Audio Effects and VST Instruments

NUENDO 2.0

media production system

PC VERSION

Built for Mac OS X

steinberg

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Table of Contents

1

The included effect plug-ins

Introduction

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

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

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	Values	Description
Mix	0-100%	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 buttons are gray) the delay time can be set freely with the Delay Time knobs, without sync to tempo.
Delay Time 1	1/1 - 1/32, 1/1 - 1/32 Triplet, 1/1 - 1/32 Dotted or 0-9999ms	This is where you specify the base note value for the delay if tempo sync is on. If tempo sync is off, it sets the delay time in milliseconds.
Delay Time 2	As above	As above.

Parameter	Values	Description
Feedback	0-100%	This sets the number of repeats for both delays.
Tempo Sync 1	x1 to x10	The note value multiplier for the first delay unit.
Tempo Sync 2	x1 to x10	As above, but for the second delay unit.
Pan1	-100 to 100%	This sets the stereo position for the first delay.
Pan2	-100 to 100%	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	Values	Description
Mix	0-100%	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 (gray button) the delay time can be set freely with the Delay Time knob, without sync to tempo.
Feedback	0-100%	This sets the number of repeats for the delay.
Delay Time	1/1 - 1/32, 1/1 - 1/32 Triplet, 1/1 - 1/32 Dotted or 0-9999ms	This is where you specify the base note value for the delay if tempo sync is on. If tempo sync is off, it sets the delay time in milliseconds.
Tempo Sync knob	x1 to x10	This is the note value multiplier for the delay when tempo sync is used.
DelayMod.	0-100%	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	Values	Description
Drive	0 - 100%	Regulates the pre-gain of the “amplifier”. Use high values if you want an overdriven sound just on the verge of distortion.
Balance	0 - 100%	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	-∞ - 0.000	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	Values	Description
Input	-15 to +15dB	Sets the input level.
Output	-15 to +15dB	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	On/Off	Simulates the sound of a speaker cabinet.
Factory Styles	Warm, Chordy, Magic OD, Fat Drive, Woody, Bluesy	Select one of six presets, which can be used as they are or as a basis for further “tweaking”.
Bass	-15 to +15dB	Tone control for the low frequencies.
Mid	-15 to +15dB	Tone control for the mid frequencies.
Hi	-15 to +15dB	Tone control for the high frequencies.
Drive	0-100%	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:

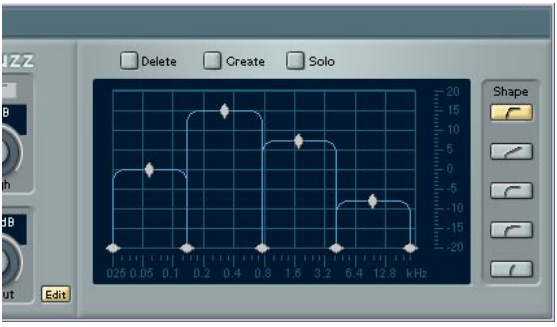
- The signal volume control *before* distortion.
  You can use the Gain control on the left side of the QuadraFuzz main window to control the overall input level of the signal that is fed into the distortion stage. The signal is split up into four frequency bands in the editor window, with adjustable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

Parameter	Description
Gain dial	This dial can be found in the lower left corner of the QudraFuzz 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.

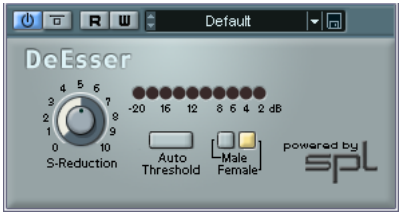
The parameters in the editor window are as follows:

Parameter	Description
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	Values	Description
S-Reduction	0 - 10	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	On/Off	See separate description below.
Male/Female	On/Off	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.

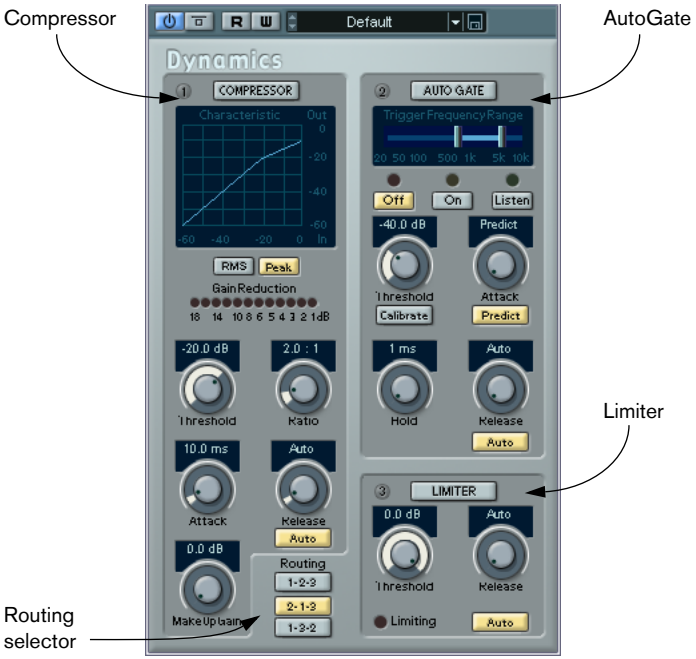
About the Auto Threshold function

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 is blue) 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 20 dB 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

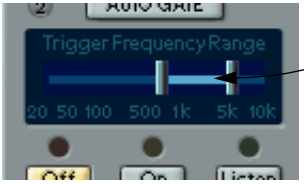
AutoGate has a feature that allows the gate to be triggered only by signals within a specified frequency range. This is a most useful feature because it lets you filter out parts of the signal that might otherwise trigger the gate in places you don't want it to, thus allowing more control over the gate 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.

