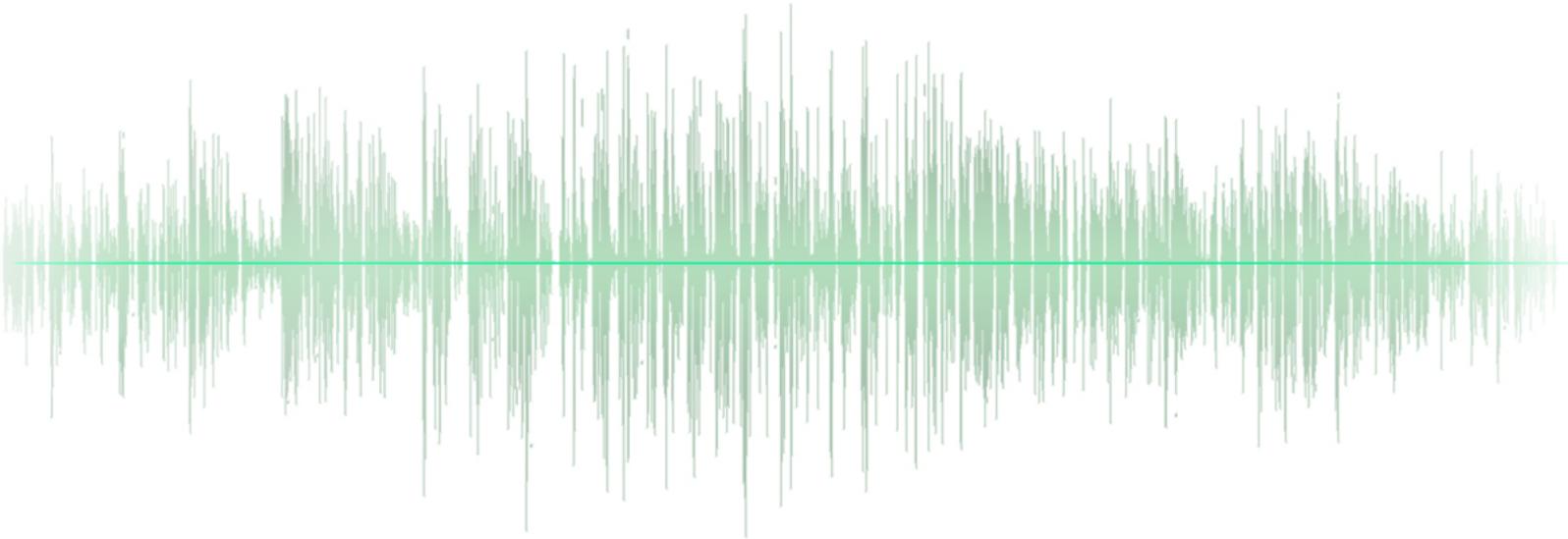


Plug-in Reference



WAVELAB ELEMENTS 9.5

Personal Audio Editing System



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This PDF provides improved access for vision-impaired users. Please note that due to the complexity and number of images in this document, it is not possible to include text descriptions of images.

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WaveLab Elements_9.5.40_en-US_2018-11-20

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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.

- WaveLab-specific plug-ins can only be used in the **Master Section**. However, some WaveLab effects are also included as VST plug-ins, available as track or clip effects in audio montages.
- You can specify which plug-ins should be available on the **Effects** pane and the **Final Effect/Dithering** pane of the **Master Section** by using the **Plug-in Settings** dialog.

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.

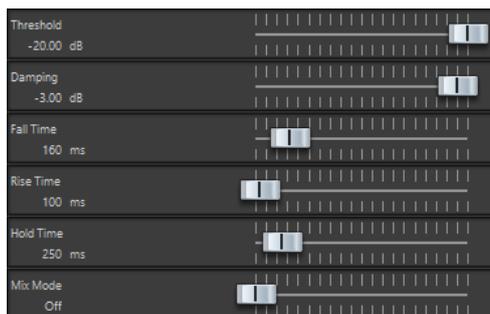
Quality

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

Ducker

This plug-in lets you control (modulate) the volume of clips placed on a track with the signal of one or more clips placed on the next adjacent track below it. The **Ducker** plug-in can only be used as a clip effect in the audio montage.

It uses the **Route to** options that can be found on the **Track** menu. You can use mono or stereo tracks for both the modulating and the upper track.



