Plug-in Reference

NUENDO 11

Premium Media Production System

steinberg

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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

Included Effect Plug-ins

The included plug-in effects are arranged according to their categories.

Ambisonics Plug-ins

VST AmbiConverter

VST AmbiConverter allows you to convert Ambisonics audio between Furse-Malham (FuMa) and AmbiX format.

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

Analyzer Plug-ins

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. Its up to 9 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.

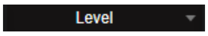
NOTE

- 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 in the paused display to set the project cursor to this position.

Hold Current Values on Stop

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

Module selector

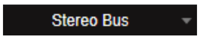


Allows you to select a module for the selected slot.

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.
- If side-chain is activated, you can select between **Main** or **Side-Chain** channels. Combined **Main & Side-Chain** views are also available for some modules.

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 playback is started.

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.

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

RELATED LINKS
Module Settings Window on page 6
Signal Modules on page 7
Spectral Domain Modules on page 14
Phase Modules on page 19
Spatial Domain Modules on page 22
Waveform Modules on page 24
Other Modules on page 27

Module Settings Window

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

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

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.

RELATED LINKS
Signal Modules on page 7
Spectral Domain Modules on page 14
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Other Modules on page 27

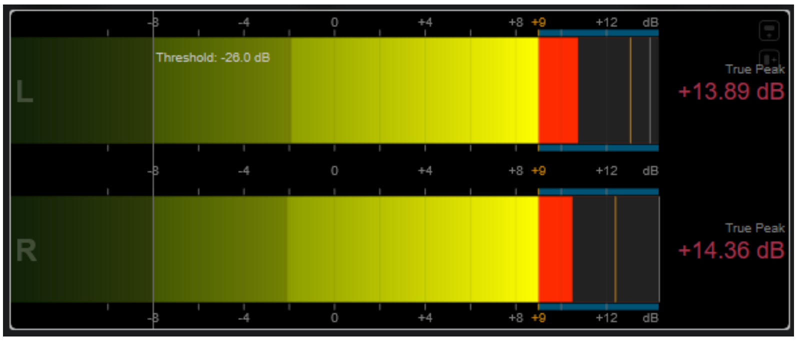
Signal Modules

The modules in this category allow you to visualize the level, loudness, or intelligibility 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 dbB Digital).

NOTE

You can customize the appearance of the meter for all scales individually in the **Preferences** dialog (**Metering**—**Appearance** page).

Peak Hold

Specifies for how long the peak levels are held in 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 **Ctrl/Cmd**, 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.

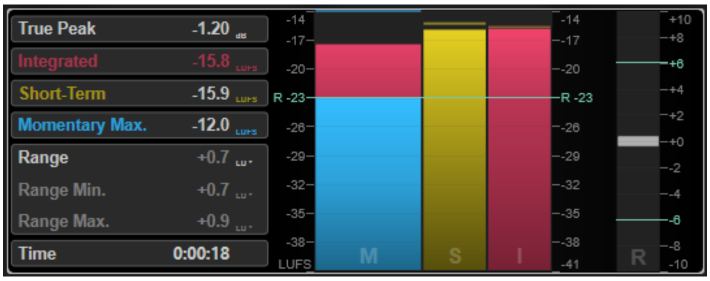
Max. Value

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.

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.

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 switch the meter between the EBU +9 scale and the EBU +18 scale.

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. The loudness meter indicates clipping, when the integrated reference value is reached.

Tol. Momentary

Sets a tolerance value for the maximum momentary loudness.

Ref. Range

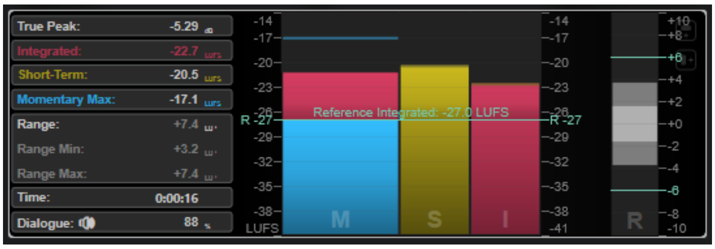
Sets a reference value for the loudness range. If higher values are detected, the loudness meter indicates clipping.

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.

Time

Shows the overall duration of the loudness measurement.

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.

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 switch the meter between the EBU +9 scale and the EBU +18 scale.

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. The loudness meter indicates clipping, when the integrated reference value is reached.

Tol. Momentary

Sets a tolerance value for the maximum momentary loudness.

Ref. Range

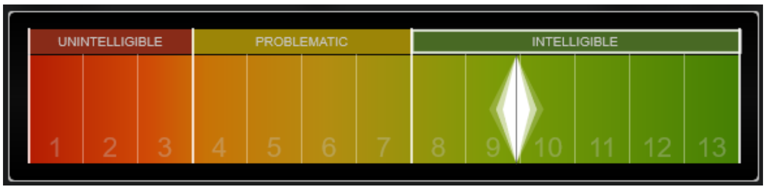
Sets a reference value for the loudness range. If higher values are detected, the loudness meter indicates clipping.

Tol. Range

Sets a tolerance value for the loudness range.

Intelligibility

This module shows the intelligibility of spoken word within your mix. The rhombus turns white if spoken word is detected and its position indicates how well it can be understood.



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 and use the mouse wheel to adjust this parameter.

Language

Allows you to specify the spoken language in your audio.

Hearing-Impaired

Adapts the intelligibility algorithm to the needs of hearing-impaired people.

Voice over Voice

Adapts the intelligibility algorithm to audio with voice over voice, for example, dubbing over the original speech in the background.

Pause Detection

Reduces the sensitivity of the speech detection by suspending the detection algorithm for the set time. This parameter is useful if the algorithm recognizes audio incorrectly as speech. You can reduce very short false positives by raising this value.

NOTE

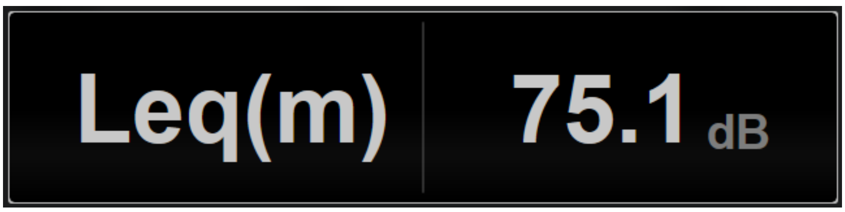
The intelligibility display is delayed by the time set for this parameter.

Hold Detection

Specifies how long the white rhombus indicates detected speech. Raising the value of this parameter allows you, for example, to bridge short pauses that a speaker makes between words.

Leq(m)

This module shows the average volume over time, using a filter that emphasizes the mid and upper range frequencies, according to the Trailer Audio Standards Association (TASA). This measurement is mainly used to ensure that a film trailer respects the volume limits for trailers in movie theaters.



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

Ref. Level

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

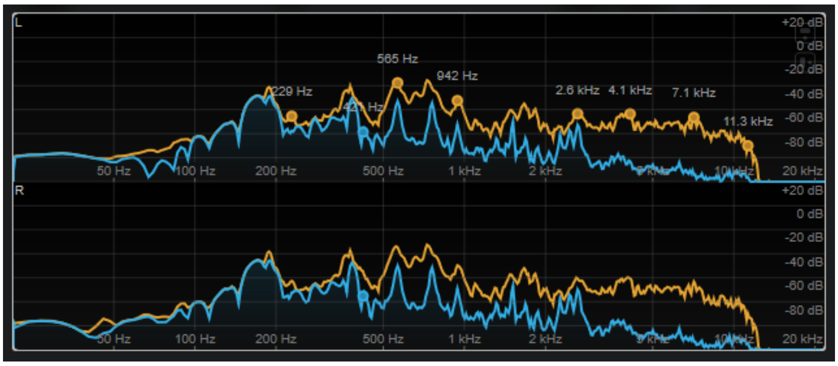
Spectral Domain Modules

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.

When using side-chaining, this module also allows you to detect the regions in your main signal that are acoustically masked by the side-chain signal.

NOTE

In addition to the signal of the track, the display can show the signal of a side-chain input. For this to work, you must select a **Main & Side-Chain** view from the channel selector. The side-chain signal is then shown in white.

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

Time Smooth

Smoothes the temporal display of the spectrum curve.

NOTE

Alternatively, you can move the mouse pointer over the selected module, hold **Alt**, 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.

Masking

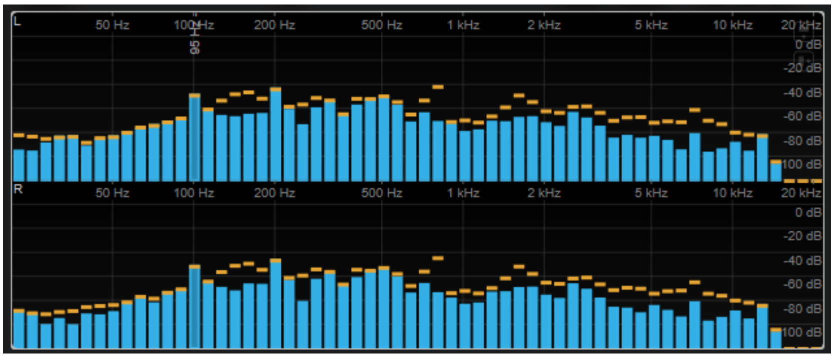
If this button is activated, frequency ranges that are affected by a side-chain signal are displayed.

NOTE

For this to work, you must activate side-chaining and select a **Main + Side-Chain** channel in the channel selector on the toolbar.

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 of the spectrum curve.

NOTE

Alternatively, you can move the mouse pointer over the selected module, hold **Alt**, 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 **Ctrl/Cmd**, 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

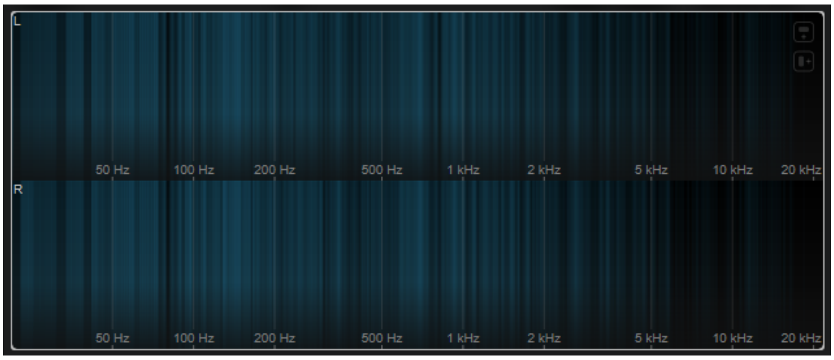
Sets the maximum value of the scale.

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 of the spectrum curve.

NOTE

Alternatively, you can move the mouse pointer over the selected module, hold **Alt**, 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

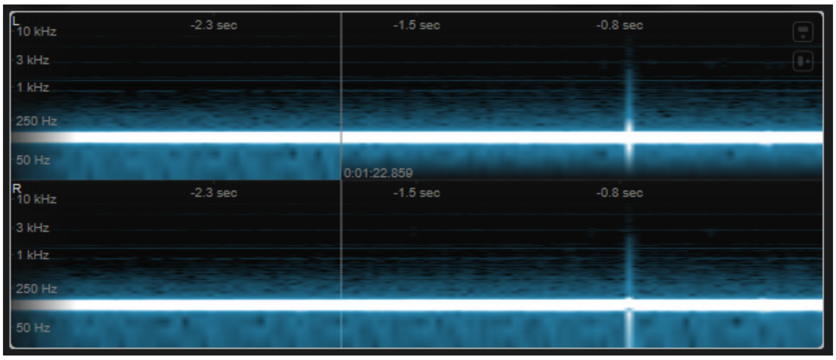
Sets the maximum value of the scale.

Slope

Adds a slope to the frequency spectrum.

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 **Ctrl/Cmd**, 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

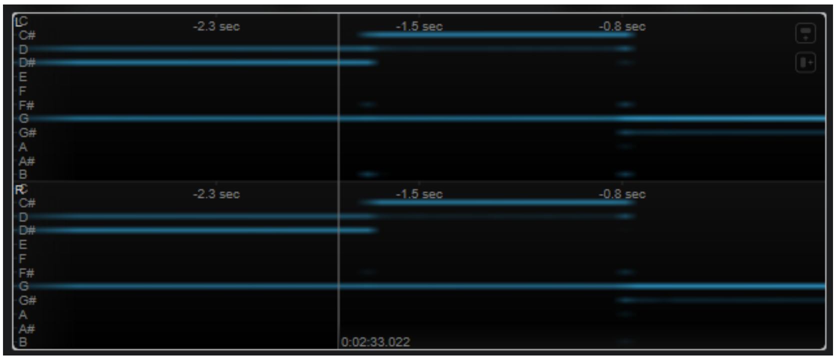
Sets the maximum value of the scale.

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 **Ctrl/Cmd**, 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.

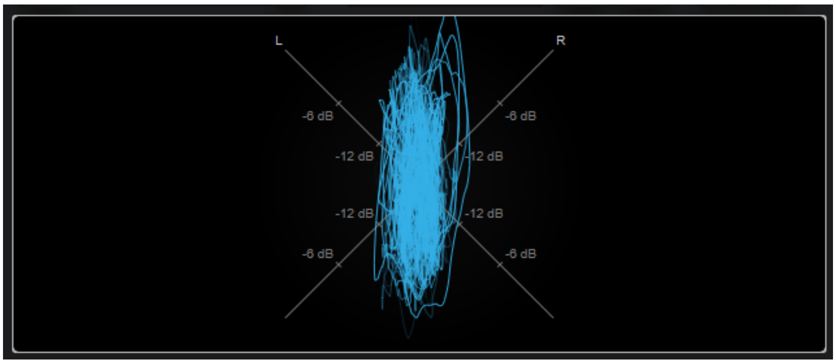
Phase Modules

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.

NOTE

In addition to the signal of the track, the display can show the signal of a side-chain input. For this to work, you must select a **Main & Side-Chain** view from the channel selector. The side-chain signal is then shown in white.

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

Zoom

Allows you to zoom in 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

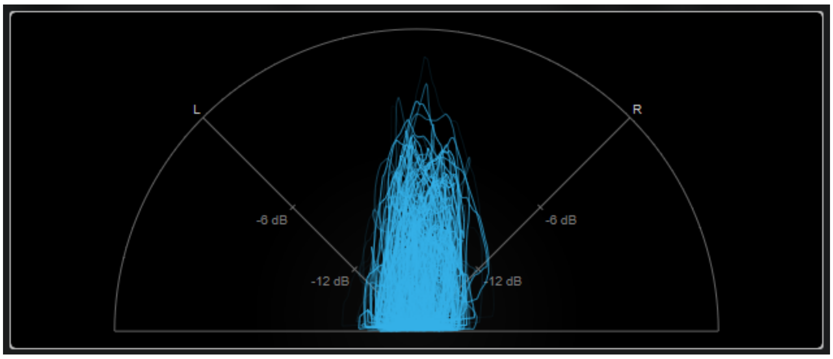
Activates/Deactivates the axis labeling.

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.



NOTE

In addition to the signal of the track, the display can show the signal of a side-chain input. For this to work, you must select a **Main & Side-Chain** view from the channel selector. The side-chain signal is then shown in white.

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

Zoom

Allows you to zoom in 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

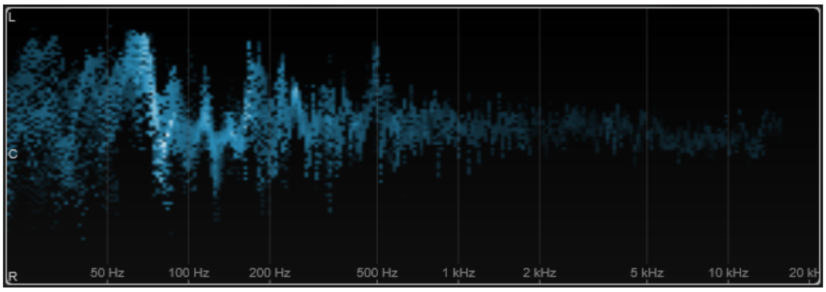
Activates/Deactivates the axis labeling.

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.

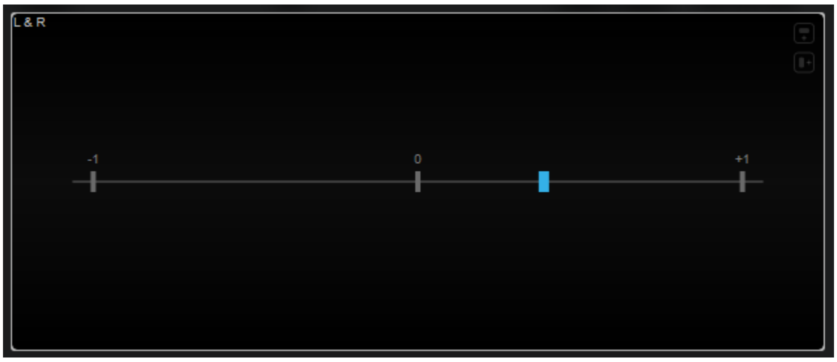
Sets the number of bands per octave.

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 settings are available in the **Module Settings** window:

Time Smooth

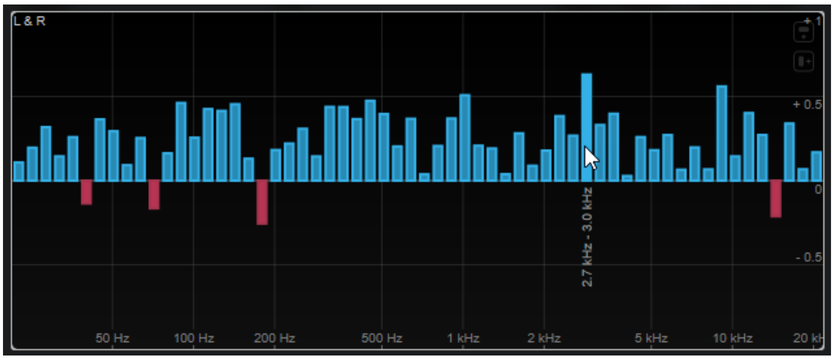
Smoothes the temporal display of the correlation.

NOTE

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

Bands/Oct.

Sets the number of bands per octave.

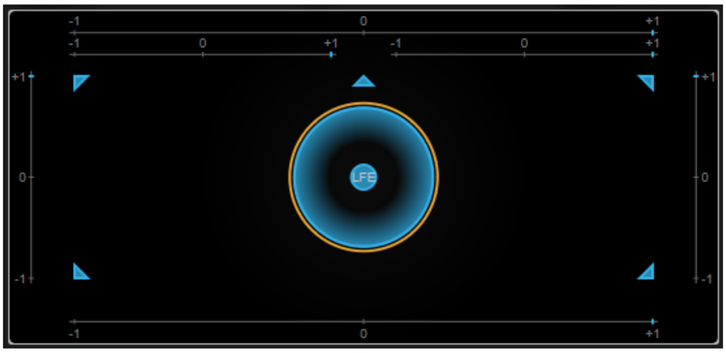
Spatial Domain Modules

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 in 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

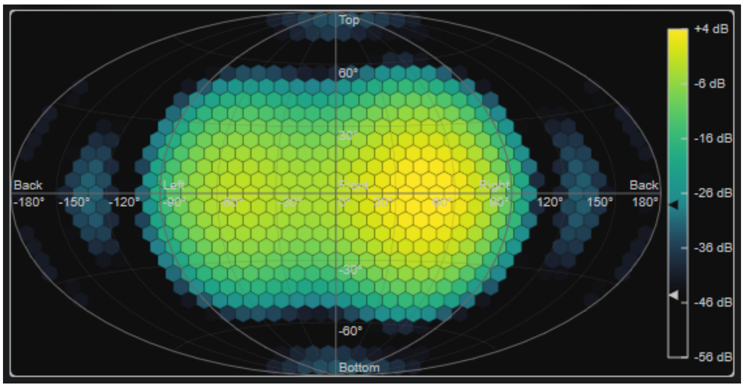
- Alternatively, you can move the mouse pointer over the selected module, hold **Ctrl/Cmd**, and use the mouse wheel to adjust this parameter.
- If this control is turned all the way to the left, the peak envelope is disabled.

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.

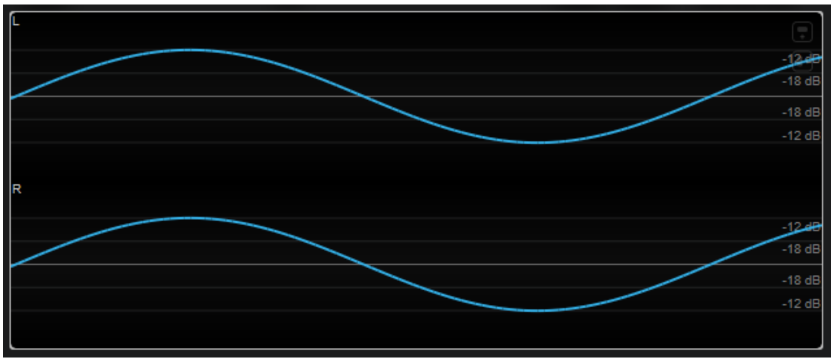
Waveform Modules

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.



NOTE

In addition to the signal of the track, the display can show the signal of a side-chain input. For this to work, you must select a **Main & Side-Chain** view from the channel selector. The side-chain signal is then shown in white.

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

Zoom

Allows you to zoom in 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 in 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

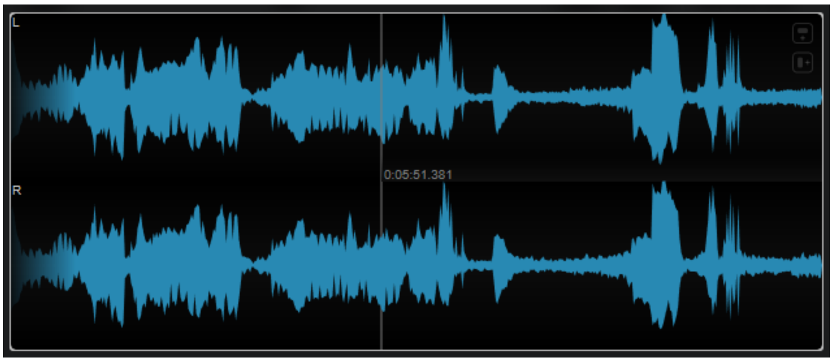
This option is only available if **Auto Zoom** is deactivated.

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.

NOTE

In addition to the signal of the track, the display can show the signal of a side-chain input. For this to work, you must select a **Main & Side-Chain** view from the channel selector. The side-chain signal is then shown in white.

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

Zoom

Allows you to zoom in 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 **Ctrl/Cmd**, 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

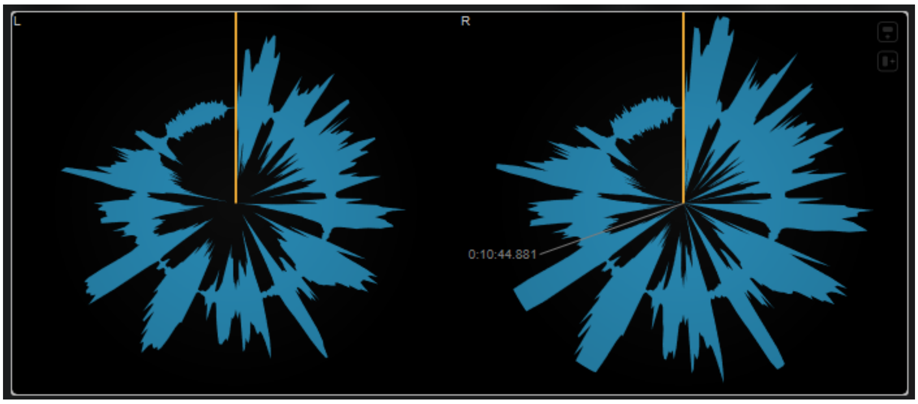
This option is only available if **Auto Zoom** is deactivated.

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.

NOTE

In addition to the signal of the track, the display can show the signal of a side-chain input. For this to work, you must select a **Main & Side-Chain** view from the channel selector. The side-chain signal is then shown in white.

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

Zoom

Allows you to zoom in 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 **Ctrl/Cmd**, 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 Modules

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 changes.

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

Mode

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

Delay Plug-ins

ModMachine

ModMachine combines delay modulation and filter modulation. Frequency and resonance of the filter are modulated by an LFO or can be set manually. An additional **Drive** parameter allows for distortion effects.

Delay

Nudge

Clicking this button once momentarily speeds up the audio coming into the plug-in, simulating the nudge command of analog tape machines.

Rate

If **Tempo Sync** is activated, **Rate** allows you to specify the base note value for tempo-syncing the effect (1/1 to 1/32, straight, triplet, or dotted).

If **Tempo Sync** is deactivated, the modulation speed can be set freely with the **Rate** dial.

Tempo Sync (Delay Modulation)

Activates/Deactivates tempo synchronization for the **Rate** parameter.

Width

Sets the amount of delay modulation. This allows you to create a vibrato or chorus-like effect.

Delay

If **Sync** is activated, this sets the base note value for the delay. If **Sync** is deactivated, the delay time can be set freely in milliseconds.

Tempo Sync (Delay)

Activates/Deactivates tempo sync for the **Delay** parameter.

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.

Drive

Adds distortion to the feedback loop. The longer the feedback, the more the delay repeats are distorted over time.

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.

Graphical display

Functional diagram

Shows the signal path, depending on the settings for **Filter Position** and **Filter Type**.

Filter Position

Allows you to select the filter position. **Loop** places the filter in the feedback loop of the delay. **Output** places it in the output path of the effect, after the **Drive** and **Feedback** parameters.

Filter Type

Allows you to select a filter type. You can choose between a **Low-Pass**, **Band-Pass**, and **High-Pass** filter.

Filter

LFO/Manual (Frequency)

Toggles between **LFO** and **Manual** mode. In **LFO** mode, you can define the modulation rate or sync it to the project tempo. In **Manual** mode, you can set the frequency manually.

Speed (Frequency)

Sets the speed of the filter frequency LFO modulation. If tempo sync is activated, this parameter sets the base note value for synchronizing the modulation to the tempo of the host application.

If tempo sync is deactivated, the speed can be set freely with the **Speed** knob.

Tempo Sync (Frequency)

Activates/Deactivates tempo sync for the **Speed** parameter. This parameter is only available in **LFO** mode.

Low Range/High Range (Frequency)

Set the range of the filter frequency modulation. These parameters are only available in **LFO** mode.

Frequency

Sets the cutoff frequency for the filter. This parameter is only available in **Manual** mode.

Spatial (Frequency)

Sets an offset between the channels to create a stereo panorama effect for the filter frequency modulation. Turn clockwise for a more pronounced stereo effect.

LFO/Manual (Q-Factor)

Toggles between **LFO** and **Manual** mode. In **LFO** mode, you can define the modulation rate or sync it to the project tempo. In **Manual** mode, you can set the resonance manually.

Speed (Q-Factor)

Sets the speed of the filter resonance LFO modulation. If tempo sync is activated, this parameter sets the base note value for tempo syncing the modulation.

If tempo sync is deactivated, the speed can be set freely with the **Speed** knob.

Tempo Sync (Q-Factor)

Activates/Deactivates tempo sync for the **Speed** parameter.

Low Range/High Range (Q-Factor)

Set the range of the filter resonance modulation. These parameters are only available in **LFO** mode.

Q-Factor

Sets the resonance of the filter. This parameter is only available in **Manual** mode.

Spatial (Q-Factor)

Sets an offset between the channels to create a stereo panorama effect for the filter resonance modulation. Turn clockwise for a more pronounced stereo effect.

MonoDelay

This is a mono delay effect. The delay line uses tempo-based or freely specified delay time settings.

Lo Filter

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

Hi Filter

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

Delay

Sets the delay time in milliseconds.

Sync

Activates/Deactivates tempo sync.

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.

NOTE

This plug-in supports external side-chaining. You can use the side-chain input to control the delay from another signal source. If the side-chain signal exceeds the threshold, the delay repeats are silenced. If the signal drops below the threshold, the delay repeats reappear. For a description of how to set up side-chain routing, see the **Operation Manual**.

MultiTap Delay

MultiTap Delay is a versatile tap effect delay with up to 8 tap repeats that allows you to create and edit taps manually, to create taps by clicking a rhythm, or to create random taps. You can set up separate effect chains for the delay loop, for delay taps, and for the overall delay output, with each chain containing up to 6 different effects.

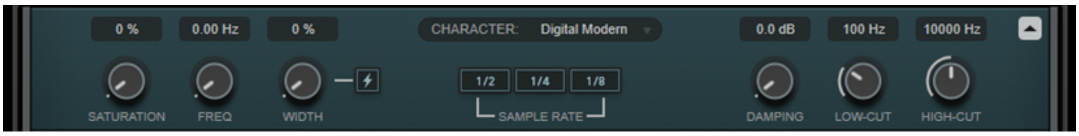
The plug-in offers predefined sound characters that you can freely customize. The delay line uses either tempo-based or freely specified delay time settings. The integrated ducker attenuates the delay output depending on the input signal level, which keeps the delayed signal rather dry during loud or intensely played passages.

NOTE

This plug-in supports external side-chaining. For a description of how to set up side-chain routing, see the **Operation Manual**.

Delay Character section

This section allows you to shape the overall sound of the delay. You can choose between 4 predefined sound characters and adjust their parameters to your liking.



Show/Hide Delay Character Section

Shows/Hides this section.

Character pop-up menu

Allows you to choose the general delay character. **Digital Modern**, **Digital Vintage**, **Tape**, and **Crazy** character are available. Modifying parameters in this section sets the delay character to **Custom**.

Saturation

Sets the amount of saturation. This effect is inserted into the delay loop, which means that the saturation of the signal is increased with each repeat.

Freq

Sets the frequency of the delay modulation.

Width

Sets the intensity of the delay modulation. If this value is set to 0, the signal is not modulated. Activating **Extreme Modulation** increases the modulation to such an extent that drastic variations in tape speed become audible.

Sample Rate

Sets the ratio to which the audio sample is decimated. Lower sample rates reduce the high frequency content and sound quality. If no button is activated, no downsampling occurs.

Damping

Sets the amount of high-frequency damping in the feedback loop.

Low-Cut

Sets the frequency below which low-frequency damping occurs.

High-Cut

Sets the frequency above which high-frequency damping occurs.

Main section

This section contains the general delay parameters and settings, and allows you to add and edit the delay taps.

Delay

If **Sync** is activated, this sets the base note value for the delay. If **Sync** is deactivated, the delay time can be set freely in milliseconds.

Sync

Activates/Deactivates tempo sync.

Lock/Unlock Delay Value and Number of Taps

Locks/Unlocks the values of the **Delay** and the **Taps** parameters when loading presets.

Erase Delay Line

Erases the delay line.

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.

Taps display

Allows you to move taps by dragging their handles and to delete them by double-clicking.

- The **Level** tab allows you to set the level of all taps individually.
- The **Panorama** tab allows you to pan the taps individually in the stereo panorama.

If the plug-in is used for tracks with a multi-channel channel configuration, the display shows the room from the top and the taps as panning handles. You can position the taps by dragging the handles or by entering **L/R** and **F/R** values for the selected handle. The **Input** pop-up menu allows you to choose the input channels for the panner:

- **L/R** uses the left and right channels of the input signal. This allows you to pan the stereo signal in the room. In this mode, no signal is routed to the front or rear center speakers.
- **Sum** uses the mono sum signal of all channels of the input signal. In this mode, you can pan the signal to any speaker in the channel configuration.

- The **Tap Parameters** tab allows you to adjust the parameters of the selected effect module in the **Tap Effects** section. The pop-up menus provide direct access to the effect modules and their parameters.
- **Spread Taps Evenly for Editing** allows easier editing of taps if they are located very close to each other.

Grid

Sets the quantize grid. Taps are quantized to the grid when added or moved.

NOTE

If you hold **Shift**, you can move taps freely even if a grid is set.

Activate/Deactivate Tapping Mode

Activates/Deactivates tapping mode so that you can click the **Tap Rhythm** button to create taps.

Tap Rhythm

Allows you to create taps by clicking a rhythm with the left mouse button.

Quantize

Quantizes all taps to the grid.

Randomize

Sets a random number of taps and tap parameter settings. The **Random Taps Options** pop-up menu allows you to specify the minimum and maximum number of taps, the timing, the panning range, and the level range of the random function.

Taps

Sets the number of taps.

Link/Unlink Taps

Allows you to move all taps simultaneously in the taps display.

Reset Taps

Resets the number of taps and all tap parameters to the default.

Output meter

Shows the level of the output signal.

Output

Adjusts the overall output level.

Ducker

This effect attenuates the delay output depending on the input signal level. If the level of the input signal is high, the effect signal is lowered, or ducked. If the level of the input signal is low, the effect signal is raised.

- **FB** suppresses feedback when the delay signal is ducked.

- **DL** erases the delay line one time as soon as the ducking of the delay signal starts.
- **Amount** sets the amount of level reduction that is applied to the delay output. The meter to the right shows the current amount of gain reduction.
- **Release** sets the time after which the effect signal returns to the original level.

NOTE

This plug-in supports external side-chaining. You can use the side-chain input to control the ducking effect from another signal source. If the side-chain signal exceeds the threshold, the delay repeats are ducked. If the signal drops below the threshold, the delay repeats reappear. For a description of how to set up side-chain routing, see the **Operation Manual**.

Play Sample Sound

Plays back a sound sample which allows you to quickly valuate the result of the current plug-in parameter settings.

Spatial

Sets the stereo width for the left/right repeats. Turn clockwise for a more pronounced 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.

Lock/Unlock Mix Value

Locks the value of the **Mix** parameter in the main section when loading presets.

Effects sections

These sections allow you to add, edit, and remove loop, tap, and post effects. You can create effect chains by adding up to 6 different effect modules from 14 effects available overall. You can modify the order of the effects in the chain by dragging the modules.

- Effects in the **Loop Effects** section feed the output signal back into the delay input, allowing for a continuously increasing effect through the loop effect chain.
- Effects in the **Tap Effects** section process the output signal of each delay tap. You can activate/deactivate each effect and set individual parameters for any single tap.
- Effects in the **Post Effects** section affect the overall output signal of the plug-in.

Bypass

Bypasses the effect chain in the corresponding effects section.

Mix

Sets the balance between dry signal and wet signal for the corresponding effect chain.

Loop Effects/Tap Effects/Post Effects

Shows/Hides the corresponding effects section. The section is highlighted if at least one effect module has been added.

Show/Hide Functional Diagram

Shows/Hides a functional diagram of the signal flow for **Loop Effects**, **Tap Effects**, or **Post Effects** in the taps display.

Add Module

Allows you to add modules to the effect chain of the corresponding section.

Tap Effects Options

Opens the options panel for tap effects. **Suspend Effect When Disabled** stops effects from being processed when they are bypassed or deactivated. This reduces the processing load.

IMPORTANT

If this option is activated, crackles may occur when bypassing or activating/deactivating an effect.

Parameter Link (only available in the Tap Effects section)

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

- If **Absolute Mode** is activated and you edit a parameter value of one tap, the corresponding parameter values of the other taps are set to the same value.
- If **Relative Mode** is activated and you edit a parameter value of one tap, the relation of the corresponding parameter values of the other taps remains the same.

Tap 1-8 (only available in the Tap Effects section)

Allows you to select a tap for editing the effect parameters.

Activate/Deactivate Effect (only available in the Tap Effects section)

Allows you to activate/deactivate the effect for the tap that is selected in the tap display.

RELATED LINKS
Effect Modules on page 36

Effect Modules

Modules allow you to create an effect chain. Each effect can be used only once in the module chain. You can drag modules in the module chain to rearrange them and change the processing order.

General Effect Settings

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.

The following effect modules are available:

Chorus

This is a single-stage chorus effect that doubles the input signal with a slightly detuned version.

Rate

If **Tempo Sync** is activated, **Rate** allows you to specify the base note value for synchronizing the modulation sweep to the tempo of the host application (1/1 to 1/32, straight, triplet, or dotted).

If **Tempo Sync** is deactivated, the sweep rate can be set freely with the **Rate** dial.

Sync

Activates/Deactivates tempo sync.

Width

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

Tone

Changes the tonal characteristic of the output signal.

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.

Flanger

This is a classic flanger effect.

Rate

If **Tempo Sync** is activated, **Rate** allows you to specify the base note value for synchronizing the flanger sweep to the tempo of the host application (1/1 to 1/32, straight, triplet, or dotted).

If **Tempo Sync** is deactivated, the sweep rate can be set freely with the **Rate** dial.

Sync
Activates/Deactivates tempo sync.

Feedback
Determines the character of the flanger effect. Higher settings produce a more metallic-sounding sweep.

Tone
Changes the tonal characteristic of the output signal.

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.

Phaser

This is a classic phasing effect.

Rate
If **Tempo Sync** is activated, **Rate** allows you to specify the base note value for tempo-syncing the phaser sweep (1/1 to 1/32, straight, triplet, or dotted).

If **Tempo Sync** is deactivated, the sweep rate can be set freely with the **Rate** dial.

Sync
Activates/Deactivates tempo sync.

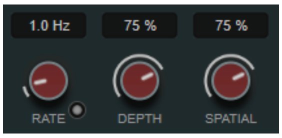
Width
Sets the intensity of the modulation effect between higher and lower frequencies.

Tone
Changes the tonal characteristic of the output signal.

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.

Vibrato

This is a pitch modulation effect.



Rate
If **Tempo Sync** is activated, **Rate** allows you to specify the base note value for tempo-syncing the effect (1/1 to 1/32, straight, triplet, or dotted).

If **Tempo Sync** is deactivated, the modulation speed can be set freely with the **Rate** dial.

Sync
Activates/Deactivates tempo sync.

Depth

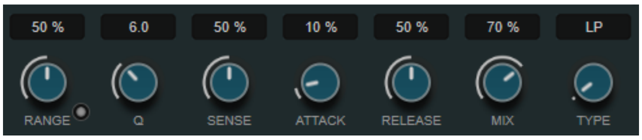
Sets the intensity of the pitch modulation.

Spatial

Adds a stereo effect to the modulation.

Envelope Filter

This is a classic envelope filter that allows for auto-wah effects.



Range

Determines the frequency range of the filter. **Sweep Downwards** reverses the filter sweep.

Q-Factor

Sets the intensity of the envelope filter effect.

Sensitivity

Determines how sensitively the effect reacts to the instrument level.

Attack

Determines how quickly an effect reacts to the input signal.

Release

Sets the gain of the release phase of the signal.

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.

Type

Sets the filter type. Low-pass (**LP**), high-pass (**HP**), band-pass (**BP**), and **Notch** filter are available.

Filter

This is a filter modulation effect. The frequency of the filter is modulated by an LFO or you can set it manually.

LFO/Manual buttons

Allows you to select **LFO** and **Manual** mode. In **LFO** mode, you can define the modulation rate or sync it to the project tempo. In **Manual** mode, you can set the frequency manually.

Freq

Sets the filter frequency. This parameter is only available in **Manual** mode.

LFO-Freq

Sets the filter frequency of the LFO modulation. If **Tempo Sync** is activated, this parameter sets the base note value for synchronizing the modulation to the tempo of the host application.

If **Tempo Sync** is deactivated, you can set the frequency freely.

This parameter is only available in **LFO** mode.

Tempo Sync

Activates/Deactivates tempo sync for the **LFO-Freq** parameter. This parameter is only available in **LFO** mode.

Q-Factor

Sets the resonance of the filter.

Type

Sets the filter type. Low-pass (**LP**), high-pass (**HP**), band-pass (**BP**), and **Notch** filter are available.

Low/High

Set the range of the filter frequency modulation.

Bit Crusher

This effect uses bit reduction to decimate and truncate the input audio signal to get a noisy, distorted sound.

Bits (0 to 24 bits)

Defines the bit resolution. A setting of 24 gives the highest audio quality, while a setting of 1 creates mostly noise.

Sample Div.

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.

Mode

Allows you to select one of the four operating modes. In each mode, the effect sounds differently. Modes **1** and **3** are nastier and noisier, while modes **2** and **4** are more subtle.

Overdrive

This effect creates a tube-like overdrive effect.

Drive

Adds harmonics to the output signal.

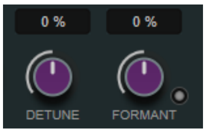
Tone

Works as a filter effect on the added harmonics.

Level
: Adjusts the output level.

Pitch Shifter

This is a pitch-shifting effect.



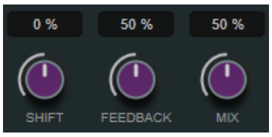
Detune
: Changes the pitch of the input signal in semitones.

Formant
: Changes the natural timbre of the input signal.

Formant Preservation
: Keeps the formants when changing the pitch with the **Detune** control.

Frequency Shifter

This effect shifts each frequency of the input signal by a fixed amount, which alters the harmonic relations. Adding feedback produces a sound similar to a phaser.



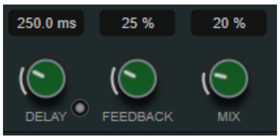
Shift
: Sets the amount of frequency shift.

Feedback
: Sets the amount of the signal that is sent from the output of the effect back to its input.

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

Delay

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



Delay
: If **Sync** is activated, this sets the base note value for the delay. If **Sync** is deactivated, the delay time can be set freely in milliseconds.

Sync
: Activates/Deactivates tempo sync.

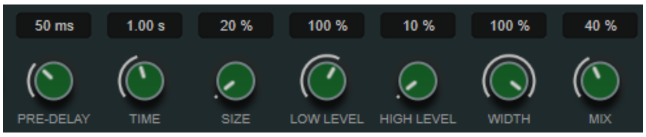
Feedback
: Sets the amount of the signal that is sent back into the delay input. The higher this value, the higher the number of repeats.

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.

Reverb

This is a versatile reverb for realistic room ambience and reverb effects.



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.

Time

Allows you to set the reverb time in seconds.

Size

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

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.

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.

AutoPan

This is an auto-pan effect that allows you to modulate the left/right stereo position.

Rate

Sets the auto-pan speed and shows the movement within the panorama. If **Tempo Sync** is deactivated, the speed is set in Hertz. If **Tempo Sync** is activated, you can set the speed in tempo values.

