BLOFELD

SYNTHESIZER

user manual

waldorf

Content

Foreword

Thank you for purchasing the Waldorf Blofeld Plug-In. You now own a virtual synthesizer that emulates the the famous Waldorf Blofeld hardware synthesizer featuring a wide range of unique sounds from analog to digital.

What to read?

The biggest problem with any manual is to find a way 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.

Anyone reading the following manual is in for a lot of fun while learning about and working with the Waldorf Blofeld Plug-In.

And now have fun with your Blofeld Plug-In!

Your Waldorf Team

Hint

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Revision:	1.0, December 2024

Please visit our website for further support and downloads for your Blofeld Plug-In:
waldorfmusic.com

We would like to thank

Julia Bach, Evi Carroll, Claudio Chiriatti, Karsten Dubsch, Willie Eckl, Joachim Flor, Kevin Junk, Roger Keller, Miroslav Pindus, Frank Schneider, Winfried Schuld, Michael von Garnier, Kurt ‘Lu’ Wangard, Rolf Wöhrmann, Haibin Wu and anyone we have forgotten.

Overview

1) Head-up section

2) Oscillators section

3) Mixer & Amp section

4) Filter section

5) Envelope & LFO section

6) FX section

About this Manual

This manual was written to help you to become familiar with your Blofeld Plug-In synthesizer. It will also aid experienced users with routine tasks.

To avoid confusion, the terminology in this manual is based on the Blofeld hardware parameter names. You will find a glossary at the end of this manual; it explains the various terms used.

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.

Instruction – Follow these guidelines to execute a desired function.

Example – Real-world examples to try out.

Highlighted Control Features and Parameters

All of Blofeld plug-in buttons, controls and parameters are highlighted in **bold** letters throughout the manual.

Examples:

- Click on the **Synth** button.
- Turn the **Cutoff** knob.

Blofeld Plug-In's different modes and parameter pages are illustrated in a depiction of the display.

Installation & Activation

System Requirements for Windows

In order to be able to use Blofeld plug-in, you will need at least:

- PC with a Intel or AMD processor.
- Windows 7 or newer.
- VST 2.4 or VST3 compatible host application. This must be correctly installed on your computer.
- AAX compatible host application. This must be correctly installed on your computer.

Please also observe the system requirements of your host application!

The Blofeld plug-in runs only within 64 bit host applications.

Installation under Windows

1. Refer to the folder where the downloaded Blofeld plug-in zip archive is located.

2. Double click on the archive file to extract it.

3. Double click on the Blofeld plug-in Installer icon. This launches a special installation program.

4. Follow the on-screen instructions.

After installing Blofeld plug-in you will have to activate the program on your computer. Please refer to the chapter “Activation of Blofeld plug-in”.

System Requirements for macOS

In order to be able to use the Blofeld plug-in, you will need at least:

- Mac with Intel processor

  or

  Mac with Apple Silicon processor.
- macOS 10.14 or newer.
- VST 2.4 compatible host application or a VST3 compatible host application. This must be correctly installed on your computer.

  or
- An AudioUnit 2.0 compatible host application. This must be correctly installed on your computer.

  or

- An AAX compatible host application. This must be correctly installed on your computer.

> ⓘ Please also observe the system requirements of your host application!
>
> ⓘ The Blofeld plug-in runs only within 64 bit host applications.

Installation under macOS

Proceed as follows to install Blofeld plug-in:

1. Refer to the folder where the downloaded Blofeld plug-in zip archive is located.

2. Double click on the archive file to extract it.

3. Double click on the Blofeld plug-in Installer DMG icon. This launches a special installation program.

4. Follow the on-screen instructions.

> ⓘ After installing Blofeld plug-in you will have to activate the program on your computer. Please refer to the next chapter „Activation of Blofeld plug-in“.

Activation of Blofeld Plug-In

Blofeld plug-in uses a copy protection system based on the users email address as well as a personalized serial number.

Proceed as follows to activate Blofeld plug-in:

1. Start your host application.

2. Load the Blofeld plug-in plug-in as instrument in your DAW.

3. An input field occurs. In the upper field, please enter the email address that was used for purchasing Blofeld plug-in. In the lower field, please enter the 20 digit serial number which you have received with your purchase.

4. Click on the OK button to confirm your data. From now on, Blofeld plug-in is authorized for this computer.

> ⓘ If you want to use Blofeld plug-in on other computers, please proceed in the same way as described above.

Basic Operations

Worth Knowing about Single and Multi Mode

The Blofeld plug-in offers a sound structure with either a single sound program or a multi program with up to 16 sound parts. All parts are always saved for each multi sound program. The 16 parts can be played simultaneously regardless of the incoming MIDI channel. More about loading/saving Single and Multi Mode with Multi programs can be found later in this manual.

General Operation

Blofeld plug-in has been optimized for a screen resolution of at least 1024x768 pixels. You can click-drag one of the resize handles at the upper left and lower right corners of the plug-in window to scale the Blofeld plug-in window to your desired size.

Blofeld plug-in has various on-screen controls. The knobs can be moved with greater accuracy by holding down the CTRL/CMD key on your computer keyboard while moving your mouse. Double-clicking the corresponding fader or knob resets the parameter to the default value. Mouse scroll wheel is supported for all continuous controls. All graphic displays (filter curve, envelopes) can be edited with your mouse.

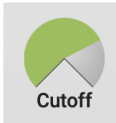
Double-click on any value field opens a text edit field for entering values with the computer keyboard.

⚠ **Warning:** Never name one of your Blofeld plug-in sound programs with the number sequence 007, otherwise you could end up in the chimney of an industrial plant or at least get a scar on your face. Always remember: You only live twice!

Control Elements

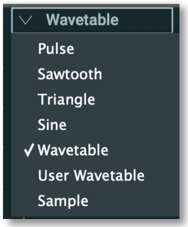
Using Blofeld Plug-In's controls is simple. There are some different types of control elements:

Dials



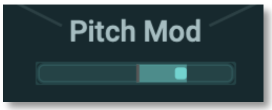
To set a value, click on the dial, hold down the mouse button and move the mouse up and down. Pressing CTRL/CMD while holding the mouse button allows finer adjustments. If you hover over a dial using the mouse, the current parameter value appears below the dial.

Pop-Down Menus



Click on the corresponding parameter/button to open a pop-down menu where you can choose the desired value/selection.

Sliders



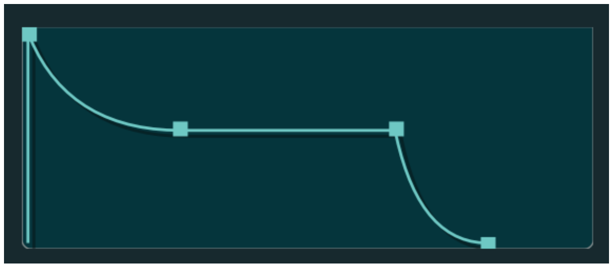
To set a value, click on the slider, hold down the mouse button and move the mouse up and down / left and right. Pressing CTRL/CMD while holding the mouse button allows finer adjustments. If you hover over a sliderusing the mouse, the current parameter value appears below the dial

Switches

If a function/parameter is active, it is highlighted and all others within the function block will be dark. To activate a function, click on its parameter name.

Graphic Displays

Click into the graphic representation and drag the mouse to continuously and smoothly change the corresponding parameters. More on the different graphic display types can be found in the corresponding manual sections.



The Parameter Pages

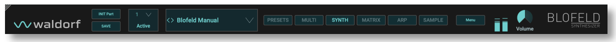
Blofeld Plug-In consists of numerous sound-shaping and utility components. The following pages describe all parameters in detail. Additional information can be found in the Appendix.

- Sound synthesis: Oscillators, Ring Modulator, Noise Generator, Mixer, Filters, Amplifier, Effects. These modules represent the audio signal flow. Sound generation actually occurs within the Oscillators. They produce square, sawtooth, triangular, sine and wavetables. The Mixer follows the Oscillators in the signal chain, which is where the Oscillators' output signals are mixed. Noise can also be added to the mix. The Filter then shapes the sound by amplifying (boosting) or attenuating (dampening) certain frequencies. The Amplifier and the Effects are located at the end of the signal chain. They determine the overall volume of the signal and add some effects like chorus, flanger, delay etc.

- Modulators LFOs, Envelopes, Modifiers, Modulation Matrix. These modules are called Modulators. These modules are called Modulators. The Modulators are designed to manipulate or modulate the sound generating components to add dynamics to sounds. The Low-frequency Oscillators (LFO) are designed for periodic or recurring modulations while Envelopes are for modulations that occur once on each note. These generators are assigned to parameters through the Modulation Matrix and influence these parameters to alter a sound. In addition, the Modifier unit can process various mathematical operations and functions on the modulation signals.

The following pages describe all parameters in detail. Additional information can be found in the Appendix.

The Head-Up Section



The Head-up section provides the global overview and includes the following options:

1. The **INIT Part** button initializes the current single sound or the current selected multi part to standard values.

2. The **Save** button opens a window to save the current Blofeld Plug-In single. You can select a desired target group for savinbg your sound.

3. The **Multi** pop-up menu is only available, if the Multi Mode option is active. Click on it to open a pop-up menu, where you can select the desired Multi part from 1 to 16.

More about Multi Mode can be found on page 59.

4. The **Sound** pop-up menu opens a list with all presets from the current selected sound bank. The **Prev / Next** buttons steps through the sound programs of this current selected bank.

> Please note that some host applications as Steinberg Cubase use an additional sound data management. It offers you an alternative way to load and save sounds. Please refer to the corresponding manual chapter of your host application.

5. The six **Selection** buttons opens the corresponding parameter or set up page oft he Blofeld plug-in.

> More information about the different Blofeld pages can be found later in that manual.

6. The **Menu** button opens a pop-up menu including the following options:

- **Blofeld Hardware** opens the Blofeld Hardware Sync option for a deeper integration in the hard-ware version.

> More information about the hardware integration option can be found later in that manual.

- **MIDI Mapping** opens a window, where you can connect external MIDI controllers to your Blofeld plug-in parameters. Click on MIDI Learn, choose your desired parameter and turn your MIDI hard-ware controller. In the MIDI Mapping window, you can also export and import MIDI Mappings.
- **Window Size** sets the plug-in window size to the corresponding fix size. Three options are available (Small, Standard & Large)
- **Show Manual** opens your favorite web browser and directly navigates to the location of this PDF manual (if you computer is connected to the inter-net).
- **Buy** opens your favorite web browser and loads our Waldorf online shop website (if you computer is connected to the internet).
- **About** opens a window with the Blofeld Plug-In version and additional information.

7. The **Volume** knob controls the overall volume of the plug-in.

8. Click on the **Blofeld logo** to open a window with Blofeld Plug-In version and additional information.

The Sound Browser Page

Click on the **Presets** button to open the integrated sound browser page. Here, you can search by name, author, and tags

The left section offers different search options:

- In the **Search** field, you can enter text or numbers. The results list in the central area will be updated immediately with the corresponding search results.
- Click on **Group Selection** to select the desired sound banks, that should be displayed in the results list.
- **Author Filter** or **Tag Filter** open a pop up menu for further search restrictions.

In the **Saved Search** section you can see all your stored searches (see also **Option** menu).

The central area contains the **Search Results** list. Here, you can double-click on a sound entry to load it into the Blofeld plug-in. Above the result list, you also see your search tags. Click on the correspondig tag to remove it from your search.

In the right area you get further informtation oft he current selected sound. If a sound is loaded, you can edit the information for Group, Author, Tags or description. Click on the corresponding pencil symbol to make your desired edits.

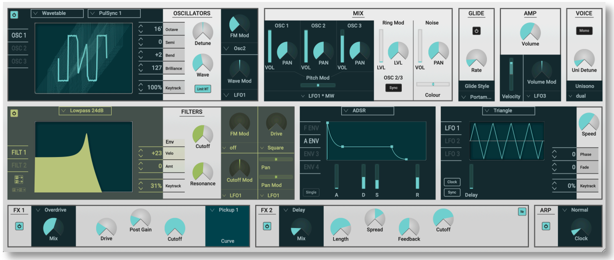
The **Option** menu at the top of this area contains the following options:

- **New Init Preset** initializes the current selected sound program and sets all parameters to standard values.
- **Delete Preset** deletes the current selected user preset.
- **Copy part** copies all settings of the current selected multi part into temporary buffer.

