KARMA·FX

SYNTH MODULAR 2

User's Manual

PREVIEW

Revision 0.46

KarmaFX Synth Modular Version 2.02+
VST2 / VST3 for PC / Windows™ and Audio Unit / VST2 / VST3 for Mac / OS X.
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Contents

1. Introduction

Thank you for purchasing KarmaFX Synth Modular!

KarmaFX Synth is an advanced simulated analog modular synthesizer that can be used in the studio either as an instrument or as an effect. It delivers superb sound quality and offers powerful modular flexibility while still being very easy to use.

Like a true modular, the Synth consists of modules that can process audio and control signals and be patched into other modules. Needless to say, this offers far more versatility than a fixed path synthesizer that only allows you to adjust predefined knobs.

Hi and Welcome ! Throughout the manual small text boxes like this one will explain words underlined in the text that may be hard to grasp.

This manual is intended to give you a basic understanding of how the Synth works. It describes the functionality of every available module in detail. After reading this, you should be able to roll your own patches with ease. A basic understanding of fundamental sound synthesis is recommended but not required.

The KarmaFX Synth is a VST plugin, meaning that it will run inside any VST2 and VST3-compatible host application (DAW) on PC/Windows. It has been tested and verified to run in many different hosts, such as Steinberg Cubase, Ableton Live and Image-Line FL-Studio, just to name a few. The synth is also available as Audio Unit, VST2 and VST3 on Mac/OS X. Audio Unit compatible hosts for Mac include software such as Apple Logic and Ableton Live..

VST is an acronym for Virtual Studio Technology: A plugin interface developed by Steinberg Corp.

In section 2 you will find a brief overview of the Synth's features and its system requirements. It also contains a detailed run-down over the most important new feature introduced in version 2. Section 3 shows you how to install KarmaFX Synth in your favorite host. Section 4 introduces the general concepts of modular synthesis. Section 5 and 6 describes the user interface and how to operate the Synth. Finally, section 7 is a reference that lets you dive into the details of every module.

So go right ahead… read on to learn the secrets of how to tame this modular beast. Then go write a killer track! ☺

We at KarmaFX sincerely hope that you will be happy with your purchase - and fall in love with synthesis all over again.

2. Features & Requirements

2.1 Feature Overview

Modular synthesizers are remarkable pieces of machinery. For readers who cannot wait to get down to the gritty details, here is a list of feature highlights in KarmaFX Synth Modular:

- Advanced Simulated Analog Modular Synthesis: Modular patching of synth modules and internal high-frequency digital simulation of analog voltage levels.
- Oscillator with Phase, Detune and Pulse-width that simulates standard analog waveforms. Dual Oscillators with Hard-Sync and Ring Mod.
- 16/24/32 bit mono/stereo Multi Sampler that imports `WAV`, `SF2` and `SFZ` files. Key/Velocity-ranges and Loop-Point Multisample editing.
- Additive and Pad Module with waveform and harmonic magnitude & phase editor. Up to 1024 harmonics + support for user-defined presets.
- Granular Module that offers sample-based granular synthesis, with variable grain size and rate and modulation/diffusion options.
- 2/4 Pole Multimode Filters with Cutoff, Resonance (LP, HP, BP and BS) + Saturation and Drive: SVF, SVF2, SKF, Zolzer, Moog, Moog2, Acid, MS20. 3/10/31 band Equalizers, Formant, Comb, Allpass, Parametric and Shelving.
- Amplifier and Stereo modules with panning, volume and velocity controls. Two channel Mixer with Ring modulation and bit operations. Inverter for Amplitude Modulation, and Mid/Side for stereo separation.
- Delay, Reverb, Phaser, Chorus/Flange, Pitchshift, BitShuffler and Distortion effects, Boost, Folder, PanSpread, Soft Clipper, Maximizer, Soft-knee Compressor and Multiband Compressor with Peak/RMS detection.
- 10-Octave / 12-Note Pitch control with detune and portamento. Choose between mono, legato, or up to 16-voice true polyphony Controllers for frequency and phase modulation (FM/PM). Up to 16 channel Unison controller with detune and stereo pan spread.
- Support for Linear FM, Phase Modulation, and Exponential FM with Exponential Frequency Sync. Generators support Through-Zero frequency, and optional High Quality (HQ) 16x oversampling.
- Scope Module with multi-layered, synced waveform oscilloscope, and frequency and phase displays for detailed sound inspection.
- Pattern controller with 1 to 32 steps, 1 to 4 octaves, hold, loop, and legato support and 8 user programmable pattern presets. Can work in Step and Arpeggiator mode, with optional scale-based note-masking.
- Bipolar/Unipolar LFO, HFO, ADSR and Multipoint Envelopes, Step Sequencer, Env.Follower, S&H, Decay, Shaper, Clock and Control modules.
- Expression, so parameters can respond to MIDI Mod.Wheel, Aftertouch, Velocity and Timbre inputs + MIDI Polyphonic Expression (MPE) support.
- Control Panel & Patch Browser for quickly navigating patches and banks.
- Keyboard module for MIDI visualization and manual MIDI triggering.
- Output module with Panning, DC removal, volume and clip control.
- Input module so that the Synth can function as an insert effect.
- Noise generator, filtered pink, white and brown noise with freq. sync.
- Full stereo support (modules can run in mono to save CPU cycles).
- Instant visual feedback of waveforms, modulation and controls.
- Up to 128 simultaneously running internal voices (Polyphonic/Unison).
- 128 user-assignable automation controls with MIDI Learn.
- SubPatch generator and effect modules for re-using patches within other patches, and doing complex generator/effect patch construction.
- 5 banks of pre-made KarmaFX patches + Extra user-banks. More than 1000+ patches total. Additional Online Banks can be downloaded and installed from inside the synth.
- Synth frequency can be offset from "Concert Pitch". (e.g., 440Hz → 432Hz)
- Fast output response, with zero latency and tight latency compensation.
- Fully skinnable GUI: Skins bundled with the installation support HiDPI/Retina GUI scaling of up to 200%.

2.2 System Requirements

In order for KarmaFX Synth Modular to run properly, the computer system must fulfill the following requirements:

PC / Windows:

- A 2 GHz Pentium class CPU with SSE (Intel or AMD) is highly recommended. Absolute minimum is a Pentium class CPU running at 1 GHz, 1 GB RAM (4 GB RAM recommended), 200 MB free disk space, Minimum 1920 x 1080 screen resolution, 32 bit color.
- Operating system: Windows XP/7/8/10/11.
- VST 2.4+ or VST3 compatible host application.
- Low latency sound card (preferably with ASIO driver).

Mac / OS X:

- Mac computer with an Intel (x86) or Apple Silicon (M1 or later) processor and 200 MB free disk space.
- Operating system: OS X 10.9 or later.
- Audio Unit (v2) compatible host application.
- VST 2.4+ or VST3 compatible host application.

2.3 What's new in Version 2

For readers already familiar with KarmaFX Synth Modular, this sections offers a run-down over some of the most important additions and changes that version 2 offers.