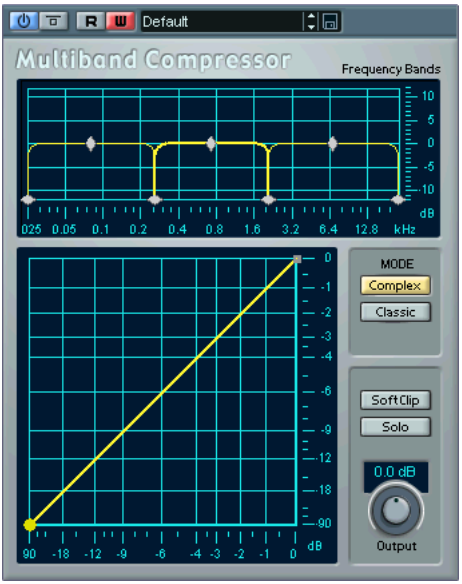
Make sure the MIDI track is selected and start playback.

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:

Parameter	Values	Description
Attack	0 - 500	This is used for determining how long it should take for the Gate to open after receiving a signal that triggers it.
Hold	0 - 3.000	Regulates how long the Gate remains open after a Note On or Note Off message (see Hold Mode below).
Release	0 - 3.000	This determines how long it takes for the Gate to close (in addition to the value set with the Hold parameter).
Note To Attack	-100 - <Off> - +100	The value you specify here determines how and to which extent the velocity values of the MIDI notes should affect the Attack. If this is set to a positive value, the Attack time will increase with higher note velocities. Negative values will give shorter Attack times with higher velocities. If you do not wish to use this parameter, set it to the <Off> position.
Note To Release	-100 - <Off> - +100	The value you specify here determines how and to which extent the velocity values of the MIDI notes should affect the Release. Positive values will increase the Release time and negative values will decrease the Release time. If you do not wish to use this parameter, set it to the <Off> position.
Velocity To VCA	0 - 127	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	Note-On/ Note-Off	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] (Win) or [Command] (Mac) 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 22 kHz.

❒ **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 +/- 12 dB. 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 0 dB. 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:

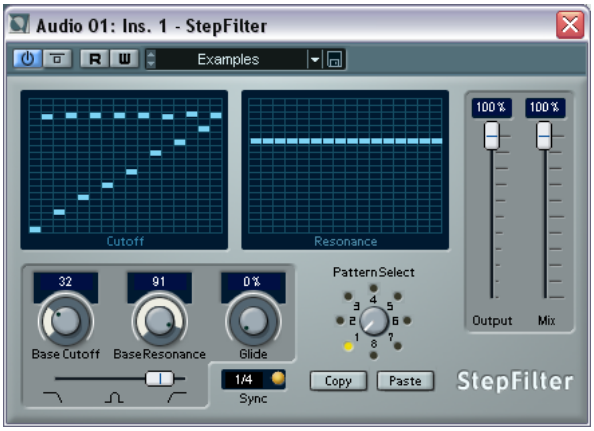
- By using the knobs.
- By clicking a value field and entering values numerically.
- By using the mouse to drag points in the EQ curve display window.
  By using this method, you control both the Gain and Frequency parameters simultaneously. The knobs turn accordingly when you drag points. In addition, 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.
Output (-20 to +20 dB)	This parameter allows you to adjust the overall output level.
Left/Stereo/Right/Mono Modes	For stereo signals you can set independent curves for the left and right channels by clicking the corresponding button. If the Stereo mode is activated, the curve will be applied to both channels. When channel independent curves have been set, the left/right channel curves will be colored green and red, respectively. The currently non-selected channel is shown with a dotted curve. If you activate Stereo mode after independent curves have been set, the currently active curve will be applied to both channels. Mono mode is automatically activated for mono signals and is otherwise unavailable.

StepFilter



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

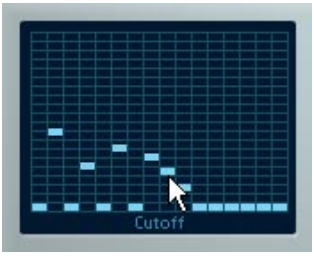
General operation

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

Setting step values

- Setting step values is done by clicking in the pattern grid windows.

- Individual step entries can be freely dragged up or down the vertical axis, or directly set by clicking in an empty grid box. By click-dragging left or right consecutive step entries will be set to the pointer position.



Setting filter cutoff values in the grid window.

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

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

Selecting new patterns

- Created patterns are saved with the song, 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.

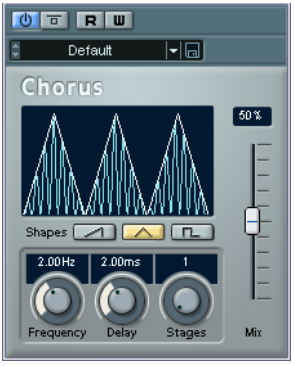
StepFilter parameters

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.

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	Values	Description
Mix	0-100%	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	Saw/ Triangle/ Pulse	This sets the modulation waveform. Triangle produces smooth modulation, saw produces ramp shaped modulation and pulse waveform produces stepped modulation.
Frequency	0-5Hz	This sets the modulation rate.
Delay	0-5ms	This controls the depth of the Chorus effect.
Stages	1-4	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	Values	Description
Mix	0-100%	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	0-100%	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 yellow when tempo sync is on, and gray when it is off.
Rate	1/1 - 1/32, 1/1 -1/32 Triplet, 1/1 -1/32 Dotted or 0 - 5Hz	If tempo sync is on, this is where you specify the base note value for tempo syncing the flanger sweep. If tempo sync is off, the sweep rate can be set freely with the Rate knob, without sync to tempo.
Tempo Sync knob	x1 to x10	This is the note value multiplier for the flanger sweep when tempo sync is used.
Shape Sync knob	1-16	This changes the shape of the modulating waveform, altering the character of the flanger sweep.
Feedback	0-100%	This determines the character of the flanger effect. Higher settings produce a more “metallic” sounding sweep.
Depth	0-100%	This sets the depth of the modulation sweep.

