Plug-in Reference

NUENDO 5

Advanced Post, Live and Audio Production System

steinberg

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Table of Contents

1

The included effect plug-ins

Introduction

This chapter contains descriptions of the included plug-in effects and their parameters.

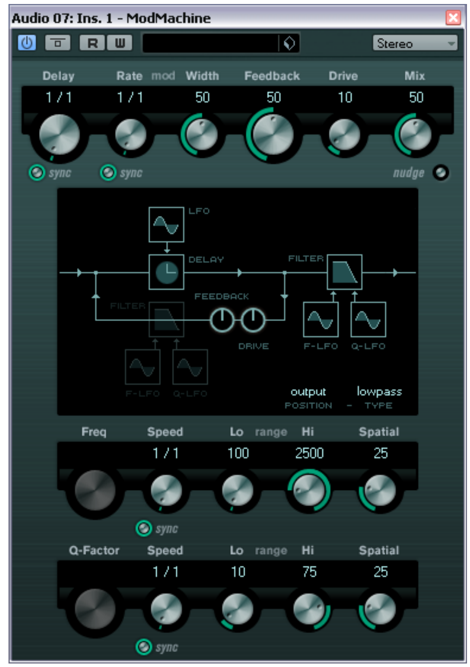
In Nuendo, the plug-in effects are arranged in a number of different categories. This chapter is arranged in the same fashion, with the plug-ins listed in separate sections for each effect category.

⇨ Most of the included effects are compatible with VST3, this is indicated by an icon in front of the name of the plug-in as displayed in plug-in selection menus (for further information, see the chapter “Audio effects” in the Operation Manual).

Delay plug-ins

This section contains descriptions of the plug-ins in the “Delay” category.

ModMachine



ModMachine combines delay modulation and filter frequency/resonance modulation and can provide many interesting modulation effects. It also features a Drive parameter for distortion effects.

The following parameters are available:

Parameter	Description
Delay	This is where you specify the base note value for the delay if tempo sync is on (1/1–1/32, straight, triplet, or dotted). If tempo sync is off, the delay time can be set freely in milliseconds.
Delay – Sync button	The button below the Delay knob is used to switch tempo sync for the Delay parameter on or off.
Rate	The Rate parameter sets the base note value for tempo syncing the delay modulation (1/1 to 1/32, straight, triplet, or dotted). If tempo sync is off, the rate can be set freely.
Rate – Sync button	The button below the Rate knob is used to switch tempo sync for the Rate parameter on or off.

Parameter	Description
Width	Sets the amount of delay pitch modulation. Note that although the modulation affects the delay time, the sound is mostly perceived as a vibrato or chorus-like effect.
Feedback	Sets the number of repeats for the delay.
Drive	Adds distortion to the feedback loop. The longer the Feedback, the more the delay repeats become distorted over time.
Mix	Sets the level balance between the dry signal and the effect. If ModMachine is used as a send effect, set this to the maximum value (100%) as you can control the dry/effect balance with the send.
Nudge button	Clicking the Nudge button once will momentarily speed up the audio coming into the plug-in, simulating an analog tape nudge type sound effect.
Signal path graphic and Filter position	The filter can either be placed in the feedback loop of the delay or in the output path of the effect (after the Drive and Feedback parameters). To switch between the “loop” and “output” positions, click on the Filter section displayed in the graphic or click on the Position field at the bottom right of the graphic.
Filter type (in graphic display)	The Type button allows you to select a filter type. A low-pass, band-pass, and high-pass filter are available.
Freq	Sets the cutoff frequency for the filter. It is only available if tempo sync for the Speed parameter (see below) is deactivated and the parameter is set to “0”.
Speed	Sets the speed of the filter frequency LFO modulation. When using tempo sync, the Speed parameter sets the base note value for tempo syncing the modulation (1/1 to 1/32, straight, triplet, or dotted). If tempo sync is off, the speed can be set freely.
Speed – Sync button	The button below the Speed knob is used to switch tempo sync for the Speed parameter on or off.
Range Lo/Hi	These knobs specify the range (in Hz) of the filter frequency modulation. Both positive (e.g. Lo set to 50 and Hi set to 10000) and negative (e.g. Lo set to 5000 and Hi set to 500) ranges can be set. If tempo sync is off and the Speed is set to zero, these parameters are inactive and the filter frequency is controlled by the Freq parameter instead.
Spatial	Introduces an offset between the channels to create a stereo panorama effect for the filter frequency modulation. Turn clockwise for a more pronounced stereo effect.
Q-Factor	Controls the resonance of the filter. It is only available if filter resonance LFO tempo sync is deactivated and the Speed parameter (see below) is set to “0”. When using tempo sync, the resonance is controlled by the Speed and Range parameters.
Speed	Sets the speed of the filter resonance LFO modulation. When using tempo sync, the Speed parameter sets the base note value for tempo syncing the modulation (1/1 to 1/32, straight, triplet, or dotted). If tempo sync is off, the speed can be set freely.
Speed – Sync button	The button below the Speed knob is used to switch tempo sync for the Speed parameter on or off.

Parameter	Description
Range Lo/Hi	These knobs specify the range of filter resonance modulation. Both positive (e.g. Lo set to 50 and Hi set to 100) and negative (e.g. Lo set to 100 and Hi set to 50) ranges can be set. If tempo sync is off and the Speed is set to zero, these parameters are inactive and the filter resonance is controlled by the Q-Factor parameter instead.
Spatial	Introduces 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 that can either be tempo-based or use freely specified delay time settings.

The following parameters are available:

Parameter	Description
Delay	This is where you specify the base note value for the delay if tempo sync is on (1/1–1/32, straight, triplet, or dotted). If tempo sync is off, it sets the delay time in milliseconds.
Sync button	The button below the Delay knob is used to switch tempo sync on or off.
Feedback	Sets the number of repeats for the delay.
Filter Lo	This filter affects the feedback loop of the effect signal and allows you to roll off low frequencies from 10 Hz up to 800 Hz. The button below the knob activates/deactivates the filter.
Filter Hi	This filter affects the feedback loop of the effect signal and allows you to roll off high frequencies from 20 kHz down to 1.2 kHz. The button below the knob activates/deactivates the filter.
Mix	Sets the level balance between the dry signal and the effect. If MonoDelay is used as a send effect, set this to the maximum value as you can control the dry/effect balance with the send.

⇨ The delay can also be controlled from another signal source via the side-chain input. When the side-chain signal exceeds the threshold, the delay repeats are silenced. When the signal drops below the threshold, the delay repeats reappear. For a description of how to set up side-chain routing, see the chapter “Audio effects” in the Operation Manual.

PingPongDelay

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

The following parameters are available:

Parameter	Description
Delay	This is where you specify the base note value for the delay if tempo sync is on (1/1–1/32, straight, triplet, or dotted). If tempo sync is off, it sets the delay time in milliseconds.
Sync button	The button below the Delay Time knob is used to switch tempo sync on or off.
Feedback	Sets the number of repeats for the delay.
Filter Lo	This filter affects the feedback loop and allows you to roll off low frequencies up to 800 Hz. The button below the knob activates/deactivates the filter.
Filter Hi	This filter affects the feedback loop and allows you to roll off high frequencies from 20 kHz down to 1.2 kHz. The button below the knob activates/deactivates the filter.
Spatial	Sets the stereo width for the left/right repeats. Turn clockwise for a more pronounced stereo “ping-pong” effect.
Mix	Sets the level balance between the dry signal and the effect. If PingPongDelay is used as a send effect, set this to the maximum value as you can control the dry/effect balance with the send.

⇨ The delay can also be controlled from another signal source via the side-chain input. When the side-chain signal exceeds the threshold, the delay repeats are silenced. When the signal drops below the threshold, the delay repeats reappear. For a description of how to set up side-chain routing, see the chapter “Audio effects” in the Operation Manual.

StereoDelay

StereoDelay has two independent delay lines which either use tempo-based or freely specified delay time settings.

The following parameters are available:

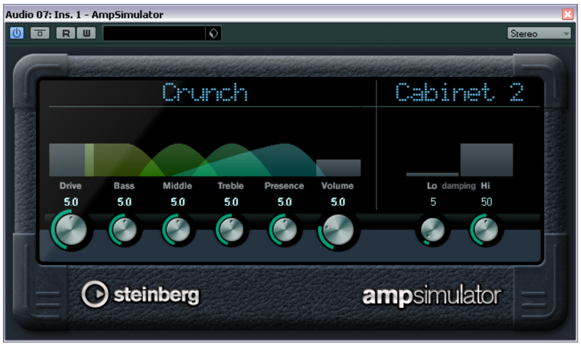
Parameter	Description
Delay 1 & 2	Using these controls you specify the base note value for the delay if tempo sync is on (1/1–1/32, straight, triplet, or dotted). If tempo sync is off, they set the delay time in milliseconds.
Sync button	The buttons below the Delay knobs are used to turn tempo sync on or off for the respective delay.
Feedback 1 & 2	The Feedback controls set the number of repeats for each delay.
Filter Lo 1 & 2	These filters affect the feedback loop and allow you to roll off low frequencies up to 800 Hz. The buttons below the knobs activate/deactivate the filter.
Filter Hi 1 & 2	These filters affect the feedback loop and allow you to roll off high frequencies from 20 kHz down to 1.2 kHz. The buttons below the knobs activate/deactivate the filter.
Pan 1 & 2	These controls are used to set the stereo position for each delay.
Mix 1 & 2	Use these controls to set the level balance between the dry signal and the effect. If StereoDelay is used as a send effect, set them to the maximum value (100%) as you can control the dry/effect balance with the send.

⇨ The delay can also be controlled from another signal source via the side-chain input. When the side-chain signal exceeds the threshold, the delay repeats are silenced. When the signal drops below the threshold, the delay repeats reappear. For a description of how to set up side-chain routing, see the chapter “Audio effects” in the Operation Manual.

Distortion plug-ins

This section contains descriptions of the plug-ins in the “Distortion” category.

AmpSimulator



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

The following parameters are available:

Parameter	Description
Amplifier pop-up menu	This pop-up menu is opened by clicking on the amplifier name shown at the top of the amp section. It allows you to select an amplifier model. The amp section can be bypassed by selecting “No Amp”.
Drive	Controls the amount of amp overdrive.
Bass	Tone control for the low frequencies.
Middle	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.
Cabinet pop-up menu	This pop-up menu is opened by clicking on the cabinet name shown at the top of the cabinet section. It allows you to select a speaker cabinet model. This section can be bypassed by selecting “No Speaker”.
Damping Lo/Hi	Further tone controls for shaping the sound of the selected speaker cabinet. Click on the values, enter a new value and press the [Enter] key.

DaTube

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

The following parameters are available:

Parameter	Description
Drive	Regulates the pre-gain of the “amplifier”. Use high values if you want an overdriven sound just on the verge of distortion.
Balance	Controls the balance between the signal processed by the Drive parameter and the dry input signal. For maximum drive effect, set this to its highest value.
Output	Adjusts the post-gain, or output level, of the “amplifier”.

Distortion

Distortion will add crunch to your tracks.

The following parameters are available:

Parameter	Description
Boost	Increases the distortion amount.
Feedback	Feeds part of the output signal back to the effect input, increasing the distortion effect.
Tone	Lets you select a frequency range to which to apply the distortion effect.
Spatial	Changes the distortion characteristics of the left and right channel, thus creating a stereo effect.
Output	Raises or lowers the signal going out of the effect.

SoftClipper

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

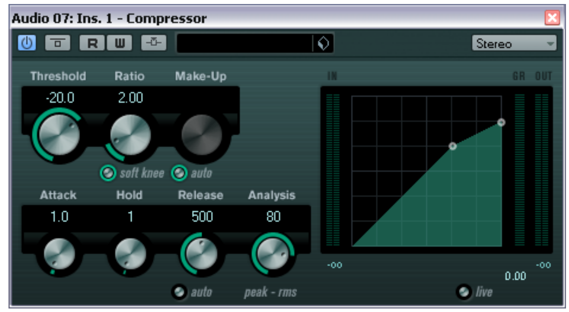
The following parameters are available:

Parameter	Description
Input	Regulates the pre-gain. Use high values if you want an overdriven sound just on the verge of distortion.
Mix	Setting Mix to 0 means that no processed signal is added to the original signal.
Output	Adjusts the post-gain, or output level.
Second	Allows you to adjust the amount of the second harmonic in the processed signal.
Third	Allows you to adjust the amount of the third harmonic in the processed signal.

Dynamics plug-ins

This section contains descriptions of the plug-ins in the “Dynamics” category.

Compressor



Compressor reduces the dynamic range of the audio, making softer sounds louder or louder sounds softer, or both. Compressor features separate controls for threshold, ratio, attack, hold, release and make-up gain parameters. Compressor features a separate display that graphically illustrates the compressor curve 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.

The following parameters are available:

Parameter	Description
Threshold (-60 to 0dB)	Determines the level where Compressor “kicks in”. Signal levels above the set threshold are affected, but signal levels below are not processed.
Ratio (1:1 to 8:1)	Sets the amount of gain reduction applied to signals over the set threshold. A ratio of 3:1 means that for every 3dB the input level increases, the output level will increase by only 1dB.
Soft Knee button	If this button is off, signals above the threshold are compressed instantly according to the set ratio (hard knee). When Soft Knee is activated, the onset of compression is more gradual, producing a less drastic result.
Make-up (0 to 24dB or Auto mode)	This parameter is used to compensate for output gain loss, caused by compression. If the Auto button is activated, the knob becomes dark and the output is automatically adjusted for gain loss.

Parameter	Description
Attack (0.1 to 100 ms)	Determines how fast Compressor will respond to signals above the set threshold. If the attack time is long, more of the early part of the signal (attack) passes through unprocessed.
Hold (0 to 5000 ms)	Sets the time the applied compression will affect the signal after exceeding the threshold. Short hold times are useful for "DJ-style" ducking, while longer hold times are required for music ducking, e.g. when working on a documentary film.
Release (10 to 1000 ms or Auto mode)	Sets the amount of time it takes for the gain to return to its original level when the signal drops below the threshold level. If the Auto button is activated, Compressor will automatically find an optimal release setting that varies depending on the audio material.
Analysis (0 to 100) (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 better for percussive material, with a lot of transient peaks.
Live button	When this button is activated, the "look ahead" feature of Compressor is disengaged. Look ahead produces more accurate processing, but adds a certain amount of latency as a trade-off. When Live mode is activated, there is no latency, which might be better for "live" processing.

⇨ The compression can also be controlled from another signal source via the side-chain input. When the side-chain signal exceeds the threshold, the compression is triggered. For a description of how to set up side-chain routing, see the chapter "Audio effects" in the Operation Manual.

DeEsser

A de-esser is used to reduce excessive sibilance, primarily for vocal recordings. Basically, it is a special type of compressor that is tuned to be sensitive to the frequencies produced by the "s" sound, hence the name de-esser. Close proximity microphone placement and equalizing can lead to situations where the overall sound is just right, but there is a problem with sibilants.

The following parameters are available:

Parameter	Description
Reduction	Controls the intensity of the de-essing effect.
Threshold	When the Auto Threshold 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.
Auto	The Auto Threshold function automatically and continually chooses an optimum threshold setting independent of the input signal. The Auto Threshold function does not work for low-level signals (< -30 db peak level). To reduce the sibilants in such a file, set the threshold manually.
Release	Sets the amount of time it takes for the de-essing effect to return to zero when the signal drops below the threshold value.
Level meters	Indicate the dB values of the input (IN) and output (OUT) signals as well as the value by which the level of the sibilant (or s-frequency) is reduced (GR). The gain reduction meter shows values between 0 dB (no reduction) and -20 dB (the s-frequency level is lowered by 20 dB).

Positioning the DeEsser in the signal chain

When recording a voice, the de-esser's position in the signal chain is usually located after the microphone preamp and before a compressor/limiter. This keeps the compressor/limiter from unnecessarily limiting the overall signal dynamics.

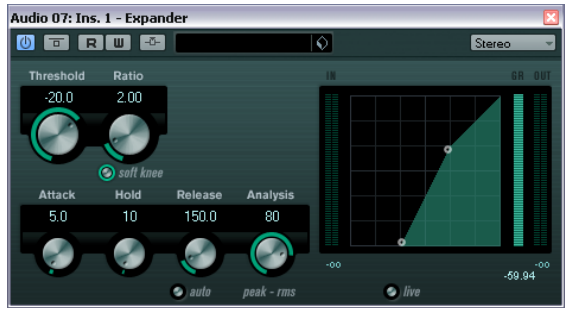
EnvelopeShaper

EnvelopeShaper can be used to cut or boost the gain of the Attack and Release phase of audio material. You can either 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.

The following parameters are available:

Parameter	Description
Attack (-20 to 20dB)	Changes the gain of the Attack phase of the signal.
Length (5 to 200ms)	Determines the length of the Attack phase.
Release (-20 to 20dB)	Changes the gain of the Release phase of the signal.
Output (-24 to 12dB)	Sets the output level.

Expander



Expander reduces the output level in relation to the input level for signals below the set threshold. This is useful when 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.

The following parameters are available:

Parameter	Description
Threshold (-60 to 0dB)	Determines the level where expansion "kicks in". Signal levels below the set threshold are affected, but signal levels above are not processed.
Ratio (1:1 to 8:1)	Determines the amount of gain boost applied to signals below the set threshold.
Soft Knee button	If this button is off, signals below the threshold are expanded instantly according to the set ratio ("hard knee"). When Soft Knee is activated, the onset of expansion is more gradual, producing a less drastic result.
Attack (0.1 to 100ms)	Determines how fast Expander responds to signals below the set threshold. If the attack time is long, more of the early part of the signal (attack) passes through unprocessed.
Hold (0 to 2000ms)	Sets the time the applied expansion will affect the signal below the Threshold.
Release (10 to 1000ms or Auto mode)	Sets the amount of time it takes for the gain to return to its original level when the signal exceeds the threshold level. If the Auto button is activated, Expander will automatically find an optimal release setting that varies depending on the audio material.

Parameter	Description
Analysis (0 to 100) (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 better for percussive material, with a lot of transient peaks.
Live button	When this button is activated, the “look ahead” feature of Expander is disengaged. Look ahead produces more accurate processing, but adds a certain amount of latency as a trade-off. When Live mode is activated, there is no latency, which might be better for “live” processing.

⇨ The expansion can also be controlled from another signal source via the side-chain input. When the side-chain signal exceeds the threshold, the expansion is triggered. For a description of how to set up side-chain routing, see the chapter “Audio effects” in the Operation Manual.

Gate

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

The following parameters are available:

Parameter	Description
Threshold (-60 to 0dB)	Determines the level where Gate is activated. Signal levels above the set threshold trigger the gate to open, and signal levels below the set threshold will close the gate.
State LED	Indicates whether the gate is open (LED lights up in green), closed (LED lights up in red) or something in between (LED lights up in yellow).
Filter buttons (LP, BP, and HP)	When the Side-Chain button (see below) is activated, you can use these buttons to set the filter type to either low-pass, band-pass, or high-pass.
Side-Chain button	This button (below the Center knob) activates the side-chain filter. The input signal can then be shaped according to set filter parameters. Internal side-chaining can be useful for tailoring how the Gate operates.
Center (50Hz to 20000Hz)	When the Side-Chain button is activated, this sets the center frequency of the filter.
Q-Factor (0.01 to 10000)	When the Side-Chain button is activated, this sets the resonance of the filter.
Monitor button	Allows you to monitor the filtered signal.
Attack (0.1 to 1000ms)	Sets the time it takes for the gate to open after being triggered. If the Live button (see below) is deactivated, it ensures that the gate will already be open when a signal above the threshold level is played back. Gate manages this by “looking ahead” in the audio material, checking for signals loud enough to pass the gate.
Hold (0 to 2000ms)	Determines how long the gate stays open after the signal drops below the threshold level.
Release (10 to 1000ms or Auto mode)	Sets the amount of time it takes for the gate to close (after the set hold time). If the Auto button is activated, Gate will find an optimal release setting, depending on the audio material.
Analysis (0 to 100) (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 better for percussive material, with a lot of transient peaks.
Live button	When this button is activated, the “look ahead” feature of Gate is disengaged. Look ahead produces more accurate processing, but adds a certain amount of latency as a trade-off. When Live mode is activated, there is no latency, which might be better for “live” processing.

⇨ The gate can also be controlled from another signal source via the side-chain input. When the side-chain signal exceeds the threshold, the gate opens. For a description of how to set up side-chain routing, see the chapter “Audio effects” in 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).

The following parameters are available:

Parameter	Description
Input (-24 to +24dB)	Allows you to adjust the input gain.
Output (-24 to +6dB)	Determines the maximum output level.
Release (0.1 to 1000ms or Auto mode)	Sets the amount of time it takes for the gain to return to its original level. If the Auto button is activated, Limiter will automatically find an optimal release setting that varies depending on the audio material.

Maximizer

Maximizer is used to raise the loudness of audio material without the risk of clipping. Optionally, there is a soft clip function that removes short peaks in the input signal and introduces a warm tube-like distortion to the signal.

The following parameters are available:

Parameter	Description
Output (-24 to +6dB)	Determines the maximum output level. Should normally be set to 0 (to avoid clipping).
Optimize (0 to 100)	Determines the loudness of the signal.
Soft Clip button	When this button is activated, Maximizer starts limiting (or clipping) the signal "softly", at the same time generating harmonics which add a warm, tube-like characteristic to the audio material.