- **Paste part** pastes all multi part settings of the multi part in temporary buffer onto the current selected instrument.
- **Group** offers three more options: **Add Group** adds a new group of seletec sounds. **Remove Group** removes a selected group of sounds. **Rename Group** renames the current selected group.
- **Save Current Search** saves the currend made search with all filters and tags. A Saved Search will be shown in the left area of the Preset browser page.
- **Import Presets** opens a window to import a Sound Bank file or a .wpc file.
- **Import Presets from BPZ** opens a window to import sound programs that were saved with the **Export** option (see below).
- **Export Selected Presets** opens a window to export the currently selected preset(s).
- **Export Presets in List** opens a window to export the complete sounds form the results list.
- **Factory Reset DB** reset the Blofeld preset databank.

The Synth Page

Click on the **Synth** button head-up section to open the main page with the synthesis parameters.

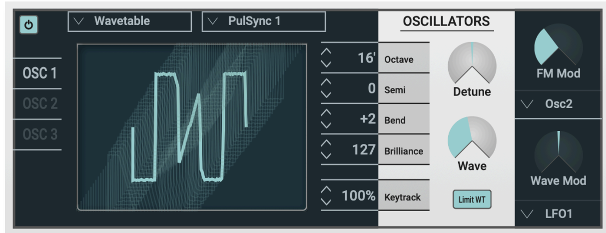


The Oscillator Section

The Blofeld offers three oscillators that nearly use the same parameters for editing.

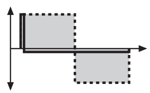
> Choose the desired oscillator by clicking on the corresponding OSC button. The current selected oscillator button will light up.

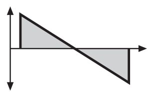
The **Osc** button disables the corresponding oscillator to reduce your CPU consumption.

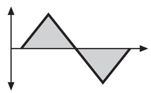


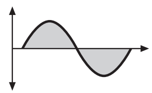
Oscillator Shape Pop-Up Menu

Sets the type of waveform to be generated by the Oscillator. The parameter is called **Shape** instead of "waveform", because it doesn't necessarily set only one waveform, but sets a specific oscillator model that could produce a number of waveforms depending on other settings. A good example for this behaviour is the *Pulse* shape. However, the term "waveform" is used interchangeably throughout the manual. The following shapes are currently available:


Pulse
(variable Pulse Width)


Sawtooth


Triangle


Sine

- *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 **PW** and **PW Mod** are used to change the pulsewidth of the waveform. Furthermore, the modulation destinations *Osc1 PW Wave*, *Osc2 PW Wave* or *Osc3 PW Wave* gain functionality, depending on which oscillator is set to *Pulse*. The **Brilliance** parameter is for adding more brilliance to the Pulse wave.

- *Saw* selects the sawtooth waveform. A Sawtooth wave has all the harmonics of the fundamental frequency in descending magnitude. It doesn't have any further parameters. This waveform is pleasing to the ear. The parameter **Brilliance** is for adding more brilliance to the Saw wave.
- *Triangle* selects the triangular waveform. The triangle mainly consists of the odd harmonics with very low magnitudes. It doesn't have any further parameters
- *Sine* consists of the fundamental frequency only. It has no harmonics at all. It doesn't have any further parameter.
- The *Wavetable* generator can create more than 65 different wavetables from earlier Waldorf synthesizers. Wavetables can be selected in the pop-up menu right besides the **Shape** pop-up menu. A complete list can be found in the appendix of this manual. Please read also the introduction to the Wavetables. When *Wavetable* is selected, the parameters **Wave** and **Wave Mod** serve to select the start point of the waves. Furthermore, the modulation sources *01PW* and *02PW* are active subject to which Oscillator is set to the wavetable.
- *User Wavetable* allows you to load your on user wavetables and use it in the same way as the factory wavetables.
- *Samples* plays a sample or multisamples from sample memory. Please note that the Sample playback is only available for Oscillator 1 and 2

Oscillators should be deactivated when not using them. This saves CPU power.

Octave

Sets the basic pitch of the corresponding oscillator in steps of an octave. The **Octave** setting is shown as register value, a common measurement based on the length of organ pipes. The reference pitch for the oscillator is generated at MIDI note A3 (note no. 69) when **Octave** is set to *8'*, **Semitone** and **Detune** are set to *0*, **Keytrack** is set to *100%* and no pitch modulation is applied. In this case, the oscillator's frequency will be the same as set in the **Global Parameters** (usually 440Hz).

Semitone

Sets the pitch of the corresponding oscillator in semitone steps. The standard setting for this parameter is *0*, but there are cases where different values are interesting as well.

Bend

Determines the pitchbend range for MIDI Pitchbend messages in semitones for the selected oscillator.

Brilliance

Determines the brilliance of the ocillator models Saw, Pulse and all Wavetables.

The models Saw and Pulse don´t play simple waveforms as a sampler. It is based on exact emulations of analog components with digital algorithms. The **Brilliance** parameter changes defined parameters of these models to point out the higher frequencies. A value of *64* is nearly similar to the brilliance of the oscillators produced by the former Waldorf Q and microQ synthesizers.

Many people can´t hear the highest frequencies of our oscillators. So don´t be surprised if you can´t hear any difference when using **Brilliance**. Ask infants, dogs or bats to help you adjusitng the oscillator brilliance.

Wavetables are using 64 harmonics. In this case **Brilliance** can add harmonics for lower frequencies. Higher **Brilliance** values come close to the sound synthesis process of the earlier wavetable synthesizers as PPG Wave and Waldorf Wave. The former Waldorf Microwave II/XT offered a similar parameter called „Quantize“.

Keytrack

Determines how much the pitch of the selected oscillator depends on the MIDI note number. The reference note for Keytrack is E3, note number 64. For positive settings, the oscillator pitch rises on notes above the reference note, for negative settings the oscillator pitch falls by the same amount and vice versa. A setting of *+100%* corresponds to a 1:1 scale, e.g. when an octave is played on the keyboard the pitch changes for the same amount. Other settings than *+100%* make sense especially when using ring modulation, FM or oscillator synchronization. Try to use values in the range *0%...+75%* or even negative settings for one oscillator while leaving the second at *+100%* Keytrack.

Detune

Fine-tunes the oscillator in steps of 1/128th 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.

- A low value of *±1* results in a slow and soft Flanger effect.
- Mid-ranged settings of *±5* are perfect for pads and other fat sounding programs.
- High values of *±12* or above will give a strong detune that can be used for accordions or effect sounds.

PW

Only available when the Pulse shape is selected. Sets the pulse width of the Pulse waveform. The value *0* is equivalent to a pulse ratio of <1%, the value *127* is equivalent to 50%.

Wave

Not available when the Pulse shape is selected. Determines the start point of a wavetable at which *0* selects the first of up to *128* Waveforms. If Sample is selected, this dial determines the startpoint of the sample. If you select any waveform, this parameter does not have any effect.

Limit WT

Determines, if the classic wavetables (from Resonant) are played with or without the additional analog waveforms. The analog waveforms are a relict from the former PPG Wave and Microwave synthesizers. If you don´t want to use these waveforms, use **Limit WT** to mask it.

FM Mod

Sets the amount of frequency modulation that is applied to the oscillator by the selected **FM source**. The sound will get more metallic and sometimes even drift out of tune, especially if Oscillator 2 is used as FM Source for Oscillator 3 and **Sync** is activated. To avoid unusable detune, use a triangular or sine waveform for the FM Source.

FM Source

Selects the source for the frequency modulation of the selected oscillator.

You can create nice E-Piano sounds when you use a high pitched oscillator as **FM Source** and set its **Keytrack** to a value between *+000%* and *+050%*.

The use of Noise is very interesting as **FM Source** on a high pitched oscillator playing a sine or triangle waveform. With a low **FM Mod**, the oscillator starts to sound dirty or airy while higher amounts create a coloured noise similar to a filter with high resonance. A side benefit is that the filters are then still free for other purposes.

If you want to bias **FM** over the keyboard so that higher notes aren't modulated as strongly as lower notes, use the **Matrix** and apply *Keytrack* to the respective oscillator FM with a negative amount.

PW Mod

Only available when the Pulse shape is selected. PW stands for pulsewidth modulation. This parameter determines the amount of modulation that is applied to the pulsewidth of the oscillator's square wave.

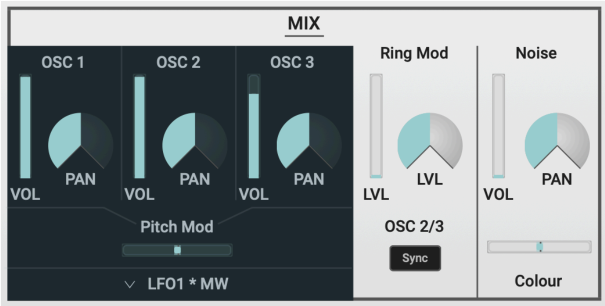
Wave Mod

Not available when the Pulse shape is selected. If a wavetable is selected, this determines the amount of the wavetable modulation. If you select any waveform other, this parameter does not have any effect.

PW Mod / Wave Mod Source

Selects the source of the pulsewidth modulation or the wavetable modulation. Common sources for pulsewidth modulation are envelopes and LFOs, but other sources like the modulation wheel or aftertouch can create nice effects as well.

The Mix Section



Volume Osc 1 / 2 / 3

Controls the volume of the corresponding Oscillator.

Pan Osc 1 / 2 / 3

Determines the ratio of the corresponding Oscillator's signal that is sent to the inputs of Filter 1 and Filter 2. If set to *F1 64*, the signal is sent to Filter 1 only. Higher values will increase the amount of signal that feeds Filter 2 and decrease the amount of signal that feeds Filter 1. If set to *mid*, both filters will receive the same signal level. If set to *F2 63*, the signal is sent to Filter 2 only.

Pitch Mod

Sets the amount of pitch modulation for all oscillators. Positive amounts will raise the pitch when positive modulation is applied, e.g. by pressing aftertouch on the keyboard. Negative amounts will lower the pitch when positive modulation is applied.

Pitch Source

Selects the source of the pitch modulation for all oscillators. A common source for pitch modulation is an LFO whose strength is controlled by the modwheel or aftertouch.

To create a common pitch vibrato that is controlled by the modwheel, set **Pitch Source** to *LFO1*MW* with **Pitch Amount** set to around *+20*.

To create a sound whose pitch glides in, set **Pitch Source** to a decaying *Envelope* with **Pitch Amount** set to around *–25*.

If you want to modulate the pitch of individual oscillators, you can do so with the **Matrix**. See the respective section for details.

Ring Modulation

The following parameters refer to the ring modulator.

Level

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:

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 100Hz and the second set to 101Hz, the resulting ring modulation is 201Hz and 1Hz, and 1Hz is very low.

Ring Modulation can be very interesting when a slow pitch modulation is applied to one oscillator, i.e. a decaying *Envelope*. This creates spacy effect sounds.

For an E-Piano sound, you might apply Ring Modulation when one high pitched oscillator's **Keytrack** is lowered to i.e. *50%*.

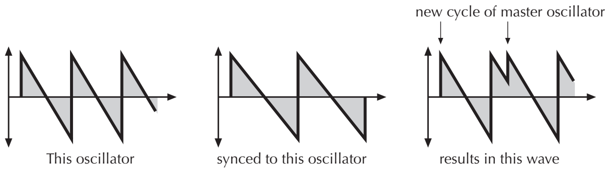
If you turn down the pitch of one oscillator markedly, you can get an effect very similar to Amplitude modulation. Use this for sounds with a periodic element if you wish.

Pan

Determines the ratio of the ring modulator's signal that is sent to the inputs of Filter 1 and/or Filter 2. If set to *F1 64*, the signal is sent to Filter 1 only. Higher values will increase the amount of signal that feeds Filter 2 and decrease the amount of signal that feeds Filter 1. If set to *mid*, both filters will receive the same signal level. If set to *F2 63*, the signal is sent to Filter 2 only.

