

Audio Effects and VST Instruments

# NUENDO 3

MEDIA PRODUCTION SYSTEM



Original Manual: Synkron

Revision and Quality Control for Nuendo 3:

Ashley Shepherd, Cristina Bachmann, Heiko Bischoff, Sabine Pfeifer, Claudia Schomburg

Thanks: Alexander Lerch

The information in this document is subject to change without notice and does not represent a commitment on the part of Steinberg Media Technologies GmbH. The software described by this document is subject to a License Agreement and may not be copied to other media except as specifically allowed in the License Agreement. No part of this publication may be copied, reproduced or otherwise transmitted or recorded, for any purpose, without prior written permission by Steinberg Media Technologies GmbH.

All product and company names are <sup>™</sup> or <sup>®</sup> trademarks of their respective owners. Windows XP is a trademark of Microsoft Corporation. The Mac logo is a trademark used under license. Macintosh and Power Macintosh are registered trademarks.

© Steinberg Media Technologies GmbH, 2004.

All rights reserved.



## **Table of Contents**

## **5 The included effect plug-ins**

---

- 6 Introduction
- 7 Delay plug-ins
- 10 Distortion plug-ins
- 16 Dynamics plug-ins
- 37 Filter plug-ins
- 46 Modulation plug-ins
- 59 Other plug-ins
- 67 Restoration
- 75 Reverb plug-ins
- 80 Surround plug-ins
- 99 Tools

## **107 The included VST Instruments**

---

- 108 A1 Synthesizer
- 118 VB-1 Bass Synth
- 120 LM-7 Drum Machine
- 123 Embracer – Surround Pad Synthesizer
- 128 Monologue – Monophonic Analog Modeling Synthesizer

## **135 Mixconvert Appendix**

---

- 136 Available conversions

## **141 Index**

---

**1**

**The included effect plug-ins**

# Introduction

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

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

# Delay plug-ins

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

## DoubleDelay



This effect provides two separate delays that can be either tempo based or use freely specified delay time settings. Nuendo automatically provides the plug-in with the tempo currently used in the project.

The parameters are as follows:

Parameter	Description
Mix	Sets the level balance between the dry signal and the effect. If DoubleDelay is used as a send effect, this should be set to maximum (100%) as you can control the dry/effect balance with the send.
Tempo sync on/off	The buttons above the two Delay Time knobs 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, without sync to tempo.
Delay Time 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 Time 2	As above.
Feedback	This sets the number of repeats for both delays.
Tempo Sync 1	The note value multiplier (x1 to x10) for the first delay unit.
Tempo Sync 2	As above, but for the second delay unit.

---

<b>Parameter</b>	<b>Description</b>
Pan1	This sets the stereo position for the first delay.
Pan2	This sets the stereo position for the second delay.

---

You can also change parameters in the graphic display window. This works as follows:

- If tempo sync is on, you can set the Tempo Sync 1 parameter by dragging the light blue handle left and right. When tempo sync is off, this sets the Delay Time 1 parameter.
- You can set the Pan 1 parameter by dragging the light blue handle up and down.
- The dark blue handle works in the same way but for the corresponding second delay parameters.



# ModDelay



This is a delay effect that can either be tempo-based or use freely specified delay time settings. The delay repeats can also be modulated.

The parameters are as follows:

---

Parameter	Description
Mix	Sets the level balance between the dry signal and the effect. If ModDelay is used as a send effect, this should be set to maximum as you can control the dry/effect balance with the send.
Tempo sync on/off	The button above 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.
Delay Time	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 knob	This is the note value multiplier (x1 to x10) for the delay when tempo sync is used.
DelayMod.	This controls the pitch modulation rate for the delay effect.

---

# Distortion plug-ins

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

## 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”.

# Overdrive



Overdrive is a distortion-type effect, emulating the sound of a guitar amplifier. A selection of factory styles is available. Note that these are not stored parameter settings, but different basic overdrive algorithms, with the style names indicating the basic character of each algorithm.

The parameters are as follows:

---

Parameter	Description
Input	Sets the input level.
Output	Sets the output level. As overdrive generates harmonics, it increases the level of the processed signal. You can use the Output fader to compensate for the level increase.
Speaker simulation	When this is activated, the effect simulates the sound of a speaker cabinet.
Factory Styles	Select one of six presets, which can be used as they are or as a basis for further “tweaking”.
Bass	Tone control for the low frequencies.
Mid	Tone control for the mid frequencies.
Hi	Tone control for the high frequencies.
Drive	Governs the amount of overdrive. You can also adjust this by clicking and dragging in the display.

---

# QuadraFuzz



QuadraFuzz is a high-quality distortion effect divided into four frequency bands allowing for control over the level both before and after distortion. This high level of control can create a very wide selection of distortion effects, ranging from subtle to extreme. The user interface consists of two windows.

- The main window features four Filterbank controls, the master Gain and Output controls and a preset selector.
- In the editor window (which is opened by clicking the “Edit” button in the lower right corner) the main feature is a frequency band display. This is where you set the width of the frequency bands as well as their level before distortion.

## How does QuadraFuzz work?

Here's a short description of the three major factors that determine how QuadraFuzz sounds, and where you find the corresponding controls:

- The signal volume control before distortion.  
You can use the Gain control on the left side of the QuadraFuzz main window to control the overall input level of the signal that is fed into the distortion stage. The signal is split up into four frequency bands in the editor window, with adjustable width and level controls. These control the input level before distortion.



- The distortion type, based on a selectable distortion characteristic.

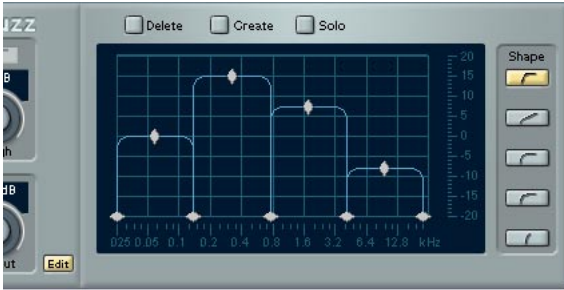


- The signal volume control after distortion.  
The Output control on the right side of the QuadraFuzz main window controls the overall output level. In addition, the Filterbank controls in the same window allow you to raise or lower the output volume of each separate frequency band that was defined in the editor window.



## Editing in the frequency band display

The signal is divided into four frequency bands before being passed to the distortion stage, as explained earlier. You adjust the level and width of these bands in the frequency band display.



The frequency band display

Two value scales as well as a number of rhomb- and diamond-shaped handles are available.

- The diamond-shaped handles at the bottom are used to define the corner frequencies of the different frequency bands.
- By using the rhomb-shaped handles on top of each frequency band you determine its relative level before distortion.
- The horizontal value scale below the Frequency band display indicates frequency. The maximum value on this scale corresponds to half the sample rate of the audio file used (Nyquist theorem).
- The vertical value scale to the right shows the approximate level of an edited frequency band.
- If you click and hold on one of the handles, its current value is displayed. Depending on the handle type, corner frequency or level is shown.
- The corner frequency handles can be moved by dragging horizontally. The level handles can be moved by dragging them up or down.
- To reset a level handle to 0 dB, hold down the [Shift] key on your computer keyboard and click on the handle.
- If you hold down the [Ctrl]/[Command] key and move a handle, the values will change in smaller steps.
- The “Solo” button above the frequency band display allows you to monitor individual frequency bands.

If Solo is activated, one of the four bands is highlighted indicating the selected band. You select other bands by clicking on them.

## The parameters

The following tables list all parameters available in QuadraFuzz.

The parameters in the main window are as follows:

---

<b>Parameter</b>	<b>Description</b>
Gain dial	This dial can be found in the lower left corner of the QuadraFuzz window. You can use it to control the level of the overall input signal before distortion.
Filterbank dials: Low/Low Mid/ High Mid/High	These dials are used to control the output level of the corresponding frequency band after distortion. Values between +/- 12 dB can be set for each band.
Presets fader	This is used to select one of the available presets. To select a new preset, click on the fader handle and drag horizontally.
Output dial	This controls the overall output level.
Over LED	When lit, this indicates that the total input signal level exceeds 0 dB. This LED does not refer to the output level but solely to the input level before distortion. Levels above 0 dB are subject to strict limiting and cause signal clipping. As this is sometimes what you want, QuadraFuzz also offers this option.
Edit button	By clicking on this button, located in the lower right corner of the main window, you can open or close the editor window.

---

The parameters in the editor window are as follows:

---

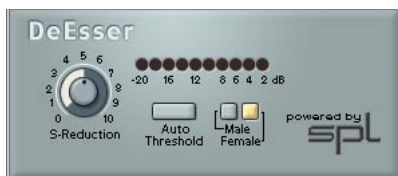
<b>Parameter</b>	<b>Description</b>
Create	If you click on this button, a dialog will open where you can add (and name) a new preset to the preset set currently in memory. The presets are stored with the project – to make a preset available in other projects you use the File pop-up menu as usual.
Delete	This deletes the selected preset from the preset set currently in memory. If you click on the button, a dialog appears where you can confirm or cancel the action.
Solo	This mutes all frequency bands except the selected band.
Shape buttons	The available distortion characteristics (from bottom to top) create effects from a slight distortion up to a trashy hardcore sound.
Frequency band display	Here you control the level and bandwidth for the four bands, see above.

---

# Dynamics plug-ins

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

## 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 0 dB (no reduction) and minus 20 dB (the s-frequency level is lowered by 20 dB). Each segment in the display represents a level reduction of 2 dB.
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 7 kHz range for the female voice and in the 6 kHz range for the male voice.



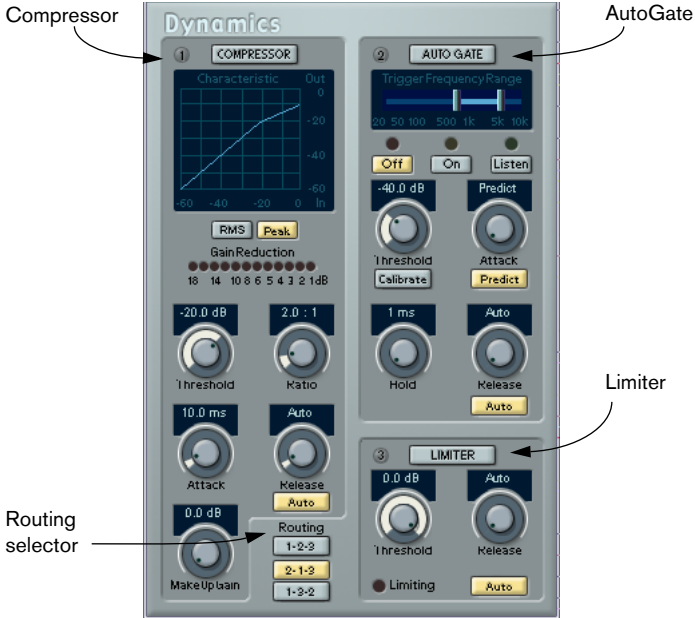
## About the Auto Threshold function

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.

