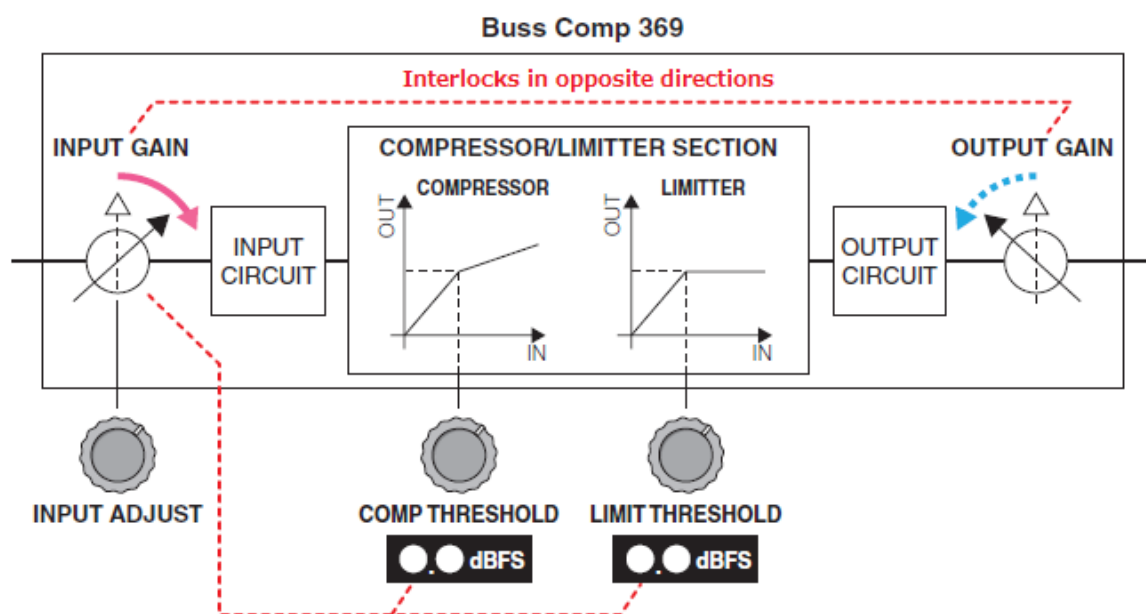
In mono mode



## [INPUT ADJUST] Knob

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

### How INPUT ADJUST Works



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

## [LINK] Switch (in stereo mode)

Activates/Deactivates the stereo link.

## COMP [IN] Switch

Activates/Deactivates the compressor.

## COMP [THRESHOLD] Knob

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

## COMP [RECOVERY] Knob

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

**a1:** Changes automatically between 100 ms and 2 sec.

**a2:** Changes automatically between 50 ms and 5 sec.

## COMP [GAIN] Knob

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

## COMP [RATIO] Knob

Adjust the ratio of the compressor.

## [METER] Select Switch (in stereo mode)

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

**[VU] Select Switch (in mono mode)**

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

**LIMIT [IN] Switch**

Activates/Deactivates the limiter.

**LIMIT [ATTACK] Switch**

Sets the attack time of the limiter.

FAST: 2 ms

SLOW: 4 ms

**LIMIT [THRESHOLD] Knob**

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

**LIMIT [RECOVERY] Knob**

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

**a1:** Changes automatically between 100 ms and 2 sec.

**a2:** Changes automatically between 50 ms and 5 sec.

## [VCM] Compressor 260

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

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



### [THRESHOLD] Knob

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

### [KNEE] Switch

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

### [ATTACK] Knob

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

### [RELEASE] Knob

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

### [RATIO] Knob

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

### [OUTPUT] Knob

Adjusts the output gain.

### [GR] Meter

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

### [OUT] Meter

Displays the output level of the effect.

## [VCM] Compressor 276

Compressor 276 emulates the characteristics of an analog compressor that is a staple in recording studios. You can obtain thick, strong sounds that are suitable for drums and bass parts. This plug-in operates in mono or stereo mode, depending on the channel configuration of the rack.



### [INPUT] Knob

Adjusts the input level.

### [RATIO] Knob

Adjust the ratio of the compressor.

### [ATTACK] Knob

Adjusts the attack time of the compressor.

### [OUTPUT] Knob

Adjusts the output gain.

### [AUTO MAKEUP] Switch

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

### [INTERNAL SC HPF] Switch

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

### [RELEASE] Knob

Adjusts the release time of the compressor.

### [VU] Meter

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

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

Selects the signal that is monitored by the VU meter.