MIDI Gate

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

Setting up

To set up MIDI Gate, proceed as follows:

1. Select the audio to be affected by MIDI Gate.
This can be audio material from any audio track, or even a live audio input (provided you have a low latency audio card).

2. Select MIDI Gate as an insert effect for the audio track.
The MIDI Gate control panel opens.

3. Select a MIDI track to control the MIDI Gate effect.
This can be an empty MIDI track or a MIDI track containing data, it does not matter. However, if you wish to use MIDI Gate in realtime – as opposed to using a recorded part – the track has to be selected for the effect to receive the MIDI output.

4. Open the Output Routing pop-up menu for the MIDI track and select the MIDI Gate option.
The MIDI output from the track is now routed to the MIDI Gate effect.

What to do next depends on whether you are using live or recorded audio and whether you are using realtime or recorded MIDI. We will assume for the purposes of this manual that you are using recorded audio, and play the MIDI in realtime.

5. Make sure the MIDI track is selected, and start playback.

6. Play a few notes on your MIDI keyboard.
As you can hear, the audio track material is affected by what you play on your MIDI keyboard.

The following MIDI Gate parameters are available:

Parameter	Description
Attack	Determines how long it takes for the gate to open after receiving a signal that triggers it.
Hold	Regulates how long the gate remains open after a note-on or note-off message (see Hold Mode below).
Release	Determines how long it takes for the gate to close (in addition to the value set with the Hold parameter).
Note To Attack	Determines to which extent the velocity values of the MIDI notes affect the attack. The higher the value, the more the attack time increases with high note velocities. Negative values give shorter attack times with high velocities. If you do not wish to use this parameter, set it to the 0 position.
Note To Release	Determines to which extent the velocity values of the MIDI notes affect the release. The higher the value, the more the release time increases. If you do not wish to use this parameter, set it to the 0 position.
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	Use this switch to set 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, and then the Hold and Release parameters are applied.

MultibandCompressor

The MultibandCompressor allows a signal to be split into a maximum of four frequency bands, each with its own freely adjustable compressor characteristic. The signal is processed on the basis of the settings that you have made in the Frequency Band and Compressor sections. You can specify the level, bandwidth and compressor characteristics for each band by using the various controls.

The 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. Two value scales and a number of handles are available. The vertical value scale to the left shows the input gain level of each frequency band. The horizontal scale shows the available frequency range.

The handles provided in the Frequency Band editor can be dragged with the mouse. You use them to set the corner frequency range and the input gain levels for each frequency bands.

- The handles at the sides are used to define the frequency range of the different frequency bands.
- By using the handles on top of each frequency band, you can cut or boost the input gain by +/- 15 dB after compression.

Bypassing frequency bands

Each frequency band can be bypassed using the B button in each compressor section.

Soloing frequency bands

A frequency band can be soloed using the S button in each compressor section. Only one band can be soloed at a time.

Using the Compressor section

By moving breakpoints or using the corresponding knobs, you can specify the Threshold and Ratio. The first breakpoint from which the line deviates from the straight diagonal will be the threshold point.

For each of the four bands the following compressor parameters are available:

Parameter	Description
Threshold (-60 to 0 dB)	Determines the level where Compressor “kicks in”. Signal levels above the set threshold are affected, but signal levels below are not processed.
Ratio (1000 to 8000) (1:1 to 8:1)	Determines the amount of gain reduction applied to signals over the set threshold. A ratio of 3000 (3:1) means that for every 3 dB the input level increases, the output level increases by only 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 (attack) will pass through unprocessed.
Release (10 to 1000 ms or Auto mode)	Sets the amount of time it takes for the gain to return to its original level when the signal drops below the threshold level. If the Auto button is activated, the compressor will automatically find an optimal release setting that varies depending on the audio material.

The Output control

The Output knob controls the total output level that the MultibandCompressor passes on to Nuendo. The range is from -24 to +24 dB.

VintageCompressor

This is modelled after vintage type compressors. This compressor features separate controls for input and output gain, attack, and release. In addition, there is a Punch mode which preserves the attack phase of the signal and a program-dependent Auto feature for the Release parameter.

The available parameters work as follows:

Parameter	Description
Input (-24 to 48 dB)	In combination with the Output setting, this parameter determines the compression amount. The higher the input gain setting and the lower the output gain setting, the more compression is applied.
Output (-48 to 24 dB)	Sets the output gain.
Attack (0.1 to 100 ms)	Determines how fast the compressor responds. If the attack time is long, more of the early part of the signal (attack) passes through unprocessed.
Punch (On/Off)	When 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 amount of time it takes for the gain to return to its original level. If the Auto button is activated, Vintage Compressor will automatically find an optimal release setting that varies depending on the audio material.

⇨ The compression can also be controlled from another signal source via the side-chain input. When the side-chain signal exceeds the threshold, the compression is triggered. For a description of how to set up side-chain routing, see the chapter “Audio effects” in the Operation Manual.

VSTDynamics

VSTDynamics is an advanced dynamics processor. It combines three separate processors: Gate, Compressor and Limiter, covering a variety of dynamic processing functions. The window is divided into three sections, containing controls and meters for each processor.

Activating the individual processors

You activate the individual processors using the buttons at the bottom of the plug-in panel.

The Gate section

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

The following parameters are available:

Parameter	Description
Threshold (-60 to 0 dB)	Determines the level where 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 something in between (LED lights up in yellow).
Side-Chain button	This button activates the internal side-chain filter. You can use this to filter out parts of the signal that might otherwise trigger the gate in places you not want it to, or to boost frequencies you wish to accentuate, allowing for more control over the gate function.
LP (low-pass), BP (band-pass), HP (high-pass)	These buttons set the basic filter mode.
Center (50 to 22000 Hz)	Sets the center frequency of the filter.

Parameter	Description
Q-Factor (0.001 to 10000)	Sets the resonance or width of the filter.
Monitor (On/Off)	Allows you to monitor the filtered signal.
Attack (0.1 to 100ms)	Sets the time it takes for the gate to open after being triggered.
Hold (0 to 2000ms)	Determines how long the gate stays open after the signal drops below the threshold level.
Release (10 to 1000ms or Auto mode)	Sets the amount of time it takes for the gate to close (after the set hold time). If the Auto button is activated, Gate will find an optimal release setting, depending on the audio material.

The Compressor section

The compressor reduces the dynamic range of the audio, making softer sounds louder or louder sounds softer, or both. It works like a standard compressor with separate controls for threshold, ratio, attack, release and make-up gain. The compressor features a separate display that graphically illustrates the compressor curve shaped according to the Threshold, Ratio and Make-Up Gain parameter settings. It also features Gain Reduction meters and a program-dependent Auto feature for the Release parameter.

The available parameters work as follows:

Parameter	Description
Threshold (-60 to 0dB)	Determines the level where the compressor “kicks in”. Signal levels above the set threshold are affected, but signal levels below are not processed.
Ratio (1:1 to 8:1)	Determines the amount of gain reduction applied to signals above the set threshold. A ratio of 3:1 means that for every 3dB the input level increases, the output level increases by only 1dB.
Make-Up (0 to 24dB)	This parameter is used to compensate for output gain loss, caused by compression. When the Auto button is activated, gain loss is being compensated automatically.
Attack (0.1 to 100ms)	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 1000ms or Auto mode)	Sets the amount of time it takes for the gain to return to its original level when the signal drops below the threshold level. If the Auto button is activated, the compressor will automatically find an optimal release setting that varies depending on the audio material.
Graphical display	Use the graphical display to graphically set the Threshold and Ratio values. To the left and right of the graphical display you will find two meters that show the amount of gain reduction in dB.

The Limiter section

The limiter is designed to ensure that the output level never exceeds a set threshold, to avoid clipping in following devices. Conventional limiters usually require very accurate setting up of the attack and release parameters to prevent the output level from going beyond the set threshold level. The limiter adjusts and optimizes these parameters automatically according to the audio material. You can also adjust the Release parameter manually.

The following parameters are available:

Parameter	Description
Output (-24 to +6dB)	Determines the maximum output level. Signal levels above the set threshold are affected, but signal levels below are left unaffected.
Soft Clip button	If this button is activated, the limiter acts differently. When the signal level exceeds -6dB, Soft Clip starts limiting (or clipping) the signal “softly”, at the same time generating harmonics which add a warm, tube-like characteristic to the audio material.
Release (10 to 1000ms or Auto mode)	Sets the amount of time it takes for the gain to return to its original level when the signal drops below the threshold level. If the Auto button is activated, the limiter will automatically find an optimal release setting that varies depending on the audio material.

The Module Configuration button

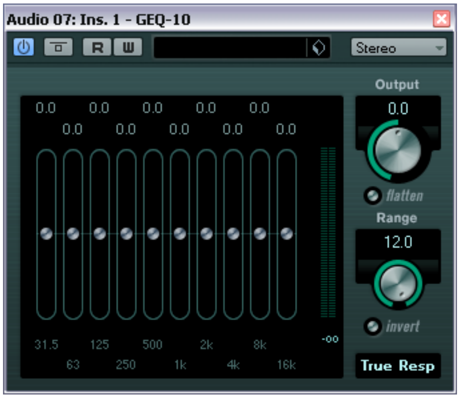
Using the Module Configuration button in the bottom right corner of the plug-in panel, you can set the signal flow order for the three processors. Changing the order of the processors can produce different results, and the available options allow you to quickly compare what works best for a given situation.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

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

EQ plug-ins

This section describes the plug-ins in the “EQ” category.

GEQ-10/GEQ-30



These graphic equalizers are identical in every respect except for the number of available frequency bands (10 and 30 respectively). Each band can be cut or boosted by up to 12dB, allowing for fine control of the frequency response. In addition there are several preset modes available which can add “color” to the sound of the GEQ-10/GEQ-30.

▪ You can draw response curves in the main display by click-dragging with the mouse.

Note that you have to click on one of the sliders first before dragging across the display. You can also point and click to change individual frequency bands, or enter values numerically by clicking on a gain value at the top of 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 cut/boost is shown in dB.

Apart from the frequency bands, the following parameters are available:

Parameter	Description
Output	Controls the overall gain of the equalizer.
Flatten button	Resets all the frequency bands to 0dB.
Range	Allows you to relatively adjust how much a set curve cuts or boosts the signal. If the Range parameter is turned fully clockwise, the range is +/-12dB.
Invert button	Inverts the current response curve.
Mode pop-up menu	The filter mode set here determines how the various frequency band controls interact to create the response curve, see below.

About the filter modes

On the pop-up menu in the lower right corner there are several different EQ modes available. These modes can add color or character to the equalized output in various ways. Here follow brief descriptions of the filter modes:

- True Response – serial filters with accurate frequency response.
- Digi Standard – resonance of last band depends on sample rate.
- Variable Q – parallel filters where the resonance depends on the amount of gain. Musical sounding.
- Constant Q u – parallel filters where the resonance of the first and last bands depends on the sample rate (u=unsymmetric).
- Constant Q s – parallel filters where the resonance is raised when boosting the gain and vice versa (s=symmetric).
- Resonant – serial filters where a gain increase of one band will lower the gain in adjacent bands.

StudioEQ

This is a high-quality 4-band parametric stereo equalizer with two fully parametric mid-range bands. The low and high bands can act as either shelving filters (three types), or as a Peak (band-pass) or Cut (low-pass/high-pass) filter.

Making settings

1. Click the corresponding On button on the left of the plug-in panel to activate any or all of the 4 equalizer bands (Low, Mid 1, Mid 2, and High).

When a band is activated, the corresponding EQ point appears in the EQ curve display.

2. Set the parameters for an activated EQ band.

This can be done in several ways:

- By using the knobs.
- By clicking on the numeric values and typing in new values.
- By using the mouse to drag points in the EQ curve display.

When using the mouse to change the parameter settings, the following modifier keys can be used:

Modifier key	Description
–	When no modifier key is pressed and you drag an EQ point in the display, the Gain and Frequency parameters are adjusted simultaneously.
[Shift]	Keep the [Shift] key pressed and drag the mouse to change the Q-factor of the corresponding EQ band.
[Alt]/[Option]	Keep the [Alt]/[Option] key pressed and drag the mouse to change the frequency of the corresponding EQ band.
[Ctrl]/[Command]	Keep the [Ctrl]/[Command] key pressed and drag the mouse to change the gain value of the corresponding EQ band.

The following parameters are available:

Parameter	Description
Band 1 Gain (-20 to +24 dB)	Sets the amount of cut/boost for the low band.
Band 1 Inv button	Inverts the gain value of the filter. Use this button to filter out unwanted noise. While 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 it, you can use the Inv button to cancel it out.
Band 1 Freq (20 to 2000 Hz)	Sets the frequency of the low band.
Band 1 Q-Factor (0.5 to 10)	Controls the width or resonance of the low band.
Band 1 Filter mode	For the low band, you can select between three types of shelving filters, a Peak (band-pass), and a Cut (low-pass/high-pass) filter. When 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.
Band 2 Gain (-20 to +24 dB)	Sets the amount of cut/boost for the mid 1 band.
Band 2 Inv button	Inverts the gain value of the filter (see the description of the Invert button for Band 1).
Band 2 Freq (20 to 20000 Hz)	Sets the center frequency of the mid 1 band.
Band 2 Q-Factor (0.5 to 10)	Sets the width of the mid 1 band: the higher this value, the “narrower” the bandwidth.
Band 3 Gain (-20 to +24 dB)	Sets the amount of cut/boost for the mid 2 band.
Band 3 Inv button	Inverts the gain value of the filter (see the description of the Invert button for Band 1).
Band 3 Freq (20 to 20000 Hz)	Sets the center frequency of the mid 2 band.
Band 3 Q-Factor (0.5 to 10)	Sets the width of the mid 2 band: the higher this value, the “narrower” the bandwidth.
Band 4 Inv button	Inverts the gain value of the filter (see the description of the Invert button for Band 1).
Band 4 Gain (-20 to +24 dB)	Sets the amount of cut/boost for the high band.
Band 4 Freq (200 to 20000 Hz)	Sets the frequency of the high band.
Band 4 Q-Factor (0.5 to 10)	Controls the width or resonance of the high band.
Band 4 Filter mode	For the high band, you can select between three types of shelving filters, a Peak, and a Cut filter. When Cut mode is selected, the Gain parameter is fixed. -Shelf I adds resonance in the opposite gain direction slightly below 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.
Output (-24 to +24 dB)	This knob on the top right of the plug-in panel allows you to adjust the overall output level.
Auto Gain button	When this button is activated, the gain is automatically adjusted, keeping the output level constant regardless of the EQ settings.

Filter plug-ins

This section contains descriptions of the plug-ins in the “Filter” category.

DualFilter

The DualFilter effect filters out certain frequencies while allowing others to pass through.

The following parameters are available:

Parameter	Description
Position	Sets the filter cutoff frequency. If you set this to a negative value, DualFilter will act as a low-pass filter. Positive values cause DualFilter to act as a high-pass filter.
Resonance	Sets the sound characteristic of the filter. With higher values, a ringing sound is heard.

PostFilter

The PostFilter is the filter plug-in to use if you are working on a post-production mix, but of course you can use it in music production, too, as an alternative to complex EQ configurations. It allows quick and easy filtering of unwanted frequencies, creating room for the important sounds in your mix.

The PostFilter plug-in combines a low-cut filter, a notch filter and a high-cut filter. You can either make settings by dragging the handles in the graphical display, or by adjusting one of the controls below the display section.

Use the Preview buttons to compare the result of your filtering and the filtered frequencies.

The following parameters are available:

Parameter	Description
Level meter	The meter to the right of the EQ display shows the output level, giving you an indication of how the filtering affects the overall level of the edited event.
Low Cut Freq (20 Hz to 1 kHz, or Off)	Use this low-cut filter to eliminate low-frequency noise. The filter is off when the handle/knob is moved all the way to the left.
Low Cut Slope pop-up menu	Allows you to choose a slope value for the low-cut filter.
Low Cut Preview button	Use the Preview button (found between the Low Cut Freq button and the graphical display) to switch the filter to a complementary high-cut filter. This deactivates any other filters, allowing you to listen only to the frequencies you want to filter out.

Parameter	Description
Notch Freq	Sets the frequency of the notch filter.
Notch Gain	Allows you to adjust the gain of the selected frequency. Use positive values to identify the frequencies that you want to filter out.
Notch Gain Invert button	Inverts the gain value of the notch filter. Use this button to filter out unwanted noise. While looking for the frequency to omit, it sometimes helps to boost it in the first place (set notch filter to positive gain). After you have found it, you can use the Invert button to cancel it out.
Notch Q-Factor	Sets the width of the notch filter.
Notch Preview button	Use the Preview button (found between the notch filter buttons and the graphical display) to create a band-pass filter with the peak filter's frequency and Q. This deactivates any other filters, allowing you to listen only to the frequencies you want to filter out.
Notches buttons (1, 2, 4, 8)	These buttons add additional notch filters to filter out harmonics.
High Cut Freq (3Hz to 20kHz, or Off)	Use this high-cut filter to eliminate high-frequency noise. Filter is Off when the handle/knob is moved all the way to the right.
High Cut Slope pop-up menu	Allows you to choose a slope value for the high-cut filter.
High Cut Preview button	Use the Preview button (found between the High Cut Freq button and the graphical display) 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.

Q

Q is a high-quality 4-band parametric stereo equalizer with two fully parametric mid-range bands. The low and high bands can act as either standard shelving filters or fixed-gain high/low-cut filters.

Making settings

1. Click the corresponding On button below the EQ curve display to activate any or all of the Low, Mid 1, Mid 2, or High equalizer bands.

When a band is activated, a corresponding EQ point appears in the EQ curve display.

2. Set the parameters for an activated EQ band.

This can be done in several ways:

- By using the knobs.
- By clicking a value field and entering values numerically.
- By using the mouse to drag points in the EQ curve display window.

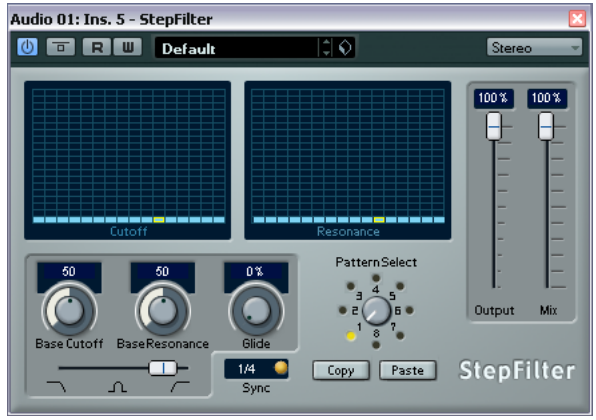
By using this method, you control both the Gain and Frequency parameters simultaneously. The knobs turn accordingly when you drag points. In addition, if the Mid 1 and Mid 2 bands (M1 and M2) are activated there will be two points on each side of the Gain/Frequency point that control the width (Q) parameter.

If you press [Shift] while dragging, values can be set in finer increments.

The following parameters are available:

Parameter	Description
Low Freq (20 to 2000 Hz)	Sets the frequency of the low band.
Low Gain (-20 to +20 dB)	Sets the amount of cut/boost for the low band.
Low Cut	If this button is activated for the low band, it acts as a low cut filter, and the Gain parameter is fixed.
Mid 1 Freq (20 to 20000 Hz)	Sets the center frequency of the mid 1 band.
Mid 1 Gain (+/-20 dB)	Sets the amount of cut/boost for the mid 1 band.
Mid 1 Width (0.05 to 5.00 Octaves)	Sets the width of the mid 1 band in octaves. The lower this value, the “narrower” the bandwidth.
Mid 2 Freq (20 to 20000 Hz)	Sets the center frequency of the mid 2 band.
Mid 2 Gain (-20 to +20 dB)	Sets the amount of cut/boost for the mid 2 band.
Mid 2 Width (0.05 to 5.00 Octaves)	Sets the width of the mid 2 band in octaves. The lower this value, the “narrower” the bandwidth.
High Freq (200 to 20000 Hz)	Sets the frequency of the high band.
High Gain (-20 to +20 dB)	Sets the amount of cut/boost for the high band.
High Cut	If this button is activated for the High band, it acts as a high cut filter, and the Gain parameter is fixed.
Output slider (-20 to +20 dB)	Allows you to adjust the overall output level.
Left/Stereo/Right/ Mono modes	For stereo signals you can set independent curves for the left and right channels by clicking the corresponding button. If the Stereo button is activated, the curve is applied to both channels. When channel-independent curves have been set, the curves for the left and right channel are colored green and red, respectively. The channel that is not selected is shown with a dotted curve. If you activate the Stereo button after independent curves have been set, the active curve is applied to both channels. Mono mode is automatically activated for mono signals and is otherwise unavailable.

StepFilter



StepFilter is a pattern-controlled multimode filter that can create rhythmic, pulsating filter effects.

General operation

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

Setting step values

- Setting step values is done by clicking in the pattern grid windows.
- Individual step entries can be freely 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.
- 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.

Selecting new patterns

- Created 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 Selector below the Resonance grid to select a new pattern.

New patterns are all set to the same step value by default.

Using pattern copy and paste to create variations

You can use the Copy and Paste buttons below the Pattern Selector to copy a pattern to another pattern slot, which is useful for creating variations on a pattern.