# Dynamics



Dynamics is an advanced dynamics processor. It combines three separate processors: AutoGate, 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 by clicking on their labels. Activated processors have highlighted labels.

## The AutoGate 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. AutoGate offers all the features of a standard noise gate, plus some very useful additional features, such as auto-calibration of the threshold setting, a look-ahead predict function, and frequency selective triggering.

The available parameters are as follows:

Parameter	Values	Description
Threshold	-60 - 0dB	This setting determines the level where AutoGate is activated. Signal levels above the set threshold trigger the gate to open, and signal levels below the set threshold will close the gate.
Attack	0,1 -100 ms or “Predict mode”	This parameter sets the time it takes for the gate to open after being triggered. If the Predict button is activated, it will ensure that the gate will already be open when a signal above the threshold level is played back. AutoGate manages this by “looking ahead” in the audio material, checking for signals loud enough to pass the gate.
Hold	0 - 1000 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, AutoGate will find an optimal release setting, depending on the audio program material.

## Trigger Frequency Range

AutoGate has a feature that allows the gate to be triggered only by signals within a specified frequency range. This is a most useful feature because it lets you filter out parts of the signal that might otherwise trigger the gate in places you don't want it to, thus allowing more control over the gate function. The Trigger Frequency Range function is set using the control in the upper part of the AutoGate panel, and the buttons located below it.

The basic operation of the Trigger Frequency Range function is as follows:

1. While playing back audio, click the “Listen” button.  
You will now monitor the audio signal, and the gate will be bypassed.
2. While listening, drag the two handles in the Trigger Frequency Range display to set the frequency range you want to use to trigger the gate.  
You will hear the audio being filtered as you move the handles.
  - Dragging the left handle to the right will progressively cut frequencies starting from the low end of the frequency spectrum.
  - Dragging the right handle to the left will progressively cut frequencies starting from the high end of the frequency spectrum.



The frequency range between the two handles will be used to trigger the gate.

3. After setting the frequency range, click the “On” button.  
AutoGate will now use the selected frequency range as the trigger input.
4. To disable the Trigger Frequency Range function, click the “Off” button.  
AutoGate will now use the unfiltered audio signal as the trigger input.

## The Calibrate function

This function, activated by using the Calibrate button located below the Threshold knob, is used to automatically set the threshold level. It is especially useful for material with consistent inherent background noise, like tape hiss. This may most of the time be masked by the audio content, but becomes noticeable during silent passages.

Use it as follows:

1. Find a part of the audio material, preferably not too short, where only the background noise is heard.

If you can only find a short background noise section, try looping it.

2. Play it back, and click on the Calibrate button.

The button will blink for a few seconds, and then automatically set the threshold so that the noise will be silenced (gated) during passages where there is no other signal present.

## 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	Values	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 3 dB the input level increases, the output level will increase by only 1 dB.
Attack	0.1-100 ms	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 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.
MakeUp Gain	0 - 24dB	This parameter is used to compensate for output gain loss, caused by compression.
Compressor Mode	RMS/Peak	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.

## 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 totally avoid the possibility of the output level 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	Values	Description
Threshold	-12 - 0dB	This setting determines the maximum output level. Signal levels above the set threshold are affected, but signal levels below are left unaffected.
Release	10-1000ms or "Auto mode"	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 Routing section



In the Routing section 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. Beside each processor label is a number. These numbers are used to represent the signal flow options shown in the Routing section. There are three routing options:

- 1-2-3 (Compressor-Gate-Limit)
- 2-1-3 (Gate-Compressor-Limit)
- 1-3-2 (Compressor-Limit-Gate)

# Magneto



Magneto brings the positive qualities of analog recording to your digital system, by providing the following capabilities:

- Simulates “tape saturation” and “tape overdrive” in a very realistic manner.
- Adds warmth, punch, and brilliance to a sound.
- Allows you to emphasize the “small details” in the sound.
- Works great on bass and guitar recordings as well as on drums, including individual samples and drum loops.
- Makes sampled drums and percussion sound much more “natural” and “warm”.
- Removes the “hardness” otherwise associated with digital audio recording.

All this makes Magneto suitable for processing both single sounds and complete recordings. In other words; practically any recording that you want to make sound warmer or more “natural”.

The algorithm behind Magneto is based on extensive studies and measurements of analog tape recorders. Special care has been taken to transfer the results of these studies into the digital domain.

- If your audio material has been recorded digitally with Emphasis, it contains a disproportionate amount of high frequencies. This will disturb the audio analysis in Magneto.

We recommend that you convert such material (removing Emphasis) before processing it with Magneto.



## About the Drive parameter and Magneto output levels

- Magneto is different from analog tape recorders in one respect: On an analog tape machine, you will get a lower output level when overdriving the tape “too far”. This is known as the “saturation” effect. In Magneto, high Drive settings do not have this effect on the Output level.
- Magneto needs headroom to perform its “magic”. For this reason you may note a decrease in Output level (compared to the Input level) when using very low Drive parameter settings (when the onTape meter shows levels below approximately +10dB). Since low Drive settings is not a normal situation (since the plug-in then practically doesn't have any audible effect), this is not something you would normally encounter. However, if for some application a low Drive setting is required, you can compensate for the loss in level with the Output Level parameter, see later in this text.

## Metering Switch

Use the “Level” buttons to switch the meters between three modes:

- **Input**  
In this mode, the level of the input signal is shown. This should never exceed 0dB, as mentioned above and described in more detail below.
- **onTape**  
In this mode, the meters show an equivalent of the level recorded on the simulated “tape”. See the description of the Drive parameter for more details.
- **Output**  
This shows the output level for the entire plug-in. This should never exceed 0dB, see below.

## Clip LEDs

The Input and Output Clip LEDs, located on the corresponding “Level” meter buttons, show if the signal is too “hot” (clipping occurs) at the input or output. The advantage of these is that they indicate clipping regardless of the mode the meters are switched to.

## Input Level

This is used to make sure the input signal is strong enough, without exceeding full level (so that clipping is avoided).

- If your input is already normalized, or sufficiently hot, leave this knob at 0.0 dB.
- If you need to adjust the input level, switch the Level metering to Input. Then adjust the knob until the signal peaks are as close as possible to 0dB without ever exceeding that level!

## Output Level

- Under normal conditions, the Output Level control should be left at 0.0dB. The DSP algorithm in Magneto includes an “auto-gain” function which tries to keep the output level as close as possible to 0dBfs, at high Drive settings.
- At very low drive settings (if onTape metering indicates peak levels at 7dB or less – see the Drive parameter description for more info) you might need to amplify the signal using the Output Level control. However, always do this with the Level metering Output button activated, so that you can check that clipping doesn’t occur.
- At very high HF-Adjust settings, you might need to back off a bit on the Output level. Again, use Output metering to check.

## If “digital clipping” occurs

If clipping occurs, (if the sound is heavily distorted), start by switching to input metering and check the input levels. If the input levels seem OK, switch to Output metering and adjust the Output Level as needed.

## The main parameters

You can change the Magneto parameters in realtime – i.e. while the audio material is played back – and the changes take effect more or less immediately (depending on your system). This allows you to experiment to get a feeling for how the settings interact.

### Input level, Output Level, “Level” buttons and Meters

These are used to adjust the level throughout the signal chain as described on the previous pages.

### Drive

This is the main parameter. It is used to set the simulated analog tape “recording level”. The value corresponds to how far above normal working level (0dB) you want to “record” on the “analog tape”. For example, a setting of 7 means the “tape” is “overdriven” by 7dB.

The higher you set this, the more of the “tape saturation” effect you will get.

Please use the following guidelines:

- Start out with a Drive setting of 10dB. Then adjust to taste.
- The effect of this parameter varies drastically with the frequency content and other characteristics of the material. There is no “best setting” for all types of recordings.
- If the material you are processing is already compressed or has been recorded on analog tape, a high Drive setting is not recommended, since it will give the sound an unnatural character.
- When processing complete mixes, you will have to be more careful with the Drive settings than when processing individual recordings. If all you want is to add some “warmth” or “punch” to a complete mix, adjust the Drive setting carefully.
- Always use the onTape meter to check out the effect of the setting on the material. This meter has to go pretty far above the 0dB level for Magneto to have any audible effect on the sound. If the meter displays levels close to, or even below 0dB, you get no “overdrive” or tape saturation effect at all! If this occurs, you need to raise the Drive setting or adjust the input level.

## Characteristics

This affects the tonal characteristics of the “tape saturation” effect controlled by the Drive parameter, as described above.

## Tape Speed

This switches the tape simulation between 15 and 30 ips (inches per second) tape speed. There are slight differences in the harmonic character of the two. How much you will actually be able to hear of this difference depends on the frequency content of the material.

## HF-Adjust

Various types of tape, recording and playback equalizers and the general design of various tape machines has an overall impact on the character of the sound. This control is used to adjust the High frequency content of the material to simulate those differences. It also has an effect on the perceived “warmth” of the sound.

This parameter can be used to compensate for the loss in high frequency that the overdrive effect introduces. Unlike on a real tape recorder it can also be used to boost the high frequency contents, compared to the original!

# MIDI Gate



Gating, in its fundamental form, silences audio signals below a certain set threshold level. I.e. 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 (“out:”) 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.

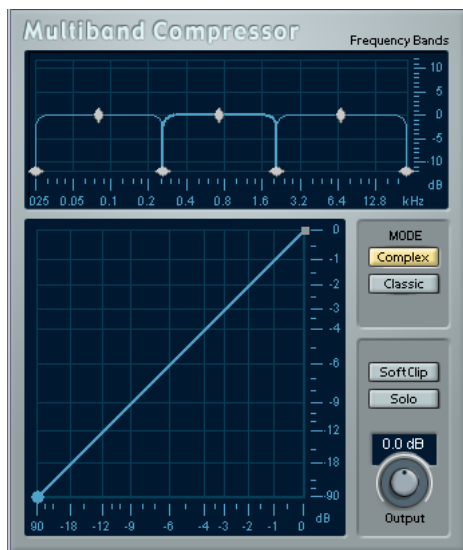
The following MIDI Gate parameters are available:

---

<b>Parameter</b>	<b>Description</b>
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 five 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 Characteristics editors. 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 is where you set the width of the frequency bands as well as their level before compression. Two value scales and a number of diamond-shaped handles are available. The vertical value scale to the right gives you a clue to the approximate input gain level of each frequency band.

