

lector VOCODER

User Manual

English



Foreword

Thank you for purchasing the Waldorf Lector Vocoder plugin.

Lector is a unique vocoder for musical alienation of audiosignals. It contains a polyphonical synthesizer based on the legendary Waldorf sound engine. Lector puts in your hands a sonic toolset for a wide range of diverse applications.

What to read?

The biggest problem with any manual is to cover both the needs of an absolute expert and a beginner alike. There are people who read a manual cover to cover while others don't even touch it. The latter is the worst choice, especially when the manual describes a Waldorf instrument.

If you decide to read the following manual, we promise you a lot of fun while reading about and working with Lector.

Your Waldorf Team

Hint

Waldorf Music GmbH is not liable for any erroneous information contained in this manual. The contents of this manual may be updated at any time without prior notice. We made every effort to ensure the information herein is accurate and that the manual contains no contradictory information. Waldorf Music GmbH extends no liabilities in regard to this manual other than those required by local law.

This manual or any portion of it may not be reproduced in any form without the manufacturer's written consent.

Waldorf Music GmbH, Lilienthalstrasse 7, D-53424 Remagen, Deutschland

Lector Development Team

Software:	Wolfram Franke, Thomas Rehaag
User interface:	Axel Hartmann
Manual:	Holger Steinbrink
Revision:	1.0, May 2011
Betatest/Sounddesign:	Boele Gerkes, Achim Gratz, Till Kopper, Dirk Krause, Jay de Miceli, Dr. Georg Müller, Kai Niggemann, Hermann Seib, Holger Steinbrink, Dr. Stefan Trippler, Dr. Ingo Weidner



Please visit our website www.waldorfmusic.de
Maybe you can download a newer version for your Lector.

We would like to thank

Karsten Dubsch, Willie Eckl, Joachim Flor, Michael von Garnier, Frank Schneider, Kurt “Lu” Wangard, Silvia Wollenweber, 吴海彬, and anyone we have forgotten.

Content

Foreword	2
Content	4
Overview	5
Introduction	6
Installation	7
Installation under Windows	7
Fehler! Textmarke nicht definiert.	
Installation under Mac OS X	8
Activation of Lector	9
Quick Start Tutorial	11
Basic Operation	12
Key Commands	12

Control Elements	12
Preset Management	13
The Controls in Detail	15
Parameter of the Input Section.....	15
The Carrier Signal of Lector	17
Parameter of the Lector Synthesizer	19
Parameter of the Lector Vocoder	28
The Effects	34
Appendix	39
What is a Vocoder?	39
Use Cases.....	39
Glossary	43
Product Support	48

Overview

User Interface of Lector



- ① Parameter of the Input section/Carrier signal
- ② Parameter of the Lector Synthesizer

- ③ Parameter of the Vocoder filterbank
- ④ Effects section

Introduction


About this Manual


This manual was written to help you to become familiar with Lector. It will also aid experienced users with routine tasks.


To avoid confusion, the terminology in this manual is based on the Lector parameter names. You will find the various terms explained in a glossary at the end of this manual.

We also used a uniform set of symbols to show you topics of particular interest or significance. Important terms are highlighted in bold letters.

Symbols

 **Caution** – The comments that follow this symbol will help you avoid errors and malfunctions.

 Info – Additional information on a given topic.

 **Example** – Real-world examples to try out.

Highlighted Control Features and Parameters

All of the Lector's buttons, controls and parameters are highlighted in **bold** letters throughout the manual.

Example:

- Turn the **Cutoff** dial.

The Lector's different modes and parameters are illustrated in a depiction of the display.

The value range of a continuous parameter is indicated from low to high with both values shown in italic letters, separated by three dots.

Example:

Resonance *0...100%*

Installation

Our software is no longer based on an eLicenser. No new installations of our software can be made with an eLicenser. Our new licensing system is completely software-based.

With one license, the software can be installed and used on up to 3 computers simultaneously. Older licenses can be converted into our new licensing system.

For older licenses purchased before 2016, please contact our support.

<https://support.waldorfmusic.com>

If you have a current license, it can be found in your "myWaldorf" customer account under: License keys.

Here you will find your activation mail and activation key as well as the download-links for MacOS and Windows to download the desired installer.

Download the version for your operating system, unpack the ZIP file and install the purchased plug-in.

Open your host program (e.g. Cubase, Logic, ProTools, Studio One etc.) and check the virtual instruments for your new software. Open it the same way you usually open virtual instruments in your host software. Here you can select the desired plug-in and open it.

The plug-in will start in demo mode. Enter the email and the key for the activation into the provided fields. This is usually done with copy/paste. If you cannot click in the second field, use the TAB key on the keyboard. Then press the Enter key to complete the activation.

Next time you will start the Waldorf Plug-In, it will start in full authorized mode.

If you have any questions, please have a look at the FAQ

www.waldorfmusic.com/produkt-faq

or write an email to: support@waldorfmusic.com


Quick Start Tutorial


Here you find a quick start for working with Lector:

- Load the Lector plug-in in an insert slot of any audio or instrument track of your host application which contains a suited signal as speech, vocals or drums.
- Start the playback of your host application and turn the **Speech** dial in Lector until you hear the original signal.
- Create a MIDI track and route the output directly to Lector.
- Activate the Lector synthesizer and turn the **Carrier** dial. While playing MIDI notes via a connected MIDI master keyboard or recorded MIDI events you should hear the Lector synthesizer. If you don't have a MIDI keyboard you can activate the **Latch** mode to trigger a permanent signal of the Lector synthesizer.
- Turn down the **Speech** and also the **Carrier** dial. Instead turn up the Vocoder dial. Start the playback of your host application and play some MIDI notes. You should now hear the typical Vocoder sounds.
- If you have loaded the VST3 or the AU version of Lector you can activate the sidechain function of

your host application as also as in Lector to route a desired signal as Carrier into Lector.

- If you are using the VST 2.4 plug-in you can load the additional Lector Carrier Plug-in as an insert effect in any desired channel to route this signal as Carrier to Lector.

 Please read the corresponding manual of your host application how to load insert effects and how to work with sidechain.

 A detailed description of all Lector parameter and its functionality can be found within the next pages.

Basic Operation

Lector has been optimized for a screen resolution of at least 1024x768 pixels. If the resolution of your screen is lower you will have to scroll your window.

Key Combinations

- If you hold down the [CTRL] key (PC) or Command key (Mac) on your computer keyboard and click on any control, its value is automatically set to its default value (e.g. Cutoff is set to *20000Hz*, Bands is set to *42* or EQ Treble is set to *0*).
- If you use a computer mouse with scroll wheel you can use it to change parameters. Move your mouse pointer over the desired value and use the scroll wheel to change it. If you hold down the [SHIFT] key you can increase the resolution.
- If the dial control mode of your host application is set to “circle mode” (you change a value by dragging around the dial in a circle), then you can temporarily change to “up/down mode” by holding down [ALT].

