Plug-in Reference

WAVELAB PRO 9.5

Audio Editing And Mastering Suite

steinberg

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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.

WaveLab Pro_9.5.40_en-US_2018-11-20

Table of Contents

WaveLab-specific Plug-ins

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

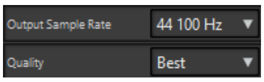
- WaveLab-specific plug-ins can only be used in the **Master Section** and in batch processes. However, some WaveLab effects are also included as VST plug-ins, available as track or clip effects in audio montages.
- You can specify which plug-ins should be available on the **Effects** pane and the **Final Effects/Dithering** pane of the **Master Section** by using the **Plug-in Settings** dialog.
- When a multichannel configuration is used in the audio montage, only certain plug-ins can be used as master effects. All channels in the **Master Section** are affected equally.

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.

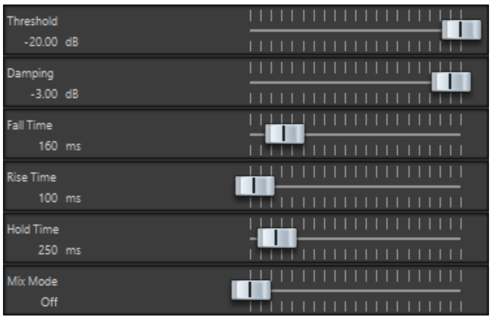
Quality

Defines the quality of the algorithm that is used (**Standard**, **High**, **Very High**, **Best**). In **Standard** mode, the CPU load is much lower than in **Best** mode but the sound quality of the resulting audio is also lower.

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**).

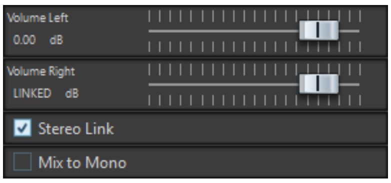
Mix Mode

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

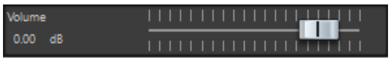
If this option is activated, **Volume Right** delivers the gain that is set for **Volume Left**.

Mix to Mono

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

Leveler Multi

This plug-in takes multichannel input and applies a fader equally to all channels.



Volume (-48 dB to 12 dB)

Governs how much gain is applied to the signal before it is routed 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 8 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

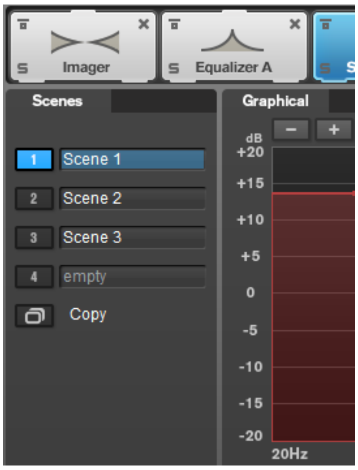
Solos the module. Only one module can be soloed at a time.

Remove

Removes the module from the module chain.

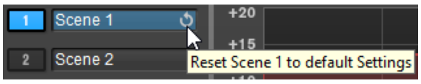
Scenes

You can save up to 4 **MasterRig** configurations as scenes. This allows you to compare different parameter settings and module combinations.



- To copy the settings of a scene to another scene, click **Copy Scene**, and then click the scene button of the scene where you want to paste the settings.

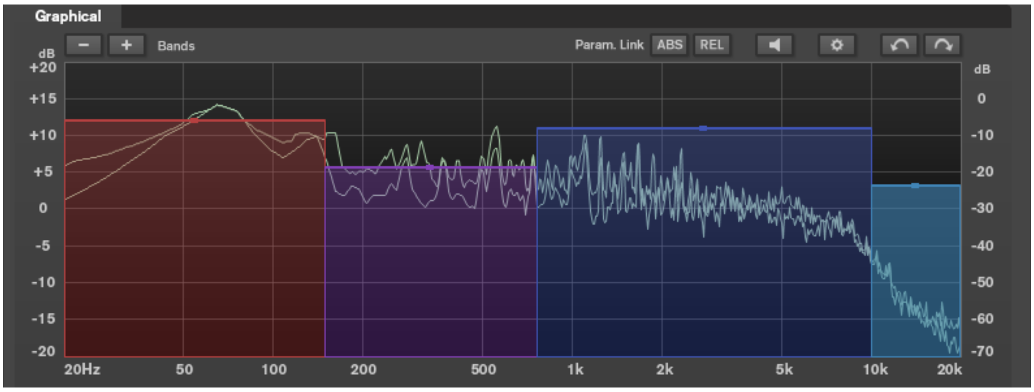
  A copy of a scene is indicated by a **(c)** behind the scene name.
- To reset the settings of the selected scene, click **Reset Scene**.



- To rename a scene, double-click the scene name and type in another name.

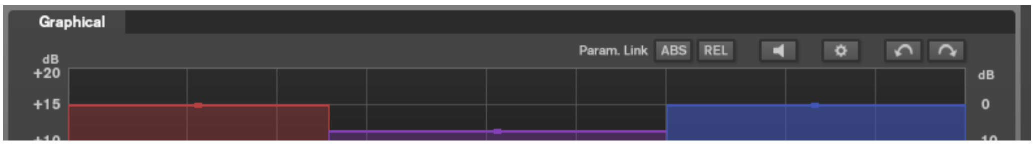
Spectrum Display

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.

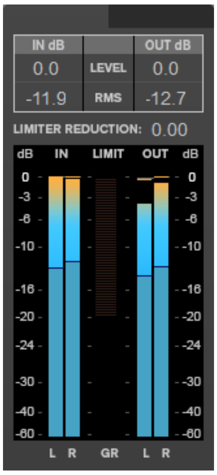
Global Settings

Allows you to make global settings for **MasterRig**.

Undo/Redo

Undoes/Redoes the last operation. The undo/redo history is deleted when you select another scene.

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 **Limiter**.

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.

Side-Chain Settings

The **Compressor** module and the **Dynamic EQ** module support side-chain. You can set up the side-chain routing for each band separately.

- To open the side-chain panel, click the **SC** button at the bottom left of each band section.

Active

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

SC FREQ

Sets the frequency of the side-chain filter.

Auto (Dynamic EQ only)

Deactivates the **SC Frequency** knob of the side-chain panel. Instead, the settings of the **Frequency** knob are used.

Listen

Allows you to solo the side-chain filter.

SC Q

Sets the resonance or width of the filter.

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.

Global Settings

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

Spectrum Display

Show Spectrum

Activates/Deactivates the spectrum display.

Smooth

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

Peak Hold

Freezes the peak values of the spectrum display.

Slope

Tilts the spectrum display around the 1 kHz pivot.

Two Channels

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

EQ Curve

Show Curve

Shows/Hides the EQ curve in the spectrum display.

Filled

If this option is activated, the EQ curve is filled.

RMS

AES17 (+3 dB)

If this option is activated, the RMS value is raised by 3 dB to follow the AES17 standard.

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.

Balance

Mid/Side

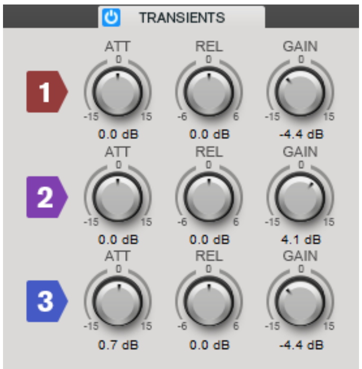
Allow you to set the gain for the mid and side signal.

Solo Mid Signal/Solo Side Signal

Allow you to solo the mid or side signal.

Transients

If the **Transients** section is activated, you can set the following parameters:



ATT

Sets the gain of the attack phase of the signal for the corresponding band.

REL

Sets the gain of the release phase of the signal for the corresponding band.

Gain

Sets the output level for the corresponding band.

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.