The diamond-shaped handles provided in the Frequency Band editor can be dragged with the mouse. You use them to set the corner frequencies and the input gain levels for up to five frequency bands. The width of each frequency band can be adjusted by dragging horizontally.

The Level handles can be moved by dragging them up or down. If you click and hold on a handle, its current value is displayed. Depending on the handle type, corner frequency or level is shown.

- The diamond-shaped handles at the bottom are used to define the corner frequencies of the different frequency bands.
- By using the diamond-shaped handles on top of each frequency band you can cut or boost the input gain by +/- 12dB before compression.
- To reset a Level handle to 0 dB, hold down the [Shift] key on your computer keyboard and click on the handle.
- If you hold down the [Shift] key and click on the corner frequency handles, they will be set to the same bandwidth (in octaves). The exact bandwidth they will be set to is dependent on the number of bands currently used.

If you hold down [Ctrl]/[Command] and move a handle, the values will change in smaller steps.

## Adding and removing frequency bands

To add a frequency band, drag the leftmost or rightmost corner frequency handle towards the middle of the window, and a new band will automatically appear (given that you have less than the maximum number of five bands active). To remove a frequency band, drag the second leftmost or second rightmost handle out of the left or right edge of the window respectively.

## About the Frequency scale

The horizontal value scale below the Frequency band display indicates frequency. The maximum value on this scale corresponds to half the sample rate of the audio file used. Hence, if a 44.1 kHz soundfile is used, the highest frequency will be 22kHz.

---

**In the digital domain, only frequencies of up to half the sample rate used can be reproduced (Nyquist theorem). The values available in the Frequency band display do therefore depend on the sample rate of the audio material used.**

---



## The Solo button

The Solo button in the lower right part of the MultibandCompressor panel can be used to separately monitor each of the frequency bands. This function is useful both when editing bandwidth settings and compressor characteristics.

- To select another band while solo is active, click somewhere in the (dark) area of the frequency band that you wish to monitor.

## Using the Characteristics editor

By adding breakpoints and drawing curves you set the compressor characteristic. Before you start using the Characteristics editor, you have to select the frequency band you want to process. This is done in the Frequency Band editor by clicking in the area inside the frequency band.

- A selected band is highlighted for editing both in the Frequency Band and the Characteristics editors.  
If you select another frequency band, the previously edited band characteristic is still shown in the Characteristics window, but it is no longer highlighted or editable until you select it again.

## About breakpoints

- Clicking anywhere on the line will add a breakpoint.
- To remove a breakpoint, hold down [Shift] and click on it.
- The first breakpoint from which the line deviates from the straight diagonal will be the threshold point.
- Creating a curve in the area below the diagonal input/output line will cause compression. Compression decreases the output level in relation to the input level.
- Creating a curve in the area above the diagonal input/output line will cause expansion. Expansion increases the output level in relation to the input level.

## About the Compressor type (MODE)

- Classic mode works like a standard compressor with fixed attack and release parameters.
- Complex mode features a new compression approach with a program adaptive circuit. The program adaptive compression automatically optimizes parameters according to 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 +/- 12dB. If the SoftClip function (see below) is active, the Output dial instead controls the amount of soft clipping.

## **The SoftClip function**

The SoftClip function is positioned at the very last stage of the internal signal path, right after the Output dial. When active, it will ensure that the total output to Nuendo never exceeds 0dB. It works by clipping the signal gently, generating harmonics which add a warm, tube-like characteristic to the signal.

# VST Dynamics



The VST Dynamics plug-in is similar to the Dynamics plug-in (see [page 18](#)), but with the following differences:

- VST Dynamics has two additional modules: Auto Level and Soft Clip.
- The signal flow is fixed, in the order AutoGate-AutoLevel-Compressor-SoftClip-Limiter.

## Activating the individual processors

You activate the individual processors by clicking on their labels. Activated processors have highlighted labels. You can activate as many processors as you want, but remember that not all processors are designed to work together. For example, “Limit” and “SoftClip” are both designed to ensure that the output never exceeds 0dB, but achieves this in different ways. To have both of them activated would be unnecessary.

- To turn off all activated VST Dynamics processors, click the lit On button to the right in the panel. Clicking the button again activates the same configuration of processors.

## Auto Gate section

This is exactly the same section as the AutoGate in the Dynamics plug-in. See [page 19](#) for details.

## Auto Level section

Auto Level reduces signal level differences in audio material. It can be used to process recordings where the level unintentionally varies. It will boost low levels and attenuate high level audio signals. Only levels above the set threshold will be processed, so low level noise or rumble will not be boosted. If the input level is greater than 0dB, Auto Level will react very fast, because it “looks ahead” in the audio material for strong signal levels and can attenuate levels before they occur, thus reducing the risk of signal clipping. Auto Level has the following parameters:

Parameter	Description
Threshold	Only levels stronger than the set threshold will be processed.
Reaction time buttons (Slow, Mid, Fast)	Here you can set the amount of time it takes for Auto Level to adjust the gain. Set this according to whether the program level changes suddenly or over a length of time.

## Compressor section

This is exactly the same section as the Compressor section in the Dynamics plug-in. See [page 22](#) for details.

## Soft Clip section

Soft Clip is designed to ensure that the output level never exceeds 0dB, like a limiter. Soft Clip, however, acts differently compared to a conventional 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. Soft Clip is simplicity itself to use as it has no control parameters. The meter indicates the input signal level, and thus the amount of “softclipping”. Levels in the green area (weaker than -6dB) are unaffected, while levels in the yellow-orange-red area indicate the degree of “softclipping”. The deep red meter area to the right indicates input levels higher than 0dB.

- **Avoid feeding Soft Clip with excessively high signal levels as audible distortion may occur, although the output level will never exceed 0dB.**

## Limiter section

This is exactly the same section as the Limiter in the Dynamics plug-in. See [page 23](#) for details.

# Filter plug-ins

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

## 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.

### 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:

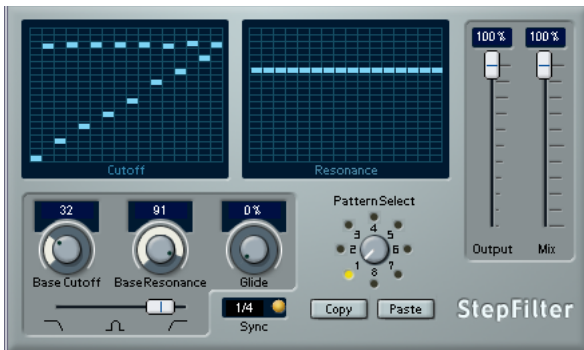
- By using the knobs.
- By clicking a value field and entering values numerically.
- By using the mouse to drag points in the EQ curve display window.  
By using this method, you control both the Gain and Frequency parameters simultaneously. The knobs turn accordingly when you drag points. In addition, 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 +20 dB)	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 +20 dB)	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 +20 dB)	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.

Parameter	Description
Output (-20 to +20 dB)	This parameter allows you to adjust the overall output level.
Left/Stereo/Right/Mono Modes	<p>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.</p> <p>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.</p> <p>Mono mode is automatically activated for mono signals and is otherwise unavailable.</p>

# StepFilter



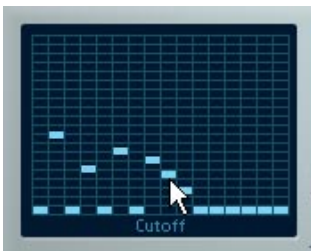
StepFilter is a pattern-controlled multimode filter that can create rhythmic, pulsating filter effects.

## 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.



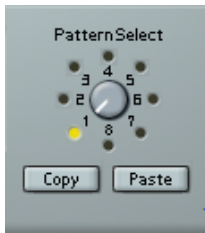
Setting filter cutoff values in the grid window.



- 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.

- Click the Copy button with the pattern you wish to copy selected, 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 1/1-1/32 (Straight, Triplet or Dotted)	This sets the pattern beat resolution, i.e. what note values the pattern will play in relation to the tempo.
Mix	Adjusts the mix between dry and processed signal.
Output	Sets the overall volume.

---

# 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).
- 24dB low pass, 18dB low pass, 12dB low pass, 6dB low pass, 12dB band pass and 12dB 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.	Chose 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...

---

## 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.

---

## X/Y Pad

---

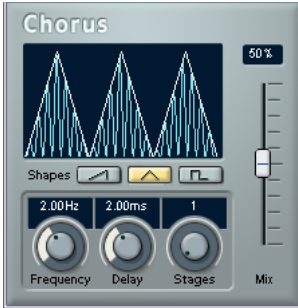
Parameter	Description
X Par	Sets the parameter to be modulated on the x axis of the XY Pad. All of Tonic’s parameters are available as destinations
Y Par	Sets the parameter to be modulated on the y axis of the XY Pad.
XY Pad	Use the mouse to control any two of Tonic’s parameters in combination. By moving 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.

---

# Modulation plug-ins

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

## Chorus



The Chorus plug-in adds short delays to the signal, and pitch modulates the delayed signals to produce a “doubling” effect.

The parameters are as follows:

Parameter	Description
Mix	Sets the level balance between the dry signal and the effect. If Chorus is used as a send effect, this should be set to maximum as you can control the dry/effect balance with the send.
Shapes	This sets the modulation waveform. Triangle produces smooth modulation, saw produces ramp shaped modulation and pulse waveform produces stepped modulation.
Frequency	This sets the modulation rate.
Delay	This controls the depth of the Chorus effect.
Stages	This adds one to three more delay taps, producing a thicker, multi-layered chorus effect.

- **Note that clicking and dragging in the display allows you to adjust the Frequency and Delay parameters at the same time!**

# Flanger



Flanger is a classic flanger effect with stereo enhancement.

The parameters are as follows:

---

Parameter	Description
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.
Output	Sets the overall volume.
Tempo sync on/off	The button above 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 - 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.
Tempo Sync knob	This is the note value multiplier (x1 to x10) for the flanger sweep when tempo sync is used.
Shape Sync knob	This changes the shape of the modulating waveform, altering the character of the flanger sweep.
Feedback	This determines the character of the flanger effect. Higher settings produce a more “metallic” sounding sweep.
Depth	This sets the depth of the modulation sweep.
Delay	This parameter affects the frequency range of the modulation sweep, by adjusting the initial delay time.
Stereo Basis	This sets the stereo width of the effect. 0% is mono, 50% original stereo, and 100% maximum stereo enhancement.