Threshold

Sets the loudness threshold that triggers the **Ducker**. Clips on the modulator track with levels above the threshold will cause the level of a clip on the upper track to be lowered.

Damping

Sets the amount of level reduction that is applied to the clip on the upper track.

Fall Time

Sets the time it takes for the level to change from 0 dB to the set damping level.

Hold Time

When the modulating signal falls below the set threshold, this setting determines how long the level will stay reduced before it starts rising to normal level again.

Rise Time

Sets the time after which the reduced level rises to the normal level when the modulating signal falls below the set threshold (after the **Hold Time**).

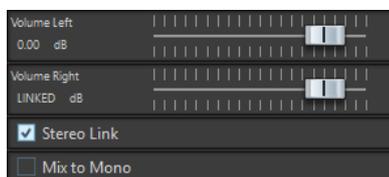
Mix Mode

If this is activated, the **Ducker** outputs a mix of the two tracks. This is only useful if the **Route to Upper Track Only** option has been activated for the modulating track. Then this feature can be used for processing several clips through the same plug-in chain if more plug-ins have been assigned after the **Ducker** on the upper track.

Note that the mixed output is controlled by the upper track. If this is not playing a clip, both of the tracks will be silent.

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.

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

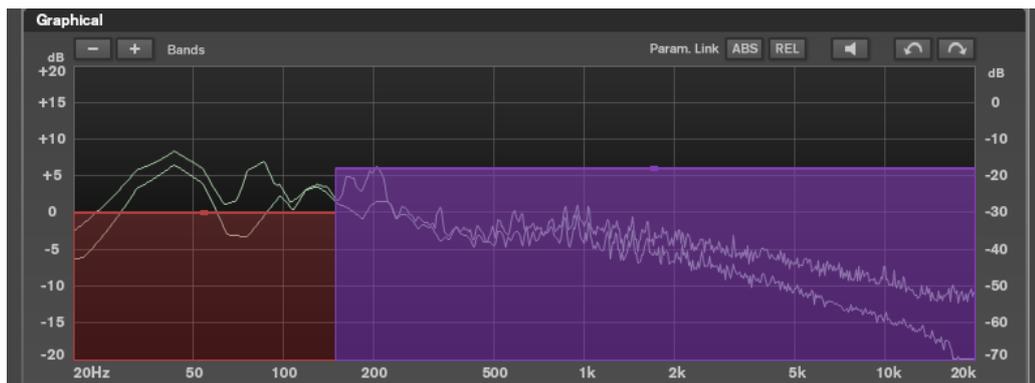
Solos the module. Only one module can be soloed at a time.

Remove

Removes the module from the module chain.

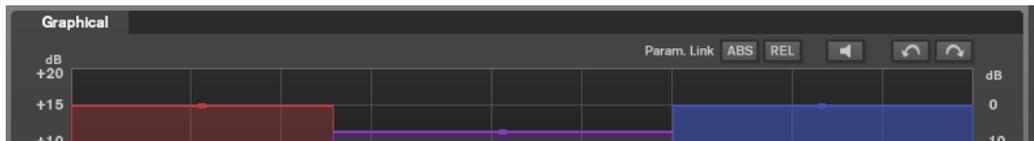
Spectrum Display

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 ± 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.

