

Plug-in Reference



# WAVELAB CAST

For Podcast and Social Media Content Creation

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# WaveLab-specific Plug-ins

WaveLab-specific plug-ins use the plug-in format of WaveLab, and cannot be used with other applications.

## Resampler

This plug-in is a professional sample rate converter providing exceptional transparency and preservation of the frequency content. It is only available in the **Master Section**.

### NOTE

This plug-in is very CPU consuming, especially in high quality modes.

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Output Sample Rate 44 100 Hz ▼

### Output Sample Rate

Defines the output sample rate while the input sample rate is determined by the sample rate of the active audio file or audio montage.

# Steinberg VST 3 Plug-ins

In WaveLab there is no limitation to the use of VST plug-ins. They can be used wherever plug-ins can be inserted.

- VST plug-ins have their own preset handling. You can save or load effect programs (presets).

## Brickwall Limiter

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



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

### Threshold (-20 to 0 dB)

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

### Release (3 to 1000 ms or Auto mode)

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

### Link

If this button is activated, **Brickwall Limiter** uses the channel with the highest level to analyze the input signal. If the button is deactivated, each channel is analyzed separately.

### Detect Intersample Clipping

If this option is activated, **Brickwall Limiter** uses oversampling to detect and limit signal levels between two samples to prevent distortion when converting digital signals into analog signals.

#### NOTE

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

## Compressor

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



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

### Threshold (-60 to 0 dB)

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

### Ratio

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

### Soft Knee

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

### High Ratio

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

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

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

### Dry Mix

Mixes the dry input signal to the compressed signal.

### Attack (0.1 to 100 ms)

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

### Hold (0 to 5000 ms)

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

### Release (10 to 1000 ms or Auto mode)

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

### Analysis (Pure Peak to Pure RMS)

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

### Live

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

## Gate

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



### Attack (0.1 to 1000 ms)

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

#### NOTE

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

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#### Hold (0 to 2000 ms)

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

#### Release (10 to 1000 ms or Auto mode)

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

#### Threshold

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

#### State LED

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

#### Analysis (Pure Peak to Pure RMS)

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

#### Range

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

#### Live

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

### Side-Chain Section

#### Side-Chain

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

#### Monitor

Allows you to monitor the filtered signal.

#### Center

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

#### Q-Factor

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



### Filter Type (Low-Pass/Band-Pass/High-Pass)

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

## Limiter

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



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

### Input (-24 to 24 dB)

Sets the input gain.

### Release (0.1 to 1000 ms or Auto mode)

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

### Output

Sets the maximum output level.

## RoomWorks SE

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



### Pre-Delay

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

### Reverb Time

Allows you to set the reverb time in seconds.

### Diffusion

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

### Low Level

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

### High Level

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

### Mix

Sets the level balance between the dry signal and the wet signal. When using **RoomWorks SE** inserted in an FX channel, you most likely want to set this to 100 %.

## StereoDelay

### NOTE

This plug-in works only on stereo tracks.

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### Feedback

Set the number of repeats for each delay.

### Delay

Sets the delay time in milliseconds.

### 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.

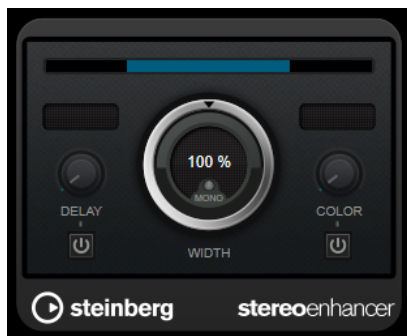
## StereoEnhancer

**StereoEnhancer** expands the stereo width of (stereo) audio material. It cannot be used with mono files.

### NOTE

This plug-in works only on stereo tracks.

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### Delay

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

### Width

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

### Mono

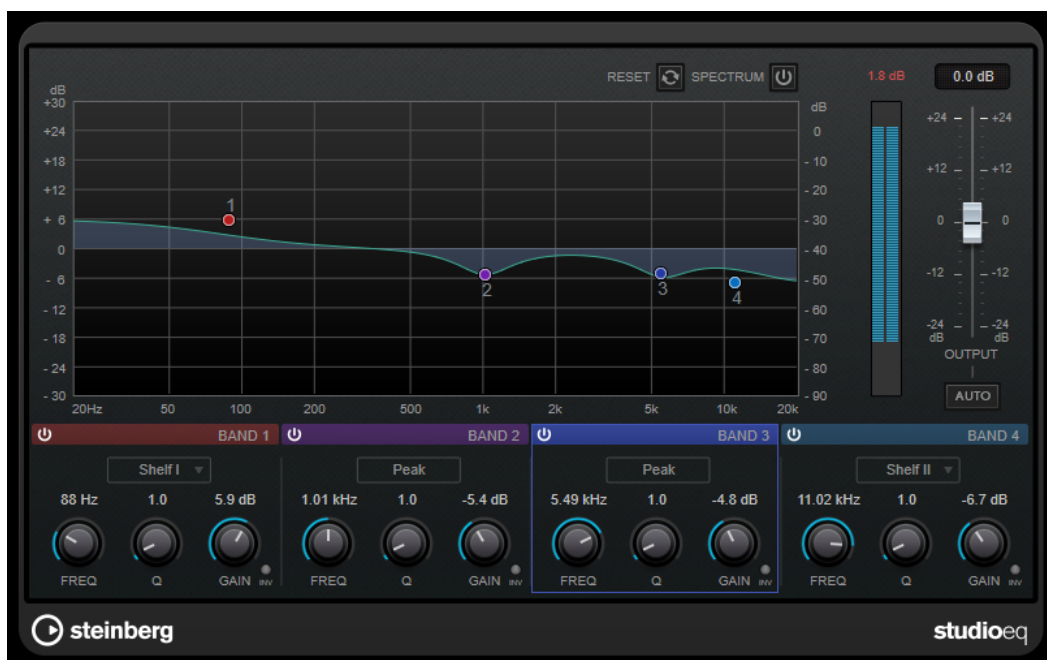
Switches the output to mono, to check for possible unwanted coloring of the sound which sometimes can occur when enhancing the stereo image.

### Color

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

## 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

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

### Show Input/Output Spectrum

Shows the spectrum before and after filtering.

### Output

Adjusts the overall output level.

### Auto Gain

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

## Band Settings



### Activate/Deactivate Band

Activates/Deactivates the corresponding band.

#### NOTE

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

### Freq

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

#### NOTE

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

### Inv

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

### Q

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

NOTE

- You can adjust the **Q** parameter of a band in the graphical editor by **l**-clicking the corresponding handle and moving the mouse up and down. Alternatively, you can point on the handle and turn the mouse wheel.

**Gain**

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

NOTE

- You can adjust the **Gain** parameter of a band in the graphical editor by **l**-clicking the corresponding handle 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 **II**.

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