---

You can also change parameters in the graphic display. This works as follows:

- If tempo sync is on, you can set the base note value by clicking the waveform and dragging left and right.  
When tempo sync is off, this sets the Rate parameter.
- You can set the Depth parameter by clicking the waveform and dragging up and down.  
This means you can freely adjust Rate and Depth at the same time by clicking and dragging.
- By click-dragging the green/blue line in the display left or right you can change the Stereo Basis parameter.



# Metalizer

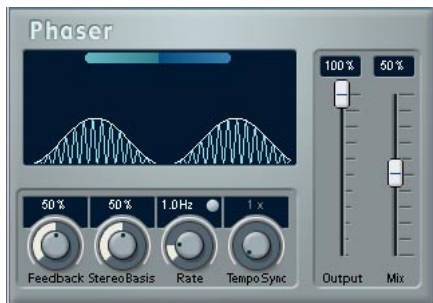


The Metalizer feeds the audio signal through a variable frequency filter, with tempo sync or time modulation and feedback control.

Parameter	Description
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.
Output	Sets the overall volume.
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.
Speed	If tempo sync is on, this is where you specify the base note value for tempo-syncing the effect (1/1 - 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.
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.
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.
Feedback	The higher the value, the more “metallic” the sound.

- **Note that clicking and dragging in the display allows you to adjust the Sharpness and Tone parameters at the same time!**

# Phaser



The Phaser plug-in produces the classic “swooshing” sound that characterizes phasing. It works by shifting the phase of the signal and adding it back to the original signal, causing partial cancellation of the frequency spectrum.

---

Parameter	Description
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.
Output	Sets the overall volume.
Tempo sync on/off	The button above 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 - 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.
Feedback	This sets the amount of feedback. A higher value produces a more pronounced effect.
Tempo Sync knob	This is the note value multiplier (x1 to x10) for the Phaser sweep when tempo sync is used.
Stereo Basis	This sets the stereo width of the effect. 0% is mono, 50% original stereo, and 100% maximum stereo enhancement.

---

You can also change parameters in the graphic display. This works as follows:

- If tempo sync is on, you can set the base note value by clicking the waveform and dragging left and right.  
When tempo sync is off, this sets the Rate parameter.
- You can set the Feedback parameter by clicking the waveform and dragging up and down.  
This means you can freely adjust the Rate and Feedback at the same time by clicking and dragging.
- By click-dragging the blue/green line in the display left or right you can change the Stereo Basis parameter.

# 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	LFO Amount controls how much the oscillator frequency is affected by the LFO.
Oscillator Env. Amount	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.
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).

<b>Parameter</b>	<b>Description</b>
LFO Waveform	Selects the LFO waveform; square, sine, saw or triangle.
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.
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.
Mix	Adjusts the mix between dry and processed signal.
Output	Sets the overall volume.

# 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 included presets provide good starting points for further tweaking of the numerous parameters.

The parameters are as follows:

Parameter	Description
Speed	This controls the speed of the Rotary in three steps: Stop/Slow/Fast.
MIDI Ctrl	Selects the MIDI continuous controller for the Speed parameter. See <a href="#">page 55</a> .
Mode	Selects whether the Slow/Fast speed setting is a switch (left button is lit), or a variable control (right button lit). 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.
Overdrive	Applies a soft overdrive or distortion.
Crossover Freq.	Sets the crossover frequency (200-3000Hz) between the low and high frequency loudspeakers.
Mic Angle	Sets the simulated microphone angle. 0 = mono, 180 = one mic on each side.
Mic Distance	Sets the simulated microphone distance from the speaker in inches.

<b>Parameter</b>	<b>Description</b>
Low Rotor Amp Mod.	Adjusts amplitude modulation depth.
Low Rotor Mix Level	Adjusts overall bass level.
Hi Rotor Amp Mod.	High rotor amplitude modulation.
Hi Rotor Freq. Mod.	High rotor frequency modulation.
Phasing	Adjusts the amount of phasing in the sound of the high rotor.
Hi Slow	Fine adjustment of the high rotor Slow speed.
Hi Rate	Fine adjustment of the high rotor acceleration time.
Hi Fast	Fine adjustment of the high rotor Fast speed.
Lo Slow	Fine adjustment of the low rotor Slow speed.
Lo Rate	Fine adjustment of the low rotor acceleration time.
Lo Fast	Fine adjustment of the low rotor Fast speed.
Output	Adjusts the overall output level.
Mix	Adjusts the mix between dry and processed signal.

## 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 (“out:”) 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.

# Symphonic



The Symphonic plug-in combines a stereo enhancer, an auto-panner synchronized to tempo and a chorus-type effect. For best results, apply the Symphonic effect to stereo signals.

The parameters are as follows:

Parameter	Description
Mix	Sets the level balance between the dry signal and the effect. If Symphonic is used as a send effect, this should be set to maximum as you can control the dry/effect balance with the send.
Tempo sync on/off	The button below the Temp sync knob is used to switch tempo sync on or off. The button is lit when tempo sync is on.
Tempo Sync pop-up	If tempo sync is on, this is where you specify the base note value for tempo syncing the auto-panning (1/1 - 1/32, straight, triplet or dotted).
Tempo Sync knob	This is the note value multiplier (x1 to x10), determining the timing of the auto-panning.
Delay	This determines the delay time and thus the character of the chorus effect, if activated.
Depth	This controls the depth of the chorus effect. If you only want to use Symphonic as an auto-panner or a stereo enhancer, set this to 0%.
Rate	This sets the modulation rate for the chorus effect, if activated.
Stereo Basis	When the Auto-panner is activated, this sets the stereo width of the panning. When the Auto-panner is deactivated (Tempo sync off), this determines the depth of the stereo enhancer effect. 0% is mono, 50% original stereo, and 100% maximum stereo enhancement.
Output	Adjusts the output level of the effect.



You can also change parameters in the graphic display. This works as follows:

- You can set the Rate parameter by clicking the waveform and dragging left and right.
- You can set the Depth parameter by clicking the waveform and dragging up and down.

This means you can freely adjust Rate and Depth at the same time by clicking and dragging.

- By click-dragging the green/blue line in the display left or right you can change the Stereo Basis parameter.

# 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
Mix	Sets the level balance between the dry signal and the effect.
Output	Adjusts the output level of the effect.
Tone	Sets the frequency (pitch) of the modulating oscillator (1 to 5000 Hz).
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.
Speed	If tempo sync is on, this is where you specify the base note value for tempo-syncing the effect (1/1 - 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.
On button	Turns modulation of the pitch parameter on or off.
Mono button	Governs whether the output will be stereo or mono.
Depth	Governs the depth of the pitch modulation.
Waveform buttons	Sets the pitch modulation waveform.

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

## Other plug-ins

This section contains descriptions of the plug-ins in the “Other” 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 different sounding result. Modes I and III are nastier and noisier, while modes II and IV are more subtle.
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.
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.
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.
Output	Governs the output level from the Bitcrusher. Drag the slider upwards to increase the level.

---

# 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
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.
Tempo sync on/off	The button above the Speed knob is used to switch tempo sync on (the button lights up) or off.
Speed	If tempo sync is on, this is where you specify the base note value for tempo-syncing the effect (1/1 - 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.
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").
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.

# Apogee 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, please refer to the chapter “Audio Effects” in the Operation Manual). You can use the UV22 HR plug-in for all dithering situations, except when working with surround audio. This is because the UV22 HR is a standard “stereo in” – “stereo out” plug-in (as opposed to the SurroundDither plug-in, see [page 96](#)).

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

Option	Description
Normal	Try this first, it is the most “all-round” setting.
Low	This applies a lower level of dither noise.
Autoblack	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.**

---

# Vocoder



The Vocoder can apply sound/voice characteristics taken from one signal source, called the “modulator” and apply this to another source, called the “carrier”. A typical application of a vocoder is to use a voice as a modulator and an instrument as a carrier, making the instrument “talk”. A vocoder works by dividing the source signal (modulator) into a number of frequency bands. The audio attributes of these frequency bands can then be used to modulate the carrier.

The Vocoder has a built-in carrier (basically a simple polyphonic synthesizer) but you can also use an external carrier, see [page 63](#).

## Setting up – using MIDI

In this mode, the Vocoder is set up slightly differently than other plug-in effects. This is because this setup requires both an audio signal (as the modulator source) and a MIDI input (to play the carrier) to function. To set up for using an external carrier, see [page 63](#).

To set up for use, proceed as follows:

1. Select a source for the modulator.  
The modulator source can be audio material from any audio track, or even a live audio input routed to an audio track (provided you have a low latency audio card).
- Good modulator source material are talking or singing voices or percussive sounds, e.g. drum loops.  
Static pads or soft ambient material are generally less appropriate for use as modulators, but there are no absolute rules as to what could be used as a modulator source.

2. Select the Vocoder as an insert effect for the audio channel with the modulator signal.
3. Make sure that the Vocoder Mode is set to “MIDI”.
4. Select a MIDI track.  
This can be an empty MIDI track, or a MIDI track containing data, it doesn't matter. However, if you wish to play the Vocoder in real-time – as opposed to having a recorded part playing it – the track has to have monitoring activated (or be record enabled) for the Vocoder to receive the MIDI output.
5. Select “Vocoder” from the MIDI “out:” pop-up menu for the MIDI track. The MIDI Output from the track is now routed to the Vocoder. There is an indicator on the Vocoder panel below the Mode switches that blinks when receiving MIDI.

That concludes setting up – you are now ready to start vocoding!

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

6. Make sure the MIDI track is record enabled and start playback.
7. Now play a few notes on your MIDI keyboard.  
As you can hear, the audio track material, or rather its formant characteristics, is now applied to the Vocoder's built-in sound source!

### **Setting up – using an external carrier**

There are two modes for using an external carrier:

- “Ext” mode is when the carrier and the modulator can be any two audio sources. The synth section is disabled and grayed out when this mode is selected. MIDI input and the Gap Thru Vocoder parameter are also disabled.
- “MIDI+Ext” mode mixes the audio carrier with the Vocoder's synth sound. This is described on [page 64](#).

To use an external carrier instead of the built-in synth (“Ext mode”), you set up as follows:

1. Create a Group channel from the Add Track submenu on the Project menu.
2. Open an audio file you wish to use as the carrier source and place it on an empty audio track.
3. Pan the audio channel full right in the Mixer or in the Inspector.
4. Route the output of the audio channel to the group.
5. Open an audio file you wish to use as the modulator source and place it on another empty audio track.  
Events on the two audio tracks (carrier and modulator) have to play back simultaneously for the Vocoder to work.
6. Pan the modulator audio channel full left in the Mixer or in the Inspector.
7. Route the output of the modulator audio channel to the group.
8. Select the Vocoder as an insert effect for the group channel.
9. Open the Vocoder panel and activate the “Ext.” Mode button.
10. If you now start playback, the carrier channel will be modulated by the modulator channel!  
Note that the synth section on the left half of the Vocoder panel and the “Gap Thru” parameter are now disabled.

