Plug-in Reference

NUENDO 4

Advanced Audio and Post Production System

steinberg

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Table of Contents

2

The included effect plug-ins

Introduction

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

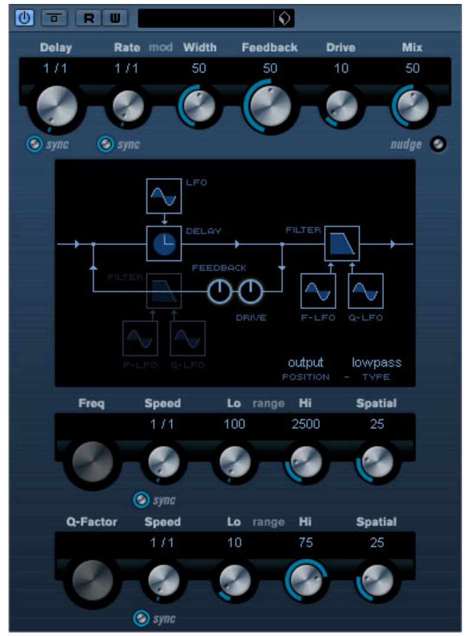
In Nuendo, the plug-in effects are arranged in a number of different categories. This chapter is arranged in the same fashion, with the plug-ins listed in separate sections for each effect category.

⇨ 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 parameters are as follows:

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.
Tempo sync Delay on/off	The button below the Delay knob turns tempo sync for the delay parameter on or off. If set to off, the delay time can be set freely with the Delay knob.
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 with the Rate knob.

Parameter	Description
Tempo sync Rate on/off	The button below the Rate knob turns tempo sync for the rate parameter on or off. If set to off, the rate can be set freely with the Rate knob.
Width	This 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	This sets the number of repeats for the delay.
Drive	This parameter adds distortion to the feeback 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, this should be set to maximum (100%) as you can control the dry/effect balance with the send.
Nudge	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	You can click on the Filter sections displayed in the graphic in the center of the plug-in to place the Filter section either before or after the Drive and Feedback parameters in the signal path.
Output/Loop	The Filter can either be placed in the feedback loop of the delay or in its output path (see above).
Filter type	This toggle button allows you to select a filter type. Lowpass/bandpass/hipass filter types are available.
Freq	This sets the cutoff frequency for the filter. This is available only, if filter frequency LFO tempo sync is deactivated and the Speed parameter (see below) is set to "0".
Speed	This sets the speed of the filter frequency LFO modulation. If tempo sync is activated 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 rate can be set freely with the Speed knob.
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 instead controlled by the Freq parameter.
Spatial	This 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	This controls the resonance of the filter. This is available only, if filter resonance LFO tempo sync is deactivated and the Speed parameter (see below) is set to "0". If tempo sync is on, the resonance is controlled by the Speed and Range parameters.
Speed	This sets the speed of the filter resonance LFO modulation. If tempo sync is activated, 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 rate can be set freely with the Speed knob.

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	This 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 delay can also be controlled from another signal source via the Side-Chain input.

The parameters are as follows:

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.
Tempo sync on/off	The button below the Delay Time knob is used to turn tempo sync on or off. If set to off the delay time can be set freely with the Delay Time knob, without sync to tempo.
Feedback	This 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, this should be set to maximum as you can control the dry/effect balance with the send.
Side-Chain on/off	When this is activated, the delay can be controlled by a signal routed to the Side-Chain input. When the side-chain signal exceeds the threshhold the delay repeats are silenced. When the signal drops below the threshold the delay repeats reappear. For a description on 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 parameters are as follows:

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.
Tempo sync on/off	The button below the Delay Time knob is used to turn tempo sync on or off. If set to off the delay time can be set freely with the Delay Time knob, without sync to tempo.
Feedback	This 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	This parameter 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, this should be set to maximum as you can control the dry/effect balance with the send.
Side-Chain on/off	When this is activated, the delay can be controlled by a signal routed to the Side-Chain input. When the side-chain signal exceeds the threshhold, the delay repeats are silenced. When the signal drops below the threshold, the delay repeats reappear. For a description on how to set up Side-Chain routing, see the chapter “Audio effects” in the Operation Manual.

StereoDelay

This effect features two independent delay lines which can either use tempo-based or freely specified delay time settings.

The parameters are as follows:

Parameter	Description
Delay 1	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.
Delay 2	As above.
Tempo sync on/off	The buttons below each respective Delay knob are used to turn tempo sync on or off for the respective delay. If set to off, the delay time can be set freely with the Delay Time knobs.
Feedback 1 & 2	This sets the number of repeats for each 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.
Pan1 & 2	This sets the stereo position for each delay.
Mix	Sets the level balance between the dry signal and the effect. If StereoDelay is used as a send effect, this should be set to maximum (100%) as you can control the dry/effect balance with the send.
Side-Chain on/off	When this is activated, the delay can be controlled by a signal routed to the Side-Chain input. When the side-chain signal exceeds the threshhold, the delay repeats are silenced. When the signal drops below the threshold, the delay repeats reappear. For a description on 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 parameters are as follows:

Parameter	Description
Drive	Governs 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	Use this to boost or damp the higher frequencies.
Volume	This controls the overall output level.
Amplifier	This allows you to select between various amplifier models. Click on the currently selected amplifier name to open a pop-up with all the available amplifier models. This section can be bypassed by selecting “No Amp".
Cabinet	Various speaker cabinet models. Click on the currently selected cabinet name to open a pop-up with all the available amplifier models.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 parameters are as follows:

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	This 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 parameters are as follows:

Parameter	Description
Drive	Increases the distortion amount.
Feedback	This parameter 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 parameters are as follows:

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	This allows you to adjust the amount of the second harmonic in the processed signal.
Third	This 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 available parameters work as follows:

Parameter	Description
Threshold (-60 to 0dB)	This setting 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)	Ratio determines 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 1 dB.
Soft Knee (On/Off)	If this is off, signals above the threshold will be compressed instantly according to the set ratio (hard knee). When Soft Knee is activated, the onset of compression will be more gradual, producing a less drastic result.
Make-up (0–24dB 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 instead automatically adjusted for gain loss.
Attack (0.1–100ms)	This 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) will pass through unprocessed.
Hold (0–2000ms)	Sets the time the applied compression will affect the signal after exceeding the Threshold.
Release (10–1000ms 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, Compressor will automatically find an optimal release setting that varies depending on the audio material.
Analysis (0–100) (Pure Peak to Pure RMS)	This parameter determines whether the input signal is analysed 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 mode (On/Off)	When activated, Live mode disengages the "look ahead" feature of the Compressor. Look ahead does produce more accurate processing but will add 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.
Side-Chain (On/Off)	When this is activated, the compression can be controlled by a signal routed to the Side-Chain input. When the side-chain signal exceeds the threshhold, the compression is triggered. For a description on how to set up Side-Chain routing, see the chapter "Audio effects" in the Operation Manual.

SPL 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. Conventional compression and/or equalizing will not easily solve this problem, but a de-esser can.

The SPL DeEsser has the following parameters:

Parameter	Description
S-Reduction	Controls the intensity of the de-essing effect. We recommend that you start with a value between 4 and 7.
Level display	Indicates the dB value by which the level of the sibilant or s-frequency is reduced. The display shows values between 0dB (no reduction) and minus 20dB (the s-frequency level is lowered by 20dB). Each segment in the display represents a level reduction of 2dB.
Auto Threshold	See separate description below.
Male/Female	This sets the s-frequency and sibilant recognition to the characteristic frequency ranges of the female or male voice. The center frequency of the bandwidth at which the SPL DeEsser operates is located in the 7kHz range for the female voice and in the 6kHz range for the male voice.

About the Auto Threshold function

Conventional de-essing devices all have a threshold parameter. This is used to set a threshold for the incoming signal level, above which the device starts to process the signal. The SPL DeEsser however has been designed for utmost ease-of-use. With Auto Threshold on (the button lights up) it automatically and constantly readjusts the threshold to achieve an optimum result. If you still wish to determine for yourself at which signal level the SPL DeEsser should start to process the signal, deactivate the Auto Threshold button. The SPL DeEsser will then use a fixed threshold.

When recording a voice, usually the de-esser's position in the signal chain is located after the microphone pre-amp and before a compressor/limiter. This is useful, as it keeps the compressor/limiter from unnecessarily limiting the overall signal dynamics by reacting to excessive sibilants and s-frequencies.

The Auto Threshold function keeps the processing on a constant level. The input threshold value is automatically and constantly adjusted to the audio input level. Even level differences of say 20dB do not have a negative impact on the result of the processing. The input levels may vary, but processing remains constant.

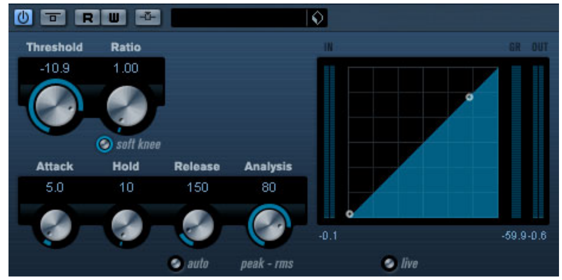
EnvelopeShaper

EnvelopeShaper can be used to cut or boost the gain of the Attack and Release phase of the audio material. You can either use the knobs or drag the breakpoints in the graphic 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 (-20–20dB)	Changes the gain of the Attack phase of the signal.
Length (5–200ms)	This determines the length of the Attack phase.
Release (-20–20dB)	Changes the gain of the Release phase of the signal.
Output (-24–12dB)	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 graphic display to change the Threshold and the Ratio parameter values.

The following parameters are available:

Parameter	Description
Threshold (-60–0dB)	This setting determines the level where expansion "kicks in". Signal levels below the set threshold are affected, but signal levels above are not processed.
Ratio (1:1–8:1)	Ratio determines the amount of gain boost applied to signals below the set threshold.
Soft Knee (On/Off)	If this is off, signals below the threshold will be expanded instantly according to the set ratio ("hard knee"). When Soft Knee is activated, the onset of expansion will be more gradual, producing a less drastic result.
Attack (0.1–100ms)	This determines how fast Expander will respond to signals below the set threshold. If the attack time is long, more of the early part of the signal (attack) will pass through unprocessed.
Hold (0–2000ms)	Sets the time the applied expansion will affect the signal below the Threshold.
Release (10–1000ms 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.
Analysis (0–100) (Pure Peak to Pure RMS)	This parameter determines whether the input signal is analysed 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.

Parameter	Description
Live mode (On/Off)	When activated, Live mode disengages the look ahead feature of Expander. Look ahead does produce more accurate processing but will add a certain amount of latency as a trade-off. When Live mode is activated, there is no latency.
Side-Chain (On/Off)	When this is activated, the expansion can be controlled by a signal routed to the Side-Chain input. When the side-chain signal exceeds the threshhold the expansion is triggered. For a description on 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 certain set threshold level. As soon as the signal level exceeds the set threshold, the gate opens to let the signal through.

