WAVELAB PRO 12

Audio Editing And Mastering Suite

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This document provides improved access for people who are blind or have low vision. Please note that due to the complexity and number of images in this document, it is not possible to include text descriptions of images.

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Table of Contents

VST Plug-ins

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

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

Analyzer

SuperVision

SuperVision is a professional tool suite for monitoring and analyzing your audio. The plug-in comes with several different modules for level, spectral, phase, or waveform analysis. Up to nine module slots allow you to create custom layouts for better overview.

SuperVision features two different processing modes: **Maximum Audio Performance** and **Sample-Accurate Display**. You can choose which mode to use for each module independently.

Toolbar

Pause Measurement

Pauses/Continues the measurement for the selected module. **Alt/Opt** -click this button to pause/continue all modules at the same time.

- You can also pause/continue the measurement for the selected module by right-clicking it.
- If a module is paused, you can still adjust the graphical display that holds the last measured values.
- For all modules that show a playback cursor, you can click on the paused display to set the project cursor to this position. This is not possible during recording.

Hold Current Values on Stop

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

Module selector
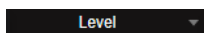
Allows you to select a module for the selected slot.

Open Module Settings

Opens the **Module Settings** window. It provides the settings for the selected module.

Channel selector
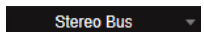
Allows you to select which channels are displayed. The available channel configurations depend on the channel configuration of the track and on the selected module. **Mixdown** allows you to display the average value of all channels within the track.

NOTE

- The channel selector is only available for configurations with two or more channels.

Reset Module Values

Resets the measured values of the selected module. **Alt/Opt** -click to reset all modules at the same time.

NOTE

You can also reset the measured values of a module by **Ctrl/Cmd** -clicking it.

Reset Module Values on Start

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

Split Horizontally

Splits the selected module slot horizontally.

NOTE

This button is not available if a module is maximized.

Split Vertically

□

Splits the selected module slot vertically.

NOTE

This button is not available if a module is maximized.

Module Slot Controls

Each module slot shows the following controls in the top right corner if you move the mouse over it:

Remove module slot

⊠

Removes the module slot from the current plug-in layout.

Split horizontally

▣

Splits the module slot horizontally.

Split vertically

▣

Splits the module slot vertically.

You can maximize a module by double-clicking it. To downsize it, double-click it again or click the standard view button ▣.

If your layout shows more than one module, you can change the focus by clicking a module or pressing **Tab** .

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

You can show the current frames per second (fps) for all modules by pressing **Alt/Opt** - **F** .

RELATED LINKS
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Module Settings Window

In the **Module Settings** window, you can make individual settings for the selected module.

● To open the **Module Settings** window, click **Open Module Settings** ⚙ on the plug-in toolbar.

The settings on the **Module Settings** window toolbar are available for all modules:

Reset Settings

↺

Resets all parameter settings to the default values of the selected module.

Maximum Audio Performance/Sample-Accurate Display

Sets the processing mode for the selected module.

If this button is activated, **Maximum Audio Performance** is selected. In this mode, the plug-in has no impact on the audio performance, but the analysis might not be sample-accurate.

If this button is deactivated, **Sample-Accurate Display** is selected. In this mode, no audio sample gets lost for analysis, but the audio performance might be slightly reduced.

NOTE

Sample-Accurate Display is not available for all modules.

Enable Warnings

If this button is activated, a red border around the affected module indicates that the displayed analysis result might not be completely sample-accurate.

NOTE

This setting is only available in **Maximum Audio Performance** mode.

Force Horizontal Display

If this button is activated, the module is always displayed horizontally when you resize it.

NOTE

This setting is not available for all modules.

Force Vertical Display

If this button is activated, the module is always displayed vertically when you resize it.

NOTE

This setting is not available for all modules.

For the specific settings of a module, see the corresponding module description.

By default, only the top row of controls is displayed. **Advanced** shows/hides all settings for a module.

RELATED LINKS

Signal Modules

The modules in this category allow you to visualize the level of the audio signal.

The following modules and module-specific settings are available:

Level

This module shows the level of your audio. It provides a multi-channel level meter and a maximum level value display.
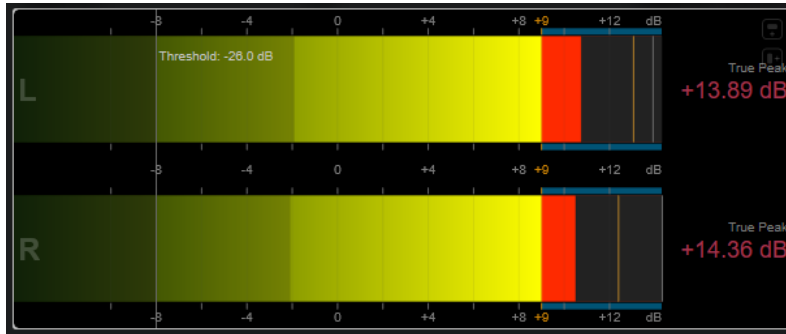
The following module-specific settings are available in the **Module Settings** window:

Scale

Allows you to select a scale according to different broadcast standards (**Internal**, **Digital**, **DIN**, **EBU**, **British**, **Nordic**, **K-20**, **K-14**, **K-12**, **+3 dB Digital**, **+6 dB Digital**, or **+12 dB Digital**).

Peak Hold

Specifies for how long the peak levels are held on the display.

Peak Fallback

Sets the speed of release for the level meters and peak indicators.

> NOTE
>
> - Alternatively, you can move the mouse pointer over the selected module, hold ⎸, and use the mouse wheel to adjust this parameter.
> - If this control is turned all the way to the left, the peak indicators are disabled.

Threshold

Sets a threshold level below which the display is masked.

> NOTE
>
> Alternatively, you can move the mouse pointer over the selected module and use the mouse wheel to adjust this parameter.

Offset

Sets the offset between measured and displayed value in dB.

This parameter is only available for the **DIN**, **EBU**, **British**, and **Nordic** scale.

Clipping

Sets the clipping value for the **Internal** scale.

Minimum

Sets the minimum value for the **Internal** scale.

Maximum

Sets the maximum value for the **Internal** scale.

Color

Sets the color of the meters. You can choose between the **Scale** color and the **Track** color.

RMS AES17

Shows the level in accordance with AES17 (RMS +3 dB).

RMS Resolution

Sets the RMS resolution for the level display in milliseconds.

Max. Value

Sets the measurement mode for the maximum level value display. The following modes are available:

- **True Peak** shows estimated real peak value for each channel.
- **Peak Max.** shows maximum sample value for each channel.
- **RMS Max.** shows the maximum RMS value for each channel.
- **RMS Max. + True Peak** shows the highest maximum RMS and the highest estimated real peak value of all channels.
- **RMS Max. + Peak Max.** shows the highest maximum RMS and the highest maximum sample value of all channels.

VU

This module shows the level of your audio on a classic VU meter. In addition to the VU meter needle and the LED peak indicator, it provides a peak level indicator needle and a numeric maximum level value display.
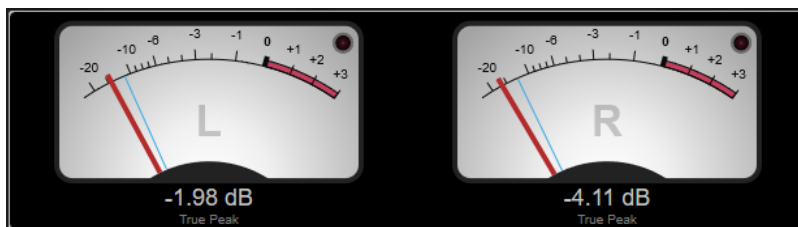
The following module-specific settings are available in the **Module Settings** window:

Scale

Allows you to select a scale according to different broadcast standards (**Internal**, **Digital**, **DIN**, **EBU**, **British**, **Nordic**, **K-20**, **K-14**, **K-12**, **+3 dB Digital**, +6 dB Digital, **+12 dB Digital**, **VU dB**, and **VU dBFS**).

Peak Hold

Specifies for how long the peak levels are held on the display.

Peak Fallback

Sets the speed of release for the level meters and peak indicators.

> NOTE
>
> - Alternatively, you can move the mouse pointer over the selected module, hold ⬛, and use the mouse wheel to adjust this parameter.
> - If this control is turned all the way to the left, the peak indicators are disabled.

Meter Mode

Sets the behavior of the needle.

- **VU** mode imitates the physical behavior of an analog VU meter that shows the current peak value.
- **Peak** mode shows the current peak value.
- **RMS** mode shows the current RMS value.

Offset

Sets the offset between measured and displayed value in dB.

This parameter is only available for the **DIN**, **EBU**, **British**, and **Nordic** scale.

Clipping

Sets the clipping value for the **Internal** scale.

Minimum

Sets the minimum value for the **Internal** scale.

Maximum

Sets the maximum value for the **Internal** scale.

Color

Sets the color of the meters. You can choose between the **Track** color and a **Dark** or **Light** scheme.

RMS AES17

Shows the level in accordance with AES17 (RMS + 3 dB).

RMS Resolution

Sets the RMS resolution for the level display in milliseconds.

Max. Value

Sets the measurement mode for the maximum level value display. The following modes are available:

- **True Peak** shows the estimated real peak value for each channel.
- **Peak Max.** shows the maximum sample value for each channel.
- **RMS Max.** shows the maximum RMS value for each channel.
- **RMS Max. + True Peak** shows the highest maximum RMS and the highest estimated real peak value of all channels.
- **RMS Max. + Peak Max.** shows the highest maximum RMS and the highest maximum sample value of all channels.

Level Histogram

This module shows a histogram for the peak or RMS value of the input level.
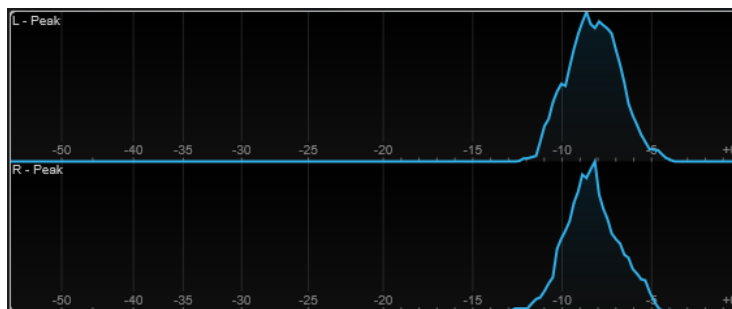
The following module-specific settings are available in the **Module Settings** window:

Scale

Allows you to select a scale according to different broadcast standards (**Internal**, **Digital**, **DIN**, **EBU**, **British**, **Nordic**, **K-20**, **K-14**, **K-12**, **+3 dB Digital**, **+6 dB Digital**, or **+12 dB Digital**).

Meter Mode

Sets the displayed level value.

- **Peak** mode shows a histogram of the peak value.
- **RMS** mode shows a histogram of the RMS value.

Peak Fallback

Sets the speed of release for the level meters and peak indicators.

NOTE

- If you change this parameter during playback, you must click **Reset Module Values** to update the display.
- Alternatively, you can move the mouse pointer over the selected module, hold ⎵, and use the mouse wheel to adjust this parameter.
- If this control is turned all the way to the left, the peak indicators are disabled.

Offset

Sets the offset between measured and displayed value in dB.

This parameter is only available for the **DIN**, **EBU**, **British**, and **Nordic** scale.

Clipping

Sets the clipping value for the **Internal** scale.

Minimum

Sets the minimum value for the **Internal** scale.

Maximum

Sets the maximum value for the **Internal** scale.

RMS AES17

Shows the level in accordance with AES17 (RMS +3 dB).

RMS Resolution

Sets the RMS resolution for the level display in milliseconds.

Smooth

Smoothes the display of the level curve.

NOTE

Alternatively, you can move the mouse pointer over the selected module and use the mouse wheel to adjust this parameter.

Measurement Modules

The modules in this category allow you to measure the loudness and intelligibility of the audio signal.

The following modules and module-specific settings are available:

Leq(m)/Leq(a)

These modules show the average volume over time, using a filter that emphasizes the mid and upper range frequencies. The Leq(m) measurement uses an M-weighting, according to the Trailer Audio Standards Association (TASA). It is mainly used to ensure that a film trailer respects the volume limits for trailers in movie theaters. The similar Leq(a) measurement uses an A-weighting and is used to verify the conformity of the sound for broadcasting.

The following module-specific setting is available in the **Module Settings** window:

Ref. Level

Sets the reference level above which the Leq(m)/Leq(a) value turns red, indicating that the volume limit is exceeded.

LFE

Sets the calibration of the low-frequency effects (LFE) channel during the measurement. You can choose between +0 dB (default value) and +10 dB (value according to ISO 21727) boost.

Loudness

This module shows the loudness of your audio in LU (Loudness Units) or LUFS (Loudness Units referenced to Full Scale) according to EBU R 128.

TP (True Peak)

Shows the maximum true peak level in dB.

I (Integrated)

Shows the integrated loudness value. This is the average loudness value that is measured over the whole audio range in LU or LUFS.

S (Short-Term)

> Shows the short-term loudness value that is measured every second on an audio block of 3 seconds in LU or LUFS. This gives information about the loudest audio passages.

M Max. (Momentary Max.)

> Shows the maximum value of all momentary loudness values that are measured every 100 ms in an audio range of 400 ms in LU or LUFS.

R (Range)

> Shows the loudness range (LRA) that is measured over the whole audio range in LU.

> The loudness range reports the ratio between the loudest and the quietest non-silent sections. The audio is divided into small blocks. There is one audio block every second and each block lasts 3 seconds, so that the analyzed blocks overlap. The top 10 % of the quiet blocks and the top 5 % of the loud blocks are excluded from the final analysis. The calculated loudness range is the ratio between the loudest and quietest remaining audio blocks. This measurement helps you to decide how much compression or expansion must be applied to the audio.

> An asterisk (*) after a loudness range value indicates that less than 1 minute of audio was analyzed.

> **Min.** shows the minimum loudness range value in LU. **Max.** shows the maximum loudness range value in LU.

> NOTE

> EBU R 128 does not recommend loudness range measurement for audio shorter than 1 minute due to too few data points.

PLR (Peak-to-Loudness)

> Shows the peak-to-loudness ratio (PLR), also referred to as the crest factor, which is the difference between the maximum true peak level value and the integrated loudness value.

Time

> Shows the overall duration of the loudness measurement.

The following module-specific settings are available in the **Module Settings** window:

Unit

> Allows you to switch the meter scale between LUFS (absolute values) and LU (relative values).

Scale

> Allows you to set the meter to either EBU +9 scale (linear), EBU +18 scale (linear), or +23 scale (logarithmic).

Ref. Integrated

> Sets a reference value for the integrated loudness. If higher values are detected, the loudness meter indicates clipping.

Tol. Integrated

> Sets a tolerance value for the integrated loudness.

Ref. True Peak

> Sets a reference value for the true peak level. If higher values are detected, the loudness meter indicates clipping.

Tol. True Peak

Sets a tolerance value for the true peak level.

Ref. Short-Term

Sets a reference value for the short-term loudness. If higher values are detected, the loudness meter indicates clipping.

Tol. Short-Term

Sets a tolerance value for the short-term loudness.

Ref. Momentary

Sets a reference value for the maximum momentary loudness. If higher values are detected, the loudness meter indicates clipping.

Tol. Momentary

Sets a tolerance value for the maximum momentary loudness.