2nd HARM/3rd HARM

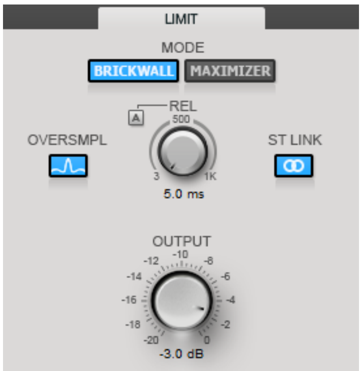
Allow you to control the second and third harmonic independently.

Drive

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

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.

Oversample

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

Stereo Link

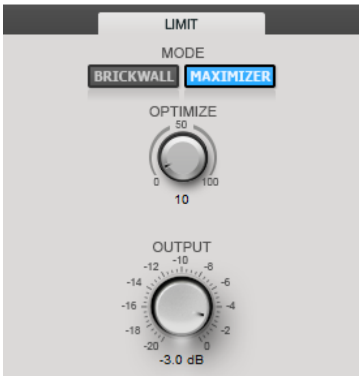
If this option is activated, **Brickwall Limiter** uses the channel with the highest level to analyze the input signal. If the button is deactivated, each channel is analyzed separately.

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.

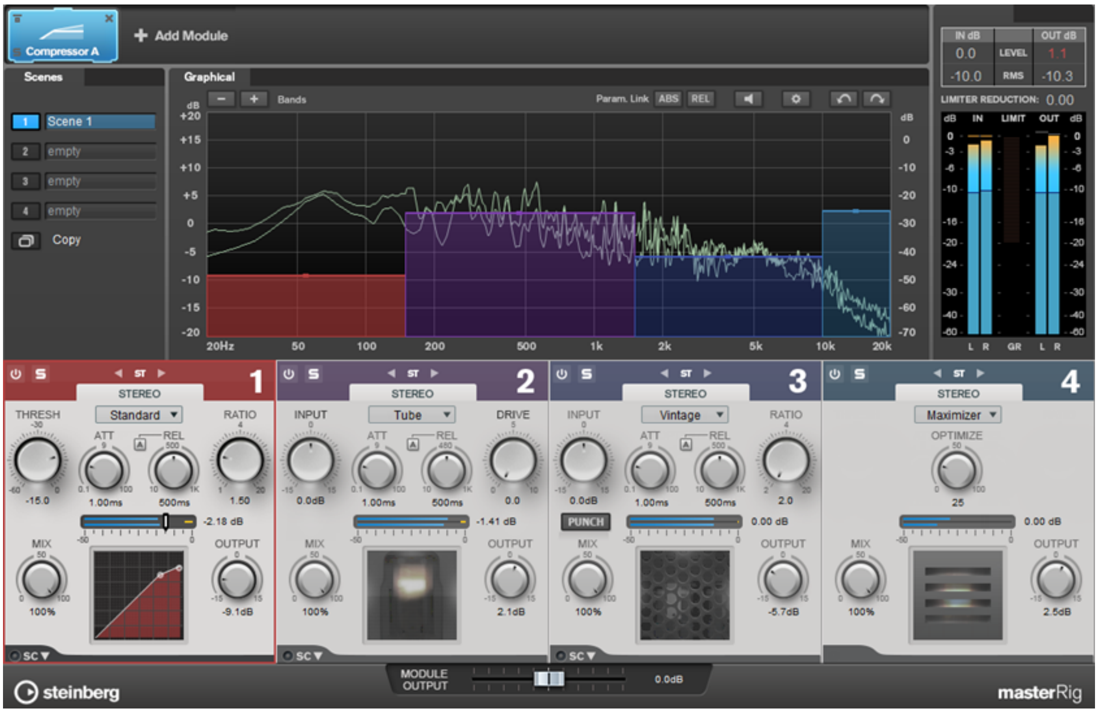
Output

Sets the output level.

Compressor

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

You can add two **Compressor** modules to the module chain, **Compressor A** and **Compressor B**.



Band Settings

On/Off

Activates/Deactivates the corresponding section.

Soloing Frequency Bands

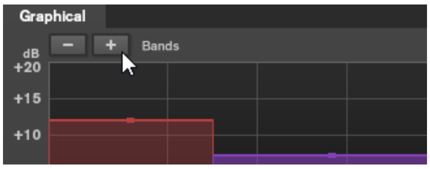
To solo a frequency band, activate the **S** button in each section. Only one band can be soloed at a time.

Channel Settings

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

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.

Side-Chain

Opens the **Side-Chain** settings.

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.

Side-Chain

Opens the **Side-Chain** settings.

Vintage

Vintage Compressor is modeled after vintage type compressors.

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 material.

Ratio

Sets the amount of gain reduction that is applied to signals above the set threshold.

Attack Mode (Punch)

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

Mix

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

Output

Sets the output gain.

Side-Chain

Opens the **Side-Chain** settings.

Maximizer

Optimize

Determines the loudness of the signal.

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 8-band parametric stereo equalizer with 8 fully parametric mid-range bands. The low and high bands can act as either shelving filter, as peak filter (band-pass), or as cut filter (low-pass/high-pass, band 1 and 8 only).

You can add two **Equalizer** modules to the module chain, **Equalizer A** and **Equalizer B**.

Band Settings

On/Off

Activates/Deactivates the corresponding section.

Channel Settings

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

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 and 8, 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 (band 1) or above (band 8) the set frequency. You can choose between different slopes: 6 dB, 12 dB, 24 dB, 48 dB, or 96 dB per octave.

FREQ (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.

Dynamic EQ

Dynamic EQ allows you to adjust frequencies and lets you determine when and how the EQ is applied depending on the dynamics of the audio material.

You can add two **Dynamic EQ** modules to the module chain, **Dynamic EQ A** and **Dynamic EQ B**.

Band Settings

On/Off

Activates/Deactivates the corresponding section.

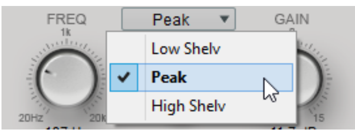
Channel Settings

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

Equalizer Section

Type pop-up menu

Allows you to select the EQ types.



- **Low Shelf** boosts or attenuates frequencies below the cutoff frequency by the specified amount.

- **Peak** boosts or attenuates frequencies at the set frequency value with a bell shaped filter.
- **High Shelf** boosts or attenuates frequencies above the cutoff frequency by the specified amount.

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.

THRESH (-50 to 0 dB)

Determines the threshold level. Only signal levels above the threshold are processed.

ATT (0.1 to 100 ms)

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

REL (10 to 1000 ms)

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

Ratio

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

Side-Chain

Opens the **Side-Chain** settings.