The available parameters are as follows:

Parameter	Description
Threshold (-60–0 dB)	This setting 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	This 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	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 (Off/On)	This button (below the Center knob) activates the filter. The input signal can then be shaped according to set Center and Q-Factor parameters which may be useful in tailoring how the Gate operates.
Center (50 Hz–20000 Hz)	Sets the center frequency of the filter.
Q-Factor (0.01–10000)	Sets the Resonance of the filter.
Monitor (Off/On)	Allows you to monitor the filtered signal.
Attack (0.1–1000 ms)	This parameter sets the time it takes for the gate to open after being triggered. If the Live button (see below) is deactivated, it will ensure 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 (0–2000 ms)	This determines how long the gate stays open after the signal drops below the threshold level.
Release (10–1000 ms or “Auto”)	This parameter 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 program material.
Analysis (0–100) (Pure Peak to Pure RMS	This parameter determines whether the input signal is analysed 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 mode (On/Off)	When activated, Live mode disengages the “look ahead” feature of the Gate. Look ahead does produce more accurate processing but will add 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.

Limiter

Limiter is designed to ensure that the output level never exceeds a certain 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 available parameters are the following:

Parameter	Description
Input (-24–+24 dB)	Allows you to adjust the input gain.
Output (-24–+6 dB)	This setting determines the maximum output level.
Release (0.1–1000 ms or Auto mode)	This parameter 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 can be 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 tubelike distortion to the signal.

The available parameters are the following:

Parameter	Description
Output (-24–+6 dB)	This setting determines the maximum output level. Should normally be set to 0 (to avoid clipping).
Optimize (0–100)	This setting determines the loudness of the signal.
Soft Clip (On/Off)	Soft Clipper starts limiting (or clipping) the signal "softly", at the same time generating harmonics which add a warm, tubelike characteristic to the audio material.

MIDI Gate

Gating, in its fundamental form, silences audio signals below a certain set threshold level. That means, 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 a Gate effect that is not triggered by threshold levels, but instead by MIDI notes. Hence it needs both audio and MIDI data to function.

Setting up

MIDI Gate requires both an audio signal and a MIDI input to function.

To set it up, proceed as follows:

1. Select the audio to be affected by the 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 the 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.
This can be an empty MIDI track, or a MIDI track containing data, it doesn't matter. However, if you wish to play the MIDI Gate in real-time – as opposed to having a recorded part playing it – 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.

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

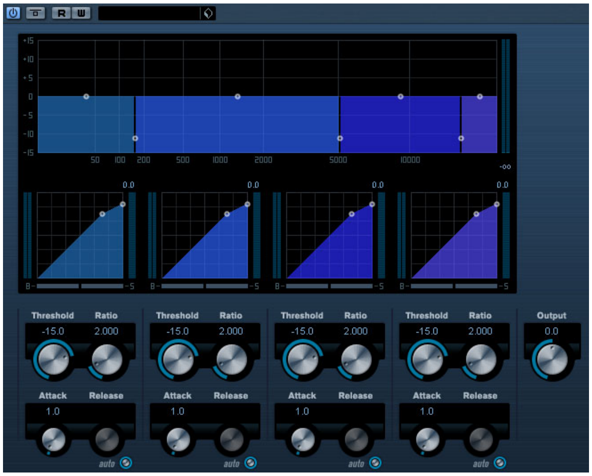
Make sure the MIDI track is selected and start playback.

5. Now 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	This is used for determining how long it should take 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	This determines how long it takes for the Gate to close (in addition to the value set with the Hold parameter).
Note To Attack	The value you specify here determines to which extent the velocity values of the MIDI notes should affect the Attack. The higher the value, the more the Attack time will increase with high note velocities. Negative values will 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	The value you specify here determines to which extent the velocity values of the MIDI notes should affect the Release. The higher the value, the more the Release time will increase. If you do not wish to use this parameter, set it to the 0 position.
Velocity To VCA	This controls to which extent the velocity values of the MIDI notes determine the output volume. A value of 127 means that the volume is controlled entirely by the velocity values, while a value of 0 means that velocities will have no effect on the volume.
Hold Mode	Use this switch to set the Hold Mode. In Note-On mode, the Gate will only remain 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 on the other hand, the Gate will remain open for as long as the MIDI note plays, and then apply the Hold and Release parameters.

MultibandCompressor



The MultibandCompressor allows a signal to be split in up to 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 +/- 15dB 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. The compressor parameters for each of the four bands are as follows:

Parameter	Description
Threshold (-60–0dB)	This setting determines the level where Compressor “kicks in”. Signal levels above the set threshold are affected, but signal levels below are not processed.
Ratio (1000–8000) (1:1 to 8:1)	Ratio determines the amount of gain reduction applied to signals over the set threshold. A ratio of 3000 (3:1) means that for every 3dB the input level increases, the output level will increase by only 1dB.
Attack (0.1–100ms)	This determines how fast the compressor will respond 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 (10–1000ms or “Auto”)	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 dial

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

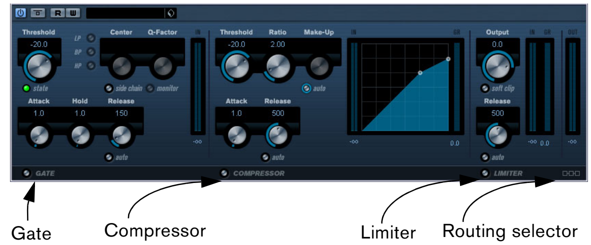
VintageCompressor

This is modelled after vintage type compressors. Compressor features separate controls for input gain, attack, release and output gain parameters. 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 gain (-24–48 dB)	This setting, together with the Output gain parameter determines the compression amount. The higher the Input gain setting, and the lower the Output gain setting, the more compression is applied.
Output gain (-48–24 dB)	Sets the output gain.
Attack (0.1–100 ms)	This determines how fast Compressor will respond. If the attack time is long, more of the early part of the signal (attack) will pass through unprocessed.
Punch (Off/On)	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 (10–1000 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.
Side-Chain (On/Off)	When this is activated, the compression can be controlled by a signal routed to the Side-Chain input. When the side-chain signal exceeds the threshhold the compression is triggered. For a description on 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 certain 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 available parameters are as follows:

Parameter	Description
Threshold (-60–0 dB)	This setting 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	This 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 (On/Off)	This button activates the internal side-chain filter. This lets you filter out parts of the signal that might otherwise trigger the gate in places you don’t want it to, or to boost frequencies you wish to accentuate, allowing for more control over the gate function.
LP (Lowpass), BP (Bandpass), HP (Highpass)	These buttons set the basic filter mode.
Center (50–22000 Hz)	This sets the center frequency of the filter.
Q-Factor (0.001–10000)	This sets the resonance or width of the filter.

Parameter	Description
Monitor (Off/On)	Allows you to monitor the filtered signal.
Attack (0.1–100ms)	This parameter sets the time it takes for the gate to open after being triggered.
Hold (0–2000ms)	This determines how long the gate stays open after the signal drops below the threshold level.
Release (10–1000ms or "Auto")	This parameter 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 program material.

The Compressor section

Compressor reduces the dynamic range of the audio, making softer sounds louder or louder sounds softer, or both. Compressor functions like a standard compressor with separate controls for threshold, ratio, attack, release and make-up gain parameters. Compressor features a separate display that graphically illustrates the compressor curve shaped according to the Threshold, Ratio and MakeUp Gain parameter settings. Compressor also features a Gain Reduction meter that shows the amount of gain reduction in dB, and a program dependent Auto feature for the Release parameter.

The available parameters work as follows:

Parameter	Description
Threshold (-60–0dB)	This setting determines the level where Compressor "kicks in". Signal levels above the set threshold are affected, but signal levels below are not processed.
Ratio (1:1–8:1)	Ratio determines 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.
Make-Up (0–24dB)	This parameter is used to compensate for output gain loss, caused by compression. When Auto is on, gain loss will be compensated automatically.
Attack (0.1–100ms)	This 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) will pass through unprocessed.
Release (10–1000ms or "Auto")	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.
Graphic display	Use the graphic display to graphically set the Threshold or the Ratio value.

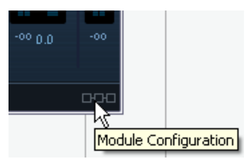
The Limiter section

Limiter is designed to ensure that the output level never exceeds a certain set output level, 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. Limiter adjusts and optimizes these parameters automatically, according to the audio material. You can also adjust the Release parameter manually.

The available parameters are the following:

Parameter	Description
Output (-24–+6dB)	This setting determines the maximum output level. Signal levels above the set threshold are affected, but signal levels below are left unaffected.
Soft Clip (On/Off)	Soft Clipper acts differently compared to the limiter. When the signal level exceeds -6dB, SoftClip starts limiting (or clipping) the signal "softly", at the same time generating harmonics which add a warm, tubelike characteristic to the audio material.
Release (10–1000ms or "Auto")	This parameter 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, Limiter will automatically find an optimal release setting that varies depending on the audio material.

The Module Configuration button



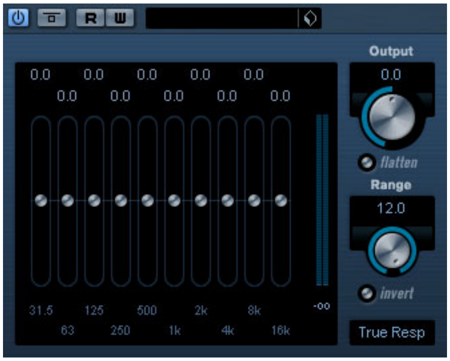
In the bottom right corner of the plug-in panel you will find a button with which you can set the signal flow order for the three processors. Changing the order of the processors can produce different results, and the available options allow you to quickly compare what works best for a given situation. 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 respective 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:

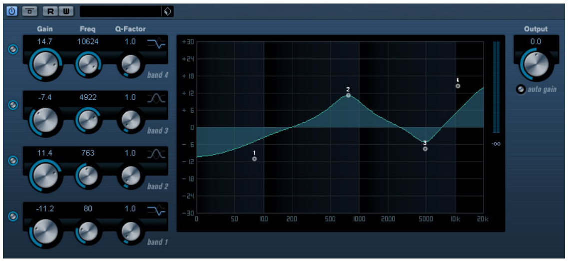
Parameter	Description
Output	This controls the overall gain of the equalizer.
Range	This allows you to relatively adjust how much a set curve cuts or boosts the signal. If the Range parameter is turned fully clockwise, +/- 12dB is the available range.
Flatten button	Resets all the frequency bands to 0dB.
Invert range	This will invert the current response curve.
Mode	The filter mode set here determines how the various frequency band contrrols interact to create the response curve. See also below.

About the filter modes

On the pop-up 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, which is sometimes desirable. As always, let your ears be the judge! 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 – parallell filters where the resonance depends on the amount of gain. Musical sounding.
- Constant Q u – parallell filters where the resonance of the first and last bands depends on the sample rate (u=unsymmetric).
- Constant Q s – parallell 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 midrange bands. The low and high bands can act as either shelving filters (three types) or as a Peak (bandpass) or Cut (lowpass/highpass) filter.