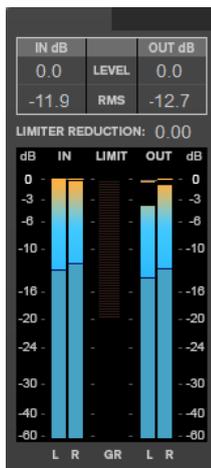
Auto Listen for Filters

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

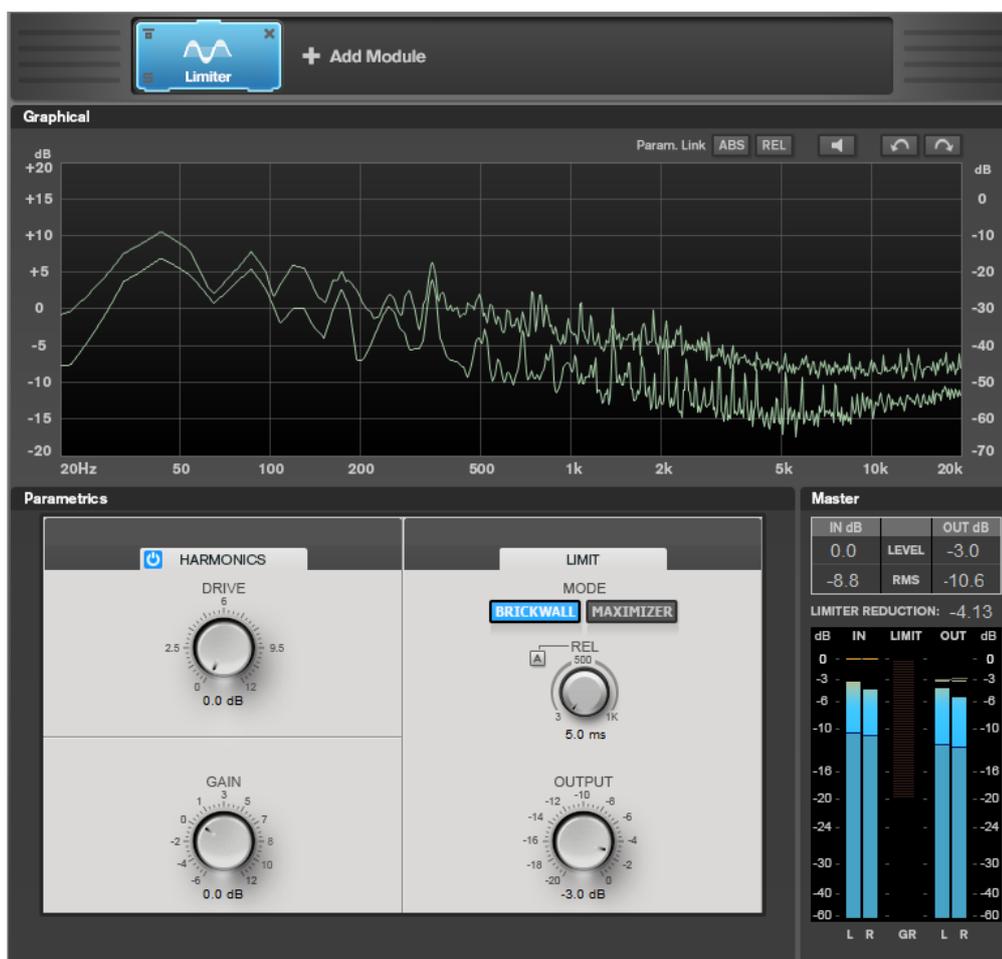
Modules allow you to create a mastering chain. Some modules can be used only once and some in two instances in the module chain. You can rearrange modules in the module chain to change the processing order.

- 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 makes sure that the output level never exceeds a set output level, to avoid clipping in following devices.



Band Settings



On/Off

Activates/Deactivates the corresponding section.

Harmonics

If the **Harmonics** section is 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 amount of attenuation.

Brickwall

Due 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 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.

Output

Sets the output level.

Maximizer

Maximizer raises the loudness of audio material without the 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

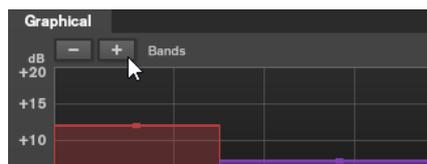
Activates/Deactivates the corresponding section.

Soloing Frequency Bands

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

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 signal 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 signal and the wet signal.

Output

Sets the output gain.

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.

Linear Phase

Activates/Deactivates linear phase mode for the corresponding band.

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

NOTE

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

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

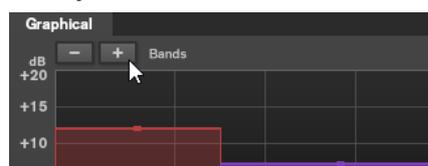
Activates/Deactivates the corresponding section.

Soloing Frequency Bands

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

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 signal 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

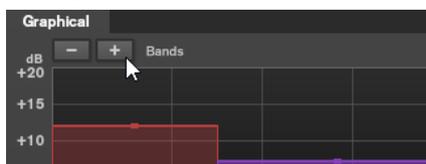
Activates/Deactivates the corresponding section.

Soloing Frequency Bands

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

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.

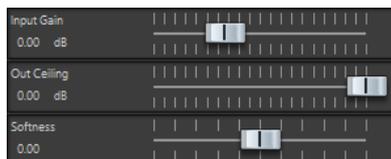
Output

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.