Parameter	Values	Description
Delay	0-100ms	This parameter affects the frequency range of the modulation sweep, by adjusting the initial delay time.
Stereo Basis	0-100%	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	Values	Description
Mix	0-100%	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	0-100%	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 yellow when tempo sync is on, and gray when it is off.
Speed	1/1 - 1/32, 1/1 - 1/32 Triplet, 1/1 - 1/32 Dotted or 0 - 10Hz	If tempo sync is on, this is where you specify the base note value for tempo-syncing the effect. 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	On/Off	Turns filter modulation on and off. When turned off, the Metalizer will work as a static filter.
Mono button	On/Off	Determines whether the output of the Metalizer will be in stereo or mono.
Sharpness	0-100%	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.

Parameter	Values	Description
Tone	0-100%	Governs the feedback frequency. The effect of this will be more noticeable with high Feedback settings.
Feedback	0-100%	Sets the amount of feedback. Higher values produce a more “metallic” 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	Values	Description
Mix	0-100%	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	0-100%	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 yellow when tempo sync is on, and gray when it is off.
Rate	1/1 - 1/32, 1/1 - 1/32 Triplet, 1/1 - 1/32 Dotted or 0 - 5Hz	If tempo sync is on, this is where you specify the base note value for tempo syncing the Phaser sweep. If tempo sync is off, the sweep rate can be set freely with the Rate knob, without sync to tempo.
Feedback	1-100%	This sets the amount of feedback. A higher value produces a more pronounced effect.
Tempo Sync knob	x1 to x10	This is the note value multiplier for the Phaser sweep when tempo sync is used.
Stereo Basis	0-100%	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).

Parameter	Description
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	Values	Description
Speed	Stop/Slow/Fast	This controls the speed of the Rotary.
MIDI Ctrl	Controllers [1] to [16]	Selects the MIDI continuous controller for the Speed parameter. See page 53.
Mode	Switched/ Variable	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	0-100%	Applies a soft overdrive or distortion.
Crossover Freq.	200-3000Hz	Sets the crossover frequency between the low and high frequency loudspeakers.
Mic Angle	0-180 degrees	Sets the simulated microphone angle. 0 = mono, 180 = one mic on each side.

Parameter	Values	Description
Mic Distance	1-36 Inches	Sets the simulated microphone distance from the speaker.
Low Rotor Amp Mod.	0-100%	Adjusts amplitude modulation depth.
Low Rotor Mix Level	0-200%	Adjusts overall bass level.
Hi Rotor Amp Mod.	0-100%	High rotor amplitude modulation.
Hi Rotor Freq. Mod.	0-100%	High rotor frequency modulation.
Phasing	-100 to 100	Adjusts the amount of phasing in the sound of the high rotor.
Hi Slow	0.0-720 rpm	Fine adjustment of the high rotor Slow speed.
Hi Rate	0.0-720 rpm/sec	Fine adjustment of the high rotor acceleration time.
Hi Fast	0.0-720 rpm	Fine adjustment of the high rotor Fast speed.
Lo Slow	0.0-720 rpm	Fine adjustment of the low rotor Slow speed.
Lo Rate	0.0-720 rpm/sec	Fine adjustment of the low rotor acceleration time.
Lo Fast	0.0-720 rpm	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	Values	Description
Mix	0-100%	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 yellow when tempo sync is on, and gray when it is off.
Tempo Sync pop-up	1/1 - 1/32, 1/1 - 1/32 Triplet, 1/1 - 1/32 Dotted	If tempo sync is on, this is where you specify the base note value for tempo syncing the auto-panning.
Tempo Sync knob	x1 to x10	This is the note value multiplier, determining the timing of the auto-panning.
Delay	0-100 ms	This determines the delay time and thus the character of the chorus effect, if activated.
Depth	0-100%	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	0-100%	This sets the modulation rate for the chorus effect, if activated.

Parameter	Values	Description
Stereo Basis	0-100%	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	0-100%	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	Values	Description
Mix	0-100%	Sets the level balance between the dry signal and the effect.
Output	0-100%	Adjusts the output level of the effect.
Tone	1 - 5000Hz	Sets the frequency (pitch) of the modulating oscillator.
Tempo sync on/off		The button above the Speed knob is used to switch tempo sync on or off. The button is yellow when tempo sync is on, and gray when it is off.
Speed	1/1 - 1/32 (straight, triplet or dotted) or 0 - 10Hz	If tempo sync is on, this is where you specify the base note value for tempo-syncing the effect. 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	On/Off	Turns modulation of the pitch parameter on or off.
Mono button	On/Off	Governs whether the output will be stereo or mono.
Depth	0-100%	Governs the depth of the pitch modulation.
Waveform buttons	Sine, Square, Saw, Reverse Saw, Triangle	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 as follows:

Parameter	Values	Description
Mode	I, II,I II, IV	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	0-24	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	1-65	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.

Parameter	Values	Description
Mix	0-100%	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	0-100%	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	Values	Description
Mix	0-100%	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 (yellow button) or off (gray button).
Speed	1/1 - 1/32, 1/1 - 1/32 Triplet, 1/1 - 1/32 Dotted or 0 - 50Hz	If tempo sync is on, this is where you specify the base note value for tempo-syncing the effect. 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	Stereo/Mono	Determines whether the Chopper will work as an auto-panner (button set to “Stereo”) or a tremolo effect (button set to “Mono”).
Waveform buttons	Sine, Square, Saw, Reverse Saw, Triangle	Sets the modulation waveform.
Depth	0-100%	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 85).

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 62.

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 62.

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 63.