**GR:** Displays the amount of gain reduction.

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

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

**OFF:** Deactivates the VU meter.



# EQ Plug-ins

## [VCM] EQ-1A

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



### [IN] Switch

Activates/Deactivates the processor.

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

### [LOW FREQUENCY] Knob

Adjusts the frequency of the low frequency filter.

### (LOW) [BOOST] Knob

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

### (LOW) [ATTEN] Knob

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

### [BAND WIDTH] Knob

Sets the bandwidth to operate with the high frequency filter.

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

### [HIGH FREQUENCY] Knob

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

### (HIGH) [BOOST] Knob

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

### (HIGH) [ATTEN] Knob

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

**(HIGH) [ATTEN SEL] Knob**

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

## [VCM] Equalizer 601

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

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



### Frequency Response Curve Graph

The frequency response curve for all bands and the frequency response curve for each band are displayed.

You can drag your mouse on the graph to change the frequency and gain of the filter.

### [+]/[-] Buttons

Enlarge or reduces the vertical axis of the graph display.

### [TYPE] Switch

Selects one of the two equalizer types with different effects.

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

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

### [INPUT] Knob

Adjusts the input level.

### Level Meter

Displays the output level of the effect.

### [OUTPUT] Knob

Adjusts the output gain.

### [FLAT] Button

Resets the gain of all bands to 0 dB.

**[Q/TYPE] Knob**

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

**[F] (Frequency) Knob**

Center frequency of the filter.

**[G] (Gain) Knob**

Gain value of the filter.

**[ON] Button**

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

## Dynamic EQ

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

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



### [BAND ON/OFF] Button




Activates/Deactivates the frequency band.

### SIDECHAIN [LISTEN] Button

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

### [FILTER TYPE] Buttons

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

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

**[FREQUENCY] Knob**

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

**[Q] Knob**

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

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

**[THRESHOLD] Knob**

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

**[RATIO] Knob**

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

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

**[ATTACK/RELEASE] Buttons**

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

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

**[MODE] Button**

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

**EQ GAIN Meter**

Displays the dynamically changing EQ gain.

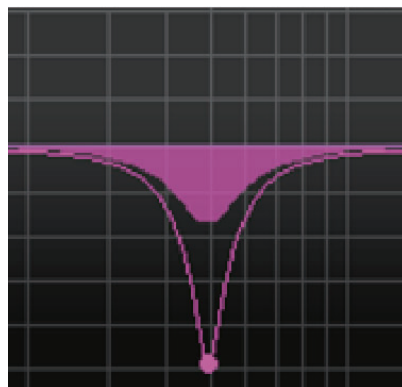
**THRESHOLD Meter**

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

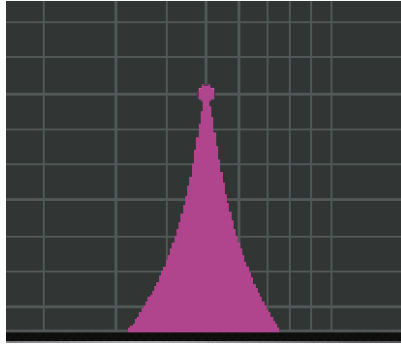
**Graphical Display**

Displays the characteristics of the equalizer.