Sync Osc 2/3

Enables or disables oscillator synchronization. When enabled, Oscillator 2 acts as a slave that is controlled by Oscillator 3, the master. Each time Oscillator 3 starts a new cycle, it sends a trigger signal to Oscillator 2, forcing it to restart its waveform cycle, too. As a result, interesting sound effects may be generated, especially when both oscillators are operating at different pitch settings. Using additional pitch modulation by envelopes, LFO, or Pitch bend will lend further movement to sync sounds. The following picture illustrates the principle of oscillator synchronization in a simplified way:



> Use **Sync** for Lead or Solo sounds. Set Oscillator 2 to play one octave and 7 semitones higher, apply an envelope to its pitch with positive amount and you get a screaming sync sound.

> **Sync** can also be very interesting on arpeggio sounds. Apply a slow clocked LFO to Oscillator 2 pitch and the arpeggio starts to move.

Noise

The following parameters refer to the noise generator.

Volume

Volume of the noise generator Noise is a fundamental source for any kind of analog-type percussion. Also, wind and other sound effects can be created by using the noise generator.

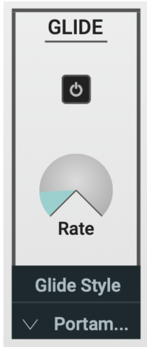
Pan

Determines how loud the noise signal is sent to the inputs of Filter 1 and Filter 2. If set to *F1 64*, the signal is sent to Filter 1 only. Higher values will increase the amount of signal that feeds Filter 2 and decrease the amount of signal that feeds Filter 1. If set to *mid*, both filters will receive the same signal level. If set to *F2 63*, the signal is sent to Filter 2 only.

Colour

Colourizes the noise signal. A value of 0 produces White Noise while positive values lower the bass area. Negative values dampens the higher noise frequencies.

The Glide Section



"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 Active

Enables or disables the Glide effect.

Rate

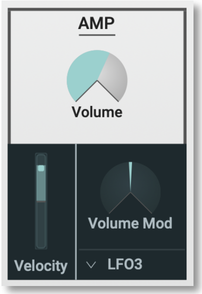
Determines the glide time. Low values will give a short glide time in a range of milliseconds that gives a special character to the sound. High values will result in a long glide time of up to several seconds which can be useful for solo and effect sounds.

Glide Style

Determines the way the Glide effect works.

- *Portamento* means that a continuous glide is performed on all new notes.
- *Fingered Port* means that a continuous glide is performed only when notes are played legato. Staccato played notes start on the exact pitch of their note.
- *Glissando* makes the normal Glissando effect in the same manner by changing the pitch in semitones.
- *Fingered Gliss* is similar to Glissando but generates a pitchchange only when notes are played legato.

The Amp Section



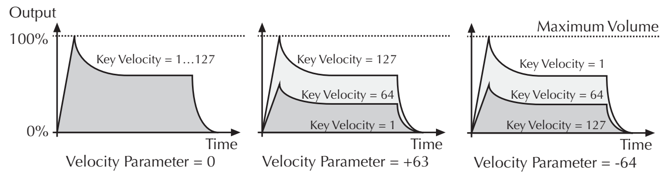
To understand the operation of this unit, it is important to know that the Amplifier Envelope is always acting as a modulation source for the volume. This means that an audio signal can only pass through if the Amplifier Envelope is triggered and opened.

Volume

Determines the master volume of the sound program.

Velocity

Specifies how much volume will be affected by keyboard velocity. Use this feature to give more expression to the sound. With a setting of *0*, velocity will have no effect on the volume. Classic organs work in this way because they do not have dynamic response. For positive settings, the volume rises with higher velocities. This is the most commonly used setting, which gives a piano-like character. With negative settings, the volume decreases at higher velocities. This gives an atypical character suitable for effect sounds. As the Amplifier always works in conjunction with the Amplifier Envelope, this parameter actually determines the envelope velocity amount. The following picture illustrates this functionality:



Volume Mod Source

Selects the source of the volume modulation.

Volume Mod

Determines the amount of volume modulation.

The Voice Section

The following parameters refer to the complete oscillator section.

Mono

Determines if the Sound can be played monophonically. If not active, each note triggers its own voice or voices, as on a piano.

If active, only the last played note sounds. All other notes are stored in an internal list but aren't played. As soon as you release the note that is currently played, the second last note is played and so on. When you play legato, only the first note that was played triggers the envelopes. All later notes use these envelopes, but sound in the pitch you've played. This mode is for sustained sounds like typical 70's solo sounds.

When *Mono* is selected and you have set up a decaying volume envelope for the selected Sound, you might not hear anything after playing several notes because of the envelopes decaying to 0.

Uni Detune

Controls the detuning of the Unison voices. Each voice is detuned differently; with **Uni Detune**, you control the overall amount.

Uni Detune is perfectly suited to thickening the tone. Arpeggios benefit too from the detune function.

Unisono

Controls how many voices are triggered when a note is played.

- *off* means that a note triggers one voice. This is the standard mode.
- *dual* means that a note triggers two voices. Both voices have high priority so they can cut off other voices that are played.
- *3...6* means that this number of voices is triggered when a note is played. Only the first voice has high priority, meaning that it can cut off other played notes. The other voices can only sound if any voices are free or if there are other unison voices with lower priority that could be cut off. This ensures that older notes play

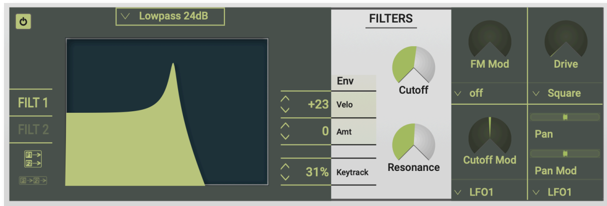
at least one voice as long as the voice allocation isn't forced to steal even this voice for a new note.

Please note that the chosen number of unisono voices will increase the CPU consumption.

The Filter Section

The Blofeld plug-in offers two filters that use the same parameters for editing.

Choose the desired filter by pressing the button. The LED of the corresponding filter will light up.



The **Filter Bypass** button deactivates the corresponding filter. This is useful if you want to disable the filter temporarily and listen to the oscillators' pure signals.

Filter Routing

These buttons switches between serial and parallel filter routing.

The Routing function is one of the advanced features of the Blofeld plug-in. Its purpose is to control the signal flow of the filters. In comparison with many other synthesizers where signal flow is static, the Blofeld offers a more flexible configuration.

The Blofeld offers two independent filters and panning units. In fact, the panning units are part of the filters in the Blofeld. The routing control makes it possible to change the signal flow from a parallel to a serial filter configuration and vice versa.

This is how the Routing section works in detail:

- The Oscillator section uses two separate outputs – one connected to the input of Filter 1, the other connected to the input of Filter 2. Each sound source, i.e. the oscillators, the ring modulator, and the noise generator has an individual **Balance** control. By means of these controls you can determine what portion of each source's signal is routed to the inputs of Filter 1 and Filter 2. E.g. this makes it possible to send the signal of Oscillator 1 and 2 to Filter 1 and the ring modulation signal to Filter 2.
- When the **Routing** parameter is set to *serial*, the whole output signal of Filter 1 is routed to the input of Filter 2, and added to the amount of signal that is already coming from the Oscillators through its dedicated output. This setting is equivalent to a serial connection of the two filters.
- When the **Routing** parameter is set to *parallel*, the whole output signal of Filter 1 is routed to the filter's panning unit. This setting is equivalent to a parallel routing of two filters, each filtering the input signals independently.

After passing the panning units, both signals are summed up again and sent to the Amplifier and FX section.

Filter Type

Selects the filter type.

- *Lowpass 24dB / 12dB* are suitable for most normal applications. Use the 24dB slope if you want to create sounds with a typical audible filtered character; use the 12dB slope if you want to get softer results.
- *Bandpass 24dB / 12dB* remove frequencies both below and above the cutoff point. As a result, the sound character gets thinner. Use these filter types when programming effect and percussion-like sounds.
- *Highpass 24dB / 12dB* are useful to thin out a sound's bass frequencies. This may also give interesting results in conjunction with cutoff frequency modulation. By doing this you can e.g. "fly-in" a sound starting at its high harmonics and then coming up to its full frequency range. Use the 24dB slope if you want to create

sounds with a typical audible filtered character; use the 12dB slope if you want to get softer results.

- *Notch 24dB / 12dB* are the opposites of the band pass types. They dampen frequencies around the cutoff point. Frequencies below or above the cutoff point are passed through. Use these filter types for programming effect sounds. On Notch filter types, the Resonance parameter is almost useless by definition because the resonance frequency is exactly the frequency that is blocked by the filter. However, you will still be able to hear slight differences when you change the Resonance because of phase changes. Anyway, the effect isn't that spectacular.

- *Comb+ / Comb- Filter* differ from the other filter types greatly, because they don't actually damp any part of the signal, but instead add a delayed version of the input signal to the output.

- *PPG Lowpass* is a resonance lowpass filter with a slope rate of 24dB per octave. Its characteristics were modeled after the legendary PPG Wave synthesizer and its integrated SSM 2044 chip. The resonance of the SSM 2044 had a very special tonal character, which has not been implemented in this way in any other filter circuit or IC. If you have the chance to directly compare the original with the Blofeld, you will find the resonance (or Emphasis, as it is called in the PPG) of both to be nearly identical.

What exactly is a Comb filter? A Comb filter is basically a very short delay that can be controlled in length and feedback. The delay time is so short that you can't hear its individual taps but a colorization of the original signal created by peaks or holes in the frequency spectrum. The frequency of the colorization is set by the delay length, which is controlled in the Blofeld through **Cutoff**, while the amount of colorization is set by the Comb filter feedback, which is controlled in the Blofeld through **Resonance**.

Cutoff

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 delay length of the comb filter types.

- When a low pass type is selected via the **Type** parameter, all frequencies above the cutoff frequency are damped.

- When a high pass type is selected, all frequencies below the cutoff frequency are damped.

- When a band pass type is selected, only frequencies near the cutoff setting will be passed through.
- When a notch type is selected, the frequencies around the cutoff frequency are damped.
- When a comb type is selected, the frequencies near the cutoff frequency are emphasized (comb+) or attenuated (comb-).

You can bring more movement into the sound by modulating the cutoff frequency via the LFOs, the envelopes or the **Keytrack** parameter of the filter. At a value of *64* and a **Resonance** value of *114*, the filter oscillates with 440Hz, which is equal to A3 (the Comb+ type oscillates one octave higher). Tuning is scaled in semitone steps. When **Keytrack** is set to *+100%*, the filter can be played in a tempered scale.

Resonance

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. This feature can be used to create analog-style effects and percussion-like electronic toms, kicks, zaps etc.

Env Amount

Determines the amount of influence the 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.

Since there are two filters, you could use the Filter Envelope on one of them, and another envelope on the second filter, buy setting this parameter for that filter to zero and use the Modulation Matrix for the other used envelope. Specially with the two filters placed serial this can bring nice effects.

Env Velocity

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.

The overall modulation applied to the filter's cutoff frequency is calculated as the sum of both the **Env Amount** and **Env Velocity** 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.

Keytrack

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%*. On most bass sounds lower settings in the range *+50...+75%* are optimal to keep the sound smooth at higher notes.

FM Mod Source

Selects the source of the frequency modulation for the selected filter.

FM Mod

Sets the amount of frequency modulation that is applied to the filter by the selected source.

Cutoff Mod Source

Selects the source of the cutoff modulation for the selected filter.

Cutoff Mod

Controls the amount of cutoff modulation for the selected filter. Positive amounts will increase the cutoff frequency when positive modulation is applied, e.g. by pressing the aftertouch on the keyboard. Negative amounts will decrease the cutoff frequency when positive modulation is applied.

Drive

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. Lower values will add some harmonics to the signal, resulting in a warm character. Increasing the value will bring in more and more distortion, suitable for harder lead sounds and effects.

Drive Curve

Determines the character of the drive. The following drive curves are available: Clipping, Tube, Hard, Medium, Soft, Pickup 1, Pickup 2, Rectifier, Square, Binary, Overflow, Sine Shaper, Osc 1 Mod.