### **Setting up – using an external carrier plus MIDI**

Setting up is the same as for using an external carrier, except that a MIDI track with its output routed to the Vocoder should also be present. The MIDI track can either play the Vocoder synth in real time or from prerecorded parts. Make sure that monitoring (or record enable) is activated for the track so that the Vocoder synth will receive MIDI played in real time.

- Set up as described, and activate “MIDI+Ext.” mode on the Vocoder panel.  
Any incoming MIDI now triggers the Vocoder synth, and the synths output is mixed with the audio carrier signal.



## Vocoder parameters

The Vocoder parameters govern the general sound quality of the vocoded sound.

---

Parameter	Description
Number of Bands	This governs how many frequency bands the modulator signal is divided into (2-24). Fewer bands will provide a thinner more resonant sound, whereas using more bands will make the sound fuller and more intelligible.
Bandwidth	This sets the bandwidth for the frequency bands, which affects the overall timbre. Very narrow bandwidth settings will produce a thin, whistle-like sound.
Min./Max. Freq.	These parameters set the minimum and maximum frequency limits for the Vocoder, respectively.
log/lin	Log/Lin controls how the frequency bands are spaced between the minimum and maximum frequencies. Log = equal spacing in octaves, Lin = equal spacing in Hz. This affects the basic timbre of the Vocoder.
Env.Speed	This determines the attack and release times of the Vocoder envelope. Fast settings will cause the modulator signal to trigger the Vocoder instantly, longer settings will gradually increase the attack/release times, providing a more subtle Vocoder effect. If set to "HOLD" the modulator is "frozen", and doesn't affect the carrier synth at all.
High Thru	This lets through high frequencies around the "S" frequency from the original input signal while notes are played.
Talk Thru	Adjusts the level of the original input signal passed to the Vocoder output while notes are played.
Gap Thru	Gap Thru (only available in MIDI mode) sets the level of the original input signal that is passed to the Vocoder output when no MIDI notes are being played. This lets you apply the Vocoder to a vocal track adding vocoded parts just where you want them.
Output	This controls the output level of the Vocoder.
Emphasis	This is a highpass filter, gradually cutting lower frequencies while letting high frequencies pass.

---

## Vocoder synth parameters

If the built-in synthesizer is the carrier, it is the sound of this instrument that the modulator source is applied to. The synth is polyphonic with up to 8 voices and features 2 oscillators per voice. The synth has the following parameters:

---

<b>Parameter</b>	<b>Description</b>
Voices	This sets the number of voices for the synth (1-8).
Fine Tune	Tunes the oscillators $\pm$ a semitone, in cents (100th of a semitone) steps.
Pitch Bend	Sets the up/down range of the Pitch Bend in semitone steps (1-12).
Noise	Adds white noise to the sound.
NoiseMod	This makes the oscillators modulate the noise level. This gives the noise a rasping sound, turning "sss" into "zzz".
P.Drift	Adds random pitch variation to the oscillators.
P.Glide	This makes the pitch glide between notes played. The parameter controls the time it takes for the pitch to glide from one note to the next.
P.Bright	This is a lowpass filter that can be used to soften the tone of the oscillators. It does not affect the white noise generator.
P.Detune	Allows you to detune one of the oscillators in cent steps.
LFO Rate	Controls the LFO rate (for vibrato).
Vibrato	Adds vibrato to the oscillators. This can also be controlled by using the Mod Wheel.

---

# Restoration

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 can not 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 44.1kHz).

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.

Parameter	Description
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 - 150Hz). 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



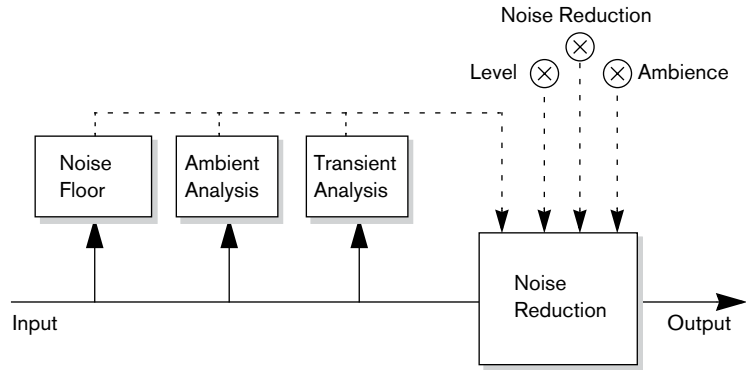
The DeNoiser plug-in lets you suppress noise without affecting the general sound quality. Or, in tech talk, the DeNoiser removes broad band noise from arbitrary audio material without leaving any “spectral finger print”. The algorithm that this plug-in is based on has the ability to track and adjust 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.



Parameter	Description
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 60Hz), 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.

## Reverb A



Reverb A is a reverb plug-in which provides smooth, dense reverb effects. Reverb A has the following parameters:

Parameter	Description
Mix	Sets the level balance between the dry signal and the effect (wet). If Reverb A is used as a send effect, this should be set to maximum wet, as you can control the dry/wet balance with the send.
Room Size	This setting determines the “size” of the simulated room environment.
Predelay	This parameter sets a delay between the direct sound and the reverb effect output. A short predelay before the reverb reduces reverb “clutter” which blurs the sound, and makes the reverb effect more natural-sounding.
Reverb Time	This parameter sets the length of the reverb time.
Filter HighCut	This filters out high frequencies for the reverb, which can make the reverb sound softer.
Filter LowCut	This filters out the lower frequencies for the reverb. It can be used to reduce low frequency “rumble”.

# Reverb B



The Reverb B provides reverb with low processor demands. It has the following parameters:

---

Parameter	Description
Mix	Sets the level balance between the dry signal and the effect. If Reverb B is used as a send effect, this should be set to maximum as you can control the dry/effect balance with the send.
Room Size	Governs the “size” of the simulated room environment.
Predelay	This parameter sets a delay between the direct sound and the reverb effect output. A short predelay before the reverb reduces reverb “clutter” which blurs the sound, and makes the reverb effect more natural-sounding.
Reverb Time	This parameter sets the length of the reverb effect.
Damp	This parameter “dampens” the higher frequencies, producing a rounder and smoother sounding reverb.

---

# 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
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.
High Shelf Gain	The amount of boost or cut for the high shelving filter.
Low Freq	Frequency at which the low shelving filter takes effect.
Low Shelf Gain	The amount of boost or cut for the low shelving filter.
PreDelay	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.
Time	Reverb Time in milliseconds.
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.
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).

<b>Parameter</b>	<b>Description</b>
Width	This controls the width of the stereo image. At 100%, you get full stereo reverb. At 0%, the reverb is all in mono.
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).
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.
High Damping Freq	This determines the frequency above which high frequency damping will occur.
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 Freq	This determines the frequency below which low damping will occur.
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.
Envelope 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.

<b>Parameter</b>	<b>Description</b>
Envelope Attack	The envelope settings in RoomWorks control how the reverb will follow the dynamics of the input signal in a fashion similar to a noise gate or downward expander. Attack determines how long 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.
Envelope 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.
Send	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.
Rotate	When active, the perspective of the room is shifted 90°.
Pos	The position 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.
Bal (Balance, only available for surround channels)	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°.

# 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.**

---

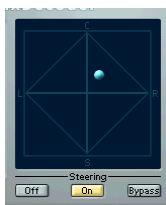
### Setting up

- 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/Decoder with the 5.0 Surround format” on [page 82](#).
- 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 0dB.

# Mixconvert



The Mixconvert plug-in is similar to the Mix6To2 and Mix8To2 plug-ins 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 [page 91](#)). You can use this fader to globally adjust gain by  $\pm 12$  dB 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  $-\infty$  and +6dB. For the LFE channel it can be changed between  $-\infty$  and +10dB, since in some mixes the LFE channel may be attenuated by 10dB (see [page 91](#)). 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 [page 92](#)).

## 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.**

---

### **Toggleing 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 120Hz, the filter slope is 12dB/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 for which keyboard shortcuts exist:

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

## Solo mode

Since there is no dedicated solo bus, all solos are in place, 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 appendix on [page 136](#) you find the tables listing all available and all unavailable combinations.

## Brief description of Mixconvert parameters

---

Parameter name	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. Click once for activating the 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

---

# 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 [page 61](#).

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 "all-round" 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.

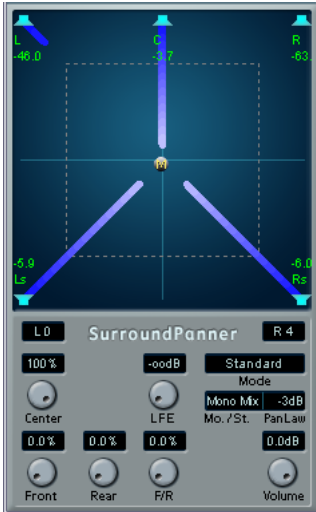


## An Example

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.

# SurroundPan



The SurroundPanner plug-in provides a graphical overview representing the speaker arrangement and the sound source, allowing you to dynamically position the audio in the surround field.

This plug-in is described in detail in the Operation Manual chapter “Surround Sound”.

# Tools

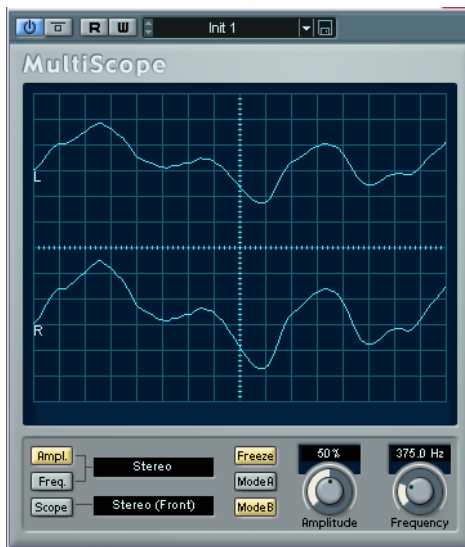
This section describes the plug-ins in the “Tools” category.

## 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.)

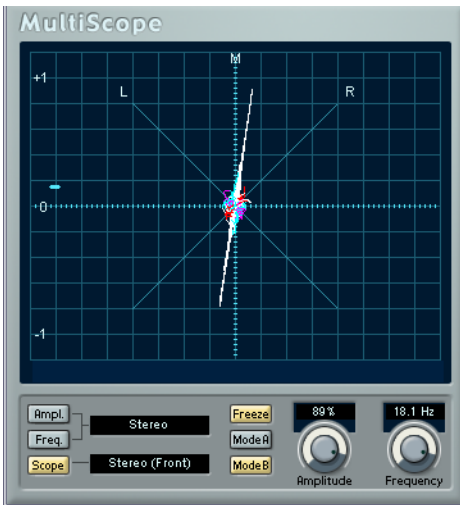
### Ampl (Oscilloscope) mode



- 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.