Sync

Activates/Deactivates tempo sync.

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.

Waveform Shape

Allows you to select presets for the modulation waveform.

- **Sine** creates a smooth sweep.
- **Triangle** creates a ramp, that is, a linear movement from full right to full left and back.
- **Square** creates an instant jump to full right, then to full left, and then back to center.

Gate

This effect silences audio signals below a set threshold. As soon as the signal level exceeds the threshold, the gate opens to let the signal through.

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.

Release

Sets the time after which the gate closes.

PingPongDelay

This is a stereo delay effect that alternates each delay repeat between the left and right channels. The delay line uses tempo-based or freely specified delay time settings.

NOTE

This plug-in works only on stereo tracks.

Lo Filter

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

Hi Filter

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

Delay

Sets the delay time in milliseconds.

Sync

Activates/Deactivates tempo sync.

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.

NOTE

This plug-in supports external side-chaining. You can use the side-chain input to control the delay from another signal source. If the side-chain signal exceeds the threshold, the delay repeats are silenced. If the signal drops below the threshold, the delay repeats reappear. For a description of how to set up side-chain routing, see the **Operation Manual**.

StereoDelay

StereoDelay has two independent delay lines which either use tempo-based or 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.

Sync
: Activates/Deactivates tempo sync for the corresponding delay.

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

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

Pan
: Sets the stereo position.

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

NOTE

This plug-in supports external side-chaining. You can use the side-chain input to control the delay from another signal source. If the side-chain signal exceeds the threshold, the delay repeats are silenced. If the signal drops below the threshold, the delay repeats reappear. For a description of how to set up side-chain routing, see the **Operation Manual**.

Distortion Plug-ins

AmpSimulator

AmpSimulator is a distortion effect that emulates the sound of various types of guitar amp and speaker cabinet combinations. A wide selection of amp and cabinet models is available.

Select Amplifier Model

This pop-up menu allows you to select an amplifier model. You can bypass this section by selecting **No Amp**.

Drive

Controls the amount of amp overdrive.

Bass

Tone control for the low frequencies.

Mid

Tone control for the mid frequencies.

Treble

Tone control for the high frequencies.

Presence

Boosts or dampens the higher frequencies.

Volume

Controls the overall output level.

Select Cabinet Model

This pop-up menu allows you to select a speaker cabinet model. You can bypass this section by selecting **No Speaker**.

Damping Low/High

These tone controls allow you to shape the sound of the selected speaker cabinet.

BitCrusher

If you are into lo-fi sound, **BitCrusher** is the effect for you. It offers the possibility of decimating and truncating the input audio signal by bit reduction, to get a noisy, distorted sound. For example, you can make a 24-bit audio signal sound like an 8 or 4-bit signal, or even render it completely garbled and unrecognizable.

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.

Mix

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

Sample Divider

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.

Depth (0 to 24 bits)

Defines the bit resolution. A setting of 24 gives the highest audio quality, while a setting of 1 creates mostly noise.

Output

Sets the output level.

DaTube

DaTube emulates the characteristic warm, lush sound of a tube amplifier.

Mix

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

Drive

Sets the pre-gain of the amplifier. Use high values if you want an overdriven sound just on the verge of distortion.

Output

Sets the output level.

Distortion

Distortion adds crunch to your tracks.

Boost

Increases the distortion amount.

Oversampling

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

NOTE

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

Mix

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

Tone

Changes the tonal characteristic of the output signal.

Feedback

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

Spatial

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

Output

Sets the output level.

Distroyer

Distroyer adds harmonics to the spectrum and allows for distortion effects from smooth overdrive to extreme clipping.

The following parameters influence only the wet signal:

Lo Filter

Changes the cuttoff frequency of the low-pass filter that is applied to the wet signal before it gets distorted.

Hi Filter

Changes the cuttoff frequency of the high-pass filter that is applied to the already distorted wet signal.

Offset

Modifies the symmetry of the distortion effect by changing the operation point of the characteristic.

Drive

Changes the characteristic of the distortion effect. Lower values lead to a smooth overdrive-like effect. Higher values change the shape of the signal towards a rectangle, leading to extreme distortion.

Oversampling

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

NOTE

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

Spatial

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

DC Filter

Removes DC offset that occurs when using high **Offset** values.

The following parameters influence both the dry and the wet signal:

Boost

Increases the distortion amount.

Mix

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

Shelf Freq

Sets the frequency of the high shelving filter.

Shelf Gain

Sets the gain of the high shelving filter.

Tone

Sets the frequency of the output low-pass filter.

Output

Sets the output level.

Grungelizer

Grungelizer adds noise and static to your recordings – like listening to a radio with bad reception, or a worn and scratched vinyl record.

Noise

Sets the amount of added static noise.

Crackle

Adds crackle to create that old vinyl record sound. The speed switch sets the virtual speed of the record in RPM (revolutions per minute).

Distort

Adds distortion.

EQ

Cuts the low frequencies, and creates a hollow, lo-fi sound.

AC

Emulates a constant, low AC hum. The frequency switch sets the virtual frequency of the AC current (50 or 60 Hz), and thus the pitch of the AC hum.

Mix

Sets the amount of overall effect.

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 signal and the wet signal.

Output (-24 to 24 dB)

Sets the output level.

Band Settings

Mute Band

Mutes/Unmutes a frequency band.

Bypass Band

Bypasses a frequency band.

Solo Band

Solos the corresponding 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

If tempo sync is activated, this is where you specify the base note value for tempo-syncing the effect (1/1 to 1/32, straight, triplet, or dotted).

If tempo sync is deactivated, the delay time can be set freely with the **Time** knob.

Sync

Activates/Deactivates tempo sync for the corresponding delay.

Duck

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

Mix

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

FBK

Determines the number of repeats for each delay.

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.

Slider

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 signal and the wet signal.

SoftClipper

SoftClipper adds soft overdrive, with independent control over the second and third harmonic.

Input (-12 to 24 dB)

Sets the pre-gain. Use high values if you want an overdriven sound just on the verge of distortion.

Mix

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

Output

Sets the output level.

Second

Controls the second harmonic.

Third

Controls the third harmonic.

VST Amp Rack

VST Amp Rack is a powerful guitar amp simulator. It offers a choice of amplifiers and speaker cabinets that can be combined with stomp box effects.

At the top of the plug-in panel, there are six buttons, arranged according to the position of the corresponding elements in the signal chain. These buttons open different pages in the display section of the plug-in panel: **Pre-Effects**, **Amplifiers**, **Cabinets**, **Post-Effects**, **Microphone Position**, **Master**, and **Configuration**.

Below the display section, the selected amplifier is shown. The color and texture of the area below the amplifier indicate the selected cabinet.

Pre/Post-Effects

On the **Pre-Effects** and **Post-Effects** pages, you can select up to six common guitar effects. On both pages, the same effects are available, the only difference being the position in the signal chain (before and after the amplifier). On each page, every effect can be used once.

Each effect features an **On/Off** button known from stompbox effects, as well as individual parameters.

Wah Wah

Pedal – Controls the filter frequency sweep.

Volume

Pedal – Controls the level of the signal passing through the effect.

Compressor

Intensity – Sets the amount by which an input signal is being compressed.

Limiter

Threshold – Determines the maximum output level. Signal levels above the set threshold are cut off.

Release – Sets the time after which the gain returns to the original level.

Maximizer

Amount – Determines the loudness of the signal.

Chorus

Rate – Allows you to set the sweep rate. This parameter can be synchronized to the project tempo.

Width – Determines the depth of the chorus effect. Higher settings produce a more pronounced effect.

Phaser

Rate – Allows you to set the sweep rate. This parameter can be synchronized to the project tempo.

Width – Determines the width of the modulation effect between higher and lower frequencies.

Flanger

Rate – Allows you to set the sweep rate. This parameter can be synchronized to the project tempo.

Feedback – Determines the character of the flanger effect. Higher settings produce a more metallic sounding sweep.

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

Tremolo

Rate – Allows you to set the modulation speed. This parameter can be synchronized to the project tempo.

Depth – Governs the depth of the amplitude modulation.

Octaver

Direct – Adjusts the mix of the original signal and the generated voices. A value of 0 means 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 signal that is generated one octave below the original pitch. A setting of 0 means that the voice is muted.

Octave 2 – Adjusts the level of the signal that is generated two octaves below the original pitch. A setting of 0 means that the voice is muted.

Delay

Delay – Sets the delay time in milliseconds. This parameter can be synchronized to the project tempo.

Feedback – Sets the number of repeats for the delay.

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

Tape Delay

Delay – Tape Delay creates a delay effect known from tape machines. The Delay parameter sets the delay time in milliseconds. This parameter can be synchronized to the project tempo.

Feedback – Sets the number of repeats for the delay.

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

Tape Ducking Delay

Delay – Tape Ducking Delay creates a delay effect known from tape machines with a ducking parameter. The Delay parameter sets the delay time in milliseconds. This parameter can be synchronized to the project tempo.

Feedback – Sets the number of repeats for the delay.

Duck – Works like an automatic mix parameter. If the level of the input signal is high, the portion of the effect signal is lowered, or ducked (low internal mix value). If the level of the input signal is low, the portion of the effect signal is raised (high internal mix value). This way the delayed signal stays rather dry during loud or intensely played passages.

Overdrive

Drive – Overdrive creates a tube-like overdrive effect. The higher this value, the more harmonics are added to the output signal of this effect.

Tone – Works as a filter effect on the added harmonics.

Level – Adjusts the output level.

Fuzz

Boost – Fuzz creates a rather harsh distortion effect. The higher this value, the more distortion is created.

Tone – Works as a filter effect on the added harmonics.

Level – Adjusts the output level.

Gate

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.

Release – Sets the time after which the gate closes.

Equalizer

Low – Changes the level of the low-frequency portion of the incoming signal.

Middle – Changes the level of the mid-frequency portion of the incoming signal.

High – Changes the level of the high-frequency portion of the incoming signal.

Reverb

Type – A convolution-based reverb effect. This parameter allows you to switch between different reverb types (**Studio**, **Hall**, **Plate**, and **Room**).

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

Sync Mode

Some parameters can be synchronized to the tempo of the host application.

The names of these parameters are underlined. Click a knob to activate or deactivate tempo sync. An LED at the top right of the knob indicates that sync mode is active. You can then select a base note value for tempo syncing from the pop-up menu above the control.

Using Effects

- To insert a new effect, click the **+** button that appears if you point the mouse at an empty plug-in slot or at one of the arrows before or after a used effect slot.
- To remove an effect from an effect slot, click the effect name and select **None** from the pop-up menu.
- To change the order of the effects in the chain, click on an effect and drag it to another position.
- To activate or deactivate an effect, click the pedal-like button below the effect name. If an effect is active, the LED next to the button is lit.

NOTE

- Pre-effects and post-effects can be mono or stereo, depending on the track configuration.
- Using quick controls you can set up an external MIDI device, such as a foot controller, to control the **VST Amp Rack** effects. For more information about quick controls, see the **Operation Manual**.

Amplifiers

The amps available on the **Amplifiers** page are modeled on real-life amplifiers. Each amp features settings typical for guitar recording, such as gain, equalizers, and master volume. The sound-related parameters Bass, Middle, Treble, and Presence have a significant impact on the overall character and sound of the corresponding amp.

Plexi
Classic British rock tone; extremely transparent sound, very responsive.

Plexi Lead
British rock tone of the 70s and 80s.

Diamond
The cutting edge hard rock and metal sounds of the 90s.

Blackface
Classic American clean tone.

Tweed
Clean and crunchy tones; originally developed as a bass amp.

Deluxe
American crunch sound coming from a rather small amp with a big tone.

British Custom
Produces the sparkling clean or harmonically distorted rhythm sounds of the 60s.

The different amps keep their settings if you switch models. However, if you want to use the same settings after reloading the plug-in, you need to set up a preset.

Selecting and Deactivating Amplifiers

To switch amps on the Amplifiers page, click the model that you want to use. Select **No Amplifier** if you only want to use the cabinets and effects.

Cabinets

The cabinets available on the **Cabinets** page simulate real-life combo boxes or speakers. For each amp, a corresponding cabinet type is available, but you can also combine different amps and cabinets.

Selecting and Deactivating Cabinets

- To switch cabinets on the Cabinets page, click the model that you want to use. Select **No Cabinet** if you only want to use the amps and effects.
- If you select **Link Amplifier & Cabinet Choice**, the plug-in automatically selects the cabinet corresponding to the selected amp model.

Microphones

On the **Microphones** page, you can choose between different microphone positions. These positions result from two different angles (center and edge) and three different distances from the speaker, as well as an additional center position at an even greater distance from the speaker.

You can choose between two microphone types: a large-diaphragm condenser microphone and a dynamic microphone. You can crossfade between the characteristics of the two microphones.

- To select one of the microphone types or blend between the two types, turn the **Mix** control between the two microphones.

Placing the Microphone

- To select a microphone position, click the corresponding ball in the graphic. The selected position is marked in red.

Master

Use the **Master** page to fine-tune the sound.

Input/Output Level Meters

The input and output level meters on the left and the right of the Master section show the signal level of your audio. The rectangle on the input meter indicates the optimum incoming level range. In compact view, the input and output levels are indicated by two LEDs at the top left and right.

Using the Master Controls

- To activate/deactivate the equalizer, click the pedal-like **On/Off** button. If the equalizer is active, the LED next to the button is lit.
- To activate/deactivate an equalizer band, click the corresponding **Gain** knob. If a band is active, the LED to the left of the **Gain** knob is lit.
- To tune your guitar strings, click the pedal-like **On/Off** button to activate the Tuner and play a string. If the correct pitch is displayed and the row of LEDs below the digital display is green, the string is tuned correctly.

  If the pitch is too low, red LEDs are lit on the left. If the pitch is too high, red LEDs are lit on the right. The more LEDs are lit, the lower/higher is the pitch.
- To mute the output signal of the plug-in, click the pedal-like **Master** button. If the output is muted, the LED is not lit. Use this to tune your guitar in silence, for example.
- To change the volume of the output signal, use the **Level** control on the Master page.

Configuration

On the **Configuration** page, you can specify whether you want to use **VST Amp Rack** in stereo or in mono mode.

- To process the pre-effects, the amplifier, and the cabinets in full stereo mode, make sure that the plug-in is inserted on a stereo track, and activate the **Stereo** button.
- To use the effect in mono-mode, make sure that the plug-in is inserted on a mono track, and activate the **Mono** button.

NOTE

In stereo mode, the effect requires more processing power.

View Settings

You can choose between 2 views: the default view and a compact view, which takes up less screen space.

In the default view, you can use the buttons at the top of the plug-in panel to open the corresponding page in the display section above the amp controls. You can horizontally resize the plug-in panel by clicking and dragging the edges or corners.

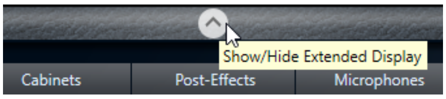
In the compact view, the display section is hidden from view. You can change the amp settings and switch amps or cabinets using the mouse wheel.

Using the Smart Controls

Smart controls become visible on the plug-in frame when you move the mouse pointer over on the plug-in panel.

Switching between Default and Compact View

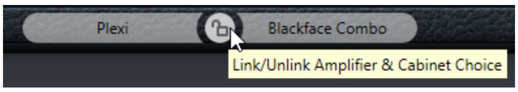
- To toggle between the different views, click the down/up arrow button (Show/Hide Extended Display) at the top center of the plug-in frame.



Changing the Amplifier and Cabinet Selection in the Compact View

In the compact view, a smart control on the lower border of the plug-in frame allows you to select different amplifier and cabinet models.

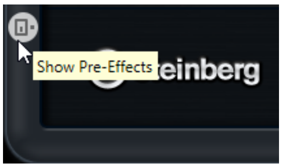
- To select a different amplifier or cabinet, click the name and select a different model from the pop-up menu.
- To lock the amplifier and cabinet combination, activate the **Link/Unlink Amplifier & Cabinet Choice** button. If you now select another amp model, the cabinet selection follows. However, if you select a different cabinet model, the lock is deactivated.



Previewing Effect Settings

In both views, you can show a preview of the pre- and post-effects that you selected on the corresponding pages:

- Click and hold the **Show Pre-Effects** or **Show Post-Effects** button at the bottom left or right of the plug-in frame.



VST Bass Amp

VST Bass Amp is a bass amp simulator. It offers a choice of amplifiers and speaker cabinets that can be combined with stomp box effects.

At the top of the plug-in panel, the following buttons open different pages in the display section of the plug-in panel: **Pre-Effects**, **Amplifiers**, **Cabinets**, **Post-Effects**, **Microphones**, **Configuration**, and **Master**.

These buttons are arranged according to the position of the corresponding elements in the signal chain.

Below the display section, the selected bass amplifier is displayed. The color and texture of the area below the bass amp indicate the selected cabinet.

Pre/Post-Effects

On the **Pre-Effects** and **Post-Effects** pages, you can select up to six common bass effects. On both pages, the same effects are available, the only difference being the position in the signal chain (before or after the bass amplifier). On each page, every effect can be used once.

Each effect features an **On/Off** button known from stompbox effects, as well as individual parameters.

Wah Wah

Pedal – Controls the filter frequency sweep.

Envelope Filter

Range – Determines the frequency range.

Q-Factor – Sets the intensity of the envelope filter effect.

Sensitivity – Determines how sensitive the effect reacts to the instrument level.

Attack – Determines how quickly an effect reacts to the input signal.

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

Type – Sets the filter type.

Release – Determines how quickly the effect fades after the input signal stops.

Volume

Pedal – Controls the level of the signal passing through the effect.

Compressor

Intensity – Sets the amount by which an input signal is being compressed.

Compressor MB

Lo Intensity – Sets the compressor effect in the low frequency band. Activate/deactivate **Auto Makeup Mode** by clicking the LED at the top right of the knob.

Hi Intensity – Sets the compressor effect in the high frequency band. Activate/deactivate **Auto Makeup Mode** by clicking the LED at the top right of the knob.

Crossover – Determines the crossover frequency between the low frequency band and the high frequency band.

Output – Sets the output level.

Limiter

Threshold – Determines the maximum output level. Signal levels above the set threshold are cut off.

Release – Sets the time after which the gain returns to the original level.

Maximizer

Amount – Determines the loudness of the signal.

Chorus

Rate – Allows you to set the sweep rate. This parameter can be synchronized to the project tempo.

Width – Determines the depth of the chorus effect. Higher settings produce a more pronounced effect.

Tone – Allows you to attenuate low frequencies.

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

Phaser

Rate – Allows you to set the sweep rate. This parameter can be synchronized to the project tempo.

Width – Determines the width of the modulation effect between higher and lower frequencies.

Tone – Allows you to attenuate the low frequencies.

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

Flanger

Rate – Allows you to set the sweep rate. This parameter can be synchronized to the project tempo.

Feedback – Determines the character of the flanger effect. Higher settings produce a more metallic sounding sweep.

Tone – Allows you to attenuate the low frequencies.

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

DI Driver

Level – Sets the output level.

Blend – Blends between normal and tube emulation circuitry. With **Blend** at 0, **Drive** and **Presence** are not active.

Bass – Boosts or attenuates low frequencies.

Treble – Boosts or attenuates high frequencies.

Presence – Boosts or attenuates upper harmonics and attacks.

Drive – Sets gain and overdrive.

Enhancer

Enhance – Simulates the classic enhancer effect.

Tone – Allows you to attenuate low frequencies.

Octaver

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

Tone – Changes the sound character of the generated signal.

Delay

Delay – Sets the delay time in milliseconds. This parameter can be synchronized to the project tempo.

Feedback – The higher this setting, the more delay repeats are created.

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

Tape Ducking Delay

Delay – The **Delay** parameter sets the delay time in milliseconds.

Feedback – The higher this setting, the more delay repeats are created.

Duck – Works like an automatic mix parameter. If the level of the input signal is high, the portion of the effect signal is lowered, or ducked (low internal mix value). If the level of the input signal is low, the portion of the effect signal is raised (high internal mix value). This way the delayed signal stays rather dry during loud or intensely played passages.

Tone – Allows you to attenuate the low frequencies.

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

Overdrive

Drive – Overdrive creates a tube-like overdrive effect. The higher this value, the more harmonics are added to the output signal of this effect.

Tone – Works as a filter effect on the added harmonics.

Level – Adjusts the output level.

Magneto II

Drive – Controls the amount of tape saturation.

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

HF-Adjust – Sets the amount of high frequency content of the saturated signal.

Gate

Threshold – Determines the level at which the gate is activated. Signal levels above the set threshold open the gate and signal levels below the set threshold close the gate.

Release – Sets the time after which the gate closes.

Equalizer

Low – Changes the level of the low-frequency portion of the incoming signal.

Middle – Changes the level of the mid-frequency portion of the incoming signal.

High – Changes the level of the high-frequency portion of the incoming signal.

Graphical EQ

Display – Consists of 8 sliders that set the level of each frequency band. Allows you to draw response curves by clicking and dragging with the mouse.

Reset Sliders – At the lower right of the Display. Flattens all values to 0 dB.

Output Slider – Allows you to control the frequency response.

Reverb

Type – A convolution-based reverb effect. This parameter allows you to switch between the reverb types **Studio**, **Hall**, **Plate**, and **Room**.

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

Sync

Some parameters can be synchronized to the tempo of the host application.

The names of these parameters are underlined. Click a knob to activate or deactivate tempo sync. An LED at the top right of the knob indicates that sync mode is active. You can then select a base note value for tempo syncing from the pop-up menu above the control.

Using Effects

- To insert a new effect, click the **+** button that appears if you point the mouse at an empty plug-in slot or at one of the arrows before or after a used effect slot.
- To remove an effect from an effect slot, click the effect name and select **None** from the pop-up menu.
- To change the order of the effects in the chain, click on an effect and drag it to another position.
- To activate or deactivate an effect, click the pedal-like button below the effect name. If an effect is active, the LED next to the button is lit.

NOTE

- Pre-effects and post-effects can be mono or stereo, depending on the track configuration.
- Using quick controls you can set up an external MIDI device, such as a foot controller, to control the **VST Bass Amp** effects. For more information about quick controls, see the **Operation Manual**.

Amplifiers

The amps available on the **Amplifiers** page are modeled on real-life amplifiers. Each amp features settings typical for bass recording, such as gain, equalizers, and master volume. The sound-related parameters bass, low mid, high mid, and treble have a significant impact on the overall character and sound of the corresponding amp. Shape 1 and Shape 2 offer predefined tone shaping.

ValveAmp300

A famous tube amplifier from the 70s, useful for rock playing styles.

Greyhound

An amplifier, well known for its typical growl, useful for several playing styles.

GreenT

A classic amplifier from the 80s, useful for funk and rock playing styles.

Paradise

An amplifier from the 90s, with a hifi-like clear tone, that makes it applicable for several styles.

Tweed

A classic vintage amplifier from the 50s, with a characterful and bright tone. Originally created for bassists, also used by many guitar players.

iTech

A modern amplifier, with a universal sound.

The different amps keep their settings if you switch models, but amp settings are lost when closing **VST Bass AMP**. If you want to use the same settings after reloading the plug-in, you need to set up a preset.

Selecting and Deactivating Amplifiers

To switch amps on the **Amplifiers** page, click the model that you want to use. Select **No Amplifier** if you only want to use the cabinets and effects.

NOTE

To scroll through amplifiers, use the mouse wheel when hovering over the amplifier panel.

Cabinets

The cabinets available on the **Cabinets** page simulate real-life combo boxes or speakers. For each amp, a corresponding cabinet type is available, but you can also combine different amps and cabinets.

The following cabinets are available:

4x10"

10" speakers provide a punchy clear sound that is suitable for “Slap” bass and regular playing styles.

10" speakers have a cleaner sound and more punch than 15" speakers.

8x10"

Compared to 4x10", double the amount of speakers.

4x12"

12" speakers provide a mellow and full sound, making them a good choice between 10" and 15" speakers.

1x15"

15" speakers provide more low frequencies compared to the other cabinets. They are suitable for rock and vintage oriented styles.

Selecting and Deactivating Cabinets

- To switch cabinets on the **Cabinets** page, click the model that you want to use. Select **No Cabinet** if you only want to use the amps and effects.
- If you select **Link Amplifier & Cabinet Choice**, the plug-in automatically selects the cabinet corresponding to the selected amp model.

Microphones

On the **Microphones** page, you can choose between different microphone types.

57
Dynamic microphone with cardioid pickup pattern.

121
Ribbon microphone with figure-8 pattern.

409
Dynamic microphone with supercardioid pickup pattern.

421
Dynamic microphone with cardioid polar pattern.

545
Dynamic microphone with cardioid pattern that minimizes feedback.

5
Dynamic microphone with cardioid pickup pattern.

30
Reference and measurement microphone with omni directional polar pattern.

87
Condenser microphone with omni directional pattern.

You can choose between different microphone positions. These positions result from two different angles (on axis and off axis) and three different distances from the cabinet.

You can crossfade between the characteristics of the two microphones.

- To select one of the microphone types or blend between the two types, turn the **Mix** control between the two microphones.
- To select a microphone position, click the corresponding ball in front of the cabinet. The selected position is marked in red.
- To determine the ratio between **line** and **mic**, turn the **Mix** control on the left of the cabinet.

NOTE

To scroll through microphones, use the mouse wheel when hovering over a microphone.

Master

Use the **Master** page to fine-tune the sound.

Input/Output Level Meters
The input and output level meters on the left and the right of the Master section show the signal level of your audio. The rectangle on the input meter indicates the optimum incoming level range. In all other views, the input and output levels are indicated by two LEDs at the top left and right.

Using the Master Controls

- To activate/deactivate the equalizer, click the pedal-like **On/Off** button. If the equalizer is active, the LED next to the button is lit.
- To activate/deactivate an equalizer band, click the corresponding **Gain** knob. If a band is active, the LED to the left of the **Gain** knob is lit.

- To tune your guitar strings, click the pedal-like **On/Off** button to activate **Tuner** and play a string. If the correct pitch is displayed and the row of LEDs below the digital display is green, the string is tuned correctly.

  If the pitch is too low, red LEDs are lit on the left. If the pitch is too high, red LEDs are lit on the right. The more LEDs are lit, the lower/higher is the pitch.
- To mute the output signal of the plug-in, click the pedal-like **Master** button. If the output is muted, the LED is not lit. Use this to tune your guitar in silence, for example.
- To change the volume of the output signal, use the **Level** control on the **Master** page.
- NOTE

  Master EQ functions only when a cabinet is selected.

Configuration

On the **Configuration** page, you can specify whether you want to use **VST Bass Amp** in stereo or in mono mode.

- To process the pre-effects, the amplifier, and the cabinets in full stereo mode, make sure that the plug-in is inserted on a stereo track, and activate the **Stereo** button.
- To use the effect in mono-mode, make sure that the plug-in is inserted on a mono track, and activate the **Mono** button.

NOTE

In stereo mode, the effect requires more processing power. Use mono configuration on a stereo track to save processing power.

View Settings

You can choose between 2 views: the default view and a compact view, which takes up less screen space.

In the default view, you can use the buttons at the top of the plug-in panel to open the corresponding page in the display section above the amp controls. You can horizontally resize the plug-in panel by clicking and dragging the edges or corners.

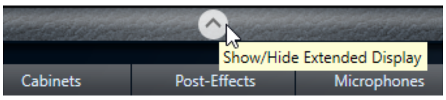
In the compact view, the display section is hidden from view. You can change the amp settings and switch amps or cabinets using the mouse wheel.

Using the Smart Controls

Smart controls become visible on the plug-in frame when you move the mouse pointer over the plug-in panel.

Switching between Default and Compact View

- To toggle between the different views, click the down/up arrow button (**Show/Hide Extended Display**) at the top center of the plug-in frame.

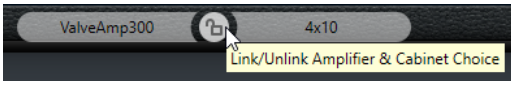


Changing the Amplifier and Cabinet Selection in the Compact View

In the compact view, a smart control on the lower border of the plug-in frame allows you to select different amplifier and cabinet models.

- To select a different amplifier or cabinet, click the name and select a different model from the pop-up menu.

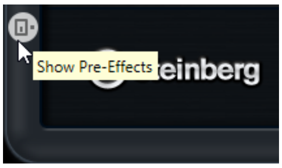
- To lock the amplifier and cabinet combination, activate the **Link/Unlink Amplifier & Cabinet Choice** button. If you now select another amp model, the cabinet selection follows. However, if you select a different cabinet model, the lock is deactivated.



Previewing Effect Settings

In both views, you can show a preview of the pre- and post-effects that you selected on the corresponding pages:

- Click and hold the **Show Pre-Effects** or **Show Post-Effects** button at the bottom left or right of the plug-in frame.



RELATED LINKS
Tuner on page 184

Dynamics Plug-ins

Brickwall Limiter

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

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

Threshold (-20 to 0 dB)

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

Release (3 to 1000 ms or Auto mode)

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

Link

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

Detect Intersample Clipping

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

NOTE

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

NOTE

This plug-in supports external side-chaining. For a description of how to set up side-chain routing, see the **Operation Manual**.

Compressor

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

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

Threshold (-60 to 0 dB)

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

Ratio

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

Soft Knee

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

High Ratio

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

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

Compensates for output gain loss caused by compression. If **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.

NOTE

This plug-in supports external side-chaining. For a description of how to set up side-chain routing, see the **Operation Manual**.

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.

NOTE

This plug-in supports external side-chaining. For a description of how to set up side-chain routing, see the **Operation Manual**.

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

Attack (-20 to 20 dB)

Sets the gain of the attack phase of the signal.

Length (5 to 200 ms)

Sets the length of the attack phase.

Release

Sets the gain of the release phase of the signal.

Output

Sets the output level.

NOTE

This plug-in supports external side-chaining. You can use the side-chain input to control the effect from another signal source. If the side-chain signal exceeds the threshold, the effect is triggered. For a description of how to set up side-chain routing, see the **Operation Manual**.

Expander

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

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

Threshold

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

Ratio

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

Soft Knee

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

Fall (0.1 to 100 ms)

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

Hold (0 to 2000 ms)

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

Rise (10 to 1000 ms or Auto mode)

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

Analysis (Pure Peak to Pure RMS)

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

Live

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

NOTE

This plug-in supports external side-chaining. You can use the side-chain input to control the expansion from another signal source. If the side-chain signal exceeds the threshold, the expansion is triggered. For a description of how to set up side-chain routing, see the **Operation Manual**.

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

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.

NOTE

This plug-in supports external side-chaining. You can use the side-chain input to control the gate from another signal source. If the side-chain signal exceeds the threshold, the gate opens. For a description of how to set up side-chain routing, see the **Operation Manual**.

Limiter

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

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

Input (-24 to 24 dB)

Sets the input gain.

Release (0.1 to 1000 ms or Auto mode)

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

Output

Sets the maximum output level.

Maximizer

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

Classic

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

Modern

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

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

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

Optimize

Determines the loudness of the signal.

Mix

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

Output

Sets the maximum output level.

Soft Clip

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

MIDI Gate

This plug-in gates audio signals. The gate is triggered by MIDI notes.

Gating, in its fundamental form, silences audio signals below a set threshold. If a signal rises above the set level, the gate opens to let the signal through. Signals below the set level are silenced. **MIDI Gate**, however, is not triggered by threshold levels, but MIDI notes. Therefore, it needs both audio and MIDI data to function.

Attack (0 to 500 ms)

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

Hold

Determines how long the gate remains open after a note-on or note-off message. The **Hold Mode** settings are taken into account.

Release (0 to 3000 ms)

Sets the time after which the gate closes after the set **Hold** time.

Note To Attack

Determines to which extent the velocity values of the MIDI notes affect the attack time. The higher the value, the more the attack time increases with high note velocities. Negative values result in shorter attack times with high velocities. If you do not want to use this parameter, set it to 0.

Note To Release

Determines to which extent the velocity values of the MIDI notes affect the release time. The higher the value, the more the release time increases. If you do not want to use this parameter, set it to 0.

Velocity To VCA

Controls to which extent the velocity values of the MIDI notes determine the output volume. At a value of **127**, the volume is controlled entirely by the velocity values, and at a value of **0**, the velocities have no effect on the volume.

Hold Mode

Sets the **Hold Mode**.

- In **Note On** mode, the gate only remains open for the time set with the **Hold** and **Release** parameters, regardless of the length of the MIDI note that triggered the gate.
- In **Note Off** mode, the gate remains open for as long as the MIDI note plays. The **Hold** and **Release** parameters are applied once a note-off signal has been received.

Setting Up MIDI Gate

To use **MIDI Gate** for your audio, you need an audio track and a MIDI track.

PROCEDURE

1. Select the audio track to which you want to apply **MIDI Gate**.
   This can be recorded or realtime audio material from any audio track.
2. Select **MIDI Gate** as an insert effect for the audio track.
3. Select a MIDI track to control the **MIDI Gate** effect.
   You can either play notes on a connected MIDI keyboard or use recorded MIDI notes.
4. Open the **Output Routing** pop-up menu for the MIDI track and select **MIDI Gate**.

Applying MIDI Gate

PREREQUISITE

Set up the plug-in properly.

How to apply **MIDI Gate** depends on whether you are using realtime or recorded MIDI. We assume for the purposes of this manual that you are using recorded audio and play the MIDI in realtime.

PROCEDURE

1. If you use realtime MIDI to trigger the plug-in, make sure the MIDI track is selected.
2. Start playback.
3. If you use realtime MIDI, play a few notes on your keyboard.

RESULT

The MIDI notes trigger the Gate. The plug-in gates the audio signals.

RELATED LINKS

Setting Up MIDI Gate on page 81

MultibandCompressor

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

NOTE

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

Frequency Band Editor

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

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

Live

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

Bypass Band

Bypasses a frequency band.

Solo Band

Solos the corresponding frequency band.

Output (-24 to 24 dB)

Sets the output level.

Compressor Section

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

Threshold (-60 to 0 dB)

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

Ratio

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

Attack (0.1 to 100 ms)

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

Release (10 to 1000 ms or Auto mode)

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

Side-Chain

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

Frequency

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

Q-Factor

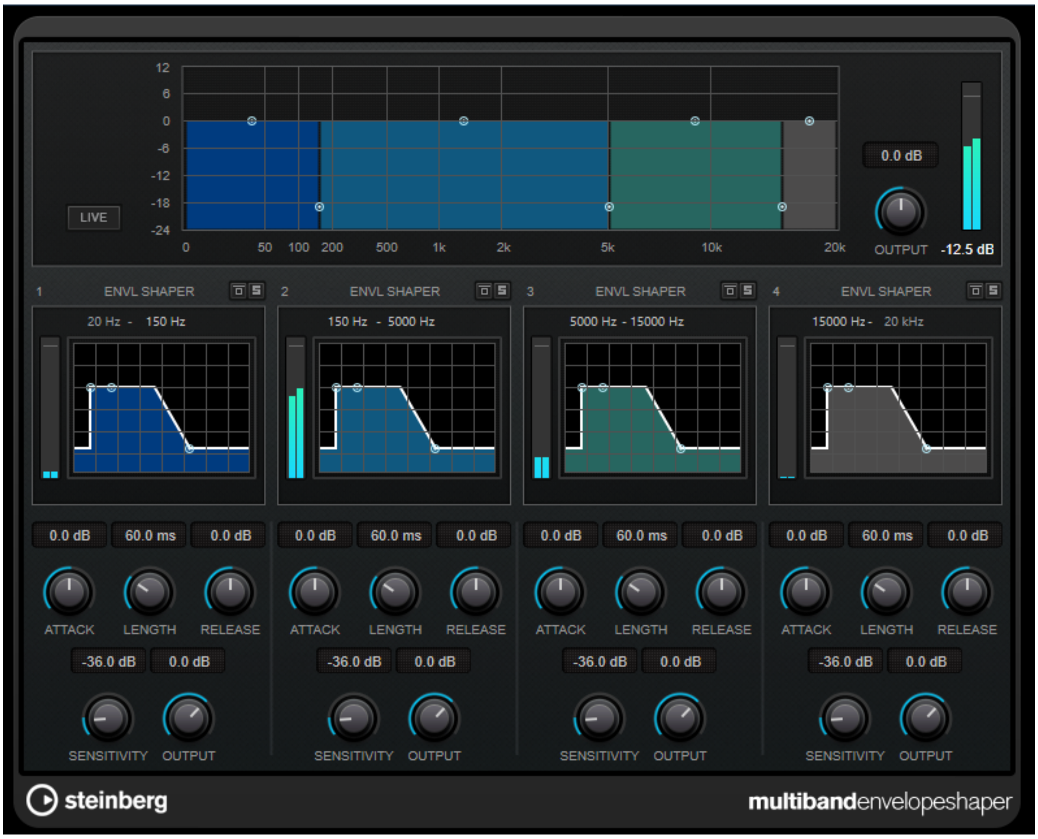
If **Side-Chain** is activated, this sets the resonance or width of the filter.

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.

Bypass Band

Bypasses a frequency band.

Solo Band

Solos the corresponding frequency band.

Output (-24 to 24 dB)

Sets the output level.

Shaper Section

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

Attack (-20 to 20 dB)

Sets the gain of the attack phase of the signal.

Length (5 to 200 ms)

Sets the length of the attack phase.

Release

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.

Bypass Band

Bypasses a frequency band.

Solo Band

Solos the corresponding frequency band.

Output (-24 to 24 dB)

Sets the output level.

Expander Section

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

Threshold

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

Ratio

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

Maximum Reduction

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

Fall (0.1 to 100 ms)

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

Hold (0 to 2000 ms)

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

Rise (10 to 1000 ms or Auto mode)

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

Output

Sets the output level.

Side-Chain Section

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.

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.

Monitor

Allows you to monitor the filtered signal.

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 3 frequency bands and use internal or external side-chaining sources 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 in 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 dry signal and 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 in the frequency display.

Side-Chain section

This section provides settings for internal and external 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 Input

This pop-up menu allows you to select the side-chain input for the corresponding band.

- **Internal** uses the input signal of the track.
- **Side-Chain 1 - Side-Chain 3** allow you to use the side-chain inputs of the plug-in.

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 (1.0 to 6.0 dB)

Controls the amount of tube saturation.

Output (-12 to 12 dB)

Sets the output gain.

Character

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

Attack (0.1 to 100 ms)

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

Release (10 to 1000 ms or Auto mode)

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

Mix

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

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.

NOTE

This plug-in supports external side-chaining. For a description of how to set up side-chain routing, see the **Operation Manual**.

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 dry signal and wet signal, preserving the transients of the input signal.

Output (-48 to 24 dB)

Sets the output gain.

NOTE

This plug-in supports external side-chaining. For a description of how to set up side-chain routing, see the **Operation Manual**.

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

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 (-60 to 0 dB)
Determines the level where the compressor kicks in. Only signal levels above the set threshold are processed.

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

Make-Up (0 to 24 dB or Auto mode)
Compensates for output gain loss caused by compression. If **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 Plug-ins

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.

DJ-EQ

DJ-EQ is an easy-to-use 3-band parametric equalizer that resembles the EQs found on typical DJ mixers. This plug-in is designed for quick sound fixes.

Graphical display

Allows you to set the amount of boost or attenuation for the low, mid, or high band by dragging.

- To set the low, mid, and high frequency gain, click and drag the corresponding band handle.
- To fine-adjust the gain, press **Shift** and drag.
- To set a parameter to zero, press **Ctrl/Cmd** and click it.

Low Frequency Gain/Mid Frequency Gain/High Frequency Gain

Set the amount of boost or attenuation for the low, mid, and high band.

Cut Low Frequencies/Cut Mid Frequencies/Cut High Frequencies

Cut the low, mid, and high band.

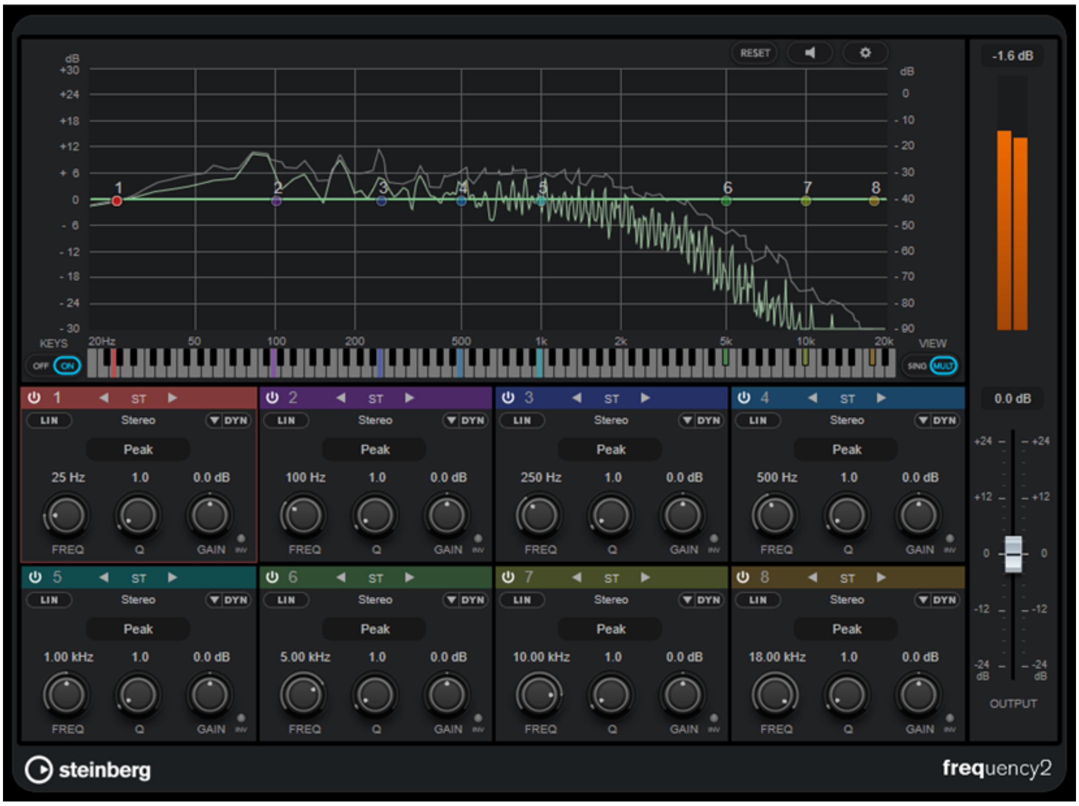
Output meter

Shows the level of the output signal.