Pan

Determines the position in the stereo panorama. When the setting is *left 64*, the sound is panned far left; when the setting is *right 63*, it is panned far right. If you want to situate the sound in the middle of the stereo panorama, use the *center* setting. To give further movement to the sound, set this parameter to a basic value and apply some modulation to it via the **Pan Source** parameter.

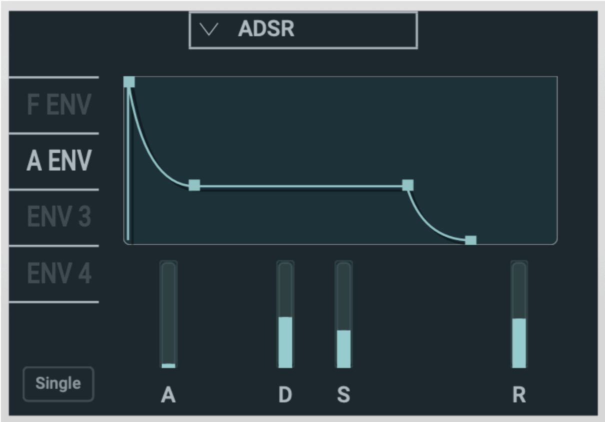
Pan Source

Selects the source of the panorama modulation for the selected filter.

Pan Amount

Determines the amount of panorama modulation for the selected filter.

The Envelopes



The Blofeld plug-in envelopes allow you to manipulate sound parameters via rate or timed modulations. The Blofeld offers four independent programmable envelopes for every sound program:

- A Filter Envelope (F Env). This envelope is fixed to control the filter but can also be used for other modulations.
- An Amplifier Envelope (A Env). This envelope is fixed to control the sound volume, but can also be used for other modulations.
- Two additional Envelopes, Env 3 and Env 4. These envelopes can be used freely to perform additional modulations on any module.

(!) The parameter controls of the envelopes are nearly similar.

(!) To select the desired envelope for editing, click on the corresponding ENV button.

Single

Determines the triggering of the corresponding envelope.

If not active, every Note starts the envelope of its own voice.

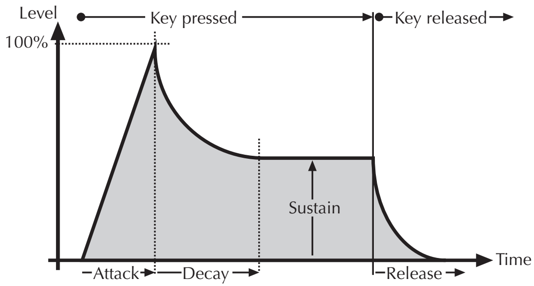
If active, the envelopes of all voices of a selected program behave like a single envelope. This common envelope starts as soon as the first note is played. The sustain level remains until the last note is released. Afterwards the release phase is active. *Single* is only active in monophonic voice-mode.

Envelope Mode

Switches between the various envelope types.

ADSR

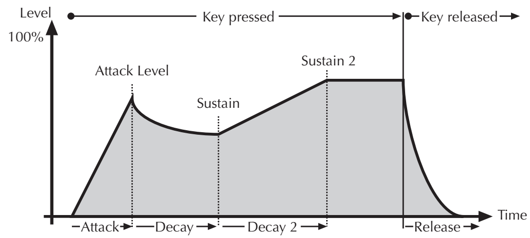
Most traditional synthesizers feature ADSR envelopes. These envelopes are made up of four parameters that determine their response: **Attack, Decay, Sustain** and **Release.** The other parameters of the Envelope section have no function, so they can not be edited. The following picture illustrates the structure of an ADSR envelope:



The envelope is started by pressing a key. It ascends to its maximum value at the rate determined by the **Attack** parameter. It then descends at the rate determined by the **Decay** value until it reaches the predetermined **Sustain** value. It remains at this value until the key is released. The envelope then descends to zero at the rate determined by the **Release** parameter.

ADS1DS2R

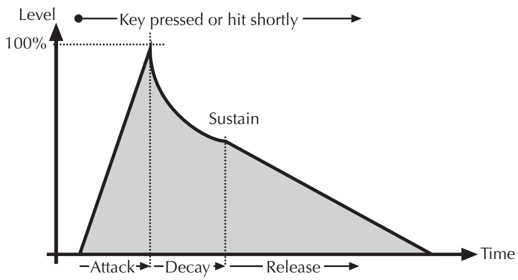
A difficult name for an envelope that is quite easy to understand. Besides the parameters an ADSR envelope features, it offers an adjustable attack level and a second Decay and Sustain pair. With these additional parameters, you can create much more complex envelopes.



One Shot Envelope

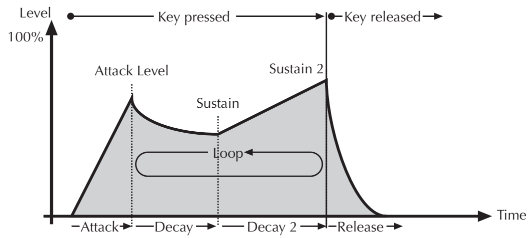
The One Shot envelope is designed for percussive sounds that don't need a stage maintained as long as a note is played. In other words: the envelope goes through all its stages, no matter how long a key is pressed. This includes even the Attack phase. It uses the parameter set of the ADSR envelope type in which the **Sustain** parameter is used to set a level breakpoint. This allows creation of One

Shot envelopes with a very percussive attack or with a "Gate" effect. Some parameters of the Envelope section have no function so they can not be edited.



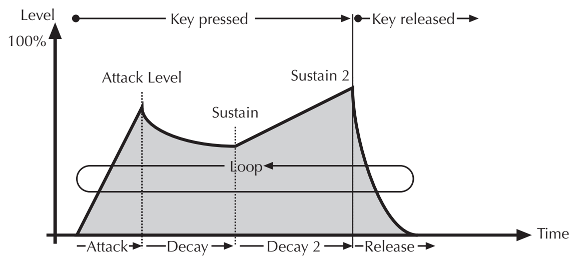
Loop S1S2 Envelope

The Loop S1S2 envelope loops the envelope between **Sustain 1** and **Sustain 2** as long as a note is held, after has been through the **Attack** phase once. This means that when **Sustain 2** is reached, **Decay 1** is used to reach **Sustain 1** again, then **Decay 2** is used to reach **Sustain 2,** and so on. As soon as the note is released, the envelope proceeds with the **Release** phase. It uses the parameter set of the ADS1DS2R envelope.



Loop All Envelope

The Loop All envelope is similar to the Loop S1S2 envelope type but it loops through all envelope stages as long as a note is held. This means that it goes through all envelope stages first, and if it ended with the **Release** phase, the envelope restarts from zero and goes through all its phases again. As soon as the note is released, the looping stops and the envelope goes into its **Release** phase.



ADSR Envelope Parameters

Attack

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

Decay

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

Sustain

Determines the sustain level which is held until a note ends.

Release

Once the note has ended, the release phase begins. During this phase, the envelope fades to zero at the rate determined by the Release value.

Additional Envelope Parameters

Attack Level

Controls the level at each the **Attack** phase ends and the **Decay** phase starts. This parameter affects the envelope types *ADS1DS2R, One Shot, Loop S1S2* and *Loop All* only.

Decay 2

Determines the decay rate or amount of time it takes for a signal to reach the **Sustain 2** level. This parameter affects the envelope types *ADS1DS2R, One Shot, Loop S1S2* and *Loop All* only.

Sustain 2

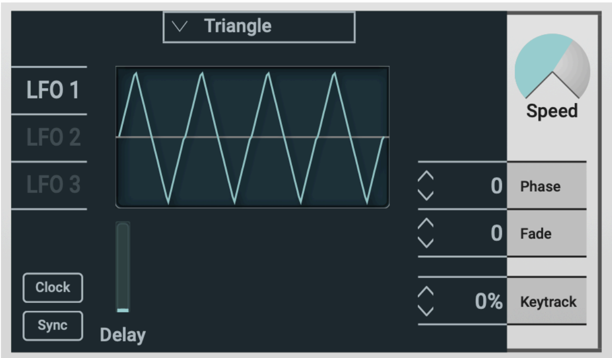
Sets the second **Sustain** level. As soon as this level is reached, the envelope goes into the **Release** phase. This parameter affects the envelope types *ADS1DS2R, One Shot, Loop S1S2* and *Loop All* only.

The LFOs

In addition to the main oscillators, the Blofeld plug-in is equipped with three low frequency oscillators (LFO) that can be used for modulation purposes. Each LFO generates a periodic waveform with adjustable frequency and shape.

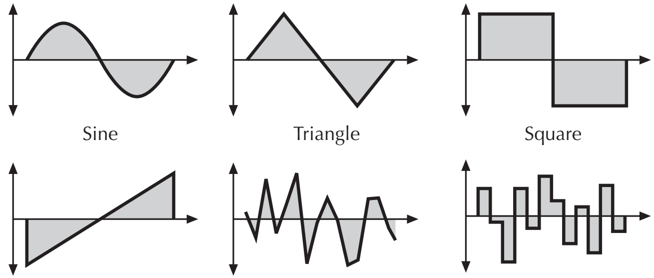
The parameter controls of the LFOs are nearly similar.

To select the desired LFO for editing, click on the corresponding LFO button.



LFO Shape

Sets the type of waveform generated by the corresponding LFO. The following picture shows the available shapes:



- The *Sine* shape is best suited for oscillator FM or pan modulations.
- The *Triangle* shape is perfect for smooth pitch, filter or volume modulations.
- The *Square* shape can be interesting for hard pan modulations or special effects.
- The *Sawtooth* shape can generate interesting filter or volume changes. If you need a modulation with inverted slope, just apply the Sawtooth shape with negative amount.

- The *Random* shape generates random values and glides to them linearly.
- *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

Determines the frequency of the corresponding LFO. At low values, it might take several minutes for the LFO to perform a complete cycle while higher values are in the audible range. Very high values are scaled in semitone steps. With **LFO Keytrack** set to *100%*, a **Speed** setting of *122* delivers an 8' LFO oscillation. 16' oscillation can therefore be generated with a **Speed** setting of *110*, and so on.

When the LFO **Clock** parameter is active, you can adjust the **Speed** in musical values. The lowest possible value is *1280 bars*, meaning that a complete LFO cycle would need 1280 bars.

Clock

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

Sync

When **Sync** is active, the LFO phases of all voices are synced so that they sound as one LFO. This can be interesting when the LFO is applied to modulate **Filter Cutoff** or **Panning**.

When **Sync** is deactivated, the LFOs run independently; this is better suited for pitch modulation to obtain thicker sounds.

> **Sync** does not mean that the LFO is synced to the host tempo or to note start. This is done with the **Clock** and the **Phase** parameters.

Delay

The **Delay** parameter works in different ways depending on the setting of the **Fade** parameter:

- When **Fade** is set to *+00…+63*, the LFO signal output is zero for the time set with the **Delay** parameter. After this time, the LFO is faded in and then runs with full magnitude.
- When **Fade** is set to *–64…-01*, the LFO runs with full magnitude for the time set with the **Delay** parameter. After this time, the LFO is faded out to zero.

Start Phase

Controls the initial phase of the LFO when a new note is started. *Free* means that the LFO isn't restarted on a new note but runs freely while other values set the LFO phase to the respective offset in degrees.

Fade

Controls the speed with which the LFO is faded in or out. With this parameter you can create slowly rising or falling modulations that might create interest when routed to pitch or volume.

Keytrack

Determines how much the speed of the LFO depends on MIDI note number. The reference note for Keytrack is E3, note number 64. For positive settings, the LFO speeds up on notes above the reference note, for negative settings the LFO slows down when higher notes are played and vice versa. A setting of *+100%* corresponds to a 1:1 scale, e.g. when the keyboard is played an octave higher, the LFO speed is doubled.

The Effects

The Blofeld offers two effect units (FX 1 and FX 2). The first effect unit is always part of the Sound Program. The second effect can be assigned either globally.

The only three parameters that are common to all types of effects:

Mix

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.

Effect Bypass

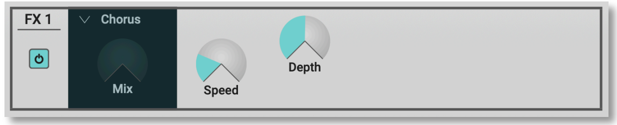
Disables the corresponding effect unit.