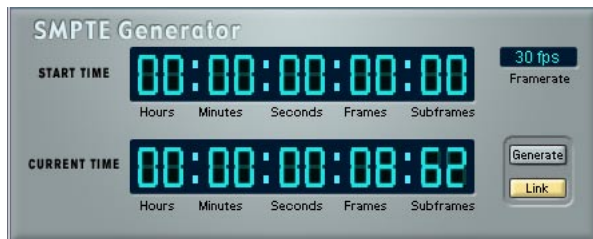
## Frequency Spectrum Analyzer



- 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.

# SMPTE Generator



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:

- **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.



# Test Generator



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 Test Generator 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 Test Generator 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.



**2**

**The included VST Instruments**

# A1 Synthesizer



The A1 is a dual oscillator software synthesizer with the following main features:

- The A1 is polyphonic with up to 16 voices.
- Multimode filter.  
Lowpass, bandpass, highpass and notch filter types are available.
- PWM (Pulse Width Modulation).
- FM (Frequency Modulation).
- Ring Modulator.
- Built-in stereo chorus/flanger effect.
- The A1 receives MIDI in Omni mode (on all MIDI channels).  
You don't need to select a MIDI channel to direct MIDI to the A1.
- The A1 responds to MIDI Controller messages.  
See [page 117](#).

# A1 Parameters

## Oscillator 1 and 2 section



This section contains parameters affecting the oscillators.

Parameter	Description
Octave	Clicking on the outer ring of the dial allows you to tune the oscillator in octave steps.
Semitone	Clicking on the inner ring of the dial allows you to tune the oscillator in semitone steps.
Detune	Tunes the oscillator in cent (100th of a semitone) steps.
Shape	This sets the waveform for the oscillator (sine, triangle, sawtooth or pulse).
PW	Sets the width of the waveform when a Pulse waveform is selected. Turning the dial clockwise gradually produces a narrower pulse waveform. Note that a PW setting of 100% will lead to complete cancellation of the waveform (i.e. silence), if no modulation (see PW Mod) is applied.
PW Mod	This parameter determines the amount of Pulse Width Modulation (PWM) by the LFO. Positive and negative values can be set. A Pulse waveform must be selected for PW Mod to function.
Pitch Mod	This parameter determines the amount of oscillator 1 pitch modulation (or vibrato) by the LFO. Positive and negative values can be set.
FM (Osc 1 only)	Governs the amount of frequency modulation. See <a href="#">page 117</a> .
FM Env (Osc 1 only)	This governs how much the Filter Envelope parameters affects the FM amount. Positive and negative values can be set. See <a href="#">page 117</a> .

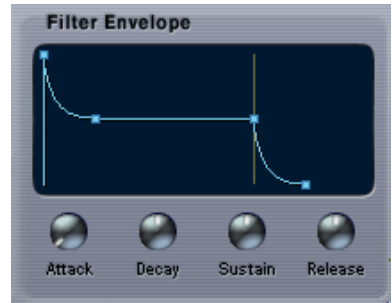
## LFO section



This section contains the LFO (Low Frequency Oscillator) parameters. LFOs are used to modulate parameters like pitch (vibrato) or the filter cutoff.

Parameter	Description
LFO Wave	This sets the LFO waveform for modulating parameters: Sine and triangle waves have a smooth waveform, suitable for normal vibrato. Saw produces a ramp up or down cycle. S&H produces stepped random modulation. Square waves produce cycles that abruptly change between two values. Random produces smooth random modulation.
LFO Sync	If this is activated, the LFO rate will be synchronized to the sequencer tempo in various bar/beat divisions that can be set with the LFO Speed parameter.
LFO Speed	Governs the modulation rate of the LFO.
LFO Speed (tempo sync on)	If the "LFO Sync" parameter is activated, the LFO rate will be synchronized to the sequencer tempo, according to the different beat divisions that can be specified here.

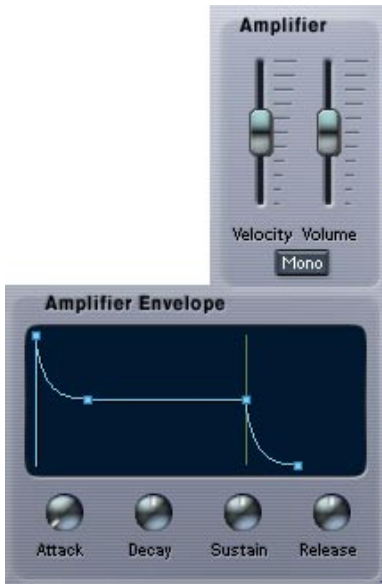
## Filter section



This section contains the filter parameters:

Parameter	Description
Filter type	Sets the filter type to either lowpass, highpass, bandpass or notch. The filter types are described on <a href="#">page 116</a> .
Cutoff	Controls the filter frequency or "cutoff". If a lowpass filter is used, it could be said to control the opening and closing of the filter, producing the classic "sweeping" synthesizer sound. How this parameter operates is governed by the filter mode (see <a href="#">page 116</a> ).
Resonance	The Resonance control for the filter. Raise this for a more pronounced filter sweep effect.
Drive	This parameter can overdrive the filter to produce distortion effects.
Filter Envelope	Controls how much the filter cutoff should be affected by the Filter Envelope. Negative values will invert the filter envelope settings.
Filter Velocity	Determines how the filter cutoff will be affected by velocity, i.e. how hard or soft you strike a key. Positive values will increase the cutoff frequency the harder you strike a key. Negative values will invert this relationship.
Filter Envelope Attack/Decay/Sustain/Release	The Filter Envelope Attack, Decay, Sustain and Release parameters. Use these parameters to determine how the filter cutoff should open and close with time, when a note is played. Values can be changed using the dials or by dragging the breakpoints in the graphic display.
Cutoff Mod	This controls how much the filter cutoff is modulated by the LFO (low frequency oscillator).
Keytrack	If this parameter is set to values over 0, the filter cutoff frequency will increase the further up on the keyboard you play. Negative values invert this relationship.

## Amplifier section



This section contains the Amplifier parameters:

Parameter	Description
Amplifier Attack/ Decay/Sustain/ Release	The Amplifier Attack, Decay, Sustain and Release parameters. Use these parameters to determine how the volume should change with time, when a note is played. Values can be changed using the dials or by dragging the breakpoints in the graphic display.
Velocity	This determines how much the Amplifier Envelope should be affected by velocity, i.e. by how hard or soft you strike a note on the keyboard.
Mono	When this is activated, the A1 will be monophonic, i.e. only play one voice at a time.



## The Chorus/Flanger section



Adding chorus will introduce a wide stereo effect and generally “fatten” sounds. With higher Feedback settings, more metallic sounding flanging effects are produced. The section contains the following parameters.

Parameter	Description
Speed	Controls the modulation rate of the effect.
Feedback	Increasing the Feedback parameter value results in a more pronounced sweeping metallic sound. Positive and negative feedback values can be set.
Depth	Controls the depth of the modulation.
Quad	Adds more delay taps, producing richer chorus/flanger effects.
On	This turns the chorus/flanger effect on or off.

## The Glide section



This section contains the glide parameters.

Parameter	Description
On	If set to “On” the pitch will glide up or down between notes played.
Speed	Controls the time it takes for the pitch to glide from one note to the next when using Glide.

## The Mixer section



This section controls the relative levels of Oscillator 1 and 2. Here you also set the levels of the Ring Modulator and Noise Generator outputs.

Parameter	Description
Osc 1	Sets the volume of oscillator 1.
Ring Mod	Controls the level of the ring modulator. See <a href="#">page 117</a> .
Osc 2	Sets the volume of oscillator 2.
Noise	Noise is commonly used to create wind and percussion type sounds. To hear the noise generator output on its own, turn down the osc 1 and 2 output in the Mixer.

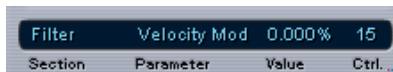
## Mod Wheel section



This section controls how the modulation wheel affects certain parameters. Positive and negative values can be set. For example, this can be used to set up so that moving the mod wheel gradually removes Filter Cutoff LFO modulation and instead introduces vibrato.

Parameter	Description
Pitch Mod	Governs the amount of LFO modulation of the oscillator frequency (vibrato) using the mod wheel.
Cutoff Mod	Governs the amount of LFO modulation of the Filter Cutoff parameter using the mod wheel.
Cutoff	Governs how much the mod wheel affects the Filter Cutoff frequency. Positive values raise the cutoff frequency when moving the mod wheel forward. Negative values invert this relationship.

## The Parameter display



The Parameter display is located in the middle of the A1 panel. The Parameter display shows information about any A1 parameter control when you point at it with the mouse. The information is presented in the following way (from left to right):

- "Section" shows what A1 section the parameter belongs to.
- "Parameter" shows the name of the parameter.
- "Value" shows the current value of the parameter.
- "Ctrl" shows the MIDI Controller assigned to the parameter, see [page 117](#).

## Setting the number of voices

A1 can have up to 16 voices, but you can freely set the number of voices for each program by changing the value in the “Voices” field.

## Keyboard section

The keyboard shows incoming MIDI note data as played by “invisible hands”. The keyboard can be “played” by clicking on it with mouse. Note that the velocity produced will be fixed and that you cannot record anything by clicking the keyboard.

- “Bend Range” is the only parameter that can be set in this section. A value of “1” equals a semitone bend range, “2” equals a range of two semitones etc.

## About the Filter types

The A1 features a multimode filter. The various filter modes are selected with the Filter Type buttons, and are as follows:

- **Lowpass**  
Lowpass filters let low frequencies pass and cuts out the high frequencies. This is the most commonly used filter type in analog synthesizers.
- **Bandpass**  
A bandpass filter cuts frequencies above and below the cutoff frequency, allowing a specific range of frequencies to pass while attenuating all others.
- **Highpass**  
A highpass filter is the opposite of a lowpass filter, cutting out the lower frequencies and letting the high frequencies pass.
- **Notch**  
A notch filter cuts out frequencies in a narrow midrange band, letting the frequencies below and above through.

## Filter Slope

You can also select between 12 or 24 dB filter slopes for all filter types. A 12 dB Lowpass filter leaves more of the harmonics in the filtered sound compared to a 24 dB Lowpass filter.

