

Plug-in Reference



# WAVELAB ELEMENTS 12

Personal Audio Editing System

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# VST 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.

- You can specify which VST plug-ins should be available in the **Effects** pane and **Final Processing/Dithering** pane of the **Master Section** in the the **Organize** tab of the **Plug-ins Preferences**.
- VST plug-ins have their own preset handling. You can save or load effect programs (presets).

## Delay

### MonoDelay

This is a mono delay effect. The delay line uses freely specified delay time settings.



#### Lo Filter

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

#### Hi Filter

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

#### Delay

Sets the delay time in milliseconds.

#### 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.

### NOTE

This plug-in works only on stereo tracks.

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### Lo Filter

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

### Hi Filter

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

### Delay

Sets the delay time in milliseconds.

### 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

**StereoDelay** has two independent delay lines with freely specified delay time settings.

### 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 control 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 control activates/deactivates the filter.

# Distortion

## Distortion

**Distortion** adds crunch to your tracks.



### Boost

Increases the distortion amount.

### Oversampling

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

#### NOTE

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

### Mix

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

### Tone

Changes the tonal characteristic of the output signal.

### Feedback

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

### Spatial

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

### Output

Sets the output level.

## Dynamics

### Brickwall Limiter

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



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

#### Threshold

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

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

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

#### Link Channels

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

#### Detect Intersample Clipping

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

#### NOTE

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

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## 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

Determines the level where the compressor kicks in.

### Ratio

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

### 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.

## DeEsser

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



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.

#### NOTE

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

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### 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

### Reduction

Controls the intensity of the de-essing effect.

### Threshold (-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

### 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.

### 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.

#### NOTE

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

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### 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 located 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 controls or drag the breakpoints on 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.

## Expander

**Expander** reduces the output level in relation to the input level for signals below the set threshold. This is useful if you want to enhance the dynamic range or reduce the noise in quiet passages.

You can either use the controls or drag the breakpoints on the graphical display to change the **Threshold** and the **Ratio** parameter values.



### Threshold

Determines the level where the expansion kicks in.

### Ratio

Sets the amount of gain boost applied to signals below the threshold.

### Soft Knee

If this button is deactivated, signals below the threshold are expanded instantly according to the set ratio (hard knee). If **Soft Knee** is activated, the onset of expansion is more gradual, producing less drastic results.

### Fall (0.1 to 100 ms)

Determines how fast the expander responds to signals below the set threshold. If the fall time is long, more of the early part of the signal passes through unprocessed.

### Hold (0 to 2000 ms)

Sets the time the applied expansion affects the signal below the threshold.

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

Sets the time after which the gain returns to its original level when the signal exceeds the threshold. If the **Auto Rise** button is activated, the plug-in automatically finds the best rise 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 closed. If **Range** is set to minus infinite , the gate is completely closed. The higher the value, the higher the level of the signal that passes through.

### Live

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

## Side-Chain Section

### Side-Chain

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

### Monitor

Allows you to monitor the filtered signal.

### Center

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

### Q-Factor

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

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

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

## Limiter

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



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

### Input (-24 to 24 dB)

Sets the input gain.

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

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

### Output

Sets the maximum output level.



## Maximizer

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



### Classic

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

### Modern

In **Modern** mode, the algorithm allows for more loudness than in **Classic** mode. This mode is particularly suited for contemporary styles of music.

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

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

### Optimize

Determines the loudness of the signal.

### Mix

Sets the level balance between the dry 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.

## Tube Compressor

This versatile compressor with integrated tube-simulation allows you to achieve smooth and warm compression effects. The VU meter shows the amount of gain reduction. **Tube Compressor** features an internal side-chain section that lets you filter the trigger signal.



### VU Meter

Shows the amount of gain reduction.

### In/Out Meters

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

### Input

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

### Drive

Controls the amount of tube saturation.

### Output

Sets the output gain.

### Character

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

### Attack (0.1 to 100 ms)

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

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

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

### Mix

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

### Ratio

Toggles between a low and a high ratio value.

### Side-Chain

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

## Side-chain section

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

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

### Center

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

### Q-Factor

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

### Monitor

Allows you to monitor the filtered signal.

## VSTDynamics

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



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

## Gate

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

The following parameters are available:

### Input meter

Shows the level of the input signal.

### Attack (0.1 to 100 ms)

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

### Threshold

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

### State LED

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


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

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

### Hold (0 to 2000 ms)

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

### Range

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

### Side-Chain

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

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

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

### Center

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

### Q-Factor

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

### **Monitor**

Allows you to monitor the filtered signal.