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 (band-pass), or as cut filter (low-pass/high-pass).

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

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.

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.

NOTE

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

Output

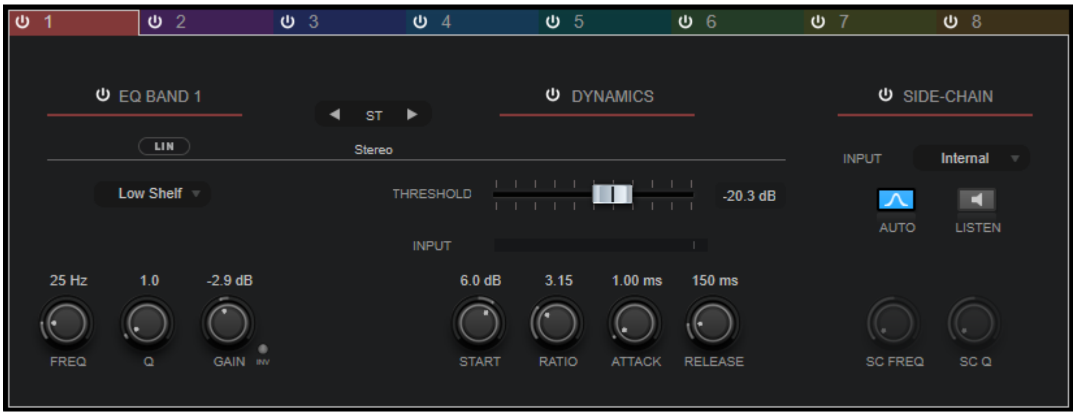
Adjusts the overall output level.

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.

NOTE

- 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, enter A5 -23 or C4 +49.

NOTE

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

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. For **Cut** filters, it adds a resonance.

NOTE

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

- This parameter is not available for **Cut 6** 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**, and **High Shelf**.

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**, and **High Shelf**.

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.

Input

Sets the side-chain input for the corresponding band. If **Internal** is selected, the input signal of the plug-in is used as side-chain source. If **Side-Chain 1** to **Side-Chain 8** is selected, the signal of tracks that are routed to the corresponding plug-in side-chain input is used.

NOTE

This setting is only available in single-band view.

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 filter for the corresponding band.

NOTE

This setting is only available in single-band view.

NOTE

This plug-in supports external side-chaining. For a description of how to set up side-chain routing, see the **Operation Manual**.

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 in the spectrum display.

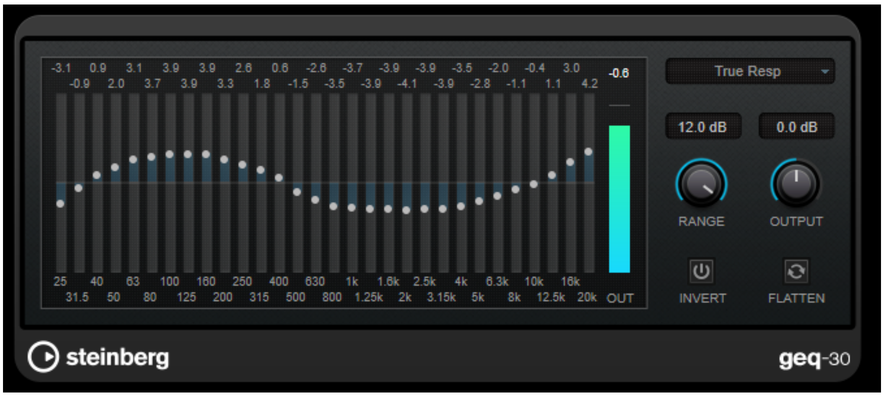
Filled

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

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

GEQ-10



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**/**GEQ-30**.

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

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

Output

Sets the overall gain of the equalizer.

Flatten

Resets all the frequency bands to 0 dB.

Range

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

Invert

Inverts the current response curve.

Mode pop-up menu

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

EQ Modes

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

True Response

Applies serial filters with an accurate frequency response.

Digital Standard

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

Classic

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

VariableQ

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

ConstQ asym

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

ConstQ sym

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

Resonant

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

StudioEQ

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

Main Layout

Reset

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

Show Input/Output Spectrum

Shows the spectrum before and after filtering.

Output

Adjusts the overall output level.

Auto Gain

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

Band Settings

Activate/Deactivate Band

Activates/Deactivates the corresponding band.

NOTE

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

Freq

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

NOTE

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

Inv

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

Q

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

NOTE

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

Gain

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

NOTE

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

Filter type

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

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

Filter Plug-ins

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.

MorphFilter

MorphFilter lets you mix low-pass, high-pass, band-pass, and band-reduction filter effects, allowing for creative morphings or mixtures between two filters.

Filter A buttons

Allow you to select the characteristic of the first filter.

- **Low Pass**

  Eliminates high-frequency signal components. Filter slopes of 6, 12, 18, and 24 dB per decade are available.

- **Band Pass**

  Allows signals falling within a certain frequency range to pass through. Filter slopes of 12 and 24 dB per decade are available.

Filter B buttons

Allow you to select the characteristic of the second filter.

- **High Pass**

  Eliminates low-frequency signal components. Filter slopes of 6, 12, 18, and 24 dB per decade are available.

- **Band Rejection**

  Lets all frequencies pass, except those in the stop band. Filter slopes of 12 and 24 dB per decade are available.

Resonance Factor

Sets the resonance value of both filters simultaneously.

Frequency

Sets the cutoff frequency of both filters simultaneously.

Graphical display

Visualizes the settings for all parameters. The handle allows you to adjust the **Morph Factor** and the **Frequency** parameters simultaneously.

Output meter

Shows the level of the output signal.

Morph Factor

Allows you to mix the output between both filters.

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 in 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.

StepFilter

StepFilter is a pattern-controlled multimode filter that can create rhythmic, pulsating filter effects. You can also trigger pattern steps individually via MIDI.

General Operation

StepFilter can produce two simultaneous 16-step patterns for the filter cutoff and resonance parameters, synchronized to the sequencer tempo.

The horizontal axis shows the pattern steps 1 to 16 from left to right, and the vertical axis determines the (relative) filter cutoff frequency and resonance settings. The higher up on the vertical axis a step value is entered, the higher the relative filter cutoff frequency or filter resonance setting.

By starting playback and editing the patterns for the cutoff and resonance parameters, you can hear how your filter patterns affect the sound source connected to **StepFilter**.

If **Sync** is deactivated, **StepFilter** allows you to trigger pattern steps individually via MIDI notes.

Setting Step Values

- To enter a step, click in the pattern grid windows.
- Individual step entries can be dragged up or down the vertical axis, or directly set by clicking in an empty grid box. By click-dragging left or right, consecutive step entries are set at the pointer position.
- Change the value for a step by dragging it up or down.
- Draw in a curve by clicking and dragging in the display.

Selecting New Patterns

- Patterns are saved with the project, and up to 8 different cutoff and resonance patterns can be saved internally. Both the cutoff and resonance settings are saved together in the 8 pattern slots.

- Use the **Pattern** slots to select a new pattern. New patterns are all set to the same step value by default.

StepFilter Parameters

Filter Type

Sets the filter type. A low-pass, a band-pass, and a high-pass filter are available.

Filter Mode

Allows you to choose between two filter modes. **Classic** mode is compatible with previous versions of this plug-in. **Modern** mode provides an additional **Hard Clip** parameter.

Mix

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

Base Cutoff

Sets the base filter cutoff frequency. Values set in the **Cutoff** grid are relative to the **Base Cutoff** value.

Base Resonance

Sets the base filter resonance. Values set in the **Resonance** grid are relative to the **Base Resonance** value.

NOTE

Very high **Base Resonance** settings can produce loud ringing effects at certain frequencies.

Rate

If **Sync** is activated, **Rate** allows you to specify the base note value for synchronizing the pattern playback to the tempo of the host application (1/1 to 1/32, straight, triplet, or dotted).

If **Sync** is deactivated, you can trigger pattern steps individually via MIDI.

Hard Clip

Adds more high frequency harmonics and distortion to the signal. This parameter is only available in **Modern** mode.

Glide

Applies glide between the pattern steps, causing values to change more smoothly.

Output

Sets the output level.

RELATED LINKS

Creating Variations for StepFilter Patterns on page 112
Controlling StepFilter via MIDI on page 113

Creating Variations for StepFilter Patterns

You can copy a pattern of the **StepFilter** to another pattern slot. This is useful for creating variations on a pattern of this plug-in.

PROCEDURE

1. Select the pattern that you want to copy.
2. Click the **Copy** button.
3. Select another pattern slot.

4. Click the **Paste** button.

RESULT

The pattern is copied to the new slot and can now be edited to create variations.

Controlling StepFilter via MIDI

StepFilter allows you to trigger steps individually via MIDI notes.

PREREQUISITE

- Your project contains a MIDI track that is routed to the MIDI input of a track that uses **StepFilter** as an insert plug-in.
- In **StepFilter**, **Sync** is deactivated.

PROCEDURE

- Do one of the following:
    - Use the note C0 to increase the step number by one.
    - Use the notes from C1 to D#2 to trigger the steps 1 to 16 directly.

ToneBooster

ToneBooster is a filter that allows you to raise the gain in a selected frequency range. It is particularly useful if it is inserted before **AmpSimulator** in the plug-in chain, greatly enhancing the tonal varieties available.

Gain

Adjusts the gain of the selected frequency range by up to 24 dB.

Tone

Sets the center filter frequency.

Width

Sets the resonance of the filter.

Mode selector

Sets the basic operational mode of the filter: **Peak** or **Band** Mode.

RELATED LINKS

AmpSimulator on page 46

WahWah

WahWah is a variable slope band-pass filter that can be auto-controlled by a side-chain signal or via MIDI modeling the well-known analog pedal effect.

You can independently specify the frequency, width, and gain for the **Low** and **High** Pedal positions. The crossover point between the Lo and Hi Pedal positions lies at 50.

NOTE

This plug-in supports external side-chaining. You can use the side-chain input to control the **Pedal** parameter from another signal source. The louder the signal, the more the filter frequency is raised so that the plug-in acts as an auto-wah effect. For a description of how to set up side-chain routing, see the **Operation Manual**.

WahWah Parameters

Pedal

Controls the filter frequency sweep.

Pedal Control (MIDI)

Allows you to choose the MIDI controller that controls the plug-in. Set this to **Automation** if you do not want to use MIDI realtime control.

Freq Low/Freq High

Set the frequency of the filter for the Lo and Hi pedal positions.

Width Low/Width High

Set the width (resonance) of the filter for the Lo and Hi pedal positions.

Gain Low/Gain High

Set the gain of the filter for the Lo and Hi pedal positions.

Filter Slope selector

Allows you to choose between two filter slope values: 6 dB or 12 dB.

MIDI Control

For realtime MIDI control of the **Pedal** parameter, MIDI must be directed to the **WahWah** plug-in.

If **WahWah** is used as an insert effect (for an audio track or an FX channel), it is available on the **Output Routing** pop-up menu for MIDI tracks.

If **WahWah** is selected on the **Output Routing** menu, MIDI data is directed to the plug-in from the selected track.

Mastering Plug-ins

UV22HR

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

8, 16, 20, 24 bit

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

Hi

Applies a normal dither gain.

Lo

Applies a lower level of dither noise.

Auto black

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

IMPORTANT

Dithering should always be applied post-fader on an output bus.

Modulation Plug-ins

AutoPan

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

NOTE

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

Waveform display

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

Factor

If **Sync** is activated, this parameter multiplies the sync rate by the selected factor. This allows you to create very slow movements in panorama.

Rate

Sets the auto-pan speed and shows the movement within the panorama. If **Sync** is deactivated, the speed is set in Hertz. If **Sync** is activated, you can set the speed in tempo values.

Sync

Activates/Deactivates tempo sync.

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.

NOTE

This plug-in supports external side-chaining. You can use the side-chain input to control the **Width** parameter from another signal source. For a description of how to set up side-chain routing, see the **Operation Manual**.

Smooth

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

Chopper

Chopper allows you to create a tremolo with or without an additional panning effect.

Waveform buttons

Allow you to select the modulation waveform.

Depth

Sets the intensity of the effect. This can also be set by clicking and dragging in the graphical display.

Sync

Activates/Deactivates tempo sync.

Speed

If tempo sync is activated, this is where you specify the base note value for tempo-syncing the effect (1/1 to 1/32, straight, triplet, or dotted).

If tempo sync is deactivated, the tremolo speed can be set freely with the **Speed** knob.

Mono

If this option is activated, **Chopper** acts as a tremolo effect only. If this option is deactivated, the modulation waveforms of the left and the right channel are phase-shifted, creating an additional panning 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.

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

If **Tempo Sync** is activated, **Rate** allows you to specify the base note value for synchronizing the modulation sweep to the tempo of the host application (1/1 to 1/32, straight, triplet, or dotted).

If **Tempo Sync** is deactivated, the sweep rate can be set freely with the **Rate** dial.

Sync

Activates/Deactivates tempo sync.

Waveform Shape

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

Lo Filter/Hi Filter

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

NOTE

This plug-in supports external side-chaining. You can use the side-chain input to control the modulation from another signal source. If the side-chain signal exceeds the threshold, the modulation is controlled by the side-chain signal's envelope. For a description of how to set up side-chain routing, see the **Operation Manual**.

Cloner

Cloner adds up to 4 detuned and delayed voices to the signal for rich modulation and chorus effects.

Graphical display

Shows the panning position of each voice in the stereo spectrum.

Voices

Sets the number of voices. Each voice has a **Detune** and a **Delay** slider.

Detune slider (1 to 4)

Sets the relative amount of detune for each voice. Positive and negative values can be set. If set to zero, no detune takes place for that voice.

Delay slider (1 to 4)

Sets the relative delay amount for each voice. If set to zero, no delay takes place for that voice.

Detune

Sets the overall amount of detune for all voices. If set to zero, no detune takes place regardless of the **Detune** slider settings.

Natural

Changes the pitch algorithm that is used for detune.

Humanize (Detune)

Sets the amount of detune variation if **Static Detune** is deactivated. With **Humanize**, the detune is continuously modulated for a more natural effect.

Static (Detune)

Activate this button to use a static amount of detune.

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

Spreads the voices across the stereo spectrum. Turn the control clockwise for a deeper stereo effect.

Output (-12 to 12 dB)

Sets the output gain.

Delay

Governs the overall depth of the delay for all voices. If set to zero, no delay takes place regardless of the **Delay** slider settings.

Humanize (Delay)

Controls the amount of delay variation if **Static Delay** is deactivated. With **Humanize**, the delay is continuously modulated for a more natural effect.

Static (Delay)

Activate this button to use a static delay amount.

Flanger

This is a classic flanger effect with added stereo enhancement.

Delay

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

Feedback

Determines the character of the flanger effect. Higher settings produce a more metallic-sounding sweep.

Mode

Allows you to toggle between **LFO** and **Manual** mode.

In **LFO** mode, you can define the sweep rate or sync it to the project tempo. In **Manual** mode, you can change the sweep manually.

Rate

If **Tempo Sync** is activated, **Rate** allows you to specify the base note value for synchronizing the flanger sweep to the tempo of the host application (1/1 to 1/32, straight, triplet, or dotted).

If **Tempo Sync** is deactivated, the sweep rate can be set freely with the **Rate** dial.

Sync

Activates/Deactivates tempo sync.

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.

Lo Range/Hi Range

Set the frequency boundaries for the flanger sweep.

Waveform Shape

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

Lo Filter/Hi Filter

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

NOTE

This plug-in supports external side-chaining. You can use the side-chain input to control the modulation from another signal source. If the side-chain signal exceeds the threshold, the modulation is controlled by the side-chain signal's envelope. For a description of how to set up side-chain routing, see the **Operation Manual**.

Metalizer

Metalizer feeds the audio signal through a variable frequency filter, with tempo sync or time modulation and feedback control.

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.

On/Off

Activates/Deactivates filter modulation. If it is deactivated, **Metalizer** works as a static filter.

Speed

If tempo sync is activated, this is where you specify the base note value for synchronizing the effect to the tempo of the host application (1/1 to 1/32, straight, triplet, or dotted).

If tempo sync is deactivated, the modulation speed can be set freely with the **Speed** knob.

Sync

Activates/Deactivates tempo sync.

Mono

Switches the output to mono.

Feedback

Determines the character of the metal effect. Higher settings produce a more metallic sound.

Sharpness

Sets the character of the filter effect. The higher the value, the narrower the affected frequency area, which produces a sharper sound and a more pronounced effect.

Tone

Sets the feedback frequency. The effect of this is more noticeable with high **Feedback** settings.

Output

Sets the output level.

Phaser

Phaser produces the well-known swooshing phasing effect with additional stereo enhancement.

Feedback

Determines the character of the phaser effect. Higher settings produce a more pronounced effect.

Width

Sets the intensity of the modulation effect between higher and lower frequencies.

Mode

Allows you to toggle between **LFO** and **Manual** mode.

In **LFO** mode, you can define the sweep rate or sync it to the project tempo. In **Manual** mode, you can change the sweep manually.

Rate

If **Tempo Sync** is activated, **Rate** allows you to specify the base note value for tempo-syncing the phaser sweep (1/1 to 1/32, straight, triplet, or dotted).

If **Tempo Sync** is deactivated, the sweep rate can be set freely with the **Rate** dial.

Sync

Activates/Deactivates tempo sync.

Spatial

If you are using multi-channel audio, the **Spatial** parameter creates a 3-dimensional impression by delaying modulation in each channel.

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/Hi Filter

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

NOTE

This plug-in supports external side-chaining. You can use the side-chain input to control the modulation from another signal source. If the side-chain signal exceeds the threshold, the modulation is controlled by the side-chain signal’s envelope. For a description of how to set up side-chain routing, see the **Operation Manual**.

RingModulator

RingModulator can produce complex, bell-like enharmonic sounds.

Ring modulators work by multiplying two audio signals. The ring-modulated output contains added frequencies generated by the sum of, and the difference between, the frequencies of the two signals.

RingModulator has a built-in oscillator that is multiplied with the input signal to produce the effect.

Oscillator

Waveform buttons

Allow you to select the oscillator waveform: square, sine, saw, or triangle.

LFO Amount

Controls how much the LFO affects the oscillator frequency.

Env. Amount

Controls how much the oscillator frequency is affected by the envelope that is triggered by the input signal. Left of center, a loud input signal decreases the oscillator pitch, whereas right of center the oscillator pitch increases if it is fed a loud input.

Frequency

Sets the oscillator frequency ± 2 octaves within the selected range.

Roll-Off

Attenuates high frequencies in the oscillator waveform to soften the overall sound. This is best used with harmonically rich waveforms, for example, square or saw.

Range

Determines the frequency range of the oscillator in Hz.

LFO

Waveform buttons

Allow you to select the LFO waveform: square, sine, saw, or triangle.

Speed

Sets the LFO speed.

Env. Amount

Controls how much the input signal level – via the envelope generator – affects the LFO speed. With negative values, a loud input signal slows down the LFO, whereas positive values speed it up at loud input signals.

Invert

Inverts the LFO waveform for the right channel of the oscillator, which produces a wider stereo effect for the modulation.

Envelope Generator

The envelope generator parameters control how the input signal is converted to envelope data, which can then be used to control oscillator pitch and LFO speed.

Attack

Controls how fast the envelope output level rises in response to a rising input signal.

Decay

Controls how fast the envelope output level falls in response to a falling input signal.

Lock R>L

If this button is activated, the L and R input signals are merged, and produce the same envelope output level for both oscillator channels. If the button is deactivated, each channel has its own envelope that affects the two channels of the oscillator independently.

Level Settings

Mix

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

Output

Sets the output level.

Rotary

This modulation effect simulates the classic effect of a rotating speaker.

A rotary speaker cabinet features speakers rotating at variable speeds to produce a swirling chorus effect, commonly used with organs.

Speed settings

Speed Mod Control (MIDI)

This pop-up menu allows you to select the MIDI controller that controls the rotary speed. If you do not want to use MIDI realtime control, set this to **Automation**. If you select **PitchBend** as MIDI controller, the speed changes with an up or down flick of the bender. If other MIDI controllers are used, the speed changes at MIDI value 64.

Speed selector (stop/slow/fast)

Allows you to control the speed of the rotary speaker.

Speed Mod

If the **Set Speed Change Mode** setting is set to the right, this knob allows you to modulate the rotary speed.

Set Speed Change Mode

If this is set to the left, the speed selector settings are taken into account. If this is set to the right, you can modulate the speed with the **Speed Mod** knob and/or with a MIDI controller that you can select on the **Speed Mod Control (MIDI)** pop-up menu.

Additional settings

Overdrive

Applies a soft overdrive or distortion.

Crossover

Sets the crossover frequency (200 to 3000 Hz) between the low and high frequency loudspeakers.

Horn

Slow

Allows for a fine adjustment of the high rotor **slow** speed.

Fast

Allows for a fine adjustment of the high rotor **fast** speed.

Accel.

Allows for a fine adjustment of the high rotor acceleration time.

Amp Mod

Controls the high rotor amplitude modulation.

Freq Mod

Controls the high rotor frequency modulation.

Bass

Slow

Allows for a fine adjustment of the low rotor **slow** speed.

Fast

Allows for a fine adjustment of the low rotor **fast** speed.

Accel.

Allows for a fine adjustment of the low rotor acceleration time.

Amp Mod

Adjusts the modulation depth of the amplitude.

Level

Adjusts the overall bass level.

Mics

Phase

Adjusts the phasing amount in the sound of the high rotor.

Angle

Sets the simulated microphone angle. A value of 0 ° corresponds to a mono miking setup with a single microphone in front of the speaker cabinet, 180 ° corresponds to a stereo miking setup with a microphone on each side of the cabinet.

Distance

Sets the simulated microphone distance from the speaker in inches.

Final Settings

Output

Sets the output level.

Mix

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

Directing MIDI to the Rotary

For realtime MIDI control of the **speed** parameter, MIDI must be directed to **Rotary**.

- If **Rotary** is used as insert effect (for an audio track or an FX channel), it is available on the **Output Routing** pop-up menu for MIDI tracks. If **Rotary** is selected on the **Output Routing** pop-up menu, MIDI is directed to the plug-in from the selected track.

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

If **Tempo Sync** is activated, **Rate** allows you to specify the base note value for synchronizing the modulation sweep to the tempo of the host application (1/1 to 1/32, straight, triplet, or dotted).

If **Tempo Sync** is deactivated, the sweep rate can be set freely with the **Rate** dial.

Sync

Activates/Deactivates tempo sync.

Waveform Shape

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

Lo Filter/Hi Filter

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

NOTE

This plug-in supports external side-chaining. You can use the side-chain input to control the modulation from another signal source. If the side-chain signal exceeds the threshold, the modulation is controlled by the side-chain signal’s envelope. For a description of how to set up side-chain routing, see the **Operation Manual**.

Tranceformer

Tranceformer is a ring modulator effect that modulates incoming audio by an internal, variable frequency oscillator, producing new harmonics. You can use a second oscillator to modulate the frequency of the first oscillator, in sync with the song tempo if needed.

Mix

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

Waveform buttons

Allow you to select a pitch modulation waveform.

Waveform display

Allows you to modify the **Pitch** and **Depth** parameters simultaneously by dragging.

Pitch

Sets the frequency of the modulating oscillator.

Activate/Deactivate Pitch Modulation

Activates/Deactivates the modulation of the pitch parameter.

Speed

If tempo sync is activated, this is where you specify the base note value for synchronizing the effect to the tempo of the host application (1/1 to 1/32, straight, triplet, or dotted).

If tempo sync is deactivated, the modulation speed can be set freely with the **Speed** knob.

Sync

Activates/Deactivates tempo sync.

Depth

Sets the intensity of the pitch modulation.

Mono

Switches the output to mono.

Output

Sets the output level.

Tremolo

Tremolo produces amplitude modulation.

Rate

If **Tempo Sync** is activated, **Rate** allows you to specify the base note value for tempo-syncing the effect (1/1 to 1/32, straight, triplet, or dotted).

If **Tempo Sync** is deactivated, the modulation speed can be set freely with the **Rate** dial.

Sync

Activates/Deactivates tempo sync.

Depth

Governs the depth of the amplitude modulation.

Spatial

Adds a stereo effect to the modulation.

Output

Sets the output level.

NOTE

This plug-in supports external side-chaining. You can use the side-chain input to control the modulation from another signal source. If the side-chain signal exceeds the threshold, the modulation is controlled by the side-chain signal's envelope. For a description of how to set up side-chain routing, see the **Operation Manual**.

Vibrato

Vibrato creates pitch modulation.

Depth

Sets the intensity of the pitch modulation.

Rate

If **Tempo Sync** is activated, **Rate** allows you to specify the base note value for tempo-syncing the effect (1/1 to 1/32, straight, triplet, or dotted).

If **Tempo Sync** is deactivated, the modulation speed can be set freely with the **Rate** dial.

Sync

Activates/Deactivates tempo sync.

Spatial

Adds a stereo effect to the modulation.

NOTE

This plug-in supports external side-chaining. You can use the side-chain input to control the modulation from another signal source. If the side-chain signal exceeds the threshold, the modulation is controlled by the side-chain signal's envelope. For a description of how to set up side-chain routing, see the **Operation Manual**.

Network Plug-ins

VST Connect CUE Mix

This plug-in is described in detail in the separate document **VST Connect SE**.

VST Connect SE

This plug-in is described in detail in the separate document **VST Connect SE**.

Other Plug-ins

LoopMash FX

LoopMash FX is a live performance effect offering DJ effects that can be controlled by a MIDI keyboard.

Quantize Note

Sets the note value on which the quantize grid of the effects is based.

Performance Controls

By clicking these buttons during playback, you can apply effects to your overall performance.

An effect is applied as long as you keep the button pressed.

NOTE

The effects can be automated. The automation of effect parameters is described in the **Operation Manual**.

Backspin

Simulates a turntable backspin.

Reverse

Plays the slice in reverse.

Tapestart

Simulates a tapestart, that is, speeds the slice up.

Scratch

Plays the slice as if scratched.

Slowdown

Applies a slowdown.

Tapestop 1

Simulates a tapestop, that is, slows the slice down, first lightly then abruptly.

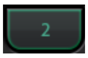
Tapestop 2

Simulates a tapestop, that is, slows the slice down, smoothly.

Stutter

Plays only the initial portion of a slice and repeats it 2, 3, 4, 6, or 8 times during one slice length.

Slur



Stretches the slice over 2 or 4 slice lengths.

Cycle

Sets up a short cycle over 4, 2, or 1 slice. This short cycle is always set up within the loop range that is set in the ruler. Setting up a cycle over 1 slice means that this slice is repeated until you release the button.

Staccato

Shortens the slice.

Mute

Mutes the slice.

Triggering the Performance Controls with Your MIDI Keyboard

You can trigger the performance controls with your MIDI keyboard starting from C3 upwards. You can also use the virtual keyboard for triggering the performance controls (for information about the virtual keyboard see the **Operation Manual**).

Controlling LoopMash FX with a MIDI Keyboard

PROCEDURE

1. Create an audio track and import an audio file, a drum loop, for example.
2. Insert **LoopMash FX** as an insert effect.

3. Play back the drum loop in a cycle.
4. Create a MIDI track.
5. On the **Output Routing** pop-up menu for the MIDI track, select **LoopMash FX**.

RESULT

You can trigger the different **LoopMash FX** performance effects with a MIDI keyboard.

Randomizer

Randomizer allows you to create variations of a sound. By setting specific parameter values, you can determine how much these parameters are randomized. This is helpful if you want to use an audio file, for example, the sound of a closing door, several times in your project and let it sound different every time.

NOTE

This plug-in is recommended for use in offline processing. Using it as a realtime plug-in might increase the latency.

Randomizer allows you to define the range within its parameters are allowed to change. **Pitch**, **Impact**, **Color**, and **Timing** act as metaparameters. This means that when you modify one of these metaparameters, a whole set of parameters is modified in the background.

Pitch

Sets the maximum pitch change.

This parameter allows for very basic and effective sound variations. It is especially suitable for voices and tonal sounds but can also give good results on atonal noises.

Impact

Sets the maximum variations of attack and envelope settings.

This parameter allows for variations of the onsets and transitions of sounds. Even sounds without hard attacks may get interesting variations.

Color

Sets the maximum amount of coloration.

This parameter allows you to change the tonal balance of the fundamental frequency and its harmonics. This can give good results on rich sounds, for example, squeaks and effects but also on voices.

Timing

Sets the maximum timing change.

This parameter allows you to vary the timing of sounds that have a recognizable pattern. At extreme settings, this may change the order of segments within a sound.

NOTE

Setting the above parameters to a value of 5 causes a noticeable yet natural sounding variation. Values higher than 8 alter the sound drastically.

RELATED LINKS

Creating Variations of Audio Events on page 134

Creating Variations of Audio Events

Randomizer allows you to create several variations of audio events in one go.

PREREQUISITE

In the **Direct Offline Processing** window, **Auto Apply** is activated.

NOTE

For detailed information about **Direct Offline Processing**, see the **Operation Manual**.

PROCEDURE

1. Create as many copies of the event as you need.
2. Select all event copies.
3. Select **Audio** > **Plug-ins** > **Other** > **Randomizer**.
4. In the dialog, select **New Version**.

   NOTE

   You can also make this a permanent setting in the **Preferences** dialog (**Editing**-**Audio** page).
5. In the **Direct Offline Processing** window, set the values for **Pitch**, **Impact**, **Color**, and **Timing**.

   The selected events are modified on a random basis within the defined parameter values.

RESULT

You have created different sounding variations of your audio event.

AFTER COMPLETING THIS TASK

Create audio assets by using the **Export Selected Events** dialog and transfer them to a game audio engine, for random playback. In case of Audiokinetic Wwise, use Game Audio Connect to transfer audio assets.

NOTE

For detailed information about Game Audio Connect, see the **Operation Manual**.

Pitch Shift Plug-ins

Doppler

Doppler allows you to emulate the physical characteristics that occur if a sound source, for example, the siren of an ambulance car, is passing by.

The pitch of a sound source increases when moving towards the listener, changes when passing by, and decreases when moving away. The effect is determined by several parameters. Most important is the speed of the sound source: the faster the sound source moves the higher the changes of pitch and volume. Depending on the distance between the sound source and the listener, frequencies with lower energy are more strongly absorbed by the air than frequencies with higher energy, and the volume changes.

Doppler allows you to emulate this effect. You can set the range and the amount of pitch change, the direction of the movement, and the panorama range. Furthermore, you can adjust the distance between the listener and the sound source, and between the listener and the start/end of the movement.

Doppler offers two different modes:

- In **Automatic** mode, the movement of the sound source is created automatically.

  This mode is recommended for offline processing. If your audio track contains more than one event between start and end position of the movement, you must bounce these events to one continuous event before adding **Doppler** as offline process.

  Alternatively, you can load the plug-in as an insert effect and use automation to record the movement. In this case, you must switch to **Manual** mode when reading the automation to assure a correct reproduction.
- In **Manual** mode, you can move the sound source manually.

  If you use this mode, you must load the plug-in as an insert effect and use automation to record the movement.

  NOTE

  Manual mode is not suited for offline processing.

RELATED LINKS
Creating a Doppler Effect by Using Automatic Mode as Offline Process on page 137
Creating a Doppler Effect by Using Automatic Mode as Insert Process on page 138
Creating a Doppler Effect by Using Manual Mode on page 139

Plug-in Panel

The **Doppler** plug-in panel features the following sections: **Mode/Display** section, **Parameters** section, and **Panorama** section.

The following controls are available in the **Mode/Display** section:

Automatic/Manual

Allow you to select **Automatic** or **Manual** mode.

Reset

Resets all plug-in settings to default values.

Graphical display

Visualizes the parameter settings.

Start/Transition/Listener/End (only available in Automatic mode)

Set Start Position allows you to define the position where the movement of the sound source starts.

Set Transition Start Position allows you to define the position where pitch change begins. If this position is not defined, a default value is used.

Set Listener Position allows you to define the position where the sound source passes the listener.

Set End Position allows you to define the position where the movement of the sound source ends.

To adjust a position, move the mouse over the corresponding timecode display and move the mouse wheel.

Object Position slider (only available in Manual mode)

Allows you to follow the movement of the sound source manually using the slider.

The following controls are available in the **Parameters** section:

Panning Direction

Allows you to toggle between a movement from left to right and a movement from right to left.

Locate (only available in Automatic mode)

Sets the left and the right locator to the start and end position, and moves the project cursor to the start position.

L-R Distance

Sets the horizontal distance between the listener and the start/end position.

Pitch

Sets the initial pitch offset of the sound source.

Transition

Sets the range of pitch change. This parameter influences the perception of the speed of the sound source. A short transition range results in a late and drastic pitch change near the listener position and a perception of higher speed. A long transition range results in an early and smooth pitch change and a perception of lower speed.

Depth

Sets the distance between the sound source and the listener on the vertical axis.

The following controls are available in the **Panorama** section:

Activate Panning

If this button is activated, panning is determined by the plug-in parameter settings.

NOTE

Panning in the **Doppler** plug-in works only on stereo tracks.

If this button is deactivated, panning must be done via the channel panner.

Left Panning Range/Right Panning Range

Allows you to adjust the panning range for the left and right channel independently. A value of 100% for left and right correspond to a movement from full left to full right. Smaller values narrow the panorama.

Creating a Doppler Effect by Using Automatic Mode as Offline Process

If you want to create a Doppler effect as offline process, you must use the plug-in in **Automatic** mode. In this mode, the sound source is linked to the project cursor. This allows you to create a passage with a Doppler effect by setting specific positions.

PREREQUISITE

- Your project contains a video track and an audio track of a sound source that passes in front of the listener, for example, a car that passes by, and the audio event on the track is selected.
- If the audio track contains more than one event between start and end position of the movement, you have bounced these events to one continuous event.
- You have opened the **Video Player** window.
- In the **Direct Offline Processing** window, you have activated **Auto Apply**.
- You have added the **Doppler** plug-in as offline process.
- On the plug-in panel, you have activated **Automatic** mode.

PROCEDURE

1. Optional: Click **Panning Direction** to set the direction in which the sound source moves.

2. On the **Transport Panel**, start playback.
3. At the time position where you want the movement of the sound source to start, click **Set Start Position**.
4. Optional: When the pitch change is supposed to begin, click **Set Transition Start Position**.

   If this step is skipped, the transition start position is set to a default value.
5. At the time position where you want the movement of the sound source to pass, click **Set Listener Position**.
6. At the time position where you want the movement of the sound source to end, click **Set End Position**.
7. To fine-tune the start, transition start, listener, and end positions, move the mouse over the corresponding timecode and move the mouse wheel.
8. Click **Locate** to set the left and the right locator to the defined start and end position, and to move the project cursor to the start position.

RESULT

The effect is rendered to the audio of your Doppler passage.

To audition the created effect, play back the range between left and right locator.

AFTER COMPLETING THIS TASK

In the **Parameters** and the **Panorama** section, adjust the parameter settings until the effect meets your expectations. These parameter changes are instantly rendered into the audio.

Creating a Doppler Effect by Using Automatic Mode as Insert Process

If you use **Doppler** in **Automatic** mode as an insert effect, you must use automation to record the movement. In this mode, the sound source is linked to the project cursor. This allows you to create a passage with a Doppler effect by setting specific positions.

PREREQUISITE

- Your project contains a video track and an audio track of a sound source that passes in front of the listener, for example, a car that passes by.
- You have opened the **Video Player** window.
- You have added **Doppler** as insert plug-in.
- On the plug-in panel, you have activated **Automatic** mode.

PROCEDURE

1. Optional: Click **Panning Direction** to set the direction in which the sound source moves.
2. On the **Transport Panel**, start playback.
3. At the time position where you want the movement of the sound source to start, click **Set Start Position**.
4. Optional: When the pitch change is supposed to begin, click **Set Transition Start Position**.

   If this step is skipped, the transition start position is set to a default value.
5. At the time position where you want the movement of the sound source to pass, click **Set Listener Position**.
6. At the time position where you want the movement of the sound source to end, click **Set End Position**.
7. To fine-tune the start, transition start, listener, and end positions, move the mouse over the corresponding timecode and move the mouse wheel.
8. Click **Locate** to set the left and the right locator to the defined start and end position, and to move the project cursor to the start position.
9. Activate **Write Automation** on the plug-in panel.

10. Play back the complete passage.

    Automation data for the Doppler passage is written.
11. Stop playback.

RESULT

Automation data for your Doppler passage is recorded.

To audition the created effect when using **Doppler** as insert effect, you must set the plug-in to **Manual** mode first.

NOTE

Modifying existing automation data for **Doppler** parameters can be troublesome. Therefore, we recommend to start from scratch if the automation pass does not meet your expectations.

AFTER COMPLETING THIS TASK

- Use automation to adjust the controls in the **Parameters** and the **Panorama** section, until the effect meets your expectations.
- We recommend to render the finalized Doppler passage into the audio before using the **Render in Place** function with **Channel Settings**. For more information about **Render in Place**, see the **Operation Manual**.

Creating a Doppler Effect by Using Manual Mode

In **Manual** mode, you can move the sound source with a slider. For this to work, you must load the plug-in as an insert effect and use automation to record the movement.

PREREQUISITE

- Your project contains a video track and an audio track of a sound source that passes in front of the listener, for example, a car that passes by.
- You have opened the **Video Player** window.
- You have added **Doppler** as insert plug-in.
- On the plug-in panel, you have activated **Manual** mode.
- You have activated **Write Automation** on the plug-in panel.

PROCEDURE

1. Set the **Object Position** slider to the position where the movement of the sound source in the video starts.
2. Optional: Click **Panning Direction** to set the direction in which the sound source moves.
3. On the **Transport Panel**, start playback.
4. On the plug-in panel, move the **Object Position** slider corresponding to the movement of the sound source.

   Automation data for the Doppler passage is written.
5. Stop playback.

RESULT

Automation data for your Doppler passage is recorded. To audition the created effect, play back the passage with automation.

NOTE

Modifying existing automation data for **Doppler** parameters can be troublesome. Therefore, we recommend to start from scratch if the automation pass does not meet your expectations.

AFTER COMPLETING THIS TASK

- Use automation to adjust the controls in the **Parameters** and the **Panorama** section, until the effect meets your expectations.

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.

Pitch Correct

Pitch Correct automatically detects, adjusts, and fixes slight pitch and intonation inconsistencies in monophonic vocal and instrumental performances in realtime. The advanced algorithms preserve the formants of the original sound and allow for natural sounding pitch correction without the typical Mickey Mouse effect.

You can use **Pitch Correct** creatively. You can create backing vocals, for example, by modifying the lead vocals or vocoder sounds by using extreme values. You can use an external MIDI controller, a MIDI track, or the virtual keyboard to play a note or a scale of target pitches that determine the current scale notes to which the audio is shifted. This allows you to change your audio in a very quick and easy way, which is extremely useful for live performances. In the keyboard display, the original audio is displayed in blue while the changes are displayed in orange.

Correction

Speed

Determines the smoothness of the pitch change. Higher values cause the pitch shift to occur immediately. 100 is a very drastic setting that is designed mainly for special effects.

Tolerance

Determines the sensitivity of analysis. A low Tolerance value lets Pitch Correct find pitch changes quickly. If the Tolerance value is high, pitch variations in the audio (for example, vibrato) are not immediately interpreted as note changes.

Transpose

With this parameter you can adjust, or retune, the pitch of the incoming audio in semitone steps. A value of zero means that the signal is not transposed.

Scale source

Internal

If you choose the **Internal** option from the **Scale Source** pop-up menu, you can use the pop-up menu next to it to decide to which scale the source audio is adapted.

- **Chromatic**: The audio is pitched to the closest semitone.
- **Major/Minor**: The audio is pitched to the major/minor scale specified on the pop-up menu to the right. This is reflected on the keyboard display.
- **Custom**: The audio is pitched to the notes that you specify by clicking keys on the keyboard display. To reset the keyboard, click the orange line below the display.

External MIDI Scale

Select this option if you want the audio to be shifted to a scale of target pitches, using an external MIDI controller, the virtual keyboard, or a MIDI track.

NOTE

You have to assign the audio track as the output of your MIDI track and the **Speed** parameter has to be set to a value other than **Off**.

External MIDI Note

Select this option if you want the audio to be shifted to a target note, using an external MIDI controller, the Virtual Keyboard or a MIDI track.

NOTE

You have to assign the audio track as the output of your MIDI track and the **Speed** parameter has to be set to a value other than **Off**.

Chord Track – Chords

Select this option if you want the audio to be shifted to target chords, using the chord information from the Chord track.

NOTE

You have to add a MIDI track in addition to the Chord track and assign **Pitch Correct** as output of the MIDI track.

Chord Track – Scale

Select this option if you want the audio to be shifted to a scale of target pitches, using the scale information from the Chord track.

NOTE

You have to add a MIDI track in addition to the Chord track and assign **Pitch Correct** as output of the MIDI track.

Formant

Shift

Changes the natural timbre, that is, the characteristic frequency components of the source audio.

Optimize (General, Male, Female)

Allows you to specify the sound characteristics of the sound sources. **General** is the default setting, **Male** is designed for low pitches and **Female** for high pitches.

Preservation

If this parameter is set to **Off**, formants are raised and lowered with the pitch, provoking strange vocal effects. Higher pitch correction values result in Mickey Mouse effects, lower pitch correction values in Monster sounds. If this parameter is set to **On**, the formants are kept, maintaining the character of the audio.

Master Tuning

Detunes the output signal.

PitchDriver

PitchDriver allows for sound design in postproduction. It can be used for extreme up or down pitching of voices or effect samples (for example, to create eerie monster sounds). Shifting the pitch with this plug-in does not keep the formants.

Detune

Lets you detune the pitch of the incoming audio.

Mix

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

Spatial

Creates an ambience effect. It introduces a light pitch offset to the incoming signal. Different offset values are used for the individual input channels in order to create a panorama effect.