Ref. Range

Sets a reference value for the loudness range. If higher values are detected, the loudness meter indicates clipping.

Tol. Range

Sets a tolerance value for the loudness range.

Loudness (Netflix)

This module provides a dialogue-gated loudness measurement that uses the Dolby Dialogue Intelligence algorithm according to ITU-R BS.1770 and shows the loudness of your audio in LU (Loudness Units) or LUFS (Loudness Units referenced to Full Scale). The **Dialogue** value allows you to assess the percentage of speech-based sequences within the audio.
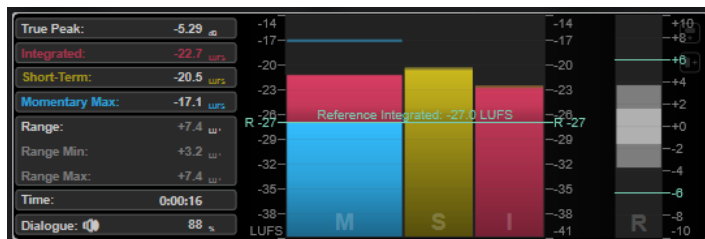
NOTE

When using the **Loudness (Netflix)** measurement, the Dolby Dialogue Intelligence algorithm adds a latency of 2.048 seconds compared to the standard **Loudness** measurement.

TP (True Peak)

Shows the maximum true peak level in dB.

I (Integrated)

Shows the integrated loudness value. This is the average loudness value that is measured over the whole audio range in LU or LUFS.

S (Short-Term)

Shows the short-term loudness value that is measured every second on an audio block of 3 seconds in LU or LUFS. This gives information about the loudest audio passages.

M Max. (Momentary Max.)

Shows the maximum value of all momentary loudness values that are measured every 100 ms in an audio range of 400 ms in LU or LUFS.

R (Range)

Shows the loudness range (LRA) that is measured over the whole audio range in LU.

The loudness range reports the ratio between the loudest and the quietest non-silent sections. The audio is divided into small blocks. There is one audio block every second and each block lasts 3 seconds, so that the analyzed blocks overlap. The top 10 % of the quiet blocks and the top 5 % of the loud blocks are excluded from the final analysis. The calculated loudness range is the ratio between the loudest and quietest remaining audio blocks. This measurement helps you to decide how much compression or expansion must be applied to the audio.

An asterisk (*) after a loudness range value indicates that less than 1 minute of audio was analyzed.

Min. shows the minimum loudness range value in LU. **Max.** shows the maximum loudness range value in LU.

NOTE

EBU R 128 does not recommend loudness range measurement for audio shorter than 1 minute due to too few data points.

Dlg. (Dialogue)

Shows the overall percentage of speech that is detected in the measured audio. The speaker icon indicates that speech is detected at the current cursor position during playback.

NOTE

If at least 15 % of speech is detected, a dialogue-gated measurement according to ITU-R BS.1770-1 is used for visualization. If less speech is detected, a program-gated measurement according to ITU-R BS.1770-3 is used.

Time

Shows the overall duration of the loudness measurement.

The following module-specific settings are available in the **Module Settings** window:

Unit

Allows you to switch the meter scale between LUFS (absolute values) and LU (relative values).

Scale

Allows you to set the meter to either EBU +9 scale (linear), EBU +18 scale (linear), or +23 scale (logarithmic).

Ref. Integrated

Sets a reference value for the integrated loudness. If higher values are detected, the loudness meter indicates clipping.

Tol. Integrated

Sets a tolerance value for the integrated loudness.

Ref. True Peak

Sets a reference value for the true peak level. If higher values are detected, the loudness meter indicates clipping.

Tol. True Peak

Sets a tolerance value for the true peak level.

Ref. Short-Term

Sets a reference value for the short-term loudness. If higher values are detected, the loudness meter indicates clipping.

Tol. Short-Term

Sets a tolerance value for the short-term loudness.

Ref. Momentary

Sets a reference value for the maximum momentary loudness. If higher values are detected, the loudness meter indicates clipping.

Tol. Momentary

Sets a tolerance value for the maximum momentary loudness.

Ref. Range

Sets a reference value for the loudness range. If higher values are detected, the loudness meter indicates clipping.

Tol. Range

Sets a tolerance value for the loudness range.

Loudness Curve

This module shows loudness values as a time curve.
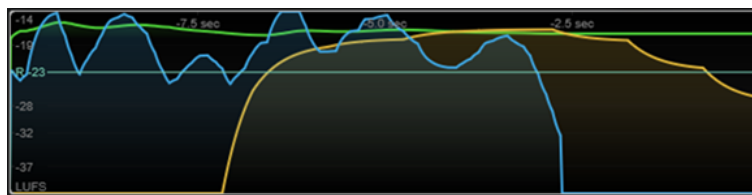
The following module-specific settings are available in the **Module Settings** window:

Duration

Sets the duration of the audio stream that is displayed.

NOTE

Alternatively, you can move the mouse pointer over the selected module, hold ⬚, and use the mouse wheel to adjust this parameter.

Unit

Allows you to switch the meter scale between LUFS (absolute values) and LU (relative values).

Scale

Allows you to set the meter to either EBU +9 scale (linear), EBU +18 scale (linear), or +23 scale (logarithmic).

Smooth

Smoothes the display of the loudness curve.

Momentary

Shows/Hides the maximum momentary loudness curve.

Short-Term

Shows/Hides the short-term loudness curve.

Integrated

Shows/Hides the integrated loudness curve.

Range

Shows/Hides a gray area around the integrated loudness curve that represents the loudness range.

Ref. Integrated

Sets a reference value for the integrated loudness.
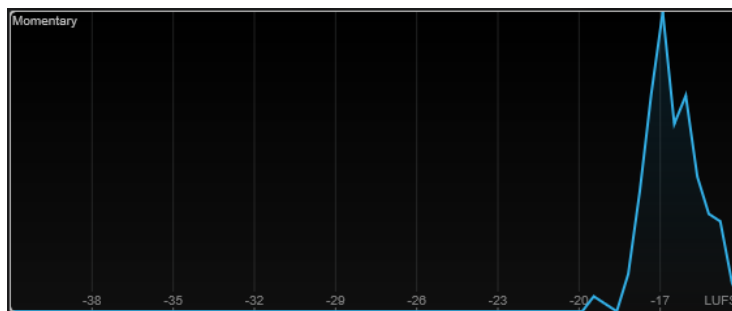
Loudness Histogram

This module shows a histogram for the loudness or the loudness ratio value.

The following module-specific settings are available in the **Module Settings** window:

Unit

Allows you to switch the meter scale between LUFS (absolute values) and LU (relative values).

Scale

Allows you to set the meter to either EBU +9 scale (linear), EBU +18 scale (linear), or +23 scale (logarithmic).

Meter Mode

Sets the displayed loudness or loudness ratio value.

- **Momentary** mode shows a histogram of the maximum value of all momentary loudness values that are measured every 100 ms in an audio range of 400 ms.

- **Short-Term** shows a histogram of the short-term loudness value that is measured every second on an audio block of 3 seconds.

- **Integrated** shows a histogram of the integrated loudness value.

- **PLR** shows a histogram of the peak-to-loudness ratio, also referred to as the crest factor, which is the difference between the maximum true peak level value and integrated loudness value.

- **PSR** shows a histogram of the peak-to-short-term-loudness ratio according to AES Convention e-Brief 373.
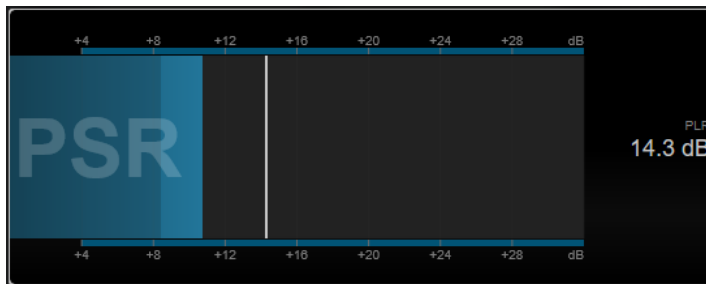
Smooth

Smoothes the display of the loudness curve.

> NOTE

Alternatively, you can move the mouse pointer over the selected module and use the mouse wheel to adjust this parameter.

Loudness Ratio

This module shows the peak-to-loudness ratio (PLR) and the peak-to-short-term-loudness ratio (PSR) values according to AES specifications.

PSR

Shows the peak-to-short-term-loudness ratio according to AES Convention e-Brief 373. The darker area of the meter indicates the minimum PSR value.

PLR

Shows the peak-to-loudness ratio, also referred to as the crest factor, which is the difference between the maximum true peak level value and integrated loudness value. The current PLR value is displayed numerically and also indicated by a thin bar on the meter.

The following module-specific setting is available in the **Module Settings** window:

Ref. Level

Sets the reference level below which the PSR display turns red.

Time Smooth

Smoothes the temporal display of the PSR value.

> NOTE
>
> Alternatively, you can move the mouse pointer over the selected module, hold ⬚, and use the mouse wheel to adjust this parameter.
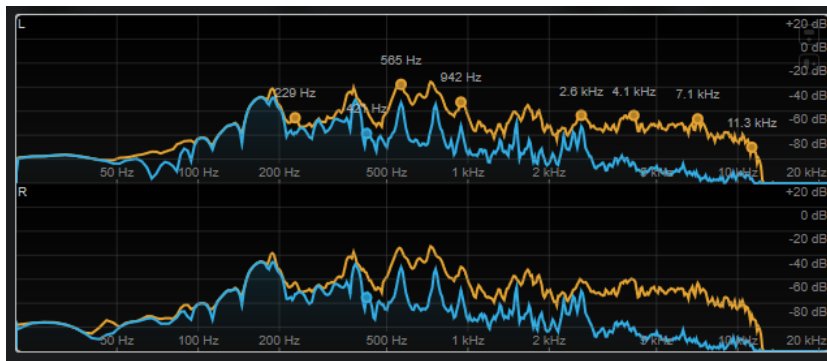
Spectral Domain Modules

The modules in this category allow you to visualize spectral information about the audio signal.

The following modules and module-specific settings are available:

Spectrum Curve

This module uses FFT (Fast Fourier Transform) techniques to show a frequency graph, providing a precise and detailed real-time frequency analysis.

The display shows the frequency spectrum as a linear graph. If you move the mouse pointer over the display, a peak curve is shown in orange. Move the mouse pointer over the curves to show the local maximum values in Hz. Press **Ctrl/Cmd** to show the maximum values in dB or press **Shift** to show their pitch.

The following module-specific settings are available in the **Module Settings** window:

Time Smooth

Smoothes the temporal display.

NOTE

Alternatively, you can move the mouse pointer over the selected module, hold ˌ, and use the mouse wheel to adjust this parameter.

Peak Fallback

Sets the speed of release for the spectrum curve and the peak curve.

NOTE

- Alternatively, you can move the mouse pointer over the selected module, hold **Ctrl/Cmd**, and use the mouse wheel to adjust this parameter.
- If this control is turned all the way to the left, the peak curve is disabled.

Freq. Smooth

Smoothes the frequency display of the spectrum curve.

NOTE

Alternatively, you can move the mouse pointer over the selected module and use the mouse wheel to adjust this parameter.

FFT Window

Sets the block size of the window that is used for the analysis. If **Multi** is selected, 3 different block sizes are used at the same time.

Minimum
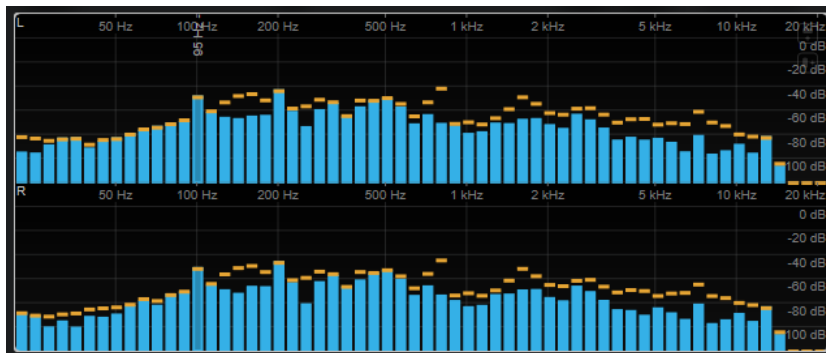
Sets the minimum value of the scale.

Maximum

Sets the maximum value of the scale.

Slope

Adds a slope to the frequency spectrum.

Spectrum Bar

This module shows a graphical representation of the frequency spectrum, analyzed into separate frequency bands, represented as vertical bars.

Move the mouse pointer over a bar to show the frequency range in Hz. Press **Ctrl/Cmd** to show the current value in dB or press **Shift** to show its pitch range.

The following module-specific settings are available in the **Module Settings** window:

Time Smooth

Smoothes the temporal display.

NOTE

Alternatively, you can move the mouse pointer over the selected module, hold ⎵, and use the mouse wheel to adjust this parameter.

Peak Fallback

Sets the speed of release for the level meters and peak indicators.

NOTE

- Alternatively, you can move the mouse pointer over the selected module, hold ⎵, and use the mouse wheel to adjust this parameter.
- If this control is turned all the way to the left, the peak indicators are disabled.

Threshold

Sets a threshold level below which the display is masked.

NOTE

Alternatively, you can move the mouse pointer over the selected module and use the mouse wheel to adjust this parameter.

Bands/Oct.

Sets the number of bands per octave.

Minimum

Sets the minimum value of the scale.

Maximum

Sets the maximum value of the scale.

Slope

Adds a slope to the frequency spectrum.

Spectrum Intensity

This module represents the frequency magnitude of the audio. The more intensely a bar is colored, the higher the magnitude at this frequency.

The following module-specific settings are available in the **Module Settings** window:

Time Smooth

Smoothes the temporal display.

> NOTE
>
> Alternatively, you can move the mouse pointer over the selected module, hold , and use the mouse wheel to adjust this parameter.

FFT Window

Sets the block size of the window that is used for the analysis. If **Multi** is selected, 3 different block sizes are used at the same time.

Color

Allows you to choose a color scheme.

Minimum
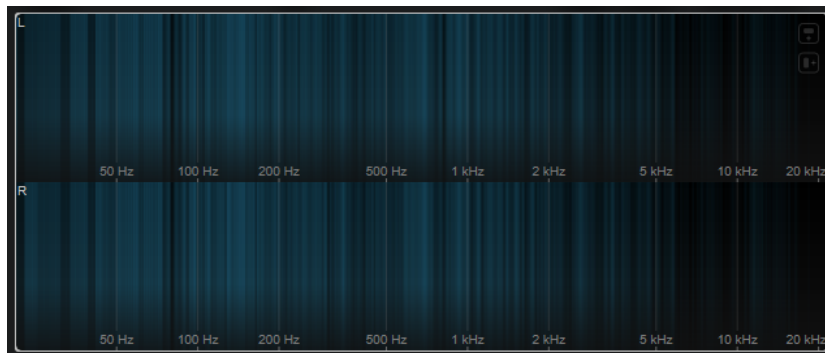
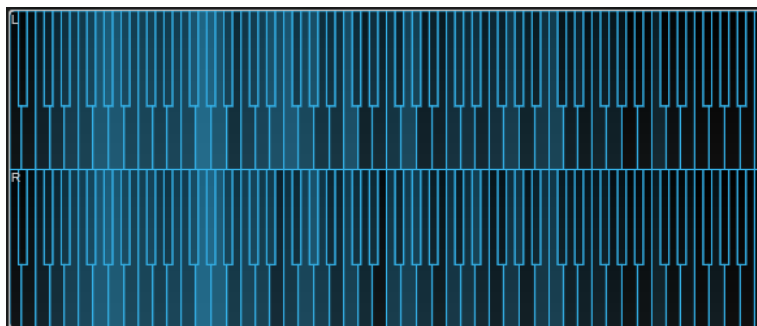Sets the minimum value of the scale.