### **Compressor**

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

#### **Input meter**

Shows the level of the input signal.

#### **Graphical display**

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

#### **Gain Reduction meter**

Shows the amount of gain reduction.

#### **Threshold**

Determines the level where the compressor kicks in.

#### **Ratio**

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

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

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

#### **Attack (0.1 to 100 ms)**

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

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

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

### **Limiter**

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

#### **Input meter**

Shows the level of the input signal.

#### **Gain Reduction meter**

Shows the amount of gain reduction.

#### **Soft Clip**

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

## Output

Sets the maximum output level.

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

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

## Output section

### Output meter

Shows the level of the output signal.

### Module Configurator

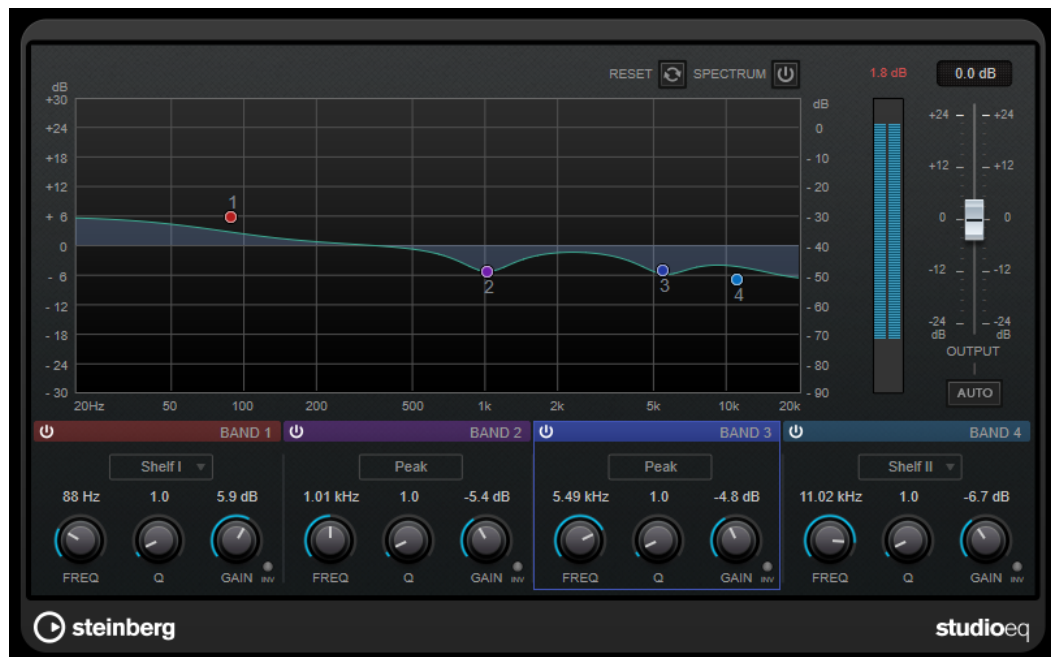
Changes the signal flow through the three effects. Changing the order of the effects can produce different results, and the available routing configurations allow you to quickly compare what works best for a given situation. Click **Module Configurator** to toggle between the following routing configurations:

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

# EQ

## 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.

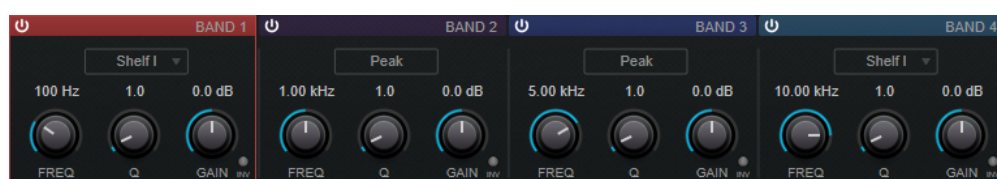
### Output

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, 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.
- To ensure that the cent offsets are taken into account, enter a space between the note and the cent offset.

### 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 to 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, and a cut filter. If **Cut** mode is selected, the **Gain** parameter is fixed.

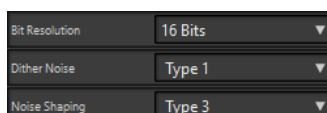
- **Shelf I** adds resonance in the opposite gain direction slightly above the set frequency.
- **Shelf II** adds resonance in the gain direction at the set frequency.
- **Shelf III** is a combination of **Shelf I** and **II**.

## Filter

## Mastering

### Internal Dithering

This is a WaveLab-specific plug-in that provides a simple way of adding a small amount of noise to the rendered signal to improve the apparent signal-to-noise ratio of the output.