- Select the pattern you wish to copy, click the Copy button, select another pattern slot, and click Paste.

The pattern is copied to the new slot, and can now be edited to create variations using the original pattern as a starting point.

StepFilter parameters

Parameter	Description
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 that very high Base Resonance settings can produce loud ringing effects at certain frequencies.
Glide	This will apply glide between the pattern step values, causing values to change more smoothly.
Filter mode	Use this slider to select a filter mode: low-pass (LP), band-pass (BP), or high-pass (HP) (from left to right).
Sync button	When the Sync button to the right of the Sync pop-up menu is activated (yellow), the pattern playback is synchronized with the project tempo.
Sync pop-up menu (1/1 to 1/32, straight, triplet, or dotted)	Use this pop-up menu to set the pattern beat resolution, i.e. what note values the pattern will play in relation to the tempo.
Output slider	Sets the overall volume.
Mix slider	Adjusts the mix between dry and processed signal.

ToneBooster

ToneBooster is a filter that allows you to raise the gain in a selected frequency range. It is particularly useful when inserted before AmpSimulator in the plug-in chain (see "AmpSimulator" on page 9), greatly enhancing the tonal varieties available.

The following parameters are available:

Parameter	Description
Tone	Sets the center filter frequency.
Gain	Allows you to adjust the gain of the selected frequency range by up to 24 dB.
Width	Sets the resonance of the filter.
Mode selector	Sets the basic operational mode of the filter; Peak or Band Mode.

Tonic

Tonic is a versatile and powerful analog modeling filter plug-in based on the filter design of the Monologue monophonic synthesizer. Its variable characteristics plus the powerful modulation functions make it an excellent choice for all current music styles. Designed to be more a creative tool rather than a tool to fix audio problems, it can add color and punch to your tracks while being light on CPU usage.

Tonic has the following properties:

- Dynamic multimode analog modeling filter (mono/stereo).
- 24 dB low-pass, 18 dB low-pass, 12 dB low-pass, 6 dB low-pass, 12 dB band-pass, and 12 dB high-pass modes.
- Adjustable drive and resonance up to self-oscillation.
- Envelope follower for dynamic filter control with an audio signal.
- Audio and MIDI trigger modes.
- Powerful step LFO with smoothing and morphing.
- X/Y matrix pad for additional realtime modulation with access to all Tonic parameters.

Filter

In the Filter section at the center of the plug-in panel, the following parameters are available:

Parameter	Description
Mode pop-up menu	Sets the filter type. Available filter types are: 24 dB low-pass, 18 dB low-pass, 12 dB low-pass, 6 dB low-pass, 12 dB band-pass, and 12 dB high-pass.
Cutoff	Sets the filter cutoff frequency. How this parameter operates is governed by the filter type.
Res	Changes the resonance of the multi-mode filter. Full resonance puts the filter into self-oscillation.
Drive	Adds a soft, tube-like saturation to the sound. As with an analog filter, the amount of saturation also depends on the input signal level.
Mix	Sets the balance between dry and effect signal.
Channel selector (Ch.).	Allows you to choose between mono or stereo operation. When set to mono, the output signal of Tonic is mono regardless of the input signal.

Env Mod

In the Env Mod section, the following parameters are available:

Parameter	Description
Mode pop-up menu	Tonic offers three types of envelope modulation: "Follow" tracks the input signal's volume envelope for dynamic control of the filter cutoff. "Trigger" uses the input signal to trigger the envelope and have it run through a single envelope cycle. "MIDI" uses any MIDI note to trigger the envelope. The filter cutoff tracks the keys played on the keyboard. In addition, velocities higher than 80 add an accent to the envelope by increasing the envelope depth and reducing the decay time. For MIDI control, set up a separate MIDI control track and select "Tonic" from the Output Routing pop-up menu for the track.
Attack	Controls the attack time of the envelope. Higher attack times result in slower rise times when the envelope is triggered.
Release	Controls the release time of the envelope. Higher release times result in slower envelope tails.
Depth	Controls the amount of envelope control applied to the filter cutoff level.
LFO Mod	Using this parameter, the envelope level modulates the LFO speed. A rather stunning effect.

X/Y Pad

In the X/Y Pad at the bottom left of the plug-in panel, the following parameters are available:

Parameter	Description
X Par pop-up menu	Sets the parameter to be modulated on the x-axis of the XY Pad. All of Tonic's parameters are available as destinations.
Y Par pop-up menu	Sets the parameter to be modulated on the y-axis of the XY Pad.
XY Pad	Use the mouse to control any two of Tonic's parameters in combination. By moving the mouse horizontally you control the x parameter, and by moving it vertically you control the y parameter. You can also record controller movements as automation data.

LFO Mod

In the LFO Mod section, the following parameters are available:

Parameter	Description
Mode pop-up menu	Sets the direction of the step LFO modulation. The available modes are: Forward, Reverse, Alternating, and Random.
Depth	Controls the amount of LFO modulation applied to the filter cutoff level.
Rate	Controls the speed of the LFO modulation. The LFO rate is always in sync with the project tempo. An example: at a rate of 4.00 steps per beat in a 4/4 time signature, the step sequencer advances in 16th notes. At a rate of 4.00 beats per step in a 4/4 time signature the LFO advances only one step per bar. Note that the current LFO Rate is shown in the field below the Env Mod section.
Smooth	Controls the smoothing of the LFO steps. This works like a glide effect applied to the filter cutoff.
Morph	Controls the playback value of the LFO step sequencer. It makes the LFO steps drift about randomly. Experiment freely with the Morph parameter. As you return the knob to its zero position, the step pattern returns to its original setting.
Steps pop-up menu	Sets the number of steps played in sequence. Deactivated steps are grayed out in the Step Matrix.
Presets pop-up menu	Offers a number of step LFO waveform patterns. Choices include: Sine, Sine+, Cosine, Triangle, Sawtooth, Square, Random, and User (which is the pattern saved with the respective program).
Step Matrix	Click into the Step Matrix to set the level for each of the 16 LFO steps. A higher amount results in a deeper filter cutoff modulation. Click and drag along the matrix to "draw" a waveform.

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 (see below). You can independently specify the frequency, width and the gain for the Lo and Hi Pedal positions. The crossover point between the Lo and Hi Pedal positions lies at 50.

The following parameters are available:

Parameter	Description
Pedal	Controls the filter frequency sweep.
Pedal Control (MIDI) pop-up menu	Allows you to choose the MIDI controller that is used to control the plug-in. Set this to "Automation" if you do not want to use MIDI realtime control.
Freq Lo/Hi	Set the frequency of the filter for the Lo and Hi Pedal positions.
Width Lo/Hi	Set the width (resonance) of the filter for the Lo and Hi Pedal positions.
Gain Lo/Hi	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: 6dB or 12dB.

⇨ When the side-chain input is activated, a signal routed to the side-chain input of the effect can control the Pedal parameter. The louder the signal, the more the filter frequency (Pedal) is raised so that the plug-in acts as an "auto-wha" effect. For a description of how to set up side-chain routing, see the chapter "Audio effects" in the Operation Manual.

MIDI control

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

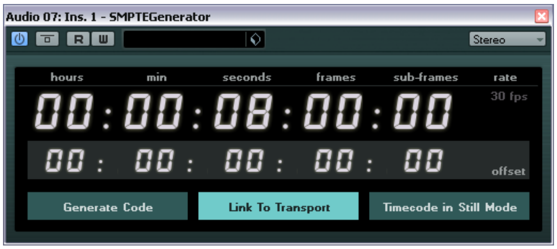
- Whenever WahWah has been added as an insert effect (for an audio track or an FX channel), it is available on the Output Routing pop-up menu for MIDI tracks.

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

Generator plug-ins

This section contains descriptions of the plug-ins in the “Generator” category.

SMPTEGenerator



This plug-in is not a real audio effect. It sends out SMPTE timecode to an audio output, allowing you to synchronize other equipment to Nuendo (provided that the equipment can sync directly to SMPTE timecode). This can be very useful if you do not have access to a MIDI-to-timecode converter.

The following parameters are available:

Parameter	Description
Main timecode display	This display shows the current timecode. When “Link to Transport” is deactivated, the generator is in “free run” mode. You can then use the timecode display to set the SMPTE start time. When “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 (see below).
Frame rate display and pop-up menu	The frame rate shown to the right of the timecode display defaults to the frame rate set in the Project Setup dialog. To generate timecode in a different frame rate (e.g. to stripe a tape), select another format on the pop-up menu (only available if “Link to Transport” is deactivated). Note that for another device to synchronize correctly to Nuendo, the same frame rate has to be set in the Project Setup dialog, the SMPTE Generator 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 Nuendo. The offset affects the generated SMPTE signal, the current cursor position in Nuendo 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 Nuendo. A scenario could be as follows: Your have placed the same video several times on the Nuendo 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 Nuendo with the (unchanging) start position on the external machine.
Generate Code button	When you activate this button, 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 button	When you activate this button, the timecode is synchronized to the Transport panel.
Timecode in Still Mode button	When you activate this button, the plug-in also generates SMPTE timecode in stop mode. However, note that this will not be 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.

⇨ 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.

Example – Synchronizing a device to Nuendo

1. Use the SMPTE Generator 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. You should also disable any EQ.

2. Connect the corresponding output on the audio hardware to the timecode input on the device you wish to synchronize to Nuendo.
Make all necessary settings for the external device so that it synchronizes to incoming timecode.

3. If needed, adjust the level of the timecode, either in Nuendo 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 SMPTE Generator.

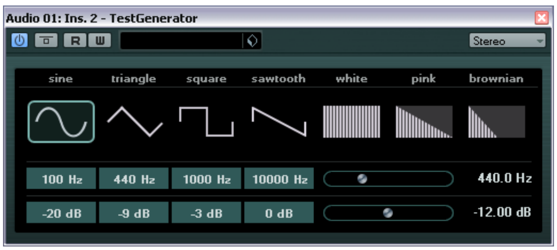
5. Activate the “Link to Transport” button.

The SMPTE Generator now outputs timecode that corresponds to the Nuendo time display.

6. On the Nuendo Transport panel, click Play.

The external device is now synchronized and will follow any position changes set with the Nuendo 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:

- For testing the specifications of audio equipment.
- For measurements of various kinds, such as calibrating tape recorders.
- For testing signal processing methods.
- For 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:

Parameter	Description
Waveforms and noise section	Allows you to set the basis for the signal generated by the waveform generator. You can select 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 select one of the preset values (100, 440, 1000, or 10000 Hz), or use the slider to set a value between 1 Hz and 20000 Hz.
Gain section	Allows you to set the amplitude of the signal. The higher the value (up to 0 dB), the stronger the signal. You can select one of the preset values (e.g. -20 dB), or use the slider to set a value between -81 and 0 dB.

Mastering – UV22HR

The UV22HR is a dithering plug-in, based on an advanced algorithm developed by Apogee. For an introduction to the concept of dithering, see the chapter “Audio effects” in the Operation Manual.

The following parameters are available:

Option	Description
Bit Resolution	The UV22HR supports dithering to multiple resolutions: 8, 16, 20 or 24 bits. You select the desired resolution by clicking the corresponding button.
Hi	Try this first, it is the most “all-round” setting.
Lo	This applies a lower level of dither noise.
Auto black	When this is activated, the dither noise is gated (muted) during silent passages in the material.

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

Modulation plug-ins

This section contains descriptions of the plug-ins in the "Modulation" category.

AutoPan

This is a simple auto-pan effect. It can use different waveforms to modulate the left-right stereo position (pan), either using tempo sync or manual modulation speed settings.

The following parameters are available:

Parameter	Description
Rate	If tempo sync is on, 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 off, the auto-pan speed can be set freely with the Rate knob.
Sync button	The button below the Rate knob is used to switch tempo sync on or off.
Width	Sets the depth of the auto-pan effect.
Waveform Shape selector	Allows you to select the modulation waveform. A sine and a triangle waveform are available.

⇨ The Width parameter can also be controlled from another signal source via the side-chain input. For a description of how to set up side-chain routing, see the chapter "Audio effects" in the Operation Manual.

Chorus

This is a single stage chorus effect. It works by doubling whatever is sent into it with a slightly detuned version (see also "StudioChorus" on page 34).

The following parameters are available:

Parameter	Description
Rate	If tempo sync is on, this is where you specify the base note value for tempo syncing the chorus sweep (1/1 to 1/32, straight, triplet, or dotted). If tempo sync is off, the sweep rate can be set freely with the Rate knob.
Sync button	The button below the Rate knob is used to switch tempo sync on or off.
Width	Determines the depth of the chorus effect. Higher settings produce a more pronounced effect.
Waveform Shape selector	Allows you to select the modulation waveform, altering the character of the chorus sweep. A sine and a triangle waveform are available.
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 effect. If Chorus is used as a send effect, set this to the maximum value as you can control the dry/effect balance with the send.
Delay	Affects the frequency range of the modulation sweep by adjusting the initial delay time.
Filter Lo/Hi	Allow you to roll off low and high frequencies of the effect signal.

⇨ The modulation can also be controlled from another signal source via the side-chain input. When 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 chapter "Audio effects" in the Operation Manual.

Cloner

The Cloner plug-in adds up to four detuned and delayed voices to the signal, for rich modulation and chorus effects.

The following parameters are available:

Parameter	Description
Voices	Allows you to select the number of voices (up to four). For each added voice, a Detune and a Delay slider are added in the right half of the panel.
Spatial	Spreads the added voices across the stereo spectrum. Turn clockwise for a deeper stereo effect.
Mix	Sets the level balance between the dry signal and the effect. If Cloner is used as a send effect, set this to the maximum value as you can control the dry/effect balance with the send.
Output	Allows you to reduce or increase the output gain by up to 12dB.
Detune slider 1–4	Controls the relative detune amount for each voice. Positive and negative values can be set, from -100 to 100. A value of zero means no detune for that voice.
Delay slider 1–4	Controls the relative delay amount for each voice. A value of zero means no delay for that voice.
Detune	Governs the overall depth of the detuning for all voices. If this is set to zero, no detuning takes place, regardless of the Detune slider settings.
Natural button	By clicking the Natural button below the Detune knob, you can change the pitch algorithm.
Detune – Humanize	Controls the amount of detune variation when Static Detune is deactivated. With Humanize, the detune is constantly modulated for a more natural effect. The value range is from 0 to 100 (strongest detune variation).
Static Detune button	Use this button to activate/deactivate the Static Detune function. If activated, the set detune amount is static, and the Humanize knob is grayed out.
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.
Delay – Humanize	Controls the amount of delay variation when Static Detune is deactivated. With Humanize, the delay is constantly modulated for a more natural effect. The value range is from 0 to 100 (strongest delay variation).
Static Delay button	Use this button to activate/deactivate the Static Delay function. If activated, the set delay amount is static, and the Humanize knob is grayed out.

Flanger

Flanger is a classic flanger effect with added stereo enhancement.

The following parameters are available:

Parameter	Description
Rate	If tempo sync is on, this is where you specify the base note value for tempo syncing the flanger sweep (1/1 to 1/32, straight, triplet, or dotted). If tempo sync is off, the sweep rate can be set freely with the Rate knob.
Sync button	The button below the Rate knob is used to switch tempo sync on or off.
Range Lo/Hi	Set the frequency boundaries for the flanger sweep.
Feedback	Determines the character of the flanger effect. Higher settings produce a more "metallic" sounding sweep.
Spatial	Sets the stereo width of the effect. Turn clockwise for a wider stereo effect.

Parameter	Description
Mix	Sets the level balance between the dry signal and the effect. If Flanger is used as a send effect, set this to the maximum value as you can control the dry/effect balance with the send.
Waveform Shape selector	Allows you to select the modulation waveform, altering the character of the flanger sweep. A sine and a triangle waveform are available.
Delay	Affects the frequency range of the modulation sweep by adjusting the initial delay time.
Manual knob	Allows you to change the sweep position manually when the Manual button is deactivated. The value range is from 0 to 100.
Manual button	Use this button to activate/deactivate the Manual function. If activated, the flanger sweep is static, i.e. no modulation takes place.
Filter Lo/Hi	Allow you to roll off low and high frequencies of the effect signal.

⇨ The modulation can also be controlled from another signal source via the side-chain input. When 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 chapter "Audio effects" in the Operation Manual.

Metalizer

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

Parameter	Description
Feedback	The higher the value, the more "metallic" the sound.
Sharpness	Governs the character of the filter effect. The higher the value, the narrower the affected frequency area, producing a sharper sound and a more pronounced effect.
Tone	Governs the feedback frequency. The effect of this will be more noticeable with high Feedback settings.
On button	Turns filter modulation on and off. When turned off, Metalizer works as a static filter.
Mono button	When this is activated, the output of Metalizer is mono.
Speed	If tempo sync is on, this is where you specify the base note value for tempo-syncing the effect (1/1 to 1/32, straight, triplet, or dotted). Note that there is no note value modifier for this effect. If tempo sync is off, the modulation speed can be set freely with the Speed knob.
Sync button	The button above the Speed knob is used to switch tempo sync on (button lights up) or off.
Output slider	Sets the overall volume.
Mix slider	Sets the level balance between the dry signal and the effect. If Metalizer is used as a send effect, set this to the maximum value as you can control the dry/effect balance with the send.

Phaser

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

The following parameters are available:

Parameter	Description
Rate	If tempo sync is on, this is where you specify the base note value for tempo syncing the phaser sweep (1/1 to 1/32, straight, triplet, or dotted). If tempo sync is off, the sweep rate can be set freely with the Rate knob.
Sync button	The button below the Rate knob is used to switch tempo sync on or off.
Width	Determines the width of the modulation effect between higher and lower frequencies.
Feedback	Determines the character of the phaser effect. Higher settings produce a more pronounced effect.
Spatial	When 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 effect. If Phaser is used as a send effect, set this to the maximum level as you can control the dry/effect balance with the send.
Manual knob	Allows you to change the sweep position manually when the Manual button is deactivated. The value range is from 0 to 100.
Manual button	Use this button to activate/deactivate the Manual function. If activated, the flanger sweep is static, i.e. no modulation takes place.
Filter Lo/Hi	Allow you to roll off low and high frequencies of the effect signal.

⇨ The modulation can also be controlled from another signal source via the side-chain input. When 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 chapter "Audio effects" in 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.

The following parameters are available:

Parameter	Description
Oscillator – LFO Amount	Controls how much the oscillator frequency is affected by the LFO.
Oscillator – Env. Amount	Controls how much the oscillator frequency is affected by the envelope (which is triggered by the input signal). Positive and negative values can be set, with center position representing no modulation. Left of center, a loud input signal will decrease the oscillator pitch, whereas right of center the oscillator pitch will increase when fed a loud input.
Oscillator – Waveform buttons	Allows you to select the oscillator waveform; square, sine, saw, or triangle.
Oscillator – Range slider	Determines the frequency range of the oscillator in Hz.

Parameter	Description
Oscillator – Frequency	Sets the oscillator frequency +/- 2 octaves within the selected range.
Oscillator – Roll-Off	Cuts high frequencies in the oscillator waveform, to soften the overall sound. This is best used when harmonically rich waveforms are selected (e.g. square or saw).
LFO – Speed	Sets the LFO speed.
LFO – Env. Amount	Controls how much the input signal level – via the envelope generator – affects the LFO speed. Positive and negative values can be set, at 0% no modulation is applied. With negative values, a loud input signal slows down the LFO, whereas positive values are used to speed it up at loud input signals.
LFO – Waveform	Allows you to select the LFO waveform; square, sine, saw, or triangle.
LFO – Invert Stereo	Inverts the LFO waveform for the right channel of the oscillator, which produces a wider stereo perspective for the modulation.
Envelope Generator section – Attack and Decay	The Envelope Generator section controls how the input signal is converted to envelope data, which can then be used to control oscillator pitch and LFO speed. It has two main controls: 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 L<R button	When this button is enabled, the L and R input signals are merged, and produce the same envelope output level for both oscillator channels. When disabled, each channel has its own envelope, which affects the two channels of the oscillator independently.
Output slider	Sets the overall volume.
Mix slider	Adjusts the mix between dry and processed signal.

Rotary

The Rotary plug-in 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. Rotary features all the parameters associated with the real thing.

The following parameters are available:

Parameter	Description
Speed selector (Stop/ Slow/Fast)	Allows you to control the speed of the Rotary in three steps.
Speed Change Mode	Allows you to select whether the Slow/Fast setting is a switch (left) or a variable control (right). When switch mode is selected and Pitchbend is the controller, the speed will switch with an up or down flick of the bender. Other controllers switch at MIDI value 64.
Speed Mod	When the Slow/Fast setting is set to variable control, this allows you to select the rotary speed, from 0 (Stop) to 100 (Fast).
MIDI controller pop-up menu	Allows you to choose the MIDI controller that is used to control the plug-in. Set this to “Automation” if you do not want to use MIDI realtime control.
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.
Horn – Fast	Allows for a fine adjustment of the high rotor Fast speed.
Horn – Accel.	Allows for a fine adjustment of the high rotor acceleration time.
Horn – Amp Mod	Controls the high rotor amplitude modulation.
Horn – Freq Mod	Controls the high rotor frequency modulation.
Bass – Slow	Allows for a fine adjustment of the low rotor Slow speed.
Bass – Fast	Allows for a fine adjustment of the low rotor Fast speed.
Bass – Accel.	Allows for a fine adjustment of the low rotor acceleration time.
Bass – Amp Mod	Adjusts the modulation depth of the amplitude.
Bass – Level	Adjusts the overall bass level.
Microphones – Phase	Allows you to adjust the phasing amount in the sound of the high rotor.
Microphones – Angle	Sets the simulated microphone angle. 0 = mono, 180 = one mic on each side.
Microphones – Distance	Sets the simulated microphone distance from the speaker in inches.
Output	Allows you to adjust the overall output level.
Mix	Allows you to adjust the mix between dry and processed signals.