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	Values	Description
Number of Bands	2-24	This governs how many frequency bands the modulator signal is divided into. Fewer bands will provide a thinner more resonant sound, whereas using more bands will make the sound fuller and more intelligible.
Bandwidth	0-100%	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.	40-8000Hz	These parameters set the minimum and maximum frequency limits for the Vocoder, respectively.
log/lin	0-100%	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	10-19699ms/ HOLD	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	0-100%	This lets through high frequencies around the “S” frequency from the original input signal while notes are played.
Talk Thru	0-100%	Adjusts the level of the original input signal passed to the Vocoder output while notes are played.
Gap Thru	0-100%	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	0-100%	This controls the output level of the Vocoder.
Emphasis	0-100%	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:

Parameter	Values	Description
Voices	1-8	This sets the number of voices for the synth.
Fine Tune	-100 to +100 Cents	Tunes the oscillators in cent (100th of a semitone) steps.
Pitch Bend	1-12 Semitones	Sets the up/down range of the Pitch Bend in semitone steps.
Noise	0-100%	Adds white noise to the sound.
NoiseMod	0-100%	This makes the oscillators modulate the noise level. This gives the noise a rasping sound, turning “sss” into “zzz”.
P.Drift	0-100%	Adds random pitch variation to the oscillators.
P.Glide	0-100%	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	0-100%	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	12.00 to 0.00	Allows you to detune one of the oscillators in cent steps.
LFO Rate	1-23Hz	Controls the LFO rate (for vibrato).
Vibrato	0-100%	Adds vibrato to the oscillators. This can also be controlled by using the Mod Wheel.

Restoration plug-ins

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

DeClicker

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

- **Note that the DeClicker module is not optimized for crackles (a series of short clicks).**
  However, as it is often hard to distinguish between clicks and crackles, you might also be able to use it to improve your recording in this respect.
- **If the recording also contains background noise (hiss), you may want to combine DeClicker with the DeNoiser plug-in.**

How DeClicker works

The Declicker process is divided in two tasks:

- Analysis – when the audio signal passes through DeClicker, the selected analysis algorithm finds the clicks in the recording. You provide input to the analysis parameters by selecting a Mode and the Threshold and DePlop parameters.
- Removal – a de-click algorithm is applied to the audio, removing the clicks. In many cases, the original audio material "hidden" underneath a click 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

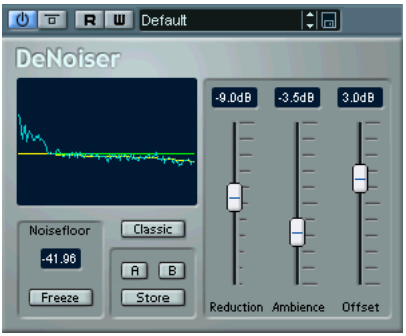
Parameter	Description
Audition	When this is activated, only the removed material will be heard. The Output display will also show the waveform image of the removed material in this mode.
Classic	When this is activated, the DeClicker attempts to remove both audible clicks and crackle noise. When it's 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.

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

Tips and Tricks

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

DeNoiser



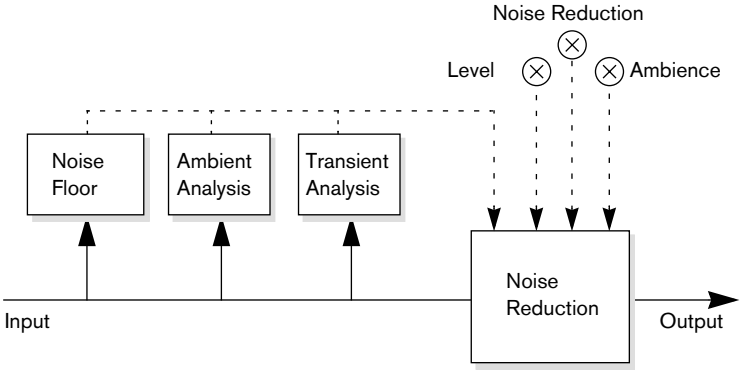
The DeNoiser plug-in lets you suppress noise without affecting the general sound quality. Or, in tech talk, the DeNoiser removes broad band noise from arbitrary audio material without leaving any “spectral finger print”. The algorithm that this plug-in is based on has the ability to track and adjust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

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

How DeNoiser works

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

The figure below shows the signal flow:



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

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

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

The Noisefloor Display

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

- The dark green spectral graph.
  This shows the spectrum of the audio currently being played back. The horizontal axis shows the frequency (linear scale). The low frequencies are visible on the left side, the high ones on the right side. The vertical axis shows the signal amplitudes, thus the level (displayed as a logarithmic dB scale).
- The yellow line.
  This is a spectral estimation of the noise floor. The average of this value is shown numerically below the display.
- The light green line.
  This is simply a graphic representation of the Offset parameter.

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

Parameters

Parameter	Description
Freeze	If you activate this button, you “freeze” the noise floor detection process. The yellow noise floor graph in the display will hold its current value (as will the numeric noise floor value display below) until you deactivate Freeze. This allows you to take a closer look at the readings.
Reduction	Governs the amount of noise reduction. The display above this fader shows the amount of dB by which the noise level is being reduced. The final result also depends on the Ambience parameter, and on the automatic Ambience and Transient analysis of the original material, as described above.
Ambience	This parameter is used to specify a balance between the noise suppression and the amount of natural ambience, which is essential for a natural result. With a low Ambience setting, the sound can become somewhat lifeless and sterile. A high setting, on the other hand, preserves more of the ambient character of the sound, but the noise suppression is less effective.