NOTE

The created panorama effect can be unstable. For a stable panorama, deactivate the **Spatial** parameter. Then, the incoming signals are summed up to a mono signal.

Output

Sets the output level.

To avoid hearing artifacts, it is recommended to set the ASIO buffer for your audio card to at least 128 samples. The buffer size can be set on the control panel for the card (opened via the **Device Setup** dialog).

VoiceDesigner

VoiceDesigner is a versatile sound design plug-in that provides extreme pitch-shifting and morphing effects, using an external side-chain signal or the integrated sound generator. You can also use the plug-in to create robotic voices with just one click. The mixing section allows you to set different levels and frequency ranges for the dry, wet, side-chain, and generator signal.

Robot

Activates/Deactivates the robotic voice effect. The **Whisper** parameter allows you to soften the sound of the robotic voice.

Morph

Activates/Deactivates the morphing effect. The input signal is processed using characteristics of a side-chain signal or of the integrated sound generator. The **Mode** selector allows you to toggle between the two morphing modes **A** and **B**.

FX

Activates/Deactivates the **Delay** and the **Feedback** sound effects.

Delay

Adds a delay to the signal.

Feedback

Adds feedback to the signal.

Transition

Allows you to morph the input signal into the side-chain or generator signal. The slider sets the morphing amount. This parameter is only available in morphing mode **A**.

Response

Sets the response time of the morphing algorithm. Fast response settings preserve fast changing transients of the input signals, such as consonants in speech signals. Slower response settings lead to blurry, pad-like sounds. This parameter is only available in morphing mode **B**.

Swap

Swaps source and target for the morphing effect. This parameter is only available in morphing mode **B**.

Resolution
Sets the resolution of the morphing signal. Lower values result in a more rhythmical sound. Higher values preserve the intelligibility of speech signals.

Generator shape selector
Lets you choose the characteristic of the internal sound generator. **White Noise** and **Pink Noise**, and **Square** and **Sawtooth** waveforms are available.

Frequency
Sets the frequency for the **Square** and **Sawtooth** waveforms of the internal sound generator.

Detune
Changes the pitch of the input signal.

Formant
Changes the natural timbre of the input signal.

Preserve
Keeps the formants when changing the pitch with the **Detune** control.

Spatial
Adds an ambience effect by using slightly different settings on all channels.

Dry
Sets the level for the dry input signal. The slider below allows you to set a low-cut and high-cut filter for the input signal.

Generator/Side-Chain
If external side-chain is activated, this control sets the level of the side-chain input. If external side-chain is deactivated, it sets the level for the internal sound generator. The slider below allows you to set a low-cut and high-cut filter for the generator or side-chain signal.

Wet
Sets the level for the effect signal. The slider below allows you to set a low-cut and high-cut filter for the effect signal.

Output
Sets the output level.

Output meter
Shows the level of the output signal.

Reverb Plug-ins

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

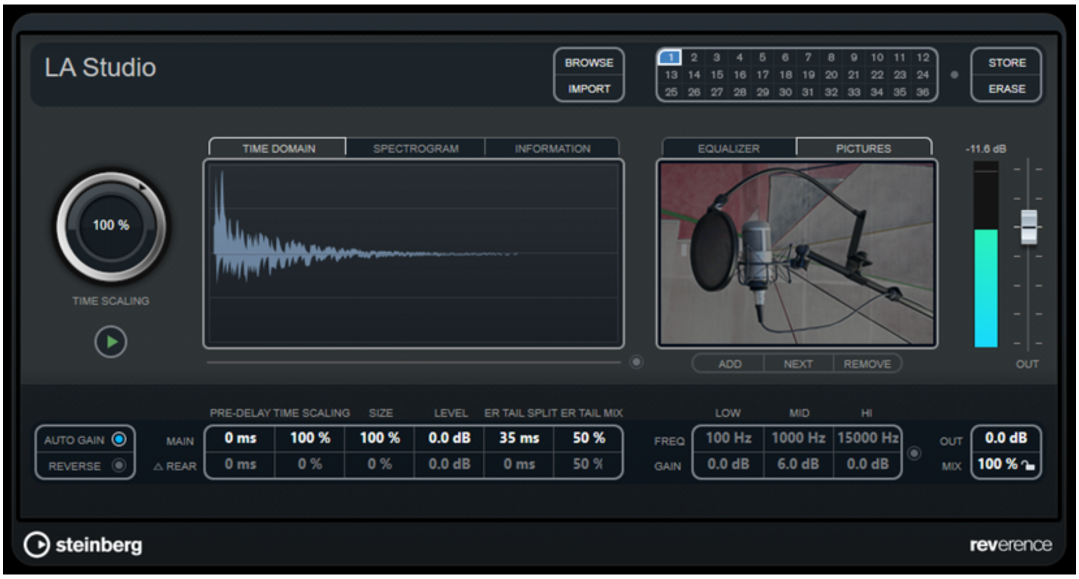
Sets the intensity of the pitch modulation.

Modulation Activate

Activates/Deactivates the chorusing effect.

REVerence

REVerence is a convolution tool that allows you to apply room characteristics (reverb) to the audio.



This is done by processing the audio signal according to an impulse response – a recording of an impulse in a room or another location that recreates the characteristics of the room. As a result, the processed audio sounds as if it were played in the same location. Included with the plug-in are top quality samples of real spaces to create reverberation.

NOTE

REVerence can be very demanding in terms of RAM. This is because the impulse responses that you load into the program slots are preloaded into RAM to guarantee artifact-free switching between programs. Therefore, you should always load only those programs that you need for a given task.

Program Matrix

A program is the combination of an impulse response and its settings. These include reverb settings, EQ settings, pictures, and output settings. The program matrix allows you to load programs or to view the name of the impulse response.

Program name

In the upper left corner of the plug-in panel, either the name of the loaded impulse response file or the name of the program is shown. After loading an impulse response, its number of channels and the length in seconds are displayed for a few seconds.

Browse

This button opens a browser window showing the available programs. If you select a program in the browser, it is loaded into the active program slot. To filter the list of impulse responses by room type or the number of channels, for example, activate the **Filters** section (by clicking the **Set Up Window Layout** button at the top right of the browser window).

Import

Click this button to load your own impulse response files from disk. The files should have a maximum length of 10 seconds. Longer files are automatically cut.

Program slots

Use these slots to load all the impulse responses that you want to work with in a session. The selected program slot is indicated by a white frame. Used slots are shown in blue. A red program slot indicates that the impulse response file is missing. Double-clicking an empty program slot opens a browser window, showing the available programs. Clicking a used program slot recalls the corresponding program and loads it into **REVerence**. If you move the mouse over a used slot, the corresponding program name is displayed below the active program name.

Smooth Parameter Changes

This button is located between the program slots and the **Store**/**Erase** buttons. If it is activated, a crossfade is performed when switching programs. Leave this button deactivated while looking for a suitable program or an appropriate setting for an impulse response. Once you have set up the program matrix to your liking, activate the button to avoid hearing artifacts when switching between programs.

Store

Stores the active impulse response and its settings as a program.

Erase

Removes the selected program from the matrix.

Programs vs. Presets

You can save your **REVerence** settings as VST plug-in presets or programs. Both presets and programs use the file extension `.vstpreset` and appear in the same category in the MediaBay, but they are represented by different icons.

Preset

A **REVerence** preset contains all settings and parameters for the plug-in, that is, a link to all loaded impulse responses along with their parameter settings and positions in the program matrix. Presets are loaded via the presets pop-up menu at the top of the plug-in panel.

NOTE

Manually imported impulse responses themselves are not part of the program or preset. If you want to move your project to a different computer, you have to move the impulse responses as well.

Program

A **REVerence** program only contains the settings related to a single impulse response. Programs are loaded and managed via the program matrix.

Presets

Presets are useful in the following situations:

- To save a complete setup with different impulse responses for later use (for example, different setups for explosion sounds that can be reused for other scenes or movies).
- When you want to save different parameter sets for the same impulse response so that you can later choose the set that best suits your needs.

Programs

Programs offer the following advantages:

- Up to 36 programs can be loaded into the program matrix for instant recall.
- A program provides a quick and easy way to save and recall the settings for a single impulse response, allowing for short loading times.
- When automating a project and loading a **REVerence** program, only one automation event is written.

  If you load a plug-in preset instead (which contains a lot more settings than a program), a lot of unnecessary automation data (for the settings that you did not use) is written.

RELATED LINKS
Reverb Settings on page 151
EQ Settings on page 153
Pictures Section on page 154
Custom Impulse Responses on page 155
Relocating Content on page 157

Setting up Programs

PROCEDURE

1. In the program matrix, click on a program slot to select it.

   A blinking white frame indicates that this program slot is selected.
2. Click the **Browse** button or click the empty slot again to load one of the included programs.

   You can also import a new impulse response file.
3. In the browser, select the program containing the impulse response that you want to use and click **OK**.

   The name of the loaded impulse response is shown in the upper left corner of the **REVerence** panel.
4. Set up the **REVerence** parameters as and click the **Store** button to save the impulse response with the current settings as a new program.
5. Set up as many programs as you need by following the steps above.

   NOTE

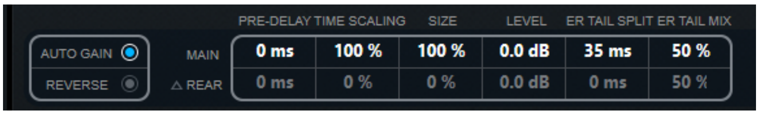
   If you want to use your set of programs in other projects, save your settings as a plug-in preset.

RELATED LINKS

Importing Impulse Responses on page 155

Reverb Settings

The reverb settings allow you to change the characteristics of the room.



Main

All values shown in the top row apply to all speakers or to the front channels if you are working with surround tracks.

Rear

If you are working with surround tracks up to 5.1, you can use this row to set up an offset for the rear channels.

Auto Gain

If this button is activated, the impulse response is automatically normalized.

Reverse

Reverses the impulse response.

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.

Time Scaling

Controls the reverb time.

Size

Determines the size of the simulated room.

Level

A level control for the impulse response. This governs the volume of the reverb.

ER Tail Split

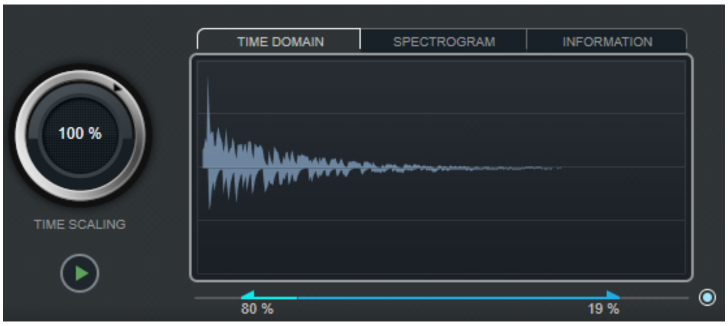
Sets a split point between the early reflections and the tail, allowing you to determine where the reverb tail begins. A value of 60 means that the split point is set to 60 ms.

ER Tail Mix

Allows you to set up the relation of early reflections and tail. Values above 50 attenuate the early reflections and values below 50 attenuate the tail.

Impulse Response Display

The display section allows you to view the impulse response details and to change the length of the response.



Time Scaling

This wheel lets you adjust the reverb time.

Play

When clicking this button to apply the loaded impulse response, a short click is played. This provides a neutral test sound that makes it easier for you to know how different settings influence the reverb characteristics.

Time Domain

This display shows the waveform of the impulse response.

Spectrogram

This display shows the analyzed spectrum of the impulse response. Time is displayed along the horizontal axis, frequency along the vertical axis, and volume is represented by the color.

Information

This display shows additional information, such as the name of the program and the loaded impulse response, the number of channels, the length, and Broadcast Wave File information.

Activate Impulse Trimming

Use this button at the bottom right of the impulse display section to activate impulse trimming. The Trim slider is shown below the Impulse display.

Trim

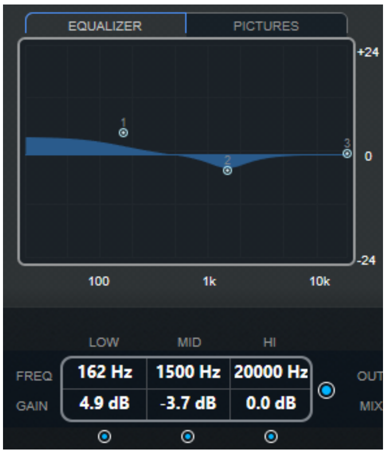
This slider allows you to trim the start and end of the impulse response. Drag the front handle to trim the start of the impulse response, and the end handle to trim the reverb tail.

NOTE

The impulse response is cut without any fading.

EQ Settings

In the **Equalizer** section, you can tune the sound of the reverb.



EQ curve

Shows the EQ curve. You can use the EQ parameters below the display to change the EQ curve, or modify the curve manually by dragging the curve points.

Activate EQ

This button to the right of the EQ parameters activates the EQ for the effect plug-in.

Low Shelf On

Activates the low shelf filter that boosts or attenuates frequencies below the cutoff frequency by the specified amount.

Low Freq (20 to 500)

Sets the frequency of the low band.

Low Gain (-24 to +24)

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

Mid Peak On

Activates the mid peak filter that creates a peak or notch in the frequency response.

Mid Freq (100 to 10000)

Sets the center frequency of the mid band.

Mid Gain(-12 to +12)

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

Hi Shelf On

Activates the high shelf filter that boosts or attenuates frequencies above the cutoff frequency by the specified amount.

Hi Freq (5000 to 20000)

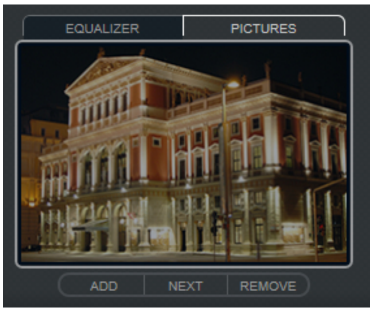
Sets the frequency of the high band.

Hi Gain (-24 to +24)

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

Pictures Section

In the **Pictures** section, you can load graphics files to illustrate the setting, for example, the recording location or microphone arrangement of the loaded impulse response. Up to five pictures can be loaded.



NOTE

Pictures are only referenced by the plug-in and are not copied to the project folder.

Add

Opens a file dialog where you can navigate to the graphics file that you want to import. JPG, GIF, and PNG file formats are supported.

Next

If several pictures are loaded, you can click this button to display the next image.

Remove

Deletes the active picture.

NOTE

This does not remove the graphics file from your hard disk.

Output Settings

In the output section, you can control the overall level and determine the dry/wet mix.

Output activity meter

Indicates the overall level of the impulse response and its settings.

Out

Adjusts the overall output level.

Mix

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

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 and programs.

Custom Impulse Responses

In addition to working with the impulse responses included with **REVerence**, you can import your own impulse responses and save them as programs or presets. WAVE and AIFF files with a mono, stereo, true-stereo, or multi-channel (up to 5.0) configuration are supported. If a multi-channel file contains an LFE channel, this channel is ignored.

REVerence uses the same channel width as the track it is inserted on. When importing impulse response files with more channels than the corresponding track, the plug-in only reads as many channels as needed. If the impulse response file contains fewer channels than the track, **REVerence** generates the missing channels (for example, the center channel as a sum of the left and right channels). If the rear channels are missing (when importing a stereo response file onto a 4.0 track, for example), the left and right channels are also used for the rear channels. In this case you can use the Rear offset parameter to create more spatiality.

Importing Impulse Responses

REVerence allows you to import your own impulse response files. Before importing these impulse response files, you can preview their effect.

PREREQUISITE

To preview the effect of impulse response files during the import process, do one of the following:

- If you use **REVerence** as an insert effect, play back the event to which you want to add the effect in a loop.
- If you use **REVerence** in the **Direct Offline Processing** window, activate **Audition** and **Audition Loop**.

PROCEDURE

1. In the program matrix, click **Import**.
2. In the file dialog that opens, navigate to the location of your impulse response files.
3. Optional: Select an impulse response file to preview it.
4. Select the file that you want to import and click **Open**.

   The file is loaded into **REVerence**. The channels from an interleaved file are imported in the same order as in other areas of the program (for example, the **Audio Connections** window), see below.
5. Make the appropriate settings and add a picture, if available.

   Pictures residing in the same folder as the impulse response file or in the parent folder are automatically found and displayed.
6. Click the **Store** button to save the impulse response and its settings as a program.

   That way you can recall the setup at any time.

RESULT

The program slot turns blue, indicating that a program is loaded.

NOTE

When saving a program, the impulse response file itself is only referenced. It still resides in the same location as before and is not modified in any way.

AFTER COMPLETING THIS TASK

Repeat these steps for any impulse response files that you want to work with.

Reading Order of Input Channels

REVerence reads input channels in the following order.

Number of input channels	Channel order in REVerence
1	L
2	L/R
3	L/R/C
4	L/R/LS/RS (if inserted on a track with a 4.0 channel configuration)
4	LL/LR/RL/RR (if inserted on a track with a stereo configuration)
5	L/R/C/LS/RS
6	L/R/C/LFE/LS/RS (LFE is being ignored.)

True Stereo

Impulse responses recorded as true-stereo files allow you to create a very realistic impression of the corresponding room.

REVerence can only process true-stereo impulse response files with the following channel configuration (in exactly that order): LL, LR, RL, RR.

The channels are defined as follows:

Channel	The signal from this source…	…was recorded with this microphone
LL	left source	left microphone
LR	left source	right microphone
RL	right source	left microphone
RR	right source	right microphone

NOTE

If your true-stereo impulse responses are only available as separate mono files, you can use the **Export Audio Mixdown** function to create **REVerence** compliant interleaved files (see the **Operation Manual**).

REVerence automatically works in true-stereo mode if the plug-in is inserted on a stereo track and you load a 4-channel impulse response.

Therefore, if you are working with surround files, that is, 4-channel impulse responses recorded with a Quadro configuration (L/R, LS/RS), you need to insert the plug-in on an audio track with a 4.0 configuration. On a stereo track, these files would be processed in true-stereo mode, too.

So how can you prevent **REVerence** from unintentionally processing surround files in true-stereo mode? The answer is a **Recording Method** attribute that can be written to the iXML chunk of the corresponding impulse response file. Whenever you load an impulse response with a 4-channel configuration on a stereo track, **REVerence** searches the iXML chunk of the file. If the plug-in finds the **Recording Method** attribute, the following happens:

- If the attribute is set to **TrueStereo**, the plug-in works in true-stereo mode.
- If the attribute is set to **A/B** or **Quadro**, the plug-in works in normal stereo mode and processes only the L/R channels of the surround file.

NOTE

You can use the **Attribute Inspector** in the **MediaBay** to tag your own impulse response files with the **Recording Method** attribute. For more information, see the **Operation Manual**.

Relocating Content

Once you have imported your own impulse responses to **REVerence**, you can comfortably work with them on your computer. But what if you need to transfer your content to another computer, for example, because you work sometimes with a PC and sometimes with a notebook, or you need to hand over a project to a colleague in the studio?

The factory content is not a problem because it is also present on the other computer. For these impulse responses, you just need to transfer your **REVerence** programs and presets to be able to access your setups.

User content is a different matter, though. If you have transferred your audio files to an external drive or a different hard disk location on the other computer, **REVerence** cannot access the impulse responses any more because the old file paths have become invalid.

PROCEDURE

1. Transfer you audio files to a location that you can access from the second computer (for example, an external hard disk).

   If you keep the files in the same folder structure as on the first computer, **REVerence** automatically finds all files contained in this structure.
2. Transfer any **REVerence** presets or programs that you need to the second computer.

   If you are unsure where the presets need to be stored, you can find the paths in the **MediaBay** (see the **Operation Manual**).
3. Open **REVerence** on the second computer and try to load the preset or program that you want to work with.

   The **Locate Impulse Response** dialog opens.
4. Navigate to the folder that contains your impulse responses.
5. Click **Open**.

RESULT

REVerence is now able to access all the impulse responses stored in this location.

IMPORTANT

The new path to these audio files has not been saved yet. To make the files permanently available without having to use the locate dialog, you need to save your programs or presets under a different name.

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

Allows you to set the reverb time in seconds.

Diffusion

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

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

Allows you to set the reverb time in seconds.

Diffusion

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

Low Level

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

High Level

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

Mix

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

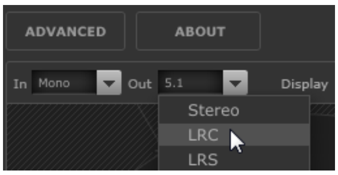
Spatial + Panner Plug-ins

Anymix Pro

The **Anymix Pro** plug-in from IOSONO is a sophisticated surround panner and a powerful upmix/downmix processor that converts any given audio material into output formats ranging from mono to 8.1.

Input/Output Configuration

The input/output configuration of the plug-in can be selected from the **In** and **Out** pop-up menus in the top left corner of the plug-in panel.



If **Anymix Pro** is used as an insert effect, the maximum input and output configuration cannot exceed the track width of the current track.

If **Anymix Pro** is used as a panner, the maximum input configuration cannot exceed the track width of the current track. The maximum output configuration cannot exceed the width of the output bus that the track is routed to.

Channel Order

The plug-in uses the channel order of the host application unless the selected output configuration differs from the track configuration.

NOTE

Choosing an output configuration that differs from the current track configuration results in channel oddities.

If the output configuration of the track is not a subset of the plug-in output configuration, for example, track = 6.1 cine and plug-in output = 7.0 music, the channels are routed as follows:

1	2	3	4	5	6	7	8	9
L	R	C	LFE	LS	RS	RSS/ RC	LSS/ LC	CS

NOTE

Channels that are missing in the output configuration are automatically skipped.

	1	2	3	4	5	6	7	8	9
Track configuration: 6.1 Cine	L	R	C	LFE	LS	RS	CS		
Plug-in output configuration: 7.0 Music	L	R	C	LS	RS	LSS	RSS		
Result	OK			Mismatch					

Latency Compensation

Anymix Pro causes a processing delay. The amount of latency depends on the buffer size of the audio card and the processing mode of the plug-in, that is, panning or upmix. Steinberg host applications can compensate this delay automatically.

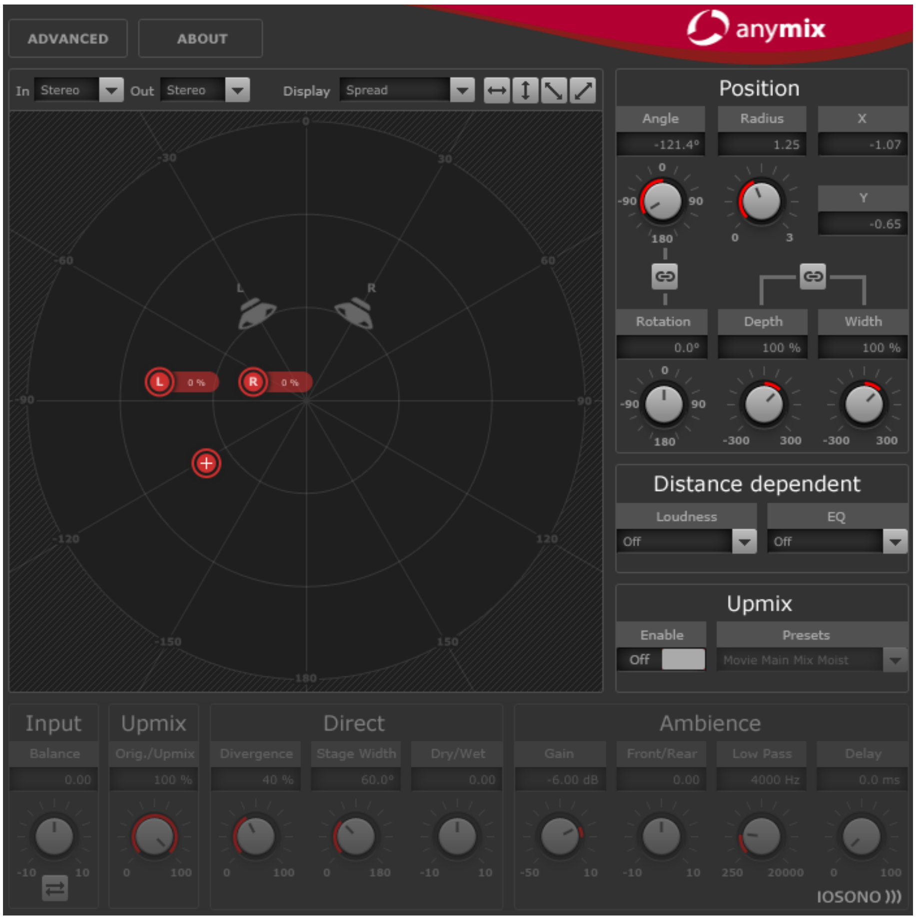
The Plug-in Panel

The panel of **Anymix Pro** is divided into several sections, with the stage view taking the most space to display the position and movement of the input channels, output configuration, and distance-dependent filter values. On the right side, there are the controls for position and movement, and the lower section of the plug-in panel contains the upmix controls.

NOTE

The plug-in panel has two different display modes: panning and upmix.

Panning Mode



In the stage view, input channels are represented by red icons, output channels by gray speakers in the background.

Moving the input channels outside the loudspeaker setup results in panning between the two nearest output speakers. The input channels that are placed at smaller distances are distributed to several output speakers.

- To change the position of the input group, click and drag anywhere in the stage view, or right-click in the stage view. Right-clicking causes the channels to jump to the new position.

NOTE

The distance between the input channels automatically shrinks if they are moved to the border of the stage. This lets you create the illusion of depth when moving stereo or multi-channel material.

Position Section

In addition to using the stage view, the input channels can also be moved using the controls at the top right of the plug-in panel.

Rotation

Rotates the input group around its center point.

Link Angle & Rotation

Changes the rotation of the input group from self-centered to stage-centered.

Depth
Scales the input group vertically.

Width
Scales the input group horizontally.

Link Depth & Width
Keeps the aspect ratio between Depth and Width scaling.

NOTE

To fine-adjust the parameters, press **Shift** while using the controls.

Individual Channel Adjustment

You can change the positions of the input channels individually by double-clicking the corresponding input icon in the stage view. A separate panel with channel-specific parameters opens.

Radius/Angle
Control the position of the selected input channel, relative to the center of the input group.

X/Y
Move the selected input channel horizontally and vertically.

Volume
Applies gain to the selected input channel.

LFE Volume
Controls the amount of LFE for the selected input channel.

Spread
Distributes the audio from the selected input channel to more than two output channels.

- At 0 % the audio source is rendered where the channel icon is placed.
- At 100 % the audio is evenly distributed to all speakers of the output configuration.

Manual Delay
Adds a delay to the selected input channel.

Link
Activate these buttons to link the corresponding parameters in the current plug-in instance. Adjusting the value of a linked parameter changes the other linked parameters, too.

IMPORTANT

The individual input channel parameters cannot be automated from the host application, but the adjustments you make for each input channel are saved for each plug-in instance and panner in the session.

Restricting Movement

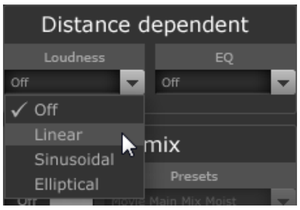
You can use the double-arrow buttons at the top right of the stage view to restrict the direction of movement of the object in the stage view to orthogonal or diagonal, for easy automation.

NOTE

In most cases, objects move on very simple routes around the audience. By restricting the direction of movement, you can quickly create accurate movements.

Distance-Dependent Filters

To create immersive mixes even faster, **Anymix Pro** is equipped with a distance-dependent filter unit that lets you adjust the volume and air damping of moving objects automatically.



Loudness

Lowers the volume for objects that are further away.

EQ

Dampens the high frequencies of objects that are further away.

For both of the filters, you can select one the following options from the corresponding pop-up menu:

- **Off**
  Deactivates the distance-dependent filter.
- **Linear**
  The filtering starts right from the center point and is applied linearly. Select this curve type if even tiny movements should have an impact on the distance-dependent filter.
- **Sinusoidal**
  The filtering starts approximately at loudspeaker distance and increases exponentially with distance. Select this curve type if movements in the center circle should have no audible impact on the distance-dependent filter.
- **Elliptical**
  The filtering starts approximately at two thirds of the stage with an exponential attenuation curve. Select this curve type if only movements along the border of the stage should have an impact on the distance-dependent filter.

NOTE

The current values can be shown in the speaker icon labels, using the **Display** pop-up menu above the stage view.

The distance-dependent filters can be further adjusted using the advanced options.

RELATED LINKS
Advanced Options on page 170

Upmix

The upmix feature of **Anymix Pro** is very useful if rearranging tracks with fewer input channels into a specific surround format is not enough.

The upmix algorithm analyzes the incoming audio signal and separates it into parts of direct sound and ambient sound. While the direct sound parts are sent to the direct sound stream and can be placed at the virtual front speaker configuration, the ambient sound parts can be modified and arranged around the virtual stage. Note that this does not add any additional information to the audio stream. All sound parts that you hear from the ambient sound were already part of the original audio material.

IMPORTANT

If your audio does not contain spatial information, there cannot be an ambient sound stream. For example, you cannot extract ambient sound from a dry recording of a narrator sitting in a recording booth.

IMPORTANT

Lossy compression, such as in MP3 files, or other deficiencies of the incoming audio cannot be remedied using the upmix mode. For example, compression artifacts can easily be misinterpreted and redistributed to the ambient sound stream.

Switching to Upmix Mode

- To switch to upmix mode, activate the **Enable** option in the **Upmix** section to the right of the stage view.

NOTE

The upmix algorithm is very sophisticated and can cause a high CPU load. Therefore, you cannot automate the **Enable** option.

Stage View

In upmix mode, the parameters are represented by segments of a circle in the stage view.

NOTE

The position parameters for the input group and any created automation are preserved when the upmix is enabled. In upmix mode, the sound image created by the upmix algorithm can be moved around the stage and is also fully automatable. The parameters that you have adjusted for a single channel have no influence on the upmix, but they are kept and are automatically reloaded when the upmix is disabled.

Upmix Presets

Anymix Pro comes with a set of preconfigured upmix presets. If a preset is loaded, the upmix and advanced parameters are set accordingly and can be further adjusted.

An upmix preset contains settings for the following upmix parameters:

- **Divergence**
- **Stage Width**
- **Direct Dry/Wet**
- **Ambience Gain**
- **Ambience Front/Rear**
- **Ambience Low Pass**
- **Ambience Delay**.

Furthermore, the following parameters on the **Advanced** panel are affected by the preset:

- **LFE Gain**
- **LFE Low Pass Enable**

- **LFE Low Pass Order**
- **LFE Low Pass Cutoff Frequency**
- **Output Gain**.

NOTE

Upmix presets from the **Cinema** category are designed for the use with X-curve tuned speaker systems. The other presets are designed for listening environments with a flat speaker tuning.

Input – Balance

Adjusts the balance of the input signal if the input signal is stereo or higher.

Upmix – Orig./Upmix

Adjusts the plug-in output between original and processed signal.

Direct Sound Stream Parameters

Divergence

Controls the strength of the center signal.

- At 0 % the mono components of the direct sound stream are distributed to the center channel.
- At 100 % the mono components of the direct sound stream are distributed to the front left and right channels.

Stage Width

Controls the position of the front channels to adjust the stereo base.

Dry/Wet

Controls the amount of ambience that remains in the direct sound stream after the ambience extraction.

Ambient Sound Stream Parameters

Gain

Applies gain to the ambient sound stream to either emphasize (high gain) or dampen (low gain) the amount of ambience in the mix.

Front/Rear

Adjusts the front/rear balance of the ambient sound stream.

Low Pass

Controls the ambient sound stream with a low-pass filter to prevent hissing.

Delay

Adds extra delay to the ambient sound stream to create the illusion of a very large space.

Proximity

Anymix Pro supports the Proximity surround format. Proximity allows you to take a sound from the speakers and place it very close to the listener.

To make this effect audible, you need an appropriate reproduction device, for example, a Proximity headphone system.

If you use the Proximity format as input or output configuration, you can select one of the following patterns from the **In** and **Out** pop-up menus:

Variable Input, Proximity Output

Converts incoming signals to the Proximity format. This allows you to balance the incoming signals between a standard 5.1 setup and two additional Proximity channels.

Proximity Input, Proximity Output

Allows you to adjust the FX Level in the Proximity signal.

Proximity Input, Variable Output

Decodes the incoming Proximity signal and converts it to any standard output configuration other than 5.1.

NOTE

Convert 8.0 and 8.1 tracks to 7.1 before you encode to 7.1 Proximity. This prevents errors in the channel order.

Proximity FX Level

Provides the Proximity effect settings.

Maximum

Allows you to set the maximum Proximity effect level that is applied to the input sources.

Distance Control

Allows you to select how the position of the input channels affects the effect level.

You can select an attenuation curve from the pop-up menu:

- **Off**

  Provides a constant effect level.

- **Linear**

  The attenuation starts right from the center position and is applied linearly.

- **Sinusoidal**

  The attenuation starts approximately at loudspeaker distance and increases exponentially with distance.

- **Elliptical**

  The attenuation starts at two thirds of the stage with an exponential attenuation curve.

Reference

Allows you to select if the distance-based attenuation of the Proximity level depends on the position of the center icon or on the position of each input channel.

Advanced Options

The advanced options can be opened using the top left button in the main plug-in panel. These options can be set for the plug-in instance by adjusting the values as needed.

Distance Dependent Parameters

Loudness

Allows you to select whether the volume change that is to be applied depends on the position of the center point, or if the volume change is calculated for each input channel separately.

Sets the maximum gain reduction that is applied if the group input or channel reaches the stage border.

EQ Gain

Allows you to select whether the filtering that is to be applied depends on the position of the center point, or if the amount of filtering is calculated for each input channel separately.

Sets the maximum gain reduction of the filter that is applied if the group or input channel reaches the stage border.

EQ Cutoff

Sets the cutoff frequency of the distance-dependent EQ.

Upmix – Matrix

Activates matrix decoding for matrix-encoded input signals.

NOTE

Matrix decoding is only applied in upmix mode.

LFE Parameters

LFE Gain

Sets a separate gain level for the LFE channel that is applied to the plug-in output.

LP Enable

Enables a low-pass filter that is applied to the LFE output channel after summing the signals from the input channels.

LP Cutoff

Sets the cutoff frequency for the generated LFE channel.

LP Order

Allows you to select the order, or slope, of the low-pass filter.

- 2nd order = 12 dB/octave
- 3rd order = 18 dB/octave
- 4th order = 24 dB/octave

NOTE

The amount of LFE can be adjusted for each input channel individually.

NOTE

If the selected input configuration includes an LFE channel, but the selected output configuration does not, the LFE input channel is distributed to front left and front right at a level of -3 dB automatically. The low-pass filter is applied to the incoming LFE signal before it is distributed to the front speakers.

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 you can see a thread, the more bass is in the signal, and 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.

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 7.1 cinema surround format to a 5.1 home theater format.

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

MonoToStereo

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

NOTE

This plug-in works only on stereo tracks.

Delay

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

Width

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

Mono

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

Color

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

Renderer for Dolby Atmos

The **Renderer for Dolby Atmos** plug-in allows you to monitor and downmix ADM projects for Dolby Atmos® without using an external renderer.

For a description of **Renderer for Dolby Atmos**, see the **Operation Manual**.

StereoEnhancer

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

NOTE

This plug-in works only on stereo tracks.

Delay

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

Width

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

Mono

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

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

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**.

VST MultiPanner

VST MultiPanner is a surround panner that allows you to position a sound source in the surround field or to modify existing premixes.

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

Surround Plug-ins

Bass Manager

If your main speakers are small, **Bass Manager** allows you to achieve full-range sound by using the LFE speaker of the studio as a subwoofer. If your LFE is small but the main speakers provide full-range sound, the plug-in allows you to strengthen the LFE sound by routing higher frequencies from the LFE to the main speakers.

If your monitoring system is connected directly to your audio interface and lacks a bass management system, you can use **Bass Manager** in the Control Room inserts to adjust the signal levels and signal routing of the LFE and main speakers. Thus, small speakers can achieve full-range sound.

Input/Output Level Meters

Level In
Shows the level of the input signal.

Level Out
Shows the level of the output signal.

Mains

Subwoofer
If this is activated, the low part of the main speakers is routed to the subwoofer.

Freq
Sets the crossover frequency. Frequencies below this value are routed to the subwoofer.

Gain
Sets the amount of attenuation/boost of the signal that is routed to the subwoofer.

LFE

Center
If this is activated, the LFE signal is routed to the center channel.

L/R
If this is activated, the LFE signal is routed to the left and right channels.

Freq
Sets the crossover frequency. Frequencies above this value are routed to the left and right channels or to the center channel.

Gain
Sets the amount of attenuation/boost of the signal that is routed to the left and right channels or to the center channel.

+10 dB
Boosts the level of the LFE signal by 10 dB.

Subwoofer

Solo
Solos the subwoofer signal.

Mute
Mutes the subwoofer signal.

Low Cut
Additional low-cut filter for the subwoofer.

High Cut
Additional high-cut filter for the subwoofer.

+10 dB
Boosts the level of the subwoofer signal by 10 dB.

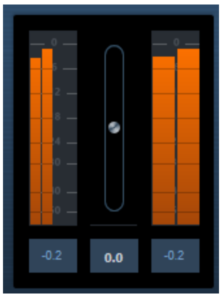
NOTE

You can also use the graphical displays to set the parameters or enter the parameter values manually in the value fields.

MatrixEncoder/MatrixDecoder

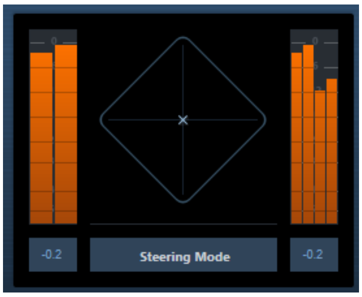
MatrixEncoder is intended for the Pro Logic compatible encoding of multi-channel files. **MatrixDecoder** allows you to monitor how an encoded mix sounds when played back on a Pro Logic compatible system.

MatrixEncoder



MatrixEncoder is intended for the Pro Logic compatible encoding of multi-channel files. This is a process where a 4-channel surround mix is packed into two channels for broadcasting or a two-channel version for DVDs, for example. **MatrixEncoder** takes four separate inputs (LRCS = Left, Right, Center, and Surround) and creates two final outputs: Left-total and Right-total (Lt and Rt).

MatrixDecoder



When using **MatrixDecoder** to play back the encoded mix, the Lt/Rt channels are again converted to four outputs (LRCS).

NOTE

This manual does not attempt to explain the full background on how Pro Logic works, but focuses on how you can use **MatrixEncoder**/**Matrix Decoder** to produce a mix that is compatible with this standard.

RELATED LINKS
Setting Up on page 177
Using MatrixEncoder with the 5.0 Surround Format on page 179
Using MatrixDecoder with the 5.0 Surround Format on page 179

Setting Up

PROCEDURE

1. In the **Audio Connections** window, create an output bus with the LRCS channel configuration and route it to the physical outputs of your audio hardware.

   This is if you want to make a four-channel surround mix. If you want to make a five-channel mix, use **MatrixEncoder** with the 5.0 Surround Format.

2. Place the **MatrixEncoder** in the first post-fader insert slot (#7) for the output bus, followed by the **MatrixDecoder** (#8).

RELATED LINKS
Using MatrixEncoder with the 5.0 Surround Format on page 179

Using MatrixEncoder/MatrixDecoder

PROCEDURE

1. Set up the mix roughly the way you want it.

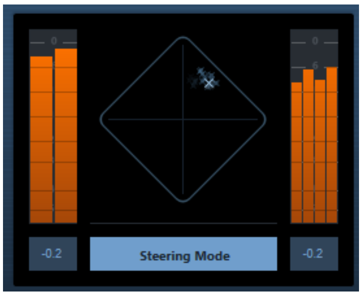
   Use **VST MultiPanner** to place channels in the surround mix, or assign channels to the individual LRCS outputs.

2. Activate **MatrixEncoder**.

   What you now hear is the encoded stereo mix, the way it sounds when it is played back on a normal stereo reproducer. On the **MatrixEncoder** control panel, you can adjust the **Gain** of the Lt/Rt output by using the fader.

3. Activate **MatrixDecoder**, open the control panel and click **Steering Mode**.

   Now you can hear how the mix is reproduced in surround on a Pro Logic compatible system.

   The steering display shows an **x** within the surround field. The position of this **x** indicates the dominant direction of the mix, sometimes referred to as the dominance vector. Part of the processing that is applied results in the dominant channel being enhanced and the non-dominant channels being reduced in gain.

4. By activating and deactivating **Bypass** in **MatrixDecoder**, you can compare the decoded mix with the encoded stereo mix, and make adjustments in the **MixConsole** as necessary.

   The main goal is to produce a mix that sounds good in both the encoded and the decoded version. To compare the encoded or decoded mix with the unprocessed mix, deactivate both **MatrixEncoder** and **MatrixDecoder**.

   IMPORTANT

   The encoding/decoding process produces significant signal loss compared to the unprocessed mix. This is normal, and does not indicate that something is not working properly. However, with careful tweaking of the mix, you can decrease the signal degradation to a much more acceptable level. You have to adjust levels and other settings before the signal runs through **MatrixEncoder**, because neither the encoder or decoder can control the mix in any way.

5. If you are satisfied with the result, bypass **MatrixDecoder**, or remove it from its effect slot.

6. Connect a master recording device to the stereo mix output and perform a mixdown as usual.

RESULT

The resulting encoded stereo mix is compatible with common home systems that use the Pro Logic standard.

Using MatrixEncoder with the 5.0 Surround Format

There are situations where you may want to mix for several surround formats. For example, you might need to mix the same material for 5.1 and LRCS.

5.1 is similar to LRCS. Omitting the LFE channel is easy, but more of a problem is that LRCS only has one surround channel whereas 5.1 has two.

For this reason, the **MatrixEncoder** sums up the surround channels to a mono signal.

PROCEDURE

1. Create your mix for 5.1.
2. In the **Audio Connections** window, create an output bus with a 5.0 channel configuration and route it to the physical outputs of your audio hardware.
3. Run the mix through the **MatrixEncoder**.

RESULT

First, the two surround channels are merged to make the mix compatible with LRCS. Then, the four resulting signals are encoded as usual. This way, far fewer adjustments are necessary when working with 5.1 and LRCS at the same time.