Making settings

1. Click the corresponding On button to the left of 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.

The following parameters are available:

Parameter	Description
Low Freq (20 to 2000Hz)	This sets the frequency of the Low band.
Low Gain (-20 to +24dB)	This sets the amount of cut/boost for the Low band.
Low Q-Factor	This parameter controls the width or resonance of the Low band.
Low Filter mode	For the Low band, you can select between three types of shelving filters or Peak (bandpass) or Cut (lowpass/highpass) filters. The Gain parameter will be fixed if Cut mode is selected. -Shelf I adds resonance in the opposite gain direction slightly over 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.
Mid 1 Freq (20 to 20000Hz)	This sets the center frequency of the Mid 1 band.
Mid 1 Gain (+/- 24dB)	This sets the amount of cut/boost for the Mid 1 band.
Mid 1 Q-Factor (0.5 to 10)	This sets the width of the Mid 1 band. The higher this value, the “narrower” the bandwidth.
Mid 2 Freq (20 to 20000Hz)	This sets the center frequency of the Mid 2 band.
Mid 2 Gain (-20 to +24dB)	This sets the amount of cut/boost for the Mid 2 band.
Mid 2 Q-Factor (0.5 to 10)	This sets the width of the Mid 2 band. The higher this value, the “narrower” the bandwidth.
High Freq (200 to 20000Hz)	This sets the frequency of the High band.
High Gain (-20 to +24dB)	This sets the amount of cut/boost for the High band.
High Q-Factor	This parameter controls the width or resonance of the High band.
High Filter mode	For the High band, you can select between three types of shelving filters, and Peak or Cut filters. The Gain parameter will be fixed if Cut mode is selected. -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 +24dB)	This parameter allows you to adjust the overall output level.
Auto Gain	When this 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

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

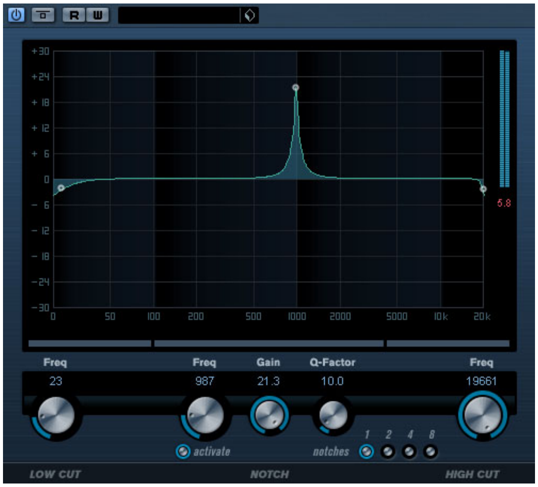
The following parameters are available:

Parameter	Description
Position	This parameter 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.

NuendoEQ2

The NuendoEQ2 plug-in is identical to the EQ section in the Channel Settings window. As a plug-in, NuendoEQ2 can be applied in different areas than the Channel EQ. For example, you could use it as an insert effect, to EQ the output of another effect plug-in, etc. See the Operation Manual chapter “The mixer” for a description of the EQ parameters.

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 graphic display, or by adjusting one of the parameter 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	Displays the output level, giving you an indication of how the filtering affects the overall level of the edited event.
Lo Cut Freq (20–1000 Hz)	Use this low-cut filter to eliminate low-frequency noise. Filter is Off when the handle/knob is moved all the way to the left.
Lo Cut Preview	Use the Preview button (found between the Lo Cut Freq button and the graphic 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.
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 Q-Factor	Sets the width of the notch filter.
Notch filter Preview	Use the Preview button (found between the notch filter buttons and the graphic 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 (1, 2, 4, 8)	These buttons add one or more additional notch filters to filter out harmonics.
Hi Cut Freq (3–20 kHz)	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.
Hi Cut Preview	Use the Preview button found between the Hi Cut Freq button and the graphic 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 midrange 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.

Parameters

Parameter	Description
Low Freq (20–2000Hz)	This sets the frequency of the Low band.
Low Gain (-20 to +20dB)	This sets the amount of cut/boost for the Low band.
Low Cut	If this button is activated for the Low band, it will act as a Low Cut filter. The Gain parameter will be fixed.
Mid 1 Freq (20–20000Hz)	This sets the center frequency of the Mid 1 band.
Mid 1 Gain (+/- 20dB)	This sets the amount of cut/boost for the Mid 1 band.
Mid 1 Width (0.05–5.00 Octaves)	This sets the width of the Mid 1 band, in octaves. The lower this value, the “narrower” the bandwidth.
Mid 2 Freq (20–20000Hz)	This sets the center frequency of the Mid 2 band.
Mid 2 Gain (-20 to +20dB)	This sets the amount of cut/boost for the Mid 2 band.
Mid 2 Width (0.05–5.00 Octaves)	This sets the width of the Mid 2 band, in octaves. The lower this value, the “narrower” the bandwidth.
High Freq (200–20000Hz)	This sets the frequency of the High band.
High Gain (-20 to +20dB)	This sets the amount of cut/boost for the High band.
High Cut	If this button is activated for the High band, it will act as a High Cut filter. The Gain parameter will be fixed.
Output (-20 to +20dB)	This parameter 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 mode is activated, the curve will be applied to both channels. When channel independent curves have been set, the left/right channel curves will be colored green and red, respectively. The currently non-selected channel is shown with a dotted curve. If you activate Stereo mode after independent curves have been set, the currently active curve will be 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.

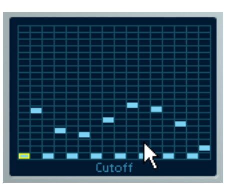
General operation

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 will be set to the pointer position.



Setting filter cutoff values in the grid window.

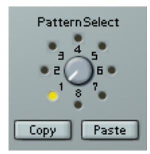
- The horizontal axis shows the pattern steps 1–16 from left to right, and the vertical axis determines the (relative) filter cutoff frequency and resonance setting.
The higher up on the vertical axis a step value is entered, the higher the relative filter cutoff frequency or filter resonance setting.

- 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 directly.

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 patterns are saved together in the 8 Pattern memories.

- To select new patterns you use the pattern selector.
New patterns are all set to the same step value by default.



Pattern Selector

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 memory location, which is useful for creating variations on a pattern.

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

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

StepFilter parameters

Parameter/ Value	Description
Base Cutoff	This sets the base filter cutoff frequency. Cutoff values set in the Cutoff grid window are values relative to the Base Cutoff value.
Base Resonance	This sets the base filter resonance. Resonance values set in the Resonance grid window are values 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	This slider selects between lowpass (LP), bandpass (BP) or highpass (HP) filter modes (from left to right respectively).
Sync 1/1 to 1/32 (Straight, Triplet or Dotted)	This sets the pattern beat resolution, i.e. what note values the pattern will play in relation to the tempo.
Output	Sets the overall volume.
Mix	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	This sets the center filter frequency.
Gain	Allows you to adjust the gain of the selected frequency range by up to 24 dB.
Width	This sets the resonance of the filter.
Mode	This sets the basic operational mode of the filter; Peak or Bandpass.

Tonic – Analog Modeling Filter

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.

The Tonic Analog Modeling Filter 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

Parameter	Description
Mode	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	Drive adds a soft, tube-like saturation to the sound. Like for an analog filter, the amount of saturation also depends on the input signal level.
Mix	Sets the balance between dry and effect signal.
Ch.	Choose between mono or stereo operation. When set to mono, the output signal of Tonic will be mono regardless of the input signal.

Env Mod

Parameter	Description
Mode	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 will 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 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, envelope level modulates the LFO speed. A rather stunning effect.

X/Y Pad

Parameter	Description
X Par	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	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 can control the x parameter, by moving it vertically, you can control the y parameter. You can also record controller movements as automation data.

LFO Mod

Parameter	Description
Mode	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 song tempo. For example: a rate of 4.00 steps per beat advances the step sequencer in 16th notes at a 4/4 time signature. A rate of 4.00 beats per step would advance the LFO at only one step per bar in a 4/4 time signature.
Smooth	The smooth parameter controls the smoothing of the LFO steps. This works like a glide effect applied to the filter cutoff.
Morph	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 will return to its original setting.
Steps	Sets the number of steps played in sequence. Deactivated steps are grayed out in the step window.
Preset	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 bandpass 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 is at 50.

The parameters are as follows:

Parameter	Description
Pedal	This controls the filter frequency sweep.
Freq Lo/Hi	Sets the frequency of the filter for the Lo and Hi Pedal positions.
Width Lo/Hi	Sets the width (resonance) of the filter for the Lo and Hi Pedal positions.
Gain Lo/Hi	Sets the gain of the filter for the Lo and Hi Pedal positions.
Slope	Specifies the slope of the filter; 6dB or 12dB.
Side-Chain On/Off	A signal routed to the Side-Chain input of the effect can control the Pedal parameter when this is activated. The louder the signal, the more the filter frequency (Pedal) is raised so the plug-in acts as an “auto-wha” effect.For a description on how to set up Side-Chain routing, see the chapter “Audio effects” in the Operation Manual.

MIDI control

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

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

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

Mastering – UV 22 HR

The UV22 HR 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 Manunal.

The following options can be set in the UV 22 HR control panel:

Option	Description
Hi	Try this first, it is the most “all-round” setting.
Low	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.
Bit Resolution	The UV22 HR supports dithering to multiple resolutions: 8, 16, 20 or 24 bits. You select the desired resolution by clicking the corresponding button.

⚠ Dither should always be applied post output bus fader.

Modulation plug-ins

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

AutoPan

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

The parameters are as follows:

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, without sync to tempo.
Tempo sync on/off	The button below the Rate knob is used to switch tempo sync on (the button lights up) or off.
Width	Sets the depth of the Autpan effect.
Shape	Sets the modulation waveform. Sine and Triangle waveforms are available.
Side-Chain On/Off	A signal routed to the Side-Chain input of the effect can control the Width parameter when this is activated. For a description on 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 33.

The parameters are as follows:

Parameter	Description
Tempo sync on/off	The button below the Rate knob is used to switch tempo sync on or off. The button is lit when tempo sync is on.
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, without sync to tempo.
Width	This determines the depth of the chorus effect. Higher settings produce a more pronounced effect.
Spatial	This 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, this should be set to maximum as you can control the dry/effect balance with the send.
Delay	This parameter affects the frequency range of the modulation sweep, by adjusting the initial delay time.
Shape	This changes the shape of the modulating waveform, altering the character of the chorus sweep. Sine and triangle waveforms are available.
Filter Lo/Hi	These parameters allow you to roll off low and high frequencies of the effect signal, respectively.
Side-Chain On/Off	When this is activated, the modulation can be controlled by a signal routed to the Side-Chain input. When the side-chain signal exceeds the threshhold the modulation will be controlled by the side-chain signal's envelope. For a description on 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 parameters are as follows:

Parameter	Description
Voices	This 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	This 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, this should be set to maximum 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	This 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	This controls the relative delay amount for each voice. A value of zero means no delay for that voice.
Master Detune	This parameter 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.
Humanize Delay knob	Humanize is turned on and off with the Static Delay button button below this knob. When activated the delay settings are subtly varied, for a richer effect. Values range from 0 to 100 (strongest delay variation). If deactivated, the set delay amount is static, and the knob is blacked out.
Humanize Detune knob	Humanize is turned on and off with the Static Detune button below this knob. When activated, the detune settings are subtly varied, for a richer effect. Values range from 0 to 100 (strongest detune variation). If deactivated, the set detune amount is static, and the knob is blacked out.
Master Delay	This parameter governs the overall depth of the delay for all voices. If this is set to zero, no delay takes place, regardless of the Delay slider settings.

Flanger

Flanger is a classic flanger effect with added stereo enhancement.