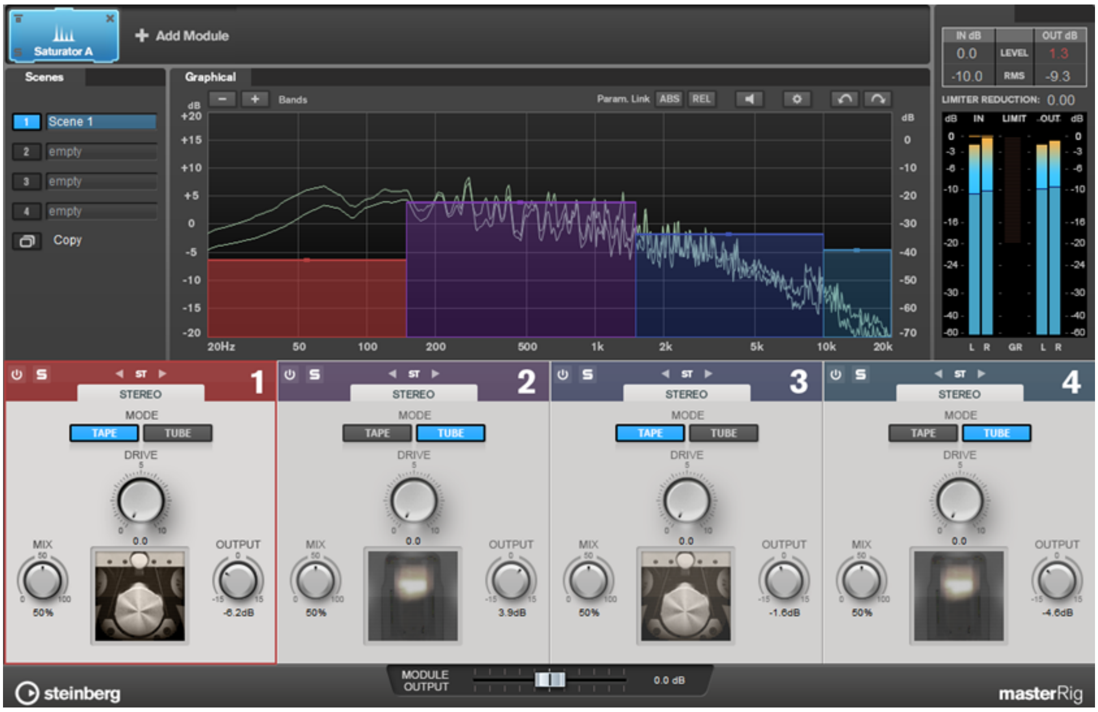
RELATED LINKS

Side-Chain Settings on page 8

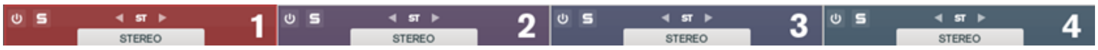
Saturator

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

You can add two **Saturator** modules to the module chain, **Saturator A** and **Saturator B**.



Band Settings



On/Off

Activates/Deactivates the corresponding section.

Soloing Frequency Bands

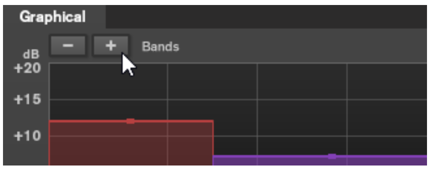
To solo a frequency band, activate the **S** button in each section. Only one band can be soloed at a time.

Channel Settings

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

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

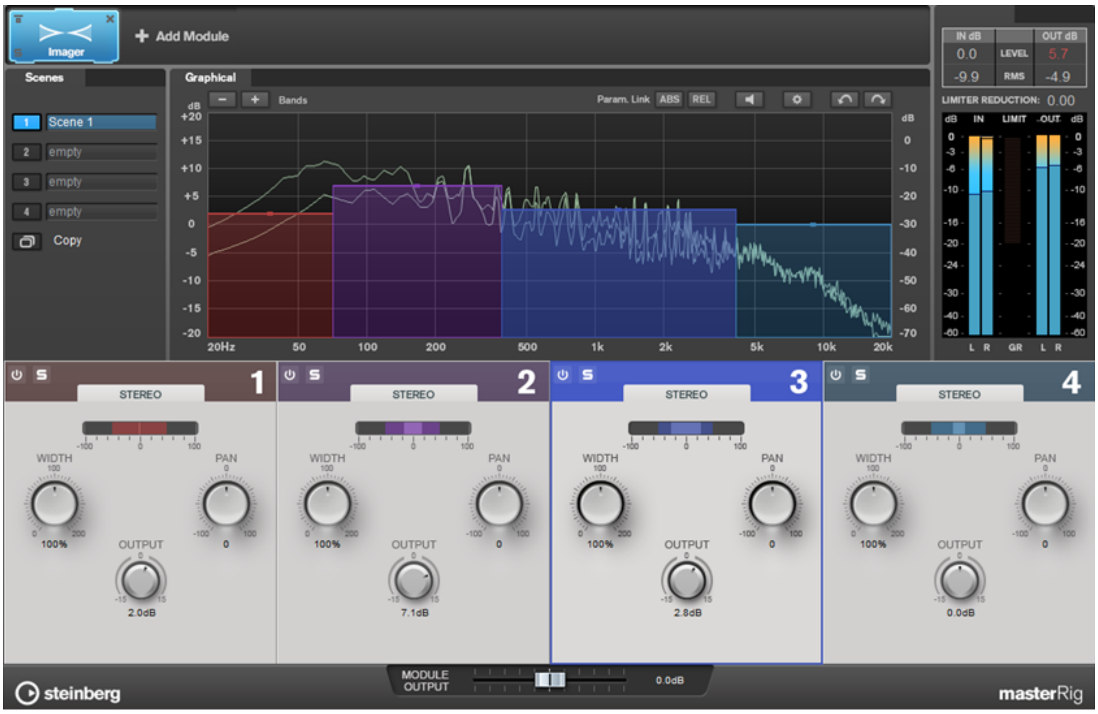
Sets the level balance between the dry signal and the wet signal.

Output

Sets the output gain.

Imager

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



Band Settings

On/Off

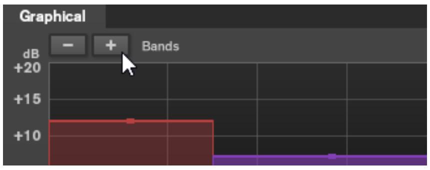
Activates/Deactivates the corresponding section.

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.

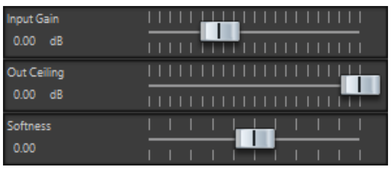
Output

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.

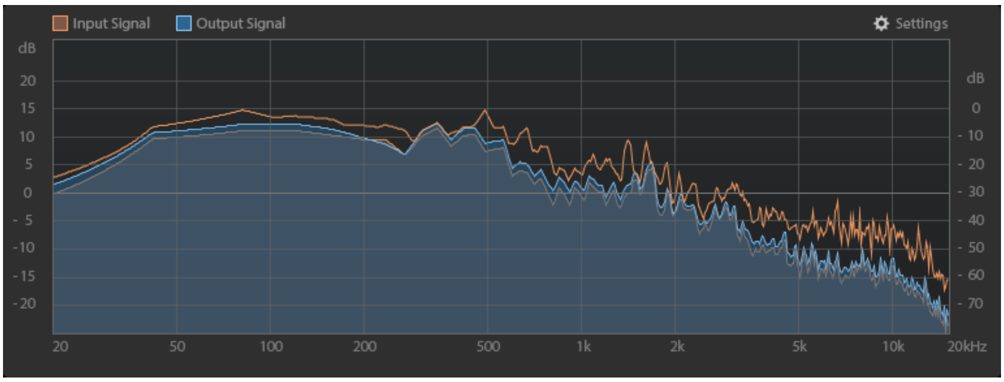
RestoreRig

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**)

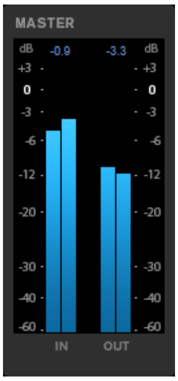
Main Layout

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.



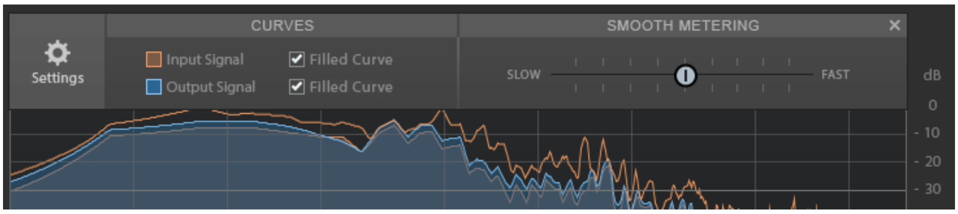
Master



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



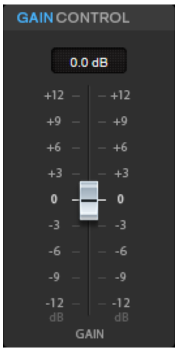
Filled Curve

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.