- Version 2 is backwards compatible with version 1.x, meaning that all existing patches and patch-banks will load in version 2. In most cases they also sound the same, but due to the many internal changes this is not 100% guaranteed.
- A set of new UI features have been added, including circular-knob-modulation-meters that show the current knob modulation, 20% larger UI overall, drop-shadows, glowing displays, gradient curves, semi-transparent menus and motion-blurred trail markers. Three brand new skins have been added, that support high quality scaling for HiDPI and Retina displays, while the classic Blue skin is also still available.
- A new **Granular** module (p.61) has been added that offers sample-based granular synthesis, with variable grain size and rate and modulation/diffusion options.
- A new **Patch Browser** (p.36) has been added, which allows for quick browsing of banks as well as patches, and also offers keyboard browsing and keyboard filtering.
- Version 2 introduces *Subpatches*: The **SubPatch** generator and effect modules (p.108) allows for patches to exist within other patches. This means that existing patches can be reused inside other patches, and that modules can be grouped into subpatches for more elegant wiring. Two levels of subpatches are supported, which means that the synth per patch technically offers an unlimited amount of modules.
- New Effects modules: A **Folder** module (p.96) for wavefolding through classic 0Hz Frequency Modulation, a **SoftClip** module (p.106) for variable softclipping, a **Repeater** module (p.107): a beat-synced, repeated delay, a **Limiter** module (p.104) for soft-knee peak limiting, the **MultiComp** module (p.98)**:** a 3 band multiband soft-knee compressor and a **Boost** module (p.105) for analog boost emulation.
- New Controller modules: The **Phase** module (p.80) which offers control over the phase control signal, and complements the updated **Frequency** module (p.79), a **Scope** module (p.88): a synced oscilloscope, frequency and phase analyzer of up to two separate input signals and internal controls signals, and a MIDI **Keyboard** trigger/visualization module (p.89). **Unison** has been reworked to offer more modes, **Phase** / **LFO** control and **Keyboard Tracking** (p.83) and Pattern module has been updated with **Steptime**, **Velocity** and **Min**/**Max** **Note** range-settings (p.85). **Notepitch** & **Pattern** modules can now be detuned from *Concert Pitch* (A=440Hz).

- New Amplifier modules have been added: An **Inverter** module (p.76) for audio and modulation inversion and a **Mid/Side** module (p.77) for mono/stereo separation. All panning operations are now constant powered, -3dB.
- Filters and been reworked in part to be virtual analog, which gives better stability and allows for higher frequency filter modulation. Three new filters have been added: The **SVF2**, the **SKF** (Sallen-Key Filter) and the **Moog2** (p.64), plus all Resonant Multimode filters now have **Saturation** and **Drive** controls.
- **Exponential Tuning** of all Filter-Cutoffs (p.64) gives better control over both high and low- filter frequencies and sounds more natural when modulated. A **Parabolic Tuning** option is available for backwards compatability. Similarly, a **Resonance** tuning option has been added (p.64), featuring a flat **Digital** response and a simulated **Analog** response that attenuates resonance at low frequencies. Cutoff has a new **Mod Link** option that couples Mod to the Cutoff value at an adjustable rate and curve. The Filters **Keyboard Tracking Base Key** is now also adjustable.
- Modulation is now allowed to be in stereo and parameters that are stereo-capable now have a **Stereo** option added to their right-click-menu.
- New Modulator modules: A **Shaper** module for customizable waveshaping of modulation signals (p.121), a **Control** module that exposes internal control signals (p.122), and a **Clock** for free-running, digital pulse signals based on host PPQ time. **LFO** now supports optional **Random Poly Phase** and **Keyboard Tracking**.
- All Generators now support so called **Through-Zero** frequency, meaning that they accept negative frequencies that generate output in reverse phase. An extra **Phase Control Signal** has been added, to complement the existing Frequency, Trigger, and Note Control Signals. This means better control over phase, which is useful for phase modulation. The FM module has been reworked to support **Linear FM** and **Exponential FM**, with **Through-Zero** support as well as **Phase Modulation** (p.81).
- All Control Signals can now be 16x oversampled in **High Quality** (HQ) mode, forcing generators to run oversampled internally. Essential for high quality FM/PM.
- An **Expression** option has been added to knobs that typically need performance tuning, such as Cutoff, Resonance, Mod, Mod.Index, Sustain, LFO rate & amount, Amplification, etc. responds to Mod.Wheel, Aftertouch, Timbre and Velocity inputs.
- To quickly clean-up patches, the right click menu now offers an **Auto Arrange** option that neatly re-orders modules, and a **Remove Unused** option that deletes disabled or disconnected modules from the patch. A **Swap** menu option quickly swaps modules backwards and forwards in the signal chain. Finally, **Insert** offers a quick menu-based way to wire a specific module in front of another module.
- Control Panel automation controls Min- and Max-values can now be *macro* programmed, to limit parameter values to user-defined ranges (p.35).
- Many parameters now have extra **Range Menu Options** added to their right click menu, allowing for more fine-grained control over, e.g., Attack, Decay, and Release time-ranges in all envelopes, all Portamento time ranges, Modulation Index ranges, Delay Finetune ranges, LFO rates, Chorus Depth ranges, etc.
- Modules can now be connected using a simple one-click mouse-action + connections now highlight when mouse-hovering in connect mode.
- A set of new patch banks have been added. Patch banks can now be read-only, and loaded while compressed on disk. New patch banks can now also be downloaded and installed instantly from within the synth using the **Bank Install** menu, available under Options (p.32).