Directing MIDI to the Rotary

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

▪ Whenever Rotary has been added 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 Rotary is selected on the Output Routing menu, MIDI is directed to the plug-in from the selected track.

StudioChorus

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

For each stage the following parameters are available:

Parameter	Description
Rate	If tempo sync is on, this is where you specify the base note value for tempo syncing the chorus sweep (1/1 to 1/32, straight, triplet, or dotted). If tempo sync is off, the sweep rate can be set freely with the Rate knob.
Sync button	The button below the Rate knob is used to switch tempo sync on or off.
Width	Determines the depth of the chorus effect. Higher settings produce a more pronounced effect.
Waveform Shape selector	Allows you to select the modulation waveform, altering the character of the chorus sweep. A sine and a triangle waveform are available.
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 effect. If StudioChorus is used as a send effect, set this to the maximum value as you can control the dry/effect balance with the send.
Delay	Affects the frequency range of the modulation sweep by adjusting the initial delay time.
Filter Lo/Hi	Allow you to roll off low and high frequencies of the effect signal.

⇨ The modulation can also be controlled from another signal source via the side-chain input. When 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 chapter “Audio effects” in the Operation Manual.

Tranceformer

Tranceformer is a ring modulator effect, in which the incoming audio is ring modulated by an internal, variable frequency oscillator, producing new harmonics. A second oscillator can be used to modulate the frequency of the first oscillator, in sync with the Song tempo if needed.

The following parameters are available:

Parameter	Description
Waveform buttons	Allow you to select a pitch modulation waveform.
Tone	Sets the frequency (pitch) of the modulating oscillator (1 to 5000 Hz).
Depth	Governs the depth of the pitch modulation.
Speed	If tempo sync is on, this is where you specify the base note value for tempo-syncing the effect (1/1 to 1/32, straight, triplet, or dotted). Note that there is no note value modifier for this effect. If tempo sync is off, the modulation speed can be set freely with the Speed knob.
Sync button	The button above the Speed knob is used to switch tempo sync on (button lights up) or off.
On button	Turns modulation of the pitch parameter on or off.
Mono button	Governs whether the output is stereo or mono.

Parameter	Description
Output slider	Allows you to adjust the output level of the effect.
Mix slider	Sets the level balance between the dry signal and the effect.

⇨ Note that clicking and dragging in the display allows you to adjust the Tone and Depth parameters at the same time!

Tremolo

Tremolo produces amplitude (volume) modulation. The following parameters are available:

Parameter	Description
Rate	If tempo sync is on, 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 off, the modulation speed can be set freely with the Rate knob.
Sync button	The button below the Rate knob is used to switch tempo sync on or off.
Depth	Governs the depth of the amplitude modulation.
Spatial	Adds a stereo effect to the modulation.
Output	Allows you to adjust the output volume.

⇨ The modulation can also be controlled from another signal source via the side-chain input. When 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 chapter “Audio effects” in the Operation Manual.

Vibrato

The Vibrato plug-in produces pitch modulation. The following parameters are available:

Parameter	Description
Rate	If tempo sync is on, 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 off, the modulation speed can be set freely with the Rate knob.
Sync button	The button below the Rate knob is used to switch tempo sync on or off.
Depth	Governs the depth of the pitch modulation.
Spatial	Adds a stereo effect to the modulation.

⇨ The modulation can also be controlled from another signal source via the side-chain input. When 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 chapter “Audio effects” in the Operation Manual.

Other plug-ins

This section contains descriptions of the plug-ins in the “Others” category.

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. You can for example make a 24-bit audio signal sound like an 8 or 4-bit signal, or even render it completely garbled and unrecognizable.

The following parameters are available:

Parameter	Description
Mode	Allows you to select one of the four operating modes of BitCrusher. In each mode the plug-in sounds differently. Modes I and III are nastier and noisier, while modes II and IV are more subtle.
Sample Divider	Sets the amount by which the audio samples are decimated. At the highest setting (65), nearly all of the information describing the original audio signal is eliminated, turning the signal into unrecognizable noise.
Depth	Defines the bit resolution. A setting of 24 gives the highest audio quality, while a setting of 1 creates mostly noise.
Output slider	Governs the output level from BitCrusher. Drag the slider upwards to increase the level.
Mix slider	Regulates the balance between the output from BitCrusher and the original audio signal. Drag the slider upwards for a more dominant effect, and downwards if you want the original signal to be more prominent.

Chopper

Chopper is a combined tremolo and autopan effect. It can use different waveforms to modulate the level (tremolo) or left-right stereo position (pan), either using tempo sync or manual modulation speed settings. The following parameters are available:

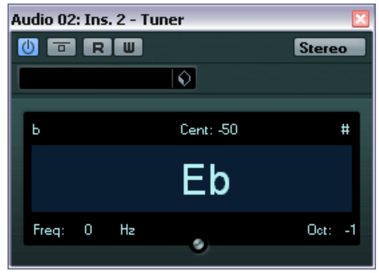
Parameter	Description
Waveform buttons	Set the modulation waveform.
Depth	Sets the depth of the Chopper effect. This can also be set by clicking in the graphical display.
Speed	If tempo sync is on, this is where you specify the base note value for tempo-syncing the effect (1/1 to 1/32, straight, triplet, or dotted). Note that there is no note value modifier for this effect. If tempo sync is off, the tremolo/auto-pan speed can be set freely with the Speed knob.
Sync button	The button above the Speed knob is used to switch tempo sync on (button lights up) or off.
Stereo/Mono button	Determines whether the Chopper works as an auto-panner (button set to “Stereo”) or a tremolo effect (button set to “Mono”).
Mix	Sets the level balance between the dry signal and the effect. If Chopper is used as a send effect, this should be set to the maximum value.

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, respectively. Octaver is best used with monophonic signals. The following parameters are available:

Parameter	Description
Direct	Adjusts the mix of the original signal and the generated voice(s). 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 generated signal one octave below the original pitch. Set to 0 means the voice is muted.
Octave 2	Adjusts the level of the generated signal two octaves below the original pitch. Set to 0 means the voice is muted.

Tuner



This is a guitar tuner. Simply connect a guitar or other instrument to an audio input and select the Tuner as an insert effect (make sure you deactivate any other effect that alters pitch, like chorus or vibrato). When the instrument is connected, proceed as follows:

- Play a note.

The key is shown in the middle of the display. In addition, the frequency in Hz is shown in the bottom left corner and the octave range in the bottom right corner. If the key is wrong (e.g. if you wish to tune the E string and the key is shown as Fb), first tune the string so that the correct key is shown.

- The two arrows indicate any deviation in pitch by their position. If the pitch is flat, they will be positioned in the left half of the display, if the pitch is sharp they will be in the right half.

The deviation is also shown (in Cent) in the upper area of the display.

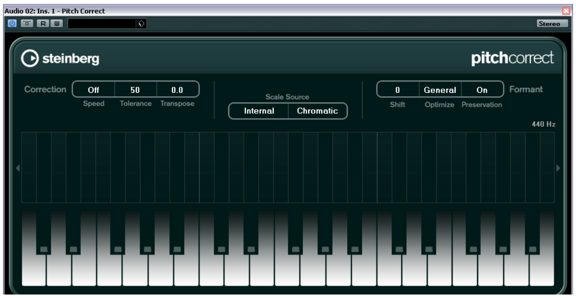
- Tune the instrument so that the two arrows are in the middle.

Repeat this procedure for each string.

Pitch Shift plug-ins

This section contains descriptions of the plug-ins in the "Pitch Shift" category.

PitchCorrect



PitchCorrect automatically detects, adjusts and fixes slight pitch and intonation inconsistencies in monophonic vocal and instrumental performances in realtime. The advanced algorithms of this plug-in preserve the formants of the original sound thus allowing for natural sounding pitch correction without the typical "Micky Mouse" effect.

Furthermore, you can use PitchCorrect 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 will be displayed in blue while the changes are displayed in orange.

The following parameters are available:

Parameter	Description
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 (e.g. the famous "Cher" effect).
Correction – Tolerance	Determines the sensitivity of analysis. A low Tolerance value lets PitchCorrect find pitch changes quickly. When the Tolerance value is high, pitch variations in the audio (e.g. vibrato) will not be immediately interpreted as note changes.
Correction – Transpose (-12 to 12)	With this parameter you can adjust (or "retune") the pitch of the incoming audio in semitone steps. You can set positive and negative values from -12 to 12. A value of zero means 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 will be adapted. The following options are available: Chromatic: The audio will be pitched to the closest semitone. Major/Minor: The audio will be pitched to the major/minor scale specified in the pop-up menu to the right. This will be reflected on the keyboard display. Custom: The audio will be pitched to the notes that you specify by clicking the desired keys on keyboard display. To reset the keyboard, click on the orange line below the display.
Scale Source – 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 that you have to assign the audio track as the output of your MIDI track and that the Speed parameter has to be set to a value other than Off.
Scale Source – 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 that you have to assign the audio track as the output of your MIDI track and that the Speed parameter has to be set to a value other than Off.
Formant – Shift (-60 to 60)	Changes the natural timbre, i.e. the characteristic frequency components of the source audio.
Formant – Optimize (General, Male, Female)	Allows you to specify the sound characteristics of the sound sources. While General is the default setting, Male is designed for low pitches and Female for high pitches.
Formant – Preservation (On/Off)	When set to Off, formants are raised and lowered with the pitch, provoking strange vocal effects. Higher pitch correction values result in "Micky Mouse" effects, lower pitch correction values in "Monster" sounds. When set to On, the formants are kept, maintaining the character of the audio.
Master Tuning	Detunes the output signal. The default setting is 440 Hz.

PitchDriver

PitchDriver was created for sound design purposes in post-production. This plug-in can be used for extreme up or down pitching of voices or effect samples (e.g. to create eerie monster sounds). Shifting the pitch with this plug-in will not keep the formants.

The following parameters are available:

Parameter	Description
Detune	Lets you detune the pitch of the incoming audio. Positive and negative values can be set.
Mix	Sets the level balance between the dry signal and the effect.
Spatial	The Spatial parameter is used to create 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 that the created panorama effect can be unstable. For a stable panorama, turn of the Spatial parameter. In that case the incoming signals are summed up to a mono signal.
Output	Allows you to adjust the output volume.

⇨ 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 card driver's control panel (opened via the Device Setup dialog in Nuendo).

Restoration plug-ins

This section contains descriptions of the plug-ins in the "Restoration" category.

DeClicker

The DeClicker plug-in is specifically designed to eliminate single "clicks" or "pops" in a recording. A typical application is to clean up recordings made from vinyl records, but you may also find it useful for removing pops from microphone switches, oxidized connector noises, clicks from sync problems when transferring material digitally, etc.

⇨ Note that the DeClicker module is not optimized for crackles (a series of short clicks). However, as it is often hard to distinguish between clicks and crackles, you might also be able to use it to improve your recording in this respect.

⇨ If the recording also contains background noise (hiss), you may want to combine DeClicker with the DeNoiser plug-in.

How DeClicker works

The DeClicker process is divided in two tasks:

- Analysis – when the audio signal passes through DeClicker, the selected analysis algorithm finds the clicks in the recording. You provide input to the analysis parameters by selecting a Mode and setting the Threshold and DePlop parameters.

- Removal – a de-click algorithm is applied to the audio, removing the clicks.
In many cases, the original audio material "hidden" underneath a click cannot be restored. This means there will be a gap once the click has been removed. DeClicker has the ability to automatically "redraw" the hence missing parts of the waveform. This feature can also be used to remove tape dropouts with a length of up to 60 samples (just above one millisecond at 44.1 kHz).

The whole de-clicking process can be visually monitored in the Input and Output displays in the DeClicker panel (showing the incoming audio and the processed, i.e. de-clicked, audio). This helps you adjust the parameters. Furthermore, if you activate the Audition button, only the removed material will be heard (and shown in the Output display).

⚠ Make sure that no low-pass filter has been applied to your audio material before you edit it with DeClicker. This may affect the detection of clicks.

Parameters

Parameter	Description
Audition button	When this button is activated, only the removed material will be heard. The Output display will also show the waveform image of the removed material in this mode.
Classic button	When this button is activated, DeClicker attempts to remove both audible clicks and crackle noise. When deactivated, only single clicks are removed while crackles (rapidly repeated clicks) are ignored. Which mode to choose, depends on the source material. Note also that Classic mode requires less CPU power.
Quality section	Here you can determine the quality of the click removal and audio restoration, with "4" being the best quality setting. Please note that selecting higher quality settings means that more processing power is consumed. Also, note that in some situations it might be more productive to use a lower Quality value. One example of this is when two clicks follow each other in quick succession or when you tackle a click in a low level part that is followed by a loud part.
Mode section	Which mode to select depends on the source material. Standard mode is suitable for a wide variety of source material – try this option first. Vintage mode is suitable for restoring "antique" recordings (with limited high frequency content), while Modern mode is best suited for contemporary recordings with a wide frequency range (putting greater emphasis on distinguishing clicks from other strong impulses in the audio material).
Threshold slider	Determines the amplitude (level) required for a click to be detected. In many cases, DeClicker's sensitive algorithms identify a lot more clicks than you can actually hear. To avoid wasting processing power for removing inaudible clicks, raise this parameter to a high value, and then lower it until all the artifacts that you actually want removed are detected. The lower the setting, the more clicks will be detected, but also the higher the risk of audible artifacts. If in doubt, activate Audition mode and check that the removed material does not contain any actual musical or rhythmical information, etc.
DePlop slider	Controls a special high-pass filter which works on signals below 150 Hz. It cuts away the "plop noise" which sometimes appears after eliminating a click. The slider adjusts the filter frequency (Off–150 Hz). Note: this function is best applied to older recordings, which often use a narrow frequency range. Be careful when applying it to modern recordings, as you may risk removing parts of the useful signal!

Tips and Tricks

- By combining Vintage Mode and extreme Threshold and DePlop settings, you can create an interesting effect which "softens" material with particularly sharp attacks, e.g. percussion or brass.
- If you have material with digital distortion (clipping), try applying DeClicker. While it cannot do miracles, it can at least make some improvement to the overall "hardness" introduced by the distortion.

DeNoiser

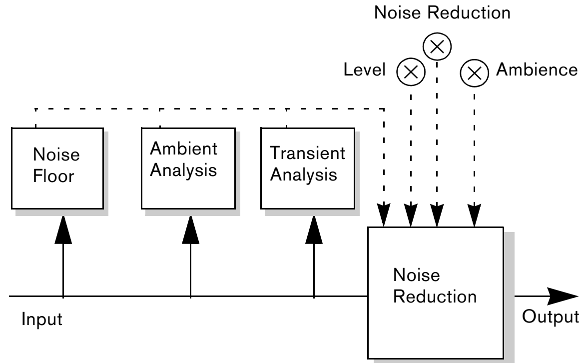
The DeNoiser plug-in lets you suppress noise without affecting the general sound quality. Or, in tech talk, the DeNoiser removes broad band noise from arbitrary audio material without leaving any "spectral finger print". The algorithm that this plug-in is based on has the ability to track and adjust itself to variations in background noise. This means the noise can be diminished without side effects, preserving the spatial impression, and without letting the result become "colorless". Many years of research were invested in developing the methods used.

Typical applications for the DeNoiser plug-in include cleaning or remastering recordings from old tape or vinyl, or noisy live recordings.

How DeNoiser works

DeNoiser is based on spectral subtraction. Each section of the frequency spectrum that has an amplitude below the estimated noise floor is reduced in intensity by use of a spectral expander. The result is a noise reduction that does not affect the phase of the signal.

The figure below shows the signal flow:



The solid line represents the actual audio signal, while the dotted lines represent control signals.

The signal is continuously analyzed by the first module in the chain, to estimate the noise floor at any given time. This is sufficient when the noise level is constant or modulates slowly. When the noise level varies rapidly, the Ambience and Transient analysis helps adjust the response of the noise reduction unit, allowing transient-rich material to maintain its liveliness and natural ambience.

⇨ When processing audio with DeNoiser, the plug-in needs a short time (less than a second) to analyze the material and set its internal parameters. Since you would not want to include this short "startup sequence" in the final result, you should make it a habit to first play back a short section of the audio, thereby letting DeNoiser "learn" the noise floor, and then stop and start over again from the beginning. The plug-in then remembers the settings internally.

The Noisefloor Display

The display on the left of the DeNoiser panel is crucial when making settings. It contains the following three elements:

- The dark green spectral graph.
This shows the spectrum of the audio being played back. The horizontal axis shows the frequency (linear scale). The low frequencies are visible on the left side, the high ones on the right side. The vertical axis shows the signal amplitudes, thus the level (displayed as a logarithmic dB scale).

- The yellow line.
This is a spectral estimation of the noise floor. The average of this value is shown numerically below the display.

- The light green line.
This is simply a graphical representation of the Offset parameter.

The light green Offset line should be adjusted so that it appears as close above the yellow noise floor graph as possible. The dark green spectrum plot is there to help you fine-tune the Offset setting, so that only the noise is removed, not parts of the signal (ideally, the light green line should be between the yellow line and the spectrum plot).

Parameters

Parameter	Description
Freeze button	This button is used to "freeze" the noise floor detection process. The yellow noise floor graph in the display will hold its current value (as will the numeric noise floor value display below) until you deactivate Freeze. This allows you to take a closer look at the readings.
Classic button	When this is activated, a less CPU-intensive version of the DeNoiser algorithm is used. Use Classic mode if you are short on processing power. However, for optimum noise suppression, we recommend that you deactivate Classic mode.
A/B/Store buttons	These buttons are described below this table.
Reduction slider	Governs the amount of noise reduction. The display above this slider shows the amount of dB by which the noise level is being reduced. The final result also depends on the Ambience parameter, and on the automatic Ambience and Transient analysis of the original material, as described above.
Ambience slider	This parameter is used to specify a balance between the noise suppression and the amount of natural ambience, which is essential for a natural result. With a low Ambience setting, the sound can become somewhat lifeless and sterile. A high setting, on the other hand, preserves more of the ambient character of the sound, but the noise suppression is less effective.
Offset slider	This parameter serves as a threshold, governing the overall level at which the noise reduction is performed. For optimal noise reduction with a minimum of sound coloration, this parameter should be set to a value slightly above the noise floor level. To help you do this, the offset value is shown as a light green line in the noisefloor display, while the noise floor is shown as a yellow line.

Using the A/B setups

With the A/B buttons you can make instantaneous switches between two different DeNoiser setups, allowing you to quickly try out and compare different configurations. You can also use this feature for separate settings for two different sections of an audio recording. Proceed as follows:

1. Make the settings you want for setup A.
2. Click the Store button and then the A button.
3. Make the settings you want for setup B.
4. Click the Store button and then the B button.

Now the two setups are stored, and you can switch between them simply by clicking the A or B button.

Grungelizer

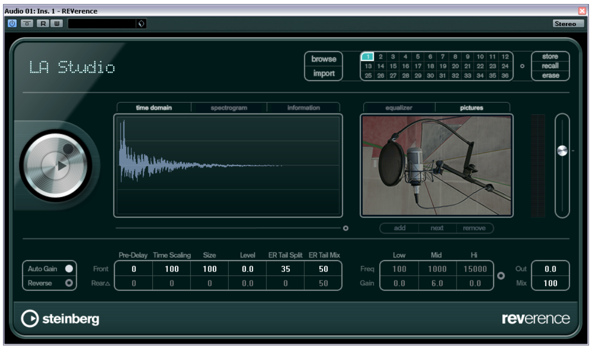
Grungelizer adds noise and static to your recordings – kind of like listening to a radio with bad reception, or a worn and scratched vinyl record. The following parameters are available:

Parameter	Description
Crackle	Adds crackle to create that old vinyl record sound. The farther to the right you turn the knob, the more crackle is added.
RPM switch	When emulating the sound of a vinyl record, this switch lets you set the RPM (revolutions per minute) speed of the record (33/45/78 RPM).
Noise	Regulates the amount of static noise added.
Distort	Adds distortion.
EQ	Turn this knob to the right to cut off the low frequencies, and create a more hollow, lo-fi sound.
AC	Emulates a constant, low hum of AC current.
Frequency switch	Sets the frequency of the AC current (50 or 60 Hz), and thus the pitch of the AC hum.
Timeline	Regulates the amount of overall effect. The farther to the right (1900) you turn the knob, the more noticeable the effect.

Reverb plug-ins

This section contains descriptions of the plug-ins in the "Reverb" category.

REVerence

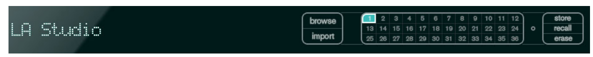


REVerence is a convolution tool that allows you to apply room characteristics (reverb) to the audio. This is done by processing the audio signal according to an impulse response – a recording of an impulse in a room or another location that is used to recreate the characteristics of the room. As a result, the processed audio will sound as if it were played in the same location. Included with the plug-in are top quality samples of real spaces to create reverberation.