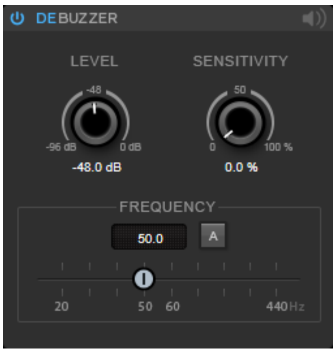
Algorithm

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

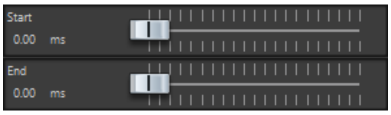
If this option is activated, **DeBuzzer** automatically detects the fundamental frequency of the current most prominent harmonic tone.

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

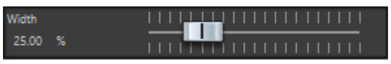
Use the slider to insert from 0 to 60,000 ms of silence at the start of the file.

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

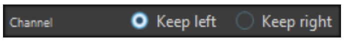
If this option is activated, **Brickwall Limiter** uses oversampling to detect and limit signal levels between two samples to prevent distortion when converting digital signals into analog signals.

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.

Rate

Allows you set the sweep rate freely with the **Rate** knob.

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.

CurveEQ

Voxengo **CurveEQ** is a spline equalizer for professional music and audio production applications. **CurveEQ** shows the filter response you are designing by means of a spline, that is, a smooth curvy line. This way you can see how the EQ alters the sound.

CurveEQ implements spectrum matching technology that allows you to transfer the spectral shape of one recording to another. In other words, you can copy the frequency balance of existing time-proven mixes so that other mixes can be improved. The filters of **CurveEQ** can be switched between linear-phase and minimum-phase modes. **CurveEQ** also features a customizable spectrum analyzer. Furthermore, you can display, save, and load static spectrum plots for comparison and matching purposes.

For detailed information about **CurveEQ** and its parameters, refer to the documentation provided by Voxengo at http://www.voxengo.com.

DeEsser

DeEsser reduces excessive sibilance, primarily for vocal recordings. It is a special type of compressor that is tuned to be sensitive to the frequencies produced by the s-sound.

Close proximity microphone placement and equalizing can lead to situations where the overall sound is just right, but there is a problem with sibilants.

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.

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 help 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

Freq (25 Hz to 20 kHz)

If the **Side-Chain** button 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.

Q-Factor

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

Side

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.

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.

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.

DualFilter

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

Position

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

Resonance

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

EnvelopeShaper

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

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

Attack (-20 to 20 dB)

Sets the gain of the attack phase of the signal.

Length (5 to 200 ms)

Sets the length of the attack phase.

Release (-20 to 20 dB)

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 knobs or drag the breakpoints in the graphical display to change the **Threshold** and the **Ratio** parameter values.

Threshold (-60 to 0 dB)

Determines the level where the expansion kicks in. Only signal levels below the set threshold are processed.

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.

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

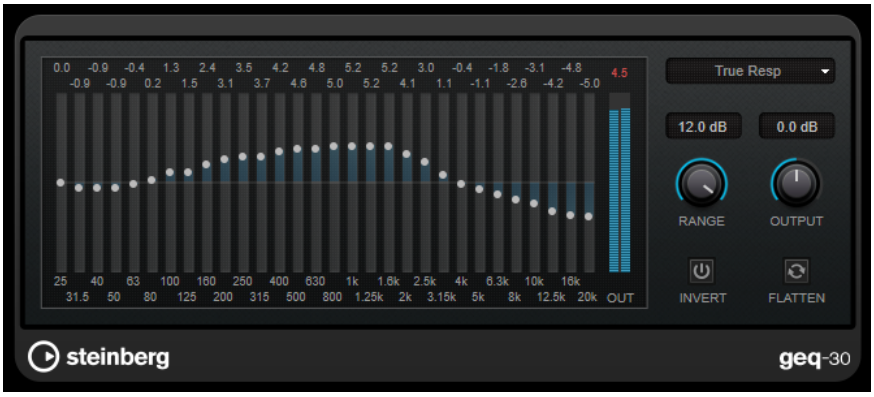
If the **Side-Chain** button is activated, this sets the resonance or width of the filter.

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.

GEQ-10/GEQ-30

These graphic equalizers are identical, except for the number of available frequency bands (10 and 30).



Each band can be attenuated or boosted by up to 12 dB, allowing for fine control of the frequency response. In addition, there are several preset modes available that can add color to the sound of **GEQ-10**/**GEQ-30**.

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

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

Output

Sets the overall gain of the equalizer.

Flatten

Resets all the frequency bands to 0 dB.

Range

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

Invert

Inverts the current response curve.

Mode pop-up menu

Allows you to set the filter mode that determines how the various frequency band controls interact to create the response curve.

EQ Modes

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

True Response

Applies serial filters with an accurate frequency response.

Digital Standard

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

Classic

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

VariableQ

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

ConstQ asym

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

ConstQ sym

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

Resonant

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

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.

L/R to M/S, M/S to L/R

This plug-in allows you to convert a stereo signal into a M/S signal and vice versa.

The **L/R to M/S** tool converts a L/R signal that is divided into a left and a right signal into a M/S signal that is divided into a mid signal (L+R) and side signals (L-R).

The **M/S to L/R** tool reconverts the M/S signal into a L/R signal.

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 signal and the wet signal.

Output

Sets the maximum output level.

Soft Clip

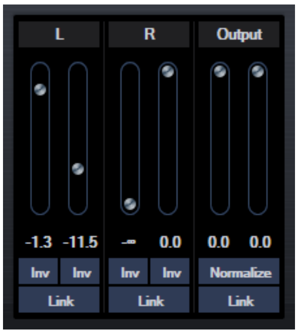
If this button is activated, **Maximizer** starts limiting or clipping the signal softly. At the same time, harmonics are generated, adding a warm, tube-like characteristic to the audio material.

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.

NOTE

This plug-in does not simulate a surround mix or add any psycho-acoustical artifacts to the resulting output – it is simply a mixer. The plug-in is only available in the **Master Section** and if a surround audio montage is active.



Surround Channels

Volume

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

Link

Link the volume faders.

Invert

Invert the phase of the left and right channel of the surround bus.

Output Bus

Volume

Set the volume of the of the mixed output.

Link

Links the **Output** faders.

Normalize

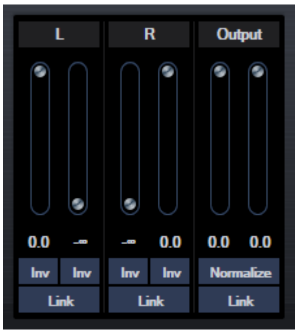
If this option is activated, the mixed output is normalized. For example, the output level is automatically adjusted so that the loudest signal is as loud as possible without clipping.

Mix8to2

Mix8to2 lets you quickly mix down your surround mix format to stereo. You can control the levels of up to eight surround channels and decide for each channel up to which level it is included in the resulting mix.

NOTE

This plug-in does not simulate a surround mix or add any psycho-acoustical artifacts to the resulting output – it is simply a mixer. The plug-in is only available in the **Master Section** and if a 8 channel audio montage is active.



Surround Channels

Volume

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

Link

Link the volume faders.

Invert

Invert the phase of the left and right channel of the surround bus.

Output Bus

Volume

Set the volume of the 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.

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.

MonoToStereo

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

NOTE

This plug-in works only on stereo tracks.

Width

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

Delay

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

Color

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

Mono

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

MultibandCompressor

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

NOTE

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

Frequency Band Editor

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

- To define the frequency range of the different frequency bands, use the handles at the sides of each frequency band.

- To attenuate or boost the gain of the frequency bands by ±15 dB after compression, use the handles at the top of each frequency band.

Live

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

Bypassing Frequency Bands

To bypass each frequency band, activate the **Bypass Band** button in each 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.

Output (-24 to 24 dB)

Sets the output level.

Compressor Section

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

Threshold (-60 to 0 dB)

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

Ratio

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

Attack (0.1 to 100 ms)

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

Release (10 to 1000 ms or Auto mode)

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

Side-Chain Section

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

IMPORTANT

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

Frequency

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

Q-Factor

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

Side-Chain

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

Monitor

Allows you to monitor the filtered signal.