NOTE

**Internal Dithering** is only available on the **Final Effects / Dithering** pane.

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The following parameters are available when selecting **Internal Dithering**.

**Noise Type**

Sets the noise type for adding to the signal.

- In **No Noise** mode, no dithering is applied.
- The **Noise Type 1** mode is the most all-round method.
- The **Noise Type 2** mode emphasizes higher frequencies more than **Noise Type 1**.



### Noise Shaping

Increases the apparent signal to noise ratio by altering the spectrum of the low-level audio signal which results from lowering the number of bits. The higher the number you select here, the more the noise is moved out of the ear's mid-range.

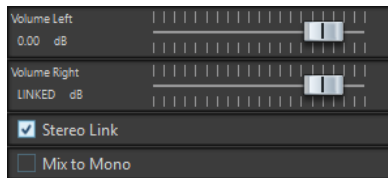
### Bit Resolution

Allows you to specify the intended bit resolution for the final audio, after dithering, regardless of whether you want to render the settings or play back in real-time.

Dithering changes the sample resolution, but not the sample size. For example, when dithering 24 bit to 16 bit, the file will still be 24 bit in size, although only 16 bits of information will have significance. When rendering to a 16-bit file, specify the file resolution to avoid wasting space.

## Leveler

This plug-in is useful for correcting an imbalance or adjusting levels between stereo channels, or for mixing down to mono.



### Volume Left/Volume Right (-48 dB to 12 dB)

Governs how much of the signal is included in the left and/or right channel of the output bus.

### Stereo Link

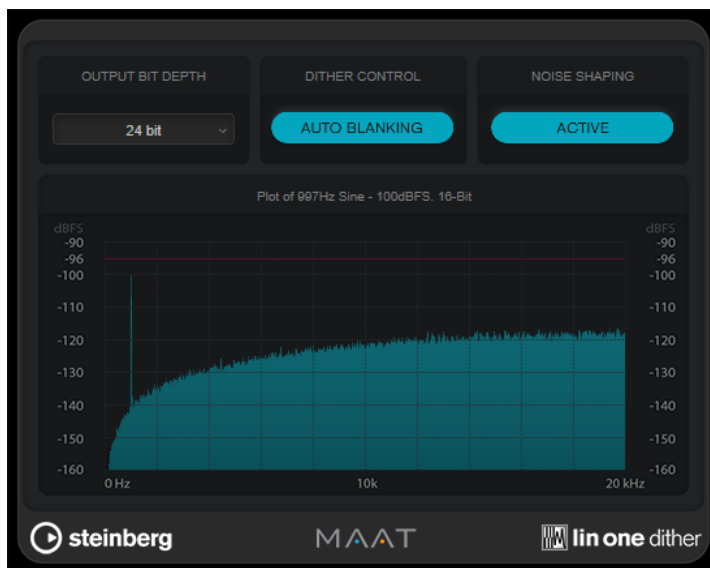
If this option is activated, **Volume Right** delivers the gain that is set for **Volume Left**.

### Mix to Mono

If this option is activated, a mono mix of the stereo channels is delivered to the output bus.

## Lin One Dither

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



### Output Bit Depth

Sets the bit depth of the output signal.

#### NOTE

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

### Dither Control

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

### Noise Shaping

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

## MasterRig

**MasterRig** allows you to master audio material in an intuitive and creative way. It offers high-class sound quality, accuracy, flexibility, and control.

### Main Layout

#### Module Chain

The module chain contains the mastering modules. You can add up to 5 modules.



The following settings are available for each module:

### Bypass

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

### Solo

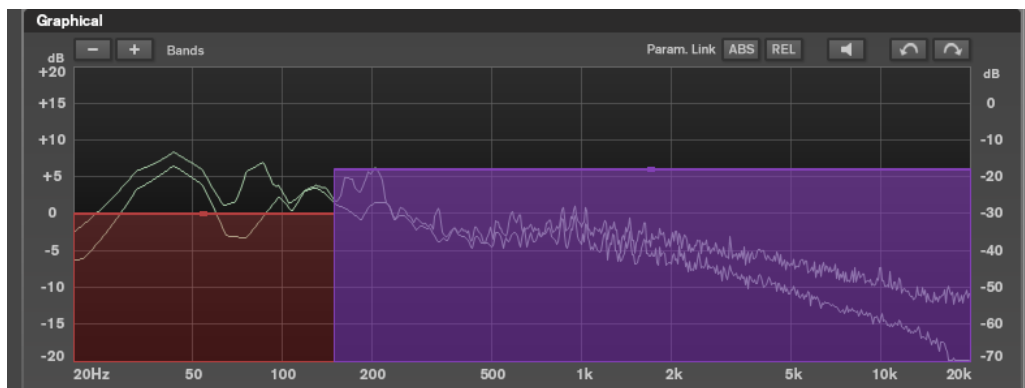
Solos the module. Only one module can be soloed at a time.