⇨ 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 an artifact-free switching between programs. Therefore you should always load only those programs that you need for a given task.

Using the program matrix

A program is the combination of an impulse response and its settings. These include reverb settings (see "Changing the reverb settings" on page 44), EQ settings (see "Making EQ settings" on page 45), pictures (see "Loading pictures" on page 46), and output settings (see "Making output settings" on page 46). The program matrix allows you to load programs and to view the name of the current program, i.e. the impulse response (see "Working with custom impulse responses" on page 46).



The following parameters are available:

Parameter	Description
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 button	This button opens a browser window showing the available programs. When you select a program in the browser, it is loaded into the active program slot. To be able to filter the list of impulse responses in the browser window, e.g. by room type or the number of channels, you can activate the Filters section (by clicking the "Set Up Window Layout" button at the bottom left of the window).
Import button	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. For more information, see "Working with custom impulse responses" on page 46.
Program slots (1 to 36)	Into these slots you can load all the impulse responses (programs) that you want to work with in a session. The selected program slot is indicated by a (blinking) white frame. Occupied slots are shown in a different color. Double-clicking an empty program opens a browser window, showing the available programs. Double-clicking an occupied program slot loads the corresponding impulse response into REVerence ("Recall").
Smooth Parameter Changes button	The "Smooth Parameter Changes" button is located between the program slots and the Store/Recall/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 button	Stores the active impulse response and its settings as a program.

Parameter	Description
Recall button	Reloads the selected program. Use this to reset a program to its default settings.
Erase button	Removes the selected program from the matrix.

Programs vs. presets

You can save your REVerence settings as VST plug-in presets or programs. The differences between the two and the advantages are described in the following.

Both presets and programs use the file extension .vstpreset and appear in the same category in the MediaBay (Plug-In Presets), but they are represented by different icons:

Icon	Description
	A REVerence preset contains all settings and parameters for the plug-in, that is all the 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.
	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:

- When you want to save a complete setup with different impulse responses for later use (e.g. 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 a subset of the plug-in parameters (i.e. the settings for a single impulse response), allowing for short loading times.
- When automating a project and loading a REVerence program, only two automation events are written.

  If 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.

Setting up programs

Proceed as follows:

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 on the empty slot again to load one of the included programs.

   You can also import a new impulse response file, see "Importing impulse responses" on page 47.

3. In the browser that appears, 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.

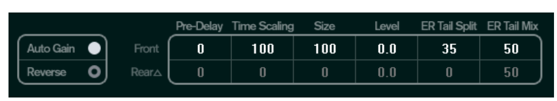
4. Set up the REVerence parameters as needed and click the Store button to save the impulse response with the current settings as a new program.

5. Set up as many programs as you need (up to 36) by following the steps above.

⇨ If you want to use your set of programs in other projects, save your settings as a plug-in preset using the Presets pop-up menu at the top of the plug-in panel.

Changing the reverb settings

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



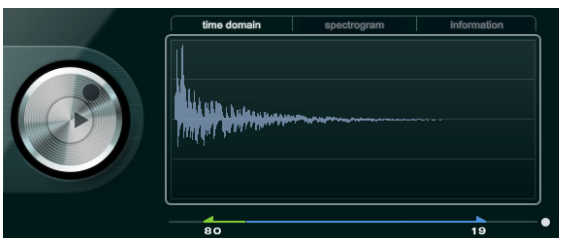
The following parameters are available:

Parameter	Description
Front	All values shown in the top row are for the front speakers.
Rear Δ	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 button	When this button is activated, the impulse response is automatically normalized.

Parameter	Description
Reverse button	Reverses the impulse response.
Pre-Delay	Controls the amount of time between the dry signal and the onset of the reverb. With higher pre-delay values you can simulate larger rooms.
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 early reflections will be heard for 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.

The impulse response display

The Display section allows you to view the impulse response details and to change the length of the response (trimming).



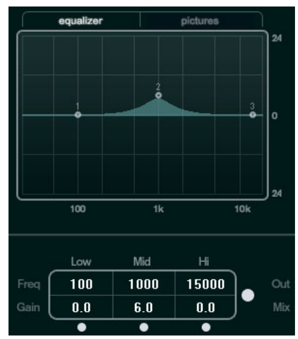
The following parameters are available:

Parameter	Description
Play button/ Time Scaling wheel	When clicking the play 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. The Time Scaling wheel lets you adjust the reverb time.
Time Domain display	Shows the waveform of the impulse response.
Spectrogram 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 display	Shows additional information, e.g. the name of the program and the loaded impulse response, the number of channels, the length, and Broadcast Wave File information.

Parameter	Description
Activate Impulse Trimming button	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 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, or the end handle to trim the reverb tail. You can also use the mouse wheel for trimming. Note that the impulse response will be cut without any fading.

Making EQ settings

In the Equalizer section you can tune the sound of the reverb.



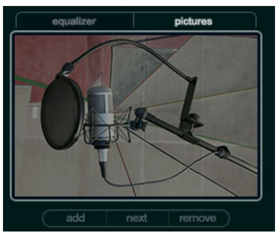
The following parameters are available:

Parameter	Description
EQ curve display	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 button	This button to the right of the EQ parameters activates the EQ for the effect plug-in.
Low Shelf On button	Activates the low shelf filter that boosts or cuts 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 cut/boost for the low band.
Mid Peak On button	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 cut/boost for the mid band.

Parameter	Description
Hi Shelf On button	Activates the high shelf filter that boosts or cuts 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 cut/boost for the high band.

Loading pictures

In the Pictures section you can load graphics files to illustrate the setting, i.e. the recording location or microphone arrangement of the loaded impulse response. Up to five pictures can be loaded.



The following parameters are available:

Parameter	Description
Add button	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 button	If several pictures are loaded, you can click this button to display the next image.
Remove button	Deletes the active picture. Note that this will not remove the graphics file from your hard disk.

⇨ Pictures are only referenced by the plug-in and will not be copied to the project folder.

Making output settings

In the Output section you can control the overall level and determine the dry/wet mix.

The following parameters are available:

Parameter	Description
Output activity meter	Indicates the overall level of the impulse response and its settings.
Output slider	Allows you to adjust the overall output level.
Out (-24 to +12)	Raises or lowers the signal output of the plug-in.
Mix (0 to 100)	Sets the level balance between the dry and the wet signal.

Working with custom impulse responses

In addition to working with the impulse responses included with REVerence, you can import your own impulse responses and save these as programs or presets. WAV, AIF, 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 less channels than the track, REVerence generates the missings channels (e.g. 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

To import impulse responses, proceed as follows:

1. In the program matrix, click the Import button.

2. Navigate to 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 Nuendo (e.g. the VST Connections window), see below.

3. 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.

4. 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.
The program slot turns blue, indicating that a program is loaded.

⇨ 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.

5. Repeat these steps for any impulse response files that you want to work with.

REVerence reads input channels in the following order:

No. 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, see below)
4	LL/LR/RL/RR (if inserted on a track with a stereo configuration, see below)
5	L/R/C/LS/RS
6	L/R/C/LFE/LS/RS (LFE is being ignored.)

True stereo

Impulse responses recorded as trues-stereo files enable 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

⇨ If your true-stereo impulse responses are only available as separate mono files, you can use the Export Audio Mixdown function in Nuendo to create REVerence compliant interleaved files (see the chapter "Export Audio Mixdown" in the Operation Manual).

By default, REVerence automatically works in true-stereo mode when 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 unintenionally 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.

⇨ 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 chapter "MediaBay" in the Operation Manual.

Relocating content

Once you have imported your own impulse responses in 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 will not be a problem since 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 since the old file paths have become invalid.

To access your impulse responses again, proceed as follows:

1. Transfer you audio files to a location that you will be able to access from the second computer (i.e. an external hard disk).
If you keep the files in the same folder structure as on the first computer, REVerence will automatically find all files contained in this structure.

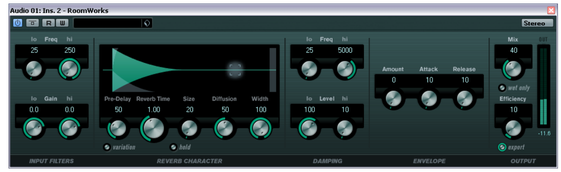
2. Transfer any REVerence presets or programs that you need to the second computer.
If you are unsure where the presets need to be stored, you can find the paths in the MediaBay (see the chapter "The MediaBay" in the Operation Manual).

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. Click Open.
REVerence is now able to access all the impulse responses stored in this location.

⚠ 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.

The following parameters are available:

Parameter	Description
Input – Lo Freq	Determines the frequency at which the low-shelving filter takes effect. Both the high and low settings filter the input signal prior to reverb processing.
Input – Hi Freq	Determines the frequency at which the high-shelving filter takes effect. Both the high and low settings filter the input signal prior to reverb processing.
Input – Lo Gain	Controls the amount of boost or cut for the low-shelving filter.
Input – Hi Gain	Controls the amount of boost or cut for the high-shelving filter.
Reverb – Pre-Delay	Controls how much time passes before the reverb is applied. This allows you to simulate larger spaces by increasing the time it takes for first reflections to reach the listener.
Reverb – Reverb Time	Allows you to set the reverb time in seconds.
Reverb – Size	Alters the delay times of early reflections to simulate larger or smaller spaces.
Reverb – 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.
Reverb – Width	Controls the width of the stereo image. 100% gives you full stereo reverb. At 0%, the reverb is all in mono.
Reverb – Variation button	Pressing this button generates a new version of the same reverb program using altered reflection patterns. This is helpful when certain sounds are causing odd ringing or undesirable results. Creating a new variation will often solve these issues. There are 1000 possible variations.
Reverb – Hold button	Pressing this button freezes the reverb buffer in an infinite loop (yellow circle around button). You can create some interesting pad sounds using this feature.
Damping – Lo Freq	Determines the frequency below which low-frequency damping will occur.

Parameter	Description
Damping – High Freq	Determines the frequency above which high-frequency damping will occur.
Damping – Low Level	Affects the decay time of 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.
Damping – High Level	Affects the decay time of 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.
Envelope – Attack	The envelope settings in RoomWorks control how the reverb will follow 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.
Envelope – Release	Determines how long after a signal peak the reverb can be heard before being cut off, similar to a gate's release time.
Surround – Distance	This control is only available for surround configurations. With this parameter you can control where the virtual listening position is within the room. Positive values position the listener closer to the front of the room and negative values place the listener towards the rear of the room.
Surround – Rotate button	This button is only available for surround configurations. When active, the perspective of the room is shifted 90°.
Surround – Balance	This control is only available for surround configurations. Balance controls the relative levels between the forward and rear speakers. Positive values favor the front speakers and negative values favor the rear speakers. When the Rotate option is activated, these relationships will shift 90°.
Output – Mix	Determines the balance of dry (unprocessed) and wet (processed) signal. When RoomWorks is used as an insert for an FX channel, you will most likely want to set this to 100% or use the Send button.
Output – Wet only button	This button defeats the mix parameter, setting the effect to 100% wet or affected signal. This button should normally be pressed when RoomWorks is being used as a send effect for an FX or group channel.
Output – Efficiency	Determines how much processing power is used for RoomWorks. The lower the value, the more CPU resources will be used, and the higher the quality of the reverb. Interesting effects can be created with very high Efficiency settings (>90%). Experiment for yourself.
Output – Export button	Determines if during audio export RoomWorks will use the maximum CPU power for the highest quality reverb. During export you may wish 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 – Output meter	Indicates the level of the output signal.

RoomWorks SE

RoomWorks SE is a "lite" version of the RoomWorks plug-in. This plug-in delivers high quality reverberation, but has fewer parameters and is less CPU demanding than the full version. The following parameters are available:

Parameter	Description
Pre-Delay	Controls how much time passes before the reverb is applied. This allows you to simulate larger spaces by increasing the time it takes for 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.
Hi Level	Affects the decay time of 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.
Lo Level	Affects the decay time of 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.
Mix	Determines the blend of dry (unprocessed) signal to wet (processed) signal. When using RoomWorks SE inserted in an FX channel, you will most likely want to set this to 100% or use the Send button.

Spatial + Panner plug-ins

This section contains descriptions of the plug-ins in the “Spatial + Panner” category.

MonoToStereo

This effect will turn a mono signal into a “pseudo-stereo” signal. The plug-in must be inserted on a stereo track playing a mono file.

The following parameters are available:

Parameter	Description
Width	Controls the width or depth of the stereo enhancement. Turn clockwise to increase the enhancement.
Delay	Increases the amount of differences between the left and right channels to further increase the stereo effect.
Color	Generates additional differences between the channels to increase the stereo effect.
Mono button	Switches the output to mono, to check for possible unwanted coloring of the sound which sometimes can occur when creating an artificial stereo image.

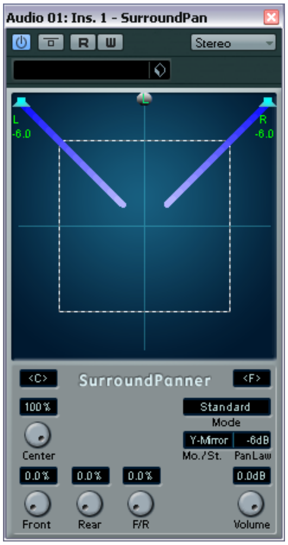
StereoEnhancer

This plug-in will expand the stereo width of (stereo) audio material. It cannot be used with mono files.

The following parameters are available:

Parameter	Description
Width	Controls the width or depth of the stereo enhancement. Turn clockwise to increase the enhancement.
Delay	Increases the amount of differences between the left and right channels to further increase the stereo effect.
Color	Generates additional differences between the channels to increase the stereo enhancement.
Mono button	Switches the output to mono, to check for possible unwanted coloring of the sound which sometimes can occur when enhancing the stereo image.

SurroundPan



The SurroundPanner plug-in allows you to position mono or stereo audio in the surround field. It consists of an image of the speaker arrangement, as defined by the output bus selected on the Output Routing pop-up menu, with the sound source indicated as a gray ball.

Although the plug-in can be used as an insert effect, it is most often inserted in the output of a track or channel. By default the SurroundPanner V5 is used for new tracks or channels, but you can switch to the SurroundPan plug-in, if needed. For more information about this, see the chapter “Surround sound” in the Operation Manual.

⇨ The SurroundPan plug-in was used as the default panner before Nuendo 5. In has now been replaced by the SurroundPanner V5 plug-in. However, projects created with a previous version of Nuendo still use the old SurroundPan plug-in.

Mode – Standard/Position/Angle

The Standard Mode/Position Mode/Angle Mode switch allows you to work in three modes:

- In both Standard and Position mode, the speakers in the front are aligned, as they would normally be in a cinema-type situation. This means that the front speakers are at a varying distance from the center. Standard mode (default) is the best mode for moving sources between speakers without level attenuation.
- Angle Mode is the traditional surround sound mixing definition. Note that here the speakers are defined as being at equal distance from the center. This is not really a true representation of for example a cinema, but has still proven to work well in many situations.

Speakers

The speakers in the panel represent the chosen surround configuration.

You can turn speakers on and off by clicking them with [Alt]/[Option] pressed. When a speaker is turned off, no audio will be routed to that surround channel.

Positioning and levels

⚠ The text below assumes that the Mono/Stereo pop-up is set to “Mono Mix”. For information on the other modes, see below.

A sound source is positioned either by clicking or by dragging the gray “ball” around in the panel (or by using key commands, see below).

- In Standard Mode, the signal levels from the individual speakers are indicated by colored lines from the speakers to the center of the display.

Exactly how levels are handled may require some explanation:

- When you move a source around, a number will indicate the loudness in each speaker.

- This is a value in dB (decibel) and is relative to the nominal level of the source. In other words, 0.0 (dB) represents full level.

- If you position the source far enough away from a speaker, its level will drop to zero (indicated by a negative infinity symbol).

▪ In Standard Mode, the signal levels from the individual speakers are indicated by colored lines from the speakers to the center of the display.

▪ In Position Mode, the concentric circles will help you determine the level of the signal at a certain position.

▪ The yellow circle represents -3 dB below nominal level, the red circle is at -6 dB and the blue is located at -12 dB. These are affected by attenuation, see below.

▪ In Angle Mode, a white arc helps you determine the perceived "range" of a source (white and blue for stereo tracks). The sound will be at its loudest in the middle of the arc and will have dropped in level towards the ends.

You can use modifier keys to restrict movement in various ways:

In Standard and Position Mode:

Key	Movement restriction
[Ctrl]/[Command]	Vertically only
[Ctrl]/[Command]-[Shift]	Horizontally only
[Alt]/[Option]	Diagonally (up left, down right)
[Ctrl]/[Command]-[Alt]/[Option]	Diagonally (up right, down left)
[Shift]	Mouse movements are scaled to allow very fine movements.

In Angle Mode:

Key	Movement restriction
[Shift]	From center to perimeter only
[Ctrl]/[Command]	Along the perimeter only (at current distance from center)

There is also a special set of key commands for working in the SurroundPanner window.

⚠ For a complete list of the available key commands, click on the SurroundPanner logo and then click again!

The LFE encoder (all modes)

If the selected surround setup includes an LFE channel, a separate LFE level encoder is available in the Surround-Panner window. Use this to set the signal amount sent to the LFE channel. For further possibilities to set the LFE level, see the chapter "Surround sound" in the Operation Manual.

Mono/Stereo pop-up menu (all modes)

If you have a mono channel, the Mono/Stereo (Mo./St.) pop-up menu is set to Mono Mix by default. The panner will then behave as described above.

If you have a stereo channel, you have the option of using one of the three Mirror modes. Two gray balls will then appear, one for each channel (L/R). This will allow you to move the two channels symmetrically, by dragging one of them. The three modes allow you to select which axis should be used for mirroring.

- The default mode for stereo channels is the Y-Mirror mode.
- If you run a stereo signal through the panner in Mono Mix mode, the two channels are mixed together before entering the plug-in.
- If you run a mono signal through the plug-in using one of the stereo modes, the signal is split before entering the plug-in.

Additional parameters (Standard mode)

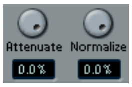
▪ Center level.

The Center control determines how center source signals are reproduced by the front speakers. With a value of 100%, the center speaker provides the center source. With a value of 0%, the center source is provided by the ghost image created by the left and right speakers. Other values will produce a mix between these two methods.

- Divergence controls.

The three divergence controls determine the attenuation curves used when positioning sound sources for X-axis front (Front), X-axis back (Rear), and Y-axis (F/R, front/rear), respectively. If all three divergence controls are set to 0% (default), positioning a sound source on a speaker sets all other speakers to zero level (-∞) (except for the center speaker which depends on the center level). With higher values, the other speakers receive a percentage of the sound source.

Additional parameters (Position and Angle modes)



- Attenuate.

Attenuate can be used to amplify or weaken the source. Exactly what effect this has on the level in each speaker can be determined by the level readouts, the concentric circle (Position mode) and the arc (Angle mode).

- Normalize.

Normalize is a function for controlling the overall loudness from all speakers. When this is set to 1.0 (full normalization), the level from all speakers together is always exactly 0dB. The individual levels will then be boosted or attenuated accordingly.

⚠ Please note that this is not a dynamic feature, like compression or limiting. It is instead just a tool for scaling the nominal output levels from the surround channels.

SurroundPanner V5

For a description of the SurroundPanner V5 plug-in, see the chapter “Surround sound” in the Operation Manual.

Surround plug-ins

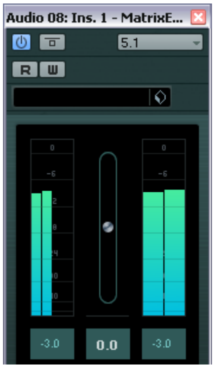
This section describes the plug-ins in the “Surround” category.

MatrixDecoder

The MatrixDecoder reverses the Encoder process performed by the MatrixEncoder (see below). It is used for monitoring how an encoded mix will sound when played back on a Pro Logic compatible system. When an encoded mix is played back via the decoder, the Lt/Rt channels are again converted to four outputs (LRCS).

⚠ This manual does not attempt to explain the full background on how Pro Logic works, but focuses on how you can use the MatrixEncoder/Decoder to produce a mix that is compatible with this standard.

MatrixEncoder



The 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. The MatrixEncoder takes four separate inputs (LRCS = Left, Right, Center, and Surround) and creates two final outputs: Left-total and Right-total (Lt and Rt).

Setting up

1. In the VST 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, see "Using the MatrixEncoder with the 5.0 surround format" on page 55.

2. Place the MatrixEncoder in the first "post fader" insert slot (#7) for the output bus, followed by the MatrixDecoder (#8).

Using the MatrixEncoder/Decoder

1. Set up the mix roughly the way you want it.

Use the SurroundPanner V5 to place channels in the surround mix, or assign channels to the individual LRCS outputs.

2. Activate the MatrixEncoder.

What you now hear is the encoded stereo mix, the way it will sound when 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 the MatrixDecoder, open the control panel and click the Steering Mode button.

Now you can hear how the mix will be reproduced in surround on a Pro Logic compatible system.

- The "Steering" display shows an 'x' within the surround field. The position of this x sign indicates the dominant direction of the mix, sometimes referred to as the "dominance vector". Part of the processing that is applied for various technical reasons results in the dominant channel being enhanced and the non-dominant channels being reduced in gain.