Effects Type Pop-Up Menu

Sets the type of effect produced by both effect units. All further parameters depend on the selected effect type.

The Effect Types in Detail

Chorus



A Chorus effect is generated by using 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.

A Mix setting of 48 to 96 produces the strongest effect because both the unaffected signal and the processed signal are mixed together.

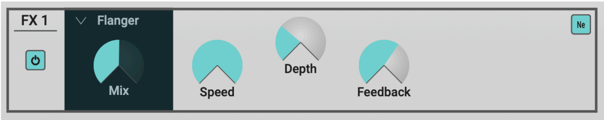
Speed

Sets the LFO speed of the Chorus effect.

Depth

Sets the modulation depth of the Chorus effect.

Flanger



The Flanger effect is very similar to the Chorus effect, but features feedback circuitry to feed the generated signal back into the comb filter. This generates a deeper detuning and colorizes the signal. With extreme settings you can hear a whistling sound which is very characteristic of a Flanger effect.

> A Mix setting of 48 to 96 produces the strongest effect because both the unaffected signal and the processed signal are mixed together.

Speed

Sets the LFO speed of the Flanger effect.

Depth

Sets the modulation depth of the Flanger effect.

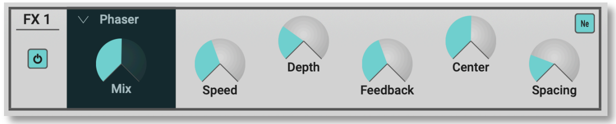
Feedback

Controls the feedback amount of the Flanger signal.

Ne Polarity

Determines whether the feedback signal is fed back into the Flanger as is, or is inverted beforehand.

Phaser



A Phaser is a combination of several "allpass" filters working in parallel. This generates an effect with equally spaced frequency peaks or troughs. The result is a strongly colorized signal with a “spacy” character.

> A Mix setting of 48 to 96 produces the strongest effect because both the unaffected signal and the processed signal are mixed together.

Speed

Sets the LFO speed of the Phaser effect.

Depth

Sets the modulation depth of the Phaser effect.

Center

Controls the basic delay length of the allpass filters. Lower settings produce a high pitched Phaser effect while higher settings enable the Phaser to cover deeper frequencies.

Spacing

Controls the relative frequencies of the allpass filters. A setting of *0* produces a classic phaser, while higher settings spread out the frequencies of the allpass filters.

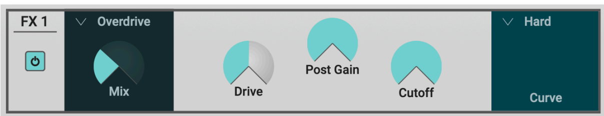
Feedback

Controls the feedback amount of the delay signal.

Ne Polarity

Determines whether the feedback signal is fed back into the Phaser as is, or is inverted beforehand.

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 sections is that Drive affects one single voice, while this effect type distorts the summed output of the whole instrument. 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. The Overdrive effect, for example, works great on Organ or E-Piano sounds.

Drive

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

Post Gain

Controls the output level of the distorted signal.

Cutoff

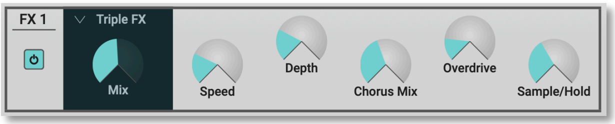
Dampens the high frequency output of the Overdrive effect.

Curve Pop-Up Menu

Determines the character of the drive. The following drive curves are available: Clipping, Tube, Hard, Medium, Soft, Pickup 1, Pickup 2, Rectifier, Square, Binary, Overflow, Sine Shaper.

Please note that the setting of the **Mix** parameter doesn't affect the strength of the overdrive effect but only the volume of it. Therefore, you can get a very strong overdrive with a low volume when you turn up **Drive** and turn down **Mix**.

Triple FX



This effect type is a combination of three different effects. You can easily use this as a replacement for one of the above mentioned effect types. The quality of Triple FX is the same as if you use a single effect, only its parameter set is reduced. The effects and their order is as follows:

- Chorus is the same as the above described effect.
- Overdrive is the same as the above described effect.
- Sample/Hold is a sample rate reduction effect.

You should turn the **Mix** control fully up with this effect type because you probably want the sample rate reduction to process the whole signal. The Overdrive and the Chorus effect have their own mix control.

Speed

Controls the LFO speed of the Chorus effect.

Depth

Controls the modulation depth of the Chorus effect.

Chorus Mix

Controls the mix level of the Chorus effect.

Overdrive

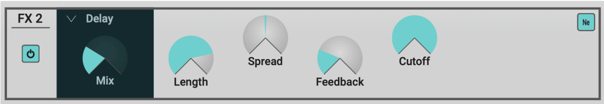
Controls the amount of distortion of the signal. Please note that the signal might become a little louder when you increase this parameter.

Sample / Hold

Controls the output sample rate. 44.1kHz means that the signal is unaffected, while other values reduce the sample rate of the sound to the respective rate. You will hear a lot of aliasing when you lower the sample rate, but this is great for so-called "lo-fi" sounds.

Delay

This effect type is only available for the **FX 2** unit.



A delay produces echoes of the input signal. An important feature of the Blofeld's delay effect is that the delay length can be changed without clicks or pitch changes. This allows you to experiment with different delay lengths without getting annoying side effects in the output signal.

Length

Sets the length of the delay tap.

Spread

Spreads the left and right delay output to half of the delay time maximum. Settings from -64 or +63 create a typical ping pong delay.

Feedback

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

Cutoff

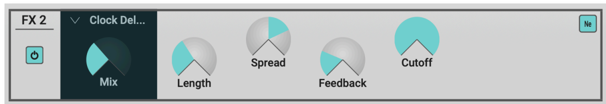
Dampens the signal produced by the delay effect. This filter is routed before the feedback circuitry meaning that adjacent taps of the Delay will be dampened further. This creates the typical "high frequency loss" that often happens in natural echoes. A setting of 127 means that the signal isn't filtered, while lower settings filter the high frequencies of the feedback signal.

Ne Polarity

Determines whether the feedback signal is fed back into the delay line as is, or is inverted beforehand.

Clock Delay

This effect type is only available for the **FX 2** unit.



The Clock Delay is an effect that produces echoes of the input signal. To make this effect musically useful in a synthesizer, the parameters to adjust the delay length are scaled in note values. The Clock Delay effect depends on the internal tempo of the Blofeld which can be found in the Arpeggiator menu.

An important feature of the Blofeld's Delay effect is that the Delay length can be changed without clicks or pitch changes. This allows you to experiment with different Delay lengths without getting annoying side effects in the output signal.

Length

Sets the length of the Delay tap in note values. A "*t*" behind the number means a triplet note value while a "." behind it means a dotted note.

Spread

Spreads the left and right delay output to half of the delay time maximum. Settings from -64 or +63 create a typical ping pong delay.

Feedback

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

Cutoff

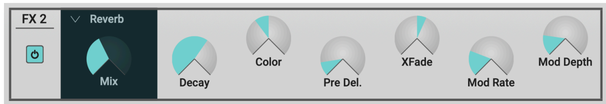
Dampens the signal produced by the Delay effect. This filter is routed before the feedback circuitry meaning that adjacent taps of the Delay will be dampened further. This creates the typical "high frequency loss" that often happens in natural echoes. A setting of 127 means that the signal isn't filtered, while lower settings filter the high frequencies of the feedback signal.

Ne Polarity

Determines whether the feedback signal is fed back into the delay line as is, or is inverted beforehand.

Reverb

This effect type is only available for the **FX 2** unit.



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 Blofeld's reverb effects don't intend to simulate the perfect natural room, rather they are an addition to the Blofeld´s sound synthesis to make it more 3 dimensional and expressive.

Decay

Determines the length of the reverb reflections. To simulate a big room choose higher Decay settings, to simulate a smaller room choose lower settings.

Color

Determines the spectral colorization of the reverb sound. Negative values dampen the higher frequencies while positive settings dampen the lower frequencies.

Pre Del.

Determines the delay between the direct sound and the reverb effect output. Lower settings connect the reverb more to the original signal while higher settings separate the effect signal to produce a more spacious sound.

Xfade

Crossfades the left and right channel of the reverb effect. It is possible to crossfade from left to right and vice versa.

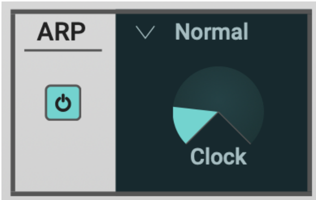
Mod Rate

Modulation allows you to enrich the reverb flag over subtle pitch modulations. **Mod Rate** determines the frequency of this pitch modulation.

Mod Depth

Determines the intensity of the pitch modulation. If no pitch modulation is desired, set this parameter to *0.*

The Arp Section



An Arpeggiator is a device that splits an incoming MIDI chord into its single notes and repeats them rhythmically. Different sequence modes can be defined for the Arpeggiator to cover a wide range of applications. In addition to the synthesis features, the Blofeld offers a deeply programmable Arpeggiator for every sound program. It can play a wide range of different rhythm patterns including accents and different timings, and allows creating sophisticated custom rhythm patterns.

The arpeggiator uses a so-called note list that can store up to 16 notes. This list is set up depending on the Arpeggiator parameter settings. Refer to the individual parameter descriptions to see if and how a parameter alters the list.

Here, you have access to the most basic arpeggiator parameters. Click on the **Arp** button in the Head-up section to get access to the full parameter set.

Arp Active

If deactivated, the arpeggiator is not active. If activated, the arpeggiator is active.

Mode

This parameter sets the way the Arpeggiator works.

- If *Normal* is seleted, the arpeggiator works in its regular way. When you press a note or a chord on the keyboard, it is split up and repeated rhythmically. As soon as you release a note, it is removed from the arpeggio rhythm. Conversely, as soon as you add another note to the existing chord, it is inserted into the arpeggio. When you release all notes, the arpeggiator stops.
- If *One Shot* is selected, the arpeggiator splits up all played notes and plays back an arpeggio. The actual length of this arpeggio is set by the **Pattern Length** parameter. After the arpeggio rhythm is played once, it is stopped automatically unless you hit a new chord. This mode is especially useful in a live performance where you might have to “synchronize” yourself, for example, to a drummer. Just hit a chord at each new bar.
- If *Hold* is selected, the arpeggiator splits up all played notes and generates a continuous arpeggio even when

the chord is released. This gives you two ways of entering a chord:

- Press all keys of the chord simultaneously. This is the normal procedure you would follow with the other Arpeggiator Modes, too.

  or

- Press and hold the first key of the chord. While holding this key, enter the other keys sequentially. After playing all keys, you can release the first key. This method is practical for playing difficult chords. It allows you to create arpeggios in the sequence of played notes. You can even hit the same note several times and it will appear in the note list accordingly.

Clock

Sets the note value for the steps of the rhythm pattern in a range from whole notes to thirty-second triplet notes. Triplets (e.g. 1/8T) and dotted notes (e.g. 1/16.) are available for every note value.

The Modulation Matrix

A modulation can be described as a signal-generating unit's influence upon a sound parameter. The terms used in this context are "Source" and "Destination". The Blofeld offers 16 independent modulation assignments (slots) each with individual settings of source, destination and amount. The Modulation Matrix (Matrix) is the key of the power of each Waldorf synthesizer, so start experimenting with it *right now*.

> ⓘ To open the Modulation Matrix, click on the **Matrix** button in the Head-up section.

Source (Src) Pop-up Menu

Defines the modulation source.

Destination (Dest) Pop-up menu

Defines the modulation destination.

> ⓘ A complete table of all available sources and destinations can be found in the Appendix.