### Remove

Removes the module from the module chain.

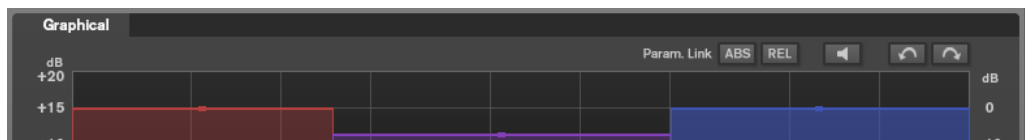
## Spectrum Display

The spectrum display in the upper half of the panel is where you set the width of the frequency bands. The vertical value scale to the left shows the gain level of each frequency band. The horizontal scale shows the 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.

## Settings



### Parameter Linking

Links the parameters of the same type in all bands in a module. This allows you to edit parameter values of all bands in a module simultaneously. Two link modes are available: **Absolute** and **Relative**.

- If **Absolute Mode** is activated and you edit a parameter value in one band, the corresponding parameter values in the other bands are set to the same value.
- If **Relative Mode** is activated and you edit a parameter value in one band, the corresponding parameter values in the other bands keep their relation.

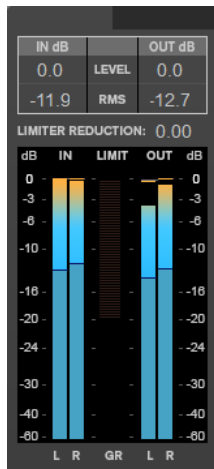
### Auto Listen for Filters

If this option is activated and you edit a parameter of a module, the corresponding filter or band is soloed. This allows you to locate unwanted frequencies in your audio and helps you to focus on a particular band or filter. Once you stop editing the parameter, **Solo** is deactivated.

## Undo/Redo

Undoes/Redoes the last operation.

## Input/Output Meter



The input/output meter provides a combined peak level, with peak-hold functionality and RMS meter. Between the meters for input and output is the gain reduction meter for the **Limitter**.

The maximum values for input/output peak level, RMS, and gain reduction are displayed above the meter display. To reset all maximum values, click any of the values.