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	Values	Description
Crackle	N/A	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	33-45-78	When emulating the sound of a vinyl record, this switch lets you set the RPM (revolutions per minute) speed of the record.
Noise	N/A	This dial regulates the amount of static noise added.
Distort	N/A	Use this dial to add distortion.
EQ	N/A	Turn this dial to the right to cut off the low frequencies, and create a more hollow, lo-fi sound.
AC	N/A	This emulates a constant, low hum of AC current.
Frequency switch	50-60 Hz	This sets the frequency of the AC current, and thus the pitch of the AC hum.
Timeline	Today - 1900	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	Values	Description
Mix	Dry/Wet	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	20-100	This setting determines the “size” of the simulated room environment.
Predelay	0-100ms	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	0,2s - forever	This parameter sets the length of the reverb time.
Filter HighCut	-15 to 0dB	This filters out high frequencies for the reverb, which can make the reverb sound softer.
Filter LowCut	-15 to 0dB	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	Values	Description
Mix	0-100%	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	0-100%	Governs the “size” of the simulated room environment.
Predelay	0-100%	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	0-100%	This parameter sets the length of the reverb effect.
Damp	0-100%	This parameter “dampens” the higher frequencies, producing a rounder and smoother sounding reverb.

Surround plug-ins

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

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).

Matrix Decoder

The Matrix Decoder reverses the 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

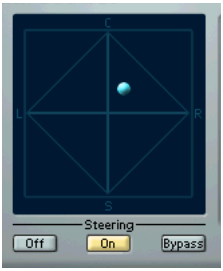
- Create an output bus with the “LRCS” speaker arrangement, in the VST Connections window, and route it to the physical outputs on your audio hardware.
  This is if you want to make a four-channel surround mix. If you want to make a five-channel mix, see “Using the Matrix Encoder/Decoder with the 5.0 Surround format” on page 80.
- 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.

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 the 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.

MixerDelay

The MixerDelay is a tool that allows you to adjust and manipulate each individual channel in a surround track or bus. The channels have 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 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.
- The channel routing section lets you select/switch the desired outputs for the channels quickly. 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 mennu 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 or Mix8to2 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 eight 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 60.

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.

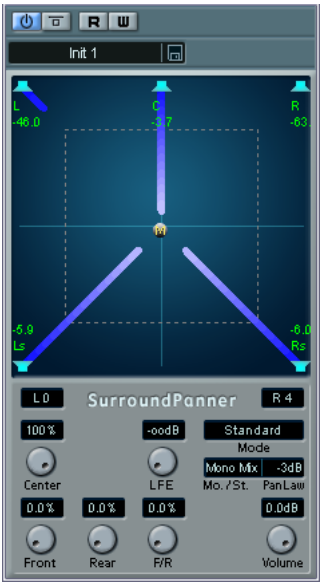
- The section has 8 buttons, one for each channel.
- Above each button there are 8 corresponding value fields that display the bit resolution the files will be converted to.
  Clicking a button several times cycles through the available bit resolution values.

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 SurroundPan 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 plug-ins

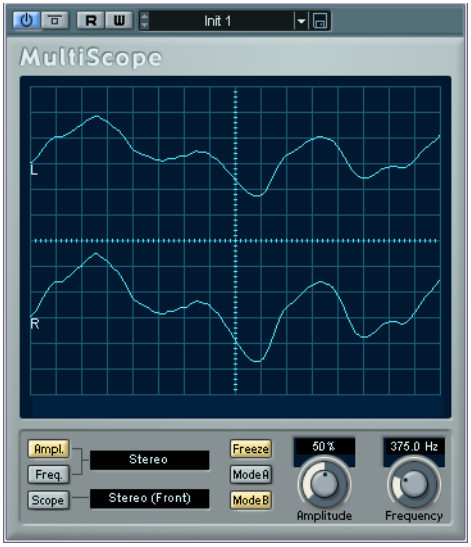
This section contains descriptions of the plug-ins in the “Tools” category.

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.)

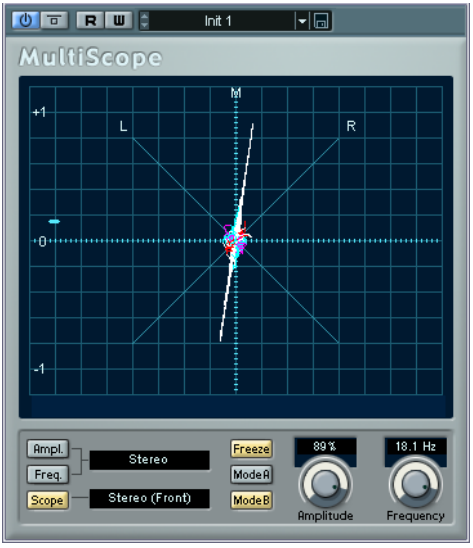
Ampl (Oscilloscope) mode



- 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:

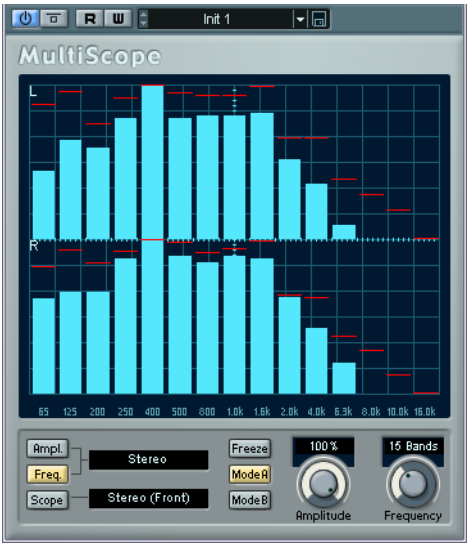
- A vertical line indicates a perfect mono signal (the left and right channels are the same).
- A horizontal line indicates that the left channel is the same as the right, but with an inverse phase.

- 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, as described above.
- 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.