4. By activating and deactivating the Bypass button in the MatrixDecoder, you can compare the decoded mix with the encoded stereo mix, and make adjustments in the Mixer 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, switch off both the MatrixEncoder and the Decoder.

⚠ The encoding/decoding process will produce 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 the MatrixEncoder, since neither the encoder or decoder can "control" the mix in any way.

5. When you are satisfied with the result, bypass the 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.

The resulting encoded stereo mix will be compatible with common home systems that use the Pro Logic standard.

Using the MatrixEncoder with the 5.0 surround format

There are situations when 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.

Proceed as follows:

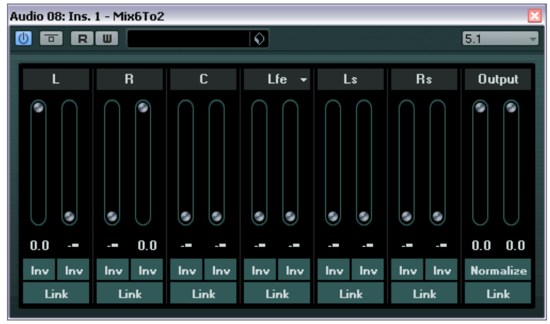
1. Create your mix for 5.1.
2. In the VST Connections window, create an output bus with a "5.0" channel configuration and route it to the physical outputs of your audio hardware.
3. Run the mix through the MatrixEncoder.

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 the 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 will be included in the resulting mix.

⇨ Mix6To2 does not simulate a surround mix or add any psycho-acoustical artifacts to the resulting output – it is simply a mixer. The plug-in should be placed in one of the post-fader insert effect slots for the output bus.

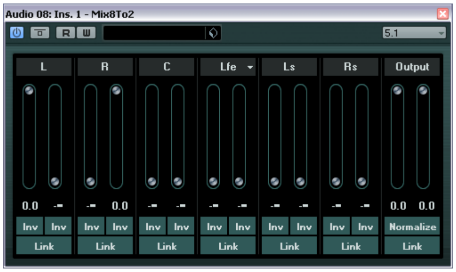
For each of the surround channels the following parameters are available:

- Two volume faders that govern how much of the signal will be included in the left and/or right channel of the output bus.
- A Link button that links the two volume faders.
- Two Invert buttons that allow you to invert the phase of the left and right channel of the surround bus.

For the Output bus the following parameters are available:

- A Link button that links the two Output faders.
- A Normalize button. If activated, the mixed output is normalized, i.e. 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 will be included in the resulting mix.

⇨ Mix8To2 does not simulate a surround mix or add any psycho-acoustical artifacts to the resulting output – it is simply a mixer. The plug-in should be placed in one of the post-fader insert effect slots for the output bus.

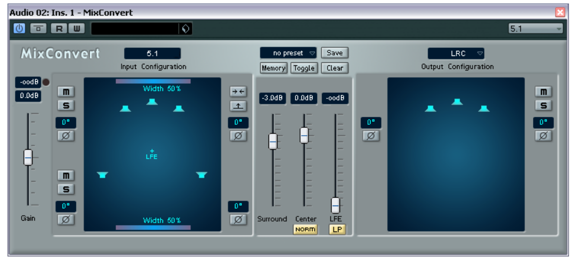
For each of the surround channels the following parameters are available:

- Two volume faders that govern how much of the signal will be included in the left and/or right channel of the output bus.
- A Link button that links the two volume faders.
- Two Invert buttons allow you to invert the phase of the left and right channel of the surround bus.

For the Output bus the following parameters are available:

- A Link button that links the two Output faders.
- A Normalize button. If activated, the mixed output is normalized, i.e. the output level is automatically adjusted so that the loudest signal is as loud as possible without clipping.

MixConvert



The MixConvert plug-in is similar to the Mix6To2 plug-in in that it can be used to quickly convert a multi-channel mix into another format that uses less channels when used as insert (for example converting a 5.1 surround mix to a stereo mix). MixConvert converts surround mixes into other surround formats, for example to mix down a 7.1 Cinema surround format to a 5.1 home theater format.

There are several obvious applications for this:

- Auditioning what an automatically generated downmix will sound like at the customer's location.
- Quickly generating an additional mix that uses a different number of channels or a different speaker configuration.
- Outputting several mix configurations simultaneously in various surround formats for broadcast purposes.

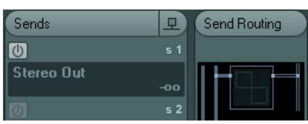
Users can use presets with standard upmix/downmix setups for specific configurations. It is possible to save up to 64 user-defined presets for each input/output configuration.

MixConvert is unique as a plug-in since it is used automatically by Nuendo in certain situations (like SurroundPanner). Nuendo will substitute MixConvert for the panner in either the main channel or in the aux send panner position when an upmix or a downmix is needed. These are the possible scenarios:

- Whenever a multi-channel audio track, group channel, or FX channel (with more than three audio paths) is routed to an output bus or group channel with a different number of audio paths (e.g. 5.1 to stereo), the MixConvert plug-in is inserted instead of the panner in that channel.

Indicates that MixConvert is inserted instead of the panner.

- Whenever a multi-channel audio track, group channel, FX channel, or output bus has an aux send that is routed to a group channel or output bus with a different number of audio paths, MixConvert will be inserted instead of the aux send's panner.



Indicates that MixConvert is inserted in the aux send panner position.

Interface

The plug-in panel has three different sections. On the left you will find the Input Configuration section with all corresponding parameters. In the middle section the level parameters for the upmix/downmix are displayed. Above this, the preset controls can be found. On the right you will find the Output Configuration section with all corresponding parameters. Additionally, on the far left there is a Gain fader.

In the following sections all controls are explained in detail. Note that when you move the mouse pointer over a control, a tooltip is displayed at the bottom of the MixConvert window.

Gain section

In this section the following parameters are available:

Parameter	Description
Global Gain fader	Attenuates or increases all channels to compensate for clipping or low levels in the converted signal. Gain depends on the input signal, the number of loudspeakers and a number of downmix parameters (see "Upmix/ Downmix parameters" on page 58). You can use this fader to globally adjust the gain by ±12dB for all channels.
Max Output Level field	This field above the Gain slider shows the maximum output level.
Max Output Level LED	The LED to the right of the field indicates whether this maximum level is above 0dB (clipping). Click the LED to reset the value field and the indicator.

Input Configuration

The input configuration is determined by the channel width of the track, group or output bus MixConvert is inserted in.

In this section the following parameters are available:

Parameter	Description
Mute button – front or surround channels	Mutes all front or surround channels.
Solo button – front or surround channels	Soloes all front or surround channels ("Solo mode" on page 59).
Phase Shift buttons (0°, 90°, 180°, 270°)	Shift the phase of the front left or right channel, or the surround left or right channel. Click the corresponding button to increase the phase by 90°. Right-click/[Ctrl]-click to reset to 0°. (For more information on phase shifting, see "Phase shifting" on page 59).
Solo to Center button	When this button is enabled, all speakers that are soloed are heard on the center channel (if available). If no center channel is present (as with stereo), the signal from the soloed channel is distributed equally to the left and right speakers.
Rear to Front button	Solos the rear channels and routes them to the front speakers.

Parameter	Description
Speaker symbols and LFE	Click on a speaker symbol to solo the speaker. If you hold down [Alt]/[Option] while clicking, the channel is muted. Holding down [Ctrl]/[Command] activates the exclusive solo (mutes all other channels even if they are also solo). Clicking again (without a modifier key) resets the channel.
Width controls	The front and back Width controls are used to set the width of the audible panorama. At minimal width (0%) the panorama is very narrow. In most cases, a setting of 50% will be appropriate as it results in unaltered signals. Values above 50% create an artificial widening of the panorama; similar to phase shifting. Be careful about modifying the panorama width when you want to generate matrixed downmixes. Drag the Width controls (the colored lines at the top and bottom of the Input Configuration display) to set the width. You can also click on the name of the control to open a pop-up menu from which you can select set values (0%, 25%, 50% and 100%)

⚠ Any signals that are equally in either the surround channels or the main left and right channels will be completely out of phase (180°) when the Width parameter is set to 100%. This will cause those signals to be completely cancelled when played over a mono system, such as AM radio broadcast or mono television. Always check for mono compatibility with mixes that are to be broadcast.

Upmix/Downmix parameters

The faders in the middle section of the plug-in panel control the levels for the surround channels, front center channel and LFE channel in the upmix/downmix. The surround channels cannot be modified individually. For center and surround channels, the level can be changed between -× and +6dB. For the LFE channel it can be changed between -× and +10dB, since in some mixes the LFE channel may be attenuated by 10dB (see "LFE channel" on page 59). The names Surround, Center and LFE refer to the corresponding channels in the input configuration.

In this section the following parameters are available:

Parameter	Description
Preset pop-up menu	Allows you to load a preset (see "Loading and saving presets" on page 58).
Save Preset button	Allows you to save a preset or delete the preset shown in the Preset pop-up menu.

Parameter	Description
Memory button	You can use the Memory, Toggle, and Clear buttons to toggle between two different sets of downmix parameters for direct comparison. Click the Memory button to write all current parameters to the temporary parameter buffer. Note this does not include the output configuration, which must be identical for both parameter sets.
Toggle button	Using the Toggle button you can switch between the buffered parameter set and the (changed) current parameter set.
Clear Memory	Clears the temporary parameter buffer.
Surround fader	Sets the level of the surround channel.
Center fader	Sets the level of the center channel.
LFE fader	Sets the level of the LFE channel.
Norm button	Normalizes all speaker channels.
LP button	Enables/disables the low-pass filter (120Hz) applied to the LFE channel.

Output Configuration

When Nuendo automatically replaces the panner by MixConvert, the output configuration is determined by the destination of the channel or aux send. However, the output configuration can be modified when used as an insert effect. You either change it directly in the pop-up menu at the top of the Output Configuration section or indirectly by loading a preset.

In this section the you will find the same parameters as in the Input Configuration section (see above), except for the Width controls, and the "Solo to Center" and "Rear to Front" buttons.

General Notes

Loading and saving presets

Full presets are only available for MixConvert when it is used as an insert effect. When Nuendo automatically places MixConvert instead of a panner, the preset menu displays only presets for the current input/output configuration.

Presets are selected and managed at the top of the middle section of the plug-in panel. The name of the selected preset is displayed in the text field. Click the symbol next to the text field to open a pop-up menu from which you can select a different preset. Which presets are available from this pop-up menu, depends on the downmix options available for the current input configuration. You save a new set of parameters by entering a new name in the text field and selecting Save Preset from the pop-up menu that appears when you click the Save button. You can save up to 64 presets for every input/output configuration. To delete a user preset, select Delete Preset from the Save pop-up menu. Note that the factory-defined presets cannot be deleted.

Phase shifting

Phase shifting can be used for various purposes. In a downmix from 2 channels to 1 channel it may be useful to introduce a 90° phase shift on one channel to avoid level increases in the downmix signal (caused by frequencies present in both channels). Also, phase shifts can be used to create "virtual" reverberation by cancelling all center information, leaving the resulting ambience.

⚠ As a general rule, be careful when using phase shifts, as they might have negative repercussions on the frequency spectrum and the level of the downmix. Also, when you generate matrixed downmixes, avoid introducing additional phase shifts, since these prevent the decoding of the mix for different speaker configurations.

Level

The volume of the downmixed signal can be different from the volume of the original mix. There are several reasons for this:

- The input signals must be scaled to avoid clipping.
- The number of speakers used influences the overall volume.
- The level of the downmixed signal depends on the correlation of all added signals, which is why phase shifting can influence the volume level.

LFE channel

The LFE channel is automatically filtered using a low-pass filter. The cutoff frequency of this low-pass filter is 120 Hz, the filter slope is 12 dB/Oct. An LFE channel present in the input configuration, but not present in the output configuration, is mixed evenly to the front-left and front-right channels since it is assumed that these will be the channels using the speakers with the widest frequency range.

Solo mode

Since there is no dedicated solo bus, all solos are inplace, i.e. all other (non-solo) channels are muted.

Available conversions

Not all theoretically possible combinations are actually available in MixConvert since the plug-in is limited to channels with 8 audio paths (this means that 10.2 or 8.1 are not supported). For a list of all available combinations, see "MixConvert Appendix" on page 85.

MixConvert-ControlRoom

The MixConvert-ControlRoom plug-in is identical to the MixConvert plug-in. It can convert surround mixes into other surround formats such as mixing a 7.1 Cinema surround format down to a 5.1 home theater format. The decisive difference to the MixConvert plug-in is, that this plug-in has no latency.

MixerDelay

MixerDelay allows you to adjust and manipulate each individual channel in a surround track, group or bus.

- Above the individual channel controls you will find global buttons for turning off Mute, Solo and Input Phase switches for all channels.

For each channel the following controls are available:

Parameter	Description
Mute button	Allows you to mute individual channels.
Solo button	Allows you to solo individual channels.
Inv button	Lets you invert the phase or polarity for individual channels.
Delay slider	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, etc.
Level slider	Allows you to fine-tune the volume balance between the surround channels.

Parameter	Description
Volume meter	Shows the level of the input signal.
Routing section	Lets you select/switch the desired outputs for the channels quickly. You can assign the same output to several channels by holding down the [Alt]/[Option] key while selecting. Note that there are also several channel routing presets available.

⇨ 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. In cases like this, 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 the sound from it 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.

⚠ 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 plug-ins.

SurroundDither

SurroundDither is not an "effect" as such. Dithering is a method for controlling the noise produced by quantization errors in digital recordings. The theory behind this is that during low-level passages, only a few bits are used to represent the signal, which leads to quantization errors and hence distortion. For example, when "truncating bits" as a result of moving from 24- to 16-bit resolution, quantization errors are added to an otherwise immaculate recording. By adding a special kind of noise at an extremely low level, the effect of these errors is minimized. The added noise could be perceived as a very low-level hiss under exacting listening conditions. However, this is hardly noticeable and much preferred to the distortion that otherwise occurs.

When should I use SurroundDither?

- Basically anytime you mix down to a lower resolution, either in realtime (playback) or with the Export Audio Mixdown function, you should consider dithering.
- Since SurroundDither is capable of dithering up to eight channels at the same time, it is recommended to use this plug-in for surround channels.
If not, you may want to use the UV22HR instead, see "Mastering – UV22HR" on page 28.

The following options can be set in the SurroundDither control panel:

Dithering Type

There are no hard and fast rules for the following options, it all depends on the type of material you are processing. We recommend that you experiment and let your ears be the final judge:

Option	Description
Off	No dithering is applied.
Type 1	Try this first, it is the most "allround" type.
Type 2	This method emphasizes higher frequencies more than Type 1.

Noise Shaping Options (Off, Type 1–3)

This parameter alters the character of the noise added when dithering. Again, there are no fixed general rules, but you may notice that the higher the number selected here, the more the noise is moved out of the ear's most sensitive range, the mid-range.

Ditherbits

This is used to specify the intended bit resolution for the final result.

- The section has eight buttons, one for each channel.
If the selected channel has less than eight sub-channels, the additional channel buttons are grayed out.

- Above each button there is a value field that displays the bit resolution the file will be converted to.
Clicking a button several times cycles through the available bit resolution values.

An example

Say you have set up a project to record 24-bit files. After completion, you want to create a digital 16-bit master for CD burning. Proceed as follows:

1. For the output bus, add SurroundDither to a post-fader insert effect slot.
This can be one of the last two slots.

2. Open the SurroundDither control panel, and select the Dithering and Noise Shaping type.

3. Set the Ditherbit destination to "16" for all the master mix outputs currently used, as defined in the VST Connections dialog.
If you are not using surround channels, this will be channels 1 and 2.

4. When you now play back the project, the digital outputs of your audio hardware will output the mix with 16-bit resolution, with dithering applied.

Tools – MultiScope

MultiScope can be used for viewing the waveform, phase linearity or frequency content of a signal. There are three different modes:

- Oscilloscope (Ampl.)
- Phase Correlator (Scope)
- Frequency Spectrum Analyzer (Freq.)

⇨ The Freeze button can be used to freeze the display in all three modes. Click it again to exit freeze mode.

Oscilloscope mode (Ampl.)

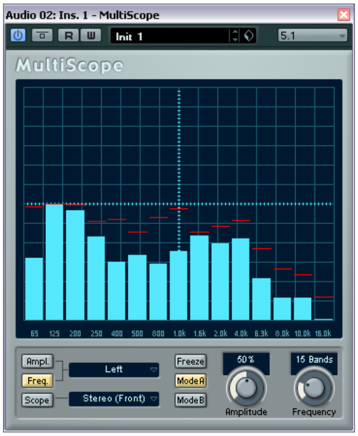
- To view a signal waveform, open the MultiScope control panel and make sure that the "Ampl." button in the lower left corner is lit.

- If the source signal is stereo you can now select either the Left or Right channel for viewing, or Stereo for both channels to be shown in the window. If it is a mono signal, this does not matter.

- If MultiScope is used with a multi-channel track or output bus, you can select any speaker channel for viewing, or All Channels to view them all at once.

- You can now adjust the Amplitude knob to increase/decrease the vertical size of the waveform, and the Frequency knob to select the frequency area for viewing.

Frequency Spectrum Analyzer mode (Freq.)



- Click the Freq button so that it lights up.

MultiScope now divides the frequency spectrum into separate vertical bands, which allows you to get a visual overview of the different frequencies' relative amplitude. The frequency bands are shown left to right, starting with the lower frequencies.

- If the source signal is stereo you can now select either the Left or Right channel for viewing, or Stereo for both channels to be shown in the window. If it is a mono signal, this does not matter.

- If MultiScope is used with a multi-channel track or output bus, you can select any speaker channel for viewing, or All Channels to view them all at once.

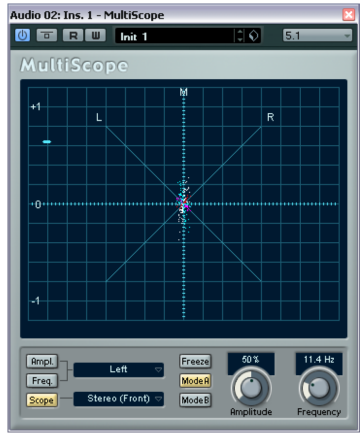
- Adjust the Amplitude knob to increase/decrease the vertical range of the bands.

- By adjusting the Frequency knob, you can divide the frequency spectrum into 8, 15, or 31 bands, or you set it to "Spectrum", which gives you a high-resolution view.

- Use the Mode A and Mode B buttons to switch between different view modes.

Mode A is more graphically detailed, showing a solid, blue amplitude bar for each band. Mode B is less detailed, showing a continuous blue line that displays the peak levels for each band. These view modes do not have any effect if you have set the Frequency knob to "Spectrum".

Phase Correlator mode (Scope)



- Click the Scope button so that it lights up.

The phase correlator indicates the phase and amplitude relationship between channels in a stereo pair or a surround configuration.

For stereo pairs, the indications work in the following way:

- 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 random but fairly round shape indicates a well balanced stereo signal. If the shape "leans" to the left, there is more energy in the left channel and vice versa (the extreme case of this is if one side is muted, in which case the phase meter will show a straight line, angled 90° to the other side).
- A perfect circle indicates a sine wave on one channel, and the same sine wave shifted by 90° on the other.
- Generally, the more you can see a "thread", the more bass in the signal, and the more "spray-like" the display, the more high frequencies in the signal.

When MultiScope is used with a surround channel in Scope mode, the pop-up menu to the right of the Scope button determines the result:

- If "Stereo (Front)" is selected, the display will indicate the phase and amplitude relationship between the front stereo channels.

- If "Surround" is selected, the display indicates the energy distribution in the surround field.

2

MIDI effects

Introduction

This chapter describes the included MIDI realtime effects and their parameters.

How to apply and handle MIDI effects is described in the chapter "MIDI realtime parameters and effects" in the Operation Manual.

Arpache 5

A typical arpeggiator accepts a chord (a group of MIDI notes) as input, and plays back each note in the chord separately, with the playback order and speed set by the user. The Arpache 5 arpeggiator does just that, and more. Before describing the parameters, let's look at how to create a simple, typical arpeggio:

1. Select a MIDI track and activate monitoring (or record enable it) so that you can play "thru" the track.
Make sure that the track is properly set up for playback to a suitable MIDI instrument.

2. Select and activate the arpeggiator.
For now, use it as an insert effect for the selected track.

3. In the arpeggiator panel, use the Step Size setting to set the arpeggio speed.
The speed is set as a note value, relative to the project tempo. For example, setting Step Size to "16" means the arpeggio will be a pattern of sixteenth notes.

4. Use the Length setting to set the length of the arpeggio notes.
This allows you to create staccato arpeggios (Length value smaller than the Step Size setting) or arpeggio notes that overlap each other (Length value greater than Step Size).

5. Set the Key Range parameter to 12.
This will make the notes arpeggiate within an octave.

6. Play a chord on your MIDI instrument.
Now, instead of hearing the chord, you will hear the notes of the chord played one by one, in an arpeggio.

7. Try the different arpeggio modes by clicking the Play Order buttons.
The symbols on the buttons indicate the playback order for the notes (Invert, Up Only, etc.). The settings are described below.