Maximum

Sets the maximum value of the scale.

Slope

Adds a slope to the frequency spectrum.

Spectrum Keyboard

This module represents the frequency magnitude of the audio mapped to the keys of a piano keyboard. The more intensely a key is colored, the higher the magnitude at its frequency.

The following module-specific settings are available in the **Module Settings** window:

Time Smooth

Smoothes the temporal display.

NOTE

Alternatively, you can move the mouse pointer over the selected module, hold ⎵, and use the mouse wheel to adjust this parameter.

Color

Allows you to choose a color scheme.

Minimum

Sets the minimum value of the scale.

Maximum

Sets the maximum value of the scale.

Slope

Adds a slope to the frequency spectrum.

Spectrogram

This module shows the last seconds of the audio stream. This allows you to detect disturbances in the spectrogram and to monitor the noise level and frequencies, for example.

NOTE

This module runs in **Maximum Audio Performance** mode.

The following module-specific settings are available in the **Module Settings** window:

FFT Window

Sets the block size of the window that is used for the analysis. This allows you to adjust the trade-off between temporal resolution and frequency resolution. If you specify a higher value, more frequencies are analyzed but they are located less accurately in the time domain.

Duration

Sets the duration of the audio stream that is displayed.

NOTE

Alternatively, you can move the mouse pointer over the selected module, hold ⎵, and use the mouse wheel to adjust this parameter.

Color

Allows you to choose a color scheme.

Minimum

Sets the minimum value of the scale.

Maximum

Sets the maximum value of the scale.

NOTE

Alternatively, you can move the mouse pointer over the selected module and use the mouse wheel to adjust the **Minimum** and **Maximum** parameters simultaneously.

Chromagram

This module displays a chromagram of your audio.

NOTE

This module runs in **Maximum Audio Performance** mode.

The following module-specific settings are available in the **Module Settings** window:

Duration

Sets the duration of the audio stream that is displayed.

NOTE

Alternatively, you can move the mouse pointer over the selected module, hold , and use the mouse wheel to adjust this parameter.

Color

Allows you to choose a color scheme.

Minimum

Sets the minimum value of the scale.

Maximum
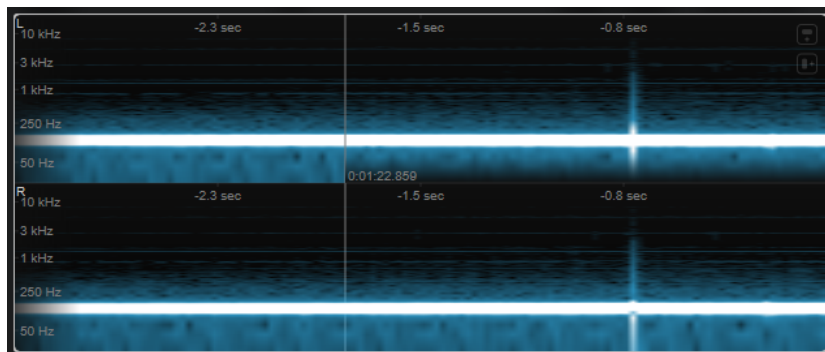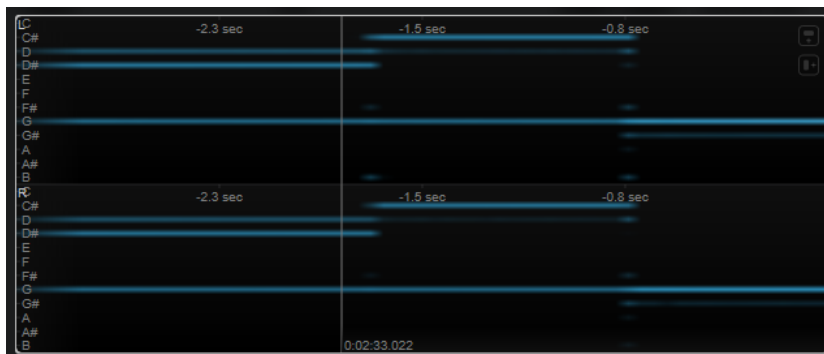
Sets the maximum value of the scale.

NOTE

Alternatively, you can move the mouse pointer over the selected module and use the mouse wheel to adjust the **Minimum** and **Maximum** parameters simultaneously.

Phase Modules

The modules in this category allow you to visualize the phase or directional relationship between the channels of the audio signal.

The following modules and module-specific settings are available:

Phasescope

This module uses a vectorscope display to show the phase and amplitude relationship between the left and right stereo channels. This provides you with directional information about a stereo audio signal.
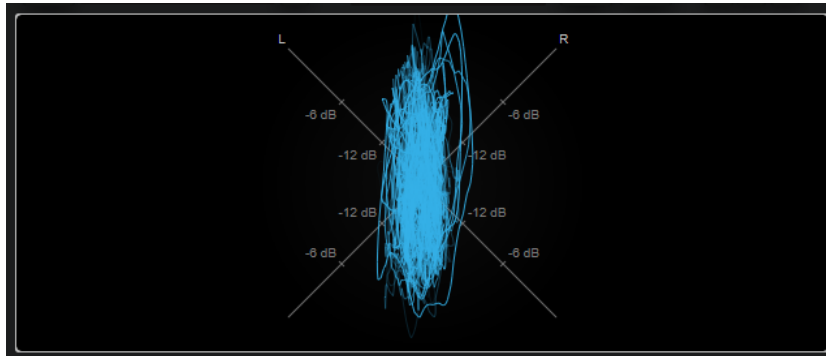
Hold **Shift** and move the mouse pointer over the display to measure the angle.

The following module-specific settings are available in the **Module Settings** window:

Zoom

Allows you to zoom on the graphical display.

NOTE

Alternatively, you can move the mouse pointer over the selected module and use the mouse wheel to adjust this parameter.

Auto Zoom

If this button is activated, the zoom factor is adapted automatically.

Mode

Sets the display mode. The following modes are available: **Lines**, **Dots**, **Envelope**.

Peak Fallback

Sets the speed of release for the peak envelope in **Envelope** mode.

NOTE

If this control is turned all the way to the left, the peak envelope is disabled.

Scale

Activates/Deactivates the axis labeling.

NOTE

This option is only available if **Auto Zoom** is deactivated.

Panorama

This module uses a polar coordinate display to show the phase and amplitude relationship between the left and right stereo channels. This provides you with directional information about a stereo audio signal.

The following module-specific settings are available in the **Module Settings** window:

Zoom

Allows you to zoom on the graphical display.

NOTE

Alternatively, you can move the mouse pointer over the selected module and use the mouse wheel to adjust this parameter.

Auto Zoom

If this button is activated, the zoom factor is adapted automatically.

Mode

Sets the display mode. The following modes are available: **Lines**, **Dots**, **Envelope**.

Peak Fallback

Sets the speed of release for the peak envelope in **Envelope** mode.

NOTE

If this control is turned all the way to the left, the peak envelope is disabled.
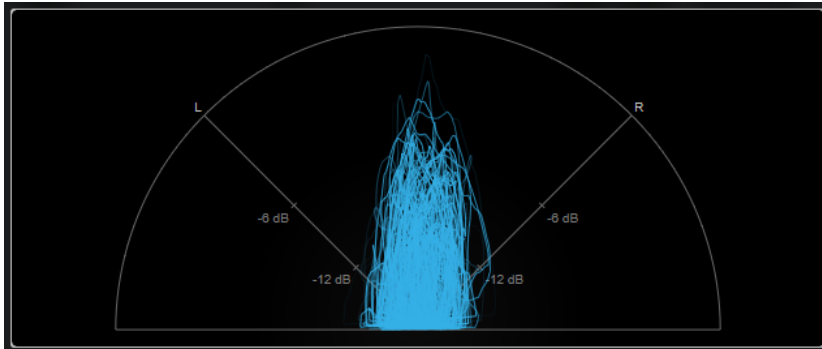
Scale

Activates/Deactivates the axis labeling.

NOTE

This option is only available if **Auto Zoom** is deactivated.

Multipanorama

This module provides you with frequency-dependent directional information about a stereo audio signal.
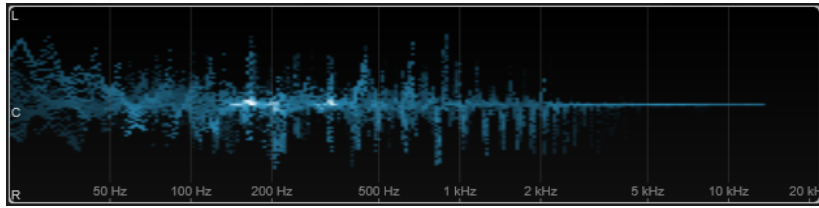
The following module-specific settings are available in the **Module Settings** window:

Time Smooth

Sets the time for which an energy impulse is displayed.

Bands/Oct.

Sets the number of bands per octave.

Color

Allows you to choose a color scheme.

Correlation

This module visualizes the phase-correlation between the left and right channel. This allows you, for example, to check the mono compatibility of a stereo recording.

The following module-specific setting is available in the **Module Settings** window:

Time Smooth

Smoothes the temporal display of the correlation.

NOTE

Alternatively, you can move the mouse pointer over the selected module, hold ▮, and use the mouse wheel to adjust this parameter

Multicorrelation

This module visualizes the phase-correlation between the left and right channel for different frequency bands.

Move the mouse pointer over a bar to display its frequency range in Hz. Hold **Ctrl/Cmd** to display its current value. Hold **Shift** to display its pitch range.

The following module-specific settings are available in the **Module Settings** window:

Time Smooth

Smoothes the temporal display of the correlation.

NOTE

Alternatively, you can move the mouse pointer over the selected module, hold ⬜, and use the mouse wheel to adjust this parameter

Bands/Oct.

Sets the number of bands per octave.

Balance

This module visualizes the balance between the left and right channel.

The following module-specific setting is available in the **Module Settings** window:

Time Smooth

Smoothes the temporal display of the correlation.

NOTE

Alternatively, you can move the mouse pointer over the selected module, hold ⬜, and use the mouse wheel to adjust this parameter

Spatial Domain Modules

The modules in this category allow you to visualize the spatial dimension of the audio signal.
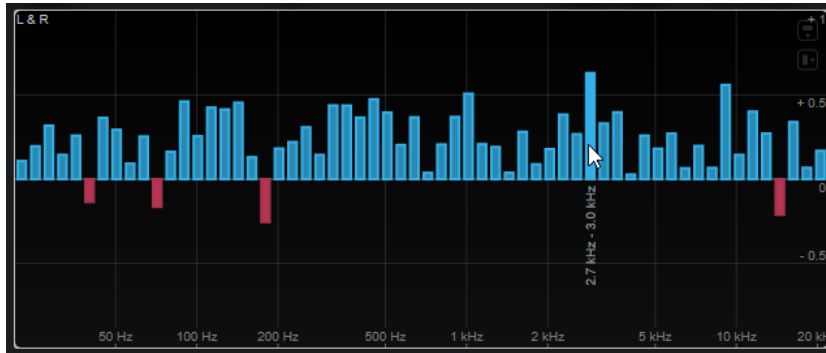
The following modules and module-specific settings are available:

Surround

This module visualizes the level and correlation of the different speakers in a surround speaker setup.

If all channels have the same level, a perfect circle is shown in the center of the display.

> NOTE
>
> This module is only available for channel-based surround configurations. Speaker configurations with top speakers and Ambisonics channels are not supported.

The following module-specific settings are available in the **Module Settings** window:

Zoom

Allows you to zoom on the graphical display.

> NOTE
>
> Alternatively, you can move the mouse pointer over the selected module and use the mouse wheel to adjust this parameter.

Peak Fallback

Sets the speed of release for the peak envelope.

> NOTE
>
> - Alternatively, you can move the mouse pointer over the selected module, hold **Ctrl/Cmd**, and use the mouse wheel to adjust this parameter.
> - If this control is turned all the way to the left, the peak envelope is disabled.

Scale

Activates/Deactivates the axis labeling for the correlation display.

Ambisonics

This module allows you to visualize the energy distribution of an Ambisonics signal.
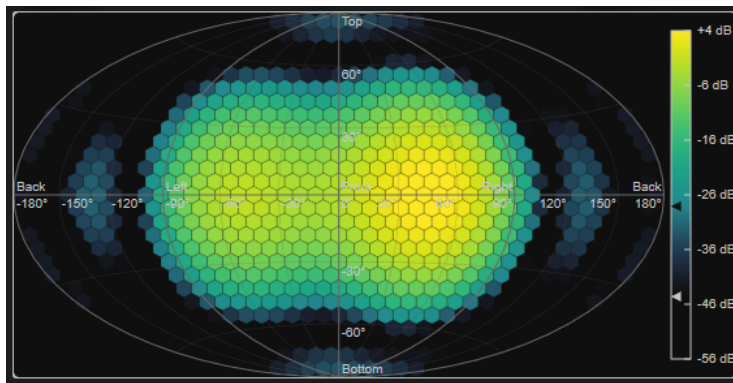
A plane representation of the spherical Ambisonics sound field shows a grid of hexagons. The color of a hexagon indicates the RMS level at this position. A filter allows you to smooth the visualization.

NOTE

This module is only available for Ambisonics channel configurations.

The following module-specific settings are available in the **Module Settings** window:

Attack

Sets the attack time of the smoothing filter.

Release

Sets the release time of the smoothing filter.

Minimum

Sets the minimum value of the signal intensity scale.

Maximum

Sets the maximum value of the signal intensity scale.

Threshold

Sets the minimum signal level that is displayed. This value is indicated by the lower triangle in the color legend on the right. If you change this value, the **Fade Range** value is adjusted accordingly.

Fade Range

Sets the level at which a hexagon is displayed fully opaque. This value is indicated by the upper triangle in the color legend on the right.

Color

Allows you to choose a color scheme.

Resolution

Sets the resolution of the grid.

Waveform Modules

The modules in this category allow you to visualize the waveform of the audio signal.

The following modules and module-specific settings are available:

Oscilloscope

This module shows a highly magnified view of the waveform.
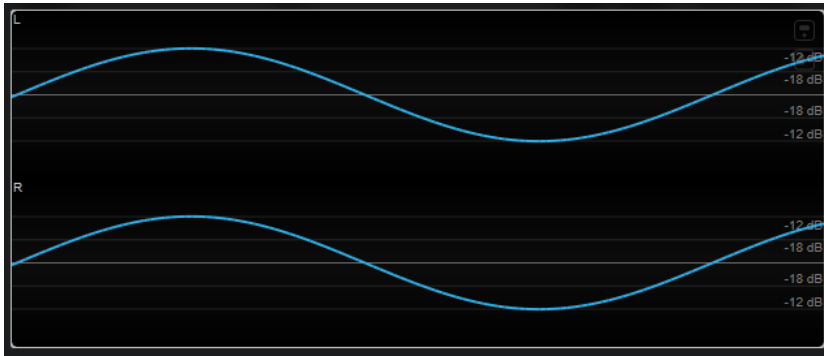
The following module-specific settings are available in the **Module Settings** window:

Zoom

Allows you to zoom on the graphical display by adjusting the amplitude.