Control Elements

Using Lector’s controls is simple. There are some different types of control elements:

Dials



To set a value, click on the dial, hold down the mouse button and drag around the dial in a circle or move the mouse up and down. This depends on the setting of your host application. Alternatively you can edit values by using your mouse scroll wheel.

Pop-up Menu



Click on the corresponding parameter to open a pop-up menu where you can choose the desired parameter.

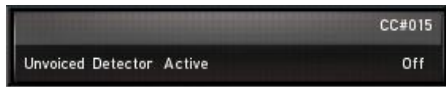
Switches



Switches are represented by “LEDs”. If a function is active, its “LED” will be lit red. To activate/deactivate a function, click on its “LED”.

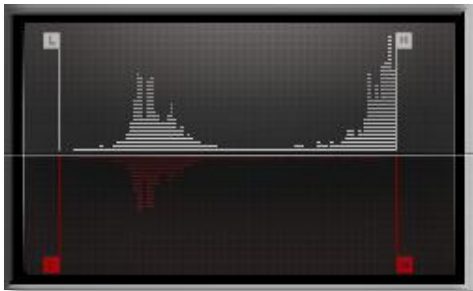
Display

The information display shows parameter information and a brief description of the parameter function. In addition, MIDI controller allocations (CTRL) and the current numerical value of the parameter are displayed.



Filter Bank Display

By clicking on the “L” or “H” handles, you can move the parameters directly in the display.



Lector Logo



If you click on the Lector logo, a “program information” window opens.

Preset Management

To load and save a Preset Program within Lector, please proceed as follows:

- Left click on the **File** menu button to open a pop-down menu with the following parameters:
 - **Load Bank** opens a window to load a Lector preset bank.
 - **Save Bank** opens a window to save the current Lector preset bank.
 - **Load Program** opens a window to load a single Lector preset program.
 - **Save Program** opens a window to save the current Lector preset program.
 - **Save All Programs up to this Program** saves all presets from preset program 1 to the current sound program as single FXP presets.
 - **Save All Programs up to this Program as Audio Unit Preset (only Mac)** saves all presets from preset program 1 to the current sound program as single AudioUnit presets.

- If your host application offers a separate Load/Save dialogue you can also use it to load and save sound programs.

The Controls in Detail

The analog user interface offers direct access to the dials on the Lector. Besides the Input/Speech section will find three separate sections: The internal synthesizer for creating a carrier signal, the Vocoder section and at least the Effects section.

Parameter of the Input Section

Here you will find a compressor for the input signal (Speech Input) and the carrier signal as also the Unvoiced Detector.



Compression

The compressor optimizes the input and/or the carrier signal for the further processing by the vocoder. It based on an automatic gain adaption which leads to a more

homogeneous audio signal. Both compressors feature the same control elements.

Compressor Active

On / Off

Switches the corresponding compressor on or off. The switch lights red when activated.

Gain

-20...+20 dB

Gain controls the level of the compressor. Higher settings raise up the level of the speech or carrier signal. Use Gain to avoid clippings of the vocoder by reducing high level signals.



Bear in mind that this compressor features no separate threshold function. It is controlled automatically and depends on the input signal.

Ratio

1:1...1:100

Ratio determines the compression rate between input and output signal. A value of 1:3 means a reduction of the output signal by factor 3.

Swap

Off / Swapped

Swap inverts the speech and carrier signal by one mouse click. Use this function for experimental purposes.

Unvoiced Detector

An Unvoiced Detector adds noise to the non tonal vocoder signal to increase the articulation of speech and its transients.

* The word "Waldorf" for example consists of consonants and vowels. The W in the beginning sounds more tonal while the d and the f sounds noisy. Between these characters you will find vowels. The Unvoiced Detector recognizes the sibilants and generates the following output signal (C stands for Carrier and R stands for UV signal):

W a l d o r f
R C C R C C R

The transition between Carrier and UV signal is smoothed to avoid skips.



You can prelistening the signal of the Unvoiced Detector when turning up the **Carrier** parameter within the Output section. To control the recognized transients in relation to the input signal you can also turn up the **Speech** parameter.

Unvoiced Detector Active

On / Off

Switches the Unvoiced Detector on or off. The switch lights red when activated.

Sensitivity

1.000:0.000...0.000:1.000

Controls the sensitivity of the UV detector. Higher values recognize more transients so that a higher amount of UV signals will be added.

Gain

-20...+20 dB

Controls the level of the UV signal. Higher settings increase the gain of the UV signals.

UV Signal

Noise / Speech / Noise Mix

Determines the signal which is used for creating the UV signal:

- *Noise* selects White Noise which contains all frequencies with the same gain level. It is perfectly suited to increase the articulation of human speech that is used as speech signal source.
- *Speech* selects the speech signal itself as signal source for the UV detector. It is suited for signals with a high amount of transients as drum loops.
- *Noise Mix* selects also White Noise. Compared to *Noise* the noise will be mixed with the carrier signal instead of replacing it. Use this setting when you want to keep the tonal character of your carrier signal, e.g. for robotic voices.

The Carrier Signal of Lector

In most use cases the integrated Lector synthesizer will be your carrier signal. Alternatively you can use any desired signal as carrier that your host application delivers. There are two different cases:

- If you load Lector as VST3 plug-in (e.g. in newer Cubase version) or as AU plug-in (e.g. in Logic), you can take advantage of the host integrated sidechain function. In this way you can route any signal source (e.g. audio tracks or instrument tracks) directly to the carrier input of Lector by using the Aux send function or the direct routing function of your host. In Lector click on the **Sidechain** function to activate it. The integrated synthesizer is deactivated automatically.
- If your host doesn't offer a sidechain function and if you are working with the VST2.4 plug-in version of Lector, you can use the separate Lector Carrier plug-in for routing any source signal directly to Lector. Load up to four Carrier plug-ins in an insert slot of a desired track (audio track or instrument track). In Lector click on the **Sidechain** function to activate the corresponding sidechain input. The integrated synthesizer is deactivated automatically. The Lector

Carrier plug-in offers an additional Thru function for the track signal.



- i** In both cases an active Sidechain function in Lector will lit in red.
- i** Keep in mind that the Lector Carrier plug-in can look different based on your used host application.

Parameter of the Lector Synthesizer

The Lector synthesizer contains a 16 voices synthesiser based on the Waldorf DNA with a lot of useful parameters.



Lector offers two oscillators that nearly use the same parameters for editing.