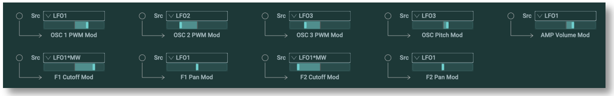
Amount Slider

Determines the amount of modulation applied to the destination. Since the modulation is in fact a multiplication of the source signal and this parameter, the resulting amplitude depends on the type of modulation source you select:

- For the so-called unipolar modulation sources, the resulting amplitude lies within the range of 0...+1, if Amount is positive or 0...-1, if Amount is negative. These sources are: all envelopes, all MIDI controllers including Modwheel, Foot control etc., Velocity, Release Velocity, Aftertouch (Pressure) and Polyphonic Pressure.
- For the so-called bipolar modulation sources, the resulting amplitude lies within the range of -1...0...+1. These sources are: all LFOs, Keytrack, Pitchbend and the Modifiers.

Synth Page Modulations

All modulations that can be found on the Synth page are mirrored in the lower section of the Matrix page.

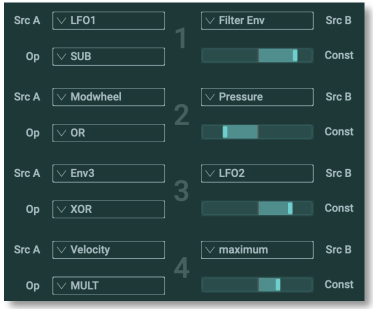


(!) Decide by yourself if you want to edit the modulations via Synth page or on the Matrix page – the result is always the same.

Modifier

Modifiers allow you to apply mathematical functions on modulation signals. Depending on the function type selected, calculation will affect two source signals or a source signal and a constant parameter. You can use up to four independent modifier units. The result of each operation is not directly audible, but is used as input source for the Modulation Matrix. Moreover, you can use it as source for yet another modifying process.

The parameter for the modifiers can be found in the right area oft he Matrix page.



Source (Src) A

Selects the first source signal used for the calculation.

Source (Src) B

Selects the second source signal when two sources are required for the calculation. See description of modifier functions for further details. The possible settings are the same as for **Source A** with one exception: *off* is replaced by *Constant* meaning that the calculation is performed with a constant value that you can set up with the **Constant** parameter.

Operation (Op)

Determines which kind of operation will be performed on the selected input sources. The following types are available:

Setting	Description
ADD	Addition
SUB	Subtraction
MULT	Multiplication
XOR	Exclusive OR function
OR	OR function
AND	AND function
min	Minimum value
max	Maximum value

Modifier Functions

The result of a modifier operation always lies within the range - max…0…+max. When it is assigned to a parameter in the Modulation Matrix, it is scaled to the range of the selected parameter.

The following paragraph describes the function and the result of each modifier function in detail:

- *ADD* returns the sum of **Source A** and **Source B**.
- *SUB* returns the difference of **Source A** and **Source B**.
- *MULT* returns the product of **Source A** and **Source B**.
- *AND* returns the binary „and“ operation of **Source A** and **Source B**.
- *OR* returns the binary „or“ operation of **Source A** and **Source B**.
- *XOR* returns the binary „exclusive-or“ operation of **Source A** and **Source B**.
- *min* returns the minimum value of either **Source A** or **Source B**. If **Source A** is smaller than **Source B**, the value of **Source A** is returned and vice versa.
- *max* returns the maximum value of either **Source A** or **Source B**. If Source A is greater than **Source B**, the value of **Source A** is returned and vice versa.

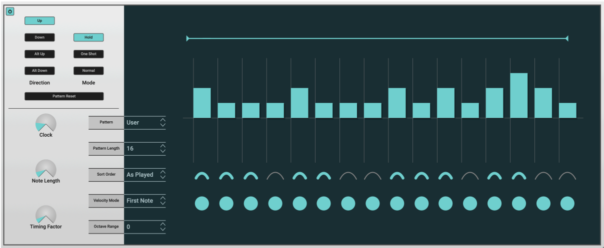
Constant

Defines a value for modifier functions that require a constant parameter. See the **Operation** parameter described above for further details.

The Arpeggiator Page

In addition to the very reduced parameter set on thew Synth page, the Arpeggiator page offers extensive options to create your own arpeggoator patterns.

> To open Arpeggiator page, click on the **Arp** button in the Head-up section.



Arp Active

Click on this button to activate the arpggiator. This button is mirrored with the corresponding button on the Synth page.

Direction

These buttons set the direction that is used to play back the arpeggio.

- If *Up* is selected, the note list is played forward and the octaves are transposed upward. The arpeggio starts in the original octave and goes up to the highest octave. Then the arpeggio is repeated.
- If *Down* is selected, the note list is played backward and the octaves are transposed downward. The arpeggio starts in the highest octave and goes down to the original octave. Then the arpeggio is repeated.
- If *Alt Up* is selected, the note list is first played forward and the octaves are transposed upward. After reaching the last note of the note list in the highest octave to play, the note list is played backward and the octaves are transposed downward down to the first note of the note list in the original octave. Then the arpeggio is repeated.
- If *Alt Down* is selected, the note list is first played backward and the octaves are transposed downward. The arpeggio starts in the highest octave. After reaching the first note of the note list in the original octave, the note list is played forward and the octaves are transposed upward up to the last note of the note

list in the highest octave to play. Then the arpeggio is repeated.

Mode

These buttons set the way the Arpeggiator works.

- If *Normal* is seleted, the arpeggiator works in its regular way. When you press a note or a chord on the keyboard, it is split up and repeated rhythmically. As soon as you release a note, it is removed from the arpeggio rhythm. Conversely, as soon as you add another note to the existing chord, it is inserted into the arpeggio. When you release all notes, the arpeggiator stops.
- If *One Shot* is selected, the arpeggiator splits up all played notes and plays back an arpeggio. The actual length of this arpeggio is set by the **Pattern Length** parameter. After the arpeggio rhythm is played once, it is stopped automatically unless you hit a new chord. This mode is especially useful in a live performance where you might have to "synchronize" yourself, for example, to a drummer. Just hit a chord at each new bar.
- If *Hold* is selected, the arpeggiator splits up all played notes and generates a continuous arpeggio even when the chord is released. This gives you two ways of entering a chord:
  - Press all keys of the chord simultaneously. This is the normal procedure you would follow with the other Arpeggiator Modes, too.
  - Press and hold the first key of the chord. While holding this key, enter the other keys sequentially. After playing all keys, you can release the first key. This method is practical for playing difficult chords. It allows you to create arpeggios in the sequence of played notes. You can even hit the same note several times and it will appear in the note list accordingly.

Pattern Reset

When all steps of an arpeggio pattern are played back, the pattern is repeated from the beginning so that the arpeggio is looped. With **Pattern Reset**, you can decide if the note list is also restarted from the beginning when the rhythm pattern is reset.

If deactived, the note list is not restarted, so that there is no synchronization between rhythm and note list. E.g., when you have a pattern where four steps are set and you play three notes, the pattern and the note list are repeated differently. The pattern restarts after the fourth step while the note list restarts after the third step. The arpeggio might look like this:

Pattern Step	1	2	3	4	1	2	3	4
Note	C1	E1	G1	C1	E1	G1	C1	E1

Pattern Step	1	2	3	4	1	2	3	4
Note	G1	C1	E1	G1	C1	E1	G1	C1

If activated, the note list will be restarted as soon as the rhythm pattern is restarted. The same arpeggio might now look like this (note the two C1s in sequence):

Pattern Step	1	2	3	4	1	2	3	4
Note	C1	E1	G1	C1	C1	E1	G1	C1

Pattern Step	1	2	3	4	1	2	3	4
Note	C1	E1	G1	C1	C1	E1	G1	C1

Clock

Sets the note value for the steps of the rhythm pattern in a range from whole notes to thirty-second triplet notes. Triplets (e.g. 1/8T) and dotted notes (e.g. 1/16.) are available for every note value.

Note Length

Sets the length of the generated arpeggio notes. However, when **Note Length** is set to *Legato*, all arpeggio notes are played without pauses between each step and **Arpeggiator Length** therefore has no effect.

Timing Factor

Determines how much the **Arp Timing** parameter affects an arpeggio step. If **Timing Factor** is set to *0*, the settings in **Arp Timing** are completely ignored and the arpeggio is played back without any shuffled timing. Settings from *1* to *127* increase the shuffling of the notes depending on the setting in the **Arp Timing** parameter. **Timing Factor** also works on ROM patterns; these are set up with standard swing rhythm.

Pattern

Sets the rhythm pattern that is used for generating the arpeggio. **Pattern** can set to *None*, *User* or to one of the 15 ROM patterns.

If *None* is selected, the arpeggiator plays back a continuous sequence of notes with the current **Clock** setting.

User gives you the ability to create your own custom rhythm pattern. This pattern is stored in in the sound. See the section "Arpeggiator Step Data" below about the pattern settings you can create.

2…16 selects one of the 15 internal ROM rhythm patterns. See the table below for an overview of each ROM rhythm pattern:

Pattern	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16
1	●		●	●	●		●	●	●		●	●	●		●	●
2	●		●		●			●	●		●		●			●
3	●		●		●		●	●	●		●		●		●	●
4	●		●	●	●		●		●		●	●	●		●	
5	●		●		●	●		●	●		●		●	●		●
6	●	●		●		●	●		●	●		●		●	●	
7	●		●		●		●		●	●		●		●		●
8	●		●		●		●	●		●		●	●		●	
9	●	●	●		●	●	●		●	●	●		●	●	●	
10	●	●		●	●		●	●		●	●		●	●	●	
11	●	●		●	●		●	●		●	●		●		●	
12	●	●		●	●		●		●	●		●	●		●	
13	●		●		●		●		●	●		●		●	●	●
14	●			●			●			●			●			●
15	●		●		●		●		●			●	●		●	

Note that you can edit a ROM pattern to create your own rhythms starting from an existing ROM pattern. As soon as you do so, the ROM pattern is copied to the User pattern and the previous User pattern is overwritten.

Pattern Length

Sets the length of the rhythm pattern. This parameter is also valid when **Pattern** is set to *Off* or if one of the ROM patterns is selected. Note that you can only edit a particular step in the right section when its position is within the range of the **Pattern Length** parameter.

Sort Order

Sets the order in which the note list is arranged. With this parameter, you can determine how the notes you originally play are split up for the arpeggio.

- If *as played* is selected, the notes are sorted in the exact order you played them. If you e.g. press E1, G1 and C1, the note list looks exactly like that.
- If *reversed* is selected, the notes are sorted to the reverse order in which you played them. If you e.g. press E1, G1 and C1, the note list is sorted to C1, G1 and E1.
- If *Key Lo>Hi* is selected, the notes are sorted from the lowest note to the highest note. If you e.g. press E1, G1 and C1, the note list is sorted to C1, E1 and G1.
- *Key Hi>Lo* is the opposite of *Key Lo>Hi*. The example would be sorted as G1, E1 and C1.

- If *Vel Lo>Hi* is selected, the notes are sorted from the softest to the loudest velocity. If you press notes with velocities 64, 120 and 96, the note list will be sorted to 64, 96 and 120.
- *Vel Hi>Lo* is the opposite of *Vel Lo>Hi*. The above notes would be sorted as 120, 96 and 64.

Velocity Mode

Determines how velocity is interpreted in the arpeggio. Note that each arpeggio step might have an additional positive or negative offset set by the **Arp Accent** parameter.

- If *Each Note* is selected, each note of the arpeggio is played back with the velocity that you originally played.
- If *First Note* is selected, the first note you played sets the velocity for all arpeggio steps.
- If *Last Note* is selected, the last note you played sets the velocity for all arpeggio steps.
- If one of the fix values is selected, all notes will be played with the selected velocity.