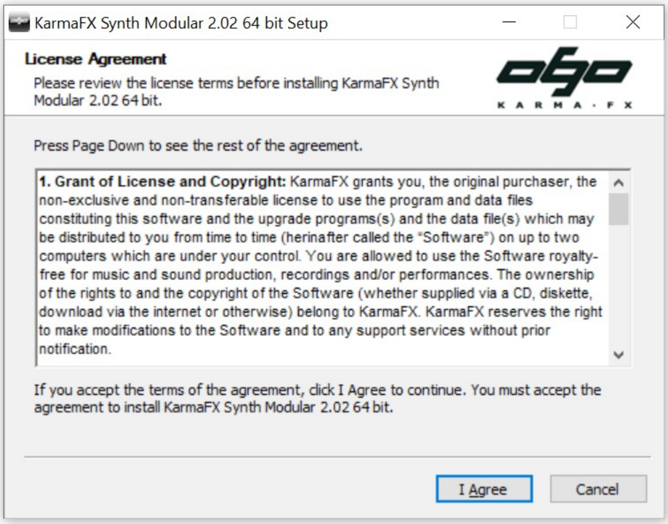
These additions are just the most prominent features. Please refer to the full changelog online for more technical details.

3. Installation

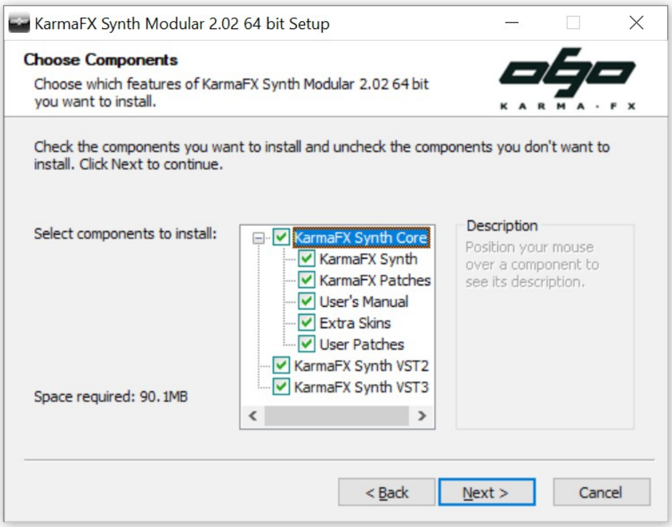
3.1 Installation on PC/Windows

This section will show you how to install the Synth on your Windows PC. Make sure to close all running sound applications before starting the installation. Then…