Oscillator Shape *Triangle / Sine / Pulse / Saw / S & H / Noise / Sidechain / Sampleplayback*

Click on a shape symbol of the corresponding oscillator to select the desired waveform. The following shapes are currently available:

- *Triangle* selects the triangular waveform. The triangle mainly consists of the odd harmonics with very low magnitudes.
- *Sine* consists of the fundamental frequency only. It has no harmonics at all.
- *Pulse* selects the pulse waveform. A pulse waveform with a pulse width of 50% has only the odd harmonics of the fundamental frequency present. This waveform produces a hollow / metallic sound. If the Pulse waveform is selected, the parameters **FM/PW** (only Osc 1) and **PWM** are used to change the pulsewidth of the waveform.
- *Saw* selects the sawtooth waveform. A Sawtooth wave has all the harmonics of the fundamental frequency in descending magnitude.
- *S & H* (Sample & Hold) samples a random value and holds it until the next value is generated.

- *Noise* selects White Noise which contains all frequencies with the same gain level.
- *Sidechain* selects an external audio signal as oscillator signal. You have to use the sidechain function of your host application (VST3, AU) or the Lector Carrier plug-in (VST2.4). Please read also the chapter „The Carrier Signal of Lector“.
- Click on *Sampleplayback* to open a pop-up menu for selecting an audio samples as source for the corresponding oscillator. Choose the Load function to load own samples.

Semitone

C-1...C7

Sets the pitch of the oscillator in semitone steps. The standard setting for this parameter is C3, but there are cases where different values are interesting as well.

Detune (Osc 2 only)

0...100 cents

Fine-tunes the oscillator in steps of 1/100 of a semitone. The audible result of detuned oscillators is a Chorus or Flanger effect. Use a positive setting for one oscillator and an equivalent negative setting for another.

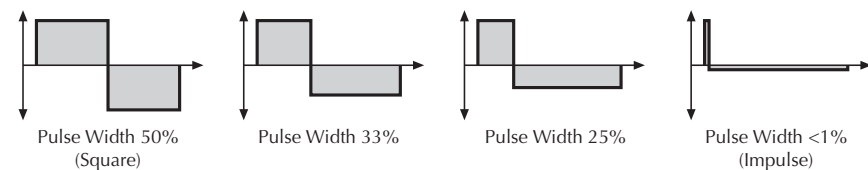
* Low value results in a slow and soft Flanger effect.

- * Mid-ranged settings are perfect for pads and other fat sounding programs.
- * High values will give a strong detune that can be used for accordions or effect sounds.

FM / PW (nur Osc 1)

0...100%

Sets the amount of frequency modulation that is applied to oscillator 2. The sound will get more metallic and sometimes even drift out of tune. When Pulse is selected as Oscillator shape, it controls only the pulsewidth of this waveform. The value 100 is equivalent to a pulse ratio of 0%, the oscillator is producing no audible signal. The value 0 is equivalent to 50% and produces a symmetric square wave without even harmonics. If you select any waveform other than *Pulse* or the wavetables, this parameter does not have any effect. The following picture illustrates the effect of the pulsewidth parameter:



PWM (also FM Env)*-100...+100%*

PWM means Pulse Width Modulation. If you have selected *Pulse* as Oscillator shape this parameter determines the amount of modulation that is applied to the pulsewidth of the oscillator's square wave by the LFO. If you select any waveform other than *Pulse*, this parameter controls the amount for the frequency modulation through the filter envelope. For this purpose the filter doesn't need to be active.

Mod*-100...+100%*

Determines the amount of influence the LFO has on the pitch. For positive settings, the pitch is increased periodically by the modulation, for negative settings, the pitch is decreased.

Osc Level*0...100%*

Controls the volume of the corresponding Oscillator.

Filter





Lector offers a multimode filter which is located after the oscillators.

Filter Active*On / Off*

Switches the Filter on or off. The switch lights red when activated.

Cutoff*0...100%*

Controls the cutoff frequency for the low pass and high pass filter types, the center frequency for the band pass and notch filter types, and the intensity of the whitening filter.

- When a low pass type (24 or 12 dB LP) is selected via the **Type** parameter, all frequencies above the cutoff frequency are damped. 
- When a high pass type (24 or 12 dB HP) is selected, all frequencies below the cutoff frequency are damped. 
- When a band pass type (24 or 12 dB BP) is selected, only frequencies near the cutoff setting will be passed through. 
- When a notch type (24 or 12 dB Notch) is selected, the frequencies around the cutoff frequency are damped. 

- The Whitening filter offers a unique function: It removes formant frequencies from choir or voices samples. The vocoder itself creates a formant based signal so that too much formants within the carrier signal can interact. The Whitening filter compensates this effect. To reduce the brilliance of the Whitening filter the Cutoff parameter controls a downstreamed 6 dB lowpass filter. When Whitening is selected, all filter parameters (except **Drive**) do not have any effect.



i Using the Whitening filter could reduce the volume of your oscillator signal.

i To understand how the Whitening filter works, create a synthesizer sound with only one sawtooth oscillator. Select the Whitening filter and deactivate the filter. Turn up the Carrier parameter in the Output section to hear the synthesizer sound while playing MIDI notes. If you activate the filter the signal will get more high frequencies. There is no additional effect because the sawtooth signal doesn't contain any outstanding formants. Deactivate the filter again and choose the Sine shape for your oscillator. This waveform offers only one harmonic which is basically one formant. Activate the filter.

You will hear: nothing! If you play some notes you maybe hear some clicks because the Whitening filter needs a little time period to locate the formants. Now choose a choir sample as Oscillator Shape. Here you can hear the effect very drastically because all formants are removed from the signal. You will also notice a loss of volume and also some brilliance.

Resonance

0...100%

Controls the emphasis of the frequencies around the cutoff point. Use lower values in the range of *0...80* to give more brilliance to the sound. At higher values of *80...113* the sound gets the typical filter character with a strong boost around the cutoff frequency. When the setting is raised to values above *113*, the filter starts to self-oscillate, generating a pure sine wave.

Keytrack

-100%...+100%

Determines how much the cutoff frequency depends on the MIDI note number. The reference note for Keytrack is E3, note number 64. For positive settings, the cutoff frequency rises on notes above the reference note, for negative settings the cutoff frequency falls by the same amount, and vice versa. A setting of *+100%* corresponds to a 1:1 scale, so e.g. when an octave is played on a

keyboard the cutoff frequency changes by the same amount. If you want to play the filter in a tempered scale, e.g. for a solo sound with self-oscillation, set the value to +100%.

Filter LFO -100%...+100%

Determines the amount of influence the LFO has on the cutoff frequency. For positive settings, the filter cutoff frequency is increased by the modulation, for negative settings, the cutoff frequency is decreased.