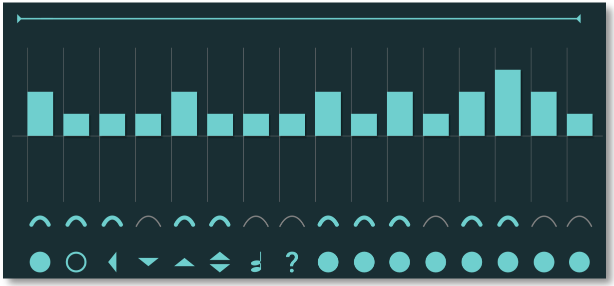
Octave Range

Determines the range of the single notes in octaves. When it is set to *1 Oct*, the note list will be played back in the same octave as originally entered Greater values mean that the note list is repeated in higher or lower octaves. The octave in which the arpeggio starts is determined by the **Direction** parameter. If you play notes that span more than one octave, they are still kept in the note list and played back before the note list is transposed. The following table shows some possible arpeggios:

Note input	Range	Dir	Resulting Arp
C1 E1 G1	1 Oct	Up	C1 E1 G1 | C1 E1 G1
C1 E1 G1	2 Oct	Up	C1 E1 G1 | C2 E2 G2 | C1 E1 G1 | C2
E1 G1 C1	3 Oct	Up	E1 G1 C1 | E2 G2 C2 | E3 G3 C3 | E1
C1 G1 E2	3 Oct	Up	C1 G1 E2 | C2 G2 E3 | C3 G3 E4 | C1
C1 E1 G1	3 Oct	Down	G3 E3 C3 | G2 E2 C2 | G1 E1 C1 | G3
C1 E1 G1	2 Oct	Alt Down	G2 E2 C2 | G1 E1 | C1 E1 G1 | C2 E2

Arpeggiator Step Data Input

On the right area of the Arpeggiator page, you can create own patterns. You can set steps with accent and glide, and define the timing and length.



> You can only select the steps that are in the range set by **Pattern Length**. E.g., when you have set **Pattern Length** to *8*, you can't change a value of step 9 or above.

Step

This parameter can have a pronounced effect on the resulting arpeggio, so you should read the following paragraphs carefully. **Arp Step** basically determines which note of the note list is played at a particular step. You can also force the Arpeggiator to play a whole chord or part of a chord or set it up to play a random note from the list.

> To change a step behaviour, click and drag your mouse up or down.

- If • *normal* is selected, the Arpeggiator plays the step unaltered. The note list is advanced beforehand, except when you press a new chord.
- If o *pause* is selected, the Arpeggiator plays nothing at this step position. When **Length** is set to *legato*, the previous step that isn't set to *pause* is still held to create the legato effect. The note list is not advanced.
- If ◀ *previous* is selected, the Arpeggiator plays the same note as it had to play in the previous step that was set to • *normal* or *? random*. With this setting, you can repeat a particular note of the note list several times. The note list is not advanced.
- If ▼ *first* is selected, the Arpeggiator plays the very first note of the note list. This might be interesting if you want to only play the "root note" of a chord in a bass sound. The note list is not advanced.
- If ▲ *last* is selected, the Arpeggiator plays the very last note of the note list. The note list is not advanced.

- If ▼▲ *first+last* is selected, the Arpeggiator plays a chord with two notes, the first and the last one of the note list. This means that you have to play at least two notes to hear the effect. Otherwise, you would hear only one note anyway. The note list is not advanced.
- If ı *chord* is selected, the Arpeggiator plays a chord with all notes from the note list. This means that you have to play at least two notes to hear the effect. The note list is not advanced.
- If *? random* is selected, the Arpeggiator plays a random note from the note list. This doesn't mean that it creates a note randomly; instead it uses one note of the existing note list. The note list is not advanced.

Accent

Sets the accent of a particular step. This accent is interpreted as a velocity offset that is added or subtracted from the original velocity stored in the note list. However, the generated velocities can never exceed the maximum MIDI velocity 127 or minimum MIDI velocity 1. This means when you have already played notes with a high velocity, **Accent** might not be able to further offset them positively, so you will only hear differences with negative accent offsets. Conversely, low velocities might not be able to be lowered by **Accent**. The only exception is *silent*.

> To change a accent, click and drag your mouse up or down.

See the following description:

- If *silent* is selected, the current step is actually played, but inaudible. This means that the note list is advanced by one note, but you are not able to hear it. This feature is the opposite to **Step** set to o *pause*, which doesn't generate any note and therefore doesn't advance the note list.
- If **1* is selected, the velocity of the current step is not altered. The arpeggio steps are played back with their original velocity.
- If **Arp Accent** is set to any divisor *(/4, /3, /2*) or multiplicator (*2, *3, *4) the velocity is raised or lowered accordingly. For example a played velocity of 64 generates a final velocity of 32 when set to */2* and 128 when set to **2*.

Glide

For each step in the arpeggio pattern you can activate the glide effect individually. This gives you the ability to create the classic "Bass Line" melody character. Make sure that you set up a reasonable glide effect.

- If deactivated, the glide effect is disabled for this step.
- If activated, the glide effect is enabled for this step. This means that the previous note glides to the note that has to be played at this particular position in the arpeggio.

Please note that **Glide** on the Synth page must be deactivated, when you want to set **Arp Glide** for individual pattern notes. Otherwise, the glide effect will occur on all notes.

Timing

Moves the playback time of a step forward or backward. Forward means that a step is played later while backward means that it is played earlier. The overall strength of this parameter is set by **Timing Factor**. If **Timing Factor** is set to *0*, **Arp Timing** has no effect on the rhythm at all. If **Timing Factor** is set to maximum, **Arp Timing** can move the step by a maximum of half the clock division. This means that it can move the step by 1/32 forward or backward when clock is set to 1/16.

- *Random* moves the step forward or backward at random. It might also be played without being moved.
- Negative values *(-3, -2 oder -1*) move the step backward so that it is played earlier.
- If *0* is selected, the step isn't moved at all.
- Positive values *(+1, +2, +3*) move the step forward so that it is played later.

Length

Changes the length of the note of a particular step. The overall length of the arpeggio depends on the **Length** setting. If **Length** is set to *legato*, **Arp Length** doesn't have any effect at all. Also, if **Length** is set to a very small value, **Arp Length** might not have an audible effect when you set it to a negative value. You can create very nice staccato and legato effects with this parameter.

- If *legato* is selected, the notes of this step are held until the next step is played. Empty steps force any step notes set to legato to remain held.
- Negative values *(-3, -2 oder -1*) shorten the length of the notes of this step.
- If *0* is selected, the step is held for the time set in the **Length** parameter.
- Positive values *(+1, +2, +3*) extend the note duration of this step.

The Multi Mode

The Waldorf Blofeld offers a 16 part Multi mode. As soon as you want to do multi track recordings in your DAW, you should start to use Multi parts. Each sound in a Multi setup based on a socalled **Part**.

To open the Multi Mode page, click on the **Multi** button in the Head-up section.

Activation of the Multi Mode

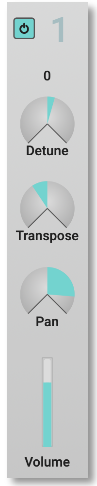
To take use of the Multi Mode, you have to activate it first in the Global Parameters area on the Multi page. The upper area of this page shows youall 16 multi parts.

Selection of Multi Parts

Before you can adjust the sound parameters of a particular Part, you have to select it. The Multi Mode offers 16 Parts that can be played at a time via MIDI.

Click on the corresponding Multi part to select it. You can also cklick on the Part dropdwon menu in the head-up section and select the desired part 1 to 16.

Multi Part Parameter



The following parameters can be found in the Multi part chennel strip.

Part Active

Activates the corresponding Multi part.

Volume

Volume sets the output volume of the selected Part.

Pan

Determines the panning position of the selected instrument. The setting *-64* stands for full left, *63* for full right. In case you want to have the sound in the mid position select *center*.

If stereo effects are active, e.g. delay, the effect will still sound in both outputs even if the basic sound is set to full left or full right.

Transpose

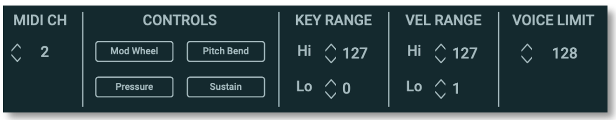
Transposes the Mulit part in semitone steps. I.e., a value of *–12* means that the Instrument sounds one octave lower than it was originally programmed.

Detune

Detunes the Instrument in steps of 1/64th of a semitone.

You can set up nice layered sounds with **Transpose** and **Detune**. Activate two Multi parts and set their parameters to identical values including the sound number. Then simply transpose one Instrument by one octave, and you have a fat layer sound. Or set them both to the same octave and set **Detune** of one Instrument to *–05* and the other to *+05*

The following Multi part parameters can be found in the bottom left are and depends on the selected Multi part.



Channel

Channel sets the MIDI Channel for the selected Multi part. This MIDI channel is used to receive MIDI messages for this Part

- *Omni* means that the selected Multi part receives data from all MIDI channels.
- *1…16* means that the selected Multi part receives on this MIDI channel. This setting is recommended for a Multi that is intended for multi track playback with DAW.

Mod Wheel

Determines if incoming modulation wheel data is received or ignored.

Pitch Bend

Determines if incoming pitch bend is received or ignored.

Pressure

Determines if incoming pressure data (Aftertouch) is received or ignored.

Sustain

Determines if incoming sustain pedal data is received or ignored.

Key Range Hi / Lo

The key range can be restricted for the multi part's tone generation. Only notes with a key number higher/lower or equal to the selected values are passed through. Set the **Lo** parameter to *C-2* and **Hi** to *G8* if you want to use the full keyboard range.

Vel Range Hi/Lo

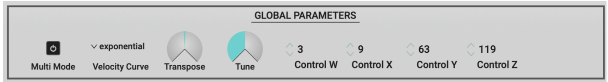
The Velocity range allows you to limit the velocity range in which the multi part is played. Only notes with a velocity higher/lower or equal to the selected value are passed through. Set the Hi parameter to *1*, if you want to turn velocity switching off.

Voice Limit

Limits the number of maximum voices for the corrsponding multi part.

Global Parameters

In the right bottom area oft he Multi page, you find the global parameters.



Velocity Curve

Determines the velocity performance of a your MIDI keyboard.

Transpose

Allows a global pitch transposition. Incoming MIDI notes are shifted by the number of semitones.

Tune

Controls the Blofeld's overall pitch in Hertz. The value specified here is the reference pitch for MIDI note A3. The default setting is 440 Hz, which is commonly used by most instruments.

⚠ **You should only change this setting if you really know what you're doing. You will have to adjust all your other instruments, too. Don't forget to set it back again!**

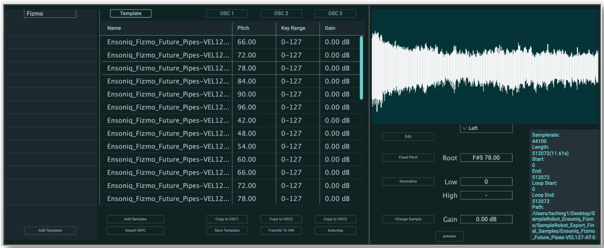
Control W...Z *0...120*

These parameters are used to define arbitrary MIDI Continuous Controllers as modulation sources for Sounds. You can setup four Controls, W, X, Y and Z, for this purpose. Each value represents a MIDI Continuous Controller number that is used when you assign its parameter as modulation source in the Modifiers or the Modulation Matrix. The highest possible Controller number is *120* because all higher numbers are reserved for non-real-time purposes.

The Sample Page

The Blofeld´s oscillators can load and play samples. On the Sample page, you organize your samples.

To open the Sample page, click on the **Sample** button in the Head-up section.



Samples are organized into Templates. In the left area of the Sample page, you can add new templates. The central area shows the samples of the current Template. Here, you see some more information about the current pitch of each sample, the key range and the gain.

Template List

The left area oft he Sample page contains the Template list.

Here, you can click on the **Add Template** button to create a new Template. With a right-click on a Template you can rename it or delete it.

Click on a Template in your Template list to select it. The corresponding samples are shown in the central Template list.

Template Sample List

The central part of the Sample page contains the Template sample list. From here, you can import and organize your samples within a template.

Add Samples

Opens your computer´s file system to loacate your audio sample(s) you want to load into the current temple.

Import WPC/XML

Imports sample sets in the Blofeld hardware format or as XML file (as created e.g. by the Skylife Sample Robot application).

Copy to OSC1/2/3

Copies the current template to the corresponding Blofeld oscillator. If copied, the samples of that template can be used as alternative oscillator „waveform“.

Save Template

Here, you can save all you edits made in your template.

Transfer to HW

Here, you can send the current seleted template to a connected Blofeld hardware unit.

Automap

Here, you can automap your samples over the keyboard. If clicked, you can choose an *Equal Distance Key* or the default settings of the template list. This will change the Key Range for each sample.

Sample Information Section

In the right area, you get more information on the currently selected sample in the Templates list. Here, you can change basic sample parameters.