RestoreRig

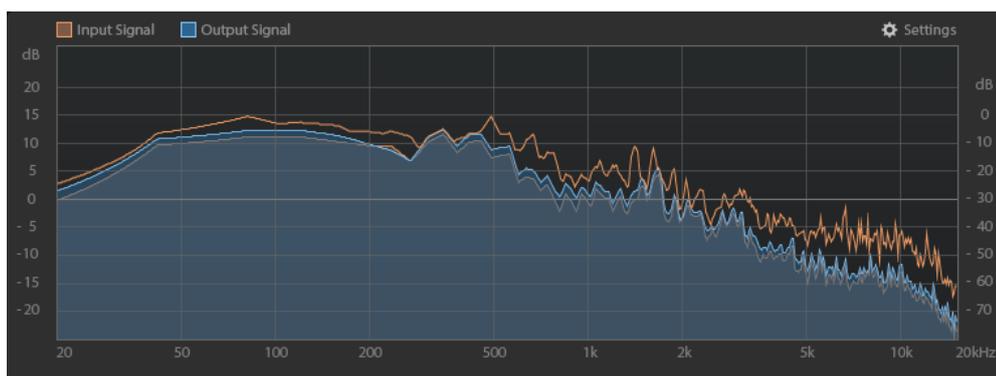
RestoreRig allows you to remove noise from an audio recording with different restoration modules. The noise can be an impulsive noise (**DeClicker**), an ambient noise (**DeNoiser**), or a low tonal noise (**DeBuzzer**)



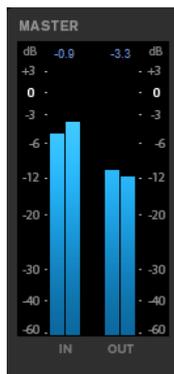
Main Layout

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.

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**.

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.

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.

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.

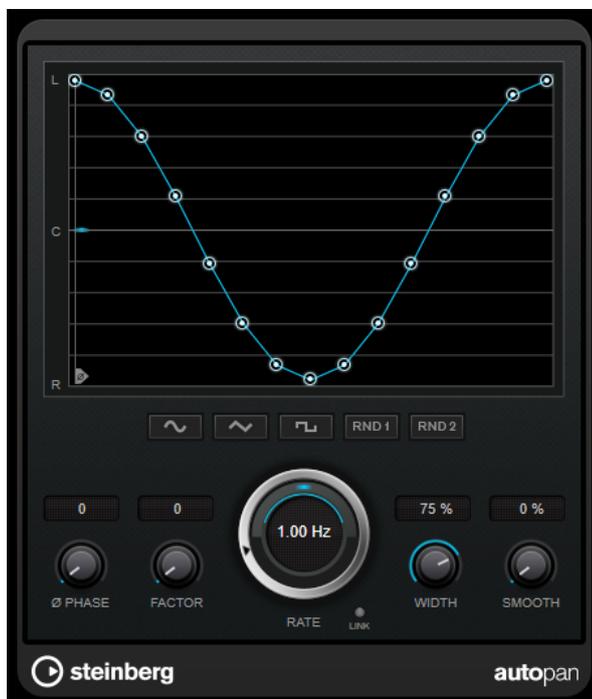
- You can specify which VST plug-ins should be available in the **Effects** pane and **Final Processing/Dithering** pane of the **Master Section** by using the **Plug-in Settings** dialog.
- VST plug-ins have their own preset handling. You can save or load effect programs (presets).

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.

NOTE

The panning effect of this plug-in works only on stereo tracks.



Waveform display

Shows the shape of the modulation waveform and allows you to manually adjust the shape of the waveform. 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 presets

Allows 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.
- **Pulse** 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, left and right channel are modulated simultaneously. This results in a chopping effect instead of auto-panning.

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

Width

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

Smooth

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

Brickwall Limiter

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



Due to its fast attack time, **Brickwall Limiter** can reduce even short audio level peaks without creating audible artifacts. However, this plug-in creates a latency of 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.

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.

Chorus

This plug-in is a single-stage chorus effect. It works by doubling 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.

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.

NOTE

If side-chaining is supported, the modulation can also be controlled from another signal source via the side-chain input. If the side-chain signal exceeds the threshold, the modulation is controlled by the side-chain signal's envelope. For a description of how to set up side-chain routing, see the **Operation Manual**.