Drive 0...54 dB

Determines the amount of saturation that is added to the signal. If set to 0, no saturation will be added or, in other words, the signal will remain clean. Increasing the value will add some harmonics to the signal, resulting in a warm character up to complete distortion.

Drive Curve *Tube / Diode / Clip*

Determines the character of the drive. The following drive curves are available:

- **Tube** simulates the asymmetric distortion of a tube circuit.

- **Diode** generates a typical transistor distortion.
- **Clip** generates a typical digital clipping while cutting the signal at a certain gain level.

Filter Env -100...+100%

Determines the amount of influence the simplified filter envelope has on the cutoff frequency. For positive settings, the filter cutoff frequency is increased by the modulation of the envelope, for negative settings, the cutoff frequency is decreased. Use this parameter to change the timbre of the sound over time. Sounds with a hard attack usually have a positive envelope amount that makes the start phase bright and then closes the filter to get a darker sustain phase. String sounds, on the other hand, usually use a negative envelope amount that gives a slow and dark attack before the cutoff rises in the sustain phase.

Filter Velo (Velocity) -100...+100%

Determines the amount of influence the filter envelope has on the cutoff frequency, based on key velocity. This parameter works similarly to the **Env** parameter with the difference that its intensity is velocity based. Use this feature to give a more expressive character to the sound.

When you hit the keys smoothly, only minimal modulation is applied. When you hit them harder, the modulation amount also gets stronger.

i The overall modulation applied to the filter's cutoff frequency is calculated as the sum of both the **Env** and **Velo** parameters. Therefore you should always bear this total in mind, especially when the filter does not behave as you expect. You can also create interesting effects by setting one parameter to a positive and the other to a negative amount.

Attack 0...60 s

Determines the attack rate or amount of time it takes for a signal to go from zero to maximum level.

Decay 0...60 s

Determines the decay rate or amount of time it takes for a signal to reach the sustain level.

Glide

"Glide" or "Portamento" describes the continuous gliding from one note to another. This effect can be created on fretless stringed instruments or some brass instruments

(e.g. trombone). It is very common on synthesizers and used throughout all music styles. Please note that Glide affects the pitch of all oscillators.

Glide *Off / On*

Enables or disables the Glide effect.

Speed *0.001...60 s*

Determines the glide time. Low values will give a short glide time in a range of milliseconds that gives a special character to the sound. High values will result in a long glide time of up to several seconds which can be useful for solo and effect sounds.

Legato *Normal / Legato*

Normal means that a continuous glide is performed on all new notes. *Legato* means that a continuous glide is performed only when notes are played legato. Staccato played notes start on the exact pitch of their note.

Glide Mode *Portamento / Glissando*

Determines the way the Glide effect works

- *Portamento* is the continuous gliding from one note to another.
- *Glissando* activates quantizing on the glide steps to allow a glissando effect in semitones like a guitar.

LFO

In addition to the main oscillators, Lector is equipped with a low frequency oscillator (LFO) that can be used for modulation purposes. The LFO generates a periodic waveform with adjustable frequency and shape.

LFO Shape *Sine / Triangle / Square / Saw / Random / S&H*

Sets the type of LFO waveform. The following shapes are available:

- *Triangle* is perfect for smooth pitch or filter modulations.
- Even *Sine* is suited for smooth pitch or filter modulations.
- *Square* can be interesting for hard modulations or special effects.
- *Sawtooth* can generate interesting filter changes.

- *Random* generates random values and glides to them linearly.
- *S&H* (Sample & Hold) samples a random value and holds it until the next value is generated. If **Speed** is set to 0, a random value is generated on each new note.

Speed *Sample & Hold...1000 Hz or 16 bars...1/64*

Determines the frequency of the LFO. At low values, it might take some time for the LFO to perform a complete cycle.

When the LFO **Sync** parameter is set to *On*, you can adjust the **Speed** in musical values. These values are sorted in regular divided measures as also in dotted and triplet divided measures. Full measures from 16 to 1 are possible as also divided measures from 1/2 to 1/64 plus dotted and triplet variants.

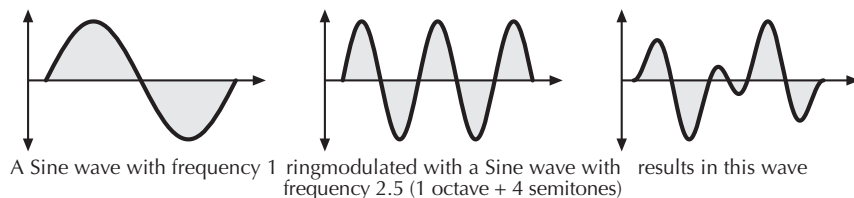
Sync *Off / On*

When **Sync** is activated, the LFO is synced to the song tempo of the host application. The **LFO Speed** setting changed to offer musically meaningful values.

Ringmod Level

0...100%

Volume of the ring modulation between Oscillator 1 and 2. From a technical point of view ring modulation is the multiplication of two oscillators' signals. The result of this operation is a waveform that contains the sums and the differences of the source frequency components. Since the ring modulation generates disharmonic components, it can be used to add metallic distorted sound characteristics. This is useful e.g. when generating synth percussion. Please note that in a complex waveform all harmonic components behave like interacting sine waves, resulting in a wide spectral range of the ring modulated sound. The following pictures show the results of two ringmodulated sine waves:



i Ring Modulation can result in unwanted low frequencies when the pitches of oscillator 1 and 2 don't differ very much. This is logical because when you use i.e. one oscillator set to 100 Hz and the second set to 101 Hz, the resulting ring modulation is 201 Hz and 1 Hz.

- * Ring Modulation can be very interesting when a slow pitch modulation is applied to one oscillator. This creates spacy effect sounds.
- * If you turn down the pitch of one oscillator markedly, you can get an effect very similar to Amplitude modulation. Use this for sounds with a periodic element if you wish.

Amplifier

Attack

0...60s

Determines the attack rate or amount of time it takes for the signal volume to go from zero to maximum level.

Decay

0...60s

Determines the decay rate or amount of time it takes for the signal volume to reach the sustain level.

Velocity

Off / On

Specifies if volume will be affected by keyboard velocity. Use this feature to give more expression to the sound.

Trigger

Normal / Single / Latch

Selects how incoming MIDI data will trigger the Lector synthesiser:

- **Normal** will respond to incoming MIDI notes to control the synthesizer polyphonically. You need to use a MIDI track within your host application to route MIDI data to Lector.
- **Single** will respond to incoming MIDI notes to control the synthesizer monophonically. You need to use a MIDI track within your host application to route MIDI data to Lector.
- **Latch** will simply trigger the Lector synthesizer permanently with the MIDI note C3. It does not need any MIDI input signal. Select Latch if you don't have a MIDI keyboard or for diagnostic purposes.