MultibandEnvelopeShaper

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

Frequency Band Editor

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

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

Live

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

Bypassing Frequency Bands

To bypass each frequency band, activate the **Bypass Band** button in each 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.

Output (-24 to 24 dB)

Sets the output level.

Shaper Section

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

Attack (-20 to 20 dB)

Sets the gain of the attack phase of the signal.

Length (5 to 200 ms)

Sets the length of the attack phase.

Release (-20 to 20 dB)

Sets the gain of the release phase of the signal.

Sensitivity (-40 to -10 dB)

Sets the sensitivity of the detection.

Output

Sets the output level.

MultibandExpander

MultibandExpander allows a signal to be split into four frequency bands. You can reduce the output level in relation to the input level for signals below the set threshold for each band. This is useful if you want to enhance the dynamic range or reduce the noise in quiet passages.

Frequency Band Editor

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

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

Live

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

Bypassing Frequency Bands

To bypass each frequency band, activate the **Bypass Band** button in each 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.

Output (-24 to 24 dB)

Sets the output level.

Expander Section

You can specify the **Threshold** and **Ratio** by moving breakpoints or using the corresponding knobs. The first breakpoint from which the line deviates from the straight diagonal is the threshold point.

Threshold (-60 to 0 dB)

Determines the level where the expansion kicks in. Only signal levels below the set threshold are processed.

Ratio

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

Maximum Reduction

Sets the maximum amount by which the level is reduced when the signal falls below the set threshold.

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.

Output

Sets the output level.

Side-Chain Section

Frequency

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

Q-Factor

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

Side-Chain

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

Monitor

Allows you to monitor the filtered signal.

Octaver

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

Direct

Sets the level balance between the dry signal and the wet signal. A value of 0 means that only the generated and transposed signal is heard. By raising this value, more of the original signal is heard.

Octave 1

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

Octave 2

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

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.

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.

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.

PostFilter

This effect allows quick and easy filtering of unwanted frequencies, creating room for the important sounds in your mix.

PostFilter combines a low-cut filter, a notch filter, and a high-cut filter. You can make settings by dragging the curve points in the graphical display, or by adjusting the controls below the display section.

Level meter

Shows the output level, giving you an indication of how the filtering affects the overall level of the edited audio.

Low Cut Freq (20 Hz to 1 kHz, or Off)

Allows you to eliminate low-frequency noise. The filter is inactive if the curve point is located all the way to the left. You can set the frequency either in Hz or as a note value. If you enter a note value, the frequency is automatically changed to Hz. For example, a note value of A3 sets the frequency to 440 Hz. When you enter a note value, you can also enter a cent offset. For example, enter A5 -23 or C4 +49.

NOTE

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

Low Cut Slope

Allows you to choose a slope value for the low-cut filter.

Low Cut Preview

Use this button between the **Low Cut Freq** button and the graphical display to switch the filter to a complementary high-cut filter. This deactivates any other filters, allowing you to listen only to the frequencies that you want to filter out.

Notch Freq

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

NOTE

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

Notch Gain

Adjusts the gain of the selected frequency. Use positive values to identify the frequencies that you want to filter out.

Notch Gain Invert

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

Notch Q-Factor

Sets the width of the notch filter.

Notch Preview

Use the **Preview** button between the notch filter buttons and the graphical display to create a band-pass filter with the peak filter's frequency and Q. This deactivates any other filters, allowing you to listen only to the frequencies you want to filter out.

Notches buttons (1, 2, 4, 8)

These buttons add additional notch filters to filter out harmonics.

High Cut Freq (3 Hz to 20 kHz, or Off)

This high-cut filter allows you to remove high-frequency noise. The filter is inactive if the curve point is located all the way to the right. You can set the frequency either in Hz or as a note value. If you enter a note value, the frequency is automatically changed to Hz. For example, a note value of A3 sets the frequency to 440 Hz. When you enter a note value, you can also enter a cent offset. For example, enter A5 -23 or C4 +49.

NOTE

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

High Cut Slope

Allows you to choose a slope value for the high-cut filter.

High Cut Preview

This button between the **High Cut Freq** button and the graphical display allows you to switch the filter to a complementary low-cut filter. This deactivates any other filters, allowing you to listen only to the frequencies you want to filter out.

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

Lo Freq

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

Hi Freq

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

Lo Gain

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

Hi 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.

Reverb Time

Allows you to set the reverb time in seconds.

Size

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

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

Lo Freq

Determines the frequency below which low-frequency damping occurs.

Hi Freq

Determines the frequency above which high-frequency damping occurs.

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.

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

Indicates the level of the output signal.

RoomWorks SE

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

Pre-Delay

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

Reverb Time

Allows you to set the reverb time in seconds.

Diffusion

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

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

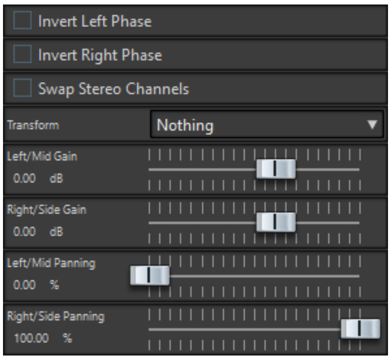
Switches the output to mono, to check for possible unwanted coloring of the sound which sometimes can occur when enhancing the stereo image.

Color

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

Stereo Tools

Stereo Tools allows you to pan or position both the left and right channels independently of one another. You can use it for stereo files that you do not want to convert to mono or for fixing a problem with the stereo file, for example.



Invert Left Phase/Invert Right Phase

Inverts the polarity of an audio channel. You can use it for removing the center information or for fixing a channel that has been inverted, for example.

Swap Stereo Channels

Swaps the left and right channels.

Transform

Determines the conversion method:

- **Nothing**: No conversion of the signal.
- **Left/Right -> Mid/Side**: Converts a stereo signal into a mid/side signal.
- **Mid/Side -> Left/Right**: Converts a mid/side signal into a stereo signal.

Left/Mid Gain (dB)

Sets the gain of the left stereo signal or the mid signal of the M/S signal.

Right/Side Gain (dB)

Sets the gain of the right stereo signal or of the side signals of the M/S signal.

Left/Mid Panning (%)

Pans the left stereo signal or the mid signal of the M/S signal.

Right/Side Panning (%)

Pans the right stereo signal or the side signals of the M/S signal.

StudioChorus

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

Delay

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

Width

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

Spatial

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

Mix

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

Rate

Allows you set the sweep rate freely with the **Rate** knob.

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**.

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.

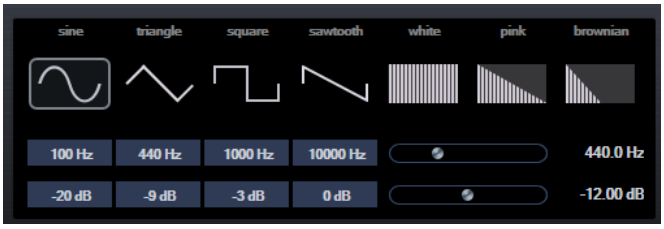
Filter type

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**.

TestGenerator

This utility plug-in allows you to generate an audio signal, which can be recorded as an audio file.



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

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

The **TestGenerator** is based on a waveform generator which can generate a number of basic waveforms such as sine and saw as well as various types of noise. Furthermore, you can set the frequency and amplitude of the generated signal. As soon as you add the **TestGenerator** as an effect on an audio track and activate it, a signal is generated. You can then activate recording as usual to record an audio file according to the signal specifications.

Waveforms and noise section

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

Frequency section

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

NOTE

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

Gain section

Allows you to set the amplitude of the signal. The higher the value, the stronger the signal. You can select one of the preset values, or use the slider to set a value between -81 and 0 dB.

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.

VintageCompressor

VintageCompressor is modeled after vintage type compressors.

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

Input

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

Output (-48 to 24 dB)

Sets the output gain.

Attack (0.1 to 100 ms)

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

Punch

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

Release (10 to 1000 ms or Auto mode)

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

VU Meter

Shows the amount of gain reduction.

In/Out Meters

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

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 -oo, 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.