The parameters are as follows:

Parameter	Description
Tempo sync on/off	The button below the Rate knob is used to switch tempo sync on or off. The button is lit when tempo sync is on.
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, without sync to tempo.
Range Lo/Hi	This sets the frequency boundaries for the flanger sweep.
Feedback	This determines the character of the flanger effect. Higher settings produce a more “metallic” sounding sweep.
Spatial	This 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 the Flanger is used as a send effect, this should be set to maximum as you can control the dry/effect balance with the send.
Shape	This changes the shape of the modulating waveform, altering the character of the flanger sweep.
Delay	This parameter affects the frequency range of the modulation sweep, by adjusting the initial delay time.
Manual	If this is activated, the flanger sweep will be static, i.e. no modulation. You can instead change the sweep position manually by turning this knob.
Filter Lo/Hi	These parameters allow you to roll off low and high frequencies of the effect signal, respectively.
Side-Chain On/Off	When this is activated, the modulation can be controlled by a signal routed to the Side-Chain input. When the side-chain signal exceeds the threshhold the modulation will be controlled by the side-chain signal’s envelope. For a description on how to set up Side-Chain routing, see the chapter “Audio effects” in the Operation Manual.

Metalizer

The 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 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, the Metalizer will work as a static filter.
Mono button	When this is on, the output of the Metalizer will be in 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, without sync to tempo.
Tempo sync on/off	The button above the Speed knob is used to switch tempo sync on or off. The button is lit when tempo sync is on.
Output	Sets the overall volume.
Mix	Sets the level balance between the dry signal and the effect. If Metalizer is used as a send effect, this should be set to maximum 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 parameters are as follows:

Parameter	Description
Tempo sync on/off	The button below the Rate knob is used to switch tempo sync on or off. The button is lit when tempo sync is on.
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, without sync to tempo.
Width	The width of the modulation effect between higher and lower frequencies.
Feedback	This determines the character of the phaser effect. Higher settings produce a more pronounced effect.
Spatial	When using multi-channel audio, Spatial creates a 3-dimensional impression by delaying modulation in each channel.
Mix	Sets the level balance between the dry signal and the effect. If the Phaser is used as a send effect, this should be set to maximum as you can control the dry/effect balance with the send.
Manual	If this is activated, the phaser sweep will be static, i.e. no modulation. You can instead change the sweep position manually by turning this knob.
Filter Lo/Hi	These parameters allow you to roll off low and high frequencies of the effect signal, respectively.
Side-Chain On/Off	When this is activated, the modulation can be controlled by a signal routed to the Side-Chain input. When the side-chain signal exceeds the threshhold the modulation will be controlled by the side-chain signal's envelope. For a description on how to set up Side-Chain routing, see the chapter "Audio effects" in the Operation Manual.

Ringmodulator

The 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.

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

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 Wave	Selects the oscillator waveform; square, sine, saw or triangle.
Oscillator Range	Determines the frequency range of the oscillator in Hz.
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, with center position representing no modulation. Left of center, a loud input signal will slow down the LFO, whereas right of center a loud input signal will speed it up.
LFO Waveform	Selects the LFO waveform; square, sine, saw or triangle.
Invert Stereo	This inverts the LFO waveform for the right channel of the oscillator, which produces a wider stereo perspective for the modulation.
Envelope Generator (Attack and Decay dials)	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 sets 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	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	Sets the overall volume.
Mix	Adjusts the mix between dry and processed signal.

Rotary

The Rotary plug-in simulates the classic effect of a rotary speaker. A rotary speaker cabinet features variable speed rotating speakers to produce a swirling chorus effect, commonly used with organs. Rotary features all the parameters associated with the real thing.

The parameters are as follows:

Parameter	Description
Speed (Stop/Slow/Fast)	This controls the speed of the Rotary in three steps.
Mode	Selects whether the Slow/Fast setting is a switch or a variable control. When switch mode is selected and Pitch Bend is the controller, the speed will switch with an up or down flick of the bender. Other controllers switch at 64.
Speed Mod	Selects the Rotary speed from 0 (Stop) to 100 (Fast).
Overdrive	Applies a soft overdrive or distortion.
Crossover Freq.	Sets the crossover frequency (200–3000 Hz) between the low and high frequency loudspeakers.
Slow	Fine adjustment of the high rotor Slow speed.
Accel.	Fine adjustment of the high rotor acceleration time.

Parameter	Description
Fast	Fine adjustment of the high rotor Fast speed.
Amp Mod	High rotor amplitude modulation.
Freq Mod	High rotor frequency modulation.
Slow	Fine adjustment of the low rotor Slow speed.
Fast	Fine adjustment of the low rotor Fast speed.
Accel	Fine adjustment of the low rotor acceleration time.
Amp Mod.	Adjusts amplitude modulation depth.
Level	Adjusts overall bass level.
Phase	Adjusts the amount of phasing in the sound of the high rotor.
Angle	Sets the simulated microphone angle. 0 = mono, 180 = one mic on each side.
Distance	Sets the simulated microphone distance from the speaker in inches.
Output	Adjusts the overall output level.
Mix	Adjusts the mix between dry and processed signals.

Directing MIDI to the Rotary

For real-time MIDI control of the Speed parameter, MIDI must be directed to the Rotary.

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

If Rotary is selected on the "out:" menu, MIDI will be 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).

The parameters for each stage are as follows:

Parameter	Description
Tempo sync on/off	The button below the Rate knob is used to switch tempo sync on or off. The button is lit when tempo sync is on.
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, without sync to tempo.
Width	This determines the depth of the chorus effect. Higher settings produce a more pronounced effect.
Spatial	This 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, this should be set to maximum as you can control the dry/effect balance with the send.
Delay	This parameter affects the frequency range of the modulation sweep, by adjusting the initial delay time.
Shape	This changes the shape of the modulating waveform, altering the character of the chorus sweep. Sine and triangle waveforms are available.
Filter Lo/Hi	These parameters allow you to roll off low and high frequencies of the effect signal, respectively.
Side-Chain On/Off	When this is activated, the modulation can be controlled by a signal routed to the Side-Chain input. When the side-chain signal exceeds the threshhold the modulation will be controlled by the side-chain signal's envelope. For a description on 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.

Parameter	Description
Waveform buttons	Sets the 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, without sync to tempo.
Tempo sync on/off	The button above the Speed knob is used to switch tempo sync on or off. The button is lit when tempo sync is on.
On button	Turns modulation of the pitch parameter on or off.
Mono button	Governs whether the output will be stereo or mono.
Output	Adjusts the output level of the effect.
Mix	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.

Parameters are as follows:

Parameter	Description
Tempo sync on/off	The button below the Rate knob is used to switch tempo sync on or off. The button is lit when tempo sync is on.
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, without sync to tempo.
Depth	This governs the depth of the amplitude modulation.
Spatial	This will add a stereo effect to the modulation.
Output	Adjusts the output volume.
Side-Chain On/Off	When this is activated, the modulation can be controlled by a signal routed to the Side-Chain input. When the side-chain signal exceeds the threshhold the modulation will be controlled by the side-chain signal's envelope. For a description on how to set up Side-Chain routing, see the chapter "Audio effects" in the Operation Manual.

Vibrato

The Vibrato plug-in produces pitch modulation.

Parameter	Description
Tempo sync on/off	The button below the Rate knob is used to switch tempo sync on or off. The button is lit when tempo sync is on.
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, without sync to tempo.
Depth	This governs the depth of the pitch modulation.
Spatial	This will add a stereo effect to the modulation.
Side-Chain On/Off	When this is activated, the modulation can be controlled by a signal routed to the Side-Chain input. When the side-chain signal exceeds the threshhold the modulation will be controlled by the side-chain signal's envelope. For a description on 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're 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 parameters are:

Parameter	Description
Mode	Select one of four operating modes for the Bitcrusher. Each mode will produce a result sounding a bit different. Modes I and III are nastier and noisier, while modes II and IV are more subtle.
Sample Divider	This 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 will be eliminated, turning the signal into unrecognizable noise.
Depth	Use this to set the desired bit resolution. A setting of 24 gives the highest audio quality, while a setting of 1 will create mostly noise.
Output	Governs the output level from the Bitcrusher. Drag the slider upwards to increase the level.
Mix	This slider regulates the balance between the output from the Bitcrusher and the original audio signal. Drag the slider upwards for a more dominant effect, and drag it 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 parameters are as follows:

Parameter	Description
Waveform buttons	Sets the modulation waveform.
Depth	Sets the depth of the Chopper effect. This can also be set by clicking in the graphic 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, without sync to tempo.
Tempo sync on/off	The button above the Speed knob is used to switch tempo sync on (the button lights up) or off.
Stereo/Mono button	Determines whether the Chopper will work 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 maximum.

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 parameters are as follows:

Parameter	Description
Direct	This 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	This adjust the level of the generated signal one octave below the original pitch. Set to 0 means the voice is muted.
Octave 2	This adjust 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.

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. One 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 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 Declicking process can be visually monitored in the Input and Output displays of the DeClicker window (showing the incoming audio and the processed – DeClicked – audio, respectively). 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.

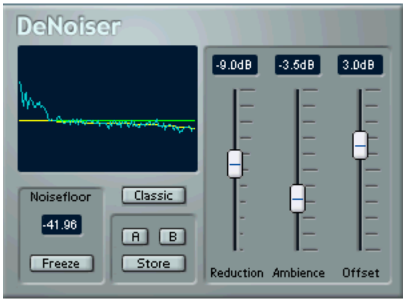
Parameters

Parameter	Description
Audition	When this 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	When this is activated, the DeClicker attempts to remove both audible clicks and crackle noise. When it's deactivated, only single clicks will be 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.
Threshold	This setting 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 to remove inaudible clicks, raise this parameter to a high value, and then lower it until all the artefacts 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 artefacts. If in doubt, activate Audition mode and check that the removed material doesn't contain any actual musical or rhythmical information, etc.
DePlop	This setting controls a special highpass 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 this function to modern recordings, as you may risk removing parts of the useful signal!
Quality	This determines the quality of the click removal and audio restoration, with "4" being the best quality setting. Please note that selecting higher quality settings also 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	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).

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 can't do miracles, it can at least make some improvement to the overall "hardness" introduced by the distortion.

DeNoiser



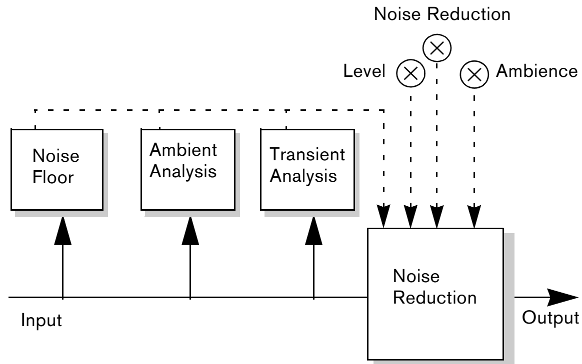
The DeNoiser plug-in lets you suppress noise without affecting the general sound quality. Or, in tech talk, the DeNoiser removes broad band noise from arbitrary audio material without leaving any "spectral finger print". The algorithm that this plug-in is based on has the ability to track and adjust 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 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 help adjust the response of the noise reduction unit, allowing transient-rich material to maintain its liveliness and natural ambience.

⇨ When you process audio in DeNoiser, the plug-in will need 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 noisefloor, and then stop and start over again from the beginning. The plug-in then remembers the settings internally.

The Noisefloor Display

The display to the left in the DeNoiser window is crucial when making settings. It contains the following three elements:

- The dark green spectral graph.
This shows the spectrum of the audio currently 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 graphic 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	If you activate this button, you "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.
Reduction	Governs the amount of noise reduction. The display above this fader 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	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	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.
A/B/Store	These buttons are described below this table.
Classic	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.