Parameter of the Lector Vocoder

The Vocoder section of Lector offers a wide range of parameters to control the analysis and synthesis signals.

Analysis Filter Bank

The Analysis Filter Bank determines which part of the frequency spectrum is being analysed by the Vocoder. Also, it includes the **Bands** parameter that sets the “resolution” of the whole Vocoding section. The single frequency bands are tuned logarithmically. If you for example set up 13 bands within the frequency range of 440 Hz and 880 Hz the bands creates a chromatic scale from standard pitch A to the octave above.



Low *20...20000 Hz*

This parameter sets the frequency range of the lower analysis band. For voice analysis, **Low** should be set to

around 220 Hz. For loops and other signals, the values should be 100 Hz.

i You can also edit the **Low** parameter by clicking on the L flag within the Filter Bank display.

High *20...20000 Hz*

This parameter sets the frequency range of the higher analysis band. For voice analysis, **High** should be set to around 14000 Hz. For loops and other signals, the values should be 16000 Hz.

i You can also edit the **High** parameter by clicking on the H flag within the Filter Bank display.

i You can change the **Low** and **High** parameter simultaneously by clicking between the flag markers and move the mouse horizontally.

Attack *0...60s*

Sets the Attack speed for the Analysis Filter Bank. The lower the value, the faster the Attack. The higher the value the slower – thus softer – the Attack. Typically, this should be set to values between 0 - 15.

Release

0...60s

Sets the Release time for the Analysis Filter Bank. The higher the value, the longer the release. Lower values will create a more percussive effect, higher values will have more “legato” results. Typically, this should be set to values between 10 - 25.

Bands

3...100

Sets the number of Filter Bands. Please note, that this parameter affects both the Analysis and Vocoder Banks. Both sections always have a matching amount of filters. The higher the number, the finer the resulting resolution, i.e. the more “intelligible” the results when used on speech signals.



For classic Vocoder effects you should set the number of bands to values between 13 and 22. If you take it further (beyond 40) it can sound increasingly “harsh”.

The Filter Bank Display

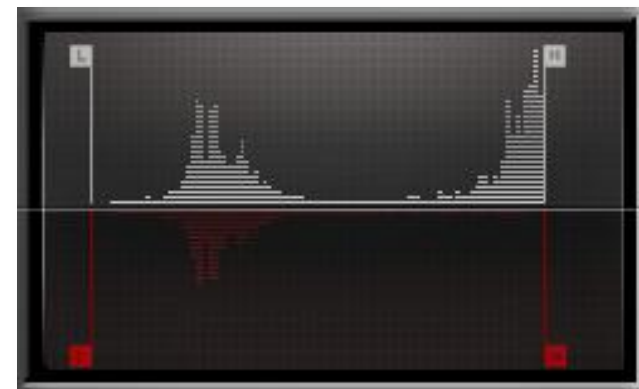
The upper half of the display shows the Analysis Band range, the lower half indicates the settings for the Vocoder Filter Band range. By clicking on the “L” or “H” handles, you can move the parameters directly in the display.

Clicking between the flag markers and moving the mouse horizontally changes both markers simultaneously.

The waveform at the center indicates if there is input signal for processing.

Active LFO modulations of the Offset parameters are shown in the lower half of the display.

Moving with the mouse over the display will stop the animation to easily set up **Low** and **High Offset**. Moving the mouse outside the display will continue the animation again.



Synthesis Filter Bank

Here you can control the parameters regarding to the synthesis functions of the Vocoder. Besides the static offset of the frequencies and the resonance of the bandpass filter you can also set up a modulation for the filter frequencies. An additional equalizer can Ein nachgeschalteter Equalizer kann optimize the articulation of the vocoder signal.



Low Offset

-100...+100%

Sets the relative offset of the lowest synthesis band compared to the lowest analysis band. A setting of 0 indicates that analysis and synthesis bands are exactly matching. Negative values indicate that the synthesis bands have been moved below the analysis bands, while positive values indicate that the highest synthesis band has been moved higher than the highest analysis band.

High Offset

-100...+100%

Sets the relative offset of the highest synthesis band compared to the highest analysis band. A setting of 0 indicates that analysis and synthesis bands are exactly matching. Negative values indicate that the synthesis bands have been moved below the analysis bands, while positive values indicate that the highest synthesis band has been moved higher than the highest analysis band.

Resonance

-100...+100%

Sets the steepness of the Vocoder filter. The higher the value, the more the Vocoder bands will resonate.

i Increasing the Vocoder **Resonance** can increase the intelligibility of the Vocoder signal.

Bandwidth -100...+100%

Controls the width of the synthesis bands. Negative values increase the width of the bands while positive values decrease the width. The default value of 0 is recommended for best vocoding results.

Modulation

The Synthesis filter bank offers a LFO for modulating the **Low** and **High Offset** parameter.

Modulation Active On / Off

Switches the modulation on or off. The switch lights red when activated.

Low Modulation -100...+100%

This parameter adjusts the amount of LFO modulation for the Low Offset position. The higher the value, the wider the modulation width. Lower setting adds a negative modulation.

i A LFO modulation of the Low Offset parameter is shown in the lower half of the Filter bank display.

High Modulation -100...+100%

This parameter adjusts the amount of LFO modulation for the High Offset position. The higher the value, the wider the modulation width. Lower setting adds a negative modulation.

i A LFO modulation of the High Offset parameter is shown in the lower half of the Filter bank display.

i By using the LFO for modulation, you can get a more natural, dynamic sound. For a more robotic sound, do not engage the LFO.

LFO Shape Diverse

Sets the type of waveform for the LFO. The following shapes are available:

- *Triangle* is perfect for smooth pitch or filter modulations.
- Even *Sine* is suited for smooth pitch or filter modulations.
- *Square* can be interesting for hard modulations or special effects.
- *Sawtooth* can generate interesting filter changes.

- *Random* generates random values and glides to them linearly.
- *S&H* (Sample & Hold) samples a random value and holds it until the next value is generated. If **Speed** is set to 0, a random value is generated on each new note.

LFO Speed *0.01...10 Hz / 8 Bar...1/64t*

Determines the frequency of the LFO. At low values, it might take some time for the LFO to perform a complete cycle.

When the LFO **Sync** parameter is set to *On*, you can adjust the **Speed** in musical values.

LFO Sync *On / Off*

When **Sync** is activated, the LFO is synced to the song tempo of the host application. The **LFO Speed** setting changed to offer musically meaningful values.