Using MatrixDecoder with the 5.0 Surround Format

Normally, two surround speakers are used even when playing back LRCS. The two speakers then simply use the same material. The **MatrixDecoder** simulates this by delivering the surround channel to two outputs. This allows you to move between formats and listening situations with less repatching of speaker channels.

Mix6to2

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

Surround Channels

Volume faders

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

Link

Links the volume faders of a surround channel.

Invert Phase

Inverts the phase of the corresponding surround bus channel.

Output Bus

Volume faders

Set the volume of the mixed output.

Link

Links the **Output** faders.

Normalize

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

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.

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.

MixerDelay

MixerDelay allows you to adjust and manipulate each individual channel in a surround track, group, or bus.

M

Allows you to mute individual channels.

S

Allows you to solo individual channels.

Inv

Lets you invert the phase or polarity for individual channels.

Above the individual channel controls, you find global buttons for turning off **M**, **S** and **Inv** for all channels.

Delay

Allows you to delay individual speaker channels. The delay times are shown in milliseconds and centimeters, making this feature very useful for distance compensation when playing back surround mixes on different speaker setups, for example.

Level

Allows you to fine-tune the volume balance between the surround channels.

Volume

Shows the level of the input signal.

Routing

Lets you select/switch the outputs for the channels quickly. You can assign the same output to several channels by holding down **Alt/Opt** while selecting. Note that there are also several channel routing presets available.

NOTE

It is common for the center channel in a 5.1 speaker configuration to be closer to the mix position in order to accommodate large video monitors or projection screens. **MixerDelay** can be used to compensate for the center channel being too close. Simply adjust the delay for the center channel by the difference in distance (in cm) between it and the other speakers to the mix position. You must delay the closer speaker so that its sound arrives at the same time as the sound from the more distant speakers. Note that **MixerDelay** has a wide range (up to 1000 ms), and fine adjustments are best made by numerically entering the delay time in centimeters for speaker alignment.

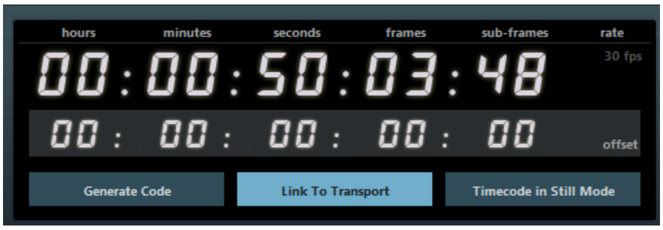
IMPORTANT

The **MixerDelay** is not a mixer – the number of outputs is the same as the number of inputs. If you need to mix down a surround signal to stereo, use the **Mix6to2**, **Mix8to2**, or **MixConvert V6** plug-ins.

Tools Plug-ins

SMPTEGenerator

SMPTEGenerator is not a real audio effect. It sends out SMPTE timecode to an audio output, allowing you to synchronize other equipment to your host application (provided that the equipment can synchronize directly to SMPTE timecode). This can be very useful if you do not have access to a MIDI-to-timecode converter.



Main timecode display

This display shows the current timecode.

If **Link to Transport** is deactivated, the generator is in free run mode. You can use the timecode display to set the SMPTE start time.

If **Link to Transport** is activated, you cannot change any of the values. This display shows the current timecode in sync with the Transport panel. Where applicable, the offset defined in the offset timecode display is taken into account.

Frame rate display and pop-up menu

The frame rate shown to the right of the timecode display corresponds to the frame rate set in the Project Setup dialog. To generate timecode in a different frame rate (for example, to stripe a tape), select another format on the pop-up menu (only available if **Link to Transport** is deactivated).

NOTE

For another device to synchronize correctly to your host, the same frame rate has to be set in the Project Setup dialog, **SMPTEGenerator**, and the receiving device.

Offset timecode display

This display is only available if **Link to Transport** is activated. It allows you to set an offset with regard to the timecode used by your host application. The offset affects the generated SMPTE signal, the current cursor position remains unaffected.

For example, use this when playing back video using an external device, where the video starts at a different timecode position than in your host. A scenario could be as follows: You have placed the same video several times on the timeline, in order to record different audio versions for that video one after the other. However, since video playback is done via an external machine (replaying the same video), you need an offset to match the different timecode positions in your host with the (unchanging) start position on the external machine.

Generate Code

If this button is activated, the plug-in generates SMPTE timecode in free run mode, meaning that it outputs continuous timecode independent from the Transport panel. Use this mode if you want to stripe tape with SMPTE.

Link to Transport

If this button is activated, the timecode is synchronized to the Transport panel.

Timecode in Still Mode

If this button is activated, the plug-in also generates SMPTE timecode in stop mode. However, note that this is not continuous timecode, but timecode generated at the current cursor position. For example, this can be useful when working with video editing software that interprets the absence of timecode as a stop command. By using this option, the video software can enter still mode instead so that a still frame is shown instead of a blank screen.

NOTE

To change one of the timecode values (main and offset timecode displays), double-click on any of the timecode fields and enter a new value.

Synchronizing a Device to Your Host

PROCEDURE

1. Use the **SMPTEGenerator** as an insert effect on an audio track, and route that track to a separate output.

   Make sure that no other insert or send effect is used on this track. Deactivate any EQ settings for this track.

2. Connect the corresponding output on the audio hardware to the timecode input on the device that you want to synchronize to your host application.

   Make all necessary settings for the external device so that it synchronizes to incoming timecode.

3. Optional: Adjust the level of the timecode, either in your host application or in the receiving device.

   Activate the **Generate Code** button (make the device send the SMPTE timecode in free run mode) to test the level.

4. Make sure that the frame rate in the receiving device matches the frame rate set in the **SMPTEGenerator**.
5. Activate the **Link to Transport** button.

   The **SMPTEGenerator** now outputs timecode that corresponds to the time display of your host application.

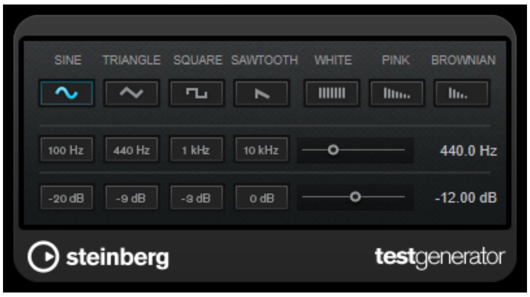
6. On the Transport panel, click **Play**.

RESULT

The external device is now synchronized and follows any position changes set with the transport controls.

TestGenerator

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



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

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

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

Waveform and noise section

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

Frequency section

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

NOTE

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

Gain section

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

Tuner

This is a guitar tuner.

To tune your instrument, connect it to an audio input, select **Tuner** as an insert effect, and activate **Monitor** for the corresponding track. Click **Mute** if you want to mute the output while tuning your instrument.

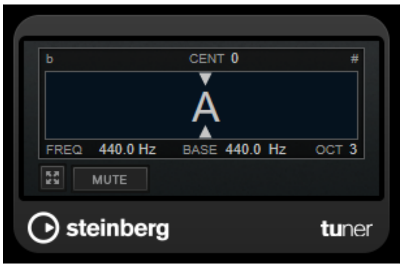
NOTE

Make sure that you deactivate any other effect that alters the pitch, such as a chorus or vibrato.

Tuner offers two different viewing modes, analog view and digital view.

- To toggle between analog view and digital view, click the **Toggle between Analog View and Digital View** button.

Analog View



The graphical display indicates the currently played pitch as a note. The two arrows indicate any pitch deviation. The deviation is also shown in the upper area of the display. If the played note is flat of the pitch, the pitch indicator is located to the left. If the played note is sharp, the pitch indicator is located to the right.

Cent

Shows the deviation in pitch. A negative value indicates that the pitch is flat. A positive value indicates that the pitch is sharp.

Frequency

Shows the frequency of the played note.

Base

Shows the frequency of the base note A. Its default value is 440 Hz. You can adjust **Base** by ± 15 Hz.

Octave

Shows the octave of the played note.

Mute

Mutes/Unmutes the output signal.

Digital View

This view provides two tuner modes: **Strobe** and **Classic**.

In **Strobe** mode, a colored moving strobe indicates any pitch deviation. If the played note is flat, the strobe moves from right to left. If the played note is sharp, the strobe moves from left to right. The higher the deviation in pitch, the faster the strobe moves. If you play the correct pitch, the strobe stops moving and turns gray.

In **Classic** mode, an indicator shows any pitch deviation. If the played note is flat, the indicator is located left of the middle. If the played note is sharp, the indicator is located right of the middle. If you play the correct pitch, the indicator is located in the middle and turns gray.

Note

Shows the currently played pitch.

Cent

Shows the deviation in pitch. A negative value indicates that the pitch is flat. A positive value indicates that the pitch is sharp.

Base

Shows the frequency of the base note A. Its default value is 440 Hz. You can adjust **Base** by ± 15 Hz.

Octave

Shows the octave of the played note.

Frequency

Shows the frequency of the played note.

Mute

Mutes/Unmutes the output signal.

Strobe/Classic

Allows you to toggle the display between **Strobe** and **Classic** mode.

MIDI Effects

This chapter describes the included MIDI realtime effects and their parameters. How to apply and handle MIDI effects is described in the **Operation Manual**.

Arpache 5

A typical arpeggiator accepts a chord as input, and plays back each note in the chord separately, with the playback order and speed set by the user.

Play Order buttons

Allow you to select the playback order for the arpeggiated notes. If you select **User**, you can set the playback order manually using the 12 Play Order slots that are now shown at the bottom of the dialog.

Step Size

Determines the speed of the arpeggio, as a note value related to the project tempo. For example, setting this to **16** means the arpeggio is a pattern of 16 notes.

Length

Sets the length of the arpeggio notes, as a note value related to the project tempo.

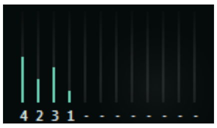
- To create staccato arpeggios, set a smaller value for **Length** than for **Step Size**.
- To create arpeggio notes that overlap each other, set a greater value for **Length** than for **Step Size**.

Key Range

Determines the arpeggiated note range, in semitones counted from the lowest key you play. This works as follows:

- Any notes you play that are outside this range are transposed in octave steps to fit within the range.
- If the range is more than one octave, octave-transposed copies of the notes you play are added to the arpeggio (as many octaves as fit within the range).

Play Order slots



If the **User** play order is selected, you can use these slots to specify a custom playback order for the arpeggio notes: Each of the 12 slots corresponds to a position in the arpeggio pattern. For each slot, you specify which note should be played on that position by selecting a number. The numbers correspond to the keys you play, counted from the lowest key.

For example, if you play the notes C3-E3-G3 (a C major chord), 1 means C3, 2 means E3, and 3 means G3.

NOTE

You can use the same number in several slots, creating arpeggio patterns that are not possible using the standard play modes. You need to begin with the leftmost slot and then fill the slots to the right.

MIDI Thru

If this button is activated, the notes that you play pass through the plug-in and are sent out together with the arpeggiated notes.

Creating an Arpeggio

PROCEDURE

1. Select a MIDI track and activate monitoring (or record enable it) so that you can play through the track.

   Make sure that the track is properly set up for playback to a suitable MIDI instrument.
2. Select the arpeggiator as an insert effect for the track.
3. Activate the arpeggiator.
4. On the arpeggiator panel, use the **Step Size** setting to set the arpeggio speed.
5. Use the **Length** setting to set the length of the arpeggio notes.
6. Set the **Key Range** parameter to 12.

   This makes the notes arpeggiate within an octave.
7. Play a chord on your MIDI instrument.

   Now, instead of hearing the chord, you hear the notes of the chord played one by one, in an arpeggio.
8. Try the different arpeggio modes by clicking the **Play Order** buttons.

   The symbols on the buttons indicate the playback order for the notes.

Arpache SX

This is a versatile and advanced arpeggiator, capable of creating anything from traditional arpeggios to complex, sequencer-like patterns.

Classic vs. Sequence Mode

Arpache SX has two different modes: **Classic** and **Sequence**. The **Classic** mode determines the basic behavior of Arpache SX. **Sequence** mode uses the events of an additional MIDI part as a pattern. This pattern forms the basis for the arpeggio, in conjunction with the MIDI input.

Classic Mode

Direction

Allows you to choose how the notes in the played chord should be arpeggiated.

One Shot Mode

Activate this option if you want the phrase to be played only once. If this option is deactivated, the phrase is looped.

Transpose

With a setting other than **Off**, the arpeggio is expanded upwards, downwards, or both (depending on the mode). This is done by adding transposed repeats of the basic arpeggio pattern.

Repeats

Sets the number of transposed repeats.

Pitch Shift

Determines the transposition of each repeat.

MIDI Thru

If this is activated, the played notes pass through the plug-in and are sent out together with the arpeggiated notes.

Step Size

Determines the resolution of the arpeggio, that is, its speed (in fixed note values or PPQ, if the **PPQ** button is activated). In Sequence mode you can also activate the **from sequence** option.

Length

Determines the length of the arpeggio notes (in fixed note values or PPQ, if the **PPQ** button is activated).

Max. Polyphony

Determines how many notes should be accepted in the input chord. The **All** setting means there are no limitations.

Sort by

If you play a chord, the arpeggiator sorts the notes in the chord in the order specified here. For example, if you play a C-E-G chord, with **Note Lowest** selected, C is the first note, E is the second and G the third. This affects the result of the **Arp Style** setting.

Velocity

Determines the velocity of the notes in the arpeggio. Using the slider you can set a fixed velocity, or you can activate the **via Input** button to use the velocity values of the notes in the chord you play. In Sequence mode you can also activate the **from sequence** option.

Sequence Mode

In Sequence mode, you can import a MIDI part into Arpache SX by dragging it from the Project window onto the **Drop MIDI Sequence** field on the right of the Arpache SX panel.

The notes in the dropped MIDI part are sorted internally, either according to their pitch if the **MIDI Seq. sort by pitch** checkbox is activated or according to their play order in the part. This results in a list of numbers. For example, if the notes in the MIDI part are C E G A E C and they are sorted according to pitch, the list of numbers reads 1 2 3 4 2 1. Here, there are 4 different notes/ numbers and 6 trigger positions.

The MIDI input (the chord you play) generates a list of numbers, with each note in the chord corresponding to a number depending on the **Sort by** setting.

Furthermore, the two lists of numbers are matched – Arpache SX tries to play back the pattern from the dropped MIDI part but using the notes from the MIDI input. The result depends on the **Play Mode** setting.

Trigger

The whole pattern from the dropped MIDI file is played back, but transposed according to one of the notes in the MIDI input. Which note is used for transposing depends on the **Sort by** setting.

Trigger Cnt.

As above, but even if all keys are released, the phrase continues playing from the last position (where it stopped), if a new key is pressed on the keyboard. This is typically used when playing live through the Arpache SX.

Sort Normal

Matches the notes in the MIDI input with the notes in the dropped MIDI part. If there are fewer notes in the MIDI input, some steps in the resulting arpeggio remain empty.

Sort First

As above, but if there are fewer notes in the MIDI input, the missing notes are replaced by the first note.

Sort Any

As above, but if there are fewer notes in the MIDI input, the missing notes are replaced by random notes.

Arp. Style

As above, but if there are fewer notes in the MIDI input, the missing notes are replaced by the last valid note in the arpeggio.

Repeat

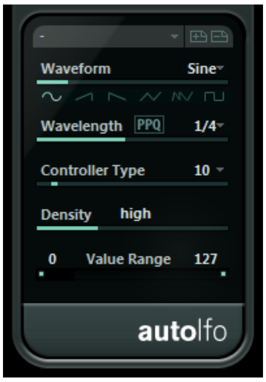
In this mode, the chords played are not separated into notes. Instead, they are used as is, and only the rhythm of the dropped MIDI part is used for playback.

NOTE

You can choose to keep the original note timing, note length, and note velocities from the dropped MIDI part, by selecting **from sequence** for the **Step Size**, **Length**, and **Velocity** options.

Auto LFO

This effect works like an LFO in a synthesizer, allowing you to send out continuously changing MIDI controller messages. One typical use for this is automatic MIDI panning, but you can select any MIDI continuous controller event type.



Waveform

Determines the shape of the controller curves that are sent out. You can click a waveform symbol or choose a value from the pop-up menu.

Wavelength

Sets the speed of Auto LFO, or rather the length of a single controller curve cycle. You can set this to rhythmically exact note values or PPQ values if the **PPQ** button is activated. The lower the note value, the slower the speed.

Controller Type

Determines which continuous controller type is sent out. Typical choices would include pan, volume, and brightness, but your MIDI instrument may have controllers mapped to various settings, allowing you to modulate the synth parameter of your choice. Check the MIDI implementation chart for your instrument for details.

Density

Determines the density of the controller curves that are sent out. The value can be set to **small**, **medium**, or **large**, or to rhythmically exact note values. The higher the note value, the smoother the controller curve.

Value Range

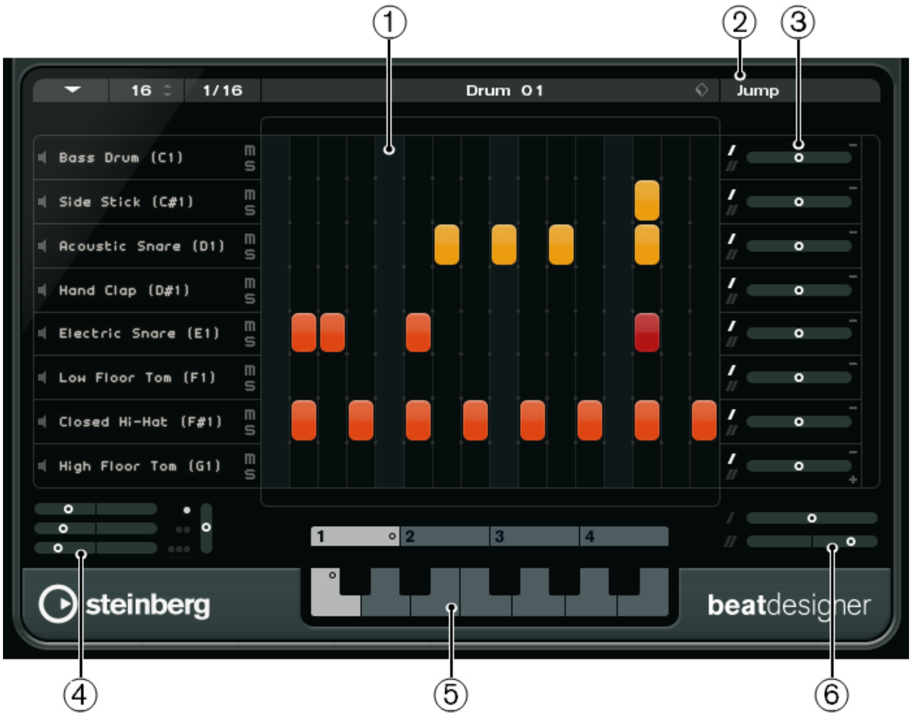
These sliders determine the range of controller values that are sent out, in other words, the bottom and top of the controller curves.

Beat Designer

Beat Designer is a MIDI pattern sequencer that allows you to create your own drum parts or patterns for a project. With **Beat Designer**, you can quickly and easily set up the drums for a project, by experimenting and creating new drum sequences from scratch.

Normally, you work on a short sequence, adjusting and modifying it while playing it back in a loop. The drum patterns can then either be converted to MIDI parts on a track or triggered using MIDI notes during playback.

Control Panel



1 Step display

2 Jump mode

3 Swing and Offset controls

4 Flam position settings

5 Pattern display

6 Swing settings

Patterns and Subbanks

Beat Designer patterns are saved as pattern banks. One pattern bank contains 4 subbanks which in turn contain 12 patterns each.

In the pattern display in the lower part of **Beat Designer**, subbanks and patterns are displayed graphically. To select a subbank, click a number (1 to 4) at the top of the display. To select a pattern within this subbank, click a key in the keyboard display below.

Initial Settings

The steps represent the beat positions in the pattern. You can specify the number of steps and the step resolution globally for a pattern.

- Click in the **Number of steps for this pattern** value field and enter a value. The maximum number of steps is 64.
- The playback length, that is, the note value for the steps, can be specified on the **Step Resolution** pop-up menu. On this menu, you can also set triplet values. These also affect the **Swing** setting.

RELATED LINKS
Triggering Patterns on page 198
Swing Setting on page 195

Selecting Drum Sounds

PROCEDURE

1. Click in the drum name field for a lane and select a drum sound from the pop-up menu.
   The available drum sounds depend on the selected drum map. If no drum map is selected for the track, the GM (General MIDI) drum names are used.
2. To find the right sound, audition the selected drum sound by clicking the **Preview Instrument** button (the speaker icon).

Entering Drum Steps

PREREQUISITE

When working on drum patterns, it is a good idea to play back a section of the project in a loop while inserting the drum sounds. This allows you to hear the result immediately.

PROCEDURE

- Enter a drum step by clicking on the step field where you want to add a beat.
  For example, add a snare drum on each downbeat for a lane and a bass drum on a second lane.

  NOTE

  You can also click and drag to enter a continuous range of drum steps.

Removing Steps

PROCEDURE

- To remove a drum step, simply click on the corresponding field again.

  NOTE

  To remove a range of drum steps, click and drag over them.

Velocity Settings

When entering a drum step, the velocity setting of this step is determined by where you click: Click in the upper part of a step for the highest velocity setting, in the middle section for a medium velocity and in the lower part for the lowest velocity setting. In the display, the different velocity settings are indicated by different colors.

- To fine-tune the velocity setting for an existing drum step, click on it and drag up or down. The current velocity is indicated numerically while you drag.
- To fine-tune the velocity for a range of drum steps, click on the first step, drag up or down to enter velocity edit mode, and then drag sideways and up or down to modify the velocity for all the steps.

  If you change the velocity for several steps at the same time, the relative velocity differences are kept for as long as possible (until the minimum or maximum setting is reached). The velocity for the steps is increased or decreased by the same amount.

  NOTE

  If you hold down **Shift** while dragging up or down, you can change the velocity for all steps on a lane.
- To create a crescendo or decrescendo for an existing range of drum steps, hold down **Alt/Opt**, click on the first step, drag up or down, and then drag to the left or right.

Editing Operations

- To move all drum steps on a lane, hold down **Shift**, click on the lane, and drag to the left or right.
- To invert a lane, that is, add drum sounds for all steps that were empty while removing all existing drum steps, hold down **Alt/Opt** and drag the mouse over the lane. This lets you create unusual rhythmic patterns.
- To copy the content of a lane onto another lane, hold down **Alt/Opt**, click in the section to the left of the lane that you want to copy, and drag.

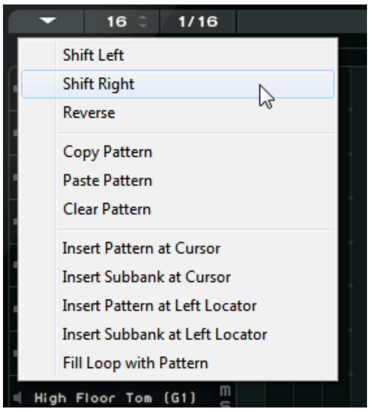
Lane Handling

- To add a lane, click the **Add Instrument Lane** button at the bottom right of the last lane.
- To remove a lane, click the **Remove Instrument Lane** button in the controls section at the far right of the lane.
- To change the order of the drum lanes, click in an empty area in the section to the left of a lane, and drag it to another position.
- To mute or solo a lane, click the corresponding buttons to the left of the step display.

IMPORTANT

Lane operations always affect all patterns in the **Beat Designer** instance.

Pattern Functions Menu



Shift Left

Moves all steps of the current pattern on all lanes to the left.

Shift Right

Moves all steps of the current pattern on all lanes to the right.

Reverse

Reverses the pattern, so that it plays backwards.

Copy Pattern

Copies the pattern to the clipboard. Copied patterns can be pasted into another pattern subbank and even directly into the project.

Paste Pattern

Allows you to paste a complete pattern, for example, into another pattern subbank, or into another instance of **Beat Designer**. This is useful if you want to create variations based on existing patterns.

Clear Pattern

Resets the current pattern.

Insert Pattern at Cursor

Creates a MIDI part for the current pattern and inserts it in the **Project** window, at the position of the project cursor.

Insert Subbank at Cursor

Creates a MIDI part for each used pattern in the subbank and inserts the parts one after the other, starting at the project cursor.

Insert Pattern at Left Locator

Creates a MIDI part for the current pattern and inserts it in the **Project** window, at the left locator.

Insert Subbank at Left Locator

Creates a MIDI part for each used pattern in the subbank and inserts the parts one after the other, starting at the left locator.

Fill Loop with Pattern

Creates a MIDI part for the current pattern and inserts it in the **Project** window as often as needed to fill the current loop area.

NOTE

In the **Key Commands** dialog, you can set up key commands for the **Insert** options and the **Fill Loop** command. How to set up and use key commands is described in the **Operation Manual**.

RELATED LINKS

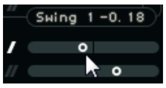
Converting Patterns into MIDI Parts on page 197

Swing Setting

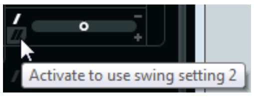
This parameter can be used to create a swing or shuffle rhythm. This adds a more human feel to drum patterns that might otherwise be too static.

Swing offsets every second drum step for a lane. If a triplet step resolution is used, every third drum step is offset instead.

In the lower right section of the **Beat Designer** panel, you can find two **Swing** sliders. You can set up two swing settings with these sliders and then quickly switch between these during playback.



- To delay every second or third drum step in the pattern, drag a slider to the right.
- To make a drum step play earlier in the pattern, drag a slider to the left.
- To switch between the swing settings, click the **Swing** buttons to the right of the step display.

- To deactivate swing for a lane, click on the selected **Swing** buttons.

Flams

The **Flam** parameter lets you add flams, that is, short secondary drum hits just before or after the actual main drum beat. You can add up to three flams for each pattern step.

In the lower left section of the **Beat Designer** panel you can make settings for the flams you created.

The first position slider specifies the flam position for all steps containing one single flam, the second slider the flam positions for all steps containing two flams, and the third slider the flam position for all steps containing three flams.

Adding Flams

PROCEDURE

1. Click in the lower left corner of the step you want to add a flam to.

   Little squares appear in the step when you point the mouse at it. If you click a step, the first square is filled to indicate that you added a flam.

2. Click again to add the second and third flam.
3. In the lower left section of the **Beat Designer** panel, make settings for the flams that you created.
   - To add the flams before or after the drum step, drag a position slider to the left or right.

     If you add flams before the first drum step in a pattern, this is indicated in the display by a small arrow in the top left corner of this step. Starting playback at the normal pattern start would result in these flams not being played.
   - To set the velocity for the flams, use the vertical sliders to the right of the flam sliders.
4. Start playback to hear the flams you created.

Offsetting Lanes

To the right of the step display, you can find the **Offset** sliders for the lanes. These allow you to offset all drum steps on this lane.

PROCEDURE

1. Drag a slider to the left to make the drum steps start a little earlier and to the right to let them start later.

   For example, playing the bass drum or snare a little earlier allows you to add more urgency to the drums, while delaying these drum sounds results in a more relaxed drum pattern.

2. Experiment with the settings to find out which fit best in your project.

   NOTE

   This function can also be used to correct faulty drum samples: If a drum sound has an attack that is slightly late, simply adjust the **Offset** slider for the lane.

Using the Drum Patterns in Your Project

Converting Patterns into MIDI Parts

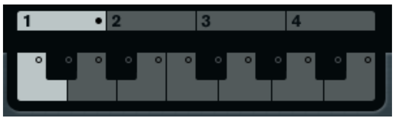
You can convert the drum patterns created in **Beat Designer** into a MIDI part by dragging them into the **Project** window.

PROCEDURE

1. Set up one or more patterns of the same subbank.

2. In the lower part of the window, click on a pattern or subbank and drag it onto a MIDI or instrument track in the **Project** window.

   - If you drag the pattern or subbank to an empty area in the **Project** window, a new MIDI track is created. This is an exact copy of the original track for which you opened **Beat Designer**.

   - If you drag a single pattern into the **Project** window, one MIDI part is created containing the drum sounds of the pattern.

   - If you drag a subbank into the **Project** window, several MIDI parts (one for each used pattern in the subbank) are created and inserted one after the other in the project.

   IMPORTANT

   Only the used patterns in a subbank are inserted. If you did not enter drum steps in a pattern, this is not converted into a MIDI part.

   You can also use the **Pattern Functions** menu to insert patterns or subbanks into the project.

   IMPORTANT

   When you have created MIDI parts for your drum patterns this way, make sure to deactivate **Beat Designer**, to avoid doubling of the drums. **Beat Designer** continues to play as long as it is activated.

   - If you import patterns that sound before the first step (due to flams or lane offsets), the MIDI part is lengthened accordingly.

   The inserted MIDI parts can now be edited as usual in the project. For example, you can fine-tune your settings in the **Drum Editor**.

NOTE

Once a pattern is converted into a MIDI part, it cannot be opened in **Beat Designer** again.

RELATED LINKS

Pattern Functions Menu on page 194

Triggering Patterns

If you want to modify your drum patterns in **Beat Designer** while working on the project, you can trigger the patterns from within the project.

You can trigger the patterns in **Beat Designer** using note-on events. These can either be events on a MIDI track or be played live via a MIDI keyboard. Which pattern is triggered depends on the pitch of the MIDI notes. The trigger range is four octaves starting with C1 (that is, C1 to B4).

PROCEDURE

1. Open **Beat Designer** for a track.
2. Activate **Jump**.

   In this mode, a MIDI note-on event triggers a new pattern.

   - To trigger the patterns using a MIDI part containing trigger events, you can specify whether the pattern is switched instantly (at the moment the event is received) or at the next bar: Activate **Now** to switch patterns immediately. If **Now** is deactivated, patterns switch at the beginning of the next bar in the project.
   - If you want to trigger the patterns live via a MIDI keyboard, the new patterns are always played when the next bar in the project is reached. Switching immediately would always produce an undesirable interruption in playback.
3. Play back the project and press a key on your MIDI keyboard to trigger the next pattern.

   The pattern starts at the next bar line.
4. Create a MIDI part and enter notes at the positions in the project where you want to switch patterns.

   Depending on the **Jump** mode setting, the new pattern is played instantly, or starts at the following bar.

   - You can also drag a pattern or subbank into the project with **Jump** mode activated to automatically create MIDI parts containing the trigger events.

   NOTE

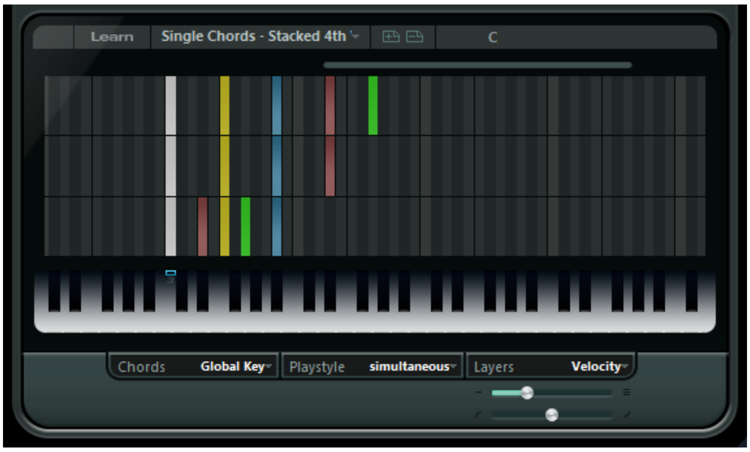
   When triggering a pattern that contains sound before the first step (due to flams or lane offsets), these are taken into account as well.

Chorder

Chorder is a MIDI chord processor, allowing you to assign complete chords to single keys in a multitude of variations. These can then be played back live or using recorded notes on a MIDI track.

There are three main operating modes: **All Keys**, **One Octave**, and **Global Key**. You can switch between these modes using the **Chords** pop-up menu.

For every key, you can record up to 8 different chords or variations on so-called layers.



Operating Modes

In the lower left section of the **Chorder** window, you can choose an option from the **Chords** pop-up menu to decide which keys in the keyboard display are used to record your chords.

All Keys

In this mode, you can assign chords to each key on the keyboard display. If you play any of these keys, you hear the assigned chords instead.

One Octave

This mode is similar to the **All Keys** mode, but you can only set up chords for each key of a single octave, that is, up to eight different chords on twelve keys. If you play a note in a different octave, you hear a transposed version of the chords set up for this key.

Global Key

In this mode, you can set up chords for a single key only. These chords (that you recorded on C3) are then played by all keys on the keyboard, but transposed according to the note you play.

Chord Indicator Lane

At the top of the keyboard display, you find a thin lane with a small rectangle for each key that you can use to record a chord. These rectangles are shown in blue for all keys that already have chords assigned to them.

NOTE

In **Global Key** mode, the C3 key has a special marking, because this is the only key used in this mode.

RELATED LINKS

Using Layers on page 200

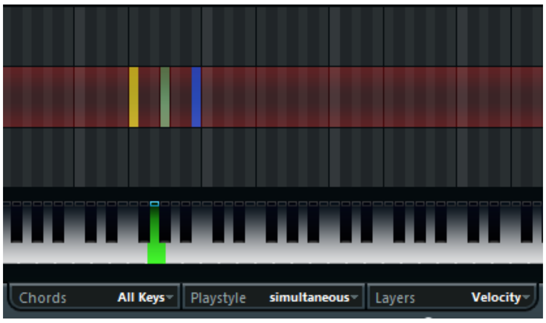
Entering Chords

PROCEDURE

1. Activate the **Learn** button at the top of the **Chorder** window to activate **Learn** mode.

The chord indicator lane is now tinted red, indicating that it is active.

The keyboard display in **Learn** mode



The second layer in **Learn** mode

2. Select the key to which you want to assign a chord by clicking on it on the keyboard display or by pressing the key on a connected MIDI keyboard.

   The red bar now moves to the first layer, indicating that you are ready to record the first chord.

   NOTE

   In **Global Key** mode, you do not have to choose a trigger key. The first layer is activated automatically.

3. Play a chord on the MIDI keyboard and/or use the mouse to enter or change the chord in the layer display.

   - Any notes you enter are immediately shown in the **Chorder** display. The notes are shown in different colors, depending on the pitch.
   - If you are entering chords via a MIDI keyboard, **Chorder** learns the chord as soon as you release all keys of your MIDI keyboard.

     As long as a key is pressed, you can continue looking for the right chord.
   - If more than one layer is shown, **Chorder** jumps automatically to the next layer where you can record another chord.

     If all layers for a key are filled, the red bar jumps back to the keyboard display so that you can choose a different trigger key (in **Global Key** mode, the **Learn** mode is deactivated).
   - If you are entering chords with the mouse, **Chorder** does not jump to the next layer automatically.

     You can select/deselect as many notes as you want and then click on another layer or deactivate the Learn mode to continue.

4. Repeat the above with any other keys you want to use.

Using Layers

The **Layers** pop-up menu at the bottom right of the window allows you to set up chord variations in the layer display above the keyboard. This works with all three modes and provides up to 8 variations for each assignable key, that is, a maximum of 8 different chords in **Global Key** mode, 12 x 8 chords in **One Octave** mode and 128 x 8 chords in **All Keys** mode.

The different layers can be triggered by velocity or interval.

PROCEDURE

1. On the **Layers** pop-up menu, select **Velocity** or **Interval**. Set this to **Single Mode** if you want to set up only one chord per key.
2. Use the slider below the **Layers** pop-up menu to specify how many variations you want to use.
3. Enter the chords.

RESULT

Now you can play the keyboard and trigger the variations according to the selected layer mode.

RELATED LINKS

Empty Layers on page 201

Layer Modes

You can play the keyboard and trigger the variations according to the selected layer mode.

Velocity

The full velocity range (1 to 127) is divided into zones, according to the number of layers you specified. For example, if you are using 2 variations, 2 velocity zones are used: 1 to 63 and 64 to 127. Playing a note with velocity 64 or higher triggers the second layer, while playing a softer note triggers the first layer.

With the **Velocity spread** slider at the bottom right of the window, you can change the velocity ranges of the layers.

Interval

In this mode, **Chorder** plays one chord at a time. If the **Interval** mode is selected, you trigger a layer by pressing 2 keys on your keyboard. The lower key determines the base note for the chord. The layer number is determined by the difference between the 2 keys. To select layer 1, press a key one semitone higher than the base note, for layer 2, press a key two semitones higher, and so on.

Single Mode

Select this if you want to use only 1 layer.

Empty Layers

If you enter fewer chords than layers are available for a key, these layers are filled automatically when you deactivate the **Learn** mode.

The following applies:

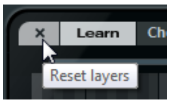
- Empty layers are filled from bottom to top.
- If there are empty layers below the first layer with a chord, these are filled from top to bottom.

An example: If you have a setup with 8 layers, and you enter the chord C in layer 3 and G7 in layer 7, you get the following result: chord C in layers 1 to 6 and G7 in layers 7 and 8.

Resetting Layers

PROCEDURE

- In **Learn** mode, click **Reset layers** at the top left of the **Chorder** window.



RESULT

For the selected trigger key, all notes in the different layers are deleted.

Playstyle

From the **Playstyle** pop-up menu at the bottom of the panel, you can choose one of seven different styles that determine in which order the individual notes of the chords are played back.

simultaneous
In this mode, all notes are played back simultaneously.

fast up
In this mode, a small arpeggio is added, starting with the lowest note.

slow up
Similar to **fast up**, but using a slower arpeggio.

fast down
Similar to **fast up**, but starting with the highest note.

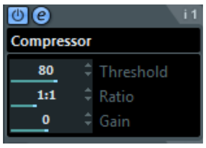
slow down
Similar to **slow up**, but starting with the highest note.

fast random
In this mode, the notes are played back in a rapidly changing random order.

slow random
Similar to **fast random**, but the note changes occur more slowly.

Compressor

This MIDI compressor is used for evening out or expanding differences in velocity.



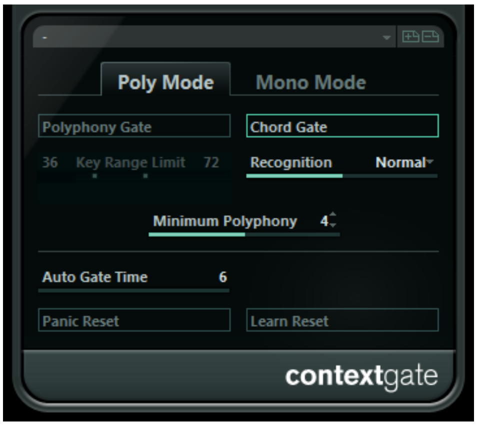
Threshold
Only notes with velocities above this value are affected by the compression/expansion.

Ratio
Sets the amount of compression applied to the velocity values above the set threshold. Ratios greater than 1:1 result in compression. Ratios lower than 1:1 result in expansion.

Gain
Adds or subtracts a fixed value from the velocities. Since the maximum range for velocity values is 0 to 127, you can use the **Gain** setting to compensate, keeping the resulting velocities within the range. Typically, negative **Gain** settings are used for expansion and positive settings for compression.

Context Gate

Context Gate allows for selective triggering/filtering of MIDI data.



This effect features two modes: In **Poly Mode**, **Context Gate** recognizes certain chords that are played. In **Mono Mode**, only certain MIDI notes are let through.

Poly Mode

Polyphony Gate

Allows you to filter MIDI according to the number of pressed keys within a given key range. This can be used independently or in conjunction with the **Chord Gate** function.

- The **Key Range Limit** sliders set the key range. Only notes within this range are let through.
- The **Minimum Polyphony** value field allows you to specify the minimum number of notes required to open the gate.

Chord Gate

If **Chord Gate** is activated, only notes in recognized chords are let through. Two **Recognition** modes are available: **Simple** and **Normal**.

- In **Simple** mode, all standard chords (major/minor/b5/dim/sus/maj7 etc.) are recognized.
- **Normal** mode takes more tensions into account.

Mono Mode

Channel Gate

If this is activated, only single note events of the specified MIDI channel are let through. This can be used with MIDI controllers that can send MIDI on several channels simultaneously, for example, guitar controllers which send data for each string over a separate channel.

- You can set **Mono Channel** to a specific channel (**1** to **16**), or to **Any**, that is, no channel gating.

Velocity Gate

This can be used independently or in conjunction with the **Channel Gate** function. Notes are played back until another note within the set range is played.

- The **Key Range Limit** sliders set the key range. Only notes within this range are let through.
- Notes below the **Minimum Velocity** threshold value are gated.

Auto Gate Time

If there is no input activity, you can specify the time, after which note-off messages are sent for the notes that are playing.

Panic Reset

Sends an “All Notes Off” message over all channels, in case of hanging notes.

Learn Reset

If this is activated, you can specify a reset trigger event via MIDI. Whenever this specific MIDI event is sent, it triggers an “All Notes Off” message. When you have set the reset event, deactivate the **Learn Reset** button.

RELATED LINKS
Application Examples on page 204

Application Examples

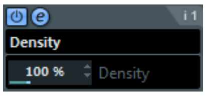
Poly Mode

In this mode, you can use **Context Gate** to accompany yourself during a live guitar performance using a VST instrument. To do this, you might use a guitar-to-MIDI converter: You could then program **Context Gate**, for example, to allow only those notes to pass the gate that are part of a four-note chord. During your performance you would then play a four-note chord every time that you want to trigger the VST instrument. The instrument plays until the **Auto Gate Time** is reached and fades out. For more complex performances this can be combined with an arpeggiator, without having to use external pedals to trigger the effect.

Mono Mode

In this mode you could use **Context Gate** to trigger variations played with a drum machine/VST instrument. To do this, you need a guitar-to-MIDI converter: You could then filter the MIDI channel using the Input Transformer (optional) and program the **Context Gate** to allow only certain notes on your guitar to pass the gate (for example, beginning at the 12th band). When you now play one of these notes, the note-off command is not send out and the corresponding note sounds until the note is played again, a new note is let through, or the **Auto Gate Time** is reached. This way you can trigger lots of different effects or notes using the high notes on you guitar without having to use an additional MIDI instrument.