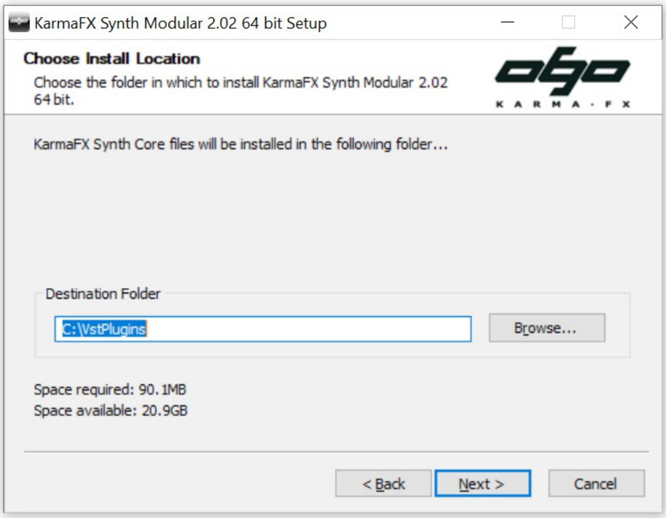
▶ Locate and run the the installation file. Use **KarmaFX Synth Modular 2 32 bit** or **KarmaFX Synth Modular 2 64 bit** for either 32-bit or 64-bit installations respectively.

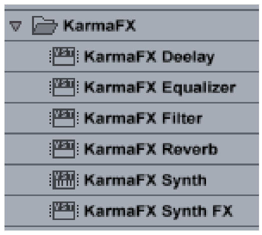


▶ After reading the License Agreement, click **I Agree**. This brings up the installation selection menu:

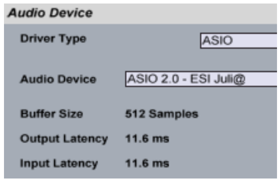


▶ Everything to be installed is checked by default, so simply click **Next** to proceed.





ASIO (Audio Stream Input Output) is a protocol for low-latency digital audio specified by Steinberg.



► Now select the folder where you would like to install the synth's core files and data. In order to save patches and presets inside the synth, <u>it is important that this folder is user-writeable</u>. We suggest:

```
C:\VstPlugins
```

Next, select the folder where your host application's VST2-plugins are placed. Some hosts may require a different path, but in most cases you can choose it yourself. Please refer to your host application to get the required folder path. The installer will attempt to auto-detect this folder. Typically it is:

```
C:\Program Files (x86)\Steinberg\VstPlugins
```

Finally, click Next to select the folder to install the VST3 plugins into. Typically this is:

```
C:\Program Files\Common Files\VST3
```

Again, the exact folder paths may vary, so please consult your host's preferences to confirm the correct locations for VST2 and VST3 plugins.

Click **Install** to install the Synth into these folders.
Afterwards, click **Close** to end the installation program.

►Start you host application. The host will usually scan for new plugins on start-up. If this does not happen, make sure to re-scan for VST-plugins. A **"KarmaFX Synth"** and **"KarmaFX Synth FX"** plugin should appear under VST-instrument and VST-effect, respectively.

►Finally, make sure your host applications sound output latency is set as low as possible. For best results, use an *ASIO driver* if possible and set the latency buffer to 128, 256 or 512 samples. The latency buffer is filled with sound samples before they are output to the sound card. This means: The longer the buffer, the longer the latency, or delay, before the sound reaches your ears. Small latency is better, since this means that the time it takes in milliseconds from a key hit on your keyboard and until your actually hear the sound, is insignificant.

That's it! You are all set.

This manual in PDF format is located in:
`<core folder>\KarmaFX\KarmaFX_Synth\Manual`

The KarmaFX Synth Modular Synth & FX VST2 plugins are:
`KarmaFX Synth.dll`
`KarmaFX Synth FX.dll`

The KarmaFX Synth Modular Synth & FX VST3 plugins are:
`KarmaFX Synth.vst3`
`KarmaFX Synth FX.vst3`

It is okay to move these plugins from their respective installation folders to another folder, without breaking the installation.