Equalizer

Lector offers an equalizer to adjust the Vocoder signal. This equalizer differs from a regular equalizer because it offers a proportional gain control for the synthesis bands.

This equalizer doesn't offer a shelving or peak characteristic, it provides a linear gain increasing or decreasing from the middle band to the lowest and highest band.

Equalizer Active *On / Off*

Switches the equalizer on or off. The switch lights red when activated.

Bass / Treble *-20...+20 dB*

Controls the overall balance of all synthesis bands by +/- 20 dB. Identical values will result in equal filter volume across the range; different values will change the low and high balance. All filters in-between are scaled proportionally.

Mid Frequency *1...numbers of bands*

Selects the band to be controlled by **Mid Gain**. The surrounding bands are scaled to match the level of the mid EQ.

Mid Gain *-20...+20 dB*

Cuts or boosts the selected **Mid Frequency** by +/- 20 dB.

i Adjusting the EQ levels will change the intelligibility of the vocoded signal. For more female tones, try higher values for **Treble**, and lower values for **Bass**. When you set **Mid Frequency** to 1 and **Mid Gain** to 0dB you can use **Treble** to receive a proportional increasing or decreasing of all bands. This is more comfortable as to work with three gain knobs.

Output

The Output section allows you to adjust the audio levels of the different sources in Lector.



Speech -∞...0dB

This fader will set the “dry” level of your analysis material. Typically, you will want to set the level to zero for classic vocoding sounds.

Carrier -∞...0dB

This fader allows you to mix the “dry” synthesizer sound in with the other two sources. Depending on the gain settings of the amplifier in the synth section, you will want to set this fader to maximum comfortable level when using Lector as a synthesizer. For classic Vocoder use, it should be set to zero.

Vocoder

$-\infty \dots 0dB$

This fader adjusts the level of your vocoding output. Typically, you will want to set this parameter close to a maximum level for those pure, classic vocoding sounds.

Effects

Lector offers an effects unit with four independent effects located behind the vocoder.

i All effects offers an Active button for switching on or off the corresponding effect. If activated the corresponding button lights red.



Overdrive

The Overdrive effect distorts the input signal by amplifying it drastically and clipping the resulting signal to a certain output level. The difference between this

Overdrive effect and the Drive parameter of the Filter section is that Drive affects one single voice, while this effect type distorts the summed output of the whole output signal. Therefore, the resulting effect is different as soon as you play more than one note and you should consider which is best suited to a particular sound program.

Overdrive Type *Diverse*

Determines the character of the drive. The following drive curves are available:

- **Tube** simulates the asymmetric distortion of a tube circuit.
- **Diode** generates a typical transistor distortion.
- **Clip** generates a typical digital clipping.

Drive *0...54dB*

Controls the amount of distortion the effect produces. Low values create no or only slight distortion while high values create heavier distortion.

Post Gain *-∞...0dB*

Controls the output level of the distorted signal.

Tone *11.56 Hz...Bypass*

Dampens the high frequency output of the Overdrive effect.

Chorus

A Chorus effect is generated by using modulated Comb filters that generate slightly detuned copies of the input signal and mix it into the output signal. The result sounds like an ensemble of several simultaneous sounds, like a choir as opposed to a single voice; hence the name Chorus. The detuning is generated by an internal LFO that can be controlled in speed and depth. The Chorus' high frequency output can be dampened with the Cutoff parameter. The Chorus effect offers besides the known 2 Stage mode (1 left, 1 right) also 4 (2 / 2) and 6 (3 / 3) Stage modes. This results in a far denser chorus effect perfectly suited for lush string or pad sounds. Furthermore, it sports feedback, which makes the Chorus also act like a Flanger.

Stages *2 / 4 / 6*

Sets the number of Chorus stages. Higher settings result in a denser effect.

Speed *0...20 Hz*

Sets the LFO speed of the Chorus effect.

Depth *0...100%*

Sets the modulation depth of the Chorus effect.

Feedback *-100...100%*

Feeds the Chorus output signal back to its input.

Delay

A Delay is an effect that produces echoes of the input signal. An additional LFO can modulate the delay times.

Sync *On / Off*

Synchronizes the Delay to the song tempo of your host application.

Time *0...2000ms oder 1/64...1t bars*

Sets the length of the Delay tap in milliseconds or (if **Sync** is activated) in useful musical values.

Mix *0...100%*

This parameter controls the volume ratio between the original signal and the effect output. If set to *0%*, the dry signal is sent to the outputs only so that no effects can be heard. Higher values will increase the effect signal. At maximum setting, the pure effect signal will be heard.

Feedback *0...100%*

Controls the amount of signal that is routed back into the Delay line. Lower values therefore produce fewer echoes than higher values.

Spread *-100...+100%*

Spreads the left and right delay output to half of the delay time maximum. Leftmost and rightmost settings (*-50 %* or *+50 %*) create a typical ping pong delay. While using settings of *±66%* or *±75%* will create a triplet or dotted ping pong delay.

Low Cut *41.18...22050 Hz*

Determines the frequency where the lower frequency parts are divided from the effect signal. A natural delay doesn't produce all frequencies equally, so the highpass

cuts the lower frequencies to produce a more realistic delay effect.

High Cut *41.18...22050 Hz*

Determines the frequency where the higher frequency parts are divided from the effect signal. A natural delay doesn't produce all frequencies equally, so the lowpass cuts the higher frequencies to produce a more realistic delay effect.

Mod Speed *0...20 Hz*

The delay effect can be modulated in its delay time with the integrated LFO. Speed determines the frequency of this LFO.

Mod Depth *0.0...100%*

Controls the modulation depth when delay time is changed by means of the LFO. The depth ranges from no delay to the amount set by the Delay parameter.

Reverb

The Reverb effect is probably the most widely used effect in music production. It is used to add a realistic ambience to clean and dry audio recorded in a studio. Very

complicated mathematical algorithms are needed to simulate the complexity of a natural reverb. As a result, good reverb processors are very expensive. The Lector's reverb effects don't intend to simulate the perfect natural room, rather they are an addition to the sound synthesis to make it more 3 dimensional and expressive

Size *0...100 m*

Determines the length of the longest room side in meters. Lower settings simulate a normal room while higher settings simulate a big hall or church.

Mix *0...100%*

This parameter controls the volume ratio between the original signal and the effect output. If set to 0%, the dry signal is sent to the outputs only so that no effects can be heard. Higher values will increase the effect signal. At maximum setting, the pure effect signal will be heard.

Diffusion *0...100 %*

Determines the consistency of different room materials. Lower settings produce a cold strong reflection like a metal or tile wall. Higher settings make a warm and thick reverb as if the signal were reflected by an uneven

surface. Settings above 100 change the room characteristic even further.