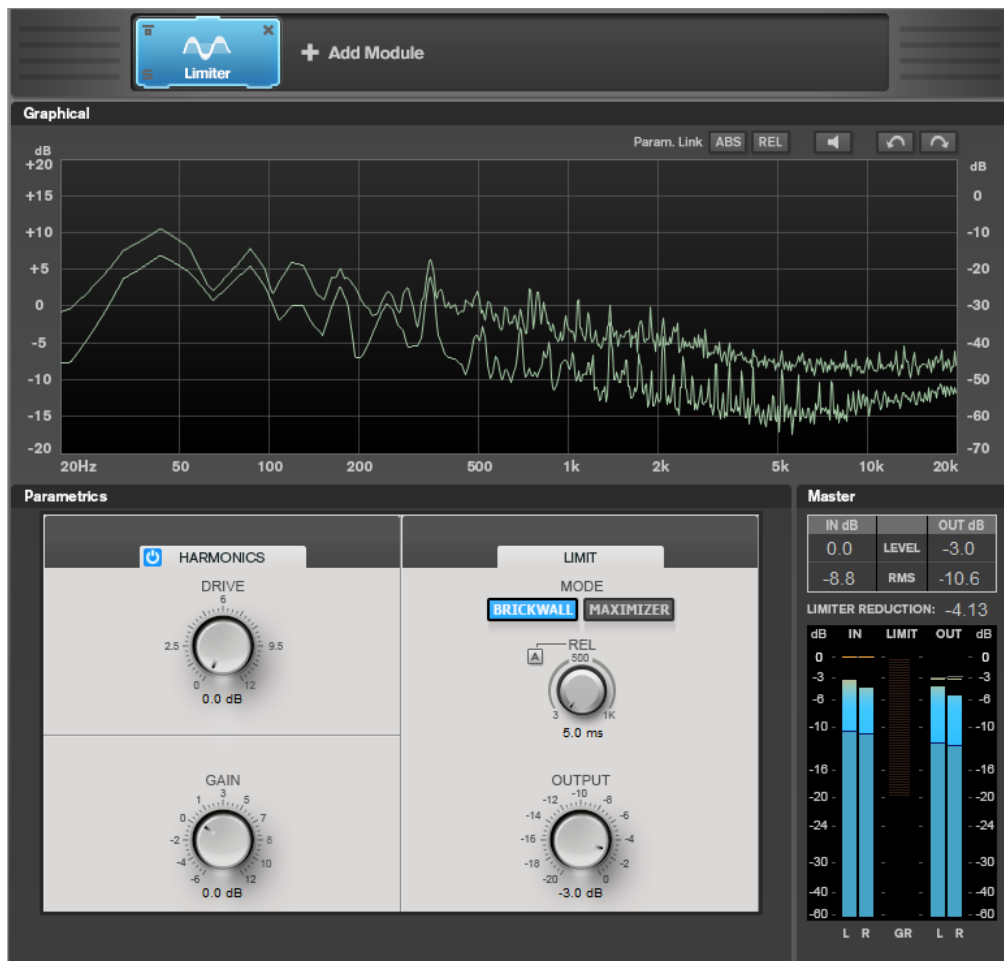
## Modules

Modules allow you to create a mastering chain. Some modules can be used only once and some in two instances in the module chain. You can rearrange modules in the module chain to change the processing order.

- To add a module to the module chain, click **Add Module** in the modules section and click a module.
- To remove a module, click the corresponding **Remove** button.
- To bypass a module, click the corresponding **Bypass** button.
- To solo a module, click the corresponding **Solo** button.
- To change the order of the modules, drag a module to another position in the module chain.

## Limiter

The **Limiter** module prevents clipping by ensuring that the output level does not exceed a set value.



### Band Settings

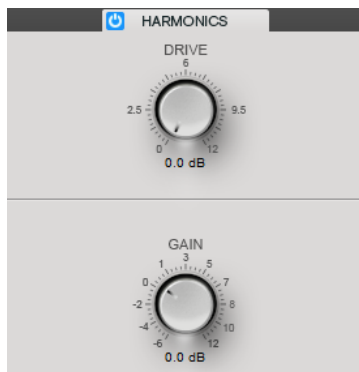


### On/Off

Activates/Deactivates the corresponding section.

### Harmonics

With the **Harmonics** section activated, the **Limiter** module starts limiting the signal softly. At the same time, harmonics are generated, adding a warm, tube-like characteristic to the audio material.



### Drive

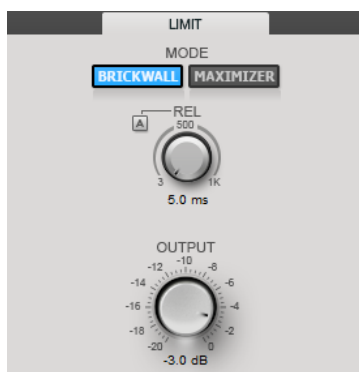
Allows you to adjust the amount of gain boost for the signal to raise the amount of soft-clipping.

### Gain

Sets the degree of attenuation.

## Brickwall

Thanks to its fast attack time, **Brickwall Limiter** can reduce even short audio level peaks without creating audible artifacts. The limiting amount is displayed between the input and the output meter.



### Release

Sets the time after which the gain returns to the original level; that is, the signal drops below the threshold. With **Auto Release** activated, the plug-in automatically identifies the best release setting for the audio material.

### Output

Sets the output level.

## Maximizer

**Maximizer** raises the loudness of audio material without any risk of clipping. The limiting amount is displayed between the input and the output meter.



### Optimize

Determines the loudness of the signal.

### Output

Sets the output level.

## Compressor

The **Compressor** module allows a signal to be split into two frequency bands. You can specify the level, bandwidth, and compressor characteristics for each band.



### Band Settings



### On/Off

Activates/Deactivates the corresponding section.

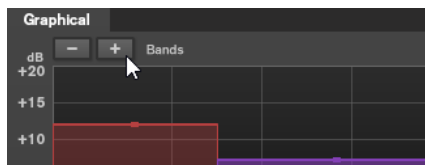
### Solo Band



Solos the corresponding frequency band.

### Add/Remove Band

Allow you to add and remove bands.



## Standard

Allows you to create smooth compression effects.



### THRESH (-60 to 0 dB)

Signal levels above the set threshold trigger the compressor.

### ATT (0.1 to 100 ms)

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

### REL (10 to 1000 ms)

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

### Ratio

Sets the amount of gain reduction applied to signal above the set threshold.

### Mix

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

### Compressor curve display

Graphically illustrates the compressor curve that is shaped according to the **Threshold** and **Ratio** parameter settings.

### Output

Sets the output gain.

## Tube

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





### Input

In combination with the **Output** setting, this parameter determines the compression amount. The higher the input gain setting and the lower the output gain setting, the more compression is applied.