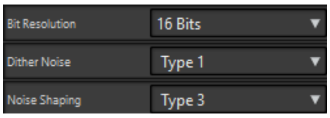
Their use with new audio projects is not recommended and they are not documented.

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

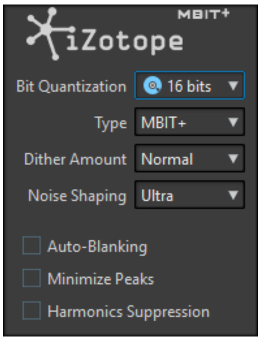
Allows you to specify the intended bit resolution for the final audio, after dithering, regardless of whether you want to render the settings or play back in real-time.

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.

MBIT+™ Dithering

This plug-in allows you to convert and dither to 24, 20, 16, 12, or 8 bits. This is useful for mastering a track for a CD (16-bit) from a 24-bit source, for example.

The **MBIT+™** dither algorithm reduces quantization distortion with minimal perceived noise and produces smooth and quiet conversions.



Bit Quantization

Sets the bit depth to which you are dithering. **MBIT+**™ produces 32-bit floating-point output, but all low-order bits will be zero and should be truncated.

Type

Sets the type of dithering. **MBIT+**™ contains two traditional dithering types and a proprietary **MBIT+**™ dithering type.

- **Type 1** is a traditional dither based on a rectangular probability distribution function (PDF).
- **Type 2** is a traditional dither based on a triangular PDF.
- **MBIT+**™ offers superior results when used with all types of source material.

Dither Amount

When using **MBIT+**™ dither, this controls the amount of dithering. The **None** and **Low** settings can leave some non-linear quantization distortion or dither noise modulation, while higher settings completely eliminate the non-linear distortion at the expense of a slightly increased noise floor. The **Normal** setting is suffice for most cases.

When using Type 1 or Type 2 dither, this controls the number of bits used to perform dithering. In most cases, 1 bit is suffice, but over-dithering with 2 bits can be useful in same cases.

Noise Shaping

When using **MBIT+**™ dither, this controls the amount of noise shaping. The choices range from no noise shaping to very aggressive noise shaping, providing approximately 14 dB of audible noise suppression, at the expense of a higher noise floor.

When using Type 1 or Type 2 dither, this controls the noise shaping. Dithering noise can be shaped in order to make it less audible. Simple noise shaping performs simple high-pass filtering on the noise. Clear noise shaping aggressively moves the noise toward the Nyquist frequency. **Psych 5** is a 5th-order filter designed to move noise away from audible bands, and **Psych 9** is a 9th-order filter with similar characteristics.

Auto-Blanking

If this option is activated, **MBIT+**™ mutes dither output when the input is completely silent for at least 0.7 seconds.

Minimize Peaks

If this option is activated, spurious peaks in the noise-shaped dither are suppressed.

Harmonics Suppression

If this option is activated, the truncation rules are slightly altered, moving the harmonic quantization distortion away from overtones of audible frequencies. This option does not create any random dithering noise floor. Instead it works more like truncation, but with better tonal quality in the resulting signal. This option is applicable only in the modes without dithering noise and without aggressive noise shaping.

UV22HR

UV22HR is an advanced version of Apogee's renowned UV22 dithering algorithm, capable of dithering to 8, 16, 20, or 24 bits.

8, 16, 20, 24 bit

These buttons allow you to select the intended bit resolution for the final audio. As when using the internal dithering, it is important to set this to the correct resolution.

Hi

Applies a normal dither gain.

Lo

Applies a lower level of dither noise.

Auto black

If this option is activated, the dither noise is gated during silent passages.

ASIO Plug-ins

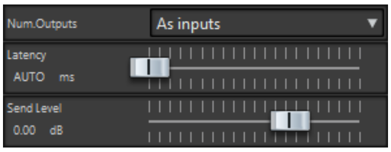
External Gear

This **Master Section** plug-in allows you to process audio files using external hardware processors. One or more ASIO outputs are used to send the audio signal to your processor, and corresponding ASIO inputs are used to return the signal from the external processor.

By default, this plug-in is located in the ASIO submenu of the **Master Section** effects. An ASIO driver must be used, and only one instance of this plug-in is allowed in the **Master Section** plug-in chain.

External Gear Plug-in

In the **Effects** pane of the **Master Section** window, select the **External Gear** plug-in from the **ASIO** submenu.



Num. Outputs

Here you can set the number of outputs to use. Normally this is the same as the number of inputs (the **As Inputs** option). However, you can use a mono out/stereo in configuration in which case you set this parameter to 2 with the slider.

Latency

External gear may introduce latency. WaveLab can automatically compensate for this if you select **Auto** (only active during rendering), or you can set this latency compensation yourself (in milliseconds). The latency introduced by the ASIO driver is automatically taken into account by WaveLab.

Send Level

Allows you to adjust the send level. This should normally be set to 0 dB. If necessary, adjust the input level on the external effect.

Using External Gear

PROCEDURE

1. Select **File** > **Preferences** > **VST Audio Connections**.
2. Set the **Audio Device** to **ASIO**.
3. Select the **ASIO Plug-ins** tab.
4. Select the channels to be used for device output (to gear) and device input (from gear), and click **OK**.

These should be different I/O channels than the ones you use for playback and recording. The available outputs in this plug-in equals the number of inputs (up to 8).

5. In the **Effects** pane of the **Master Section** window, select the **External Gear** plug-in from the **ASIO** submenu.

   The **External Gear** plug-in window opens.

6. Make your settings.

AFTER COMPLETING THIS TASK

Now you can process a signal through the external processor, just as if it was a software plug-in effect. If you render a file using the External Gear plug-in, playback is not available during the rendering.

Audio Input

This is a special **Master Section** plug-in that allows you to render a signal coming in to the inputs of a sound card together with any **Master Section** effects. This signal can be anything your sound card accepts, such as a feed from a mixer, a recorder, or a microphone.

By default, this plug-in is located in the ASIO submenu of the **Master Section** effects. An ASIO driver must be used, and only one instance of this plug-in is allowed in the **Master Section** plug-in chain.

When the Audio Input plug-in is loaded, wave playback is not possible.

Setting Up the Audio Input Plug-in

PROCEDURE

1. Select **File** > **Preferences** > **VST Audio Connections**.
2. Set the **Audio Device** to **ASIO**.
3. Select the **ASIO Plug-ins** tab.
4. Select the channels to be used for device input, and optionally name them.
5. In the top effect slot of the **Effects** pane of the **Master Section** window, select the **Audio Input** plug-in from the **ASIO** submenu.
6. In the **Audio Input** plug-in window, set the number of inputs and the sample rate.

   You have to match the number of inputs in the **VST Audio Connections** tab to the number of inputs selected here.

7. Start playback.

   The cursor does not move, but the **Play** button is lit and you can now monitor the input source. Pressing **Stop** ends input monitoring.

8. If you change the settings in the control panel, click **Stop**, and restart playback to apply them.
9. In the **Master Section**, click **Render**.
10. Select a name, an audio format, and a location for the file to be rendered.
11. Click **Start**.

    Recording/rendering starts, recording the external input from the output of the **Master Section**, including all real-time processing. You can monitor the recording as it happens.

12. Click **Stop** to stop the recording/rendering.

Batch Processing Plug-ins

In the **Batch Processor** window, you can add a sequence of plug-ins that can be used to process a batch of audio files. These plug-ins can be standard plug-ins available from the **Master Section**, offline processes available in the **Audio Editor**, and plug-ins that are only available within batch processing.