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

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

Grungelizer

The 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 available parameters are as follows:

Parameter	Description
Crackle	This adds crackle to create that old vinyl record sound. The farther to the right you turn the dial, 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	This dial regulates the amount of static noise added.
Distort	Use this dial to add distortion.
EQ	Turn this dial to the right to cut off the low frequencies, and create a more hollow, lo-fi sound.
AC	This emulates a constant, low hum of AC current.
Frequency switch	This sets the frequency of the AC current (50 or 60 Hz), and thus the pitch of the AC hum.
Timeline	This dial regulates the amount of overall effect. The farther to the right (1900) you turn this dial, the more noticeable the effect.

Reverb plug-ins

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

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. RoomWorks has the following parameters:

Parameter	Description
Low Freq	Frequency at which the low shelving filter takes effect.
High Freq	Frequency at which the high shelving filter takes effect. Both the high and low filters EQ the input signal prior to reverb processing.
Low Gain	The amount of boost or cut for the low shelving filter.
High Gain	The amount of boost or cut for the high shelving filter.
Pre-Delay	The amount of time before the onset of reverb. This allows you to simulate larger spaces by increasing the time it takes for first reflections to reach the listener.
Reverb Time	Reverb Time in milliseconds.
Size	This alters the delays times of early reflections to simulate larger or smaller spaces.
Diffusion	This affects the character of the reverb tail. Higher diffusion is smoother while less diffusion can be clearer. This emulates changing the types of surfaces in a room (brick vs. carpet for instance).
Width	This controls the width of the stereo image. At 100%, you get full stereo reverb. At 0%, the reverb is all in mono.
Variation	Pressing this button will generate 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.
Hold	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.
Low Range	This determines the frequency below which low damping will occur.
High Range	This determines the frequency above which high frequency damping will occur.
Low Damping	The amount of damping applied to the low frequencies. At 100%, no damping occurs. Values lower than 100% increase the amount of damping, reducing low frequencies over time. Values above 100% have the opposite effect.
High Damping	This affects the decay time of high frequencies. Normal room reverb decays quicker in the high and low frequency range than in the midrange. Lowering the damping percentage will cause high frequencies to decay quicker. Damping percentage values above 100% will cause high frequencies to decay longer than the midrange.
Amount	This determines how much effect the envelope attack and release controls have on the reverb itself. Lower numbers have a more subtle effect while higher numbers sound more drastic.
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 in milliseconds it takes for the reverb to reach full volume after a signal peak. This is similar to a predelay but the reverb is ramping up instead of starting all at once.
Release	The release determines how long after a signal peak the reverb can be heard before being cut off, similar to a gate's release time.
Mix	Determines the blend of dry (unprocessed) signal to wet (processed) signal. When using RoomWorks inserted in an FX channel, you will most likely want to set this to 100% or use the Send button.
Wet only	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 inserted on an FX or group channel.
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.
Rotate	This button is only available for surround configurations. When active, the perspective of the room is shifted 90°.
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. Note that when the Rotate option is activated, these relationships will shift 90°.

Parameter	Description
Efficiency	This unique control determines how much of the CPU is used for RoomWorks. The lower the percentage of efficiency, the more CPU resources will be used. This will yield a higher quality reverb than higher percentage settings. Interesting effects can be created with very high Efficiency settings (>90%). Experiment for yourself.
Export	This button determines if during audio export RoomWorks will use the maximum CPU power for the highest quality reverb or not. You may wish to keep a higher efficiency setting for a desired effect during export. If you want the highest quality reverb during export make sure this is selected (yellow circle around button).

⇨ Note that the options in the Surround section on the far right of the RoomWorks panel are available only when using the plug-in as an insert for a surround-enabled track.

RoomWorks SE

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

Parameter	Description
Pre-Delay	The amount of time before the onset of reverb. This allows you to simulate larger spaces by increasing the time it takes for first reflections to reach the listener.
Reverb Time	Reverb Time in seconds.
Diffusion	This affects the character of the reverb tail. Higher diffusion is smoother while less diffusion can be clearer. This emulates changing the types of surfaces in a room (brick vs. carpet for instance).
High Damping Amount	This affects the decay time of high frequencies. Normal room reverb decays quicker in the high and low frequency range than in the midrange. Lowering the damping percentage will cause high frequencies to decay quicker. Damping percentage values above 100% will cause high frequencies to decay longer than the midrange.
Low Damping Amount	The amount of damping applied to the low frequencies. At 100%, no damping occurs. Values lower than 100% increase the amount of damping, reducing low frequencies over time. Values above 100% have the opposite effect.
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 plug-ins

This section contains descriptions of the plug-ins in the "Spatial" 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 to work.

The parameters are as follows:

Parameter	Description
Width	This controls the width or depth of the stereo enhancement. Turn clockwise to increase the enhancement.
Delay	This parameter increases the amount of differences between the left and right channels to further increase the stereo effect.
Color	This parameter also generates differences between the channels to increase the stereo effect.
Mono	This 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 parameters are as follows:

Parameter	Description
Width	This controls the width or depth of the stereo enhancement. Turn clockwise to increase the enhancement.
Delay	This parameter increases the amount of differences between the left and right channels to further increase the stereo effect.
Color	This parameter also generates differences between the channels to increase the stereo enhancement.
Mono	This switches the output to mono, to check for possible unwanted coloring of the sound which sometimes can occur when enhancing the stereo image.

Surround plug-ins

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

Matrix Decoder

The Matrix Decoder reverses the Encoder process performed by the Matrix Encoder (see above). 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 Matrix Encoder/Decoder to produce a mix that is compatible with this standard.

Setting up

- Create an output bus with the “LRCS” speaker arrangement, in the VST Connections window, and route it to the physical outputs on 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 Matrix Encoder with the 5.0 surround format” on page 45.

- The Encoder should be placed in the first “post fader” insert slot (#7) for the output bus, followed by the Decoder.

Using the Matrix Encoder/Decoder

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

Use the Surround Panner to place channels in the Surround mix, or assign channels to the individual LRCS outputs.

2. Activate the Matrix Encoder.

What you now hear is the encoded stereo mix, the way it will sound when played back on a normal stereo reproducer. If you open the Matrix Encoder control panel you can adjust the Gain of the Lt/Rt output by using the fader.

3. Activate the Matrix Decoder, open the control panel and click on the Steering “On” button.

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

- The “Steering” display shows a ball within the LRCS axis. The position of this ball 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 switching the Matrix Decoder “Bypass” button on and off, 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. If you wish to compare the encoded or decoded mix with the unprocessed mix, you should switch off both the Matrix Encoder 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 isn’t working properly. You can however, with careful tweaking of the mix decrease the signal degradation to a much more acceptable level. You have to adjust levels and other settings before the Matrix Encoder, neither the encoder or decoder can “control” the mix in any way.

5. When you are satisfied with the result, Bypass the Matrix Decoder, 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 now be compatible with common home systems that use the Pro Logic standard.

Using the Matrix Encoder 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 one for 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 there are two Surround Channels in the Matrix Encoder, making a total of 5 Channels. This is meant to be used in conjunction with the 5.0 surround format. Proceed as follows.

1. Create your mix for 5.1.

2. Create an output bus with the "5.0" speaker arrangement, in the VST Connections window, and route it to the physical outputs on your audio hardware.

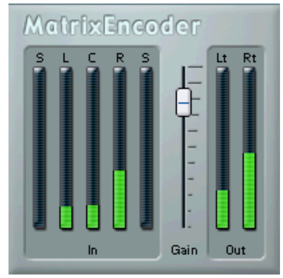
3. Run the mix through the Matrix Encoder.

Now, the two Surround channels will first be merged together to make the mix compatible with LRCS. Then the four resulting signals will be encoded as usual. This will require much fewer adjustment when moving between 5.1 and LRCS.

Using the Matrix Decoder with the 5.0 surround format

The Matrix Decoder also has five channels. This is for similar reasons. Normally two surround speakers are used even when playing back LRCS. The two speakers then simply use the same material. The Matrix decoder 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.

Matrix Encoder



The Matrix Encoder is intended for Pro Logic compatible encoding of multichannel files. This is a process where a 4 channel Surround mix is "packed" into two channels for broadcasting or distribution on video tape, for example. The Matrix Encoder takes four separate inputs; left, right, center, and surround (LRCS), and creates two final outputs, left-total and right-total (Lt and Rt).

Mix6To2

The Mix6To2 effect allows you to control the levels of up to six surround channels, and to mix these down to a stereo output. The pop-up menu contains a number of speaker arrangement presets that correspond to some default surround formats. The Mix6To2 lets you quickly mix down your surround mix format to stereo, and to include parts of the surround channels in the resulting mix.

- Note that Mix6To2 does not simulate a surround mix or add any psycho-acoustical artifacts to the resulting output – it is simply a mixer. Also note that the Mix6To 2 should be placed in one of the post fader insert effect slots for the output bus.

Each of the surround channels has the following parameters:

- Two volume faders that govern the levels of the surround bus to the left and right side of the (master) bus.
- A Link button that links the two volume faders.
- Two Invert buttons allow you to invert the phase of the left and right side of the surround bus.

The Master bus has the following parameters:

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

Mix8To2

The Mix8To2 effect allows you to control the levels of up to eight surround channels, and to mix these down to a stereo output. The pop-up menu contains a number of speaker arrangement presets that correspond to some default surround formats. The Mix8To2 allows you to quickly mix down your surround mix format to stereo, and to include parts of the surround channels in the resulting mix.

- Note that the Mix8To2 does not simulate a surround mix or add any psycho-acoustical artifacts to the resulting output – it is simply a mixer.

- Also note that the Mix8To 2 should be placed in one of the post fader insert effect slots for the output bus.

Each of the surround channels have the following parameters:

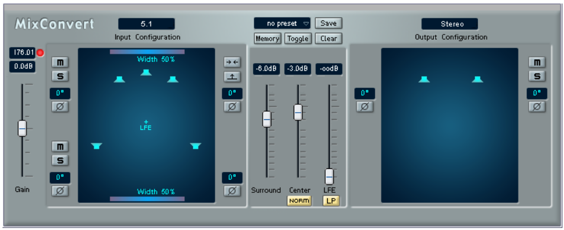
- Two volume faders that govern the levels of the surround bus to the left and right side of the (master) bus.
- A Link button that links the two volume faders.
- Two Invert buttons allow you to invert the phase of the left and right side of the surround bus.

The Master bus has the following parameters:

- A Link button that links the two Master faders.
- A Normalize button that will normalize the mixed output if activated.

Normalize is a function for controlling the overall loudness of the output. When this is activated, the level of the mixed output will be boosted to exactly 0 dB.

Mixconvert



The Mixconvert plug-in is similar to the Mix6To2 plug-in in that it is used to quickly convert a multichannel 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 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.

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.

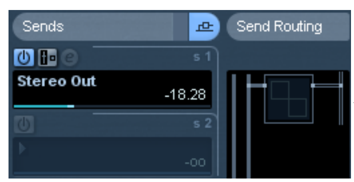
Users can use presets with standard upmix/downmix setups for specific configurations. It is possible to save up to 64 user-defined presets for each input/output configuration.

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

Indicates that Mixconvert is inserted in place of the panner.

- Whenever a multichannel 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, a Mixconvert plug-in will be inserted in place of the aux send's panner.



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

Interface

Overview

The plug-in's interface has three different sections. On the left you find the input Configuration display with all parameters that directly affect the input configuration. In the middle section the level parameters for the upmix/downmix are displayed. Above this, the preset controls can be found. On the right the output configuration is displayed with all parameters that affect the output configuration. Additionally, on the far left there is a gain fader.