Waveform Display

Here you can see the current selected sample as waveform. Use the pop-up mnenu below this section to determine the channel that is dispalyed (left, right or L+R).

Fixed Pitch

Here, you can set the current selected sample to a fixed pitch.

Normalize

Here, you can perform a normalizing of current selected sample to the maximum gain.

Change Sample

Here, you can exchange the current selected sample by loading any other sample from your hard drive.

Root

Here, you can change the root key for the current selected sample.

Low/High

Here, you can determin the key range for the current selected sample.

Gain

Here, you can determine a desired gain in dB for the current selected sample.

Preview

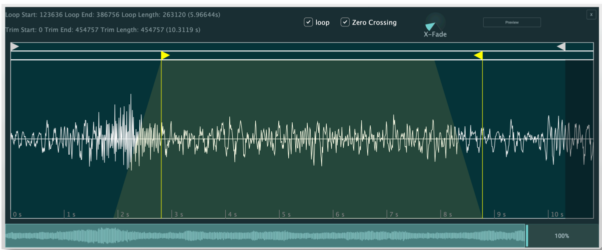
Click on this button to hear a preview of the current selected sample.

Sample Information Section

Here, you see some information about the current selected sample.

Sample Edit Page

In Edit mode, you can trim a sample and set a sample loop, if desired.



Click on the Edit button to open the **Sample Edit** page. Click on the x in the top right area oft he sample edit page to close it.

Sample Start / End handlers

By default, this white coloured handler is positioned at the extreme left of the sample (beginning of the sample). Dragging this handler will start playing the sample at the point the handler is located, instead of the beginning of the sample.

Similar to the above, but this time the change is applied to the end of the sample. By default, the handler is placed at the extreme right of the sample (sample end).

Loop Start / End Handler

This yellow coloured handler defines the beginning of the loop, which will be played when the Loop option is active. It works in conjunction with the Loop End handler, and together they define which part of the sample will be looped.

The same way the Loop Start handler defines the point where the looped region begins, this handler defines the point where the looped region ends.

X-Fade

Introduces an x-fade for the sample loop start and end to receive a smoother loop result.

Zero Crossing

Zero crossing is a point where the waveform crosses the zero level axis. To perform editing operations such as trim or looping, we recommend activath this option.

Preview

Click on this button to hear a preview of the sample and your edits.

Sample Zoom

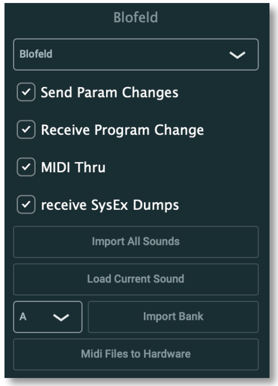
Use the down right slider to zoom into the sample. The current zoom factor is displayd in percent.

Blofeld Hardware Sync

With this option, you can use the Blofeld Plug-In to control your Waldorf Blofeld hardware synthesizer and import sounds.

The Blofeld hardware Sync window can be found in the Menu in the head-up section.

> To take use oft he hardware control options, please connect yout Blofeld hardware via USB to your computer.



Device Pop-up Menu

Here, you select your connected Blofeld hardware.

Options

Activate the corresponding option, if desired:

- *Send Param Changes* – if active, every parameter that is edited in the plug-in will sends its parameter settings immediately to a connected Blofeld hardware and changes the corresponding parameter.
- *Receive Program Change* – if active, the Blofeld hardware receives program changes from the Blofeld plug-in, when a new sound program is selected.
- *MIDI Thru* – if active, the Blofeld hardware receives the same MIDI data that is sent into the Blofeld plug-in.
- *Receive SysEx Dumps* – if active, the Blofeld hardware receives SysEx data from the Blofeld plug-in

Import All Sounds

Click on this button to import all sound data from a connected Blofeld hardware into the Blofeld plug-in. This process may take some time.

Load Current Sound

Click on this button to import only the current selected sound from a connected Blofeld hardware into the Blofeld plug-in.

Import Bank

Click on this button to import the chosen sound bank from a connected Blofeld hardware into the Blofeld plug-in. With the bank pop-up menu, you can select the desired bank A to H.

MIDI Files to Hardware

Click on this button to send a Blofeld sound bank (.mid) located on your computer hard disk to you Blofeld hardware.

Appendix

Wavetable Synthesis

A part of the sound generation in Blofeld Plug-In is based on wavetable synthesis.

A wavetable is a table consisting of single waveforms, each with its own special sound character. The main difference between wavetable synthesis and other sound-generation principles is the ability to not only play one waveform per oscillator but also to step through other waves in the wavetable employing different modulations, thereby creating so-called wavetable sweeps. The results can be dramatic – much more so than anything sample-based systems could ever produce.

This principle offers powerful capabilities, such as:

- Each note on a keyboard can access a different wave of a wavetable.
- Different waves can be played depending on key velocity.
- An LFO can modulate the position within the wavetable. You can create subtle to drastic sound changes.
- User-selected controllers, such as the mod wheel, can change the position within the wavetable. When you turn the wheel while playing a chord, each note's wave will be modified instantly.

You should keep the following sentence in mind:

A wavetable is a list of two or more (up to 64) waves, between which you can move at will.

As soon as you play a note, the envelope advances the position through the wavetable, generating different waveforms over time.

The decay stage would move through these waves in the opposite direction prior to holding a certain wave during its sustain stage. When you release the note, the envelope continues the move back through the waves to the starting point.

Most wavetables are created so that they start with a hollow wave at position 0 and go through increasingly brighter waves up to maximum position. This results in a behaviour similar to a low pass filter so that they can be conveniently controlled by wave envelope.

If Time is 0 and Level set to a medium value you get a percussive sound; if you turn up the attack, you get a soft-sounding start.

You can also use an LFO to modulate the wavetable position and, depending on the selected **LFO Shape**, you might get a wave scan that goes back and forth (triangle), in only one direction followed by a hard reset to the origin (sawtooth) or between only two waves (square).

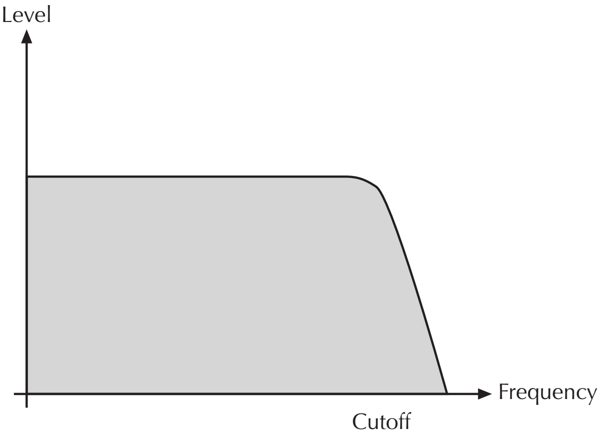
> If you like the sound and possibilities of wavetable synthesis, you should try out the our virtual instrument Waldorf Microwave 1 Plug-In.

Filter Introduction

Once the oscillator signal leaves the mixer it is sent to the filters. Blofeld Plug-In offers two filter units, each with its own individual settings. The signal flow in the filters can be controlled via the Routing function. The filters are components that have significant influence on Blofeld Plug-In's sound characteristics.

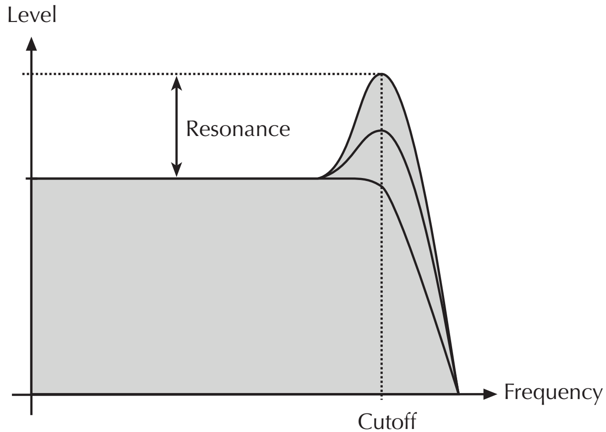
Now, we'll explain the basic function of a filter discussing the type used most commonly in synthesizers: the lowpass filter.

The lowpass filter type dampens frequencies that lie above a specified cutoff frequency. Frequencies below this threshold are hardly affected. The frequency below the cutoff point is called the pass band range; the frequencies above are called the stop band range. Blofeld Plug-In's filter dampens frequencies in the stop band with a certain slope. The slope is 24dB per octave. This means that the level of a frequency that lies an octave above the cutoff point will be 24dB less than those frequencies of the signal that fall into the pass band. The following image illustrates the basic principle of a low pass filter:



To give you an idea of the extent of damping, consider this example of a lowpass filter: a reduction of 24dB reduces the original level one octave above the cutoff point by approx. 94%. The damping factor two octaves above the cutoff point reduces the original level by more than 99%, which in most cases means this portion of the signal is no longer audible.

Blofeld Plug-In's filter also features a resonance parameter. Resonance in the context of a low, band or high pass filter means that a narrow frequency band around the cutoff point is emphasized. The following image illustrates the effect of the resonance parameter on the filter's frequency curve:



If the resonance is raised to a great extent then the filter will begin self-oscillation – i.e. the filter generates an audible sine wave even when it does not receive an incoming signal.

Glossary

Aftertouch

The majority of contemporary keyboards are capable of generating aftertouch messages. On this type of keyboard, when you press harder on a key you are already holding down, a MIDI Aftertouch message is generated. This feature makes sounds even more expressive (e.g. through vibrato).

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

The extent to which 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.

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.

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.

Coffee Filter

A coffee filter is a coffee-brewing utensil, usually made of disposable paper. It is part of an essential toolkit for survival when working with Blofeld Plug-In.

Control Change (Controllers)

MIDI messages enable you to manipulate the response of a sound generator to a significant degree.

This message essentially consists of two components:

• The Controller number, which defines the element to be influenced. It can be between 0 and 120.

• The Controller value, which determines the extent of the modification.

Controllers can be used for effects such as slowly swelling vibrato, changing the stereo panorama position and influencing filter frequency.

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. The most common type is the lowpass filter. A lowpass filter dampens all frequencies above the cutoff frequency.

Filter Cutoff Frequency

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

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.

Low Pass Filter

Synthesizers are often equipped with a lowpass filter. A lowpass filter dampens all frequencies above its cutoff frequency. Frequencies below the cutoff point are not affected.

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 different devices did not exist, so MIDI was a significant advance. It made it possible to link any MIDI-equipped device with 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 Blofeld Plug-In, then the computer is the sender and Blofeld Plug-In 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 is 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 that 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. Consequently, the sender can address specific receivers individually. MIDI Channels 1 through 16 are available for this purpose.

Modulation

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

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.

Pitch-bend

Pitch-bend is a MIDI message. Although pitch-bend 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 pitch-bend 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 pitch-bend 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 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 resonance, to a level where the filter begins self-oscillation then it will generate a relatively clean sine waveform.

Spectrum

A basic component of sounds are periodic oscillations. The perceived pitch corresponds to the fundamental frequency of this oscillation. The frequency spectrum of periodic oscillations is a line spectrum, the lowest frequency corresponds to the fundamental frequency (fundamental) and other frequencies-integer multiples of the fundamental frequency (harmonics).

Sustain

An envelope parameter. 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 to start an envelope.

Volume

The term describes a sound's output level.

Wave

In this context, a Wave is a digitally-memorized reproduction of one single wave pass insofar as it is identical to a sample that is looped after one single wave pass. In contrast to the samples in a sampler, all waves in Waldorf Wavetable synthesizers have the same lengths and are played back in the same pitch.

Wavetable

One oscillator shape in Blofeld Plug-In is based on waveform sets called wavetables. You should think of these as a sequence of up to up to 64 single waves. This can be played back in a static way or played through dynamically, which results in characteristic sound transformations. If the waves do not differ much, then the wavetable will probably sound smooth and pleasant. If they have a completely different structure then this will result in wild spectral changes.

Product Support

Any Questions?

If you have any questions about your Waldorf product, feel free to contact us. We're here to help.

① Use the support form at our website. 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.com

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

③ Visit our support area at **waldorfmusic.com**

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