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



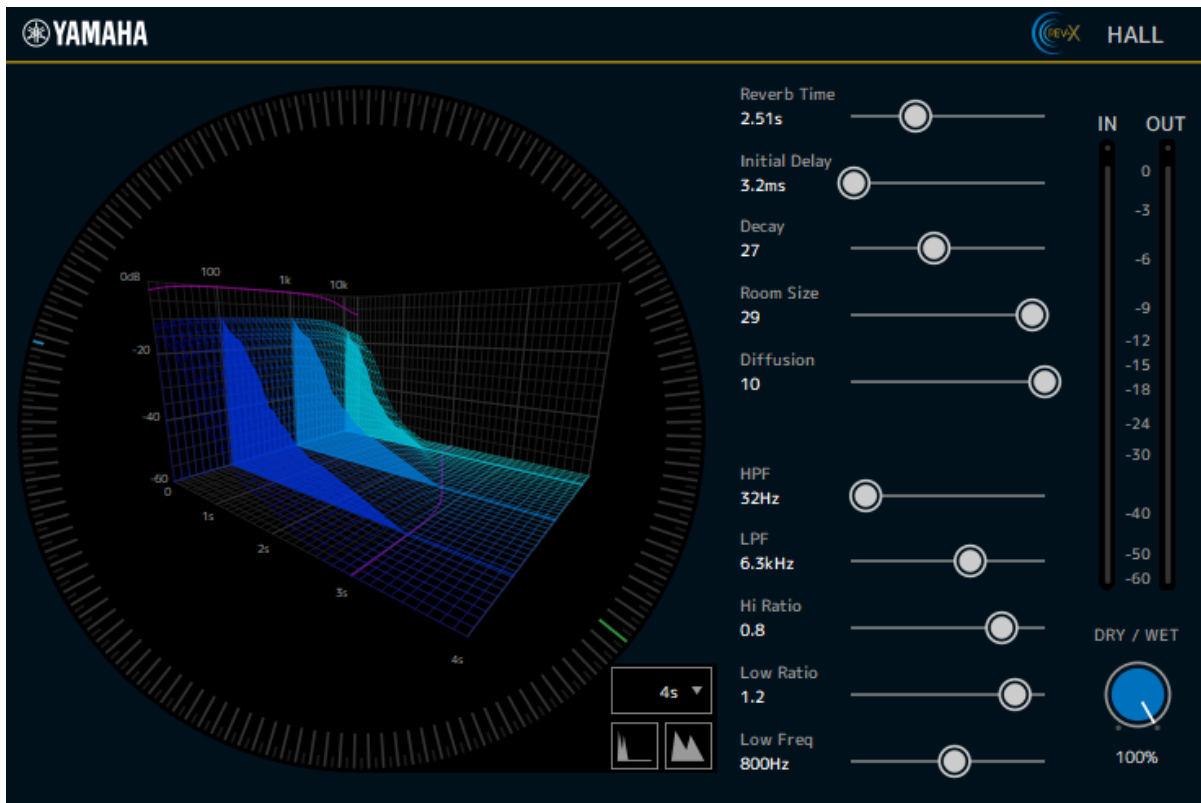
If SIDECHAIN [LISTEN] is activated, the characteristics of the side-chain filter are displayed.



# Reverb Plug-ins

## REV-X

REV-X is a reverb algorithm with features such as sound quality with high-density and rich reverberation, smooth attenuation, and breadth and depth that make the best use of the original sound. You can select from three types of reverb according to the sound field and purpose: REV-X HALL, REV-X ROOM, and REV-X PLATE.



### EFFECT TYPE

Selects the effect type.

### Reverb Time

This is the time required for the reverberation to decay and disappear. The higher the value, the longer the reverberation will last.

### Initial Delay

This is the delay from the input of the original sound to the start of reverberation. The higher the value, the later the occurrence of reverberation.

### Decay

This is the envelope shape of the reverberation. The characteristics of the reverberation will change depending on this value.

### Room Size

This is the size of the room. The higher the value, the larger the simulated space. This value is linked with [Reverb Time]. Changing this value will also change [Reverb Time].

### Diffusion

This is the density and spread of reverberation. The higher the value, the higher the density and the stronger the sense of spaciousness.



## HPF

This is a filter that cuts the low-frequency components of reverberation. Components that are below the frequency specified by this value are cut. This filter has no effect on the original sound.

## LPF

This is a filter that cuts the high-frequency components of reverberation. Components that are above the frequency specified by this value are cut. This filter has no effect on the original sound.

## Hi Ratio

This is the length of the high-frequency reverberation. The high-frequency reverberation time is specified as a ratio with [Reverb Time].

## Low Ratio

This is the length of the low-frequency reverberation. The low-frequency reverberation time is specified as a ratio with [Reverb Time].

## Low Freq

This is the reference frequency for [Low Ratio]. Frequency bands below this value are affected by [Low Ratio].

## Filter Frequency Response Curve

The curve changes depending on the HPF and LPF values.

## Reverberation Image

This is an image of the reverberation of the high band (10 kHz), mid band (1 kHz), and low band (100 Hz). The image changes depending on the value of each parameter. The vertical axis represents the level, the horizontal axis represents the reverberation time, and the shape represents the envelope.

## Reverberation Time Curve

This curve displays the reverberation time of the high band (10 kHz), mid band (1 kHz), and low band (100 Hz).

The curve changes depending on the values for [Reverb Time], [Hi Ratio], and [Low Ratio].

## Zoom

Displays the length of time allocated to the time axis (horizontal axis of the graph) in seconds.

## Zoom Out Button

Click this button to increase the number of seconds allocated to the time axis (horizontal axis of the graph). As a result, the width of the graph is reduced.

## Zoom In Button

Click this button to decrease the number of seconds allocated to the time axis (horizontal axis of the graph). As a result, the width of the graph is increased.

## MIX Slider

Adjusts the balance between the original sound and the effect. If this value is 0%, only the original sound is output. If it is 100%, only the effect is output.

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