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

Global Gain fader

Gain depends on the input signal, the number of loudspeakers and a number of downmix parameters (see "Level" on page 50). You can use this fader to globally adjust gain by ±12dB for all channels.

Max Output Level

This field shows the maximum output level. The LED display on the right hand side of the field indicates whether this maximum level is above 0 dB (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.

Output Configuration

The Output Configuration can only be modified when used as an insert effect. When Nuendo automatically replaces the panner by Mixconvert, the Output configuration is determined by the destination of the channel or aux send. When used as an insert effect, the Output configuration can be changed either directly in the pop-up menu above the Output Configuration display or indirectly by loading a preset.

Faders for Surround, Center and LFE

These faders 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 50). The names Surround, Center and LFE refer to the corresponding channels in the Input Configuration.

Solo and Mute buttons

Using the Solo and Mute buttons (on the left of the Input Configuration and the right of the Downmix Configuration sections) you can mute or solo all front or surround channels simultaneously (see "Solo mode" on page 50).

Soloing or muting individual speakers

If you want to solo or mute a single loudspeaker in the Input Configuration or Output Configuration displays, you can click on it. Simply clicking will solo the channel. When you hold down the [Alt]/[Option] key while clicking, the channel will be muted. Holding down the [Ctrl]/[Command] key while clicking will also mute all channels currently in solo mode. Clicking again (without a modifier key) will reset the channel.

Phase shift

You can shift the phase of the front left/right channels and the surround left/right channels in steps of 90°. Clicking the button once will increase the phase by a further 90°. You can reset the phase value by right-clicking (Windows) or [Ctrl]-clicking (Mac) on the button.

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, you should 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, you should avoid introducing additional phase shifts, since these would prevent the decoding of the mix for different speaker configurations.

Toggling between parameter sets

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. This buffer is cleared when clicking the Clear button. Using the Toggle button, you can switch between the buffered parameter set and the (changed) current parameter set. Note that here the Output Configuration is not a parameter, but must be identical for both parameter sets.

Modifying the width

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, the default setting will be 50%. The 50% setting results in unaltered signals. Values above 50% will create an artificial widening of the panorama; similar to phase shifting. You should be careful when 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.

Loading and saving presets

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

Presets are selected and managed at the top of the middle section of the plug-in interface. The name of the currently 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.

General Notes

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.

Keyboard shortcuts

The plug-in interface is designed for mouse operation. There are two commands with these keyboard shortcuts:

- Store Parameter Memory: [M] (for "memory")
- Toggle Parameters: [S] (for "swap")

Solo mode

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

Functionality and available conversions

The speaker configuration of the input mix (Input Configuration) is defined by the width of the channel it is inserted in. It is displayed automatically. The speaker configuration of the output mix (Output Configuration) is automatically selected when Mixconvert is inserted in the panner position of a channel or aux send. If it is used as an insert effect, the output configuration can be selected either from the corresponding menu or by loading a preset.

Note, however, that not all theoretically possible combinations are actually available. Mixconvert is limited to channels with 8 audio paths (this means that 10.2 or 8.1 are not supported). In the tables presented (see "Mixconvert Appendix" on page 72) you can find all available combinations.

Brief description of Mixconvert parameters

Parameter	Description
Width	Modifies the panorama – 0% (minimum width) – 50% (normal width, unaltered) – 100% (maximum width)
Global Gain	Attenuates or increases all channels to compensate for clipping or low levels in the converted signal.
Surround level	Level of the surround channel.
LFE level	Level of the LFE channel.
Center level	Level of the front center channel.
Phase shift	Phase shift of a channel (settings: 0°, 90°, 180°, 270°), available for front and surround left/right. Click once for shifting a further 90°. Right-click/[Ctrl]-click to reset to 0°.
Speaker	Click a speaker symbol to set the speaker to mute or solo mode. [Alt]/[Option]-click for activating the Mute mode. [Ctrl]/[Command]-click for activating the exclusive solo (mute all other channels even if they are also solo). Click again on a speaker to reset the channel.
Solo button	Soloes all front and surround channels.
Mute button	Mutes all front and surround channels.
Output Config	Only available when used as insert. Sets the output speaker configuration.
Store Memory	Temporarily saves the current parameter set.
Toggle Memory	Toggles between the current and the temporary parameter set.
Clear Memory	Clears the temporary parameter buffer.
Save Preset	Saves or deletes the preset specified in the preset text field.
Preset pop-up menu	Loads a preset.

Available conversions

For a list of the available conversions, see "Mixconvert Appendix" on page 72.

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

The MixerDelay is a tool that allows you to adjust and manipulate each individual channel in a surround track, group or bus. Each channel has the following controls:

- Level faders allow you to fine-tune the volume balance between the surround channels.
- Mute and Solo buttons are useful for listening to individual channels, etc.
- Phase switches let you invert the phase or polarity for individual channels.
- Delay controls allow 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.

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 1000ms) and fine adjustments are best made by numerically entering the delay time in centimeters for speaker alignment.

- The channel 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. (Simply click the "Select Presets" button on the common panel to open a pop-up menu listing the available presets.)

Finally there is a common panel to the right with global buttons for turning off Mute, Solo and Input Phase switches for all channels.

⚠ 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, you should 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 real-time (playback) or with the Export Audio Mixdown function, you should consider dithering.

- Since SurroundDither is capable of dithering up to six channels at the same time, it is recommended if you're using surround channels.
If not, you may want to use the UV22 HR instead, see "Mastering – UV 22 HR" 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 six buttons, one for each channel.

- Above each button there are six corresponding value fields that display the bit resolution the files 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. Add SurroundDither to a post fader insert effect slot for the output bus.
I.e. in one of the last two slots.

2. Open the control panel for SurroundDither, 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 Channel 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 plug-ins

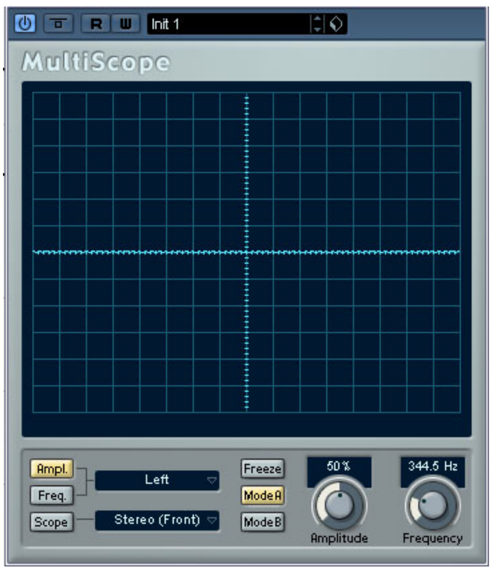
This section describes the plug-ins in the “Tools” category.

MultiScope

The 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.)

Ampl (Oscilloscope) mode



- To view a signal waveform, open the MultiScope control panel and make sure that the button “Ampl.” 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 won’t matter.

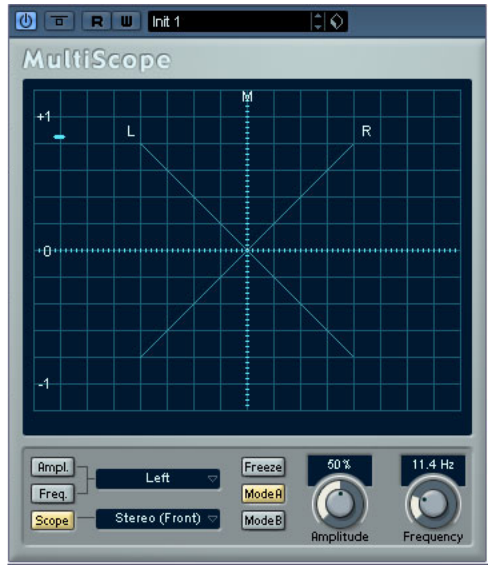
- If the MultiScope is used with a multi-channel track or output bus, you can select any speaker channel for viewing, or All Channels to view them all at once.

- 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.

- The “Freeze” button can be used to freeze the display for all three Scope modes.

Click it again to exit freeze mode.

Phase Correlator mode



To select the phase correlator, 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.

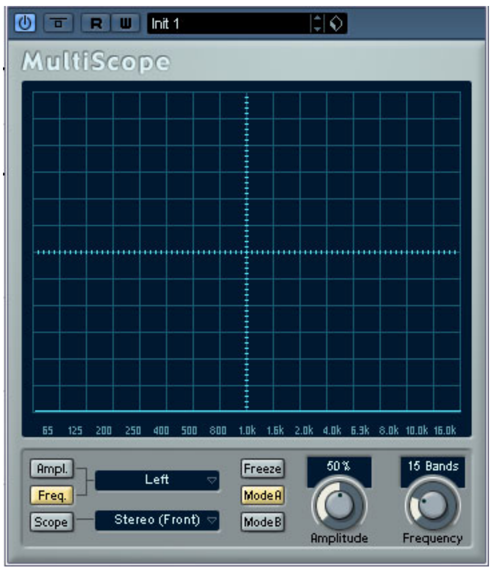
- Generally, the more you can see a “thread”, the more bass in the signal, and the more “spray-like” the display, the more high frequencies in the signal.

When the 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.

Frequency Spectrum Analyzer



- Click on the “Freq” button so that it lights up in yellow.

The MultiScope is now in Frequency Spectrum analyze mode, and will divide 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 won’t matter.

- If the 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.

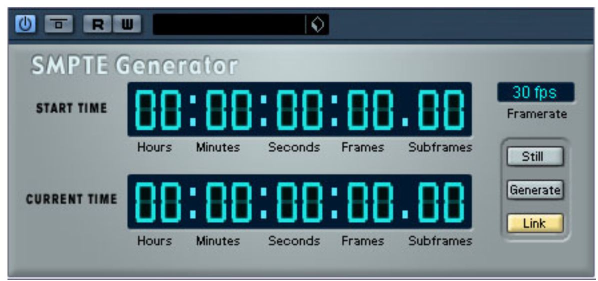
- Adjust the Amplitude knob to increase/decrease the vertical range of the bands.

- By adjusting the Frequency knob, you can divide the frequency spectrum into 8, 15, or 31 bands, or you can select “Spectrum”, which shows 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 don’t have any effect if you have selected “Spectrum” with the Frequency knob.

SMPTEGenerator



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

The following items and parameters are available:

- Still Button

Activate this to make the device generate SMPTE time code at the current cursor position in stop mode.

- Generate Button

Activate this to make the device generate SMPTE time code.

- Link Button

This synchronizes the time code output to the Transport time positions. When Link is activated, the time code output will exactly match the play position in Nuendo.
Activating the Generate button makes the device send the SMPTE time code in “free run” mode, meaning that it will output continuous time code, independently from the transport status in Nuendo. If you wish to “stripe” a tape with SMPTE, you should use this mode.

- Start Time

This sets the time at which the SMPTE Generator starts, when activated in “free run” mode (Link button off). To change the Start time, click on a digit and move the mouse up or down.

- Current Time

When Link is on this shows the current position in Nuendo. If Link is off it shows the current time of the SMPTE Generator in “free run” mode. This cannot be set manually.

- Framerate

This defaults to the frame rate set in the Project Setup dialog. If you wish to generate time code in another frame rate than the Project is currently set to (for example to stripe a tape), you can select another format on the Framerate pop-up (provided that “Link” is off).
Note, however, that for the other device to synchronize correctly with Nuendo, the framerate has to be the same in the Project Setup dialog, the SMPTE Generator and in the receiving device.