NOTE

Alternatively, you can move the mouse pointer over the selected module, hold **Alt/ Opt** , and use the mouse wheel to adjust this parameter.
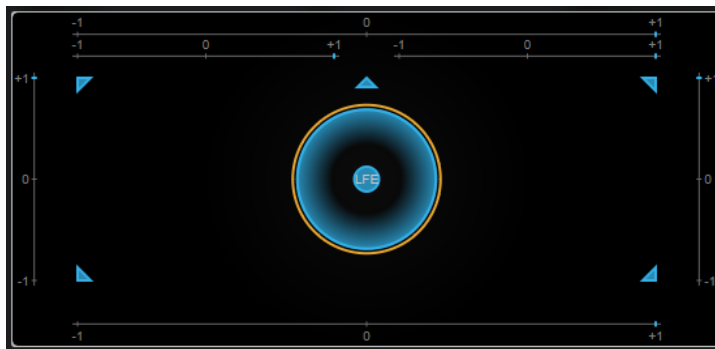
Frequency

Allows you to zoom on the graphical display by adjusting the frequency.

NOTE

Alternatively, you can move the mouse pointer over the selected module, hold **Ctrl/ Cmd** , and use the mouse wheel to adjust this parameter.

Trigger

Sets the channel that is used to synchronize the audio signal.

NOTE

Alternatively, you can click the waveform of the corresponding channel.

Scale

Activates/Deactivates the axis labeling.

NOTE

This option is only available if **Auto Zoom** is deactivated.

Phase

Allows you to shift the zero-crossing position.

Wavescope

This module displays the real-time waveform of the audio signal.

Move the mouse pointer over a waveform position to display the corresponding project time.

The following module-specific settings are available in the **Module Settings** window:

Zoom

Allows you to zoom on the graphical display.

NOTE

Alternatively, you can move the mouse pointer over the selected module and use the mouse wheel to adjust this parameter.

Duration

Sets the duration of the audio stream that is displayed.

NOTE

Alternatively, you can move the mouse pointer over the selected module, hold ⏷, and use the mouse wheel to adjust this parameter.

Tempo Sync

If this button is activated, you can set **Duration** in beats.

NOTE

The equivalent duration is limited to a minimum of 0.5 s and a maximum of 30 s.

Scale

Activates/Deactivates the axis labeling.

NOTE

This option is only available if **Auto Zoom** is deactivated.

Station. Cursor

If this button is activated, the waveform moves continuously under the stationary cursor. If this button is deactivated, the waveform is refreshed when the cursor moves over it.

Wavecircle

This module displays the real-time waveform of the audio signal as a circle.

Move the mouse pointer over a waveform position to display the corresponding project time.

The following module-specific settings are available in the **Module Settings** window:

Zoom

Allows you to zoom on the graphical display.

NOTE

Alternatively, you can move the mouse pointer over the selected module and use the mouse wheel to adjust this parameter.

Duration

Sets the duration of the audio stream that is displayed.

NOTE

Alternatively, you can move the mouse pointer over the selected module, hold , and use the mouse wheel to adjust this parameter.

Tempo Sync

If this button is activated, you can set **Duration** in beats.

NOTE

The equivalent duration is limited to a minimum of 0.5 s and a maximum of 30 s.
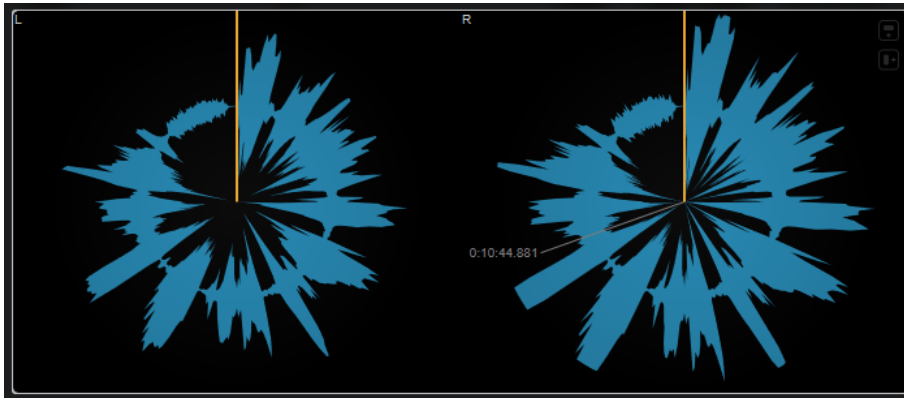
Reverse

Changes the rotation direction.

Station. Cursor

If this button is activated, the waveform moves continuously under the stationary cursor. If this button is deactivated, the waveform is refreshed when the cursor moves over it.

Other

This category provides a time display.

Time

This module shows the current time position of the project cursor.

NOTE

If the project cursor is located outside the locator range, the color of the time display turns gray.

The following module-specific setting is available in the **Module Settings** window:

Mode

Allows you to select one of the following display modes: **Time**, **Sample**, **Beats**, or **Timecode**.

Delay

MonoDelay

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

Lo Filter

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

Hi Filter

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

Delay

Sets the delay time in milliseconds.

Feedback

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

Mix

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

PingPongDelay

This is a stereo delay effect that alternates each delay repeat between the left and right channels.

NOTE

This plug-in works only on stereo tracks.

Lo Filter

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

Hi Filter

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

Delay

Sets the delay time in milliseconds.

Feedback

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

Mix

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

Spatial

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

Start Left/Start Right

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

StereoDelay

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

NOTE

This plug-in works only on stereo tracks.

Feedback

> Set the number of repeats for each delay.

Delay

> Sets the delay time in milliseconds.

Mix

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

Lo Filter

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

Pan

> Sets the stereo position.

Hi Filter

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

Distortion

Distortion

Distortion adds crunch to your tracks.

Boost

Increases the distortion amount.

Oversampling

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

NOTE

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

Mix

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

Tone

Changes the tonal characteristic of the output signal.

Feedback

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

Spatial

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

Output

Sets the output level.

Magneto II

Magneto II simulates the saturation and compression of recording on analog tape machines.

Saturation

Determines the amount of saturation and the generation of overtones. This leads to a small increase in input gain.

Saturation On/Off

Activates/Deactivates the saturation effect.

Dual Mode

Simulates the use of two machines.

Frequency Range Low/High

These parameters set the frequency range of the spectrum band to which the tape effect is applied.

For example, to avoid the saturation of lower frequencies, set the **Low** value to 200 Hz or 300 Hz. To avoid the saturation of very high frequencies, set the **High** parameter to values below 10 kHz.

Solo

Allows you to hear only the set frequency range including the tape simulation effect. This helps you to determine the appropriate frequency range.

HF-Adjust

Sets the amount of high frequency content of the saturated signal.

HF-Adjust On/Off

Activates/Deactivates the **HF-Adjust** filter.

Quadrafuzz v2

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

Frequency Band Editor

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

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

● To attenuate or boost the output level of each frequency band by ±15 dB, use the handles on top of each frequency band.

Global Settings

SB

Switches between multi band and single band mode.

Scenes

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

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

To copy the settings of a scene to another scene, select the scene that you want to copy, click **Copy**, and click one of the numbered buttons.

You can automate the selection of scenes.

Mix

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

Output (-24 to 24 dB)

Sets the output level.

Band Settings

Mute Band

Mutes/Unmutes a frequency band.

Solo Band

Solos the corresponding frequency band.

Bypass Band

Bypasses a frequency band.

In/Out meter

Display the input and output level.

Gate

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

Tape

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

Drive

Controls the amount of tape saturation.

Tape Mode Dual

Simulates the use of two machines.

Tube

This band mode simulates the saturation effects using analog tubes.

Drive

Controls the amount of tube saturation.

Tubes

Determine the number of tubes that are simulated.

Dist

This band mode adds distortion to your tracks.

Drive

Controls the amount of distortion.

FBK

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

Amp

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

Drive

Controls the amount of amp overdrive.

Amp Types

You can select the following types of guitar amps:

- Amp Clean
- Amp Crunch
- Amp Lead

Dec

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

Decimator

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

Mode

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

S&H

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

Delay

To open the **Delay** section, click the **Delay** button.

Time

Sets the delay time.

Mode

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

> **NOTE**
>
> High **FBK** values and low **Duck** values in combination with activated **Mode** can lead to unwanted noise.

FBK

Determines the number of repeats for each delay.

Duck

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

Mix

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

Sliders

Width

Sets the stereo width for the corresponding band.

Out

Sets the output gain for the corresponding band.

Pan

Sets the stereo position for the corresponding band.

Mix

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

Dynamics

Black Valve

Black Valve is a highly adaptive compressor with a vintage tube sound. It is based on a classic analog device containing a tube pre-amp in conjunction with a classic compressor.

Drive

Controls the amount of tube saturation.

Gain

Sets the input gain for the compressor section to adjust the gain after the tube stage,

Reduction

The LEDs show the gain reduction.

Peak Reduction

Combines the threshold and ratio parameters from common compressors. The higher the value, the more compression occurs.

Dry/Wet

Adjusts the mix between the dry signal and the wet signal, preserving the transients of the input signal. This allows for parallel compression.

Output

Sets the output gain.

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

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

Release (3 to 1000 ms or Auto mode)

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

Link Channels

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

Detect Intersample Clipping

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

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

Compressor

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

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

Threshold

Determines the level where the compressor kicks in.

Ratio

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

Soft Knee

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

High Ratio

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

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

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

Dry Mix

Mixes the dry input signal to the compressed signal.

Attack (0.1 to 100 ms)

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

Hold (0 to 5000 ms)

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

Release (10 to 1000 ms or Auto mode)

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

Analysis (Pure Peak to Pure RMS)

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

Live

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

DeEsser

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

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

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

Display

Shows the spectrum of the input signal.

● To adjust the frequency band, drag the border lines or click in the middle of the band and drag.

● To change the width of the frequency band, hold **Shift** and drag to the left or right.

Filter

Lo/Hi

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

NOTE

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

Solo

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

Diff

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

Dynamics

Reduction

Controls the intensity of the de-essing effect.

Threshold (-50 to 0 dB)

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

Release (1 to 1000 ms)

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

Auto

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

Side-Chain

Side-Chain

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

Freq (25 Hz to 20 kHz)

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

NOTE

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

Q-Factor

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

Monitor

Allows you to monitor the filtered signal.

Live

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

Positioning the DeEsser in the Signal Chain

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

EnvelopeShaper

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

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

Attack (-20 to 20 dB)

> Sets the gain of the attack phase of the signal.

Length (5 to 200 ms)

> Sets the length of the attack phase.

Release

> Sets the gain of the release phase of the signal.

Output

> Sets the output level.

Expander

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

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

Threshold

Determines the level where the expansion kicks in.

Ratio

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

Soft Knee

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

Fall (0.1 to 100 ms)

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

Hold (0 to 2000 ms)

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

Rise (10 to 1000 ms or Auto mode)

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

Analysis (Pure Peak to Pure RMS)

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

Live

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

Gate

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

Attack (0.1 to 1000 ms)

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

NOTE

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

Hold (0 to 2000 ms)

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

Release (10 to 1000 ms or Auto mode)

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

Threshold

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

State LED

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

Analysis (Pure Peak to Pure RMS)

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

Range

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

Live

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

Side-Chain Section

Side-Chain

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

Monitor

Allows you to monitor the filtered signal.

Center

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

Q-Factor

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

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

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

Limiter

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

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

Input (-24 to 24 dB)

Sets the input gain.

Release (0.1 to 1000 ms or Auto mode)

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

Output

Sets the maximum output level.

Maximizer

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

Classic

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

Modern

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

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

● **Release** sets the overall release time.

● **Recover** allows for a faster signal recovering at the beginning of the release phase.

Optimize

Determines the loudness of the signal.

Mix

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

Output

Sets the maximum output level.

Soft Clip

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

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. This allows you to set the plug-in into ducking mode per 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.

Output

Sets the output level.

Bypass Band

Bypasses a frequency band.

Solo Band

Solos the corresponding frequency band.

Compressor Section

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

Threshold

Determines the level where the compressor kicks in.

Ratio

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

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

Q-Factor

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

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.

Output

Sets the output level.

Bypass Band

Bypasses a frequency band.

Solo Band

Solos the corresponding frequency band.

Shaper Section

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

Attack (-20 to 20 dB)

Sets the gain of the attack phase of the signal.

Length (5 to 200 ms)

Sets the length of the attack phase.

Release

Sets the gain of the release phase of the signal.

Sensitivity (-40 to -10 dB)

Sets the sensitivity of the detection.

Output

Sets the output level.

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.

Output

Sets the output level.

Bypass Band

Bypasses a frequency band.

Solo Band

Solos the corresponding frequency band.

Expander Section

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

Threshold

Determines the level where the expansion kicks in.

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.

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

Q-Factor

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

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.

Raiser

Raiser is a versatile limiter plug-in, that can increase the loudness of the audio material to a high extent. Its range goes from smooth limiting of solo tracks and full mixes to a rather aggressive limiting, best suited for percussive material.

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

Stationary Cursor

If this button is activated, the waveform moves continuously under the stationary cursor. If this button is deactivated, the waveform is refreshed when the cursor moves over it.

Gain

Sets the input gain.

Reduction

Shows the amount of gain reduction.

Release Mode

This pop-up menu sets the mode for the release time:

- **Manual** mode allows you to set the release time freely with the **Time** control.

- **Auto** mode automatically applies the best release setting for the audio material.

- **Restricted Min.** mode automatically applies the best release setting for the audio material but allows you to define a minimum release time with the **Release** control.

- **Restricted Max.** mode automatically applies the best release setting for the audio material but allows you to define a maximum release time with the **Release** control.

- **Aggressive** mode applies aggressive limiting and preserves the transients of the input signal without creating pumping effects.

Fast

If this option is activated, an initial fast release phase is added before the second phase that is set by the **Time** parameter. We recommend using this option for percussive material.

Release Time

Sets the time after which the gain returns to the original level when the signal drops below the threshold. This parameter is only available if **Manual**, **Restricted Min.**, or **Restricted Max.** mode is selected.

Link Channels

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

Detect Intersample Clipping

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

Ceiling

Sets the maximum output level. If **Detect Intersample Clipping** is activated, the maximum output level is the true peak level.

Compare

Sets the output level to the level of the dry input signal. This allows you to monitor the limiting effect without the level boost.

Squasher

Squasher is a multi-band compressor that combines upward and downward compression, allowing you to squash the audio signal. You can set up different parameters for up to three frequency bands and use internal side-chaining to control the amount of compression for each band.

Main section

This section provides general settings for upward and downward compression for each frequency band.

Frequency/Compressor display

Displays the frequency spectrum. It allows you to edit the band range, the output level, the upward/downward ratio, and the upward/downward threshold for each band.

You can switch the display between a frequency spectrum view and a compressor characteristics view by clicking the corresponding buttons left of the display:

Frequency spectrum

Compressor characteristics

You can edit the output level of a band or the cutoff frequency between two bands by dragging the corresponding handle on the frequency spectrum display.

Show/Hide Full Frequency Band View

Shows/Hides the frequency spectrum/compressor display.

Input meter

Shows the level of the overall input signal.