Decay *0...100*

Feedback of the reverb network. Recommended settings are between 1/4 and half its range. Higher values create audible ringing especially with small **Size** settings, which don't correspond to any natural reverb response, but might be used for simulations of spring or plate reverbs. Maximum setting results in a very long reverb.

Low Cut *0...22000 Hz*

Determines the frequency where the lower frequency parts are divided from the effect signal. A natural reverb doesn't produce all frequencies equally, so the highpass cuts the lower frequencies to produce a more realistic reverb effect.

High Cut *0...22000 Hz*

Determines the frequency where the higher frequency parts are divided from the effect signal. A natural reverb doesn't produce all frequencies equally, so the lowpass cuts the higher frequencies to produce a more realistic reverb effect.

Low Damp *0...22000 Hz*

High pass filter in the feedback loop of the late reverb. Determines how fast the room simulation dampens the lower reverb frequencies. This phenomenon makes a natural echo sound somewhat hollow. Higher values of the Damping parameter mean that more low frequencies will be dampened.

High Damp *0...22000 Hz*

Low pass filter in the feedback loop of the late reverb. Determines how fast the room simulation dampens the higher reverb frequencies. This phenomenon makes a natural echo sound somewhat hollow. Higher values of the Damping parameter mean that more high frequencies will be dampened.

Appendix

What is a Vocoder?

A Vocoder is a circuitry to process the frequency content of one signal with the frequency content of another signal. This is done by splitting up the analysis signal (also known as Speech signal) with bandpass filters into several frequency bands. The volume of each frequency band is then evaluated with an envelope follower and the resulting envelope is used to control the volume of a bandpass filter that processes the synthesis signal (also known as Carrier signal). The synthesis signal therefore is filtered by as many bandpass filters as the analysis signal was so that each analysis filter finds its synthesis filter counterpart.

This technique was already developed in World War II, but the plan was to use it to encrypt speech when transmitted by radio, rather than using it to create musically useful sounds.

How to use the Vocoder in a musical context?

You should feed a Vocoder with speech, drum loops or any other collection of external audio material you like.

The results will always be complex and interesting, sometimes near to the original signal and sometimes far away.

For the Vocoder, you always need two signals, a signal that is analyzed and another signal that is used to be processed by the synthesis filters. The analyzed signal always comes one source, either an audio track or an instrument track. The synthesis signal mostly will be delivered by the integrated synthesizer, but if you like to use a different synthesis signal, you can route any desired audio or instrument signal into Lector.

Use Cases

Lector in Latch Mode

This use case is great for quickly and effectively altering the sound of a voice (especially “robot voices”) or percussion, as the pitch would not be tracked or controlled externally.

1. Choose a track where you would like the audio to be processed. Alternatively you can choose a virtual instrument track (e.g. sample player).
2. load Lector directly in one of the channel’s insert slots.

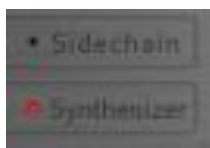
3. Check the settings in the Output section. Only “Vocoder” should be up, as shown in the screenshot.



4. Select *Latch* in the Trigger section.



5. Make sure that you have selected *Synthesizer* in the Input section.



6. Start the playback of your host application.

7. Depending on your source signal, enhance the sibilant frequencies to improve intelligibility by adjusting the UV Detector. Make sure it's on (indicated by the LED) and adjust the values to your requirements.

Classic Vocoder controlled by MIDI notes

This is the typical setup for using the Vocoder musically, i.e. creating a melodic Vocoder effect, typically using the internal synthesizer.

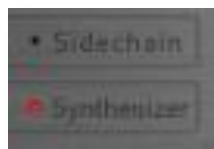
1. Choose a track where you would like the audio to be processed. Alternatively you can choose a virtual instrument track (e.g. sample player).
2. Load Lector directly in one of the channel's insert slots.
3. Check the settings in the Output section. Only “Vocoder” should be up, as shown in the screenshot.



4. Select *Normal* or *Single* in the Trigger section, depending on if you would like the Vocoder to be monophonic or polyphonic.



5. Make sure that you have selected *Synthesizer* in the Input section.



6. Create a MIDI track in your host application and make sure that it is routed to Lector. Keep also in mind that the MIDI track recognized MIDI data from a connected MIDI master keyboard.

7. Start the playback of your host application and play some MIDI notes on your master keyboards. You can also record MIDI notes for controlling the Lector vocoder.

8. Depending on your source signal, enhance the sibilant frequencies to improve intelligibility by adjusting the UV Detector. Make sure it's on (indicated by the LED) and adjust the values to your requirements.

Lector and an external Carrier Signal

This setup is great for processing the envelope of one track onto another. For best results, you should be using pad-like continuous sounds as carrier and percussive signals as analysis signal.

1. Choose a track where you would like the audio to be processed. Alternatively you can choose a virtual instrument track (e.g. sample player). Load Lector directly in this channel's insert slots.

2. Now load the Lector Carrier plug-in on the desired carrier channel (the channel where you would like the audio to be processed. This could also be any audio or instrument track. The Lector Carrier plug-in doesn't have any user interface and will look different in any application. Make sure it is active.

3. Check the settings in the Output section. Only "Vocoder" should be up, as shown in the screenshot.



4. Select *Normal* or *Single* in the Trigger section, depending on if you would like the Vocoder to be monophonic or polyphonic. You can also set it to *Latch* if you want to trigger the Lector vocoder permanently with the MIDI note C3.

5. Make sure that you have selected *Sidechain* in the Input section. If activated, Lector will not use the internal synthesizer, but the Carrier signal as sound source.
6. Start the playback of your host application.
7. Depending on your source signal, enhance the sibilant frequencies to improve intelligibility by adjusting the UV Detector. Make sure it's on (indicated by the LED) and adjust the values to your requirements.

Glossary

Aliasing

Aliasing is an audible side effect arising in digital systems as soon as a signal contains harmonics higher than half the sampling frequency.

Amount

Describes to which extent a modulation influences a given parameter.

Amplifier

An amplifier is a component that influences the volume level of a sound via a control signal. This control signal is often generated by an envelope or an LFO.

Analysis Signal

The analysis signal of a vocoder is the signal that is being analyzed. Typically, this signal can be speech or a vocal performance.

Attack

An envelope parameter. “Attack” is a term that describes the ascent rate of an envelope from its starting point to the point where it reaches its highest value. The Attack phase

is initiated immediately after a trigger signal is received, i.e. after you play a note on the keyboard.

Carrier Signal

The carrier signal is the sound source that carries the analysis sound. In the case of Lector, the carrier will either be the built-in synthesizer or an external channel when the Lector Carrier Plug-In or the host integrated sidechain function is used.