Parameters

The Arpache 5 has the following settings:

Setting	Description
Play Order buttons	Allows you to select the playback order for the arpeggiated notes. The options are Normal, Invert, Up only, Down only, Random, User. 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. The range is 32T (1/32 note triplets) to "1." (dotted note values).
Length	Sets the length of the arpeggio notes, as a note value related to the project tempo. The range is the same as for the Step Size setting.
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 will be 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 will be added to the arpeggio (as many octaves as fit within the range).

Setting	Description
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 pressed key. So, if you play the notes C3-E3-G3 (a C major chord), “1” would mean C3, “2” would mean E3, and “3” would mean G3. Note that you can use the same number in several slots, creating arpeggio patterns that are not possible using the standard play modes. Please note that you need to begin with the left-most slot and then fill the slots to the right. 
MIDI Thru	If this is activated, the notes sent to the arpeggiator (i.e. the chord you play) will pass through the plug-in (sent out together with the arpeggiated notes).

Arpache SX

This is an even more versatile and advanced arpeggiator, capable of creating anything from traditional arpeggios to complex, sequencer-like patterns. The Arpache SX has two different modes: Classic and Sequence.

Classic vs. Sequence mode

The Classic mode determines the basic behavior of the Arpache SX. When Sequence mode is selected, the Arpache SX uses the events of an additional MIDI part as a pattern. This pattern then forms the basis for the arpeggio, in conjunction with the MIDI input.

Classic mode

The following parameters are available:

Parameter	Description
Direction	This allows you to choose how the notes in the chord you play should be arpeggiated. In Classic mode you can choose a value from a pop-up menu, in Sequence mode you will find additional options, see below.
One Shot Mode	Activate this option if you want the phrase to be played only once. When this option is deactivated, the phrase will be looped.
Transpose	When a setting other than “Off” is selected, the arpeggio will be expanded upwards, downwards or both (depending on the mode). This is done by adding transposed repeats of the basic arpeggio pattern.
Repeats	The “Repeats” setting sets the number of transposed repeats.
Pitch Shift	The “Pitch Shift” setting determines the transposition of each repeat.
MIDI Thru	If this is activated, the notes sent to the arpeggiator (i.e. the chord you play) will pass through the plug-in (sent out together with the arpeggiated notes).
Step Size	Determines the resolution of the arpeggio, i.e. 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, see below.
Length	Determines the length of the arpeggio notes (in fixed note values or PPQ, if the PPQ button is activated). In Sequence mode you can also activate the “from sequence” option, see below.
Max. Polyphony	Determines how many notes should be accepted in the input chord. The “All” setting means there are no limitations.
Sort by	When you play a chord into the Arpache SX, the arpeggiator will sort 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 will be the first note, E will be 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 corresponding notes in the chord you play. In Sequence mode you can also activate the “from sequence” option, see below.

Sequence mode

In Sequence mode you can import a MIDI part into the Arpache SX by dragging it from the Project window and dropping it in the “Drop MIDI Sequence” field on the right of the Arpache SX panel.

Now, the notes in the dropped MIDI part will be sorted internally, either according to their pitch ("MIDI Seq. sort by pitch" checkbox 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 will read 1 2 3 4 2 1. Here, there are 4 different notes/numbers and 6 trigger positions.

The MIDI input (the chord you send into the Arpache SX) will generate 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 will be matched – the Arpache SX tries to play back the pattern from the dropped MIDI part but using the notes from the MIDI input (chord). The result depends on the Play Mode setting:

Option	Description
Trigger	The whole pattern from the dropped MIDI file will be 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 when all keys are released, the phrase continues playing from the last position (where it stopped), when 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 (numbers) in the MIDI input, some steps in the resulting arpeggio will be empty.
Sort First	As above, but if there are fewer notes in the MIDI input, the missing notes will be replaced by the first note.
Sort Any	As above, but if there are fewer notes in the MIDI input, the missing notes will be replaced by any (random) note.
Arp. Style	As above, but if there are fewer notes in the MIDI input, the missing notes will be replaced by the last valid note in the arpeggio.
Repeat	In this mode, the chords played will not be separated into notes. Instead they will be used as is, and only the rhythm of the dropped MIDI part is used for playback.

Note also that 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 plug-in 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. The Auto LFO effect has the following parameters:

Waveform

These settings determine the shape of the controller curves sent out. You can click on a waveform symbol, or choose a value from the pop-up menu.

Wavelength

This is where you set the speed of the Auto LFO, or rather the length of a single controller curve cycle. Using the slider or by choosing an entry from the pop-up menu, 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. For example, if you set this to "1/8", the waveform will be repeated every eighth note.

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

This determines the density of the controller curves sent out. The value can be set to “small”, “medium”, or “large”, or to rhythmically exact note values (by choosing from the pop-up menu). The higher the note value, the smoother the controller curve. For example, if you set this to “1/16”, a new controller event will be sent out at every 1/16 note position.

Value Range

These two sliders are used to determine the range of controller values sent out, i.e. the “bottom” and “top” of the controller curves.

Beat Designer (Nuendo Expansion Kit only)

The Beat Designer is a MIDI pattern sequencer that allows you to create your own drum parts or “patterns” for a project. With the 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 will work on a short sequence, adjusting and modifying it while playing it back in a loop until you get the desired result. The drum patterns can then either be converted to MIDI parts on a track or triggered using MIDI notes during playback, see “Converting patterns into MIDI parts” on page 71 and “Triggering patterns” on page 71.

To use the Beat Designer, select it as MIDI insert effect for a MIDI track (routed to a VSTi or an external device) or an instrument track.

Overview

When you open the control panel for the Beat Designer for the first time, it shows a display with 8 empty lanes, each containing 16 steps.

Patterns and subbanks

The 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 the Beat Designer, subbanks and patterns are displayed graphically. To select a subbank, click on a number (1 to 4) at the top of the display. To select a pattern within this subbank, click on 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 the desired value.
The maximum number of steps is 64. By default, 16 steps are shown.

- The playback length, i.e. the note value for the steps, can be specified in the Step resolution pop-up menu next to the Number of Steps setting.
On this menu, you can also set triplet values. These also affect the Swing setting, see “The Swing setting” on page 69. The default setting is 1/16.

Number of steps for this pattern Step resolution

Selecting drum sounds

To specify a drum sound, click in the drum name field for a lane and select the desired 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.

- To find the right sound, you can audition the selected drum sound by clicking the Preview Instrument button (the speaker icon).

Entering drum steps

To enter a drum step, click on the step field where you want to add a beat. You could e.g. add a snare drum on each downbeat for a lane and a bass drum on a second lane. When you click in an empty field, it becomes “filled”, indicating that you will hear a drum beat on this step.

You can also click and drag to enter a continuous range of drum steps.

⇨ 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, as this allows you to hear the result immediately.

Removing steps

- To remove a drum step, simply click on the corresponding field again.

- To remove a range of drum steps, click and drag over them.

Setting the velocity

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. This is a quick way of roughly setting the velocity on the fly while entering drum sounds. In the display, the different velocity settings are indicated by different colors.

- You can fine-tune the velocity setting for an existing drum step by clicking on it and dragging up or down.
The current velocity is indicated numerically while you drag, allowing you to find the desired setting easily. The available range is from 1 to 127.

- You can also fine-tune the velocity for a range of drum steps. Click on the first step, drag up or down to enter into velocity edit mode, and then drag sideways and up or down to modify the velocity for all the steps.

- If you hold down [Shift] while dragging up or down, you can change the velocity for all steps on a lane.

⇨ If you change the velocity for several steps at the same time, the relative velocity differences will be kept for as long as possible (until the minimum or maximum setting is reached).

The velocity for the steps will be increased or decreased by the same amount.

- You can also create a crescendo (or decrescendo) for an existing range of drum steps by holding down [Alt]/[Option], clicking on the first step, dragging up or down and then dragging to the left or right.

Editing operations

- You can move all drum steps on a lane by holding down [Shift], clicking on the lane and dragging to the left or right.

- You can also “invert” a lane, i.e. add drum sounds for all steps that were empty while removing all existing drum steps. This lets you create unusual rhythmic patterns. To do so, hold down [Alt]/[Option] and drag the mouse over the lane.

- You can copy the content of a lane onto another lane by holding down [Alt]/[Option], clicking in the section to the left of the lane you want to copy and dragging to the desired position.
When you drag, a vertical line and a plus symbol will be displayed.

Lane handling

If you find that you have too many or too few lanes in the Beat Designer, you can add or remove them.

- To add a lane, click on the “Add Instrument Lane” button at the bottom right of the last lane shown.

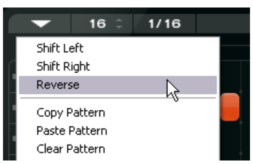
- To remove a lane, click on the “Remove Instrument Lane” button in the controls section at the far right of the lane.

- You can change the order of the drum lanes by clicking in an empty area in the section to the left of a lane (i.e. not on a button) and dragging it to another position.

- You can mute or solo a lane by clicking the respective buttons to the left of the step display.

⚠ The lane operations always affect all patterns in the Beat Designer instance, not only the one you edit.

The Pattern Functions menu



This menu contains the following editing functions:

Option	Description
Shift Left	This moves all steps of the current pattern (all steps on all lanes) to the left.
Shift Right	This moves all steps of the current pattern (all steps on all lanes) to the right.
Reverse	Reverses the pattern, so that it plays backwards.
Copy Pattern	This copies the pattern to the clipboard. Copied patterns can be pasted into another pattern subbank (see below), and even directly into the project. The default key command for this is [Ctrl]/[Command]-[C].
Paste Pattern	Allows you to paste a complete pattern, e.g. into another pattern subbank, even into another instance of the Beat Designer. This is handy when you want to create variations based on existing patterns. The default key command for this is [Ctrl]/[Command]-[V].
Clear Pattern	This resets the current pattern.
Insert Pattern at Cursor	This creates a MIDI part for the current pattern and inserts it in the Project window, at the position of the project cursor (see also “Converting patterns into MIDI parts” on page 71).
Insert Subbank at Cursor	This creates a number of MIDI parts (one for each used pattern in the subbank) and inserts them one after the other, starting at the project cursor (see also “Converting patterns into MIDI parts” on page 71).
Insert Pattern at Left Locator	This creates a MIDI part for the current pattern and inserts it in the Project window, at the left locator (see also “Converting patterns into MIDI parts” on page 71).
Insert Subbank at Left Locator	This creates a number of MIDI parts (one for each used pattern in the subbank) and inserts them one after the other, starting at the left locator (see also “Converting patterns into MIDI parts” on page 71).
Fill Loop with Pattern	This 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 (the space between the left and right locators), see also “Converting patterns into MIDI parts” on page 71.

- You can set up key commands for the Insert options and the Fill Loop command in the Key Commands dialog.

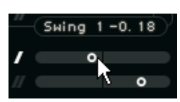
How to set up and use key commands is described in the chapter “Key Commands” in the Operation Manual.

The Swing setting

This parameter can be used to create a swing or shuffle rhythm, which allows you to add a more human feel to drum patterns that might otherwise be too static. This is done by offsetting every second drum step for a lane. If a triplet step resolution is used, every third drum step will be offset instead.

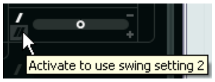
In the lower right section of the Beat Designer panel, you can find two Swing sliders. Dragging a slider to the right will delay every second (or third, see above) drum step in the pattern. Dragging to the left will make them play a little earlier.

You can set up two swing settings with these sliders and then quickly switch between these during playback. By default, the first swing setting is used (activated) in all lanes, but the slider is set to zero (middle position). Change the setting for this slider to hear how the pattern’s feel changes.



Drag the upper fader to set swing setting I and the lower fader to set swing setting II.

You can switch between the two swing settings using the Swing buttons to the right of the step display.



Click on the buttons to select the respective swing setting or click on a selected button to deactivate swing for this lane.

Adding flams

The Flam parameter lets you add flams (short secondary drum hits just before or after the actual main drum beat).

You can add up to three flams for each pattern step:

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 with the mouse at the step. After you clicked, the first square becomes filled to indicate that you added a flam.

Click here to add up to three flams to the step.

2. Click again to add the second and third flam, if needed.

3. In the lower left section of the Beat Designer panel you can make settings for the flams you created.

Here, you can specify the flam positions for all steps containing one, two and three flams, respectively.

With these sliders, you can specify the velocity for the separate flams.

- The first (topmost) 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.
- Drag a Position slider to the left to add the flams before the drum step and to the right to add them after the step.
- When you add flams before the very first drum step in a pattern, this is indicated in the display by a small arrow in the top left corner of this step. This indicates that you have to treat this pattern with special care in playback and arranging. Starting playback at the normal pattern start would result in these flams not being played.
- Use the vertical sliders to the right of the flam sliders to set the velocity for the flams.

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. Drag a slider to the left to make the drum steps start a little earlier and to the right to let them start later.

Playing e.g. the bass drum or snare a little earlier allows you to add more “urgency” to the drums, delaying these drum sounds will result in a more relaxed drum pattern. Experiment with the settings to find out which fit best in your project.

Note that 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.

Saving and loading presets

You can save all 48 Beat Designer patterns as a pattern bank. This can then be loaded in other projects. Pattern banks contain all the step and lane settings for a pattern (Mute and Solo, number and order of the lanes, pitch, etc.).

To save a pattern bank, proceed as follows:

1. In the Beat Designer, click on the Preset Management button to the right of the preset name field.

2. On the pop-up menu select “Save Preset”.
A dialog appears.

3. Enter a name for the preset and click OK.

The preset will now be available on the Preset browser, in the MediaBay and on the Load Track Preset pop-up menu in the Inspector.

Pattern banks are handled much like track presets in the MediaBay. For further information, refer to the chapters “The MediaBay” and “Working with track presets” in the Operation Manual.

Using the drum patterns in your project

You can use the drum patterns created with the Beat Designer in two ways: either by converting them to MIDI parts on a MIDI or instrument track or by triggering the different patterns using MIDI notes.

Converting patterns into MIDI parts

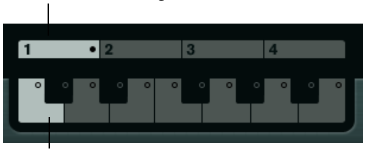
You can convert the drum patterns created in the Beat Designer into a MIDI part by dragging them into the Project window.

Proceed as follows:

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 at the desired position 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 will be an exact copy of the original track for which you opened the Beat Designer.

Click here and drag to convert this subbank into separate MIDI parts.



Click here and drag to convert this pattern into a MIDI part.

- 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.

⚠ Only the used patterns in a subbank are inserted, i.e. if you did not enter drum steps in a pattern, this will not be converted into a MIDI part.

You can also use the Pattern Functions menu to insert patterns or subbanks into the project, see “The Pattern Functions menu” on page 69.

⚠ When you have created MIDI parts for your drum patterns this way, make sure to deactivate the Beat Designer, to avoid doubling of the drums. The Beat Designer will continue 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 will be lengthened accordingly.

The inserted MIDI parts can now be edited as usual in the project. You can e.g. fine-tune your settings in the Drum Editor.

⇨ Once a pattern is converted into a MIDI part, it cannot be opened in the Beat Designer again.

Triggering patterns

When you want to be able to modify your drum patterns in the Beat Designer while working on the project, you cannot convert them into parts, as these cannot be opened again in the Beat Designer. Instead, you can trigger the patterns from within the project.

You can trigger the patterns in the 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 will be triggered depends on the pitch of the MIDI notes. The trigger range is four octaves starting with C1 (i.e. C1 to B4).

Proceed as follows:

1. Open the Beat Designer for a track.
Again, this can be a MIDI or an instrument track.

2. Click on the Jump field to activate Jump mode.
In this mode, a MIDI note-on event will trigger a new pattern.

Jump mode is activated.

▪ When you want to trigger the patterns using a MIDI part containing trigger events, you can specify whether the pattern will be switched directly (at the moment the event is received) or at the next bar: Click on the field to the right (where it says “Now”) to activate the immediate switching of patterns. When Now is deactivated, patterns will switch at the beginning of the next bar in the project.

▪ When 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.

Now, you can trigger the patterns in the following way:

1. Play back the project and press a key on your MIDI keyboard to trigger the next pattern.
The pattern will start at the next bar line.

2. 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 will be played directly or start at the following bar.

▪ You can also drag a pattern or subbank into the Project when Jump mode is active to automatically create MIDI parts containing the trigger events.

⇨ 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

The 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, see below.

For every key you can record up to eight different chords or variations on so-called “layers”. This is described in detail in the section “Using Layers” on page 73.

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 will be used to record your chords.

Global Key

In this mode, you can assign chords to each key on the keyboard display. When you play any of these keys, you will hear the assigned chords instead.

One Octave

The One Octave 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). When you play a note (e.g. C) on a different octave, you will hear a transposed version of the chords set up for this key.

Global Key

In Global Key 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.

The chord indicator lane

At the top of the keyboard display you will 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.

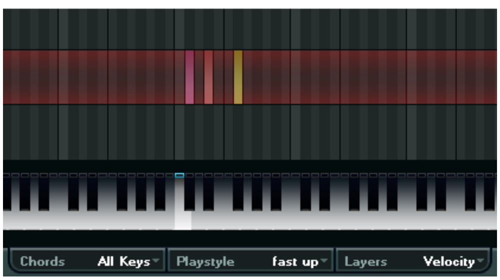
The chord indicator lane in One Octave mode with chords set up for 5 of the 12 available trigger keys.

⇨ In Global Key mode the C3 key has a special marking instead since this is the only key used in this mode.

Entering chords

To enter chords you need to switch to Learn mode. In this mode a transparent red bar indicates which element is ready for "learning" a note or chord. When you choose the trigger note for a chord, for example, the keyboard display is shown in red.

The keyboard display in Learn mode



The second layer in Learn mode

Proceed as follows:

1. Click 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.

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 will now move to the first layer, indicating that you are ready to record the first chord.

⇨ In Global Key mode you do not have to choose a trigger key. The first layer is activated directly.

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, the Chorder will learn the chord as soon as you release all keys of your MIDI keyboard simultaneously.
As long as a key is pressed, you can continue looking for the right chord.

- If more than one layer is shown, the Chorder will jump automatically to the next layer where you can record another chord.
When all the layers for a key are filled, the red bar will jump back to the keyboard display so that you can choose a different trigger key (in Global Key mode the Learn mode is deactivated instead).

- If you are entering chords with the mouse, the Chorder will not jump to the next layer automatically.
You can select/deselect as many notes as you wish and then click on another layer or deactivate the Learn mode to continue.

4. Repeat the above with any other keys you wish 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 eight 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. Proceed as follows to set up your layers:

1. Open the Layers pop-up menu and 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 (layers) you want to use.

3. Enter the chords as described above.

4. Now you can play the keyboard and trigger the variations according to the selected layer mode.

The layer modes work as follows:

Trigger mode	Description
Velocity	The full velocity range (1–127) is divided into “zones”, according to the number of layers you specified. For example, if you are using two variations (Number of Layers is set to 2) there will be two velocity “zones”: 1–63 and 64–127. Playing a note with velocity 64 or higher will trigger the second layer, while playing a softer note will trigger the first layer. Using the “Velocity spread” slider at the bottom right of the window, you can change the velocity ranges of the layers so that a different layer will be activated using the same velocity value.
Interval	In this mode, the Chorder will play one chord at a time – you cannot play several different chords simultaneously. When the Interval mode is selected, you press two keys on your keyboard to trigger the desired layer, with the lower key determining the base note for the chord. The layer number will be the difference, i.e. the interval, between the two 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 do not wish to use different layers.

Empty layers

If you enter less chords than layers present for a key, these layers will be filled automatically when you end the Learn mode.

This works according to the following rules:

- 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

In Learn mode, you can use the “Reset layers” button at the top left of the Chorder window to delete all notes in the different layers for the selected trigger key.

Playstyle

From the Playstyle pop-up menu at the bottom of the pane you can choose one of seven different styles that determine in which order the individual notes of the chords are played back.

The following options are available:

Playstyle	Description
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

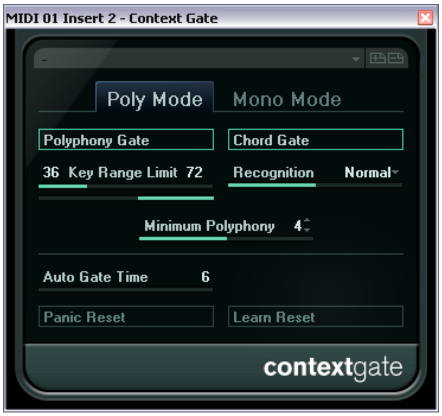
This MIDI compressor is used for evening out or expanding differences in velocity. Though the result is similar to what you get with the Velocity Compression track parameter, the Compress plug-in presents the controls in a manner more like regular audio compressors.

The following parameters are available:

Parameter	Description
Threshold	Only notes with velocities above this value will be affected by the compression/expansion.
Ratio	This determines the rate of compression applied to the velocity values above the threshold level. Ratios greater than 1:1 result in compression (i.e. less difference in velocity) while ratios lower than 1:1 result in expansion (i.e. greater difference in velocity). What actually happens is that the part of the velocity value that is above the threshold value is divided by the ratio value.

Parameter	Description
Gain	This adds or subtracts a fixed value from the velocities. Since the maximum range for velocity values is 0–127, you may need to use the Gain setting to compensate, keeping the resulting velocities within the range. Typically, you would use negative Gain settings when expanding and positive Gain settings when compressing.