### ATT (0.1 to 100 ms)

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

### REL (10 to 1000 ms)

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.

### Drive

Controls the amount of tube saturation.

### Mix

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

### Output

Sets the output gain.

### RELATED LINKS

[#unique\\_27/unique\\_27\\_Connect\\_42\\_side\\_chain\\_section](#)

## Equalizer

The **Equalizer** module is a high-quality 4-band parametric stereo equalizer with 4 fully parametric mid-range bands. The low band can act as either shelving filter, as peak filter (band-pass), or as cut filter (low-pass, band 1 only).



## Band Settings



### On/Off

Activates/Deactivates the corresponding section.

## Equalizer Section



### Type

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

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

**FREQ (20 to 20000 Hz)**

Sets the frequency of the corresponding band.

**Q**

Controls the width of the corresponding band.

**Gain (-15 to +15 dB)**

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

## Saturator

The **Saturator** module allows you to simulate the sound of analog tubes, and the saturation and compression effect when recording on analog tape machines.



## Band Settings



### On/Off

Activates/Deactivates the corresponding section.

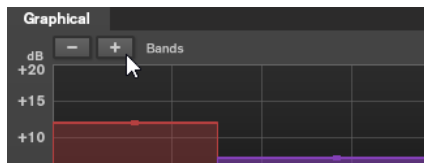
### Solo Band



Solos the corresponding frequency band.

### Add/Remove Band

Allow you to add and remove bands.



## Saturator Section



### Tape/Tube

Allows you to switch between tube saturation and tape saturation.

- Tube saturation simulates the saturation of analog tube compressors.
- Tape saturation simulates the saturation and compression effect of analog tape machine recordings.

### Drive

Controls the amount of saturation.

### Mix

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

### Output

Sets the output gain.

## Imager

The **Imager** module allows you to expand or reduce the stereo width of your audio in up to two bands. This way you can independently adjust the stereo image in defined frequency domains.



## Band Settings



### On/Off

Activates/Deactivates the corresponding section.

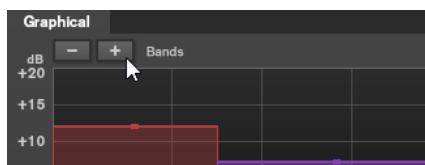
### Solo Band



Solos the corresponding frequency band.

### Add/Remove Band

Allow you to add and remove bands.



## Imager Section



### Width

Allows you to control the stereo width per band.

### Pan

Allows you to pan the signal left/right.

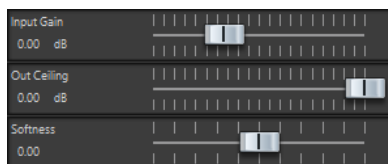
### Output

Sets the output level for each band.

## Peak Master

This is a basic plug-in that minimizes peaks in your audio file, allowing a louder mix without clipping. It is useful in taming dynamic instruments.

It is primarily used as a brickwall limiter. For example, you can limit audio peaks without altering the rest of the audio signal. In this case, set **Input Gain** to 0 dB and **Out Ceiling** to 0 dB, to achieve a clip-free audio signal. When used in this way, **Peak Master** is an excellent plug-in to succeed a resampler plug-in, and to proceed a dithering plug-in.



### Input Gain

Values range from -12 dB to 24 dB.

### Out Ceiling

This is the maximum level of the output signal. Values range from -18 dB to 0 dB.

### Softness

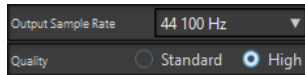
This governs the speed at which the signal becomes unaffected after limiting has been triggered on some samples. Values range from -5 to +5.

## 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.



### 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.

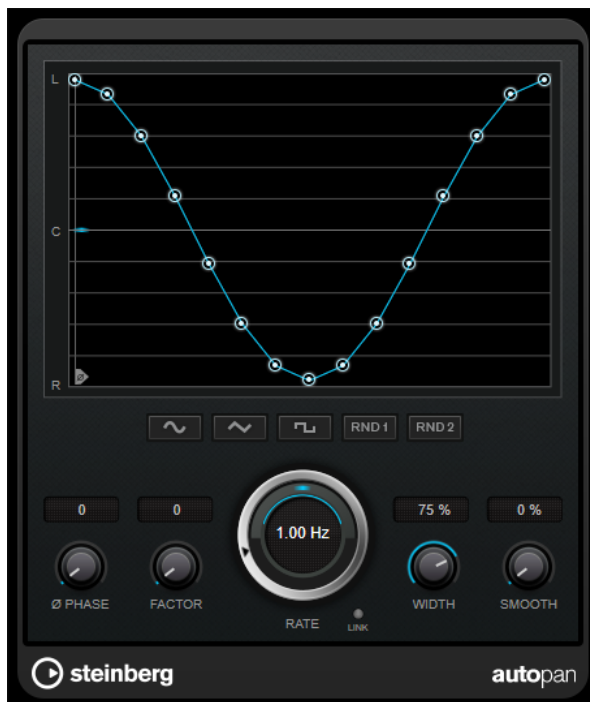
### Quality

Defines the quality of the algorithm that is used (**Standard** or **High**).