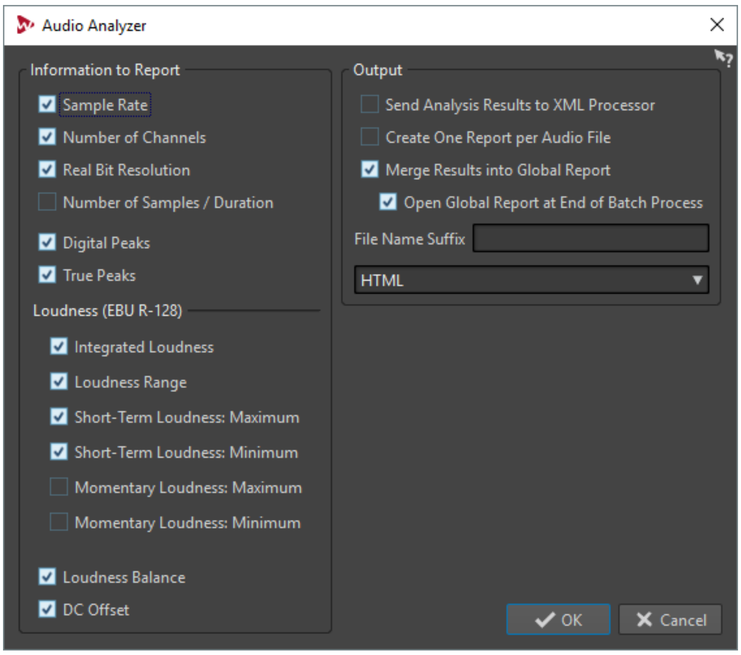
The following batch processor plug-ins are described in the WaveLab Pro **Operation Manual**:

- **Loudness Normalizer**
- **Pitch Quantize**
- **Pitch Correction**
- **Pan Normalizer**
- **Time Stretch**

Audio Analyzer

This plug-in allows you to generate text files with statistics about the audio files in a batch process.

This monopass plug-in is exclusive to the **Batch Processor** window.



If you want to analyze files without writing anything, select **No Output** in the **Output** tab of the **Batch Processor** window.

Information to Report

In this section you specify which information to include in the output. The following information can be included:

- Sample rate
- Number of channels
- Real bit resolution
- Number of samples/duration
- Digital peaks
- True peaks
- Integrated loudness
- Loudness range
- Short-term loudness (maximum)
- Short-term loudness (minimum)
- Momentary loudness (maximum)
- Momentary loudness (minimum)
- Loudness balance
- DC Offset

Output

In this section you set up the output of the **Audio Analyzer**. The following options are available:

Send Analysis Results to XML Processor
If this option is activated, the analysis results are passed as parameters to the XML or HTML output of the batch processor.

Create One Report per Audio File
If this option is activated, one report is created for each audio file in the batch process. The audio file name is used as the report file name.

Merge Results into Global Report
If this option is activated, the results of the analysis are merged into a global report. The audio file name is used as the report file name.

Open Global Report at End of Batch Process
If this option is activated, a global report opens after the batch process.

File Name Suffix
Lets you specify a file name suffix. This is necessary when you are using this plug-in multiple times in a batch process, for example, to see the stats before and after specific plug-ins.

Use different suffixes for each instance of the **Audio Analyzer** plug-in used in the processing chain.

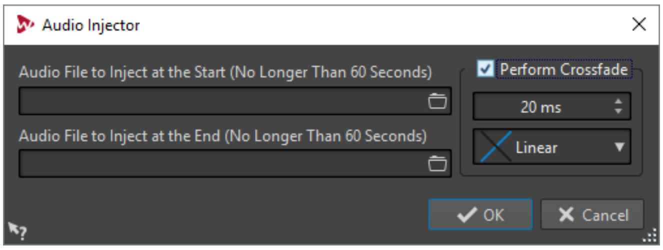
Output format
Lets you select the output format. The following formats are available:

- Pure text
- HTML
- Adobe PDF
- Open Office
- Spreadsheet

- XML

Audio Injector

This plug-in allows you to insert an audio file at the beginning and/or end of the audio file being processed. The inserted file can also be crossfaded with the original audio file, if required.



This monopass plug-in is exclusive to the **Batch Processor** window.

Audio File to Inject at the Start (No Longer Than 60 Seconds)
Specifies the audio file to be added before the main audio file.

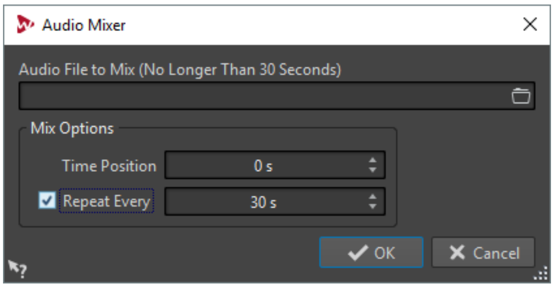
Audio File to Inject at the End (No Longer Than 60 Seconds)
Specifies the audio file to be added after the main audio file.

Perform Crossfade
Allows you to select a crossfade time and shape for the crossfade between the main audio file and the injected audio file.

Audio Mixer

This plug-in allows you to mix an audio file with other audio files. The mix-in happens from a specified time and can optionally be repeated at a specified interval.

For example, you can insert a spectral watermark into the audio spectrum or you can insert beep sounds to mark an audio file as demo material.



This monopass plug-in is exclusive to the **Batch Processor** window.

Audio File to Mix (No Longer Than 30 Seconds)
Allows you to select the audio file that you want to mix into other audio files. The audio file must be no longer than 30 seconds.

Time Position
Allows you to specify the time position at which the audio file will be mixed in.

Repeat Every
If this option is activated, you can specify the time after which the audio file is repeatedly mixed in.

DC Remover

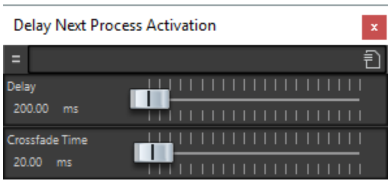
This plug-in allows you to remove any DC Offset from an audio file.

It is useful to apply this plug-in first in a batch before other plug-ins to avoid further processing a file containing any DC Offset. For example, an audio file that has a DC offset is not at its loudest possible volume when normalized, because the offset consumes headroom.

This multipass plug-in is available in the **Batch Processor** window and as an offline processor in the **Audio Editor**.

Delay Next Process Activation

This plug-in allows you to delay the processing of the next VST plug-in in the plug-in chain for a certain time.



This monopass plug-in is exclusive to the **Batch Processor** window.

Delay

Specifies the duration until the wet signal starts to fade-in the dry signal.

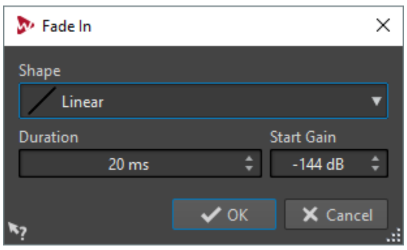
Crossfade Time

Specifies the duration of the crossfade time.

Fade In/Fade Out

This plug-in allows you to fade the beginning (**Fade In**) or the end (**Fade Out**) of a batch audio file. You can choose the length and shape of the fade, its duration, and the gain you want it to start/end at.

The fade plug-ins are exclusive to the **Batch Processor** window. **Fade In** is a monopass plug-in and **Fade Out** is a multipass plug-in.



Shape

Determines the shape of the fade.

Duration

Determines the duration of the fade.

Start Gain/End Gain

Determines the gain with which the fade starts. It ends with 0 dB.

Instructor

Instructor is a special utility plug-in that allows you to instruct the next plug-in in the batch with information about the audio it needs to process. This is useful for situations where you want to use monopass plug-ins that require an analysis stage that is not available at this point.

In effect, the **Instructor** plug-in turns a monopass plug-in into a dual pass one. Some monopass plug-ins, such as **DeNoiser** or **DeBuzzer**, need to learn about the audio they are to process before they can begin processing correctly. The **Instructor** plug-in can help in this situation, because it can teach the next plug-in in the audio chain about the audio it is about to process.

The **Instructor** plug-in must be used as a pair:

1. The first instance replicates the start of the audio stream. This means the next plug-in in the chain receives the start of the audio stream twice.
2. The second instance of the plug-in comes after the plug-in being instructed. It cuts out the extra audio injected by the first instance of the Instructor plug-in.