Example – Synchronizing a device to Nuendo

Proceed as follows:

1. Connect the SMPTE Generator as an insert effect on an audio channel, and route the output of that channel to a separate output.

Make sure that no other insert or send effects are used on the time code channel. You should also disable EQ, if this is active.

2. Connect the corresponding output on the audio hardware to the time code input on the device you wish to synchronize to Nuendo.

Make all necessary settings in the other device, so that it is set to synchronize to incoming timecode.

3. Adjust the level of the time code if needed, either in Nuendo or in the receiving device.

Activate Generate button (make the device send the SMPTE time code 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 button.

The SMPTE Generator will now output time code that matches the position of the Nuendo Transport panel.

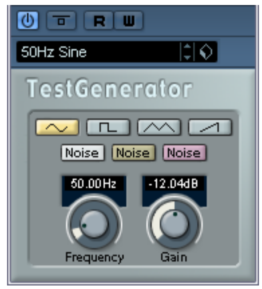
- Press Play on the Nuendo Transport panel.

The other device is now synchronized and will follow any position changes set with the Nuendo transport controls.

Drag offset for display

If you want to enter an offset, click with the mouse into the display and drag upwards or downwards to change the values. This enters a display offset - the current cursor position will not be affected. In Generate mode this offsets the Start Time, in Link mode it offsets the generated Timecode.

TestGenerator



This utility 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, including 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 and various types of noise. In addition, you can also set the frequency and amplitude of the generated signal.

As soon as you add the TestGenerator as an effect to 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	By clicking these buttons, you select the basis for the signal generated by the waveform generator. You can select between four basic waveforms: Sine, Square, Sawtooth and Triangle, or three types of noise (white, brown and pink noise – from left to right).
Frequency	This controls the frequency of the generated signal, from 1 Hz to 20000 Hz.
Gain	This controls the amplitude of the signal. The higher the value (up to 0 dB) the stronger the signal.

3

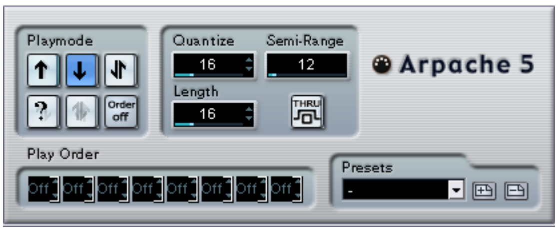
MIDI effects

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.
Check 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 Quantize setting to set the arpeggio speed.
The speed is set as a note value, relative to the project tempo. For example, setting Quantize 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 smaller than the Quantize setting) or arpeggio notes that overlap each other (Length greater than Quantize).

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

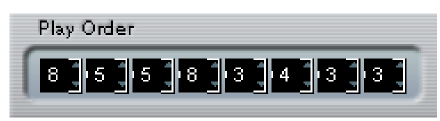
6. Play a chord on your MIDI instrument.
Now, instead of hearing the chord, you will hear the notes of the chord played one by one, in an arpeggio.

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

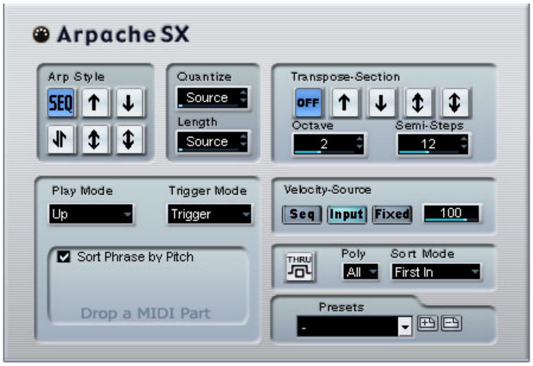
Parameters

The Arpache 5 has the following settings:

Setting	Description
Playmode buttons	Allows you to select the playback order for the arpeggiated notes. The options are down+up, up+down, up, down, random ("?" button) and "Order off", in which case you can set the playback order manually with the Play Order fields below.
Quantize	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 Quantize setting.
Semi-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).
Thru	If this is activated, the notes sent to the arpeggiator (i.e. the chord you play) will be passed through the plug-in (sent out together with the arpeggiated notes).
Play Order	If the "Order on" playmode is selected, you can use these "slots" to specify a custom playback order for the arpeggio notes: Each slot 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.



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 the following parameters:

Parameter	Description
Arp Style	Determines the basic behavior of the Arpache SX. In the Seq mode, the arpeggiator uses an imported MIDI part as a starting point for the pattern – this is described below. All other modes describe how the notes in the chord you play should be arpeggiated – up, down, up & down, mostly up or mostly down.
Quantize	Determines the resolution of the arpeggio, i.e. its “speed”. The “Source” setting is used in Seq mode, see below.
Length	Determines the length of the arpeggio notes. The “Source” setting is used in Seq mode, see below.
Transpose	When a mode 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. The “Octave” setting sets the number of transposed repeats and the “Semi-Steps” setting determines how much each repeat will be transposed.
Play Mode	See the description of Seq mode below!
Trigger Mode	See the description of Seq mode below!
Velocity Source	Determines the velocity of the notes in the arpeggio. The options are Seq (used in Seq mode only), Input (the same as the velocity values of the corresponding notes in the chord you play) or Fixed, in which case all arpeggio notes will get the velocity set in the value field to the right.
Thru	If this is activated, the notes sent to the arpeggiator (i.e. the chord you play) will be passed through the plug-in (sent out together with the arpeggiated notes).
Poly	Determines how many notes should be accepted in the input chord. The “All” setting means there are no limitations.
Sort Mode	When you play a chord into the Arpache SX, the arpeggiator will look at the notes in the chord as sorted 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.

Seq mode

When Seq mode is selected in the Arp Style section, the Arpache SX uses an additional MIDI part as a pattern. This pattern then forms the basis for the arpeggio, in conjunction with the MIDI input.

- To import a MIDI part into the Arpache SX, drag it from the Project window and drop it in the “Drop a MIDI Part” section on the Arpache SX.

Now, the notes in the dropped MIDI part will be sorted internally, either according to their pitch (“Sort Phrase 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.

Now the MIDI input (the chord you send into the Arpache SX) will also generate a list of numbers, with each note in the chord corresponding to a number depending on the Sort Mode setting.

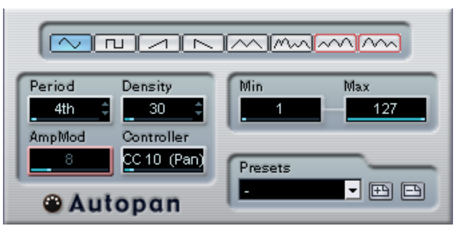
The two lists of numbers will now be matched – the Arpache SX tries to play back the pattern from the dropped MIDI file but using the notes from the MIDI input (chord). The result depends on the Trigger Mode setting:

Trigger Mode	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 Mode 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.

Trigger Mode	Description
Sort Normal	Matches the notes in the MIDI input to 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.

Finally, the Play Mode setting affects the resulting arpeggio. Note also that you can choose to keep the original note timing, note length and note velocities from the dropped MIDI part, by selecting "Source" in the Quantize and Length fields, and "Seq" in the Velocity Source section.

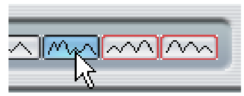
Autopan



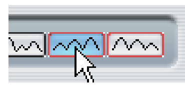
This plug-in works a bit like an LFO in a synthesizer, allowing you to send out continuously changing MIDI controller messages. One typical use for this is automatic MIDI panning (hence the name), but you can select any MIDI Continuous Controller event type. The Autopan effect has the following parameters:

Waveform selectors

These determine the shape of the controller curves sent out. The results of most of these waveforms are obvious from looking at the buttons, but a few of them require some extra explanations:



This generates a "random" controller curve.



These generate curves with a "periodical envelope". The amplitude will gradually increase or decrease over a time, set with the Period parameter (see below).

Period

This is where you set the speed of the Autopan, or rather the length of a single controller curve cycle. The value can be set in ticks (1/480ths of quarter notes), or as rhythmically exact note values (by clicking the arrow buttons next to the value). The lower the note value, the slower the speed. For example, if you set this to 240 ("8th") the waveform will be repeated every eighth note.

Density

This determines the density of the controller curves sent out. The value can be set in ticks (1/480ths of quarter notes), or as rhythmically exact note values (by clicking the arrow buttons next to the value). The higher the note value, the smoother the controller curve. For example, if you set this to 60 (shown as "32th") a new controller event will be sent out every 60th tick (at every 1/32 note position).

⚠ You should probably avoid extremely low Density values, as these will generate a very large number of events (which may cause the MIDI instrument to "choke", delaying notes etc.).

AmpMod

This is only used for the two waveforms with “periodical envelopes” (see above). The period value (set in beats) determines the length of the envelope. In the following figure, Period is set to 4th and the AmpMod is 4 beats. This results in a quarter note-based curve in which the top amplitude decreases gradually, repeated each bar.

Controller

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!

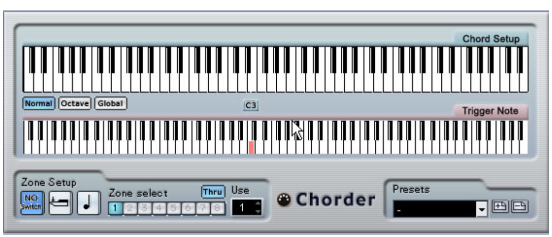
Min and Max

These determine the minimum and maximum controller values sent out, i.e. the “bottom” and “top” of the controller curves.

Chorder

The Chorder is a MIDI chord processor, allowing you to assign complete chords to single keys in a multitude of variations. There are three main modes of operation: Normal, Octave and Global. You switch between these modes by clicking the respective button to the left below the keyboard.

Normal mode



In this mode, you can assign a different chord to each single key on the keyboard. Proceed as follows:

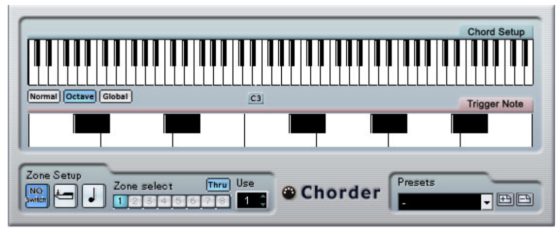
1. Select the key to which you want to assign a chord, by clicking in the lower “Trigger Note” keyboard display.

2. Set up the desired chord for that key by clicking in the upper “Chord Setup” keyboard display.
Clicking a key adds it to the chord; clicking it again removes it.

3. Repeat the above with any other keys you wish to use.

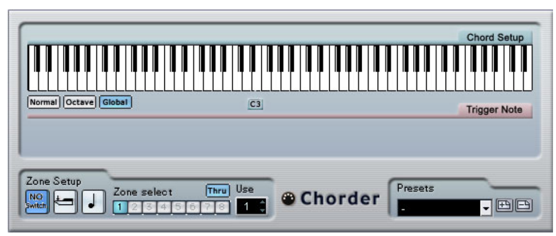
If you now play the keys you have set up, you will instead hear the assigned chords.

Octave mode



The Octave mode is similar to the Normal mode, but you can only set up one chord for each key in an octave (that is, twelve different chords). When you play a C note (regardless of whether it’s a C3, C4 or any other octave) you will hear the chord set up for the C key.

Global mode



In the Global mode, you only set up a single chord, using the Chord Setup keyboard display (the lower keyboard display is hidden). This chord is then played by all keys on the keyboard, but transposed according to the note you play.