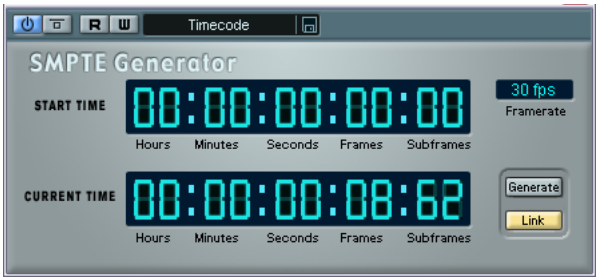
- If the MultiScope is used with a multi-channel track or output bus, you can select any speaker channel for viewing, or All Channels to view them all at once.

- 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.

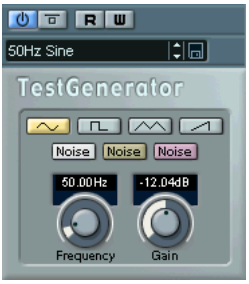
Example - Synchronizing a device to Nuendo

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 106.

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 (Oscillator 1 only)	Governs the amount of frequency modulation. See page 106.
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 page 106.

LFO section

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

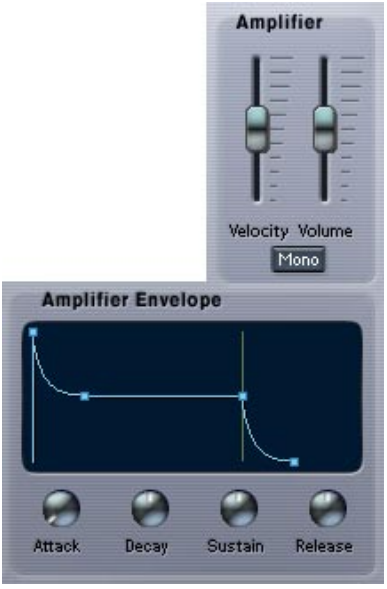
Parameter	Description
LFO Wave	This sets the LFO waveform for modulating parameters: Sine and triangle waves have a smooth waveform, suitable for normal vibrato. Saw produces a ramp up or down cycle. S&H produces stepped random modulation. Square waves produce cycles that abruptly change between two 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 page 105.
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 page 105).
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 page 105.
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 number assigned to the parameter, see page 106.

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

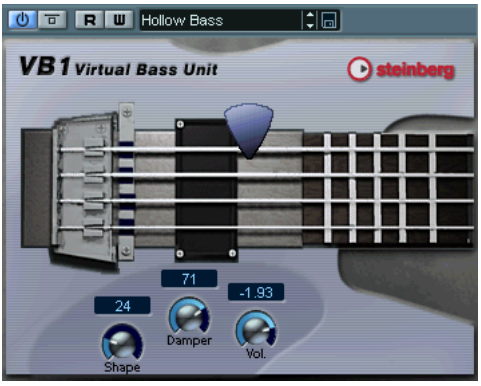
Frequency Modulation, or FM, is when the frequency of one oscillator (called the “carrier”) is modulated by the frequency of another oscillator (called the “modulator”). Using FM can produce a wide range of harmonic and non-harmonic timbres.

- 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 104).

VB-1 Bass Synth



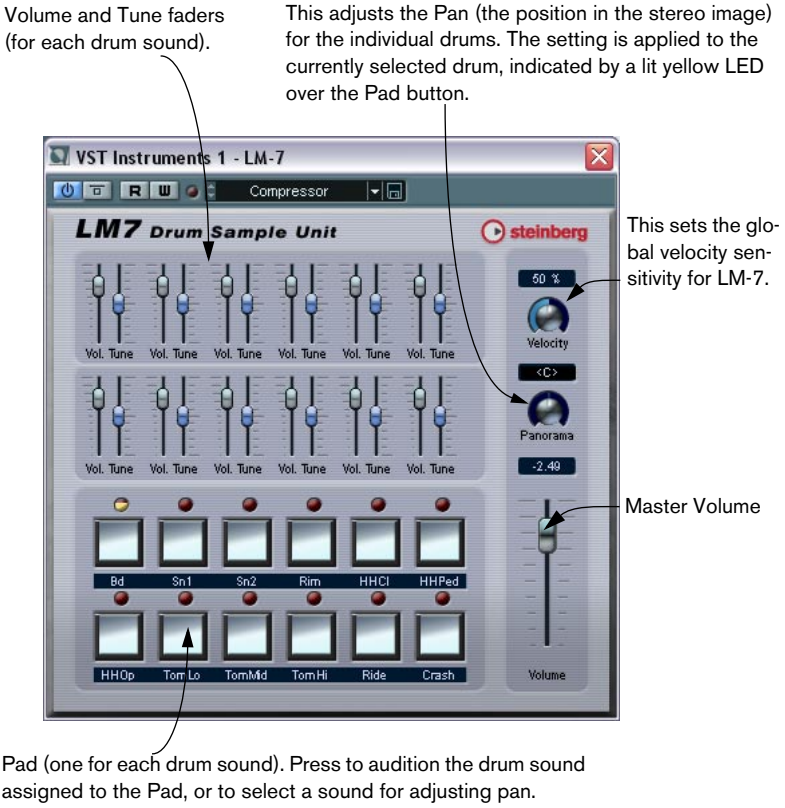
The VB-1 is a virtual bass instrument built on real-time physical modelling principles. It has the following properties:

- 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

Parameter	Description
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



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:

Drum sound	Note	Comment
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.

3

Nuendo 1.x effects and instruments

Introduction

This chapter describes the VST effects and VST instruments under the “Nuendo 1.x Plugins” heading on the effect pop-up menus. These are plug-ins from previous versions of Nuendo.

- **In most cases there are plug-ins with similar or identical functionality among the Nuendo 2 plug-ins. The Nuendo 1.x plug-ins are provided to allow opening of projects created in previous versions of Nuendo.**

  In the descriptions on the following pages, we will refer to the new Nuendo plug-ins when the parameters and functions are the same.