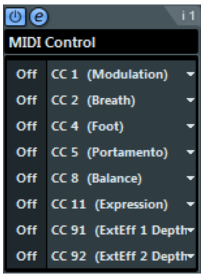
Density



This generic control panel affects the density of the notes being played from or through the track. If this is set to 100 %, the notes are not affected. Density settings below 100 % randomly filter out or mute notes. Settings above 100 % randomly add notes that were played before.

MIDI Control

This generic control panel allows you to select up to 8 different MIDI controller types and set values for these. You can then use the plug-in as a control panel to adjust the sound of a MIDI instrument from within your host application.



- To select a controller type, use the pop-up menus on the right.
- To change the value of a controller type, enter a value in the value field or click the value field and drag the cursor up or down.
- To deactivate a controller, type **Off** in the value field or click the value field and drag the cursor down until the value field displays **Off**.

MIDI Echo

This is an advanced MIDI echo, which generates additional echoing notes based on the MIDI notes it receives. It creates effects similar to a digital delay, but also features MIDI pitch shifting and much more.

The effect does not echo the actual audio, but the MIDI notes which eventually produce the sound in the synthesizer.

Velocity Offset

Allows you to raise or lower the velocity values for each repeat so that the echo fades away or increases in volume (provided that the sound you use is velocity sensitive).

Pitch Offset

If you set this to a value other than 0, the echoing notes are raised or lowered in pitch, so that each successive note has a higher or lower pitch than the previous. The value is set in semitones.

For example, setting this to -2 causes the first echo note to have a pitch two semitones lower than the original note, the second echo note two semitones lower than the first echo note, and so on.

Repeats

The number of echoes (1 to 12) of each incoming note.

Beat Align

During playback, this parameter quantizes the position of the first echo note. You can either set this to rhythmically exact values or activate the **PPQ** button and choose a **PPQ** value.

Setting this to 1/8, for example, causes the first echo note to sound on the first eighth position after the original note.

NOTE

The echo time can also be affected by the **Delay Decay** parameter.

NOTE

During live mode, this parameter has no effect since the first echo is always played together with the note event itself.

Delay

The echoed notes are repeated according to this value. You can either set this to rhythmically exact values or activate the **PPQ** button and choose a PPQ value. This makes it easy to find rhythmically relevant delay values, but still allows for experimental settings in between.

Delay Decay

Adjusts how the echo time changes with each successive repeat. The value is set as a percentage.

- If this is set to 100 % the echo time is the same for all repeats.
- If you raise the value above 100 %, the echoing notes play with gradually longer intervals, that is, the echo becomes slower.
- If you lower the value below 100 %, the echoing notes become gradually faster, like the sound of a bouncing ball.

Length

Sets the length of the echoed notes. This can either be identical with the length of the original notes (parameter set to its lowest value) or the length you specify manually. You can either set this to rhythmically exact values (displayed as note values – see the table below) or activate the **PPQ** button and choose a PPQ value.

NOTE

The length can also be affected by the **Length Decay** parameter.

Length Decay

Adjusts how the length of the echoed notes changes with each successive repeat. The higher the setting, the longer the echoed notes.

About Ticks and Note Values

The timing and position-related parameters (**Delay**, **Length**, and **Beat Align**) can all be set in ticks. There are 480 ticks to each quarter note. The parameters allow you to step between the rhythmically relevant values. The following table shows the most common note values and the corresponding number of ticks.

Note Value	Ticks
1/32 note	60

Note Value	Ticks
1/16 note triplet	90
1/16 note	120
1/8 note triplet	160
1/8 note	240
Quarter note triplet	320
Quarter note	480
Half note	960

MIDI Modifiers

This plug-in is essentially a duplicate of the **MIDI Modifiers** section in the **Inspector**. This can be useful, for example, if you need extra **Random** or **Range** settings.

The **MIDI Modifiers** effect also includes the **Scale Transpose** function that is not available among the track parameters.

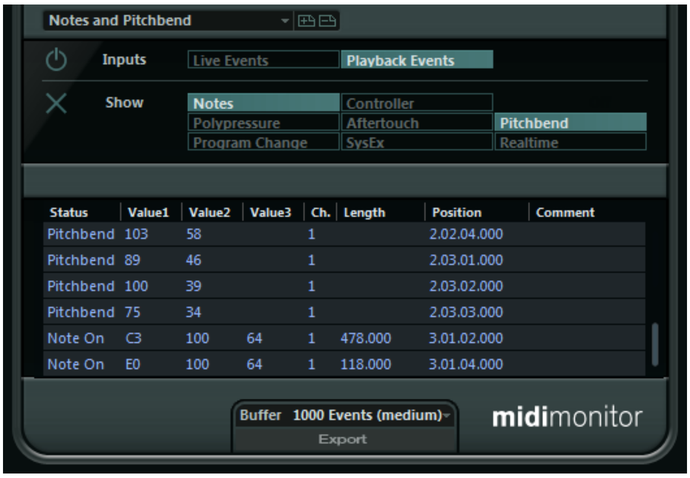
Scale Transpose

Allows you to transpose each incoming MIDI note, so that it fits within a selected musical scale. The scale is specified by selecting a key (C, C#, D, etc.) and a scale type (major, melodic or harmonic minor, blues, etc.).

- To deactivate **Scale Transpose**, select **No Scale** from the **Scale** pop-up menu.

MIDI Monitor

This effect monitors incoming MIDI events.



You can choose whether to analyze live or playback events and which types of MIDI data are to be monitored. Use this, for example, to analyze which MIDI events are generated by a MIDI track, or to find suspicious events, such as notes with velocity 0 that certain MIDI devices might fail to interpreted as note-off events.

Inputs Section

In this section, you can choose whether to monitor live events or playback events.

Show Section

Here, you can activate/deactivate the different types of MIDI events. If you choose **Controller**, you can also define which type of controller to monitor.

Data Table

In the table in the lower section of the window, you see detailed information about the monitored MIDI events.

Buffer Pop-up Menu

This is the maximum number of events that is kept in the list of monitored events. Once this list is full, the oldest entries are deleted when new events are received.

NOTE

The larger the buffer, the more processing resources are required.

Export

Allows you to export the monitoring data as a simple text file.

Record Events

This button to the left of the **Inputs** section allows you to start or stop the monitoring of MIDI events.

Clear List

This button to the left of the **Show** section allows you to clear the table of recorded MIDI events.

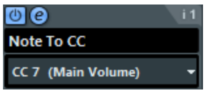
Micro Tuner

Micro Tuner lets you set up a different microtuning scheme for the instrument, by detuning each key.

- Each detune slider corresponds to a key in an octave (as indicated by the keyboard display). Adjust a detune field to raise or lower the tuning of that key, in cents (hundreds of a semitone).
- You can set the root note that is taken as a reference for the detuning.
- You can adjust all keys by the same amount by keeping **Alt/Opt** pressed.

Micro Tuner comes with a number of presets, including both classical and experimental microtuning scales.

Note to CC



This effect generates a MIDI continuous controller event for each incoming MIDI note. The value of the controller event corresponds to the velocity of the MIDI note, which is then used to control the selected MIDI controller (by default CC 7, Main Volume). For each note end, another controller event with the value 0 is sent. The incoming MIDI notes pass through the effect unaffected.

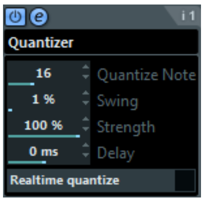
The purpose of this plug-in is to generate a gate effect. This means that the notes that are played control something else. For example, if **Main Volume** (CC 7) is selected, notes with low velocity lower the volume in the MIDI instrument, while notes with a high velocity raise the volume.

IMPORTANT

A controller event is sent out each time a new note is played. If high and low notes are played simultaneously, this may lead to confusing results. Therefore, the **Note to CC** effect is best applied to monophonic tracks.

Quantizer

This effect allows you to apply quantizing in realtime. This makes it easier to try out different settings when creating grooves and rhythms.



Quantizing is a function that changes the timing of notes by moving them towards a quantize grid. For example, this grid may consist of straight sixteenth notes, in which case the notes all get perfect sixteenth note timing.

NOTE

The main **Quantize** function in your Steinberg DAW is described in the **Operation Manual**.

Quantize Note

Sets the note value on which the quantize grid is based. Straight notes, triplets. and dotted notes are available. For example, 16 means straight sixteenth notes and 8T means eighth note triplets.

Swing

Allows you to offset every second position in the grid, creating a swing or shuffle feel. The value is a percentage – the higher you set this, the farther to the right every even grid position is moved.

Strength

Determines how close the notes should be moved to the quantize grid. If this is set to 100 %, all notes are forced to the closest grid position. Lowering the setting gradually loosens the timing.

Delay

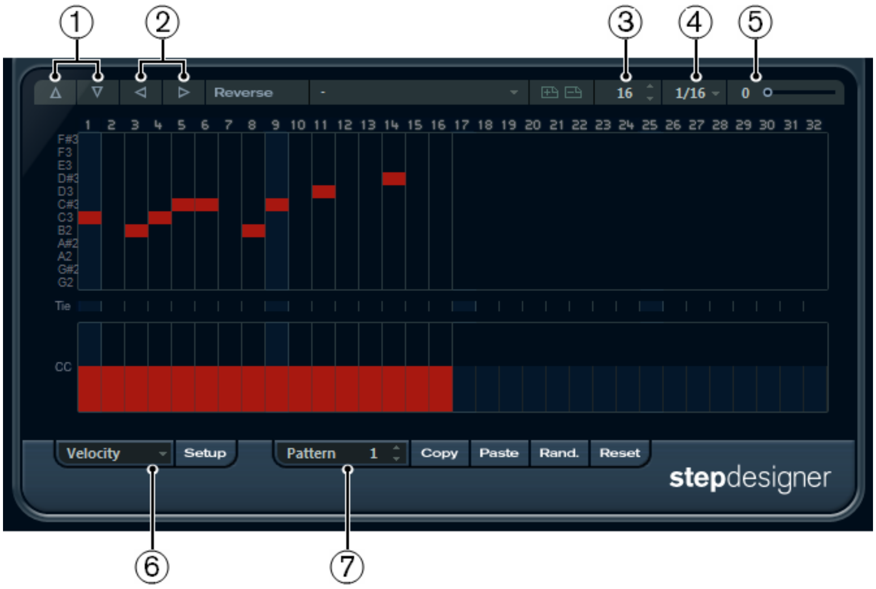
Sets the delay time in milliseconds. This delay can be automated.

Realtime quantize

During live mode, this option can be used to change the timing of the notes that are played so that they fit the quantize grid.

StepDesigner

StepDesigner is a MIDI pattern sequencer that sends out MIDI notes and additional controller data according to the defined pattern. It does not make use of the incoming MIDI, other than automation data (such as recorded pattern changes).



1 Shift octave up/down
2 Shift steps left/right
3 Number of steps
4 Step size
5 Swing
6 Controller pop-up menu
7 Pattern selector

Creating a Basic Pattern

PROCEDURE

1. Use the **Pattern** selector to choose which pattern to create.

   Each **StepDesigner** can hold up to 200 different patterns.

2. Use the **Step size** setting to specify the resolution of the pattern.

   This setting determines the step length.

3. Specify the number of steps in the pattern with the **Number of steps** setting.

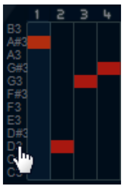
   The maximum number of steps is 32. For example, setting **Step size** to 16 and **Number of steps** to 32 creates a 2 bar pattern with sixteenth note steps.

4. Click in the note display to insert notes.

You can insert notes on any of the 32 steps, but **StepDesigner** only plays back the number of steps set with the **Step size** parameter.

- The display spans one octave (as indicated by the pitch list to the left). You can scroll the displayed octave up or down by clicking in the pitch list and dragging up or down.

  This way you can insert notes at any pitch.

- To remove a note from the pattern, click on it again.

NOTE

Each step can contain one note only – **StepDesigner** is monophonic.

RESULT

If you now start playback in your host application, the pattern plays as well, sending out MIDI notes on the track's MIDI output and channel (or, if you activated **StepDesigner** as a send effect, on the MIDI output and channel selected for the send in the **Inspector**).

Adding Controller Curves

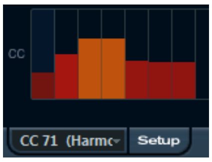
PROCEDURE

1. Open the **Controller** pop-up menu and select a controller.

   The selection is displayed in the lower controller display.

2. Click in the controller display to draw events.

   The MIDI controller events are sent out during playback along with the notes.

NOTE

If you drag a controller event bar all the way down, no controller value is sent out on that step.

Setting Up the Controller Menu

You can specify which two controller types (filter cutoff, resonance, volume, etc.) should be available on the **Controller** pop-up menu.

PROCEDURE

1. Click **Setup**.
2. Select the controllers that you want to have available in the **Controller** pop-up menu and click **OK**.

   This selection is global, that is, it applies to all patterns.

Adjusting the Step Length

- To make notes shorter, select **Gate** on the **Controller** pop-up menu and lower the bars in the controller display.

  If a bar is set to its maximum value, the corresponding note is the full length of the step.
- To make notes longer, you can tie two notes together. This is done by inserting two notes and clicking in the **Tie** column for the second note.

  If 2 notes are tied, the second note is not triggered – the previous note is lengthened instead. Also, the second note gets the same pitch as the first note. You can add more notes and tie them in the same way, creating longer notes.

Other Pattern Functions

Shift Octave up/down
Shifts the entire pattern up or down in octave steps.

Shift Steps left/right
Moves the pattern one step to the left or right.

Reverse
Reverses the pattern, so that it plays backwards.

Copy/Paste
Allow you to copy the current pattern and paste it in another pattern location (in the same **StepDesigner** instance or another).

Reset
Clears the pattern, removing all notes and resetting controller values.

Randomize
Generates a completely random pattern.

Swing
Offsets every second step, creating a swing or shuffle feel. The value is a percentage – the higher you set this, the farther to the right every even step is moved.

Presets
Allows you to load/save presets for the effect.

NOTE

A stored preset contains all 200 patterns in **StepDesigner**.

Automating Pattern Changes

You can create up to 200 different patterns in each **StepDesigner** instance.

Typically, you want the pattern selection to change during the project. You can accomplish this by automating the pattern selector, either in realtime by activating **Write** automation and switching patterns during playback or by drawing on the automation track for the MIDI track.

Note that you can also press a key on your MIDI keyboard to change patterns. For this, you have to set up **StepDesigner** as an insert effect for a record enabled MIDI track. Press C1 to select pattern 1, C#1 to select pattern 2, D1 to select pattern 3, D#1 to select pattern 4 and so on. You can record these pattern changes as note events on a MIDI track.

PROCEDURE

1. Select a MIDI track or create a new one and activate **StepDesigner** as an insert effect.

2. Set up several patterns.
3. Activate the **Record** button and press keys on your keyboard to select the corresponding patterns.

   The pattern changes are recorded on the MIDI track.
4. Stop recording and play back the MIDI track.

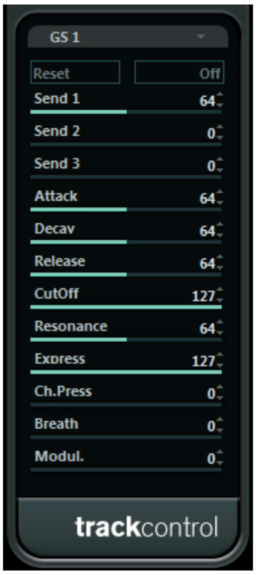
RESULT

You now hear the recorded pattern changes.

NOTE

You can only automate the first 92 patterns.

Track Control



The **Track Control** effect contains three control panels for adjusting parameters on a GS or XG compatible MIDI device. The Roland GS and Yamaha XG protocols are extensions of the General MIDI standard, allowing for more sounds and better control of various instrument settings. If your instrument is compatible with GS or XG, **Track Control** allows you to adjust sounds and effects in your instrument from within your host application.

The Available Control Panels

You select the control panel from the pop-up menu at the top of the effect panel. The following panels are available:

GS 1

Contains effect sends and various sound control parameters for use with instruments compatible with the Roland GS standard.

XG 1

Contains effect sends and various sound control parameters for use with instruments compatible with the Yamaha XG standard.

XG 2

Global settings for instruments compatible with the Yamaha XG standard.

About the Reset and Off Buttons

You find two buttons labeled **Off** and **Reset** at the top of the control panel:

- Clicking the **Off** button sets all controls to their lowest value, without sending out any MIDI messages.
- Clicking the **Reset** button resets all parameters to their default values, and sends out the corresponding MIDI messages.

GS 1

The following controls are available if the **GS 1 Controls** mode is selected:

Send 1
Send level for the reverb effect.

Send 2
Send level for the chorus effect.

Send 3
Send level for the variation effect.

Attack
Adjusts the attack time of the sound. Lowering the value shortens the attack, while raising it makes the attack time longer.

Decay
Adjusts the decay time of the sound. Lowering the value shortens the decay, while raising it makes the decay longer.

Release
Adjusts the release time of the sound. Lowering the value shortens the release, while raising it makes the release time longer.

Cutoff
Adjusts the filter cutoff frequency.

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

Express
Allows you to send out expression pedal messages on the track’s MIDI channel.

Ch. Press
Allows you to send out aftertouch (channel pressure) messages on the track’s MIDI channel. This is useful if your keyboard cannot send aftertouch, but you have sound modules that respond to aftertouch.

Breath
Allows you to send breath control messages on the track’s MIDI channel.

Modul.
Allows you to send modulation messages on the track’s MIDI channel.

XG 1

The following controls are available if the XG 1 mode is selected.

Send 1
Send level for the reverb effect.

Send 2

Send level for the chorus effect.

Send 3

Send level for the variation effect.

Attack

Adjusts the attack time of the sound. Lowering this value shortens the attack, while raising it makes the attack time longer.

Release

Adjusts the release time of the sound. Lowering this value shortens the release, while raising it makes the release time longer.

Harm.Cont

Adjusts the harmonic content of the sound.

Bright

Adjusts the brightness of the sound.

CutOff

Adjusts the filter cutoff frequency.

Resonance

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

XG 2

In this mode, the parameters affect global settings in the instruments. Changing one of these settings for a track affects all MIDI instruments connected to the same MIDI output, regardless of the MIDI channel setting of the track. Therefore, it might be a good idea to create an empty track and use this only for these global settings.

Eff. 1

Allows you to select which type of reverb effect should be used: No effect (reverb deactivated), Hall 1–2, Room 1–3, Stage 1–2, or Plate.

Eff. 2

Allows you to select which type of chorus effect should be used: No effect (chorus deactivated), Chorus 1–3, Celeste 1–3, or Flanger 1–2.

Eff. 3

Allows you to select one of a large number of variation effect types. Select **No Effect** to deactivate the variation effect.

Reset

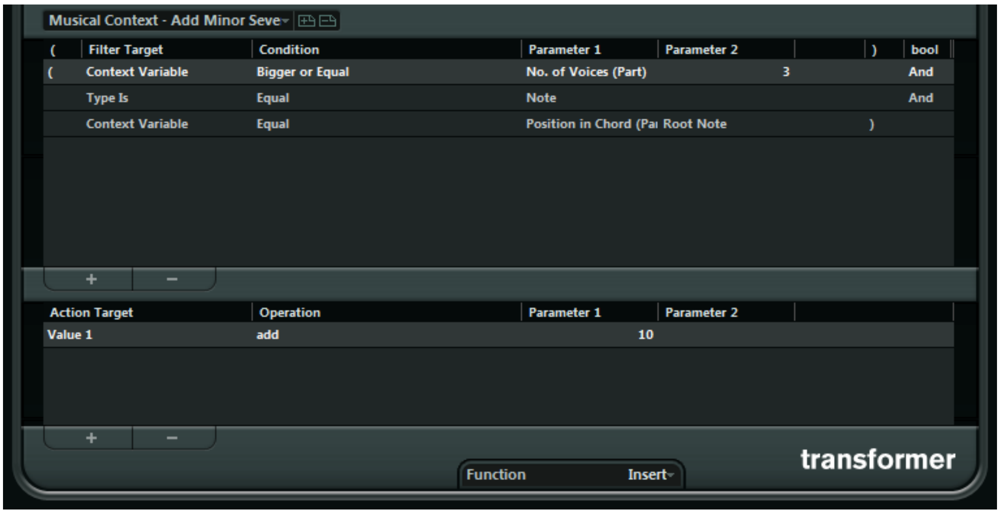
Sends an XG reset message.

MastVol

Controls the master volume of an instrument. Normally, you should leave this at its highest position and set the volumes individually for each channel (with the volume faders in the **MixConsole** or in the **Inspector**).

Transformer

Transformer is a realtime version of the **Logical Editor**. With this you can perform very powerful MIDI processing on the fly, without affecting the actual MIDI events on the track.



The **Logical Editor** is described in the **Operation Manual**. As the parameters and functions are almost identical, the descriptions for the **Logical Editor** also apply to **Transformer**. Where there are differences between the two, this is clearly stated.

Included VST Instruments

This chapter contains descriptions of the included VST instruments and their parameters.

Groove Agent SE

This VST instrument is described in detail in the separate document **Groove Agent SE**.

HALion Sonic SE

This VST instrument is described in detail in the separate document **HALion Sonic SE**.

LoopMash

LoopMash is a powerful tool for slicing and instant re-assembling of any kind of rhythmic audio material. With **LoopMash**, you can preserve the rhythmic pattern of one audio loop, but you can replace all sounds of this loop with the sounds from up to seven other loops.

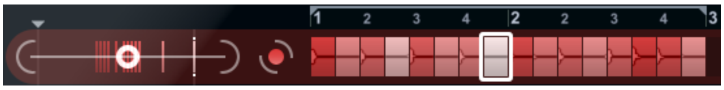
LoopMash provides dozens of possibilities to influence the way the slices are re-assembled, thus giving you full control over the results of your performance. You can choose from a variety of effects and apply them to single slices or to your overall performance. Finally, you can store your configuration as scenes on scene pads, and trigger these scene pads with your MIDI keyboard.

LoopMash is fully integrated into your host application, which allows you to drag and drop audio loops from the **MediaBay** or **Project** window directly onto the **LoopMash** panel. Furthermore, you can drag and drop slices from **LoopMash** to the sample pads of Groove Agent

SE. This allows you to extract certain sounds that you like from **LoopMash** and use them with Groove Agent SE.

The **LoopMash** window has two main areas: the track section in the upper part of the panel, and the parameter section at the bottom.

The selected track is indicated by the background color of the track and the lit button to the left of the waveform display.



The selected track holds the master loop. The rhythmic pattern of the **LoopMash** output is governed by the master loop – that is, what you hear is the rhythmic pattern of this loop.

On the left of each track, you find the similarity gain sliders. The further to the right you move the similarity gain slider of a track, the more slices are played back from this track.

Getting Started

To give you a first impression of what you can do with **LoopMash**, open the tutorial preset.

PROCEDURE

1. In your host application, create an instrument track with **LoopMash** as the associated VST instrument.
2. In the **Inspector** for the new track, click the **Edit Instrument** button to open the **LoopMash** panel.
3. At the top of the plug-in panel, click on the icon to the right of the preset field and select **Load Preset** from the pop-up menu.
4. The presets browser opens, showing presets found in the `VST 3 Presets` folder for **LoopMash**.
5. Select the preset called “A Good Start...(Tutorial) 88”.

   The preset is loaded into **LoopMash**.
6. At the bottom of the panel, make sure that the **sync** button in the transport controls is off, and start playback by clicking the **play** button.
7. Look at the 24 pads below the track section: the pad labeled **Original** is selected. Select the pad named **Clap**.

   A new loop is displayed on the second track in the track display, and you hear that the snare drum sound of the first loop has been replaced with a handclap sound.
8. Select the pad labeled **Trio**, and then the pad labeled **Section**. Each time you click, a new loop is added to the mash.

   Note how the rhythmic pattern of the music stays the same, although an increasing number of sounds is taken from the other loops.
9. Select other pads to find out how different parameter settings influence the **LoopMash** output.

   Some of the pads have the same label, for example, **Original** and **Replaced**. The scenes that are associated with these pads form the basis for variations of that scene. The variations of a scene are associated with the scene pads to the right of the original scene, that is, the scene labeled **SliceFX** is a variation of the scene labeled **Original** and shows an example for the usage of slice effects.

RELATED LINKS

LoopMash Parameters on page 220
Applying Slice Selection Modifiers and Slice Effects on page 224

How Does **LoopMash** Work?

Whenever you import a loop into **LoopMash**, the plug-in analyzes the audio material. It generates perceptual descriptors (information on tempo, rhythm, spectrum, timbre, etc.) and then slices the loop into eighth-note segments.

This means that after you have imported several loops, **LoopMash** knows the rhythmic pattern of each loop and the location of various sounds that make up this pattern within each loop. During playback, **LoopMash** uses the perceptual descriptors to determine how similar each slice is to the current slice of the master track.

NOTE

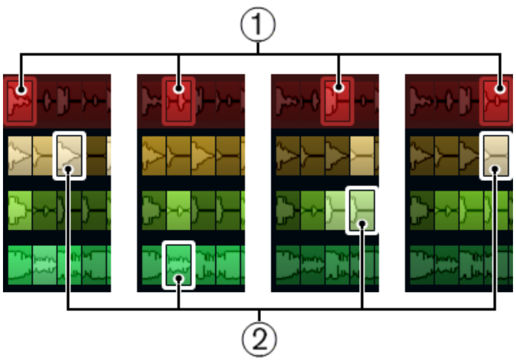
LoopMash does not categorize the sounds, but looks for overall similarity in the sound. For example, **LoopMash** might replace a low snare drum sound with a kick drum sound, even though a high snare sound is also available. **LoopMash** always tries to create a loop acoustically similar to the master loop, but using other sounds.

The similarity is shown by the brightness of each slice on each track, and also by the position of each slice on the similarity gain slider to the left of each track (if you click a slice, its position is highlighted on the similarity gain slider). The brighter a slice, the more similar a slice is to the current master track slice, and the further to the right it is displayed on the similarity gain slider. Darker slices are less similar and can be found further to the left on the slider.

The similarity gain settings of the various tracks determine which slice gets playback priority. This creates a new loop, over and over again, but with the rhythmic pattern of the original master loop.

In the following figure, you can see four tracks. The track at the top is the master track. During playback, **LoopMash** moves through the master loop step-by-step (which is indicated by a rectangle in the track's color around the current slice) and automatically selects four slices from these tracks to replace the slices of the master track. The currently playing slice is indicated by a white rectangle around the slice.

The following figure shows the result of the selection process for each playback step.



1 Master track slices for playback steps 1 to 4.

2 Slices 1 to 4 selected for playback.

For best performance, use audio files that have the same sample rate as your project (to avoid sample rate conversion when loading presets or storing scenes).

Experiment with the provided **LoopMash** presets, and with your own loops of different lengths and with different rhythms, containing many different sounds – **LoopMash** is like an instrument, and we very much encourage you to play it!

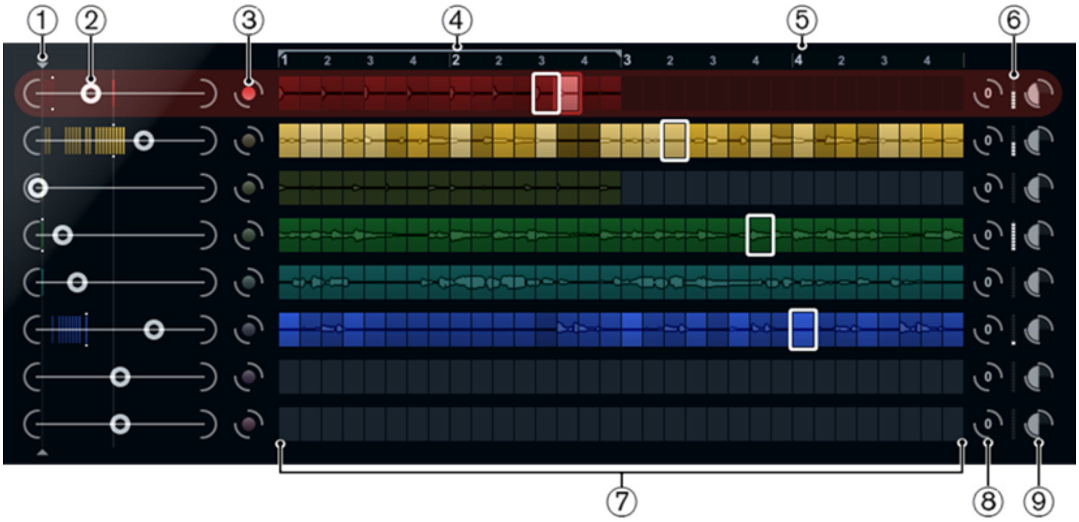
LoopMash Parameters

You can influence the process of constantly assembling a new loop with the various functions and parameter controls of **LoopMash**.

NOTE

Many of the **LoopMash** parameters can be automated. The automation of VST instrument parameters is described in the **Operation Manual**.

The Track Section



The track section contains the track display with the track controls for setting the track volume and a transposition value to the right of each track. To the left of the track display you find the similarity gain sliders. With the button between the similarity gain slider and the track, you can define the master track that serves as the reference for rhythm and timbre. At the top of the track display you find a ruler that shows bars and beats and the loop range selector.

1. Similarity threshold control
2. Similarity gain sliders
3. Master track on/off
4. Loop range selector
5. Ruler showing bars and beats
6. VU meter
7. Track display
8. Track transposition value
9. Track volume

Importing and Removing Loops

You can import up to eight audio loops onto the eight tracks in the track display.

PROCEDURE

1. Locate the audio loop that you want to import in one of the following locations: **MediaBay** and the **MediaBay** related browsers (for example, the **Loop Browser**), **Project** window, **Pool**, **Sample Editor** (regions), **Audio Part Editor**, or the File Explorer/macOS Finder.

   The quickest way to find the **LoopMash** content is to use the **MediaBay**: Navigate to the **LoopMash** content via the VST Sound node.
2. Drag the loop file onto a track in **LoopMash**.

   Dragging a loop to a track already occupied replaces the original loop.

RESULT

LoopMash separates the loop into slices, analyzes them, and displays them as a waveform on the track. One track can hold up to 32 slices. If a long loop contains more than 32 slices, **LoopMash** imports only the first 32. Ideally, you would use a loop file cut at bar boundaries. If you import your file from the **MediaBay**, **LoopMash** uses the tempo information supplied by the **MediaBay** for the slicing of the loop.

NOTE

To remove a loop from a **LoopMash** track, right-click the track and select **Clear track**.

Defining the Master Loop

One track is always selected. This is the master track: it provides the rhythmic pattern that you hear. The sounds of this loop are replaced by slices selected from the other loops in the current **LoopMash** configuration.

PROCEDURE

- To make a track the master track, activate the button to the left of it, next to the track display.

Auditioning Slices

PROCEDURE

1. Click on the slice that you want to hear.
2. Use the **Step** function in the transport controls to step through the slices.

RELATED LINKS

Transport Controls on page 225

Playback and Master Slice Indicators

A rectangle in the track color around a slice indicates the current position within the master loop, that is, the master slice. The slice selected for playback is indicated by a white rectangle.

Setting a Loop Range

At the top of the track display, a ruler showing bars and beats (using the project’s time signature) is displayed. In the ruler, you also find the loop range selector (the bracket) that defines the play length.

PROCEDURE

1. To shorten the play length, click and drag the handles of the loop range selector (the bracket) at the top of the track display.

   This allows you to select even a very small range within your master loop for playback – the rest of the loop is not taken into account.

   NOTE

   Short loop ranges (less than 1 bar) may conflict with the jump interval setting.

2. To change the playback range, click the loop range selector and drag it to a different position as a whole.

RELATED LINKS

Storing Your Configuration as Scenes on page 226

Setting Track Transposition Value and Track Volume

The track controls to the right of each track allow you to set a track transposition value and the track volume for each track individually.

PROCEDURE

1. To set a track transposition value, click the button to the right of the track and select a transposition interval from the pop-up menu.

   The set value is displayed on the button.

   NOTE

   This function is tied to the setting for the **Slice Timestretch** parameter. If **Slice Timestretch** is deactivated, transposition is created by increasing/decreasing the playback speed of the slices (transposing a track up by one octave corresponds to playing the slices twice as fast). With **Slice Timestretch** on, you get true pitch shifting, that is, there is no change in playback speed.

2. You can change the relative volumes of your tracks with the volume controls on the far right of each track.

   This is useful for level adjustments between tracks. A VU meter to the left of the volume control provides visual feedback of the current volume.

RELATED LINKS

Audio Parameters on page 228

Setting the Similarity

With the similarity gain slider (to the left of each track), you determine how important a particular track is for the mashing up of the master loop. By moving the slider, you specify that a track is more/less similar to the master track, thus overruling the result of the **LoopMash** analysis. As a result, more/less slices from this track are included in the current mash.

PROCEDURE

1. Move the slider to the right to select more slices from the corresponding track for playback, and to the left to reduce the number of slices for playback.

   The vertical lines on the similarity gain slider correspond to the slices in this loop. The changing pattern of slices indicates similarity of each slice, on all tracks, to the current master track slice. The further to the right a line is, the greater the similarity of this slice to the master slice.

2. Drag the similarity threshold control (the thin line with handles at the top and bottom intersecting all similarity gain sliders) to the left or right to determine a minimum similarity that slices must match to be considered for playback.

   Slices with a similarity below this threshold are not played.

   NOTE

   On the **Slice Selection** page at the bottom of the **LoopMash** panel, you can make further settings for influencing which slices are played.

RELATED LINKS

Slice Selection on page 227

Creating Composite Tracks

LoopMash allows you to build composite tracks.

PROCEDURE

1. Import the loop that you want to extract sounds from.
2. Audition the slices and drag the slices that you want to use onto an empty track.

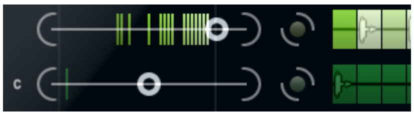
   A dialog opens asking you to confirm that you want to create a composite track, and to determine the number of slices that the track contains. If you enter a higher number of slices than the track actually contains, the track is filled up with empty slices.

3. Click **OK**.

RESULT

The destination track of the dragged slice becomes composite, indicated by a C to the left of the track.



You can use this feature in a very versatile way:

- You can assemble a combination of sounds that you like most on one track.
- You can define a certain rhythmic pattern by combining slices from different loops on a composite track and making this track the master loop.
- You can use a composite track as a clipboard, allowing you to include sounds from more than eight loops into your mash.

  You can use one track for importing and removing the loops that you want to search for sounds, and use the remaining seven tracks as composite tracks. This allows for including up to 32 sounds from up to 32 different loop files on each of the seven composite tracks.

NOTE

Composite tracks are quantized according to the set tempo.

RELATED LINKS
Transport Controls on page 225

Applying Slice Selection Modifiers and Slice Effects

Right-clicking a slice opens a context menu where you can influence the selection of individual slices and which effect is applied to them. The upper part of the context menu shows the slice selection modifiers.

Always
Only available for master track slices. The slice is played always.

Always Solo
Only available for master track slices. The slice is played always and exclusively (independent of the **Voices** parameter that you set on the **Slice Selection** page).

Exclude
The slice is never selected for playback.

Boost
Increases the similarity for this particular slice, so that it is played back more often.

Below the selection modifiers, the context menu shows the slice effects.

Mute
Mutes the slice.

Reverse
Plays the slice in reverse.

Staccato
Shortens the slice.

Scratch A, B
Plays the slice as if scratched.

Backspin 4
Simulates a turntable backspin lasting over 4 slices.

Slowdown
Applies a slowdown.

Tapestart
Simulates a tapestart, that is, speeds the slice up.

Tapestop 1, 2
Simulates a tapestop, that is, slows the slice down.

Slur 4
Stretches the slice over 4 slice lengths.

Slur 2
Stretches the slice over 2 slice lengths.

Stutter 2, 3, 4, 6, 8
Plays only the initial portion of a slice, and repeats it 2, 3, 4, 6, or 8 times during one slice length, respectively.

RELATED LINKS
Slice Selection on page 227
Performance Controls on page 228

Transport Controls

The transport controls can be found at the bottom of the **LoopMash** panel.

Play

Click the **Play** button to start or stop playback.

Locate

Click the **Locate** button to return to the beginning of the loop (bar 1/beat 1). Playback always starts automatically when clicking this button.

Step left/right

Clicking the **Step left**/**right** button steps backwards/forwards through the timeline, playing one slice at a time.

Setting the LoopMash Tempo

During playback, **LoopMash** can be synchronized to the tempo set in your host application, or can follow its own tempo setting.

- Activate the **sync** button (to the right of the **Play** button) to synchronize **LoopMash** to the project tempo set in your host application.

  If **sync** is activated, you can start playback using the transport controls of your host application. With **sync** deactivated, **LoopMash** starts playing when you click the **Play** button in **LoopMash**.
- If **sync** is deactivated, the current **LoopMash** tempo (in BPM) is displayed in the tempo field to the left of the master button. To change the local tempo, click in the tempo field, enter a new value, and press **Enter**.
- If **sync** is deactivated, you can click the master button (to the right of the tempo field) to copy the tempo of the current master loop into the tempo field.

  The **sync** on/off parameter can be automated. This is useful to control **LoopMash** in a project – with sync off, the playback of **LoopMash** within a project is paused.

Controlling Transport Functions with Your MIDI Keyboard

You can control the **start**, **stop**, **sync on**, and **sync off** functions with your MIDI keyboard.

C2

Start

D2

Stop

E2

Sync on

F2

Sync off

NOTE

If you do not have a MIDI keyboard connected to your computer, you can use the virtual keyboard (see the **Operation Manual**).

Storing Your Configuration as Scenes

On the **Slice Selection** and the **Audio Parameters** pages, you find a row of 24 pads. For each of these pads, you can save one scene, that is, a combination of up to eight tracks with all parameter settings. By triggering the pads, you can quickly change between different scenes during your performance.

1 Save scene
2 Remove scene
3 Jump interval
4 Selected scene
5 Pad with associated scene
6 Empty scene pad

- To save the current settings as a scene, click the round button and then a pad. This saves your setup to that pad.
- To recall a scene, click the corresponding scene pad.
- To remove a scene from a pad, click the **x** button and then a pad.
- To edit a scene pad label, double-click on the scene pad and enter a name.
- To rearrange the scene pads, click on a scene pad and drag it to a new position.

IMPORTANT

Once you have set up a **LoopMash** configuration, save it to a scene pad. Changing scenes without saving means discarding any unsaved changes.

Setting a Jump Interval

You can determine the point at which **LoopMash** changes to the next scene during playback when you trigger a pad.

PROCEDURE

- Click the **Jump interval** button and select an option from the pop-up menu.

RESULT

NOTE

The option **e: End** means that the current loop is played to the end before switching scenes. When you set up a short loop range, you may need to set the interval to **e: End** to ensure that the jump point is reached.

Triggering Scene Pads with Your MIDI Keyboard

The scene pads are arranged according to the keys on a MIDI keyboard. You can trigger the 24 scene pads with a connected MIDI keyboard starting from C0 and ending with B1.

Slice Selection

Click the **Slice Selection** button (above the transport controls) to open the Slice Selection page. The options on this page allow you to further influence which slices are selected for playback.

Number of Voices

Here you can set the total number of slices from all tracks that replace the master slice (according to the current similarity gain settings). The range is from one (left) to four (right) voices, that is, sounds from up to four loops can play simultaneously. Increasing the number of voices increases the CPU load.

Voices per Track

This is the maximum number of slices that can be selected from a single track. The range is from one to four. The fewer slices can be picked from the same track, the more variety you get in the **LoopMash** output.

Selection Offset

Move this slider to the right to allow slices that are less similar to be selected for playback. This setting affects all tracks of this scene.

Random Selection

Move this slider to the right to allow more variation when selecting slices for playback, adding a more random feel to the selection process. This setting affects all tracks of this scene.

Selection Grid

Determines how often **LoopMash** looks for similar slices during playback: always (left position), or only every 2nd, 4th, or 8th (right position) step. For example, if you set the Selection Grid to every 8th step (right position), **LoopMash** replaces similar slices every 8th step. Between two replacement steps it plays back the tracks of the slices that have been selected in the last replacement step, resulting in longer playback sequences on one track.

Similarity Method

Here, you can modify the criteria that **LoopMash** considers when comparing the slices for similarity. There are three similarity methods:

- **Standard** – This is the standard method, where all slices on all tracks are compared and various characteristics regarding rhythm, tempo, spectrum, etc. are taken into account.
- **Relative** – This method does not only consider the overall similarity of all slices on all tracks, but also takes the relation to the other slices on the same track into account. For example, **LoopMash** can replace the loudest, lowest sound on one track with the loudest, lowest sound on another track.
- **Harmonic** – This method only takes the analyzed tonal information into account, so that a slice is replaced by a harmonically similar slice, rather than by a rhythmically similar slice. With this method, also the track transposition value is considered, that is, a master slice with a C major chord is not replaced by a slice with a D major chord. But it is replaced if you set the transposition value of the track of the slice with the D major chord to -2. It is advisable to keep the similarity gain sliders in a low position when you work with this method, because otherwise you may produce disharmonies. You can modify the transposition values to play back more slices of a specific track.

RELATED LINKS
Storing Your Configuration as Scenes on page 226

Audio Parameters

Click the **Audio Parameters** button (above the transport controls) to open the **Audio Parameters** page. With the options on this page, you can influence the sound of the LoopMash audio output.

Adapt Mode

With this mode, you can adapt the sound of the selected slice to the sound of the master slice. The available options are:

- **Volume** – changes the overall volume of the selected slice.
- **Envelope** – modifies volume changes within the slice.
- **Spectrum** – modifies the spectrum of the slice (equalization).
- **Env + Spectrum** – this is a combination of the **Envelope** and **Spectrum** modes.

Adapt Amount

Move this slider to the right to increase the adaptation specified with the **Adapt Mode** parameter.

Slice Quantize

Move this slider to the right to apply quantizing to the slices, that is, the slices are aligned to an eighth-note grid. If the slider is all the way to the left, the slices follow the rhythmic pattern defined by the original master loop.