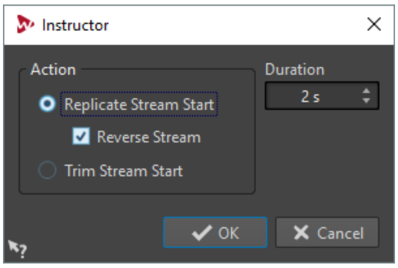
For example, this means that the **DeNoiser** plug-in has time to sufficiently analyze the audio stream before the second stream start is injected. The “badly” processed first part of the stream is skipped by the second instance of the **Instructor** plug-in.

You can set the **Instructor** plug-in to replicate up to 20 seconds of audio.

NOTE

Do not set a value that is longer than the shortest file in the batch, otherwise a short file is over truncated by the second instance of the plug-in.

This monopass plug-in is exclusive to the **Batch Processor** window.



Replicate Stream Start

Injects the start of the audio stream twice into the next plug-ins. This action must be selected for the first instance of the **Instructor** plug-in.

Reverse Stream

If this option is activated, the start of the stream is injected first in reverse sample order, then in normal sample order. This changes nothing from the spectrum analysis point of view, but it improves the transition between the repeated streams.

Trim Stream Start

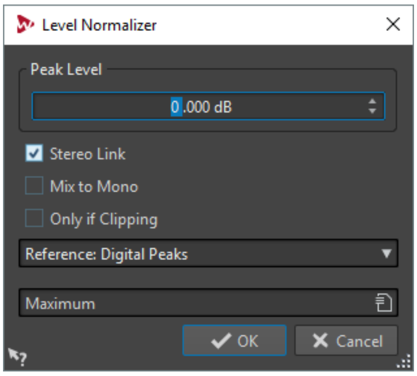
Skips the start of the audio stream. This action must be selected for the second instance of the **Instructor** plug-in.

Duration

Specifies how much audio to replicate or skip.

Level Normalizer

This multipass plug-in allows you to raise or lower levels so that the signal peaks exactly at the specified value just before it is converted to a file.



Peak Level

Specify the highest level of any audio sample.

Stereo Link

Applies the gain to both channels.

Mix to Mono

Mixes the left and right channels. The resulting mono file gets the specified peak level. This ensure a clip-less mix.

Only if Clipping

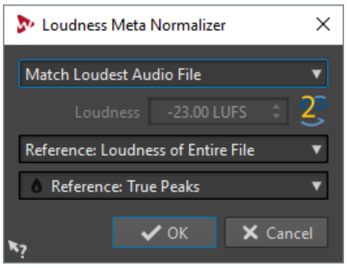
Only applies a gain change if the audio file is beyond the reference peak level at some point. If not, the signal is untouched.

Loudness Meta Normalizer

This plug-in allows you to normalize a batch of files to the same loudness, while taking the EBU R-128 loudness measurement and a true peak analysis into account.

The purpose of this plug-in is to achieve the same loudness in all files (the highest loudness found, if possible), while being certain that no file clips. For each file, a specific gain is computed by the plug-in once all files have been analyzed and prior to actually applying any gain to achieve the common loudness. If it is not possible to match the highest found loudness, the level of the file with the highest loudness is reduced, so that other files can get the same loudness. Because no peak compression is used, the dynamics is preserved and no distortion is introduced.

This metapass plug-in is exclusive to the **Batch Processor** window.



Match Loudness

Select what loudness the clip should get. The following options are available:

- Match loudest audio file

- Match maximum achievable loudness
- Match specific loudness

Loudness

Determines the specific loudness to match. For example, -23 LUFS if you want to follow the EBU R-128 recommendation for broadcast.

Reference

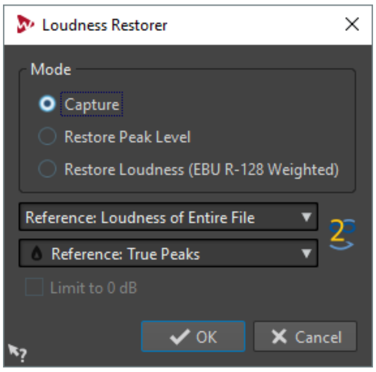
Select whether WaveLab should use as reference the loudness of the entire clip (EBU R-128 recommendation), the average loudest 3 second audio section (**Top of Loudness Range**), or the loudest 3 seconds audio section (**Maximum Short-Term Loudness**).

Peaks

Select whether WaveLab should refer to sample values (digital peaks) or to analog reconstructed values (true peaks).

Loudness Restorer

Loudness Restorer captures the loudness at a specific point in the audio chain and restores that loudness at another point. For this reason, the **Loudness Restorer** must be inserted in pairs into the signal chain: one plug-in for capturing and one plug-in for restoring.



This multipass plug-in is exclusive to the **Batch Processor** window.

Mode - Capture

The first instance in the plug-in pair must be set to this mode. This makes the plug-in read the signal at this position in the audio chain.

Mode - Restore Peak Level/Restore Loudness (EBU R-128 Weighted)

The second instance in the plug-in pair must be set to one of these modes. Select one of these options if you want to use peak levels as a basis for determining what is considered equal level. **Restore Loudness (EBU R-128 Weighted)** produces a more natural result than **Restore Peak Level**.

Reference menu

Select whether WaveLab should use as reference the loudness of the entire file (EBU R-128 recommendation), the average loudest 3 second audio section (**Top of Loudness Range**), or the loudest 3 seconds audio section (**Maximum Short-Term Loudness**).

Peak menu

Select whether WaveLab should use sample values (**Digital Peaks**) or analog reconstructed values (**True Peaks**).

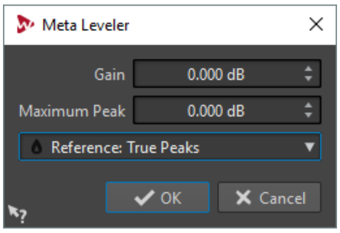
Limit to 0 dB

If this option is activated, the restoration process will never result in levels above 0 dB.

Meta Leveler

This plug-in allows you to change the level of a batch of files consistently.

The core purpose of this plug-in is to apply the same gain to all files, while being certain that a specific peak level never exceeds in any file. The unique gain that you want to apply is (possibly) reduced by the plug-in, once all files in the batch have been analyzed, and prior to actually applying the gain across the batch.



This metapass plug-in is exclusive to the **Batch Processor** window.

Gain

Applies the specified gain to each file. The actual gain can be lower and even negative, to not exceed the value specified in the **Maximum Peak** field.

Maximum Peak

Specifies the maximum peak level that any audio file should get at the end of the process.

Peaks

Select whether WaveLab should refer to sample values (digital peaks) or to analog reconstructed values (true peaks).

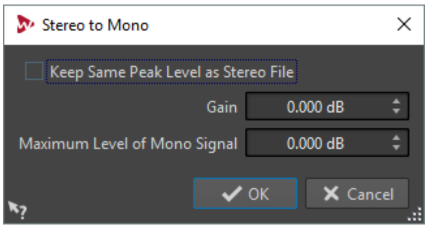
Resizer

This plug-in allows you to specify the duration of all audio files in the batch, and to choose whether to insert silence after the end of the chosen duration.

This monopass plug-in is exclusive to the **Batch Processor** window.

Stereo to Mono

This plug-in allows you to mix a stereo signal down to a mono signal while being certain not to clip while mixing channels because of the multipass approach. You can choose to use the same peak level that the stereo file contains, or set the gain to be applied and/or the maximum level to be reached in the resulting mono file.



This multipass plug-in is exclusive to the **Batch Processor** window.

Keep Same Peak Level as Stereo File

If this option is activated, the peak level of the resulting mono file is the same as the peak level of the original stereo file.

Gain

Specifies the increase or decrease in peak level for the resulting mono file, in relation to the original stereo file.

Maximum Level of Mono Signal

Specifies the peak level that the resulting mono file must not exceed. This ensure that the output file never clips. This way, the result never exceeds 0 dB, regardless of the specified **Gain** value.

Trimmer

This plug-in allows you to remove a specified duration (from 0 ms to 60 s) of audio from the head and/or tail of an audio file.

This monopass plug-in is exclusive to the **Batch Processor** window.

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