## Ring Modulator

Ring modulators basically multiply two audio signals together. In the A1, Oscillator 1 is multiplied with Oscillator 2 to produce sum and difference frequencies. Ring modulation can be used to create complex, bell-like sounds.

- To hear the output of Ring Modulator on its own, turn down the osc 1 and 2 output in the Mixer.
- If the oscillators are tuned to the same frequency, and no modulation is applied to either the oscillator 1 or 2 frequency, the ring modulated output will sound fairly similar to the “normal” sound of the oscillators. It is when the frequencies of osc 1 and osc 2 differ, that you get the more complex timbres associated with ring modulation.

## About FM

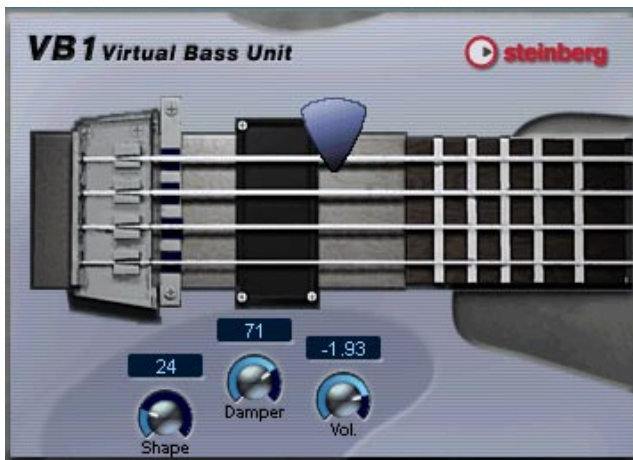
Frequency Modulation, or FM, is when the frequency of one oscillator (called the “carrier”) is modulated by the frequency of another oscillator (called the “modulator”). Using FM can produce a wide range of harmonic and non-harmonic timbres.

- In the A1, Oscillator 1 is the carrier and Oscillator 2 the modulator. When using FM, you should turn the master volume for Oscillator 2 down to zero in the Mixer to hear the “pure” sound of FM. The output of oscillator 2 is internally routed to oscillator 1 anyway when using FM.
- Changing the frequency of Oscillator 2 also changes the timbre of the FM sound.  
The waveform selected for both oscillators also affects the timbre.

## MIDI Controller Messages

The A1 responds to MIDI Controller Messages. All A1 parameters are assigned controller numbers. To find out what controller number is assigned to a parameter, simply point at the parameter and you can see the associated controller number assigned to it in the Parameter display (see [page 115](#)).

## VB-1 Bass Synth



The VB-1 is a virtual bass instrument built on real-time physical modelling principles. It has the following properties:

- VB-1 is polyphonic with up to 4 voices.
- VB-1 receives MIDI In Omni mode (on all MIDI channels). You don't need to select a MIDI channel to direct MIDI to the VB-1.
- VB-1 responds to the following MIDI messages: MIDI Note On/Off (velocity governs volume), Volume and Pan.

## VB-1 Parameters

---

<b>Parameter</b>	<b>Description</b>
Pick-up	To change the Pick-up position, click and drag the lower end of the Pick-up. Positioning the pick-up position towards the left produces a hollow sound that emphasizes the upper harmonics of the plucked string. When placed towards the right position, the tone is fuller and warmer.
Pick	This determines where along the length of the string the initial pluck is made. This controls the “roundness” of the tone, just like on a real bass. Click-drag the Pick to change position.
Shape	This knob selects the basic waveform used to drive the plucked string model. This parameter can drastically change the sound character. The control smoothly morphs through the waves. It is possible to create sounds that have no relation to a bass guitar with this control.
Volume	This knob regulates the VB-1 volume.
Damper	This parameter controls the length of time the string vibrates after being plucked.

---

# LM-7 Drum Machine

Volume and Tune faders  
(for each drum sound).

This adjusts the Pan (the position in the stereo image)  
for the individual drums. The setting is applied to the  
currently selected drum, indicated by a lit yellow LED  
over the Pad button.



This sets the  
global velocity  
sensitivity for  
LM-7.

Master Volume

Pad (one for each drum sound). Press to audition the drum sound  
assigned to the Pad, or to select a sound for adjusting pan.

The LM-7 is a 24-bit drum machine. It has the following properties:

- LM-7 is polyphonic with up to 12 voices.
- LM-7 receives MIDI in Omni mode (on all MIDI channels).  
You don't need to select a MIDI channel to direct MIDI to LM-7.
- LM-7 responds to the following MIDI messages:  
MIDI Note On/Off (velocity governs volume).



## LM-7 Parameters

---

Parameter	Description
Velocity	This sets the global velocity sensitivity for LM-7. The higher the value, the more sensitive LM-7 will be to incoming velocity data. If set to "0", the sounds will play back with a fixed velocity value.
Volume sliders	The volume sliders are used to adjust the volume for each individual drum sound.
Tune sliders	The tune sliders are used to tune each individual drum sound, up or down 1 octave.
Pad	The Pads are used for two things: To audition the individual drum sounds, and to select a sound for adjusting pan.
Panorama	This is used to position an individual sound in the stereo image. The setting applies to the currently selected sound, indicated by a lit yellow LED over the Pad button.

---

## Drum sounds

LM-7 comes with six sets of drum sounds. "Compressor", "909" and "Percussion" are loaded as the default sets when launching LM-7. "Modulation", "Fusion" and "DrumNbass" can be loaded by selecting "Load Bank" from the File menu and opening the `lm7_second_set.fxb` file (which is located in the `Vstplugins/Drums` subfolder).

- You switch between the three loaded sets by using the pop-up menu (just like you switch between effect programs).

## MIDI note mapping

The table below shows how the drum sounds are assigned to note values on your MIDI keyboard. The mapping is GM compatible:

<b>Drum sound</b>	<b>Note</b>	<b>Comment</b>
Bd	C1	
Rim	C#1	Compressor only.
Snare	D1	
Clap	D#1	909 only.
Hi-Hat	F#1	
O-Hi-Hat	A#1	
Tom 1	A1	
Tom 2	B2	
Tom 3	D2	
Crash	C#2	
Ride	D#2	Compressor only.
Tambourine	F#2	Percussion only.
Cowbell	G#2	Percussion only.
Hi Bongo	C3	Percussion only.
Lo Bongo	C3#	Percussion only.
Conga Mute	D3	Percussion only.
Conga Open	D#3	Percussion only.
Conga Lo	E3	Percussion only.
Timbale Lo	G3	Percussion only.
Timbale Hi	G#3	Percussion only.
Cabasa	A3	Percussion only.

## Embracer – Surround Pad Synthesizer

Embracer is a simple but powerful polyphonic synthesizer designed entirely for producing pads and accompaniment sounds. With its easy-to-use envelope and tone controls, it gives you fast access to the sounds you need without having to search through thousands of presets. However, the most powerful feature of Embracer is its surround output. With a single switch, you can turn the instrument from stereo to surround and the width control allows you to spread your pad sound anywhere from mono to stereo to full 360° surround. The unique “eye” controller gives you an exact idea of how the sound will be placed in a mix.

If you’ve never worked with a surround system before, now is the time to start exploring these possibilities.



The Embracer Surround Pad Synthesizer has the following properties:

- Embracer is a Polyphonic surround pad synthesizer.
- 2 oscillators with 12 waveforms.
- Independent envelope and tone controls.
- Stereo and surround outputs.
- Up to 32 voices of polyphony per instance.

- Dynamic width control for exciting 3D sounds.
- Unique “eye” controller for simultaneous tone and width control.
- Full MIDI control implementation.

## Osc 1 and 2

Parameter	Description
Wave	Selects the waveform for each oscillator. Available waveforms are: Carpet, DigiPad, Choir, Ensemble, Metal Phaze, Phase Strings, Sing Sing, Soft Wave, Spit Strynx, Stepfloor, Submerged, Wave Bellz. Note: If you want to use only one oscillator, set the waveform to OFF. In this case only one voice per key will be used.
Tone	Embracer offers a high pass and low pass filter for each oscillator. Both filters are controlled via a single Tone knob. In the 50% center position, the signal will not be filtered. Reducing the tone value adds low pass filtering. Values above 50% add high pass filtering. This parameter can also be controlled by the “eye” controller.
Width	Controls the spatial spread of the signal. A value of 0% puts the signal mono into the center position. In stereo mode, a value of 100% results in a maximum stereo width. In surround mode, a value of 100% creates a full 360° surround image. The width parameter can be controlled by a variety of modulation sources, as well as by the “eye” controller.
Coarse (Oscillator 2 only)	Changes the pitch in semitones. Maximum range is +1/24 semitones = 2 octaves.
Fine (Oscillator only)	Changes the pitch in fine steps with a range of up to +/- 50 cents. Note: If you want to create a slight detune effect between the oscillators, make sure to set the master tune parameter to a negative value of the same amount to keep the instrument in tune.

## Envelope and Level

---

Parameter	Description
Attack	Controls the attack time of each oscillator. Higher values create slower attacks.
Attack Vel	Sets the amount of velocity control of the attack time. Higher values increase the velocity sensitivity.
Level	Controls the oscillator output level.
Level Vel	Sets the amount of velocity control of the oscillator level. Higher values increase the velocity sensitivity.

---

## Master

---

Parameter	Description
Release	Controls the overall release time of the volume envelope. Higher values result in longer release times.
Mode	Sets the output mode of Embracer. You can choose between "Stereo" and "Surround". In Stereo Mode, Embracer has one stereo output in the VST Mixer. In Surround Mode, Embracer has either a quadraphonic 4-channel output or two independent stereo outputs in the VST Mixer. See below for more details on using Embracer in a surround mixer setup.
Width Ctr	Use this parameter to select a modulation source for the width parameter. Available sources are: Mod Wheel, Aftertouch, Velocity and Envelope. Both oscillators are controlled simultaneously. However, modulation depth is controlled independently by the respective width parameter of each oscillator.
Max Poly	Sets the total number of voices available. Each oscillator uses one voice per note played. Thus, a two-oscillator sound with 8 voices results in 4-voice polyphony. The default value for Max Poly is 16.
Fine Tune	Use this to adjust the pitch of the whole instrument. Range is +/- 50 Cents. Use Fine Tune in combination with the Fine Tune parameter of OSC 2 to create smooth detune effects.
Master Out	Sets the overall output volume of the instrument.

---

## The “Eye”

The Embracer’s unique “Eye” controller offers a creative new way of controlling the sound’s overall character and shape. This controller gives you access to several parameters at the same time.