Slice Timestretch

Allows you to apply realtime timestretching to the slices, filling gaps or avoiding overlaps between slices that are not played back at their original tempo, or when combining slices with different original tempos. Applying timestretch increases the CPU load and may affect the sound quality. Reduce the need for timestretching by using loops with similar original tempos.

Staccato Amount

If you move this slider to the right, the length of the slices is gradually reduced, giving the output a staccato feel.

Dry/Wet Mix

Sets the balance between the volumes of the master loop and the selected slices from the other tracks.

RELATED LINKS
Setting Track Transposition Value and Track Volume on page 222

Performance Controls

Click the **Performance Controls** button to open the **Performance Controls** page. On this page, you find a row of buttons that are arranged according to the keys on a MIDI keyboard.

- By clicking these buttons during playback, you can apply effects to your overall performance.

  An effect is applied as long as the button is activated.

Most of the available effects correspond to the effects that you can apply to single slices, with the green buttons corresponding to the stutter and slur effects and the red buttons to the Mute, Reverse, Staccato effects, etc.

NOTE

Effects triggered with the **Performance Controls** buttons override the slice effects.

With the blue buttons and the yellow button, you can apply additional effects that cannot be applied to single slices:

Cycle 4, 2, 1

Sets up a short cycle over 4, 2, and 1 slices, respectively. This short cycle is always set up within the loop range that is set in the ruler. Setting up a cycle over 1 slice means that this slice is repeated until you release the button.

Continue

Plays back the tracks of the selected slices continuously until you release the button.

NOTE

You cannot save global effects in scenes. To apply effects and save them in scenes, use slice effects.

Triggering the Performance Controls with Your MIDI Keyboard

You can trigger the performance controls with your MIDI keyboard starting from C3 upwards.

RELATED LINKS

Applying Slice Selection Modifiers and Slice Effects on page 224

Mystic

The synthesis method used by **Mystic** is based on three parallel comb filters with feedback. A comb filter is a filter with a number of notches in its frequency response, with the notch frequencies harmonically related to the frequency of the fundamental (lowest) notch.

A typical example of comb filtering occurs if you are using a flanger effect or a delay effect with very short delay time. Raising the feedback (the amount of signal sent back into the delay or flanger) causes a resonating tone – this tone is basically what the **Mystic** produces. This synthesis method is capable of generating a wide range of sounds, from gentle plucked-string tones to weird, non-harmonic timbres.

The basic principle is the following:

- You start with an impulse sound, typically with a very short decay.

  The spectrum of the impulse sound largely affects the tonal quality of the final sound.
- The impulse sound is fed into the three comb filters, in parallel. Each of these has a feedback loop.

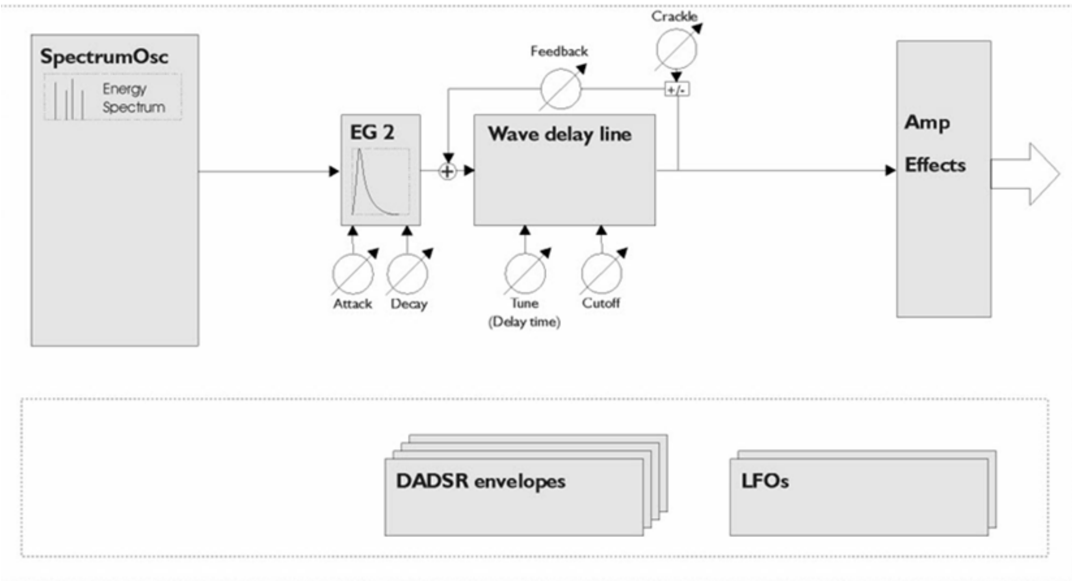
  This means the output of each comb filter is fed back into the filter. This results in a resonating feedback tone.
- When the signal is fed back into the comb filter, it goes via a separate, variable low-pass filter.

  This filter corresponds to the damping of high frequencies in a physical instrument – if this is set to a low cutoff frequency it causes high harmonics to decay faster than the lower harmonics (as when plucking a string on a guitar, for example).
- The level of the feedback signal is governed by a feedback control.

  This determines the decay of the feedback tone. Setting this to a negative value simulates the traveling wave in a tube with one open end and one closed end. The result is a more hollow, square wave-like sound, pitched one octave lower.
- A detune control offsets the fundamental frequencies of the three comb filters, for chorus-like sounds or drastic special effects.

Finally you have access to the common synth parameters – two LFOs, four envelopes and an effect section.

- By default, envelope 2 controls the level of the impulse sound – this is where you set up the short impulse decay when emulating string sounds, etc.



Functional Diagram

Sound Parameters

The Impulse Control Section

This is where you set up the impulse sound – the sound fed into the comb filters, serving as a starting point for the sound. The Impulse Control has two basic waveforms that are filtered through separate spectrum filters with adjustable base frequency. The output is an adjustable mix between the two waveform/spectrum filter signals.

Spectrum Displays

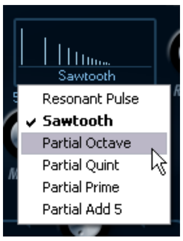
The displays allow you to draw a filter contour with your mouse for spectrum filters A and B.

- To set up the contour, click in one of the displays and drag the mouse to draw a curve. This produces the inverse contour in the other display, for maximum sonic versatility.

  To set up the contour independently for the two filters, hold down **Shift** and click and drag the mouse in either display.
- Use the Preset pop-up menu to select a preset contour.
- If you want to calculate a random spectrum filter curve, you can choose the **Randomize** function from the Preset pop-up menu.

  Each time you choose this function, a new randomized spectrum appears.

Waveform Pop-up Menu



The pop-up menu at the bottom of the waveform section (the central box at the top of the panel) allows you to select a basic waveform to be sent through filter contour A. The options are especially suited for use with the spectrum filter.

Cut

Offsets the frequency of the filter contour, working somewhat like a cutoff control on a standard synth filter. To use the filter contour in its full frequency range, set **Cut** to its maximum value.

Morph

Adjusts the mix between the two signal paths: waveform A spectrum contour A and waveform B spectrum contour B.

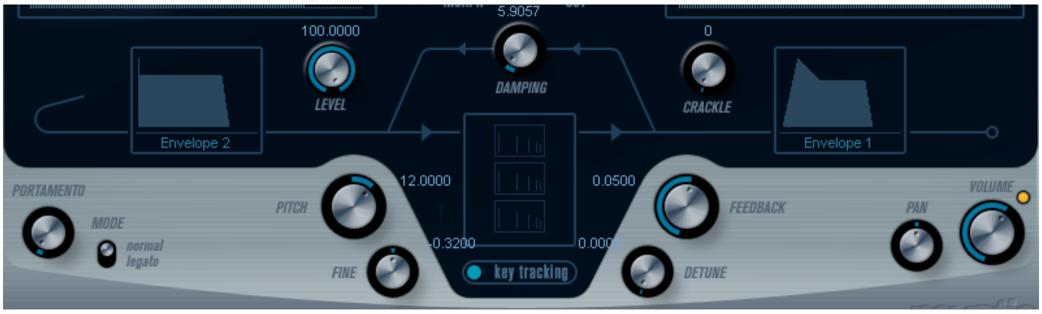
Coarse

Offsets the pitch for the impulse sound. In a typical string setup, when the impulse sound is very short, this does not change the pitch of the final sound, but the tonal color.

Raster

This removes harmonics from the impulse sound. As the harmonic content of the impulse sound is reflected in the comb filter sound, this changes the final timbre.

Comb Filter Sound Parameters



Damping

This is a 6 dB/oct low-pass filter that affects the sound being fed back into the comb filters. This means the sound becomes gradually softer when decaying, that is, high harmonics decay faster than the lower harmonics (as when plucking a string on a guitar, for example).

- The lower the **Damping**, the more pronounced this effect.

  If you open the filter completely (turn **Damping** up to max) the harmonic content is static – the sound does not get softer when decaying.

Level

Determines the level of the impulse sound being fed into the comb filters. By default, this parameter is modulated by envelope 2. That is, you use envelope 2 as a level envelope for the impulse sound.

- For a string-type sound, you want an envelope with a quick attack, a very short decay and no sustain (an impulse in other words), but you can also use other envelopes for other types of sounds.

  Try raising the attack for example, or raising the sustain to allow the impulse sound to be heard together with the comb filter sound.

Crackle

Allows you to send noise directly into the comb filters. Small amounts of noise produce a crackling, erratic effect, higher amounts give a more pronounced noise sound.

Feedback

Determines the amount of signal sent back into the comb filters (the feedback level).

- Setting **Feedback** to zero (twelve o’clock) effectively turns off the comb filter sound, as no feedback tone is produced.

- Setting **Feedback** to a positive value creates a feedback tone, with higher settings generating longer decays.
- Setting **Feedback** to a negative value creates a feedback tone with a more hollow sound, pitched one octave lower. Lower settings generate longer decays.

Detune

Offsets the notch frequencies of the three parallel comb filters, effectively changing the pitches of their feedback tones. At low settings, this creates a chorus-like detune effect. Higher settings detunes the three tones in wider intervals.

Pitch and Fine

Overall pitch adjustment of the final sound. This changes the pitch of both the impulse sound and the final comb filter sound.

Key Tracking

Determines whether the impulse sound should track the keyboard. This affects the sound of the comb filters in a way similar to a key track switch on a regular subtractive synth filter.

Portamento

Makes the pitch glide between the notes you play. The parameter setting determines the time it takes for the pitch to glide from one note to the next. Turn the knob clockwise for longer glide time.

The **Mode** switch allows you to apply glide only when you play a legato note (switch is set to **Legato**). Legato is when you play a note without releasing the previously played note. Note that **Legato** mode only works with monophonic parts.

Master Volume and Pan

The master **Volume** knob controls the master volume (amplitude) of the instrument. By default, this parameter is controlled by Envelope 1, to generate an amplitude envelope for the oscillators.

The **Pan** knob controls the position of the instrument in the stereo spectrum. You can use **Pan** as a modulation destination.

Modulation and Controllers

The lower half of the control panel displays the various modulation and controller assignment pages available, as well as the **EFX** page. You switch between these pages using the buttons above this section.

The following pages are available:

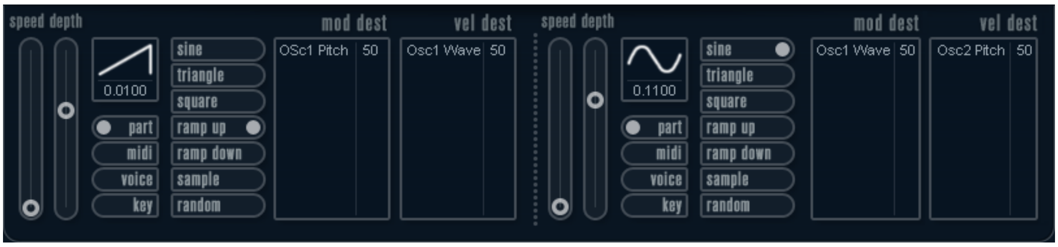
- The **LFO** page has two low frequency oscillators (LFOs) for modulating parameters.
- The **ENV** page contains the four envelope generators that can be assigned to control parameters.
- The **Event** page contains the common MIDI controllers (Mod wheel, Aftertouch, etc.) and their assignments.
- The **EFX** page offers three separate effect types: Distortion, Delay, and Modulation.

RELATED LINKS
LFO Page on page 234

Envelope Page on page 236
Event Page on page 238
Effects (EFX) Page on page 239

LFO Page

The LFO page is opened by clicking the **LFO** button at the top of the lower half of the control panel. The page contains all parameters and the modulation and velocity destinations for two independent LFOs.



Depending on the selected preset, there may already be modulation destinations assigned, in which case these are listed in the **Mod Dest** box for each LFO.

A low frequency oscillator (LFO) is used for modulating parameters, for example the pitch of an oscillator (to produce vibrato), or for any parameter where cyclic modulation is required.

The two LFOs have identical parameters.

Speed

Governs the rate of the LFO. If the sync mode is set to **MIDI**, the available rate values are selectable as note values, so the rate is synchronized to the sequencer tempo.

Depth

Controls the amount of modulation applied by the LFO. If this is set to zero, no modulation is applied.

Waveform

Sets the LFO waveform.

Sync mode (Part/MIDI/Voice/Key)

Sets the sync mode for the LFO.

RELATED LINKS
Assigning LFO Modulation Destinations on page 235

About the Sync Modes

The sync modes determine how the LFO cycle affects the notes you play.

Part

In this mode, the LFO cycle is free running and affects all the voices in sync. Free running means that the LFO cycles continuously, and does not reset when a note is played.

MIDI

In this mode, the LFO rate is synced in various beat increments to MIDI clock.

Voice

In this mode, each voice in the Part has its own independent LFO cycle (the LFO is polyphonic). These cycles are also free running – each key down starts anywhere in the LFO cycle phase.

Key

Same as **Voice** except that it is not free running – for each key down the LFO cycle starts over.

About the Waveforms

Most standard LFO waveforms are available for LFO modulation. You use sine and triangle waveforms for smooth modulation cycles, square and ramp up/down for different types of stepped modulation cycles and random or sample for random modulation. The sample waveform is different:

- In this mode, the LFO makes use of the other LFO as well.

  For example, if LFO 2 is set to use **Sample**, the resulting effect also depends on the speed and waveform of LFO 1.

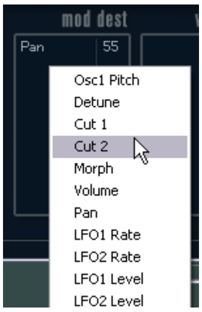
Assigning LFO Modulation Destinations

You can assign a modulation destination for an LFO.

PROCEDURE

1. Click in the **Mod Dest** box for one of the LFOs.

   A pop-up menu appears in which all possible modulation destinations are shown. All Sound parameters as well as most LFO and Envelope parameters are available as destinations.

2. Select a destination, for example, **Cut**.

   The selected modulation destination is now shown in the list. Beside the destination, a default value (50) has been set. The value represents the modulation amount.

   - You can set positive and negative modulation values by clicking on the value in the list, typing in a new value, and pressing **Enter**.

   To enter negative values, type a minus sign followed by the value.

3. Select a suitable LFO Waveform, Speed, Depth, and sync mode.

   You should now hear the **Cut** parameter being modulated by the LFO.

4. Using the same basic method, you can add any number of modulation destinations for the LFO.

   They are all listed in the **Mod Dest** box.

   - To remove a modulation destination, click on its name in the list and select **Off** from the pop-up menu.

Assigning LFO Velocity Destinations

You can also assign velocity-controlled LFO modulation.

PROCEDURE

1. Click in the **Vel Dest** box for one of the LFOs.

A pop-up menu appears in which all possible velocity destinations are shown.

2. Select a destination.

   The selected velocity destination is now shown in the list. Beside the destination, a default value (50) has been set. The value represents the modulation amount.

   - You can set positive and negative values by clicking on the value in the list, typing in a new value, and pressing **Enter**.

   To enter negative values, type a minus sign followed by the value.

3. Using the same basic method, you can add any number of velocity destinations for the LFO.

   They are all listed in the **Vel Dest** box.

   - To remove a velocity destination, click on its name in the list and select **Off** from the pop-up menu.

LFO modulation velocity control

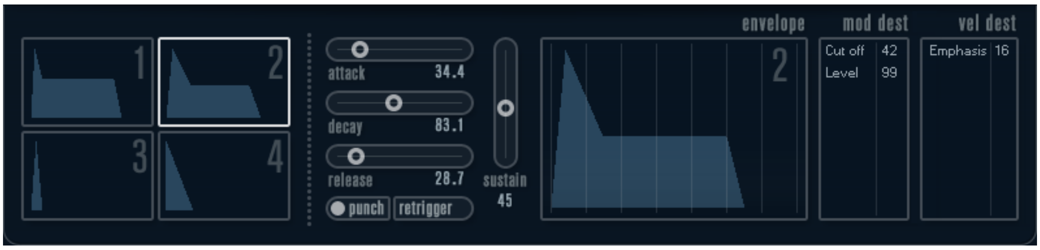
If you follow the steps above and select the **Cut** parameter as a Velocity destination, the following happens:

- The harder you strike the key, the more the **Cut** parameter is modulated by the LFO.
- If you enter a negative value for the velocity modulation amount, the opposite happens: the harder you play, the less the **Cut** parameter is modulated by the LFO.

Envelope Page

The Envelope page is opened by clicking the **ENV** button at the top of the lower half of the control panel. The page contains all parameters and the modulation and velocity destinations for the four independent envelope generators.

Envelope generators govern how a parameter value changes when a key is pressed, when a key is held and finally when a key is released.



On the Envelope page, the parameters for one of the four envelope generators is shown at a time.

- You switch between the four envelopes in the section to the left.

  Clicking on either of the four mini curve displays selects it and displays the corresponding envelope parameters to the right.
- Envelope generators have four parameters: **Attack**, **Decay**, **Sustain**, and **Release** (ADSR).
- You can set envelope parameters in 2 ways: by using the sliders or by clicking and dragging the curve in the Envelope curve display.

  You can also do this in the mini curve displays.
- By default, Envelope 1 is assigned to the master volume, and therefore acts as an amplitude envelope. The amplitude envelope adjusts how the volume of the sound changes from the time you press a key until the key is released.

  If no amplitude envelope is assigned, there is no output.
- Envelope 2 is by default assigned to the **Level** parameter.

The Envelope parameters are as follows:

Attack

The attack phase is the time it takes from zero to the maximum value. How long this takes is governed by the **Attack** setting. If the **Attack** is set to 0, the maximum value is reached instantly. If this value is raised, it takes time before the maximum value is reached. Range is from 0.0 milliseconds to 91.1 seconds.

Decay

After the maximum value has been reached, the value starts to drop. How long this takes is governed by the **Decay** parameter. The **Decay** has no effect if the **Sustain** parameter is set to maximum.

Sustain

Determines the level for the envelope after the **Decay** phase. Note that **Sustain** represents a level, whereas the other envelope parameters represent times.

Release

Determines the time it takes for the value to fall back to zero after releasing the key. Range is from 0.0 milliseconds to 91.1 seconds.

Punch

If **Punch** is activated, the start of the decay phase is delayed a few milliseconds, that is, the envelope stays at top level for a moment before moving on to the decay phase. The result is a punchier attack similar to a compressor effect. This effect is more pronounced with short attack and decay times.

Retrigger

If **Retrigger** is activated, the envelope retriggers each time you play a new note. However, with certain textures/pad sounds and a limited number of voices, it is recommended to leave the button deactivated, due to click noises that might occur.

Assigning Envelope Modulation Destinations

You can assign a modulation destination for an envelope.

PROCEDURE

1. Click in the **Mod Dest** box for one of the envelopes.

   A pop-up menu appears in which all possible modulation destinations are shown. All Sound parameters as well as most LFO and Envelope parameters are available as destinations.

2. Select a destination, for example, **Cut**.

   The selected modulation destination is now shown in the list. Beside the destination, a default value (50) has been set. The value represents the modulation amount.

   - You can set positive and negative modulation values by clicking on the value in the list, typing in a new value, and pressing **Enter**.

     To enter negative values, type a minus sign followed by the value.

3. Select a suitable envelope curve for the modulation.

   You should now hear the **Cut** parameter being modulated by the envelope as you play.

4. Using the same basic method, you can add any number of modulation destinations for the envelope.

   They are all listed in the **Mod Dest** box.

   - To remove a modulation destination, click on its name in the list and select **Off** from the pop-up menu.

Assigning Envelope Velocity Destinations

You can also assign velocity-controlled envelope modulation, that is, the modulation is governed by how hard or soft you strike a key.

PROCEDURE

1. Click in the **Vel Dest** box for one of the envelopes.

   A pop-up menu appears in which all possible velocity destinations are shown.

2. Select a destination.

   The selected velocity destination is now shown in the list. Beside the destination, a default value (50) has been set. The value represents the modulation amount. See below for an example of how velocity modulation works.

   - You can set positive and negative values by clicking on the value in the list, typing in a new value, and pressing **Enter**.

     To enter negative values, type a minus sign followed by the value.

3. Using the same basic method, you can add any number of velocity destinations for the Envelope.

   They are all listed in the **Vel Dest** box.

   - To remove a velocity destination, click on its name in the list and select **Off** from the pop-up menu.

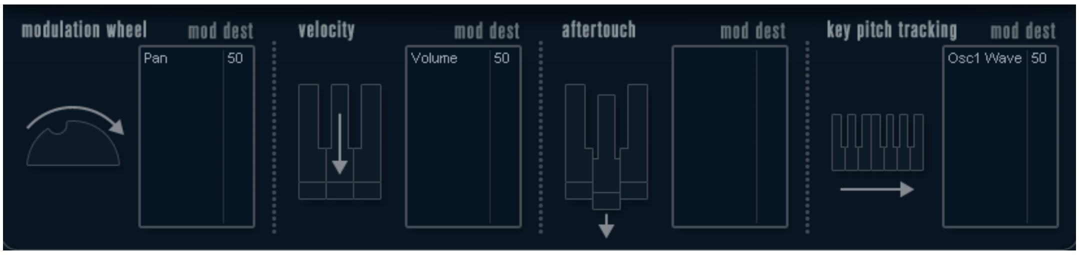
Envelope modulation velocity control

If you follow the steps above and select the **Cut** parameter as a Velocity destination, the following happens:

- The harder you strike the key, the more the parameter is modulated by the envelope.
- If you enter a negative value for the velocity modulation amount, the opposite happens; the harder you play the less the **Cut** parameter is modulated by the Envelope.

Event Page

The Event page is opened by clicking the **EVENT** button at the top of the lower half of the control panel. This page contains the most common MIDI controllers and their assignments.



Modulation Wheel

The modulation wheel on your keyboard can be used to modulate parameters.

Velocity

Controls parameters according to how hard or soft you play notes on your keyboard. A common application of velocity is to make sounds brighter and louder if you strike the key harder.

Aftertouch

Aftertouch, or channel pressure, is MIDI data sent when pressure is applied to a keyboard after the key has been struck, and while it is being held down or sustained.

Aftertouch is often routed to control filter cutoff, volume, and other parameters to add expression.

Key Pitch Tracking

This can change parameter values linearly according to where on the keyboard you play.

Assigning a Controller to a Parameter

PROCEDURE

1. Click in the **Mod Dest** box for one of the controllers.

   A pop-up menu appears in which all possible modulation destinations are shown. All Sound parameters as well as most LFO and Envelope parameters are available as destinations.

2. Select a destination.

   The selected modulation destination is now shown in the list. Beside the destination, a default value (50) has been set. The value represents the modulation amount when the controller is at its full range.

   - You can set positive and negative modulation values by clicking on the value in the list, typing in a new value, and pressing **Enter**.

     To enter negative values, type a minus sign followed by the value.

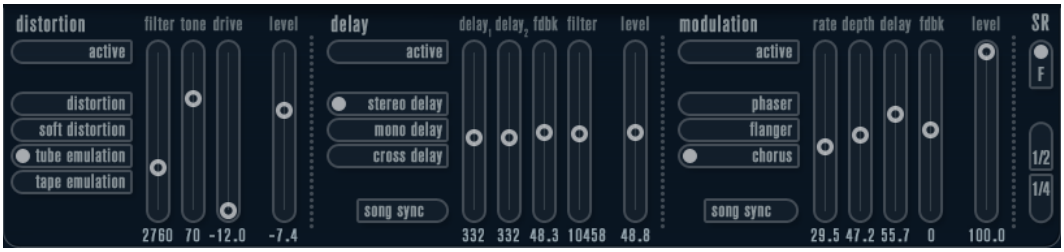
3. Using the same basic method, you can add any number of modulation destinations for the controllers.

   They are all listed in the **Mod Dest** box for each controller.

   - To remove a modulation destination, click on its name in the list and select **Off** from the pop-up menu.

Effects (EFX) Page

This page features three separate effect units: **Distortion**, **Delay**, and **Modulation** (Phaser/Flanger/Chorus). The Effect page is opened by clicking the **EFX** button at the top of the lower half of the control panel.



- Each separate effect section is laid out with a row of buttons that determine the effect type or characteristic and a row of sliders for making parameter settings.
- To activate an effect, click the **Active** button so that a dot appears.

  Clicking again deactivates the effect.

Distortion

You can choose between 4 basic distortion characteristics:

- **Distortion** provides hard clipping distortion.
- **Soft Distortion** provides soft clipping distortion.
- **Tape Emulation** produces distortion similar to magnetic tape saturation.
- **Tube Emulation** produces distortion similar to valve amplifiers.

Drive

Sets the amount of distortion by amplifying the input signal.

Filter

Sets the crossover frequency of the distortion filter. The distortion filter consists of a low-pass filter and a high-pass filter with a cutoff frequency equal to the crossover frequency.

Tone

Controls the relative amount of low-pass and high-pass filtered signal.

Level

Controls the output level of the effect.

Delay

You can choose between 3 basic delay characteristics:

- **Stereo Delay** has two separate delay lines panned left and right.
- In **Mono Delay**, the two delay lines are connected in series for monophonic dual tap delay effects.
- In **Cross Delay**, the delayed sound bounces between the stereo channels.

Song Sync

Activates/Deactivates tempo sync of the delay times.

Delay 1

Sets the delay time ranging from 0 ms to 728 ms. If **MIDI sync** is activated, the range is from 1/32 to 1/1; straight, triplet or dotted.

Delay 2

Same as **Delay 1**.

Feedback

Controls the decay of the delays. With higher settings, the echoes repeat longer.

Filter

A low-pass filter is built into the feedback loop of the delay. This parameter controls the cutoff frequency of this feedback filter. Low settings result in successive echoes sounding darker.

Level

Controls the output level of the effect.

Modulation

You can choose between 3 basic modulation characteristics:

- **Phaser** uses an 8-pole all-pass filter to produce the classic phasing effect.
- **Flanger** is composed of two independent delay lines with feedback for the left and the right channel. The delay time of both delays is modulated by one LFO with adjustable frequency.
- **Chorus** produces a rich chorus effect with 4 delays modulated by four independent LFOs.

Song Sync

Activates/Deactivates tempo sync of the **Rate** parameter.

Rate

Sets the rate of the LFOs modulating the delay time. If **Song Sync** is activated, the rate is synchronized to various beat increments.

Depth

Controls the depth of the delay time modulation.

Delay

Sets the delay time of the four delay lines.

Feedback

Controls the amount of positive or negative feedback for all four delay lines.

Level

Controls the output level of the effect.

SR Parameters

With these buttons, you can change the sample rate. Lower sample rates basically reduce the high frequency content and sound quality, but the pitch is not altered. This is useful to emulate the lo-fi sounds of older digital synths.

- If the **F** button is active, the program of the selected part plays back with the sample rate set in the host application.
- If the **1/2** button is active, the program of the selected part plays back with half the original sample rate.
- If the **1/4** button is active, the program of the selected part plays back with a quarter of the original sample rate.

A bonus effect of using lower sample rates is that it reduces the load on the computer CPU, allowing for more simultaneous voices to be played, etc.

Padshop

This VST instrument is described in detail in the separate document **Padshop**.

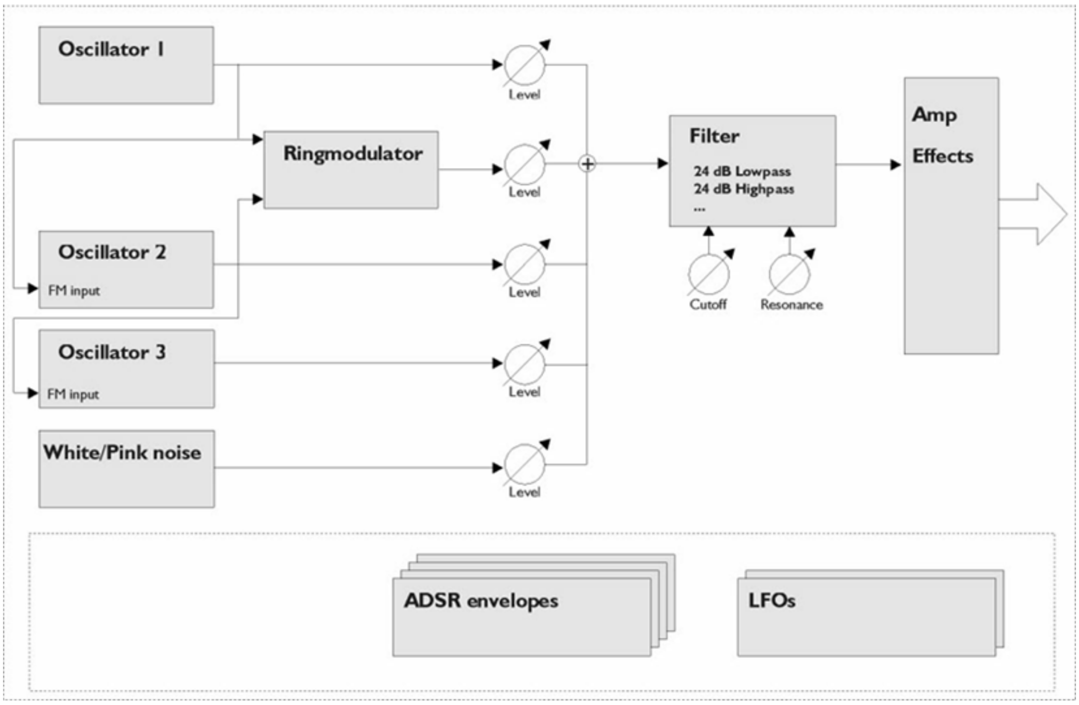
Prologue

Prologue is modelled on subtractive synthesis, the method used in classic analog synthesizers. It has the following basic features:

- Multimode filter

  Variable slope low-pass and high-pass, plus band-pass and notch filter modes.
- Three oscillators, each with 4 standard waveforms plus an assortment of specialized waveforms.
- Frequency modulation.
- Ring Modulation.
- Built-in effects.
- **Prologue** receives MIDI on all MIDI channels.

  You do not have to select a MIDI channel to direct MIDI to **Prologue**.



Functional Diagram

Sound Parameters

Oscillator Section

This section contains parameters affecting the 3 oscillators. These are located in the upper half of the instrument panel.

Selecting Waveforms

Each oscillator has a number of waveforms that can be selected by clicking on the waveform name in the box located in each oscillator section.

Sawtooth

This waveform contains all harmonics and produces a bright and rich sound.

Parabolic

This can be described as a rounded sawtooth waveform, producing a softer timbre.

Square

Square waveforms only contain odd number harmonics, which produces a distinct, hollow sound.

Triangle

The triangle waveform generates only a few harmonics, spaced at odd harmonic numbers, which produces a slightly hollow sound.

Sine

The sine wave is the simplest possible waveform, with no harmonics (overtones). The sine wave produces a neutral, soft timbre.

Formant 1–12

Formant waveforms emphasizes certain frequency bands. Like the human voice, musical instruments have a fixed set of formants, which give it a unique, recognizable tonal color or timbre, regardless of pitch.

Vocal 1–7

These are also formant waveforms, but specifically vocal-oriented. Vowel sounds (A/E/I/O/U) are among the waveforms found in this category.

Partial 1–7

Partials, also called harmonics or overtones, are a series of tones which accompany the prime tone (fundamental). These waveforms produce intervals with two or more frequencies heard simultaneously with equal strength.

Reso Pulse 1–12

This waveform category begins with a complex waveform (Reso Pulse 1) that emphasizes the fundamental frequency (prime). For each consecutive waveform in this category, the next harmonic in the harmonic series is emphasized.

Slope 1–12

This waveform category begins with a complex waveform (Slope 1), with gradually decreasing harmonic complexity the higher the number selected. Slope 12 produces a sine wave (no harmonics).

Neg Slope 1–9

This category also begins with a complex waveform (NegSlope 1), but with gradually decreasing low frequency content the higher the number selected.

- To hear the signal generated by the oscillators, the corresponding Osc controls in the oscillator sections must be set to a suitable value.

OSC 1 Parameters

Oscillator 1 acts as a master oscillator. It determines the base pitch for all three oscillators.

Osc 1 (0–100)

This controls the output level of the oscillator.

Coarse (±48 semitones)

This determines the base pitch used by all oscillators.

Fine (±50 cent)

Fine-tunes the oscillator pitch in cent increments (100th of a semitone). This also affects all oscillators.

Wave Mod (±50)

This parameter is only active if the **Wave Mod** button is activated beside the waveform selection box. Wave modulation works by adding a phase-shifted copy of the oscillator output to itself, which produces waveform variations. For example if a sawtooth waveform is used, activating WM produces a pulse waveform. By modulating the WM parameter with for example an LFO, classic PWM (pulse width modulation) is produced. However, wave modulation can be applied to any waveform.

Phase button (On/Off)

If phase synchronization is activated, all oscillators restart their waveform cycles with every note that is played. With **Phase** deactivated, the oscillators generate a waveform cycle continuously, which produces slight variations when playing as each note starts from a random phase in the cycle, adding warmth to the sound. For bass sounds or drum sounds, it is often required that the attack of every note sounds the same, therefore, for these purposes activate phase sync. Phase sync also affects the noise generator.

Tracking button (On/Off)

If **Tracking** is activated, the oscillator pitch tracks the notes played on the keyboard. If **Tracking** is deactivated, the oscillator pitch remains constant, regardless of the note that is played.

Wave Mod button (On/Off)

Activates/Deactivates wave modulation.

Waveform pop-up menu

Sets the basic waveform for the oscillator.

OSC 2 Parameters

Osc 2 (0–100)

Controls the output level of the oscillator.

Coarse (±48semitones)

Determines the coarse pitch for Osc 2. If **FM** is enabled, this determines frequency ratio of the oscillator regarding Osc 1.

Fine (±50 cent)

Fine-tunes the oscillator pitch in cent increments (100th of a semitone). If **FM** is activated, this determines the frequency ratio of the oscillator regarding Osc 1.

Wave Mod (±50)

This parameter is only active if the **Wave Mod** button next to the waveform selector is activated. Wave modulation works by adding a phase-shifted copy of the oscillator output to itself, which produces waveform variations. For example, if a sawtooth waveform is used, activating **WM** produces a pulse waveform. By modulating the **WM** parameter with an LFO, classic PWM (pulse width modulation) is produced. Wave modulation can be applied to any waveform.

Ratio (1–16)

This parameter is only active if **Freq Mod** is activate. It adjusts the amount of frequency modulation applied to oscillator 2. It is normally referred to as “FM index”.

Sync button (On/Off)

If **Sync** is activated, Osc 2 is slaved to Osc 1. This means that every time Osc 1 completes its cycle, Osc 2 is forced to start its cycle from the beginning. This produces a characteristic sound, suitable for lead playing. Osc 1 determines the pitch, and varying the pitch of Osc 2 produces changes in timbre. For classic sync sounds, try modulating the pitch of Osc 2 with an envelope or an LFO. The Osc 2 pitch should also be set higher than the pitch of Osc 1.

Tracking button (On/Off)

If **Tracking** is activated, the oscillator pitch tracks the notes played on the keyboard. If **Tracking** is deactivated, the oscillator pitch remains constant, regardless of the note that is played.

Freq Mod button (On/Off)

Activates/Deactivates frequency modulation.

Wave Mod button (On/Off)

Activates/Deactivates wave modulation.

Waveform pop-up menu

Sets the basic waveform for the oscillator.

OSC 3 Parameters

Osc 3 (0–100)

Controls the output level of the oscillator.

Coarse (±48semitones)

Determines the coarse pitch for Osc 3. If **FM** is activated, this determines the frequency ratio of the oscillator regarding Osc 1/2.

Fine (±50 cent)

Fine-tunes the oscillator pitch in cent increments. If **FM** is activated, this determines the frequency ratio of the oscillator regarding Osc 1/2.

Ratio (1–16)

This parameter is only active if the **Freq Mod** button is activated. It adjusts the amount of frequency modulation applied to oscillator 3. It is normally referred to “FM index”.

Sync button (On/Off)

If **Sync** is activated, Osc 3 is slaved to Osc 1. This means that every time Osc 1 completes its cycle, Osc 3 is forced to start its cycle from the beginning. This produces a characteristic sound, suitable for lead playing. Osc 1 determines the pitch, and varying the pitch of Osc 3 produces changes in timbre. For classic sync sounds, try modulating the pitch of Osc 3 with an envelope or an LFO. The Osc 3 pitch should also be set higher than the pitch of Osc 1.

Tracking button (On/Off)

If **Tracking** is activated, the oscillator pitch tracks the notes played on the keyboard. If **Tracking** is deactivated, the oscillator pitch remains constant, regardless of the note that is played.

Freq Mod button (On/Off)

Activates/Deactivates frequency modulation.

Wave Mod button (On/Off)

Activates/Deactivates wave modulation.

Waveform pop-up menu

Sets the basic waveform for the oscillator.

Frequency Modulation

Frequency modulation or FM means that the frequency of one oscillator, called the carrier, is modulated by the frequency of another oscillator, called the modulator.

- In Prologue, Osc 1 is the modulator, and Osc 2 and 3 are carriers.

  However, Osc 2 can be both carrier and modulator as if frequency modulation is applied to Osc 2 it is modulated by Osc 3. If Osc 2 also uses frequency modulation, Osc 3 is modulated by both Osc 1 and Osc 2.
- The pure sound of frequency modulation is output through the modulator oscillators.

  This means that you should turn off the Osc 1 output when using frequency modulation.
- The **Freq Mod** button activates/deactivates frequency modulation.
- The **Ratio** parameter determines the amount of frequency modulation.

Portamento

This parameter makes the pitch glide between the notes you play. The parameter setting determines the time it takes for the pitch to glide from one note to the next. Turn the knob clockwise for longer glide time.

The **Mode** switch allows you to apply glide only if you play a legato note. Legato mode only works with monophonic parts.

Ring Modulation

Ring modulators multiply two audio signals. The ring-modulated output contains added frequencies generated by the sum of, and the difference between, the frequencies of the two signals. In Prologue, Osc 1 is multiplied with Osc 2 to produce sum and difference frequencies. Ring modulation is often used to create bell-like sounds.

- To hear the ring modulation, turn down the output level for Osc 1 and 2, and turn up the **R.Mod** level all the way.
- If Osc 1 and 2 are tuned to the same frequency and no modulation is applied to the Osc 2 pitch, nothing happens.

  However, if you change the pitch of Osc 2, drastic changes in timbre can be heard. If the oscillators are tuned to a harmonic interval, such as fifth or octave, the ring modulated output sounds harmonic, other intervals produce inharmonious, complex timbres.
- Deactivate oscillator sync when using ring modulation.

Noise Generator

A noise generator can be used for simulating drum sounds and breath sounds for wind instruments, for example.

- To hear only the sound of the noise generator, turn down the output level for the oscillators, and turn up the **Noise** parameter.
- The noise generator level is routed to Envelope 1 by default.

RELATED LINKS

Envelope Page on page 236

Filter Section

The circle in the middle contains the filter parameters. The central control sets the filter cutoff and the outer ring the filter type.

Filter type

Sets the filter type to low-pass, high-pass, band-pass, or notch.

Cutoff

Controls the filter frequency or cutoff. If a low-pass filter is used, it can control the opening and closing of the filter, producing the classic sweeping synthesizer sound. How this parameter operates is governed by the filter type.

Emphasis

This is the resonance control for the filter. For low-pass and high-pass filters, raising the **Emphasis** value emphasizes the frequencies around the set cutoff frequency. This produces a generally thinner sound, but with a sharper, more pronounced cutoff sweep. The higher the filter **Emphasis** value, the more resonant the sound becomes until it starts to self-oscillate, generating a distinct pitch. For band-pass or notch filters, the Emphasis setting adjusts the width of the band. If you raise the value, the band where frequencies are let through (band-pass), or cut (notch) becomes narrower.

Drive

Adjusts the filter input level. Levels above 0 dB gradually introduce a soft distortion of the input signal, and decrease the filter resonance.

Shift

Internally, each filter consists of two or more subfilters connected in series. This parameter shifts the cutoff frequency of the subfilters. The result depends on the filter type: For low-pass and high-pass filter types, it changes the filter slope. For band-pass and notch filter types, it changes the bandwidth. The Shift parameter has no effect for the filter types **12 dB LP** or **12 dB HP**.

Tracking

If this parameter is set to values over the 12 o’clock position, the filter cutoff frequency increases the further up on the keyboard you play. Negative values invert this relationship.

If the **Tracking** parameter is set fully clockwise, the cutoff frequency tracks the keyboard by a semitone per key.

About the Filter Types

You select the filter type using the buttons around the filter cutoff knob. The following filter types are available (listed clockwise starting from the 9 o’clock position):

12 dB LP

Low-pass filters let low frequencies pass and cut out the high frequencies. This low-pass filter has a gentler slope (12 dB/octave above the cutoff frequency), leaving more of the harmonics in the filtered sound.

18 dB LP

This low-pass filter also has a cascade design, attenuating frequencies above the cutoff frequency with a 18 dB/octave slope, as used in the classic TB 303 synth.

24 dB LP

This filter type attenuates frequencies above the cutoff frequency with a 24 dB/octave slope that produces a warm and fat sound.

24 dB LP II

This low-pass filter has a cascade design that attenuates frequencies above the cutoff frequency with a 24 dB/octave slope, which produces a warm and dark sound.

12 dB Band

This band-pass filter cuts both high and low frequencies above and below the cutoff frequency with a 12 dB/octave slope, producing a nasal and thin sound.