Context Gate



The Context Gate allows for selective triggering/filtering of MIDI data. It features two modes: in Poly Mode the Context Gate recognizes certain chords that are played and in Mono Mode only certain MIDI notes are let through. These modes can be used for context selective control of MIDI devices and are, for example, very useful in certain live scenarios.

The following parameters are available:

Poly Mode – Polyphony Gate

This 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 are used to set the key range.

Only notes within this range will be let through.

- The "Minimum Polyphony" value field allows you to specify the minimum number of notes required to open the gate.

Poly Mode – Chord Gate

When 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, whereas Normal mode takes more tensions into account.

Mono Mode – Channel Gate

When this is activated, only single note events in a specified MIDI channel are let through, which can be used with MIDI controllers that can send MIDI over 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", i.e. no channel gating.

Mono Mode – Velocity Gate

This can be used independently or in conjunction with the Channel Gate function. Played notes will sound (no note-off message) until a note is played inside the set range (and additionally the set Channel Gate channel, if checked).

- The Key Range Limit sliders are used to set the key range.

Only notes within this range will be let through.

- Notes below the Minimum Velocity threshold value will be gated.

Auto Gate Time

If there is no input activity, all resounding notes are sent a note-off message after the set time, in seconds or milliseconds.

Panic Reset button

Sends an "All Notes Off" message over all channels, in case of hanging notes.

Learn Reset button

When 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, the Learn button should be deactivated.

Application examples

Poly Mode

In Poly mode, you could use the 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 the 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 will play until the Auto Gate Time is reached and fade 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 Mono Mode you could use the Context Gate to trigger variations played with a drum machine/VST instrument. To do this, you will 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 (e.g. beginning at the 12th band). When you now play one of these notes, the note-off command will not be send out and the corresponding note will sound 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 thru) the track. When this is set to 100%, the notes are not affected. Lowering the Density setting below 100% will randomly filter out or “mute” notes. Raising the setting above 100% will instead randomly add notes that have been played before.

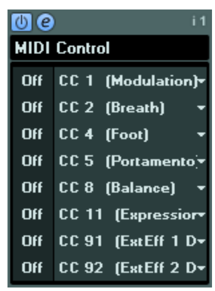
Micro Tuner

The 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).
- By keeping the [Alt]/[Option] key pressed, you can adjust all keys by the same amount.

The Micro Tuner comes with a number of presets, including both classical and experimental microtuning scales.

MIDI Control



This generic control panel allows you to select up to eight different MIDI controller types, and use the value fields or sliders (which are displayed when you click on a value field while holding down the [Alt]/[Option] key) to set values for these. A typical use for this would be if you are using a MIDI instrument with parameters that can be controlled by

MIDI controller data (e.g. filter cutoff, resonance, levels, etc.). By selecting the correct MIDI controller types, you can use the plug-in as a control panel for adjusting the sound of the instrument from within Nuendo, at any time.

- To select a controller type, use the pop-up menus to the right.
- To deactivate a controller slider, set it to “Off” (drag the slider all the way down).

MIDI Echo

This is an advanced MIDI Echo, which will generate 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. As always it is important to remember that the effect does not “echo” the actual audio, but the MIDI notes which will eventually produce the sound in the synthesizer.

The following parameters are available:

Velocity Offset

This parameter 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). For no change of velocity, set this to 0 (middle position).

Pitch Offset

If you set this to a value other than 0, the repeating (echoing) notes will be 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 will cause 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

This is the number of echoes (1 to 12) from each incoming note.

Beat Align

During playback, the Beat Align parameter quantizes the position of the first echo note. 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.

Setting this to “1/8”, for example, will cause the first echo note to sound on the first eighth position after the original note.

⇨ The echo time can also be affected by the Delay Decay parameter.

⇨ During live mode, this parameter has no effect since the first echo will always be played together with the note event itself.

Delay

The echoed notes will be repeated as set up with this parameter. 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. This makes it easy to find rhythmically relevant delay values, but still allows experimental settings in between.

Delay Decay

This parameter lets you adjust how the echo time should be changed with each successive repeat. The value is set as a percentage.

- When set to 100% (middle position) the echo time will be the same for all repeats (as set with the Delay parameter).
- If you raise the value above 100%, the echoing notes will play with gradually longer intervals (i.e. the echo will become slower).
- If you lower the value below 100%, the echoing notes will become gradually faster, like the sound of a bouncing ball.

Length

This 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.

⇨ The length can also be affected by the Length Decay parameter.

Length Decay

This parameter lets you adjust how the length of the echoed notes should change with each successive repeat. The higher the setting (25–100), the longer the echoed notes will be, compared to their original notes.

About ticks and note values

The timing and position-related parameters (Delay, Length and Beat Align) can all be set in ticks (or PPQ which denotes the same thing here). There are 480 ticks to each quarter note. While the parameters allow you to step between the rhythmically relevant values (displayed as note values), the following table can also be of help, showing you the most common note values and their corresponding number of ticks:

Note Value	Ticks
1/32 note	60
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.

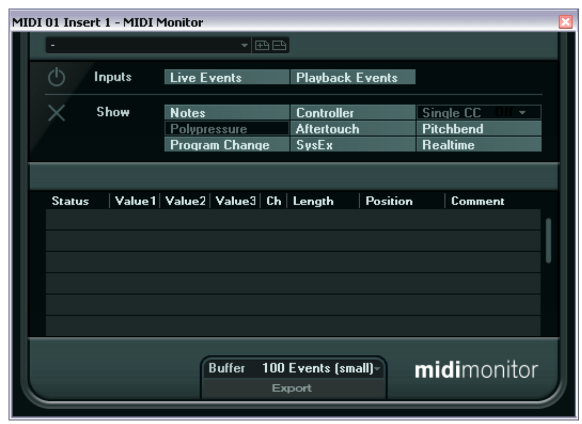
The MIDI Modifiers effect also includes an additional function that is not available among the track parameters:

Scale Transpose

This 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 turn Scale Transpose off, select “No Scale” from the Scale pop-up menu.

MIDI Monitor



The MIDI Monitor is used to monitor 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 being 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, e.g. notes or program change events. If you choose the Controller option you can also define which type of controller to monitor.

Data table

In the table in the lower section of the window, you will see detailed information about the monitored MIDI events.

Buffer pop-up menu

In the Buffer pop-up menu you can set the buffer size to 100, 1000 or 10000 events. This is the maximum number of events that is kept in the list of monitored events. Once this list is full, the oldest entries will be deleted when new events are received.

⇨ The larger the buffer, the more processing resources are required. To avoid a negative impact on your system's performance, make sure to use the smallest possible buffer size.

Export function

Click the Export button to export the monitoring data as a simple text file.

Record events button

Use this button to the left of the Inputs section to start or stop the monitoring of MIDI events.

Clear list button

The Clear List button to the left of the Show section allows you to clear the table of recorded MIDI events.

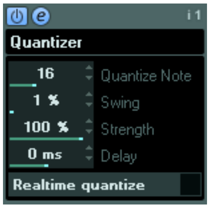
Note to CC

This effect will generate 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 played are used to control something else. For example, if Main Volume (CC 7) is selected, notes with low velocity will lower the volume in the MIDI instrument, while notes with a high velocity will raise the volume.

⚠ Note that 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 (playing one note at a time).

Quantizer



Quantizing is a function that changes the timing of notes by moving them towards a “quantize grid”. This grid may consist of e.g. straight sixteenth notes (in which case the notes would all get perfect sixteenth note timing), but could also be more loosely related to straight note value positions (applying a “swing feel” to the timing, etc.).

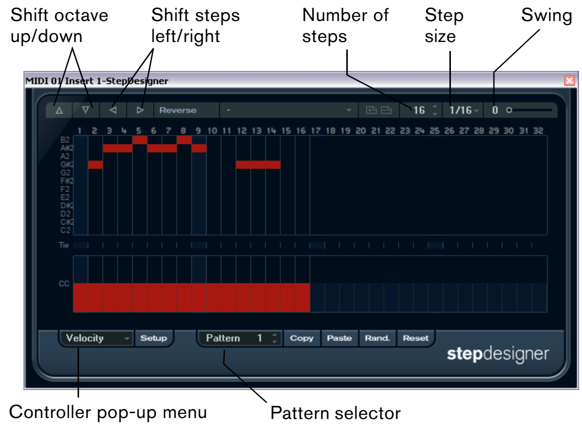
⇨ The main Quantize function in Nuendo is described in the Operation Manual.

While the Quantize function on the MIDI menu applies the timing change to the actual notes on a track, the Quantizer effect allows you to apply quantizing “on the fly”, changing the timing of the notes in real time. This makes it easier to try out different settings when creating grooves and rhythms. Note however, that the main Quantize function contains settings and features that are not available in the Quantizer.

The Quantizer has the following parameters:

Parameter	Description
Quantize Note	This 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	This 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	This determines how close the notes should be moved to the quantize grid. When set to 100%, all notes will be forced to the closest grid position; lowering the setting will gradually loosen the timing.
Delay	This delays (positive values) or advances (negative values) the notes in milliseconds. Unlike the Delay setting in the Track Parameters, this delay can be automated.
Realtime quantize	During live mode this option can be used to change the timing of the notes played so that they fit the quantize grid.

StepDesigner



The StepDesigner is a MIDI pattern sequencer that sends out MIDI notes and additional controller data according to the pattern you set up. It does not make use of the incoming MIDI, other than automation data (such as recorded pattern changes).

Creating a basic pattern

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.

In other words, this setting determines how long each step is. For example, if this is set to “1/16” each step will be a sixteenth note.

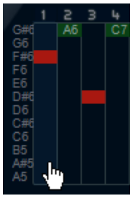
3. Specify the number of steps in the pattern with the “Number of steps” setting.

As you can see in the note display, the maximum number of steps is 32. For example, setting “Step size” to 16 and “Number of steps” to 32 would create a two 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 the StepDesigner will only play 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. Note that each step can contain one note only – the StepDesigner is monophonic.

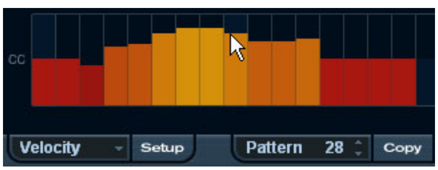


Click and drag to view other octaves.

- To remove a note from the pattern, click on it again.

5. On the Controller pop-up menu, select Velocity.
This pop-up menu determines what is shown in the lower controller display.

6. Adjust the velocity of the notes by dragging the velocity bars in the controller display.



7. To make notes shorter, select "Gate" on the Controller pop-up menu and lower the bars in the controller display.
When a bar is set to its maximum value (fully up), the corresponding note will be the full length of the step (as set with the Step size parameter).

8. 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.
When two notes are tied, the second note will not be triggered – the previous note is lengthened instead. Also, the tied (second) note will automatically get the same pitch as the first note. You can add more notes and tie them in the same way, creating longer notes.

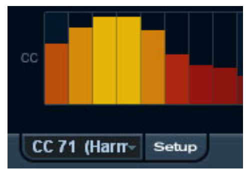
9. If you now start playback in Nuendo, the pattern will play as well, sending out MIDI notes on the track's MIDI output and channel (or, if you have activated the StepDesigner as a send effect, on the MIDI output and channel selected for the send in the Inspector).

Adding controller curves

The Controller pop-up menu has two more items: two controller types.

- You can select which two controller types (filter cutoff, resonance, volume, etc.) should be available on the pop-up menu by clicking the Setup button and selecting controllers from the lists that appears.
This selection is global, i.e. it applies to all patterns.

- To insert controller information in a pattern, select the desired controller from the pop-up menu and click in the controller display to draw events.
The MIDI controller events will be sent out during playback along with the notes.



⇨ If you drag a controller event bar all the way down, no controller value is sent out on that step.

Other pattern functions

The following functions make it easier to edit, manipulate and manage patterns:

Function	Description
Shift Octave up/down	These buttons allow you to shift 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	Allows 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 setting controller values to default.
Randomize	Generates a completely random pattern – useful for experimenting.
Swing	The Swing parameter allows you to offset 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	Handling of presets is described in the chapter "MIDI realtime settings" in the Operation Manual. Note that a stored Preset contains all 200 patterns in the StepDesigner.

Automating pattern changes

You can create up to 200 different patterns in each StepDesigner – just select a new pattern and add notes and controllers as described above.

Typically, you want the pattern selection to change during the project. You can accomplish this by automating the Pattern selector, either in real time by activating the Write automation and switching patterns during playback or by drawing in the automation track for the StepDesigner's MIDI track. Note that you can also press a key on your MIDI keyboard to change patterns. For this, you have to set up the 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. If you want, you can record these pattern changes as note events on a MIDI track.

Proceed as follows:

1. Select the desired MIDI track or create a new one and activate the StepDesigner as an insert effect.

2. Set up several patterns as described above.

3. Press the Record button and press the desired keys on your keyboard to select the corresponding patterns.
The pattern changes will be recorded on the MIDI track.

4. Stop recording and play back the MIDI track.
You will now hear the recorded pattern changes.

⇨ This will only work for the first 92 patterns.

Track Control

The Track Control effect contains three ready-made 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, the Track Controls effect allows you to adjust sounds and effects in your instrument from within Nuendo.

Selecting a control panel

At the top of the Track Controls effect window you will find a pop-up menu. This is where you select which of the available control panels to use:

Control panel	Description
GS 1	Effect sends and various sound control parameters for use with instruments compatible with the Roland GS standard.
XG 1	Effect Sends and various sound control parameters for use with instruments compatible with the Yamaha XG standard.
XG 2	Global settings (affecting all channels) for instruments compatible with the Yamaha XG standard.

About the Reset and Off buttons

Regardless of the selected mode, you will find two buttons labeled “Off” and “Reset” at the top of the control panel:

- Clicking the Off button will set all controls to their lowest value, without sending out any MIDI messages.
- Clicking the Reset button will set all parameters to their default values, and send out the corresponding MIDI messages.

For most parameters, the default values will be zero or “no adjustment”, but there are exceptions to this. For example, the default “Send 1” setting is 64.

GS 1

The following controls are available when the GS 1 Controls mode is selected:

Control	Description
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 gives a slower attack. Middle position (64) means no adjustment is made.
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	Adjusts the filter resonance.
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. The default value for this parameter is zero.
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 (just as you normally do with a modulation wheel on a MIDI keyboard).

XG 1

The following controls are available when the XG 1 mode is selected:

Control	Description
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 gives a slower attack. Middle position means no adjustment is made.
Release	Adjusts the release time of the sound. Lowering this value shortens the release, while raising it makes the release time longer. Middle position means no adjustment is made.
Harm.Cont	Adjusts the harmonic content of the sound.
Bright	Adjusts the brightness of the sound.
CutOff	Adjusts the filter cutoff frequency.
Resonance	Adjusts the filter resonance.

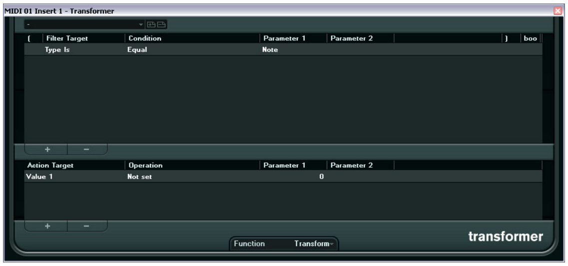
XG 2

In this mode, the parameters affect global settings in the instrument(s). Changing one of these settings for a track will in fact affect all MIDI instruments connected to the same MIDI output, regardless of the MIDI channel setting of the track. Therefore, to avoid confusion it might be a good idea to create an empty track and use this only for these global settings.

The following controls are available:

Control	Description
Eff. 1	This allows you to select which type of reverb effect should be used: No effect (the reverb turned off), Hall 1–2, Room 1–3, Stage 1–2 or Plate.
Eff. 2	This allows you to select which type of chorus effect should be used: No effect (the chorus turned off), Chorus 1–3, Celeste 1–3 or Flanger 1–2.
Eff. 3	This allows you to select one of a large number of “variation” effect types. Selecting “No Effect” is the same as turning off the variation effect.
Reset	Sends an XG reset message.
MastVol	This is used to control the Master Volume of an instrument. Normally you should leave this in its highest position and set the volumes individually for each channel (with the volume faders in the Nuendo mixer or in the Inspector).

Transformer



The 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 corresponding chapter in the Operation Manual. As the parameters and functions are almost identical, the descriptions for the Logical Editor also apply to the Transformer. Where there are differences between the two, this is clearly stated.

3

MixConvert Appendix

Available conversions

The following tables list all combinations when MixConvert is used. Each column is an output configuration and each row is an input configuration. When MixConvert is used as an insert effect, only downmix is possible. In this case, the number of outputs can be less than or equal to the number of inputs.

- D = Direct connection (1 to 1)
- M = MixConvert is used
- P = Standard Panner is used (Stereo Dual Panner/Stereo Combined Panner/Stereo Balance Panner)
- S = SurroundPanner is used
- - = Direct connection is used (trying to match the speaker configuration, for example L-> L or C->C)

Output Config. / Input Config.	Mono	Stereo	LRS	LRS +Lfe	LRC	LRC +Lfe	LRCS	LCRS +Lfe	Quadro	Quadro +Lfe	5.0	5.1	6.0 Cine	6.0 Music
Mono	**D**	P	S	S	S	S	S	S	S	S	S	S	S	S
Stereo	P	P	S	S	S	S	S	S	S	S	S	S	S	S
LRS	M	M	**D**	M	M	M	M	M	M	M	M	M	M	M
LRS+Lfe	M	M	M	**D**	M	M	M	M	M	M	M	M	M	M
LRC	M	M	M	M	**D**	M	M	M	M	M	M	M	M	M
LRC+Lfe	M	M	M	M	M	**D**	M	M	M	M	M	M	M	M
LRCS	M	M	M	M	M	M	**D**	M	M	M	M	M	M	M
LCRS+Lfe	M	M	M	M	M	M	M	**D**	M	M	M	M	M	M
Quadro	M	M	M	M	M	M	M	M	**D**	M	M	M	M	M
Quadro+Lfe	M	M	M	M	M	M	M	M	M	**D**	M	M	M	M
5.0	M	M	M	M	M	M	M	M	M	M	**D**	M	M	M
5.1	M	M	M	M	M	M	M	M	M	M	M	**D**	M	M
6.0 Cine	M	M	M	M	M	M	M	M	M	M	M	M	**D**	M
6.0 Music	M	M	M	M	M	M	M	M	M	M	M	M	M	**D**
6.1 Cine	M	M	M	M	M	M	M	M	M	M	M	M	M	M
6.1 Music	M	M	M	M	M	M	M	M	M	M	M	M	M	M
7.0 Cine	M	M	M	M	M	M	M	M	M	M	M	M	M	M
7.0 Music	M	M	M	M	M	M	M	M	M	M	M	M	M	M
7.1 Cine	M	M	M	M	M	M	M	M	M	M	M	M	M	M
7.1 Music	M	M	M	M	M	M	M	M	M	M	M	M	M	M
8.0 Cine	M	M	M	M	M	M	M	M	M	M	M	M	M	M
8.0 Music	M	M	M	M	M	M	M	M	M	M	M	M	M	M
8.1 Cine	-	-	-	-	-	-	-	-	-	-	-	-	-	-
8.1 Music	-	-	-	-	-	-	-	-	-	-	-	-	-	-
10.2	-	-	-	-	-	-	-	-	-	-	-	-	-	-

Output Config. / Input Config.	6.1 Cine	6.1 Music	7.0 Cine	7.0 Music	7.1 Cine	7.1 Music	8.0 Cine	8.0 Music	8.1 Cine	8.1 Music	10.2
Mono	S	S	S	S	S	S	S	S	S	S	S
Stereo	S	S	S	S	S	S	S	S	S	S	S
LRS	M	M	M	M	M	M	M	M	-	-	-
LRS+Lfe	M	M	M	M	M	M	M	M	-	-	-
LRC	M	M	M	M	M	M	M	M	-	-	-
LRC+Lfe	M	M	M	M	M	M	M	M	-	-	-
LRCS	M	M	M	M	M	M	M	M	-	-	-
LCRS+Lfe	M	M	M	M	M	M	M	M	-	-	-
Quadro	M	M	M	M	M	M	M	M	-	-	-
Quadro+Lfe	M	M	M	M	M	M	M	M	-	-	-
5.0	M	M	M	M	M	M	M	M	-	-	-
5.1	M	M	M	M	M	M	M	M	-	-	-
6.0 Cine	M	M	M	M	M	M	M	M	-	-	-
6.0 Music	M	M	M	M	M	M	M	M	-	-	-
6.1 Cine	**D**	M	M	M	M	M	M	M	-	-	-
6.1 Music	M	**D**	M	M	M	M	M	M	-	-	-
7.0 Cine	M	M	**D**	M	M	M	M	M	-	-	-
7.0 Music	M	M	M	**D**	M	M	M	M	-	-	-
7.1 Cine	M	M	M	M	**D**	M	M	M	-	-	-
7.1 Music	M	M	M	M	M	**D**	M	M	-	-	-
8.0 Cine	M	M	M	M	M	M	**D**	M	-	-	-
8.0 Music	M	M	M	M	M	M	M	**D**	-	-	-
8.1 Cine	-	-	-	-	-	-	-	-	**D**	-	-
8.1 Music	-	-	-	-	-	-	-	-	-	**D**	-
10.2	-	-	-	-	-	-	-	-	-	-	**D**

Index