## Modulation

### AutoPan

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



### Waveform display

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

### Waveform preset buttons

Allow you to select presets for the modulation waveform.

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

### Phase

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

### Rate

Sets the auto-pan speed in Hertz and shows the movement within the panorama.

### Link

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

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

### Width

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

### Smooth

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

## Chorus

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



### Delay

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

### Width

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



### Spatial

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

### Mix

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

### Waveform Shape

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

### Lo Filter/Hi Filter

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

## Restoration

### RestoreRig

**RestoreRig** allows you to remove noise from an audio recording with different restoration modules. The noise can be an impulsive noise (**DeClicker**), an ambient noise (**DeNoiser**), or a low tonal noise (**DeBuzzer**)



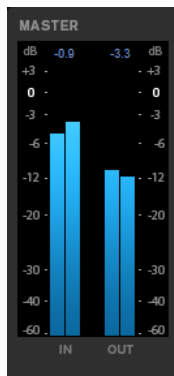
### Main Layout

#### Input Signal/Output Signal

Displays the input signal and the output signal of the restored signal. The vertical value scale to the left shows the gain level of the input and output signals. The horizontal scale shows the frequency range.



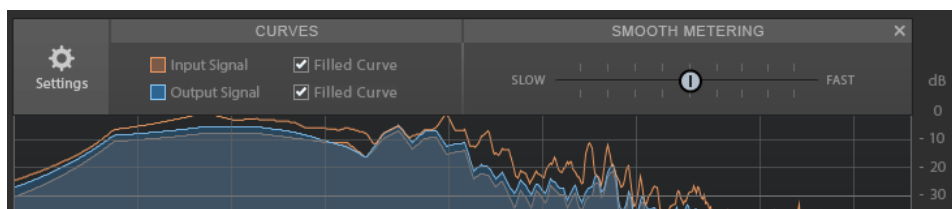
## Master



The input/output meter provides a peak level meter.

The maximum values for input/output peak level are displayed above the meter display. To reset all maximum values, click any of the values.

## Settings



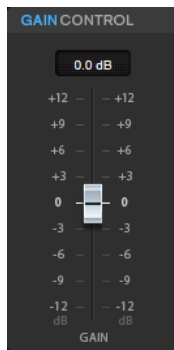
### Filled Curve

Allows you to fill the curves of the input signal and the output signal.

### Smooth Metering

Determines the reaction time of the display. Lower values result in faster reaction times.

## Gain Control



The **Gain Control** allows you to set the master gain for the modules.

## Modules

The modules **DeClicker**, **DeNoiser**, and **DeBuzzer** allow you to remove different kinds of noises.

- To activate or deactivate a module, click **Activate/Deactivate** on the left of the module name.
- To only listen to the sound that has been removed from the audio, click the **Noise Listening Mode** button of the module that you want to hear.

## DeClicker

**DeClicker** allows you to remove clicks from audio material.



### Activate/Deactivate DeClicker

Activates/Deactivates the module.

### Noise Listening Mode

Allows you to listen to the signal that has been removed from the original audio material.

### Meters

Allow you to monitor the quantity of impulsive noise that is removed from the signal.

#### NOTE

Avoid letting the meter reach the red region, as this can produce destructive artifacts.

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### Crackle

Allows you to remove very short impulsive noise from the audio signal.

### Click

Allows you to remove medium-sized impulsive noise from the audio signal.

### Pop

Allows you to remove long impulsive noise from the audio signal.

## DeNoiser

**DeNoiser** allows you to remove noise from the audio material.



### Activate/Deactivate DeNoiser

Activates/Deactivates the module.

### Noise Listening Mode

Allows you to listen to the signal that has been removed from the original audio material.

### Dynamic Level

Allows you to remove noise that evolves over time from the audio signal.

### Static Level

Allows you to remove noise that does not evolve over time from the audio signal. The **Learn** option allows you to define the stationary noise.

### Noise

The **Noise** options allow you to define a section in an audio file that contains a static noise that you want to remove. When you then render the audio file, you can remove the recorded static noise from the audio signal.