12 dB Notch

This notch filter cuts off frequencies near the cutoff frequency by 12 dB/octave, letting the frequencies below and above through. This produces a phaser-like sound.

12 dB HP

A high-pass filter cuts out the lower frequencies and lets the high frequencies pass. This high-pass filter has a 12 dB/octave slope, producing a bright and thin sound.

24 dB HP

This filter has a 24 dB/octave slope, producing a bright and sharp sound.

Master Volume and Pan

The master **Volume** knob controls the master volume (amplitude) of the instrument. By default, this parameter is controlled by Envelope 1, to generate an amplitude envelope for the oscillators.

The **Pan** knob controls the position of the instrument in the stereo spectrum. You can use **Pan** as a modulation destination.

Modulation and Controllers

The lower half of the control panel displays the various modulation and controller assignment pages available, as well as the **EFX** page. You switch between these pages using the buttons above this section.

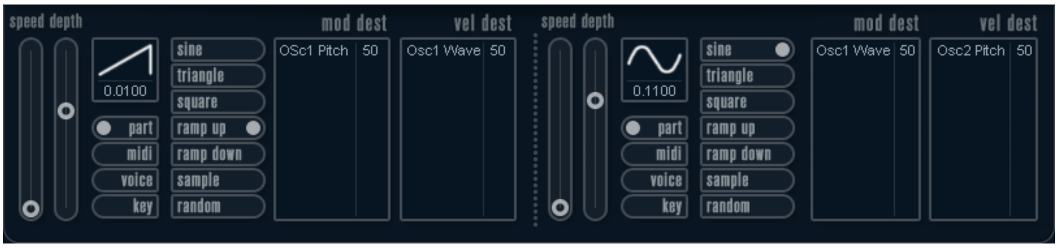
The following pages are available:

- The **LFO** page has two low frequency oscillators (LFOs) for modulating parameters.
- The **ENV** page contains the four envelope generators that can be assigned to control parameters.
- The **Event** page contains the common MIDI controllers (Mod wheel, Aftertouch, etc.) and their assignments.
- The **EFX** page offers three separate effect types: Distortion, Delay, and Modulation.

RELATED LINKS
LFO Page on page 234
Envelope Page on page 236
Event Page on page 238
Effects (EFX) Page on page 239

LFO Page

The LFO page is opened by clicking the **LFO** button at the top of the lower half of the control panel. The page contains all parameters and the modulation and velocity destinations for two independent LFOs.



Depending on the selected preset, there may already be modulation destinations assigned, in which case these are listed in the **Mod Dest** box for each LFO.

A low frequency oscillator (LFO) is used for modulating parameters, for example the pitch of an oscillator (to produce vibrato), or for any parameter where cyclic modulation is required.

The two LFOs have identical parameters.

Speed

Governs the rate of the LFO. If the sync mode is set to **MIDI**, the available rate values are selectable as note values, so the rate is synchronized to the sequencer tempo.

Depth

Controls the amount of modulation applied by the LFO. If this is set to zero, no modulation is applied.

Waveform

Sets the LFO waveform.

Sync mode (Part/MIDI/Voice/Key)

Sets the sync mode for the LFO.

RELATED LINKS
Assigning LFO Modulation Destinations on page 235

About the Sync Modes

The sync modes determine how the LFO cycle affects the notes you play.

Part

In this mode, the LFO cycle is free running and affects all the voices in sync. Free running means that the LFO cycles continuously, and does not reset when a note is played.

MIDI

In this mode, the LFO rate is synced in various beat increments to MIDI clock.

Voice

In this mode, each voice in the Part has its own independent LFO cycle (the LFO is polyphonic). These cycles are also free running – each key down starts anywhere in the LFO cycle phase.

Key

Same as **Voice** except that it is not free running – for each key down the LFO cycle starts over.

About the Waveforms

Most standard LFO waveforms are available for LFO modulation. You use sine and triangle waveforms for smooth modulation cycles, square and ramp up/down for different types of stepped modulation cycles and random or sample for random modulation. The sample waveform is different:

- In this mode, the LFO makes use of the other LFO as well.

  For example, if LFO 2 is set to use **Sample**, the resulting effect also depends on the speed and waveform of LFO 1.

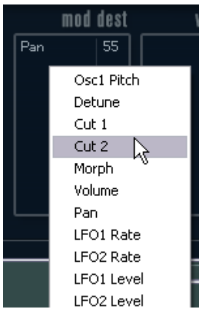
Assigning LFO Modulation Destinations

You can assign a modulation destination for an LFO.

PROCEDURE

1. Click in the **Mod Dest** box for one of the LFOs.

   A pop-up menu appears in which all possible modulation destinations are shown. All Sound parameters as well as most LFO and Envelope parameters are available as destinations.

2. Select a destination, for example, **Cut**.

   The selected modulation destination is now shown in the list. Beside the destination, a default value (50) has been set. The value represents the modulation amount.

   - You can set positive and negative modulation values by clicking on the value in the list, typing in a new value, and pressing **Enter**.

   To enter negative values, type a minus sign followed by the value.

3. Select a suitable LFO Waveform, Speed, Depth, and sync mode.

   You should now hear the **Cut** parameter being modulated by the LFO.

4. Using the same basic method, you can add any number of modulation destinations for the LFO.

   They are all listed in the **Mod Dest** box.

   - To remove a modulation destination, click on its name in the list and select **Off** from the pop-up menu.

Assigning LFO Velocity Destinations

You can also assign velocity-controlled LFO modulation.

PROCEDURE

1. Click in the **Vel Dest** box for one of the LFOs.

   A pop-up menu appears in which all possible velocity destinations are shown.

2. Select a destination.

   The selected velocity destination is now shown in the list. Beside the destination, a default value (50) has been set. The value represents the modulation amount.

- You can set positive and negative values by clicking on the value in the list, typing in a new value, and pressing **Enter**.

To enter negative values, type a minus sign followed by the value.

3. Using the same basic method, you can add any number of velocity destinations for the LFO.

They are all listed in the **Vel Dest** box.

- To remove a velocity destination, click on its name in the list and select **Off** from the pop-up menu.

LFO modulation velocity control

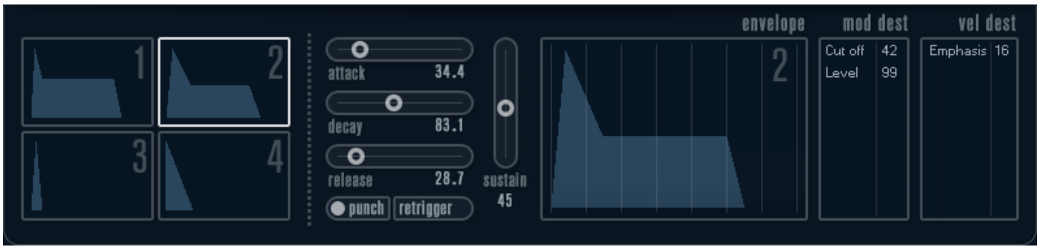
If you follow the steps above and select the **Cut** parameter as a Velocity destination, the following happens:

- The harder you strike the key, the more the **Cut** parameter is modulated by the LFO.
- If you enter a negative value for the velocity modulation amount, the opposite happens: the harder you play, the less the **Cut** parameter is modulated by the LFO.

Envelope Page

The Envelope page is opened by clicking the **ENV** button at the top of the lower half of the control panel. The page contains all parameters and the modulation and velocity destinations for the four independent envelope generators.

Envelope generators govern how a parameter value changes when a key is pressed, when a key is held and finally when a key is released.



On the Envelope page, the parameters for one of the four envelope generators is shown at a time.

- You switch between the four envelopes in the section to the left.

  Clicking on either of the four mini curve displays selects it and displays the corresponding envelope parameters to the right.
- Envelope generators have four parameters: **Attack**, **Decay**, **Sustain**, and **Release** (ADSR).
- You can set envelope parameters in 2 ways: by using the sliders or by clicking and dragging the curve in the Envelope curve display.

  You can also do this in the mini curve displays.
- By default, Envelope 1 is assigned to the master volume, and therefore acts as an amplitude envelope. The amplitude envelope adjusts how the volume of the sound changes from the time you press a key until the key is released.

  If no amplitude envelope is assigned, there is no output.
- Envelope 2 is by default assigned to the **Level** parameter.

The Envelope parameters are as follows:

Attack

The attack phase is the time it takes from zero to the maximum value. How long this takes is governed by the **Attack** setting. If the **Attack** is set to 0, the maximum value is reached instantly. If this value is raised, it takes time before the maximum value is reached. Range is from 0.0 milliseconds to 91.1 seconds.

Decay

After the maximum value has been reached, the value starts to drop. How long this takes is governed by the **Decay** parameter. The **Decay** has no effect if the **Sustain** parameter is set to maximum.

Sustain

Determines the level for the envelope after the **Decay** phase. Note that **Sustain** represents a level, whereas the other envelope parameters represent times.

Release

Determines the time it takes for the value to fall back to zero after releasing the key. Range is from 0.0 milliseconds to 91.1 seconds.

Punch

If **Punch** is activated, the start of the decay phase is delayed a few milliseconds, that is, the envelope stays at top level for a moment before moving on to the decay phase. The result is a punchier attack similar to a compressor effect. This effect is more pronounced with short attack and decay times.

Retrigger

If **Retrigger** is activated, the envelope retriggers each time you play a new note. However, with certain textures/pad sounds and a limited number of voices, it is recommended to leave the button deactivated, due to click noises that might occur.

Assigning Envelope Modulation Destinations

You can assign a modulation destination for an envelope.

PROCEDURE

1. Click in the **Mod Dest** box for one of the envelopes.

   A pop-up menu appears in which all possible modulation destinations are shown. All Sound parameters as well as most LFO and Envelope parameters are available as destinations.

2. Select a destination, for example, **Cut**.

   The selected modulation destination is now shown in the list. Beside the destination, a default value (50) has been set. The value represents the modulation amount.

   - You can set positive and negative modulation values by clicking on the value in the list, typing in a new value, and pressing **Enter**.

     To enter negative values, type a minus sign followed by the value.

3. Select a suitable envelope curve for the modulation.

   You should now hear the **Cut** parameter being modulated by the envelope as you play.

4. Using the same basic method, you can add any number of modulation destinations for the envelope.

   They are all listed in the **Mod Dest** box.

   - To remove a modulation destination, click on its name in the list and select **Off** from the pop-up menu.

Assigning Envelope Velocity Destinations

You can also assign velocity-controlled envelope modulation, that is, the modulation is governed by how hard or soft you strike a key.

PROCEDURE

1. Click in the **Vel Dest** box for one of the envelopes.
   A pop-up menu appears in which all possible velocity destinations are shown.
2. Select a destination.
   The selected velocity destination is now shown in the list. Beside the destination, a default value (50) has been set. The value represents the modulation amount. See below for an example of how velocity modulation works.
   - You can set positive and negative values by clicking on the value in the list, typing in a new value, and pressing **Enter**.
     To enter negative values, type a minus sign followed by the value.
3. Using the same basic method, you can add any number of velocity destinations for the Envelope.
   They are all listed in the **Vel Dest** box.
   - To remove a velocity destination, click on its name in the list and select **Off** from the pop-up menu.

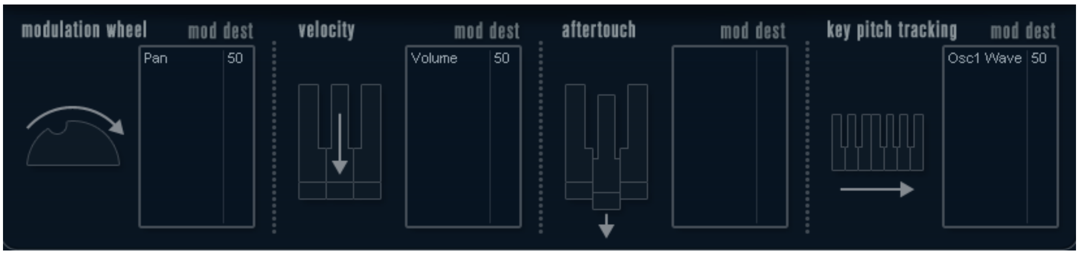
Envelope modulation velocity control

If you follow the steps above and select the **Cut** parameter as a Velocity destination, the following happens:

- The harder you strike the key, the more the parameter is modulated by the envelope.
- If you enter a negative value for the velocity modulation amount, the opposite happens; the harder you play the less the **Cut** parameter is modulated by the Envelope.

Event Page

The Event page is opened by clicking the **EVENT** button at the top of the lower half of the control panel. This page contains the most common MIDI controllers and their assignments.



Modulation Wheel
The modulation wheel on your keyboard can be used to modulate parameters.

Velocity
Controls parameters according to how hard or soft you play notes on your keyboard. A common application of velocity is to make sounds brighter and louder if you strike the key harder.

Aftertouch
Aftertouch, or channel pressure, is MIDI data sent when pressure is applied to a keyboard after the key has been struck, and while it is being held down or sustained.

Aftertouch is often routed to control filter cutoff, volume, and other parameters to add expression.

Key Pitch Tracking

This can change parameter values linearly according to where on the keyboard you play.

Assigning a Controller to a Parameter

PROCEDURE

1. Click in the **Mod Dest** box for one of the controllers.

   A pop-up menu appears in which all possible modulation destinations are shown. All Sound parameters as well as most LFO and Envelope parameters are available as destinations.

2. Select a destination.

   The selected modulation destination is now shown in the list. Beside the destination, a default value (50) has been set. The value represents the modulation amount when the controller is at its full range.

   - You can set positive and negative modulation values by clicking on the value in the list, typing in a new value, and pressing **Enter**.

     To enter negative values, type a minus sign followed by the value.

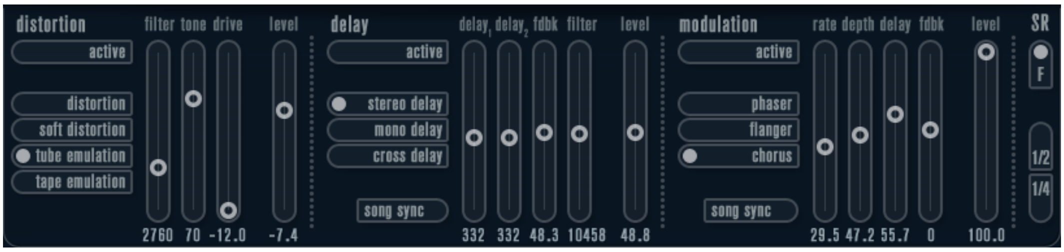
3. Using the same basic method, you can add any number of modulation destinations for the controllers.

   They are all listed in the **Mod Dest** box for each controller.

   - To remove a modulation destination, click on its name in the list and select **Off** from the pop-up menu.

Effects (EFX) Page

This page features three separate effect units: **Distortion**, **Delay**, and **Modulation** (Phaser/ Flanger/Chorus). The Effect page is opened by clicking the **EFX** button at the top of the lower half of the control panel.



- Each separate effect section is laid out with a row of buttons that determine the effect type or characteristic and a row of sliders for making parameter settings.
- To activate an effect, click the **Active** button so that a dot appears.

  Clicking again deactivates the effect.

Distortion

You can choose between 4 basic distortion characteristics:

- **Distortion** provides hard clipping distortion.
- **Soft Distortion** provides soft clipping distortion.
- **Tape Emulation** produces distortion similar to magnetic tape saturation.
- **Tube Emulation** produces distortion similar to valve amplifiers.

Drive

Sets the amount of distortion by amplifying the input signal.

Filter

Sets the crossover frequency of the distortion filter. The distortion filter consists of a low-pass filter and a high-pass filter with a cutoff frequency equal to the crossover frequency.

Tone

Controls the relative amount of low-pass and high-pass filtered signal.

Level

Controls the output level of the effect.

Delay

You can choose between 3 basic delay characteristics:

- **Stereo Delay** has two separate delay lines panned left and right.
- In **Mono Delay**, the two delay lines are connected in series for monophonic dual tap delay effects.
- In **Cross Delay**, the delayed sound bounces between the stereo channels.

Song Sync

Activates/Deactivates tempo sync of the delay times.

Delay 1

Sets the delay time ranging from 0 ms to 728 ms. If **MIDI sync** is activated, the range is from 1/32 to 1/1; straight, triplet or dotted.

Delay 2

Same as **Delay 1**.

Feedback

Controls the decay of the delays. With higher settings, the echoes repeat longer.

Filter

A low-pass filter is built into the feedback loop of the delay. This parameter controls the cutoff frequency of this feedback filter. Low settings result in successive echoes sounding darker.

Level

Controls the output level of the effect.

Modulation

You can choose between 3 basic modulation characteristics:

- **Phaser** uses an 8-pole all-pass filter to produce the classic phasing effect.
- **Flanger** is composed of two independent delay lines with feedback for the left and the right channel. The delay time of both delays is modulated by one LFO with adjustable frequency.
- **Chorus** produces a rich chorus effect with 4 delays modulated by four independent LFOs.

Song Sync

Activates/Deactivates tempo sync of the **Rate** parameter.

Rate

Sets the rate of the LFOs modulating the delay time. If **Song Sync** is activated, the rate is synchronized to various beat increments.

Depth

Controls the depth of the delay time modulation.

Delay

Sets the delay time of the four delay lines.

Feedback

Controls the amount of positive or negative feedback for all four delay lines.

Level

Controls the output level of the effect.

SR Parameters

With these buttons, you can change the sample rate. Lower sample rates basically reduce the high frequency content and sound quality, but the pitch is not altered. This is useful to emulate the lo-fi sounds of older digital synths.

- If the **F** button is active, the program of the selected part plays back with the sample rate set in the host application.
- If the **1/2** button is active, the program of the selected part plays back with half the original sample rate.
- If the **1/4** button is active, the program of the selected part plays back with a quarter of the original sample rate.

A bonus effect of using lower sample rates is that it reduces the load on the computer CPU, allowing for more simultaneous voices to be played, etc.

Retrologue

This VST instrument is described in detail in the separate document **Retrologue**.

Spector

The synthesis used by **Spector** is based around a spectrum filter. This allows you to specify the frequency response by drawing a filter contour in the spectrum display. Slightly simplified, the signal path is the following:

- The starting point is the sound generated by up to 6 oscillators.

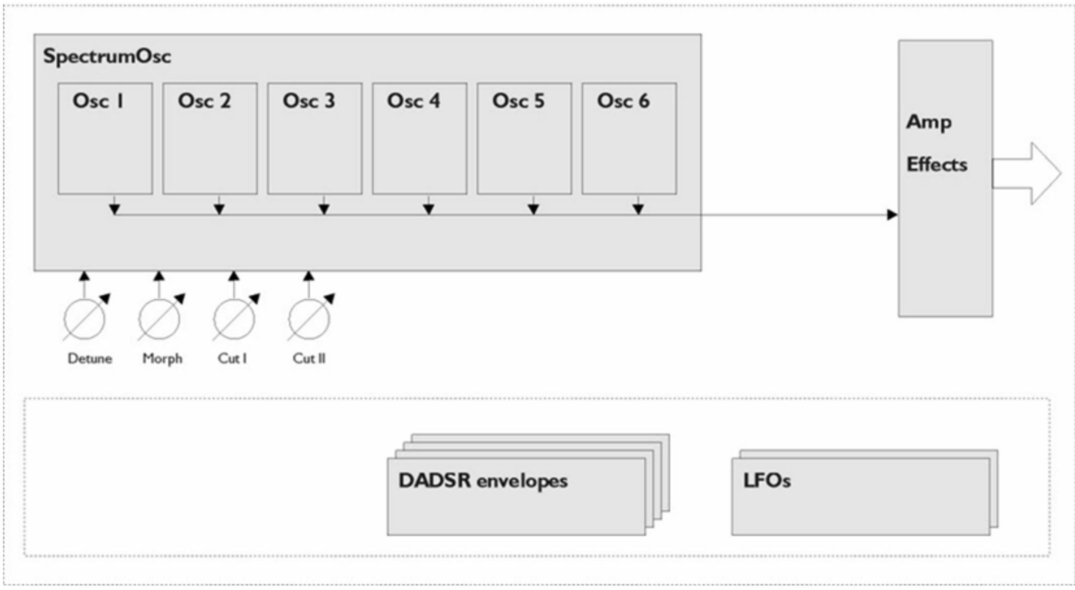
  You can choose between different numbers of oscillators in different configurations (in octaves, in unison, etc.). The oscillators can also be detuned for fat sounds or extreme special effects.
- Each oscillator produces two basic waveforms, labeled A and B.

  You can choose between six different waveforms, independently selected for A and B.
- The two waveforms pass through separate spectrum filters (A and B).

  You can draw different spectrum contours for the two filters, or select a contour from the included presets.
- The **Cut 1 & 2** parameters allow you to shift the frequency range of the spectrum filter.

  This makes it easy to create unique-sounding filter sweeps.
- A **Morph** control lets you mix the output of spectrum filters A and B.

  Since this can be controlled with envelopes, LFOs, etc. That allows you to create morphing effects.
- Controllers and modulation parameters are also available.



Functional Diagram

Sound Parameters

Oscillator Section

A/B Waveform Pop-up Menus

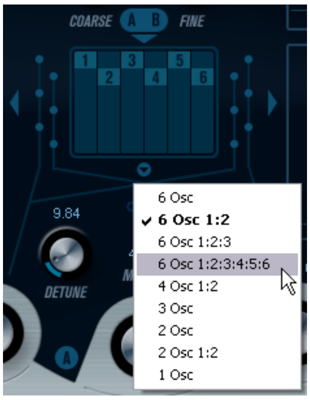
This is where you select basic waveforms for the A and B output of the oscillators. The options are best suited for use with the spectrum filter.

Coarse and Fine

These parameters provide overall transposition and tuning of the oscillators (common for all oscillators, A and B waveforms).

Oscillator Pop-up Menu

This pop-up menu is opened by clicking on the arrow below the central section (which illustrates the selected oscillator configuration).



6 Osc
: 6 oscillators with the same pitch.

6 Osc 1:2
: 3 oscillators with base pitch and 3 pitched one octave down.

6 Osc 1:2:3
: Three groups of two oscillators with the pitch ratio 1:2:3 (2 oscillators with base pitch, 2 oscillators at half the frequency of the base pitch, and 2 oscillators at a third of the frequency).

6 Osc 1:2:3:4:5:6
: 6 oscillators tuned with the pitch ratio 1:2:3:4:5:6 (known as the subharmonic series).

4 Osc 1:2
: 2 oscillators with base pitch and 2 pitched one octave down.

3 Osc
: 3 oscillators with the same pitch.

2 Osc
: 2 oscillators with the same pitch.

2 Osc 1:2
: One oscillator with base pitch and one pitched one octave down.

1 Osc
: A single oscillator. In this mode, the **Detune** and **Cut II** parameters are not active.

Detune

Detunes the oscillators. Low values give gentle chorus-like detuning. Raising the control detunes the oscillators by several semitones for special effects.

Raster

Reduces the number of harmonics present in the oscillator waveforms in the following manner:

- If **0** is selected, all harmonics are present.
- If **1** is selected, only every second harmonic is present.
- If **2** is selected, only every third harmonic is present.

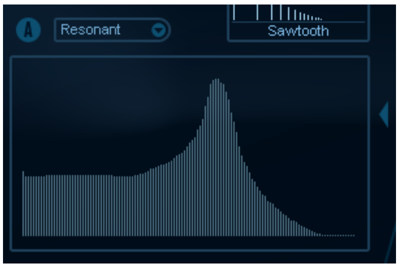
And so on.

Portamento

This parameter makes the pitch glide between the notes you play. The parameter setting determines the time it takes for the pitch to glide from one note to the next. Turn the knob clockwise for longer glide time.

The **Mode** switch allows you to apply glide only if you play a legato note. Legato mode only works with monophonic parts.

Spectrum Filter Sections



This is where you create the contours, that is, frequency response characteristics, for the two 128 pole resonant spectrum filters A and B.

- You can use the Preset pop-up menu to select a preset contour.
- To change the contour, click and draw with the mouse.
- If you want to calculate a random spectrum filter curve, select **Randomize** from the Preset pop-up menu.
  Each time you choose this function, a new randomized spectrum is calculated.

Cut I and II

These parameters work like cutoff frequency controls on a conventional filter: With the **Cut** controls at the maximum setting, the full frequency range is used for the spectrum filter. Lowering the **Cut** controls gradually moves the entire contour down in frequency, closing the filter.

NOTE

- If a 2 oscillator configuration is used, you can set different cutoffs for the two oscillators. If more than two oscillators are used, they are internally divided into two groups, for which you can set independent cutoffs with **Cut I** and **Cut II**.
- If the **Spectrum Sync** button (link symbol) between the cut controls is activated, the two knobs are linked and follow each other and are set to the same value.

Morph

Controls the mix between the sound of spectrum filters A and B. If the **Morph** knob is turned fully left, only the A sound is heard. If it is turned right only the B sound is heard. This allows you to seamlessly morph between two totally different sounds.

Master Volume and Pan

The master **Volume** knob controls the master volume (amplitude) of the instrument. By default, this parameter is controlled by Envelope 1, to generate an amplitude envelope for the oscillators.

The **Pan** knob controls the position of the instrument in the stereo spectrum. You can use **Pan** as a modulation destination.

Modulation and Controllers

The lower half of the control panel displays the various modulation and controller assignment pages available, as well as the **EFX** page. You switch between these pages using the buttons above this section.

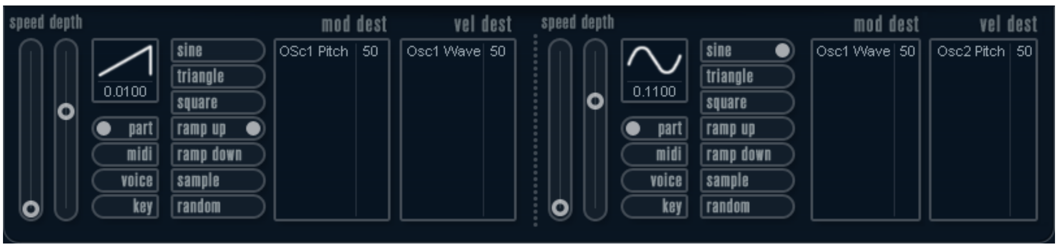
The following pages are available:

- The **LFO** page has two low frequency oscillators (LFOs) for modulating parameters.
- The **ENV** page contains the four envelope generators that can be assigned to control parameters.
- The **Event** page contains the common MIDI controllers (Mod wheel, Aftertouch, etc.) and their assignments.
- The **EFX** page offers three separate effect types: Distortion, Delay, and Modulation.

RELATED LINKS
LFO Page on page 234
Envelope Page on page 236
Event Page on page 238
Effects (EFX) Page on page 239

LFO Page

The LFO page is opened by clicking the **LFO** button at the top of the lower half of the control panel. The page contains all parameters and the modulation and velocity destinations for two independent LFOs.



Depending on the selected preset, there may already be modulation destinations assigned, in which case these are listed in the **Mod Dest** box for each LFO.

A low frequency oscillator (LFO) is used for modulating parameters, for example the pitch of an oscillator (to produce vibrato), or for any parameter where cyclic modulation is required.

The two LFOs have identical parameters.

Speed

Governs the rate of the LFO. If the sync mode is set to **MIDI**, the available rate values are selectable as note values, so the rate is synchronized to the sequencer tempo.

Depth

Controls the amount of modulation applied by the LFO. If this is set to zero, no modulation is applied.

Waveform

Sets the LFO waveform.

Sync mode (Part/MIDI/Voice/Key)

Sets the sync mode for the LFO.

RELATED LINKS

Assigning LFO Modulation Destinations on page 235

About the Sync Modes

The sync modes determine how the LFO cycle affects the notes you play.

Part

In this mode, the LFO cycle is free running and affects all the voices in sync. Free running means that the LFO cycles continuously, and does not reset when a note is played.

MIDI

In this mode, the LFO rate is synced in various beat increments to MIDI clock.

Voice

In this mode, each voice in the Part has its own independent LFO cycle (the LFO is polyphonic). These cycles are also free running – each key down starts anywhere in the LFO cycle phase.

Key

Same as **Voice** except that it is not free running – for each key down the LFO cycle starts over.

About the Waveforms

Most standard LFO waveforms are available for LFO modulation. You use sine and triangle waveforms for smooth modulation cycles, square and ramp up/down for different types of stepped modulation cycles and random or sample for random modulation. The sample waveform is different:

- In this mode, the LFO makes use of the other LFO as well.

  For example, if LFO 2 is set to use **Sample**, the resulting effect also depends on the speed and waveform of LFO 1.

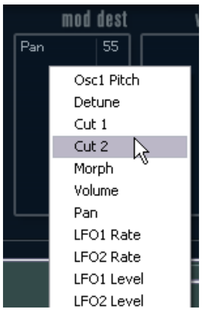
Assigning LFO Modulation Destinations

You can assign a modulation destination for an LFO.

PROCEDURE

1. Click in the **Mod Dest** box for one of the LFOs.

   A pop-up menu appears in which all possible modulation destinations are shown. All Sound parameters as well as most LFO and Envelope parameters are available as destinations.



2. Select a destination, for example, **Cut**.

   The selected modulation destination is now shown in the list. Beside the destination, a default value (50) has been set. The value represents the modulation amount.

   - You can set positive and negative modulation values by clicking on the value in the list, typing in a new value, and pressing **Enter**.

   To enter negative values, type a minus sign followed by the value.

3. Select a suitable LFO Waveform, Speed, Depth, and sync mode.

   You should now hear the **Cut** parameter being modulated by the LFO.

4. Using the same basic method, you can add any number of modulation destinations for the LFO.

   They are all listed in the **Mod Dest** box.

   - To remove a modulation destination, click on its name in the list and select **Off** from the pop-up menu.

Assigning LFO Velocity Destinations

You can also assign velocity-controlled LFO modulation.

PROCEDURE

1. Click in the **Vel Dest** box for one of the LFOs.

   A pop-up menu appears in which all possible velocity destinations are shown.

2. Select a destination.

   The selected velocity destination is now shown in the list. Beside the destination, a default value (50) has been set. The value represents the modulation amount.

   - You can set positive and negative values by clicking on the value in the list, typing in a new value, and pressing **Enter**.

   To enter negative values, type a minus sign followed by the value.

3. Using the same basic method, you can add any number of velocity destinations for the LFO.

   They are all listed in the **Vel Dest** box.

   - To remove a velocity destination, click on its name in the list and select **Off** from the pop-up menu.

LFO modulation velocity control

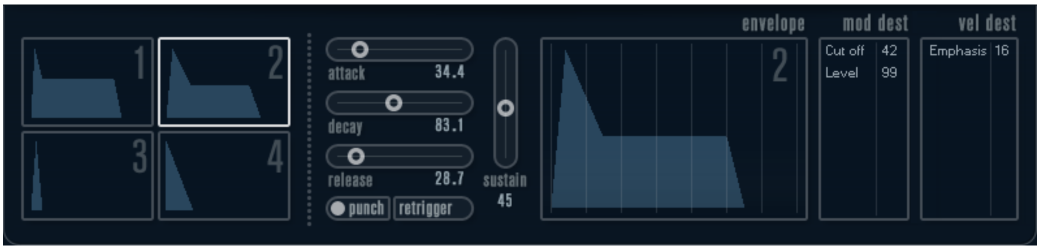
If you follow the steps above and select the **Cut** parameter as a Velocity destination, the following happens:

- The harder you strike the key, the more the **Cut** parameter is modulated by the LFO.
- If you enter a negative value for the velocity modulation amount, the opposite happens: the harder you play, the less the **Cut** parameter is modulated by the LFO.

Envelope Page

The Envelope page is opened by clicking the **ENV** button at the top of the lower half of the control panel. The page contains all parameters and the modulation and velocity destinations for the four independent envelope generators.

Envelope generators govern how a parameter value changes when a key is pressed, when a key is held and finally when a key is released.



On the Envelope page, the parameters for one of the four envelope generators is shown at a time.

- You switch between the four envelopes in the section to the left.

  Clicking on either of the four mini curve displays selects it and displays the corresponding envelope parameters to the right.
- Envelope generators have four parameters: **Attack**, **Decay**, **Sustain**, and **Release** (ADSR).
- You can set envelope parameters in 2 ways: by using the sliders or by clicking and dragging the curve in the Envelope curve display.

  You can also do this in the mini curve displays.
- By default, Envelope 1 is assigned to the master volume, and therefore acts as an amplitude envelope. The amplitude envelope adjusts how the volume of the sound changes from the time you press a key until the key is released.

  If no amplitude envelope is assigned, there is no output.
- Envelope 2 is by default assigned to the **Level** parameter.

The Envelope parameters are as follows:

Attack

The attack phase is the time it takes from zero to the maximum value. How long this takes is governed by the **Attack** setting. If the **Attack** is set to 0, the maximum value is reached instantly. If this value is raised, it takes time before the maximum value is reached. Range is from 0.0 milliseconds to 91.1 seconds.

Decay

After the maximum value has been reached, the value starts to drop. How long this takes is governed by the **Decay** parameter. The **Decay** has no effect if the **Sustain** parameter is set to maximum.

Sustain

Determines the level for the envelope after the **Decay** phase. Note that **Sustain** represents a level, whereas the other envelope parameters represent times.

Release

Determines the time it takes for the value to fall back to zero after releasing the key. Range is from 0.0 milliseconds to 91.1 seconds.

Punch

If **Punch** is activated, the start of the decay phase is delayed a few milliseconds, that is, the envelope stays at top level for a moment before moving on to the decay phase. The result is a punchier attack similar to a compressor effect. This effect is more pronounced with short attack and decay times.

Retrigger

If **Retrigger** is activated, the envelope retriggers each time you play a new note. However, with certain textures/pad sounds and a limited number of voices, it is recommended to leave the button deactivated, due to click noises that might occur.

Assigning Envelope Modulation Destinations

You can assign a modulation destination for an envelope.

PROCEDURE

1. Click in the **Mod Dest** box for one of the envelopes.

   A pop-up menu appears in which all possible modulation destinations are shown. All Sound parameters as well as most LFO and Envelope parameters are available as destinations.

2. Select a destination, for example, **Cut**.

   The selected modulation destination is now shown in the list. Beside the destination, a default value (50) has been set. The value represents the modulation amount.

   - You can set positive and negative modulation values by clicking on the value in the list, typing in a new value, and pressing **Enter**.

     To enter negative values, type a minus sign followed by the value.

3. Select a suitable envelope curve for the modulation.

   You should now hear the **Cut** parameter being modulated by the envelope as you play.

4. Using the same basic method, you can add any number of modulation destinations for the envelope.

   They are all listed in the **Mod Dest** box.

   - To remove a modulation destination, click on its name in the list and select **Off** from the pop-up menu.

Assigning Envelope Velocity Destinations

You can also assign velocity-controlled envelope modulation, that is, the modulation is governed by how hard or soft you strike a key.

PROCEDURE

1. Click in the **Vel Dest** box for one of the envelopes.

   A pop-up menu appears in which all possible velocity destinations are shown.

2. Select a destination.

   The selected velocity destination is now shown in the list. Beside the destination, a default value (50) has been set. The value represents the modulation amount. See below for an example of how velocity modulation works.

   - You can set positive and negative values by clicking on the value in the list, typing in a new value, and pressing **Enter**.

     To enter negative values, type a minus sign followed by the value.

3. Using the same basic method, you can add any number of velocity destinations for the Envelope.

   They are all listed in the **Vel Dest** box.

   - To remove a velocity destination, click on its name in the list and select **Off** from the pop-up menu.

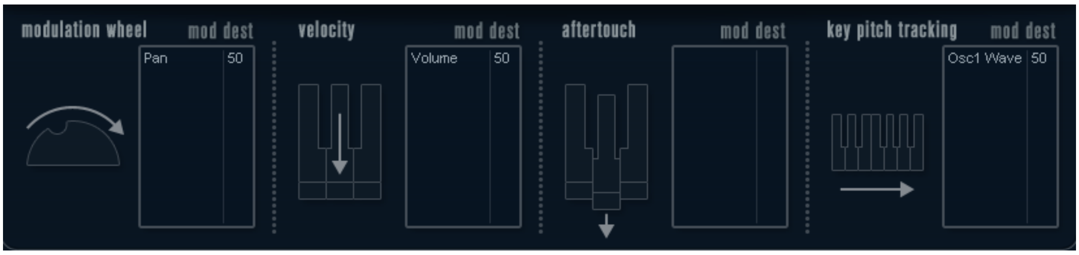
Envelope modulation velocity control

If you follow the steps above and select the **Cut** parameter as a Velocity destination, the following happens:

- The harder you strike the key, the more the parameter is modulated by the envelope.
- If you enter a negative value for the velocity modulation amount, the opposite happens; the harder you play the less the **Cut** parameter is modulated by the Envelope.

Event Page

The Event page is opened by clicking the **EVENT** button at the top of the lower half of the control panel. This page contains the most common MIDI controllers and their assignments.



Modulation Wheel

The modulation wheel on your keyboard can be used to modulate parameters.

Velocity

Controls parameters according to how hard or soft you play notes on your keyboard. A common application of velocity is to make sounds brighter and louder if you strike the key harder.

Aftertouch

Aftertouch, or channel pressure, is MIDI data sent when pressure is applied to a keyboard after the key has been struck, and while it is being held down or sustained. Aftertouch is often routed to control filter cutoff, volume, and other parameters to add expression.

Key Pitch Tracking

This can change parameter values linearly according to where on the keyboard you play.

Assigning a Controller to a Parameter

PROCEDURE

1. Click in the **Mod Dest** box for one of the controllers.

   A pop-up menu appears in which all possible modulation destinations are shown. All Sound parameters as well as most LFO and Envelope parameters are available as destinations.

2. Select a destination.

   The selected modulation destination is now shown in the list. Beside the destination, a default value (50) has been set. The value represents the modulation amount when the controller is at its full range.

   - You can set positive and negative modulation values by clicking on the value in the list, typing in a new value, and pressing **Enter**.

     To enter negative values, type a minus sign followed by the value.

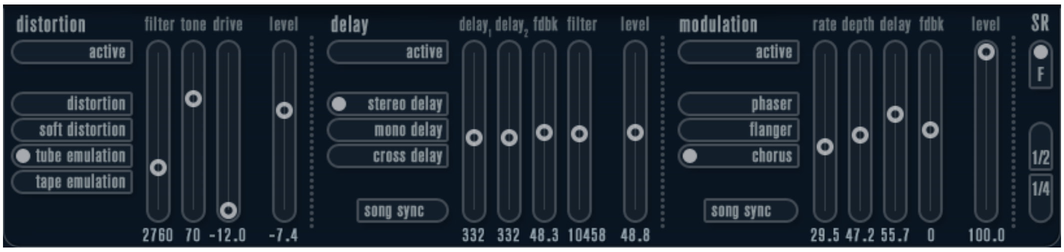
3. Using the same basic method, you can add any number of modulation destinations for the controllers.

   They are all listed in the **Mod Dest** box for each controller.

- To remove a modulation destination, click on its name in the list and select **Off** from the pop-up menu.

Effects (EFX) Page

This page features three separate effect units: **Distortion**, **Delay**, and **Modulation** (Phaser/Flanger/Chorus). The Effect page is opened by clicking the **EFX** button at the top of the lower half of the control panel.



- Each separate effect section is laid out with a row of buttons that determine the effect type or characteristic and a row of sliders for making parameter settings.
- To activate an effect, click the **Active** button so that a dot appears.

  Clicking again deactivates the effect.

Distortion

You can choose between 4 basic distortion characteristics:

- **Distortion** provides hard clipping distortion.
- **Soft Distortion** provides soft clipping distortion.
- **Tape Emulation** produces distortion similar to magnetic tape saturation.
- **Tube Emulation** produces distortion similar to valve amplifiers.

Drive

Sets the amount of distortion by amplifying the input signal.

Filter

Sets the crossover frequency of the distortion filter. The distortion filter consists of a low-pass filter and a high-pass filter with a cutoff frequency equal to the crossover frequency.

Tone

Controls the relative amount of low-pass and high-pass filtered signal.

Level

Controls the output level of the effect.

Delay

You can choose between 3 basic delay characteristics:

- **Stereo Delay** has two separate delay lines panned left and right.
- In **Mono Delay**, the two delay lines are connected in series for monophonic dual tap delay effects.
- In **Cross Delay**, the delayed sound bounces between the stereo channels.

Song Sync

Activates/Deactivates tempo sync of the delay times.

Delay 1

Sets the delay time ranging from 0 ms to 728 ms. If **MIDI sync** is activated, the range is from 1/32 to 1/1; straight, triplet or dotted.

Delay 2

Same as **Delay 1**.

Feedback

Controls the decay of the delays. With higher settings, the echoes repeat longer.

Filter

A low-pass filter is built into the feedback loop of the delay. This parameter controls the cutoff frequency of this feedback filter. Low settings result in successive echoes sounding darker.

Level

Controls the output level of the effect.

Modulation

You can choose between 3 basic modulation characteristics:

- **Phaser** uses an 8-pole all-pass filter to produce the classic phasing effect.
- **Flanger** is composed of two independent delay lines with feedback for the left and the right channel. The delay time of both delays is modulated by one LFO with adjustable frequency.
- **Chorus** produces a rich chorus effect with 4 delays modulated by four independent LFOs.

Song Sync

Activates/Deactivates tempo sync of the **Rate** parameter.

Rate

Sets the rate of the LFOs modulating the delay time. If **Song Sync** is activated, the rate is synchronized to various beat increments.

Depth

Controls the depth of the delay time modulation.

Delay

Sets the delay time of the four delay lines.

Feedback

Controls the amount of positive or negative feedback for all four delay lines.

Level

Controls the output level of the effect.

SR Parameters

With these buttons, you can change the sample rate. Lower sample rates basically reduce the high frequency content and sound quality, but the pitch is not altered. This is useful to emulate the lo-fi sounds of older digital synths.

- If the **F** button is active, the program of the selected part plays back with the sample rate set in the host application.
- If the **1/2** button is active, the program of the selected part plays back with half the original sample rate.
- If the **1/4** button is active, the program of the selected part plays back with a quarter of the original sample rate.

A bonus effect of using lower sample rates is that it reduces the load on the computer CPU, allowing for more simultaneous voices to be played, etc.

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