For each oscillator, there is a circle representing the tone and width of the sound. Click and drag the corresponding circle to change its shape. There are also two (numbered) oscillator handles. You can drag these vertically to change the tone or horizontally to change the width of the respective oscillator. When you drag a handle, the respective Tone and Width knobs of the oscillator are adjusted accordingly. Play a note while editing to hear the effect.

The “eye” can not only be used as a controller for the tone and width parameters, but also works as a surround scope for monitoring the spatial integration of the current sound. The display represents the sound’s position in the stereo or surround sound field. In stereo mode, the sound position is shown only in the upper half of the display and represents the front part of the sound field. In surround mode, the sound position is shown in the upper and lower half of the display and represents the front and rear part of the sound field.

- **You can use Embracer’s automation feature to record the movements of the mouse within the “eye” controller!**

## Using Embracer in Surround Mode

When you want to enjoy Embracer in 3D, set it up in surround mode and listen to it on a surround system. Let's assume you have a surround monitoring system set up with your VST mixer and your VST output connections are properly set up.

1. Open an instance of Embracer in the VST instruments rack and set it to surround mode.
2. When you open the VST mixer you will see two separate stereo channels for the Embracer. The first is titled "Embracer" and the second "Embracer rear".
3. Assign both channel outputs to the surround output bus.  
The two channel strips will now show independent surround panners. By default, the first output pair is assigned to the front left and right channels and the second output pair to the rear left and right channels. The surround width can be controlled with the "width" parameter.
4. Double-click on the surround panner to open its control panel. Set the "Mono/Stereo" parameter to either "Y-Mirror", "X-mirror" or "XY-mirror". You can now freely adjust the surround panning to your taste.
5. If your surround configuration includes a center or LFE channel, you can also add some of Embracer's signal to the center or LFE channels. Feel free to experiment to find out what works best in a given project and mix.

# Monologue – Monophonic Analog Modeling Synthesizer

Monologue is a monophonic analog synthesizer based on physical modeling technology. It offers full, rich and colorful sounds without consuming a lot of CPU power. The Monologue synthesizer is the perfect tool for bass, lead and sequenced sounds.



The Monophonic Analog Modeling Synthesizer has the following properties:

- 2 oscillators with sawtooth, square and triangle waveforms.
- Monologue has an additional noise generator for white noise.
- Monologue has two filters: a high pass filter and a versatile multimode filter.
- Monologue has a single LFO.
- Monologue has 4-stage ADSR mod and amp envelopes.



- Monologue has an effects section with chorus, phaser, and flanger effects, plus separate delay and overdrive units.
- Monologue has a X/Y matrix pad for additional realtime modulation with access to all Monologue parameters.

## Osc 1 and 2

Parameter	Description
Waveform (pop-up menu)	This is where you select the waveform: Saw, Square and Sub for oscillator 1 and Saw, Square and Triangle for Oscillator 2.
Coarse	Sets the coarse pitch in semitones. The available range is +/- one octave.
Fine	Allows you to fine-tune the pitch in cent increments. The available range is +/- 50 Cents.
Depth	Controls the pitch modulation depth for the mod source defined in the "mod src" field. The available range is +/- one octave.
Mod Src	Defines the pitch modulation source. Available sources are: Modwheel, Aftertouch, Pitchbend, Velocity, LFO and Mod Env.
PWM (OSC2 only)	Controls the pulse width of the square wave. In the center position, pulse width is 50/50. Turning the PWM knob clockwise or counter clockwise creates a positive or negative pulse, respectively.
Sync (OSC2 only)	Activating the sync button synchronizes the pitch of oscillator 2 to the pitch of oscillator 1. When this is active, changing or modulating the pitch of oscillator 2 will change the tone and not the pitch. For the typical sync sound, turn osc 1 down in the mix and use osc 2 only.

## Mix

Parameter	Description
Osc 1	Sets the pre-filter level for oscillator 1.
Noise	Sets the pre-filter noise level.
Osc 2	Sets the pre-filter level for oscillator 2.

## 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.
High Pass	Sets the cutoff frequency of the additional high-pass filter.
Res	Changes the resonance of the multi-mode filter. Full resonance puts the filter into self-oscillation.
Key Track	Determines the amount of key tracking applied to the filter cutoff frequency. The available range is 0-100%. A range of 100% tunes the filter cutoff frequency to the keyboards pitch 1:1.
Mod Src (A+B)	Defines the filter modulation source. The available sources are: Modwheel, Aftertouch, Pitchbend, Velocity, LFO, and Mod Env.
Depth (A+B)	Controls the filter modulation depth for the mod source set in the "mod src" field.

## Envelope

Parameter	Description
A – (Attack)	Sets the attack time.
D – (Decay)	Sets the decay time.
S – (Sustain)	Sets the sustain level.
R – (Release)	Sets the release time.
Mod Src (A+B)	Defines the envelope modulation source. You can select: Modwheel, Aftertouch, Pitchbend, Velocity, LFO and Mod Env.
Depth (A+B)	Controls the envelope modulation depth for the mod source defined in the "mod src" field.

## LFO

---

<b>Parameter</b>	<b>Description</b>
Waveform (pop-up menu)	Here, you can select the waveform for the low frequency oscillator. Available waveforms are: Triangle, Square, Sawtooth, Sample & Hold and Random.
Rate	Adjusts the frequency of the LFO, thus changing the rate of the modulation. Depending on the LFO sync parameter, you can edit the rate in Hertz or in note values.
Sync	When "Sync" is "on" the LFO speed will be synchronized to the sequencer's tempo. This also affects the LFO rate format.
Mod Src	Defines the LFO modulation source. Available sources are: Modwheel, Aftertouch, Pitchbend, Velocity, LFO and Mod Env.
Depth	Controls the LFO modulation depth for the mod source defined in the "mod src" field.

---

## X/Y Pad

---

<b>Parameter</b>	<b>Description</b>
X Par	Sets the parameter to be modulated on the x axis of the XY Pad. All of Monologue'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 Monologue'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.

---

## Effects

---

<b>Parameter</b>	<b>Description</b>
FX Type (pop-up menu)	Selects the effect type for Monologue's pitch effects. The available types are Chorus, Flanger and Phaser.
Rate	Use this to adjust the rate of the effect modulation.
Depth	Use this to adjust the depth of the effect modulation.
FBK	Controls the feedback of the effect.

---

<b>Parameter</b>	<b>Description</b>
Mix	Controls the balance between dry and wet (effect) signal. Set to 0, the effect will be off. Set to 50, the balance between dry and wet signal is 50/50.
Overdrive	Controls the amount of overdrive (distortion) added to the signal. A slight amount of overdrive will create punch and bottom. Higher amounts will add distortion.
Delay	Sets the delay time in musical values. The delay effect is always in sync with the song tempo.
Spread	Controls the stereo spread of the delay signal. If you set this to 0, the delay will be centered mono. Higher amounts of spread will shift the left and right delay channels. If you set this to 100, the delays will "ping-pong" between the left and right channels at an even rate.
Tone	Adds a low pass filter to the delay. Increasing "tone" will make every delay repetition darker in tone.
FBK	Controls the amount of feedback of the delay. High feedback levels will create infinite delays. Use this parameter with caution.
Mix	Controls the balance between dry and wet (effect) signal. Set to 0, the effect will be off. Set to 50, the balance between dry and wet signal is 50/50.

## Master

---

Parameter	Description
Glide Mode	The available modes are: "held", "on" and "off". With "held" selected, a glide effect only occurs for notes played legato.
Rate	Controls the glide rate – the time it takes for a note to reach its destination pitch.
PB Range	Controls the range of a pitch bend midi controller. Range can be set between 1 and 24 semitones for a total of two octaves.
Env Trigger	When set to "Multi", each keystroke will re-trigger the envelopes. When set to "single", legato notes will not retrigger the envelopes, effectively holding the envelopes on the sustain level until all keys are released before a new note is triggered.
Note Priority	Defines which note is played when multiple keys are held. Options are: First, Lowest, Highest, and Last
Oct	Controls the master pitch of Monologue in octave steps. Range is +/- 4 octaves.
Master Out	Controls the master output level that is sent to the VST mixer. Use it to adjust the balance between different presets. Use the VST mixer channel volume to control or automate the Monologue master volume.
Keyboard	Pressing the "keyboard" button will reveal a six octave virtual keyboard. Pressing the "keyboard" button again will hide the keyboard and display the master section again.

---



## **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.	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
Input Config.											
<b>Mono</b>	S	S	S	S	S	S	S	S	S	S	S
<b>Stereo</b>	S	S	S	S	S	S	S	S	S	S	S
<b>LRS</b>	M	M	M	M	M	M	M	M	-	-	-
<b>LRS+Lfe</b>	M	M	M	M	M	M	M	M	-	-	-
<b>LRC</b>	M	M	M	M	M	M	M	M	-	-	-
<b>LRC+Lfe</b>	M	M	M	M	M	M	M	M	-	-	-
<b>LRCS</b>	M	M	M	M	M	M	M	M	-	-	-
<b>LCRS+Lfe</b>	M	M	M	M	M	M	M	M	-	-	-
<b>Quadro</b>	M	M	M	M	M	M	M	M	-	-	-
<b>Quadro+Lfe</b>	M	M	M	M	M	M	M	M	-	-	-
<b>5.0</b>	M	M	M	M	M	M	M	M	-	-	-
<b>5.1</b>	M	M	M	M	M	M	M	M	-	-	-
<b>6.0 Cine</b>	M	M	M	M	M	M	M	M	-	-	-
<b>6.0 Music</b>	M	M	M	M	M	M	M	M	-	-	-



**Index**

## A

A1 Synthesizer 108  
Apogee UV22 HR 61

## B

Bitcrusher 59

## C

Chopper 60  
Chorus 46

## D

DaTube 10  
DeClicker 67  
DeNoiser 70  
Dither 61  
DoubleDelay 7  
Dynamics 18

## E

Embracer Synthesizer 123

## F

Flanger 47

## L

LFE channel 91  
LM-7 120

## M

Magneto 24  
Matrix Decoder 80  
Metalizer 49  
MIDI Gate 29  
Mix6To2 84  
Mixconvert 86  
ModDelay 9  
Monologue Synthesizer 128  
MultibandCompressor 31

## O

Overdrive 11

## P

Phase shift  
    Mixconvert  
        Phase 89  
Phaser 50

## Q

Q 37  
QuadraFuzz 12

## R

Reverb A 75  
Reverb B 76  
Ringmodulator 52  
Rotary 54

## S

SMPTE Generator 75  
SPL DeEsser 16  
StepFilter 40  
SurroundDither 95  
SurroundPan 98  
Symphonic 56

## T

Tonic 43  
Tranceformer 58

## U

UV22 HR 61

## V

VB-1 118  
Vocoder 62  
VST Dynamics 35