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 the **Auto** button is activated, the knob becomes dark and 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 the **Auto Release** button is activated, the plug-in automatically finds the best release setting for the audio material.

Analysis (Pure Peak to Pure RMS)

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

Live

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

Distortion

Distortion adds crunch to your tracks.



Boost

Increases the distortion amount.

Oversampling

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

NOTE

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

Mix

Sets the level balance between the dry signal 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.

Gate

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



Attack (0.1 to 1000 ms)

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

NOTE

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

Hold (0 to 2000 ms)

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

Release (10 to 1000 ms or Auto mode)

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

Threshold (-60 to 0 dB)

Determines the level where 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.

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 (50 to 20000 Hz)

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

Q-Factor

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

Filter buttons (Low-Pass/Band-Pass/High-Pass)

If the **Side-Chain** button is activated, you can use these buttons 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.

MonoDelay

This is a mono delay effect that can either be tempo-based or use 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.

Feedback

Sets the number of repeats for the 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.

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.

Lo 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.

Hi 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

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

NOTE

This plug-in works only on stereo tracks.



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.

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.



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-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 **Alt**-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 **Shift**-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 **Ctrl/ Cmd**-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**.

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.



Drive (1.0 to 6.0 dB)

Controls the amount of tube saturation.

Input

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

Ratio

Toggles between a low and a high ratio value.

Output (-12 to 12 dB)

Sets the output gain.

Character

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

Attack (0.1 to 100 ms)

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

Release (10 to 1000 ms or Auto mode)

Sets the time after which the gain returns to its original level. If the **Auto Release** 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.

In/Out Meters

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

VU Meter

Shows the amount of gain reduction.

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 buttons (Low-Pass/Band-Pass/High-Pass)

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

Center (50 to 20000 Hz)

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

Q-Factor

If the **Side-Chain** button 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 processors: **Gate**, **Compressor**, and **Limiter**, covering a variety of dynamic processing functions.



The window is divided into three sections, containing controls and meters for each processor. Activate the individual processors using the buttons **Gate**, **Compressor**, and **Limiter** at the bottom of the plug-in panel.

Gate Section

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:

Threshold (-60 to 0 dB)

Determines the level where 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).

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 buttons (Low-Pass/Band-Pass/High-Pass)

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

Center (50 to 20000 Hz)

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

Q-Factor

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

Monitor

Allows you to monitor the filtered 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 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 the **Auto Release** button is activated, the plug-in automatically finds the best release setting for the audio material.

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.

Input Gain Meter

Shows the input gain.

Compressor Section

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

Threshold (-60 to 0 dB)

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

Ratio

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

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

Compensates for output gain loss, caused by compression. If the **Auto** button is activated, the knob becomes dark and 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 the **Auto Release** button is activated, the plug-in automatically finds the best release setting for the audio material.

Graphical display

Use the graphical display to graphically set the threshold and ratio values. To the left and right of the graphical display, you find two meters that show the amount of gain reduction in dB.

Limiter Section

The limiter ensures that the output level never exceeds a set threshold, to avoid clipping in following devices. Conventional limiters usually require very accurate setting up of the attack and release parameters to prevent the output level from going beyond the set threshold level. The limiter adjusts and optimizes these parameters automatically according to the audio material.

Output

Sets the maximum output level.

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.

Release (10 to 1000 ms or Auto mode)

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

Meters

The three meters show the input gain (IN), the gain reduction (GR) and the output gain (OUT).

Module Configuration Button

Using the **Module Configuration** button in the bottom right corner of the plug-in panel, you can set the signal flow order for the three processors. Changing the order of the processors can produce different results, and the available options allow you to quickly compare what works best for a given situation. Simply click the **Module Configuration** button to change to a different configuration. There are three routing options:

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

Legacy Plug-ins

Under Windows, a set of plug-ins is provided for compatibility with audio projects that referenced these effects when using earlier versions of WaveLab. An audio montage which referenced these plug-ins would otherwise require cumbersome user intervention to open, for example.

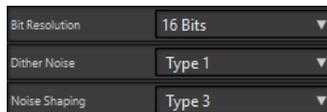
Their use with new audio projects is not recommended and they are not documented.

Dithering Plug-ins

Dithering plug-ins add small quantities of noise to a signal to reduce the audibility of low level distortion in a digital recording. A small amount of random noise is added to the analog signal before the sampling stage, reducing the effect of quantization errors.

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.



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.

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