Input

Sets the overall input level.

Bands

Sets the number of frequency bands.

Mix

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

Parameter Link

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

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

- If **Relative Mode** is activated and you edit a parameter value of one band, the relation of the corresponding parameter values of the other bands remains the same.

Activate/Deactivate Band

Activates/Deactivates the corresponding frequency band.

Solo Band

Solos the corresponding frequency band.

Up Ratio/Down Ratio

Set the amount of upward and downward compression. These parameters allow you to adjust the squashing effect.

In

Shows the input level for the corresponding frequency band.

Up Threshold/Down Threshold

The left triangle sets the threshold for upward compression. If the input signal is below this threshold, the upward compressor raises the level according to the **Up Ratio** value.

The right triangle sets the threshold for downward compression. If the input signal is above this threshold, the downward compressor lowers the level according to the **Down Ratio** value.

You can drag the darker area between both handles to adjust **Up Threshold** and **Down Threshold** simultaneously.

Output meter

Shows the level of the overall output signal.

Output

Sets the overall output level.

Squash Parameter section

This section provides additional compression and filter settings for each frequency band.

Show/Hide Squash Parameter Section

Shows/Hides the squash parameter section.

Att.

Sets the compression attack time for both the upward and downward compressor.

Rel.

Sets the compression release time for both the upward and downward compressor.

Drive

Sets the amount of saturation. This parameter adds harmonics to the output signal.

Gate

Sets the threshold for the internal gating effect. Signal levels above this threshold trigger the gate to open. Signal levels below this threshold close the gate.

NOTE

You can control this parameter via side-chaining.

Mix

Adjusts the mix between dry signal and wet signal for the corresponding band.

Output

Sets the output level for the corresponding band.

NOTE

Alternatively, you can edit this parameter by dragging the corresponding handle on the frequency display.

Side-Chain section

This section provides settings for internal side-chaining for each frequency band. It is only available if the parameter section is shown.

Show/Hide Side-Chain Section

Shows/Hides the side-chaining section.

Activate/Deactivate Side-Chaining for Band

Activates/Deactivates side-chaining for the corresponding band.

Side-Chain Filter Listen

Lets you monitor the side-chain signal and the applied filter.

Freq

Sets the frequency of the side-chain filter.

Q

Sets the Q factor of the side-chain filter.

Send to

This pop-up menu allows you to send the side-chain signal to the compressor section (**Squasher**) or to the internal gate.

Tube Compressor

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

VU Meter

Shows the amount of gain reduction.

In/Out Meters

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

Input

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

Drive

Controls the amount of tube saturation.

Output

Sets the output gain.

Character

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

Attack (0.1 to 100 ms)

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

Release (10 to 1000 ms or Auto mode)

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

Mix

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

Ratio

Toggles between a low and a high ratio value.

Side-Chain

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

Side-chain section

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

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

Center

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

Q-Factor

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

Monitor

Allows you to monitor the filtered signal.

VintageCompressor

VintageCompressor is modeled after vintage type compressors.

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

VU Meter

Shows the amount of gain reduction.

In/Out Meters

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

Input

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

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 **Auto Release** is activated, the plug-in automatically finds the best release setting for the audio material.

Mix

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

Output (-48 to 24 dB)

Sets the output gain.

VoxComp

VoxComp is a compressor designed for processing vocals. It uses a highly adaptive algorithm, which makes it easy to achieve the desired results.

Meter section

Shows the input, output, and gain reduction levels.

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.

Dry/Wet

Adjusts the mix between the dry signal and the wet signal, preserving the transients of the input signal. This allows for parallel compression.

Threshold

Determines the level where the compressor kicks in.

Output

　　Sets the output gain.

VSTDynamics

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

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

Gate

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

The following parameters are available:

Input meter

　　Shows the level of the input signal.

Attack (0.1 to 100 ms)

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

Threshold

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

State LED

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

Release (10 to 1000 ms or Auto mode)

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

Hold (0 to 2000 ms)

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

Range

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

Side-Chain

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

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

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

Center

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

Q-Factor

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

Monitor

Allows you to monitor the filtered signal.

Compressor

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

Input meter

Shows the level of the input signal.

Graphical display

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

Gain Reduction meter

Shows the amount of gain reduction.

Threshold

Determines the level where the compressor kicks in.

Ratio

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

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

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

Attack (0.1 to 100 ms)

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

Release (10 to 1000 ms or Auto mode)

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

Limiter

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

Input meter

> Shows the level of the input signal.

Gain Reduction meter

> Shows the amount of gain reduction.

Soft Clip

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

Output

> Sets the maximum output level.

Release (10 to 1000 ms or Auto mode)

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

Output section

Output meter

> Shows the level of the output signal.

Module Configurator

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

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

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

EQ

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.

EQ-M5

EQ-M5 is a characterful vintage tube EQ, especially suited for mid processing. It offers low, mid, and high peak filters.

Low Frequency

    Sets the frequency of the **Low Boost** filter.

Low Boost

    Sets the boost of the low peak filter.

Mid Frequency

    Sets the frequency of the **Mid Attenuate** filter.

Mid Attenuate

    Sets the attenuation of the mid peak filter.

High Frequency

    Sets the frequency of the **High Boost** filter.

High Boost

    Sets the boost of the high peak filter.

Output

    Sets the output gain.

EQ-P1A

EQ-P1A is unique sounding vintage tube EQ with a famous low frequency boost and attenuation, based on a classic analog equalizer. It offers low and high shelving filters and an additional high peak filter.

Low Boost

Sets the boost of the low shelving filter.

Low Attenuate

Sets the attenuation of the low shelving filter.

Low Boost/Attenuate Frequency

Sets the frequency of the **Low Boost** and **Low Attenuate** filters.

High Boost Bandwidth

Sets the bandwidth of the **High Boost** filter.

High Boost

Sets the boost of the high peak filter.

High Boost Frequency

Sets the frequency of the **High Boost** filter.

High Attenuate

Sets the attenuation of the high shelving filter.

High Attenuate Frequency

Sets the frequency of the **High Attenuate** filter.

Output

Sets the output gain.

Frequency 2

Frequency 2 is a high-quality equalizer with 8 fully parametric bands. The bands can act as shelving filter, as peak or notch filter, or as low-pass/high-pass cut filter.

The plug-in supports separate internal or external side-chaining for each band. For **Low Shelf**, **High Shelf**, and **Peak** filters, dynamic filtering lets you determine when and how the EQ is applied, depending on the dynamics of the audio material.

Main Section

Reset

RESET

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

Auto Listen for Filters

If this option is activated and you edit a parameter of a band, the corresponding frequency range is isolated. This helps you to focus on a particular frequency range and allows you to locate unwanted frequencies in your audio.

Global Settings

Opens the settings dialog for the spectrum display.

Display Gain Range

+30

Allows you to adjust the gain range that is shown in the spectral display.

Keys

Shows/Hides the keyboard below the graphical editor.

On the keyboard, color indicators reflect the center frequencies of all active equalizer bands. You can adjust the frequency of a band by dragging its color indicator. If you drag the color indicator of a band to a key, the band is set to its exact frequency.

View

Toggles between single-band and multi-band view. Single-band view provides additional parameters for each band.

To toggle between single-band and multi-band view, you can also double-click on the top of a band section.

Output

Adjusts the overall output level.

Output meter

Shows the level of the overall output signal.

Band Settings

Multi-band view
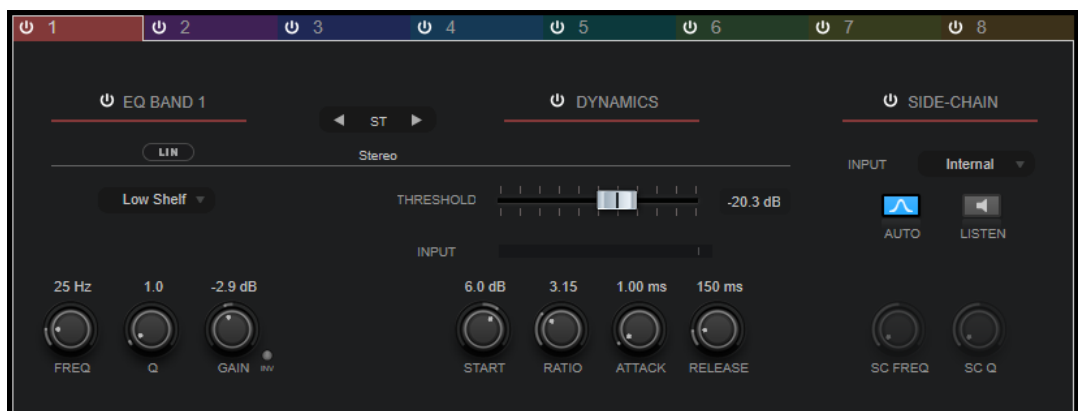
Single-band view

Activate/Deactivate Band

Activates/Deactivates the corresponding band.

- To activate/deactivate a band, you can also double-click the corresponding handle in the graphical editor.
- You can modify the parameters of deactivated bands.

Switch Processing buttons

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.

IMPORTANT

When using **Mid/Side** processing mode, we recommend that you activate **Linear Phase Processing** in order to avoid unwanted sound colorization.

NOTE

This setting is only available for stereo tracks.

Linear Phase Processing

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.

Activating this option deactivates dynamic filtering for the corresponding band.

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, you may hear an unwanted pre-ringing effect.

Filter Type

You can choose between the filter types **Low Shelf**, **Peak**, **High Shelf**, and **Notch**. For band 1 and 8, you can also select the types **Cut 6**, **Cut 12**, **Cut 24**, **Cut 48**, and **Cut 96**.

- **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.
- **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

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

NOTE

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

If the band is active, the frequency value is reflected as a highlighted key on the keyboard below the graphical editor.

Q

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

> NOTE
>
> - You can adjust the **Q** parameter of a band in the graphical editor by ▐-clicking the corresponding handle and moving the mouse up and down. Alternatively, you can point to the handle and turn the mouse wheel.
>
> - This parameter is not available for **Cut** filters.

Gain

Sets the amount of attenuation/boost for the corresponding band. If **Dynamic Filtering** is activated, this is also the target gain value.

> 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.

Invert Gain

Inverts the value of the gain parameter. Positive gain values become negative and vice versa.

Show Dynamics Parameters

Shows/Hides parameters for dynamic filtering in multi-band view.

> NOTE
>
> This setting is only available for the filter types **Low Shelf**, **Peak**, **High Shelf**, and **Notch**.

Activate/Deactivate Dynamic Filtering

Activates/Deactivates dynamic filtering for the corresponding band. If this option is activated, the filtering of the band depends on the dynamics of the input signal.

Activating this setting deactivates linear phase mode for the corresponding band.

> NOTE
>
> This setting is only available for the filter types **Low Shelf**, **Peak**, **High Shelf**, and **Notch**.

Threshold

Determines the threshold level. Only signal levels above the threshold are dynamically filtered.

Input meter

Shows the level of the input signal.

> NOTE
>
> In multi-band view, the input level meter is combined with the threshold handle.

Start

Allows you to adjust the starting point for the gain. Dynamic filtering takes place from this point to the set **Gain** of the EQ band.

NOTE

This setting is only available in single-band view.

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 target gain is reached almost immediately.

Attack

Determines how fast the 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.

Release

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

Side-Chain

Activates/Deactivates internal side-chaining for the corresponding band. This allows you to shape the input signal according to the side-chain filter parameters.

NOTE

- This setting is only available in single-band view.
- Side-chaining is only available if dynamic filtering is activated.
- If side-chaining is activated, this is indicated by showing **SC** on the top of a band section.

Side-Chain Filter Auto

Activates/Deactivates automatic filtering of the side-chain signal. If this parameter is activated, the **SC Freq** and **SC Q** parameters are deactivated. Instead, the **Freq** and **Q** values of the corresponding band are used.

NOTE

This setting is only available in single-band view.

Side-Chain Filter Listen

Allows you to solo the side-chain filter. This way, you can quickly check the part of the signal that is filtered out using the current settings.

NOTE

This setting is only available in single-band view.

SC Freq

Sets the frequency of the side-chain filter for 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

- Ensure that you enter a space between the note and the cent offset. Only in this case, the cent offsets are taken into account.
- This setting is only available in single-band view.

SC Q

Sets the resonance or width of the side-chain filter for the corresponding band.

NOTE

This setting is only available in single-band view.

Global Settings

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

Spectrum Display

Show Spectrum

Activates/Deactivates the spectrum display.

Peak Hold

Holds the peak values of the spectrum display for a short time.

Smooth

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

Bar Graph

If this option is activated, the frequency spectrum is analyzed into 60 separate bands that are displayed as vertical bars.

Two Channels

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

Slope

Tilts the spectrum display around the 1 kHz pivot.

EQ Curve

Show Curve

Shows/Hides the EQ curve on the spectrum display.

Filled

If this option is activated, the EQ curve is filled. **Amount** allows you to specify the degree of coverage between 10 and 80 %.

GEQ-10/GEQ-30
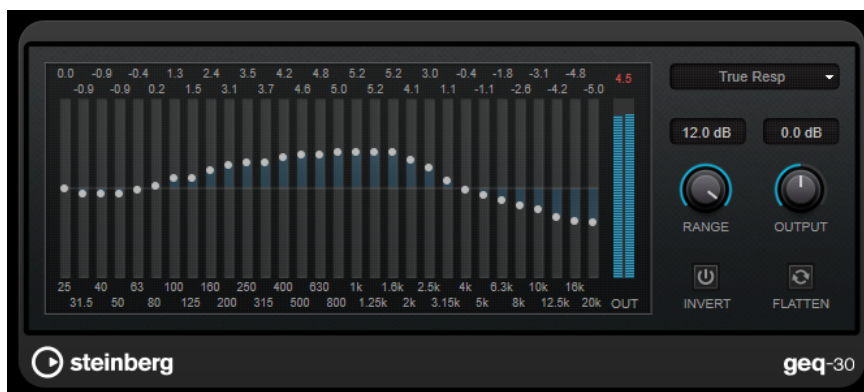
GEQ-30

GEQ-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**.

You can draw response curves on 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.

StudioEQ

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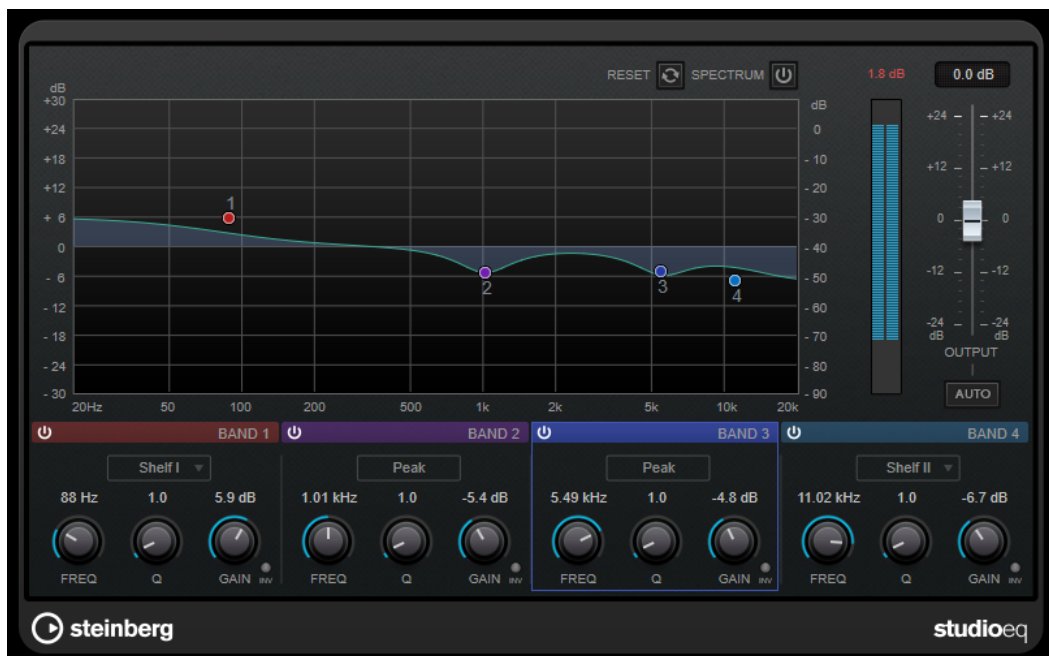
Main Layout

Reset

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

Show Input/Output Spectrum

Shows the spectrum before and after filtering.