Audio effect plug-ins

AutoPan

Makes the sound move automatically between the left and right channel. This effect would most often be used as an insert effect for a stereo track (audio, FX channel or group channel). Note however:

- **The AutoPan is a mono in-stereo out effect.**
  When using it as an insert effect for stereo channels, only the left or right channel will be processed (depending on the routing settings you make for the insert slot).

Parameter	Explanation
LFO Freq	This sets the speed of the panning effect.
Width	This sets the depth of the effect, that is, how far out to the left/right speaker the sound should move.
Waveform	This sets the shape of the LFO producing the effect. Sine and Triangle both produce a smooth sweep, but with different characteristics. Sawtooth creates a ramp (sweep from one speaker to the other and then a quick jump back). Pulse makes the signal jump back and forth between the speakers.
Output Level	The stereo output level of the effect.

Chopper2

Chopper2 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 the same as for the Nuendo 2 “Chopper” effect (see page 59) with the addition of Input and Output level controls.

Distortion

The Distortion effect plug-in is capable of producing anything from a soft “crunch” to all-out distortion. There is a selection of factory presets available. Note that these presets are not only stored parameter settings, but actually use different basic algorithms to produce distortion. The basic character of the distortion preset “models” is indicated by their names. The parameters are as follows:

Parameter	Description
Input	Sets the Input level.
Output	Sets the Output level. As distortion generates harmonics, it increases the level of the processed signal. You can use the Output fader to compensate for the level increase.
Shapes	The Shape parameter determines how much the input signal is affected by the distortion effect.
Contour	This is a selective low pass filter, altering the tonal quality of the distortion.
Drive	Governs the amount of distortion.
Factory Presets	Select one of five presets, which can be used as they are, or as a basis for further “tweaking”.

DoubleDelay2

This effect provides two separate delays that can be tempo-synced. The parameters are the same as for the Nuendo 2 “DoubleDelay” plug-in (see page 7.)

Fuzzbox

This is a simulation of the good old transistor distortion stomp box with the following parameters:

Parameter	Description
Boost	This governs the amount of distortion. If you want to increase the distortion without raising the signal level, you may have to adjust the Volume knob as well.
Clipback	Raising this parameter will “invert” the part of the signal that is above the clipping level, instead of employing hard clipping. The result is that more 2nd order harmonics are added, changing the character of the distortion.
Volume	This is a volume control for the output signal from the Fuzzbox.

JetFlange

This is a classic flanger effect with the following parameters:

Parameter	Description
Mix	Sets the level balance between the dry signal and the effect.
Delay	This parameter affects the frequency range of the modulation sweep, by adjusting the initial delay time.
Depth	This sets the depth of the modulation sweep.
Rate	This sets the speed of the modulation sweep.
Feedback	This determines the character of the flange effect. Higher settings produce a more pronounced and “metallic” sounding sweep.

Metalizer2

The Metalizer2 feeds the audio signal through a variable frequency filter, with tempo sync or time modulation and feedback control. The parameters are the same as for the Nuendo 2 “Metalizer” plug-in (see page 46).

ModDelay2

This is a delay effect that can either be tempo-based or use freely specified time settings. The delay repeats can also be modulated. The parameters are the same as for the Nuendo 2 “ModDelay” plug-in (see page 9).

NuendoCompressor

The NuendoCompressor is a multi-band compressor with up to five adjustable frequency bands. For details on the parameters and how to make settings, see the description of the Nuendo 2 “MultibandCompressor” plug-in (page 31).

NuendoDither

This is the same multi-channel dithering plug-in as the Nuendo 2 “SurroundDither” plug-in. See page 84.

NuendoDynamics and NuendoDynamics2

These plug-ins are the same. The parameters are the same as for the Nuendo 2 “Dynamics” plug-in (see page 18).

NuendoEQ

This plug-in is the same as the “NuendoEQ2” plug-in, with one difference: the maximum EQ frequency is 16 kHz (whereas the NuendoEQ2 and the EQ sections for audio channels have a max frequency of 20 kHz). Apart from that, the parameters and features are the same as for NuendoEQ2, which in turn has the same features as the regular EQ section for audio channels (described in the Operation Manual).

NuendoReverb

NuendoReverb is the same reverb as the Nuendo 2 “Reverb A”, but with a different graphical user interface. The parameters are described on page 74.

NuendoVerb3

NuendoVerb3 is a useful reverb plug-in that uses very little processor power. It offers the following reverb and delay algorithms (selected on the pop-up menu at the top of the window):

Algorithm	Description
Hall	The reverberation of a medium-sized hall.
Large Hall	The reverberation of a larger hall.
Large Room	The reverberation of a large room.
Medium Room	The reverberation of a medium-sized room.
Small Room	The reverberation of a very small room.
Plate	The slightly metallic effect of a plate reverb.
Gated	A special effect, where the reverb is abruptly cut off.
Effect 1	A special “bouncing” effect.
Echoes	An echo (delay) effect.
Effect 2	A special, resonant effect, suitable for “ringing” metal sounds.

After selecting a reverb algorithm, you can adjust the following three parameters:

Parameter	Description
Size	This is the size of the simulated room. Changing this will affect the density and character of the reverb. If you have selected a Reverb Type where you can hear the individual “bounces” (Effect 1, Echoes, etc.), raising the Size will increase the time between each “bounce”, like the time control on a delay effect.
Decay	This is the decay time for the reverb. The higher the value, the longer the reverb.
Damp	Raising this value will cause the high frequency contents of the reverb sound to die out quicker. This results in a softer, darker reverb.

- **The NuendoVerb3 is a mono in-stereo out effect.**
  When using it as an insert effect for stereo channels, only the left or right channel will be processed (depending on the routing settings you make for the insert slot).

StereoEcho

The StereoEcho is a delay with separate settings for the left and right channel. It can also be used as a single mono delay, in which case the maximum delay time will be doubled.