1. Play back the audio section that contains the noise that you want to remove and click **Learn**.

**RestoreRig** records the audio for a few seconds to detect the static noise.

2. Use the **Static Level** dial to set the level.
3. To remove the recorded static noise in the audio file, render the audio file.

If you want to record the static noise at another audio section, click **Reset**, play back another audio section, and click **Learn** again.

### Algorithm

Allows you to select different **DeNoiser** algorithms. Depending on the audio material, different modes can affect the **DeNoiser** quality.

- **Smooth** is sufficient for most uses.
- Use **Musical** for harmonic content with low rhythmic or transient components.
- Use **Rhythmic** for drum and percussive content.

- Use **Strong** if the noise level reduction is more important than the accuracy of the noise reduction.
- Use **Speech** for vocal content.

## DeBuzzer

**DeBuzzer** allows you to remove harmonic noise with a fundamental frequency that should be around 50 to 60 Hz.



### Activate/Deactivate DeBuzzer

Activates/Deactivates the module.

### Noise Listening Mode

Allows you to listen to the signal that has been removed from the original audio material.

### Level

Allows you to define the reduction of the noise in dB.

### Sensitivity

Allows you to define how sensitive the reduction will be to the current audio level. At 0 %, **DeBuzzer** reduces the current harmonic noise with the **Level** value. At higher sensitivity values, the level is dynamically defined in a range between 0 dB and the **Level** value. This reduces the buzz when the audio level is low and does not affect the audio when the audio level is high.

### Frequency

Allows you to define the value of the fundamental frequency.

### Auto

If this option is activated, **DeBuzzer** automatically detects the fundamental frequency of the current most prominent harmonic tone.

#### NOTE

Once you have detected the frequency that you want to remove, deactivate **Auto**.

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## Reverb

### 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 Frequency

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 Frequency

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

Sets 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 Frequency**

Determines the frequency below which low-frequency damping occurs.

### **High Frequency**

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. This button should normally be activated if **RoomWorks** is used as a send effect for an FX channel or a group channel.

### 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. Interesting effects can be created with very high **Efficiency** settings (>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

Sets 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 %.

## Spatial

### MonoToStereo

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

#### NOTE

This plug-in works only on stereo tracks.

---



### Delay

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

### Width

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

### Mono

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

### Color

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

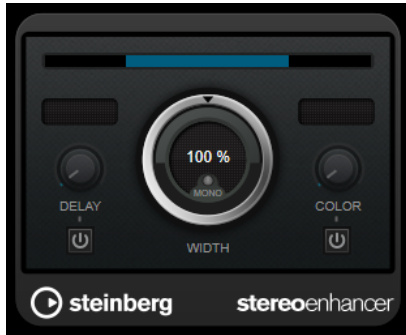
## StereoEnhancer

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

#### NOTE

This plug-in works only on stereo tracks.

---



### Delay

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

### Width

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

### Mono

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

### Color

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

## Stereo Expander

This plug-in is a stereo width enhancer that makes a stereo signal sound wider. It gives better results from real stereo material, as opposed to mono channels panned to different positions in the stereo image.



### Width

Higher values result in a greater stereo width. Usually, you set **Width** to values between 0 % and 20 %. Higher values can be used for special effects.

## Surround

### Mix6to2

**Mix6to2** lets you quickly mix down your surround mix format to stereo. You can control the levels of up to six surround channels and decide for each channel up to which level it is included in the resulting mix.



### Surround Channels

#### Volume faders

Determine how much of the signal is included in the left and/or right channel of the output bus.

#### Link

Links the volume faders of a surround channel.

#### Invert Phase

Inverts the phase of the corresponding surround bus channel.

### Output Bus

#### Volume faders

Set the volume of the mixed output.

#### Link

Links the **Output** faders.

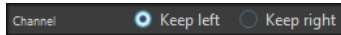
#### Normalize

If this option is activated, the mixed output is normalized. For example, the output level is automatically adjusted so that the loudest signal is as loud as possible without clipping.

## Tools

### Channel Extractor

This plug-in allows you to only keep the left or the right channel of a stereo stream.



#### Channel

Lets you select whether to keep the left or the right channel of the stereo stream.

### Silence

This plug-in provides a simple way of inserting a precise period of silence at the start or at the end of an audio file. Use this plug-in to add silence at the end of a file, so that the tail of a reverb plug-in does not cut immediately at the end of the file.



#### Start

Use the slider to insert from 0 to 60,000 ms of silence at the start of the file.

#### End

Use the slider to insert from 0 to 60,000 ms of silence at the end of the file.

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