Using switches

The Switch Setup section at the bottom of the panel allows you to set up variations to the defined chords. This works with all three modes and provides a total of eight variations for each assignable key (that is, a maximum of 8 different chords in Global mode, 12 x 8 chords in Octave mode and 128 x 8 chords in Normal mode).

The variations can be controlled by velocity or note range. Here's how you set it up:

1. Select one of the two switch modes: velocity or note.
How to use these is explained below.

The velocity switch mode selected.

2. Specify how many variations you want to use with the Use value box.

3. Click the first Switch Select button and set up the chord(s) you want for the first variation.

4. Click the next Switch Select button and set up the chord(s) you want for that variation.

5. Repeat this for the number of variations you specified with the Use setting.
Each Switch Select button corresponds to a variation.

6. Now you can play the keyboard and control the variations according to the selected switch modes.
These work as follows:

Switch mode	Description
Velocity	The full velocity range (1–127) is divided into "zones", according to the number of variations you specified. For example, if you're using two variations (Max is set to 2) there will be two velocity "zones": 1–63 and 64–127. Playing a note with velocity at 64 or higher will trigger the second variation, while playing a softer note will trigger the first variation.
Note	In this mode, the chorder will play one chord at a time – you cannot play several different chords simultaneously. When the Note switch mode is selected, you play a key to determine the base note for the chord, then press a higher key to select a variation. The variation number will be the difference between the two keys. To select variation 1, press a key one semitone higher than the base note, for variation 2, press a key two semitones higher, and so on.

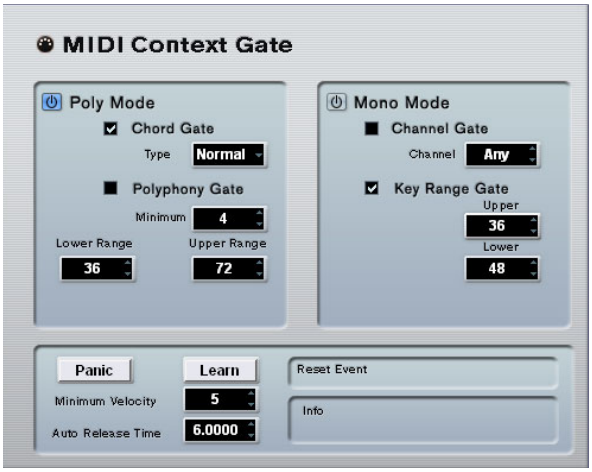
- To turn the variation switch feature off, select the "No Switch" mode.

Compress

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 parameters are:

Parameter	Description
Threshold	Only notes with velocities over 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.
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 can be used for context selective control of MIDI devices. The following parameters are available:

Poly Mode – Chord Gate

When Chord Gate is activated, only notes in recognized chords are let through. There are two modes of chord recognition available; Simple and Normal. In Simple mode, all standard chords (major/minor/b5/dim/sus/maj7 etc.) are recognized, whereas Normal mode also takes more tensions into account.

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 Minimum value field allows you to specify the minimum number of notes needed for the notes to be let through.
- The Upper/Lower Range sets the key range.

Only notes within this range will be let through.

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 either set this to a specific channel (1–16), or to “Any”, i.e. no channel gating.

Mono Mode – Key Range 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 Upper and Lower range (and additionally the set Channel Gate channel, if checked).

Panic button

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

Learn 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.

Auto Release time

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

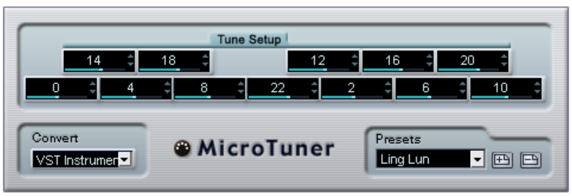
Minimum Velocity

Notes below a set velocity threshold value will be gated.

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 new notes.

Micro Tuner

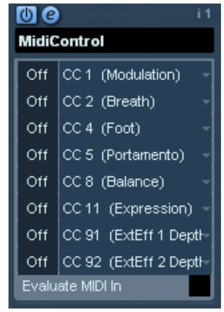


The Micro Tuner lets you set up a different microtuning scheme for the instrument, by detuning each key.

- Each Detune field 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).
- Set the Convert setting according to whether the track is routed to a VST instrument or a “real” standard MIDI instrument (capable of receiving microtuning information).

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

MIDIControl



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’re 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).

MIDIEcho

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 doesn't "echo" the actual audio, but the MIDI notes which will eventually produce the sound in the synthesizer.

The following parameters are available:

Quantize

The echoed notes will be moved in position to a quantizing grid, as set up with this parameter. You can either use the slider or type to set the value in ticks (1/480 ticks of quarter notes) or click the arrow buttons to step between the "rhythmically exact" values (displayed as note values – see the table below). This makes it easy to find rhythmically relevant quantize values, but still allows experimental settings in between.

An example: setting this to "16th" will force all echo notes to be played on exact 16th note positions, regardless of the timing of the original notes and the Echo-Quant. setting.

⇨ To disable quantizing, set this parameter to its lowest value (1).

Length

This sets the length of the echoed notes. This can either be the same as their original notes (parameter set to its lowest value, "Source") or the length you specify manually. You can either set the length in ticks or click the arrow buttons to step between the "rhythmically exact" lengths (displayed as note values – see the table below).

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

Repeat

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

Echo-Quant.

The Echo-Quant. parameter sets the delay time, i.e. the time between a played note and its first echo note. You can either use the slider or type to set the value in ticks (1/480 ticks of quarter notes) or click the arrow buttons to step between the "rhythmically exact" delay times (displayed as note values – see the table below).

For example, setting this to "8th" will cause the echo notes to sound an eighth note after their original notes.

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

Velocity Decay

This parameter allows you to add or subtract to 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 Decay

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.

Echo 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 Echo-Quant. 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 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 (Echo-Quant., Length and Quantize) can all be set in ticks. 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

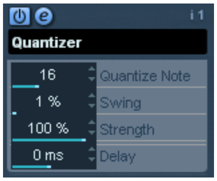
Note to CC

This effect will generate a MIDI continuous controller event for each incoming MIDI note. The value of the controller event corresponds to the note number (pitch) and the single parameter allows you to select which MIDI controller should be sent out (by default controller 7, MIDI volume). The incoming MIDI notes pass through the effect unaffected.

For example, if MIDI volume (controller 7) is selected, notes with low note numbers (pitches) will lower the volume in the MIDI instrument, while higher note numbers will raise the volume. This way you can create “keyboard tracking” of volume or other parameters.

⚠ Note that a controller event is sent out each time a new note is played. If high and low notes are played simultaneously, this could lead to somewhat confusing results. Therefore, the Note to CC effect is probably 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.).

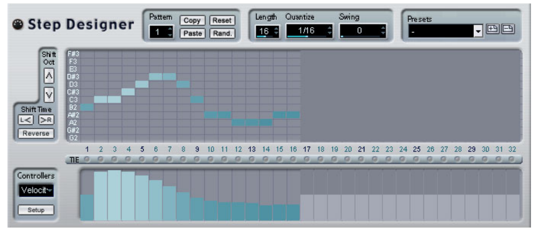
⇨ The main Quantize function in Nuendo is described in the Operation manual.

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.

Step Designer



The Step Designer 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 Step Designer can hold up to 200 different patterns.

2. Use the Quantize setting to specify the "resolution" of the pattern.

In other words, this setting determines how long each step is. For example, if Quantize is set to "16th" each step will be a sixteenth note.

3. Specify the number of steps in the pattern with the Length setting.

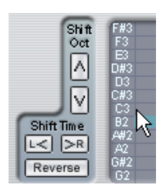
As you can see in the note display, the maximum number of steps is 32. For example, setting Quantize to 16 and Length to 32 would create a two bar pattern with sixteenth note steps.

4. Click in the note display to insert notes.

You can insert notes on any of the 32 steps, but the Step Designer will only play back the number of steps set with the Length 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 Step Designer is monophonic.

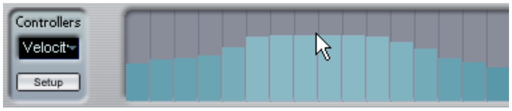


Click and drag to view other octaves.

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

5. Select "Velocity" on the Controllers pop-up menu.
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 Controllers 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 Quantize parameter).

8. To make notes longer, you can tie two notes together. This is done by inserting two notes and clicking the Tie button below the second note.
When the Tie button is lit for a note, it won't retrigger – instead the previous note will be lengthened. 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.

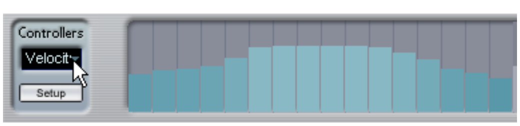
9. If you now start playback in Nuendo, the pattern will play as well, sending out MIDI notes on the track's MIDI output and channel (or, if you have activated the Step Designer as a send effect, on the MIDI output and channel selected for the send in the Inspector).

Adding controller curves

The Controllers 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 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 will be sent out on that step.

Other pattern functions

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

Function	Description
Shift Oct	These buttons allow you to shift the entire pattern up or down in octave steps.
Shift Time	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 Step Designer or another).
Reset	Clears the pattern, removing all notes and setting controller values to default.
Random	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	Note that a stored Preset contains all 200 patterns in the Step Designer.

Automating pattern changes

You can create up to 200 different patterns in each Step Designer – 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 subtrack for the Step Designer'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 Step Designer 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 Step Designer as an insert effect.

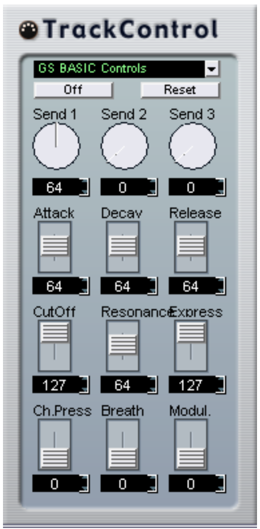
2. Set up several patterns as described above.

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 Basic Controls	Effect sends and various sound control parameters for use with instruments compatible with the Roland GS standard.
XG Effect + Sends	Effect Sends and various sound control parameters for use with instruments compatible with the Yamaha XG standard.
XG Global	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 labelled “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 Reverb Send settings are 64.

GS Basic Controls

The following controls are available when the GS Basic 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.
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 Effects + Sends

The following controls are available when the XG Effects + Sends 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 Global Settings

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).

Track FX

This plug-in is essentially a duplicate of the Track Parameter section. This can be useful if you e.g. need extra Random or Range settings, or if you prefer to have your track parameters in a separate window (to get this, [Alt]/[Option]-click the Edit button for the effect).

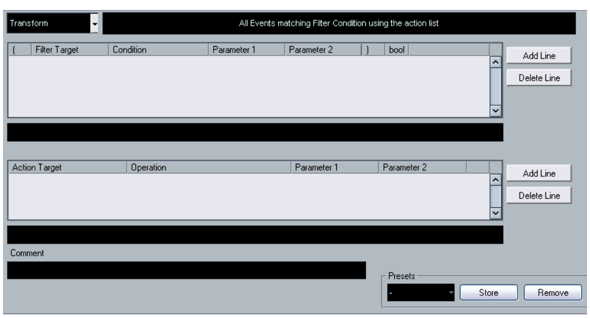
The Track FX also includes an additional function that isn't available among the track parameters:

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.

Transformer



The Transformer is a real-time 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.

4

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