Output

Adjusts the overall output level.

Auto Gain

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

Band Settings

Activate/Deactivate Band

Activates/Deactivates the corresponding band.

NOTE

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

Freq

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

NOTE

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

Inv

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

Q

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

NOTE

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

Gain

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

NOTE

- You can adjust the **Gain** parameter of a band in the graphical editor by ▯-clicking the corresponding handle and moving the mouse up and down.
- This parameter is not available for **Cut** filters.

Filter type

For the low and high band, you can choose between three types of shelving filters, a peak filter, and a cut filter. If **Cut** mode is selected, the **Gain** parameter is fixed.

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

Filter

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.

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 change the settings by dragging the curve points on the graphical display, or by adjusting the controls below the display section.

Graphical display

Visualizes the settings for all parameters.

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** controls 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 the 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 this button between the notch filter controls 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** controls 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.

Mastering

Internal Dithering

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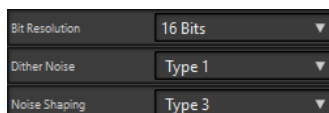
NOTE

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

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

Noise Type

Sets the noise type for adding to the signal.

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

Noise Shaping

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

Bit Resolution

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

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

Leveler

This plug-in is useful for correcting an imbalance or adjusting levels between stereo channels, or for mixing down to mono.
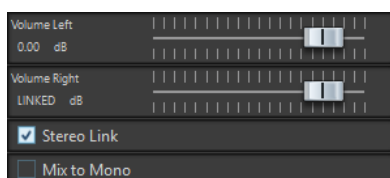
Volume Left/Volume Right (-48 dB to 12 dB)

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

Stereo Link

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

Mix to Mono

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

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.

Lin Pro Dither

Lin Pro Dither is a dithering plug-in that uses advanced algorithms and offers additional noise shaping to increase the apparent signal-to-noise ratio by altering the spectrum of the low-level audio signal. Different noise shaping types allow you to adapt the dithering effect to the audio material to be processed.
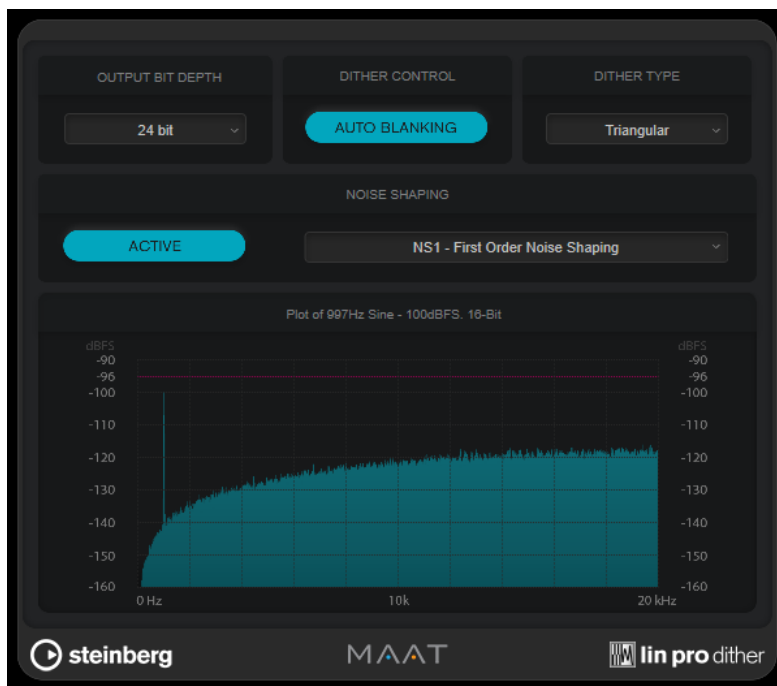
Output Bit Depth

Sets the bit depth of the output signal.

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

Dither Control

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

Dither Type

Allows you to choose between **Triangular** and **Triangular Hi-Pass** dither types.

- If **None** is selected, no dithering is applied but the signal is still rendered to the set **Output Bit Depth**.

Noise Shaping

Activates/Deactivates noise shaping. Noise shaping increases the apparent signal-to-noise ratio. The pop-up menu allows you to choose between different algorithms which move the noise to different frequency ranges.

NOTE

Noise shaping is not available if **None** is selected as **Dither Type**.

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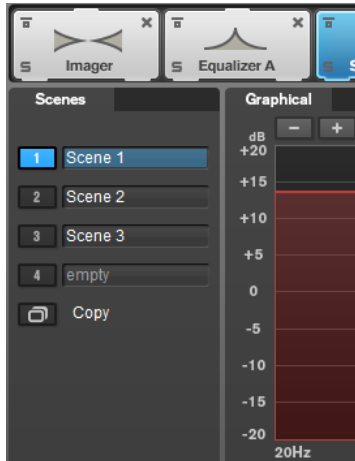
- To copy the settings of a scene to another scene, click **Copy Scene**, and then click the scene button of the scene where you want to paste the settings.

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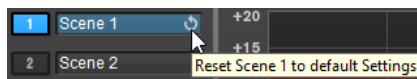
- To rename a scene, double-click the scene name and type in another name.

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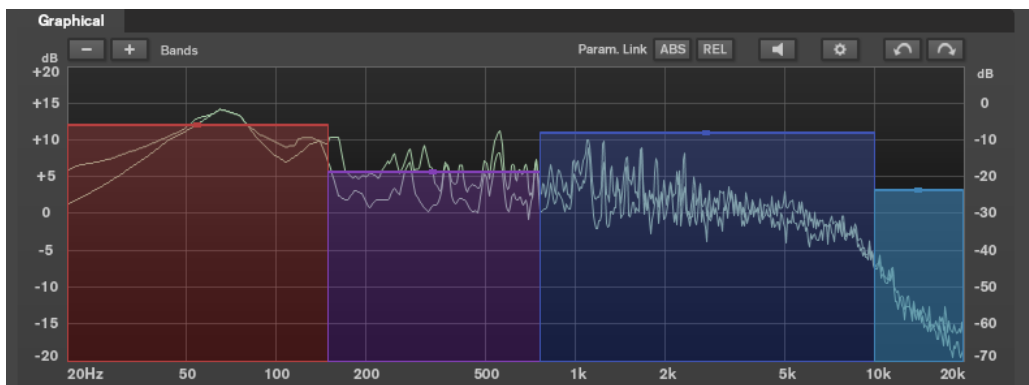
- To define the frequency range of the different frequency bands, use the handles at the sides of each frequency band.

- To attenuate or boost the output level of each frequency band by ±15 dB, use the handles on top of each frequency band.

Settings
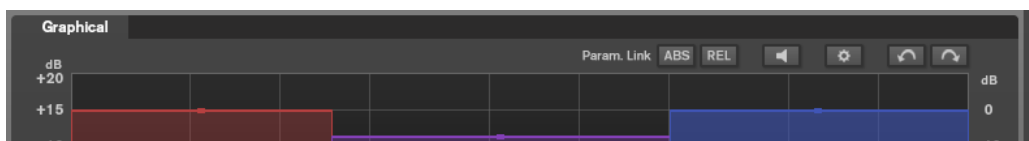
Parameter Linking

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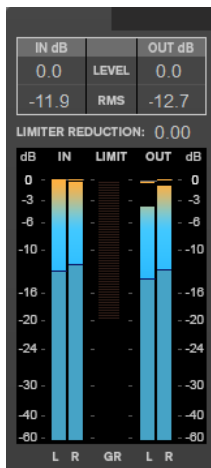
The input/output meter provides a combined peak level, with peak-hold functionality and RMS meter. Between the meters for input and output is the gain reduction meter for the **Limiter**.

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 prevents clipping by ensuring that the output level does not exceed a set value.

Band Settings

On/Off

Activates/Deactivates the corresponding section.

Balance

Mid/Side

> Allows you to set the gain for the mid and side signal.

Solo Mid Signal/Solo Side Signal

> Allows you to solo the mid or side signal.

Transients

The **Transients** section allows you to set the following parameters:

ATT

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

REL

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

Gain

> Sets the output level for the corresponding band.

Harmonics

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

2nd HARM/3rd HARM

Allows 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

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

Release

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

Oversample

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

Stereo Link

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

Output

Sets the output level.

Maximizer

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

Maximizer Type set to Classic

Maximizer Type set to Modern

Type

> **Classic** provides the algorithms of previous versions of this module and is suited for all styles of music.
>
> **Modern** is particularly suited for contemporary styles of music. The algorithm allows for more loudness than **Classic**. **Modern** provides additional settings to control the release phase:
>
> **Release** sets the overall release time.
>
> **Recover** allows for faster signal recovering at the beginning of the release phase.

Optimize

> Determines the loudness of the signal.

Output

> Sets the output level.

Mix

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

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.

Solo Band

Solos the corresponding frequency band.

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.
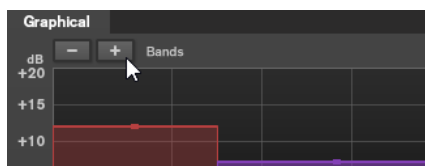
Standard

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 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 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 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 and the wet signal.

Output

Sets the output gain.

RELATED LINKS
Side-Chain Settings on page 87

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.

IMPORTANT

When using **Mid/Side** processing mode, we recommend that you activate **Linear Phase** in order to avoid unwanted sound colorization.

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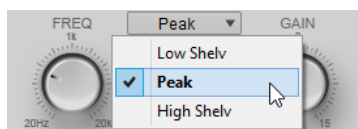
- **Low Shelf** boosts or attenuates frequencies below the cutoff frequency by the specified amount.

- **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

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**.
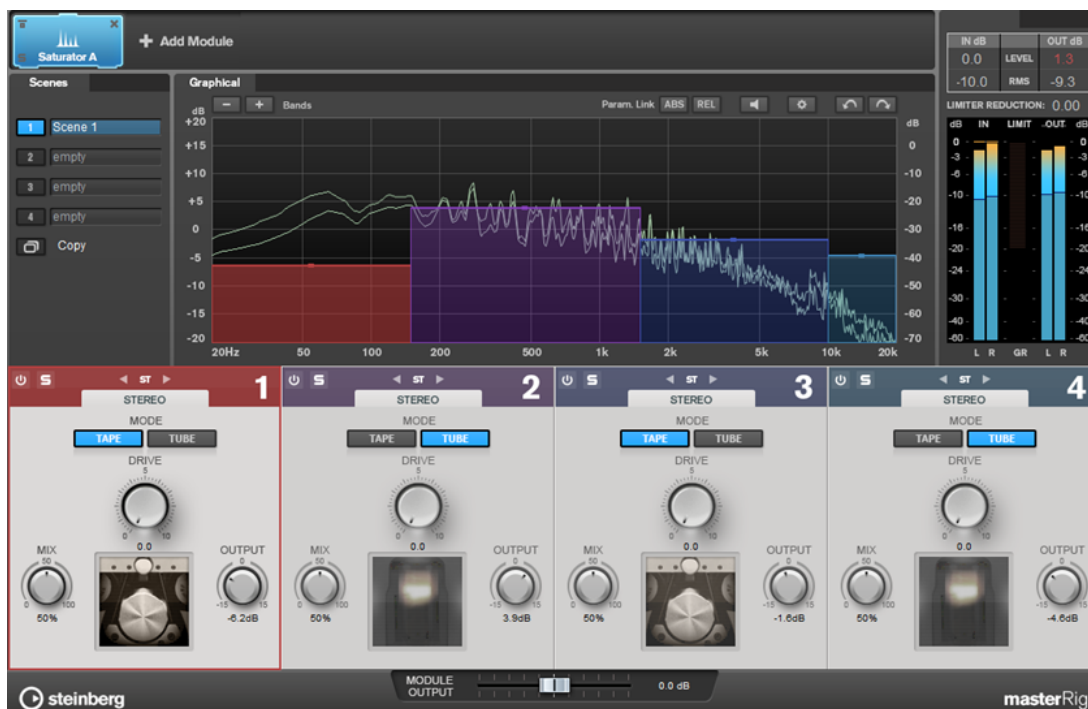
Band Settings

On/Off

Activates/Deactivates the corresponding section.

Solo Band

Solos the corresponding frequency band.

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.
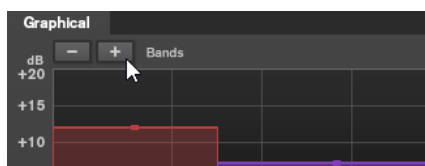
Saturator Section

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 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.

Solo Band

Solos the corresponding frequency band.

Add/Remove Band

Allow you to add and remove bands.
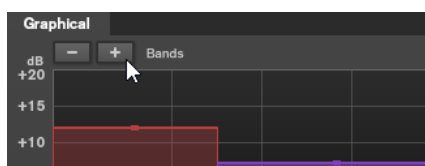
Imager Section

Width

Allows you to control the stereo width per band.

Pan

Allows you to pan the signal left/right.

Output

Sets the output level for each band.

Peak Master

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

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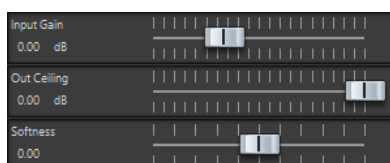
Input Gain

Values range from -12 dB to 24 dB.

Out Ceiling

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

Softness

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

Resampler

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

NOTE

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

| Output Sample Rate | 44 100 Hz ▼ |
| Quality | Best ▼ |
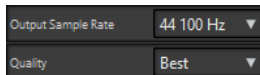
Output Sample Rate

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.

Modulation

AutoPan

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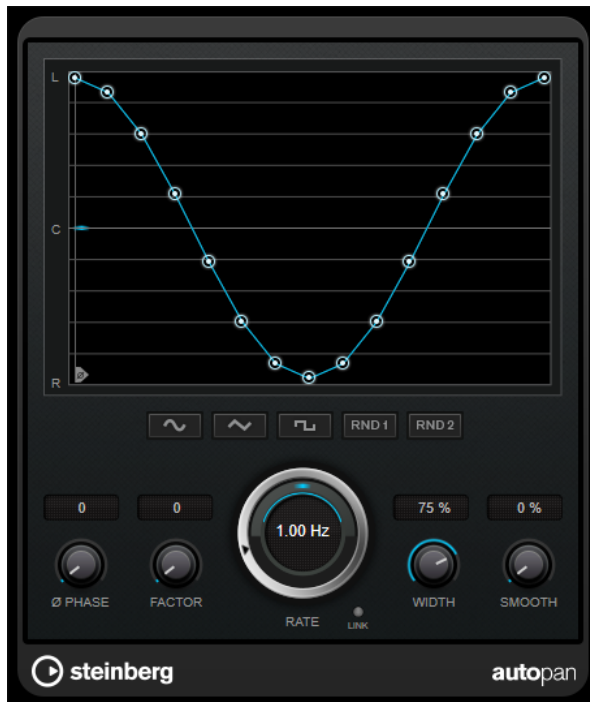
Waveform display

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

Waveform preset buttons

Allow you to select presets for the modulation waveform.