- **The StereoEcho is a mono in-stereo out effect.**
  When using it as an insert effect for stereo channels, only the left or right channel will be processed (depending on the routing settings you make for the insert slot).

The StereoEcho has the following parameters:

Parameter	Explanation
Delay1	The delay time for the left channel. The maximum delay time is 500 ms, unless you link both channels for mono operation, in which case the maximum delay time is 1000 ms - see below (1000ms = 1 second).
Feedbck1	The delay feedback for the left channel. Higher values result in a higher number of echo repeats.
Link 1-2	Activating this switch turns the effect into a mono delay. When Link is on, only the left channel parameters will be available (Delay1, Feedback1, etc).
Delay2	The delay time for the right channel.
Feedbck2	The delay feedback for the right channel.
Del2 Bal	This parameter determines how much of the left channel output is sent to the right channel input. When set to 0.0 (fully left), then none of the left channel output is added to the right channel input; when it is set to 1.0 (fully right), the right input receives both its normal source and the complete output of the left channel.
Volume L	The output level of the left channel delay.
Volume R	The output level of the right channel delay.

StereoSpread

This plug-in can be used to enlarge or narrow the stereo “width” of an existing stereo signal. It has the following parameters:

Parameter	Description
Slider	When the slider is positioned far left, the output will be mono. Settings between far left and up to just before middle position produce a narrower stereo spread. The middle position corresponds to the original signal. Values just after the middle and up to far right will enlarge the stereo image.
Center	This should be on (lit) for normal operation. If the slider is positioned somewhere in the right half of the slider range, switching the Center button off will remove the “virtual” center of the signal. This can be used as a “karaoke” effect, to remove or suppress the lead vocal (which is typically mixed dead center) from a recording. Note that if the slider is in the left half of the range, then switching this button does nothing.
Inverse	This button simply inverts the left and right channels, so that the left channel becomes the right channel and vice versa.

StereoWizard

The StereoWizard is a stereo width enhancer that takes a stereo input signal and makes it sound “wider”. It must be used as an insert effect for a stereo channel. StereoWizard will give best result if you use “real” stereo material (as opposed to mono channels panned to different positions in the stereo image).

Parameter	Explanation
Amount	Higher values result in a greater stereo width. Normally, you should set this to values between 0.00 - 0.20; higher values can be used for special effects.
Reverse	Reverses the left and right channel.

Symphonia

The Symphonia plug-in combines a stereo enhancer effect, an auto-panner and a chorus/flanger-type effect. The parameters are the same as for the Nuendo 2 “Symphonic” plug-in (but without the Output level control). See page 54.

Tranceformer2

Tranceformer2 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, synchronized to the Song tempo if you wish. The parameters are the same as for the Nuendo 2 “Tranceformer” plug-in (see page 56) with two differences: The Tranceformer2 has an Input level control and the “Tone” parameter on the Tranceformer is called “Pitch” on the Tranceformer2.

UV22

The UV22 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 Operation Manual). The plug-in has the same features as the Nuendo 2 “UV22HR” dithering plug-in (see page 60) but dithers to 16 bit resolution only.

VST Instruments

LM-9

The LM-9 is a simple drum machine. It has the following properties:

- LM-9 is polyphonic with up to 9 voices.
- LM-9 responds to MIDI Note On/Off with velocity controlling volume.
- LM-9 receives MIDI in Omni mode (on all MIDI channels).
  You don't need to select a MIDI channel to direct MIDI to LM-9.

LM-9 Parameters:

Parameter	Description
Velocity	This sets the global velocity sensitivity for LM-9. The higher the value, the more sensitive LM-9 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.
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-9 comes with two sets of drum sounds; "Acoustic" and "Beat Box". Acoustic features samples of an acoustic drum kit and Beat Box features classic analog drum machine sounds. To switch set, use the pop-up menu (just like you switch between effect programs).
The table below shows how the drum sounds are assigned to note values on your MIDI keyboard. The mapping is GM compatible:

Drum sound	Note	Drum sound	Note
Bass	C1	Tom 1	D2
Snare	D1	Tom 2	B1
Hi-Hat	F#1	Tom 3	A1
O-Hi-Hat	A#1	Crash	C#2
		Ride	D#2

Neon

The Neon is a simple software synth with the following properties:

- The Neon is polyphonic with up to 16 voices.
- The Neon receives MIDI in Omni mode (on all MIDI channels).
  You don't need to select a MIDI channel to direct MIDI to the Neon.
- The Neon responds to the following MIDI messages:
  MIDI Note On/Off (velocity governs volume), Volume, Pan, Pitch Bend (± 2 semitones) and Modulation (vibrato).

Neon Parameters:

Parameter	Description
Range	Selects an octave range for the oscillators, 16, 8 or 4 feet.
Waveform	The basic waveform for the oscillators, Triangle, Sawtooth or Square.
LFO Speed	Governs the speed of the vibrato. The vibrato depth is controlled via MIDI Modulation messages (for example, using the Mod Wheel on your MIDI controller).
Osc 2 Detune	Allows you to detune the "second oscillator" ± 7 semitones. By setting this to a value close to "twelve o'clock", you will get fine detuning, for a warmer, fatter sound.
VCF Cutoff	The Cutoff Frequency for the filter, governing the amount of high frequencies in the sound. On the Neon, the Cutoff control also serves as a Depth control for the Filter Envelope (VCF Attack, Decay, Sustain, Release), so that the lower the setting of the Cutoff parameter, the more will the filter be affected by the Filter Envelope.
VCF Resonance	The Resonance control for the filter. Raise this for a more pronounced filter effect.
VCF Attack, Decay, Sustain, Release	The Filter Envelope. Use these parameters to determine how the filter should open and close with time, when a note is played.
VCA Attack, Decay, Sustain, Release	The Amplitude Envelope. Use these parameters to determine how the amplitude (volume) should change with time, when a note is played.

Index

S

T

U

V