Clipping

Clipping is a sort of distortion that occurs when a signal exceeds its maximum value. The curve of a clipped signal is dependent of the system where the clipping takes place. In the analog domain, clipping effectively limits the signal to its maximum level. In the digital domain, clipping is similar to a numerical overflow and so the polarity of the signal’s part above the maximum level is negated.

Cutoff

The cutoff frequency is a significant factor for filters. A low pass filter dampens the portion of the signal that lies above this frequency. Frequencies below this value are allowed to pass through without being processed

Decay

“Decay” describes the descent rate of an envelope once the Attack phase has reached its zenith and the envelope drops to the level defined for the Sustain value.

Envelope

An envelope is used to modulate a sound-shaping component within a given time frame so that the sound is changed in some manner. For instance, an envelope that modulates the cutoff frequency of a filter opens and closes this filter so that some of the signal's frequencies are filtered out. An envelope is started via a trigger, usually a fixed trigger. Normally, the trigger is a MIDI Note. The classic envelope consists of four individually variable phases: Attack, Decay, Sustain and Release. This sequence is called an ADSR envelope. Attack, Decay and Release are time or slope values, and Sustain is a variable volume level. Once an incoming trigger is received, the envelope runs through the Attack and Decay phases until it reaches the programmed Sustain level. This level remains constant until the trigger is terminated. The envelope then initiates the Release phase until it reaches the minimum value.

Filter

A filter is a component that allows some of a signal's frequencies to pass through it and dampens other frequencies. The most important aspect of a filter is the filter cutoff frequency. Filters generally come in four categories: low pass, high pass, band pass, and band stop. A low pass filter dampens all frequencies above the cutoff frequency. A high pass filter in turn dampens the frequencies below the cutoff. The band pass filter allows only those frequencies around the cutoff frequency to pass, all others are dampened. A band stop filter does just the opposite, i.e. it dampens only the frequencies around the cutoff frequency. The most common type is the low pass filter.

LFO

LFO is an acronym for low-frequency oscillator. The LFO generates a periodic oscillation at a low frequency and features variable waveshapes. Similar to an envelope, an LFO can be used to modulate a sound-shaping component.

Manual

Manuals for Waldorf products are written during night-shifts by solitued people which fear the daylight. Please

appreciate this by using this manual here and there. We recommend to deposit it in your restroom within reach.

MIDI

The acronym MIDI stands for “musical instrument digital interface.” It was developed in the early '80s so that diverse types of electronic musical instruments by different manufacturers could interact. At the time a communications standard for heterogeneous devices did not exist, so MIDI was a significant advance. It made it possible to link all devices with one another through simple, uniform connections.

Essentially, this is how MIDI works: One sender is connected to one or several receivers. For instance, if you want to use a computer to play the Pulse, then the computer is the sender and the Pulse acts as the receiver. With a few exceptions, the majority of MIDI devices are equipped with two or three ports for this purpose: MIDI In, MIDI Out and in some cases MIDI Thru. The sender transfers data to the receiver via the MIDI Out jack. Data are sent via a cable to the receiver's MIDI In jack.

MIDI Thru has a special function. It allows the sender to transmit to several receivers. It routes the incoming signal to the next device without modifying it. Another device is simply connected to this jack, thus creating a chain through which the sender can address a number of

receivers. Of course it is desirable for the sender to be able to address each device individually. Consequently, there is a rule which is applied to ensure each device responds accordingly.

MIDI Channel

This is a very important element of most messages. A receiver can only respond to incoming messages if its receive channel is set to the same channel as the one the sender is using to transmit data. Subsequently, the sender can address specific receivers individually. MIDI Channels 1 through 16 are available for this purpose.

MIDI Clock

The MIDI Clock message determines the tempo of a piece of music. It serves to synchronize processes based on time.

Modulation

A modulation influences or changes a sound-shaping component via a modulation source. Modulation sources include envelopes, LFOs or MIDI messages. The modulation destination is sound-shaping component such as a filter or a VCA.

Note on / Note off

This is the most important MIDI message. It determines the pitch and velocity of every generated note. The time of arrival is simultaneously the start time of the note. Its pitch is derived from the note number, which lies between 0 and 127. The velocity lies between 1 and 127. A value of 0 for velocity is similar to “Note Off”.

Panning

The process of changing the signal's position within the stereo panorama.

Pitchbend

Pitchbend is a MIDI message. Although pitchbend messages are similar in function to control change messages, they are a distinct type of message. The reason for this distinction is that the resolution of a pitchbend message is substantially higher than that of a conventional Controller message. The human ear is exceptionally sensitive to deviations in pitch, so the higher resolution is used because it relays pitchbend information more accurately.

Program Change

These are MIDI messages that switch sound programs. Program numbers 1 through 128 can be changed via program change messages.

Release

An envelope parameter. The term “Release” describes the descent rate of an envelope to its minimum value after a trigger is terminated. The Release phase begins immediately after the trigger is terminated, regardless of the envelope's current status. For instance, the Release phase may be initiated during the Attack phase.

Resonance

Resonance or emphasis is an important filter parameter. It emphasizes a narrow bandwidth around the filter cutoff frequency by amplifying these frequencies. This is one of the most popular methods of manipulating sounds. If you substantially increase the emphasis, i.e, to a level where the filter begins self-oscillation, then it will generate a relatively clean sine waveform.

Sibilants

Sibilant sounds are characterized by a hissing sound. These include the english pronunciation of such consonants as "f", "s", "z", or "th".

Sustain

The term "Sustain" describes the level of an envelope that remains constant after it has run through the Attack and Decay phases. Sustain lasts until the trigger is terminated.

Trigger

A trigger is a signal that activates events. Trigger signals are very diverse. For instance, a MIDI note or an audio signal can be used as a trigger. The events a trigger can initiate are also very diverse. A common application for a trigger is its use to start an envelope.

Unvoiced Detector

Unvoiced signals have the tonal characteristic of the full frequency band, such as the consonant sounds that cover almost the entire frequency spectrum, such as "f", "s", "t" and so forth. The Unvoiced Detector of an vocoder will add a small amount of noise to the signal, improving the intelligibility of the sibilant frequencies.

Volume

The term describes a sound's output level.

Product Support

If you have any questions about your Waldorf product, feel free to contact us via one of the four options listed below:

1. Send us an email message. This is the most efficient and fastest way to contact us. Your questions will be forwarded immediately to the resident expert and you will quickly receive an answer.

support@waldorfmusic.de

2. Send us a letter. It will take a bit longer, but it is just as dependable as an email.

Waldorf Music GmbH
Lilienthalstr. 7
53424 Remagen, Germany