- **Sine** creates a smooth sweep.

- **Triangle** creates a ramp, that is, a linear movement from full right to full left and back.

- **Square** creates an instant jump to full right, then to full left, and then back to center.

- **Random One Shot** creates a random curve. Click this button again to create a new random curve.

- **Random Continuous** automatically creates a new random curve after each period.

Phase

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

Rate

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

Link

> If this button is activated, the left and right channel are modulated simultaneously.
> This results in a chopping effect instead of auto-panning.
>
> In this mode, **Width** sets the intensity of the volume modulation.

Width

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

Smooth

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

Chorus

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

Delay

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

Width

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

Spatial

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

Mix

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

Rate

> Sets the sweep rate.

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.

StudioChorus

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

Delay

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

Width

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

Spatial

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

Mix

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

Rate

Sets the sweep rate.

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.

Pitch Shift

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.

Restoration

DeReverb

DeReverb allows you to remove reverb from the input signal in real time. The plug-in uses an impulse response that is generated by a physical model based on adjustable parameters and a learning function.

When you activate the learning function, the plug-in records a short range of the input signal and automatically adjusts the parameters of the physical model.

Frequency display

Sets the spectral envelope of the impulse response. You can edit the four bands of the spectral envelope range and the output level for each band. You can edit the level of a band or the cutoff frequency between two bands by dragging the corresponding handle.

Algorithm

Sets the restoration algorithm. We recommend to try different algorithms and choose the algorithm that provides the best results.

Play Mode

Sets the playback mode:

- **Input** plays back the wet input signal.

- **Reverb** plays back only the removed reverb portion.

- **Output** plays back the processed output signal.

Reduction

Sets the amount of reverb that is removed from the wet input signal.

IR Physical Model section

Duration

Sets the duration of the impulse response.

Pre-Delay

Sets a pre-delay for the impulse response.

Learn

Activates the learning function.

Play IR

Allows you to playback the generated impulse response.

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.
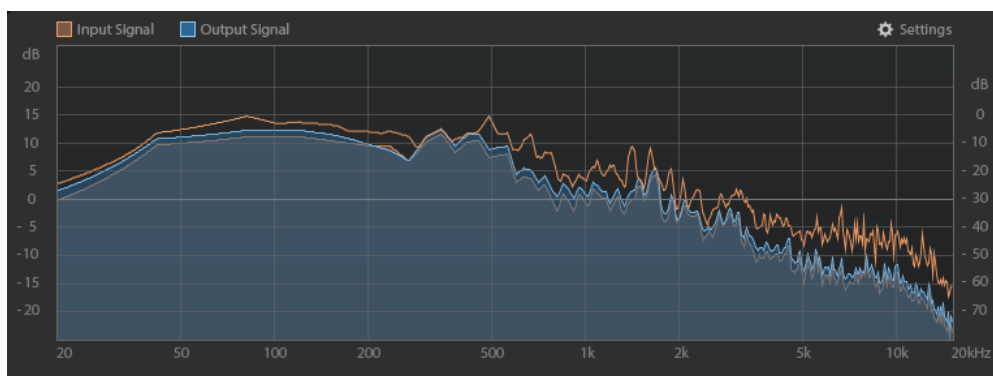
Master
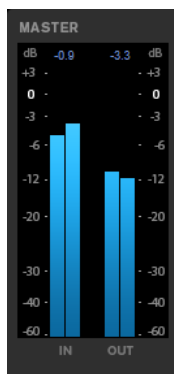
The input/output meter provides a peak level meter.

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
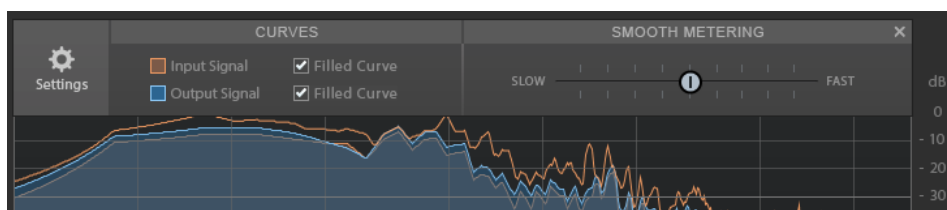
Filled Curve

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.
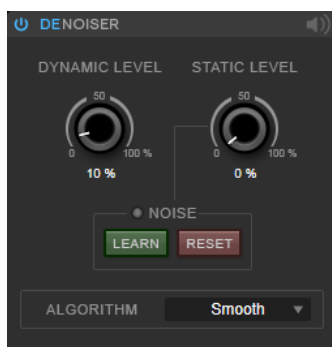
Activate/Deactivate DeNoiser

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**.

Reverb

REVelation

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

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

Pre-Delay

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

Early Reflections

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

ER/Tail Mix

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

Size

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

Low Cut

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

High Cut

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

Delay

Delays the onset of the reverb tail.

Room Size

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

Main Time

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

High Time

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

Low Time

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

High Freq

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

Low Freq

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

Shape

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

Density

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

High Cut

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

Width

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

Mix

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

Lock Mix Value

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

Modulation

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

Modulation Rate

Specifies the frequency of the pitch modulation.

Modulation Depth

Sets the intensity of the pitch modulation.

Modulation Activate

Activates/Deactivates the chorusing effect.

RoomWorks

RoomWorks is a highly adjustable reverb plug-in for creating realistic room ambience and reverb effects in stereo and surround formats. The CPU usage is adjustable to fit the needs of any

system. From short room reflections to cavern-sized reverb, this plug-in delivers high quality reverberation.

Input Filters

Low Frequency

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

High Frequency

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

Low Gain

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

High Gain

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

Reverb Character

Pre-Delay

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

Size

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

Reverb Time

Sets the reverb time in seconds.

Diffusion

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

Width

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

Variation

Clicking this button generates a new version of the same reverb program using altered reflection patterns. This is helpful if some sounds are causing odd ringing or

undesirable results. Creating a new variation often solves these issues. There are 1000 possible variations.

Hold

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

Damping

Low Frequency

Determines the frequency below which low-frequency damping occurs.

High Frequency

Determines the frequency above which high-frequency damping occurs.

Low Level

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

High Level

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

Envelope

Amount

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

Attack

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

Release

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

Output

Mix

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

Wet only

This button deactivates the **Mix** parameter, setting the effect to 100 % wet or affected signal. This button should normally be activated if **RoomWorks** is used as a send effect for an FX channel or a group channel.

Efficiency

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

Export

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

Output meter

Shows the level of the output signal.

RoomWorks SE

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

Pre-Delay

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

Reverb Time

Sets the reverb time in seconds.

Diffusion

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

Low Level

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

High Level

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

Mix

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

Spatial

VST AmbiDecoder

VST AmbiDecoder allows you to convert Ambisonics audio for playback on headphones or multi-channel speaker setups.

For a description of **VST AmbiDecoder**, see the **Operation Manual**.

MixConvert V6

The **MixConvert V6** plug-in can be used to quickly convert a multi-channel mix to a format with a different channel configuration, for example, to mix down a 5.1 surround mix into a stereo mix.

For a description of **MixConvert V6**, see the **Operation Manual**.

Imager

Imager allows you to expand or reduce the stereo width of your audio in up to 4 bands. This way, you can independently adjust the stereo image in defined frequency domains.

Bands

Sets the number of frequency bands.

Live

If this button is activated, a more analog-style filter bank is used. This mode introduces no latency and is better suited for live performances. If this button is deactivated, a more neutral-sounding linear phase filter bank is used, at the cost of introducing latency.

Frequency display

Shows a spectrum display and allows you to edit the band range and the output level for each band.

You can edit the output level of a band or the cutoff frequency between two bands by dragging the corresponding handle.

Output meter

Shows the level of the overall output signal.

Activate/Deactivate Band

Activates/Deactivates the corresponding frequency band.

Solo Band

Solos the corresponding frequency band.

Phase display

A phase scope for each band indicates the phase and amplitude relationship between the stereo channels. The phase scope works as follows:

- A vertical line indicates a perfect mono signal (the left and right channels are the same).

- A horizontal line indicates that the left channel is the same as the right, but with an inverse phase.

- A fairly round shape indicates a well balanced stereo signal. If the shape leans to one side, there is more energy in the corresponding channel.

- A perfect circle indicates a sine wave on one channel, and the same sine wave shifted by 45° on the other.

Generally, the more visible a thread, the more bass is in the signal. The more spray-like the display, the more high frequencies are in the signal.

The phase correlation meters below work as follows:

- The vertical bar shows the current phase correlation.

- With a mono signal, the meter shows +1, indicating that both channels are perfectly in phase.

- If the meter shows -1, the two channels are the same, but one is inverted.

Show/Hide Phase Scope

Shows/Hides the phase scopes and phase correlation meters for all bands.

Width

Sets the stereo width for the corresponding band.

Pan

Sets the left-right panning for the corresponding band.

Output

Sets the output level for the corresponding band.

MonoToStereo

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

NOTE

This plug-in works only on stereo tracks.
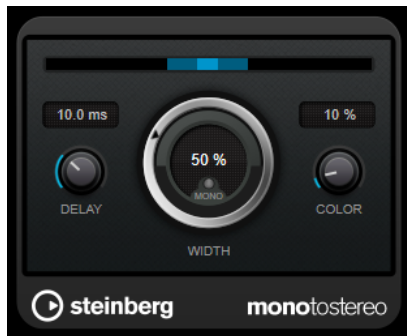
Delay

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

Width

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

Mono

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

Color

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

StereoEnhancer

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

NOTE

This plug-in works only on stereo tracks.
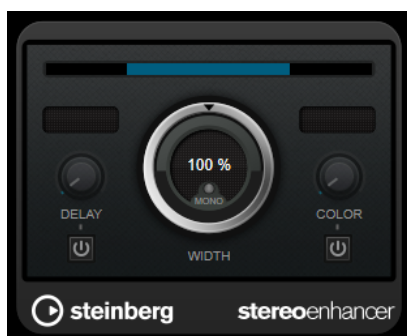
Delay

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

Width

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

Mono

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

Color

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

Stereo Expander

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

Width

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

Surround

Mix6to2

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

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 faders

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

Link

Links the volume faders of a surround channel.

Invert Phase

Inverts the phase of the corresponding surround bus channel.

Output Bus

Volume faders

Set the volume of the mixed output.

Link

Links the **Output** faders.

Normalize

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

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 faders

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

Link

Link the volume faders.

Invert Phase

Inverts the phase of the corresponding surround bus channel.

Output Bus

Volume faders

Set the volume of the mixed output.

Link

Links the **Output** faders.

Normalize

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

Tools

Channel Extractor

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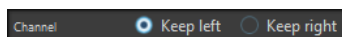
Channel

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

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.

SampleAlign

The **SampleAlign** plug-in allows you to delay one audio channel in a stereo file, relative to the other channel; that is, to modify the samples of either one of the channels or both the left and the right audio channel.

The resulting shift is applied to the entire waveform, with all frequencies shifted by the same value.

The **SampleAlign** plug-in serves the following purposes:

- To correct misalignments: If your audio sounds weak or hollow on playback, this can be the consequence of positioning microphones at varying distances from the sound source during recording: The spatial discrepancy results in time delays between the microphones, which, in turn, lead to sample issues in the mix, with the delayed channel playing back a number of silent samples to compensate for the delay. Shifting the samples of one channel enables you to correct this misalignment and to improve the sound quality.

- To create special sound effects: In audio production, sample shifting is often used for sound design, such as the creation of "swirling" or "spinning" sounds, which you can use in music or in movies and video games.

- To separate microphone channels: You can use the plug-in to emulate stereo microphone channel separation.

- To improve stereo imaging: You can use sample shifting to manipulate the stereo image. For example, by shifting the samples of one channel, you can create the illusion of a sound source being farther to the left or to the right than it actually is. This can be useful in music production, where creating a wide and engaging stereo image is often an expressed goal.

- To conduct psycho-acoustic experiments: Sample shifts are used in binaural beats, where two sine waves with slightly different frequencies are presented to each ear separately. The brain processes the difference in frequencies, which can have various effects on consciousness, from relaxation to focus stimulation.
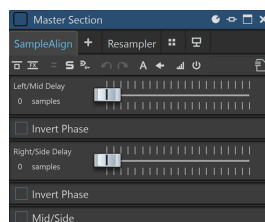
Left/Mid Delay

Allows you to modify the delay for the left or mid channel.

The farther you move the slider to the right, the more samples are shifted.

You can invert the phase by activating **Invert Phase** below the slider.

Right/Side Delay

Allows you to modify the delay for the right or side channel.

The farther you move the slider to the right, the more samples are shifted.

You can invert the phase by activating **Invert Phase** below the slider.

Mid/Side

Activating this option allows you to modify the phase of the mid and side channel.

NOTE

The plug-in is mainly designed to perform micro-shifts, which is why the editing unit is set to samples rather than milliseconds.

The tool is primarily designed for stereo files. When it is applied to mono files, it only takes effect on the delay and phase inversion parameters of the left channel.

IMPORTANT

While sample shifting can be useful, we recommend to apply it with caution. Otherwise, it can lead to phase cancelation, with the left and right channels canceling each other out, significantly reducing the overall loudness or causing other unintended negative effects.

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 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.
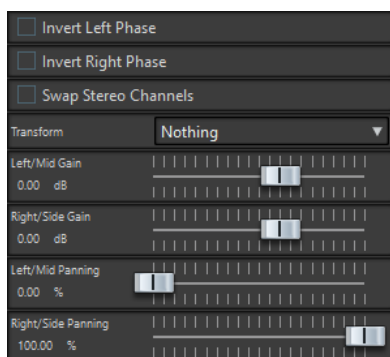
Invert Left Phase/Invert Right Phase

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.

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 put to multiple uses, such as the following:

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

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 **TestGenerator** to an audio track as an effect and activate it, a signal is generated. You can then start recording an audio file as usual, based on the signal specifications.

Interval

Activates/Deactivates the modulation of the generated signal using a pulse signal of 1 kHz with a duration of 200 ms.

Signal type section

Allows you to set the basis for the signal generated by the waveform generator. You can choose from different waveforms (**Sine**, **Triangle**, **Square**, a sweep signal, and **Sawtooth**) and types of noise (**White**, **Pink**, **Brown**, **Blue**, **Gray**, **Violet**).

Frequency section

Allows you to set the frequency of the generated signal. You can select one of the preset values, set a value by using the slider or a quick control, or enter a value into the value field. In the value field, you can set the frequency either in Hz or as a note value. If you enter a note value, the frequency is automatically changed to Hz. For example, a note value of A3 sets the frequency to 440 Hz. When you enter a note value, you can also enter a cent offset; for example, A5 -23 or C4 +49.

NOTE

Enter a space between the note and the cent offset to ensure that 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, set a value by using the slider or a quick control, or enter a value into the value field.

NOTE

The minimum gain value of -120 dB can only be set by entering the value into the value field.

Input Gain

Allows you to add the input signal to the generated signal.

Sweep section

Allows you to adjust the start frequency, the end frequency, and the duration of the sweep signal.

Output buttons

Allow you to activate/deactivate the output signal for each channel separately.

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 exclusively available in the **Batch Processor** window.
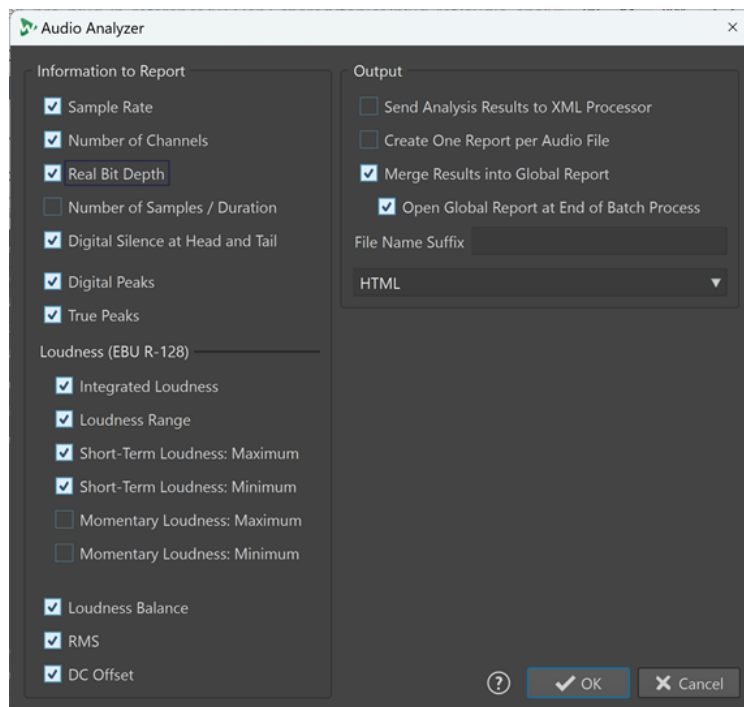
To analyze files without writing anything, select **No Audio Output** via the **Output** tab of the **Batch Processor** window.

Information to Report

This column allows you to select which of the following information to include in the output:

- **Sample Rate**
- **Number of Channels**
- **Real Bit Depth**
- **Number of Samples/Duration**
- **Digital Silence at Head and Tail**

> NOTE
>
> We recommend that you activate this option if a silent section at the beginning and at the
> end of audio material intended for public release is mandatory in your professional context,
> as it facilitates verification for quality control purposes.

- **Digital Peaks**
- **True Peaks**
- **Integrated Loudness**
- **Loudness Range**
- **Short-term Loudness: Maximum**
- **Short-term Loudness: Minimum**
- **Momentary Loudness: Maximum**
- **Momentary Loudness: Minimum**
- **Loudness Balance**
- **RMS**
- **DC Offset**

Output

This column allows you to set up the output of the **Audio Analyzer**. The following options are
available:

Send Analysis Results to XML Processor

Submits the analysis results to the XML or HTML output of the batch processor as
parameters.

Create One Report per Audio File

Generates a report for each audio file in the batch process. The audio file name is used
as the file name.

Merge Results into Global Report

Consolidates the results of the analysis into a global report. The audio file name is used
as the 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

Allows you to specify a file name suffix. This is necessary when using the **Audio
Analyzer** in a batch process multiple times; for example, to see the stats before and
after adding specific plug-ins.

We recommend using a different suffix for each instance of the **Audio Analyzer** plug-in
in the processing chain.

Output format

Allows you to choose from the following output formats:

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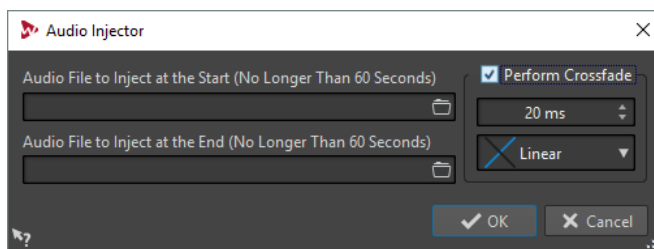
This monopass plug-in is exclusive to the **Batch Processor** window.

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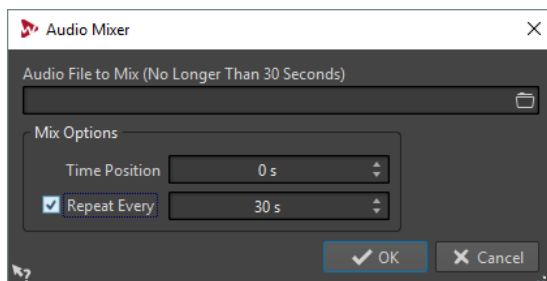
This monopass plug-in is exclusive to the **Batch Processor** window.

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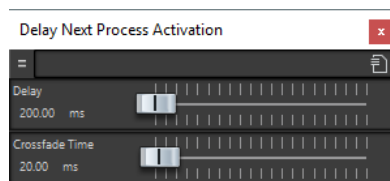
This monopass plug-in is exclusive to the **Batch Processor** window.

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.
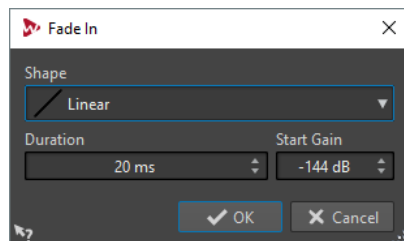
Shape

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.
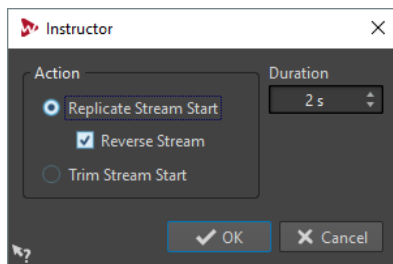
Replicate Stream Start

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.
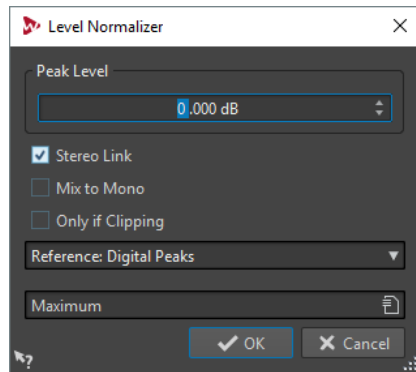
Peak Level

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

The **Loudness Meta Normalizer** batch plug-in allows you to change the loudness level of a batch of files, while maintaining a consistent loudness ratio among them.

The **Loudness Meta Normalizer** plug-in operates by modifying gain values. It does not apply any audio compression, thus ensuring that the original sound quality is not compromised.

The **Loudness Meta Normalizer** serves the following purposes:

- To apply the same gain to all files, while at the same time, ensuring that a specified peak level is not exceeded in any file. If required to prevent clipping, the gain that you specify is automatically reduced by the plug-in, after analyzing all files in the batch, and prior to applying the gain to the files in the batch.

- To set the loudness of a reference file to a specific value, and to offset the gain level of all other files in the batch by the amount that was applied to the reference file to obtain the desired loudness level.

The **Loudness Meta Normalizer** batch plug-in is exclusively available on the **Metapass Plug-ins** panel of the **Batch Processor** window.
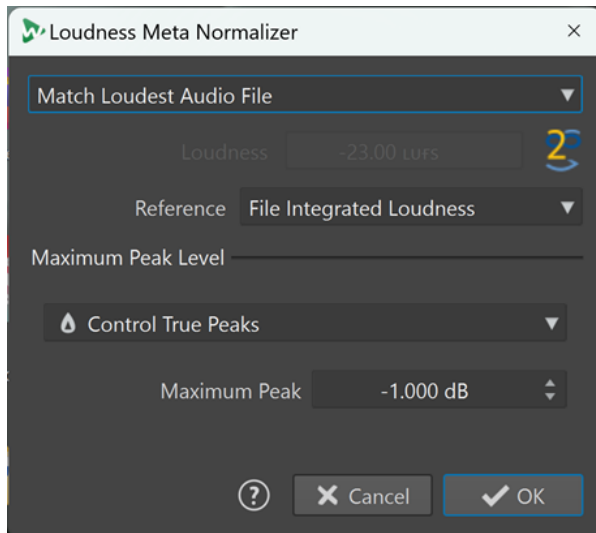
You can choose one of the following options from the **Loudness** pop-up menu at the top:

Match Loudest Audio File

> Applies the same amount of gain to all audio files, so that they match the loudness level of the loudest of them as closely as possible.

Match Maximum Achievable Loudness

> Detects the maximum achievable loudness among all files, without exceeding the specified peak value, before applying the same amount of gain to all audio files, so that they match the maximum loudness level.

Set Same Specific Loudness

> Applies a specific amount of gain to all audio files, to match the specified loudness value as closely as possible.

Set Loudness of Named File, Shift Others

> Allows you to define a single file as the reference, or "master", file. This file is set to the loudness value that you specify in the **Loudness** entry field below, and the same amount of gain is applied to all audio files, regardless of their original individual loudness levels.

> When you select this option, a new entry field appears, which prompts you to enter **Part of Master File Name**. You can either type in the entire file name or a fragment of it, so that the plug-in can identify the file and use it as the "master" or reference file.

> The option is particularly suited for stem mastering.

> **Use case example:**

> You have the following files:

> - **Main Mix**
> - **Drum Stem**
> - **Bass Stem**
> - **Guitar Stem**
> - **Keyboard Stem**

> To set the **Main Mix** file as the "master" file, you can type in the entire file name or a part of it, such as "mix" or "main".

> **NOTE**
>
> Make sure that the name fragment that you enter into the field is unique. If you intend to set **Guitar Stem** as the "master" file, for example, typing in "stem" is not sufficient for the plug-in to clearly identify a file. As a consequence, no "master" file is set.

The option allows you to set the **Main Mix** file to a specific value, such as -18 LUFS Integrated Max, before it automatically adds the same amount of gain to the loudness level of all stem files.

In the **Loudness** entry field, you can specify the loudness value to be matched; for example, -23 LUFS to adhere to the provisions of the EBU R-128 standard in this respect.

You can choose one of the following options from the **Reference** pop-up menu:

File Integrated Loudness

The loudness of all files, using the standard Integrated Loudness value, in accordance with the EBU R-128 standard

Top of Loudness Range

The peak of a loudness range (LRA); that is, the average loudness level of the loudest audio segments

> **NOTE**
>
> The loudest 10% of the audio segments with a duration of less than three seconds are excluded from the calculation, to ensure that exceptionally loud single sounds do not affect the outcome.

Maximum Short-Term Loudness

The highest value resulting from the short-term loudness analysis

You can choose from the following **Maximum Peak Level** options:

- **Control Digital Peaks**

    Limits the sample values, with the peak measured in the digital domain as the reference.

- **Control True Peaks**

    Limits the reconstructed analog sample values, with the peak measured by simulating an analog signal as the reference.

- **Maximum Peak**

    Determines the maximum peak value not to be exceeded.

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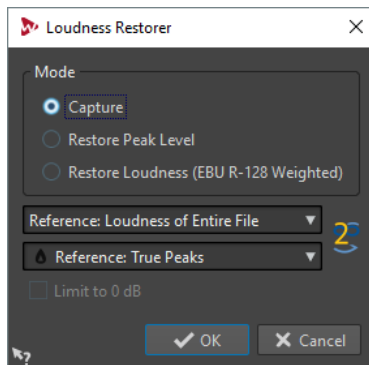
This multipass plug-in is exclusive to the **Batch Processor** window.

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

The **Meta Leveler** plug-in allows you to adjust the levels of all files in a batch by the same value.

The **Meta Leveler** operates by modifying gain values. It does not apply any audio compression, thus ensuring that the original sound quality is not compromised.

The main purpose of this plug-in is to apply the same gain to all files, while at the same time, ensuring that a specified peak level is not exceeded in any file. If required, to prevent clipping, the gain that you specify is automatically reduced by the plug-in, after analyzing all files in the batch, prior to actually applying the gain to the files.
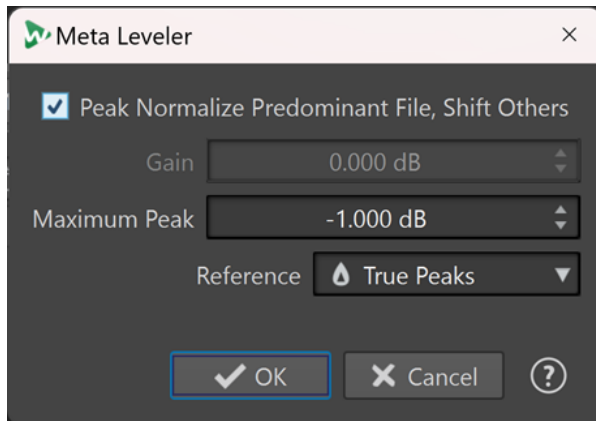
The **Meta Leveler** batch plug-in is exclusively available on the **Metapass Plug-ins** panel of the **Batch Processor** window.

Peak Normalize Predominant File, Shift Others

Adjusts the file with the highest peak (the "predominant" file), so that its peak corresponds to the **Maximum Peak** value, as defined in the entry field below.

All other files are offset by the same amount.

Use case example:

You have the following files:

- File "A" with a peak of -4 dB.
- File "B" with a peak of -6 dB.
- File "C" with a peak of -7 dB.

The "predominant" file is file "A".

With **Maximum Peak** set to -1 dB, a gain of +3 dB is applied to file "A", so that it reaches -1 dB. The same amount of gain (+3 dB) is applied to all other files.

This results in the following peak levels:

- -1 dB for file "A".
- -3 dB for file "B".
- -4 dB for file "C".

NOTE

With this option activated, it is not possible to set a specific value for **Gain**.

Gain

Allows you to define a specific gain to be applied to all audio files.

NOTE

The actual gain that is applied to individual files may be lower than the value you enter, or even negative, if required, to ensure that the value specified for **Maximum Peak** is not exceeded.

Maximum Peak

Allows you to set the maximum peak level to a specific value.

- With **Peak Normalize Predominant File, Shift Others** activated, the **Maximum Peak** value specifies the maximum peak level of the "predominant" file; that is, the file with the highest peak.

- With **Peak Normalize Predominant File, Shift Others** deactivated, the **Maximum Peak** value specifies the maximum peak level to be set for all files at the end of the process.

Reference

Allows you to choose from the following options for analyzing the batch of files:

- **True Peaks**: The analysis is based on analog reconstructed values.
- **Digital Peaks**: The analysis is based on sample values.

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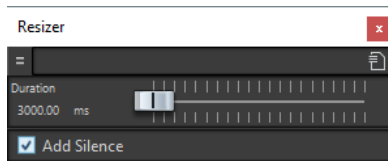
This monopass plug-in is exclusive to the **Batch Processor** window.

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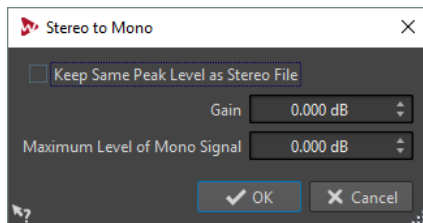
This multipass plug-in is exclusive to the **Batch Processor** window.

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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