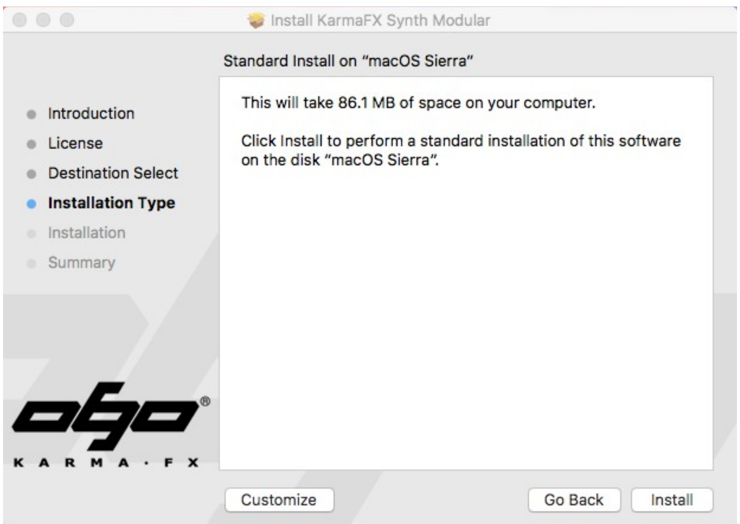
To uninstall the software, please run the supplied uninstaller:
`<core folder>\KarmaFX\UninstallSynthModular.exe`

3.2 Installation on Mac/OS X

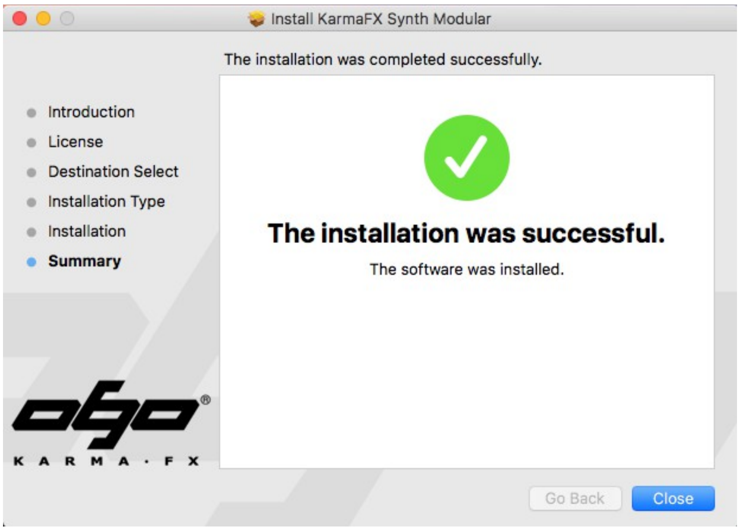
For installation on Mac/OS X, first make sure to close all running sound applications. Then...

▶ Locate and run the supplied PKG installer and follow the instructions. Use **KarmaFX Synth Modular 2 32 bit** or **KarmaFX Synth Modular 2 64 bit** for either 32-bit or 64-bit installations respectively.

▶ Click **Continue** to agree to the License Agreement. After that the installation files can be customized by clicking the **Customize** button. However, everything to be installed is checked by default, so simply click **Install** to proceed with standard installation.



▶ Once the installation completes, you will be presented with the "The installation was successful." message shown below. This concludes the installation and you can simply click **Close**.



▶Finally, you may start your sound host application. And that's it! You are all set.

The host will usually scan for new plugins on start-up. If this does not happen, or the synth doesn't show up, make sure that it is a VST- or Audio Unit compatible host and that Audio Units support is enabled. Rebooting your Mac may in some cases be needed to force a rescan of the Audio Units folder.

The KarmaFX Synth Modular Synth & FX Audio Units are called:
`KarmaFX Synth.component`
`KarmaFX Synth FX.component.`

After installation, they are located in:
`/Library/Audio/Plug-Ins/Components/`

VST plugins are located in:
`/Library/Audio/Plug-Ins/VST/`
`/Library/Audio/Plug-Ins/VST3/`

The core files for the plugins, patches, presets etc. are stored in:
`/Library/Application Support/KarmaFX`

This manual in PDF format is located in:
`/Library/Application Support/KarmaFX/Manual`

PREVIEW

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Typically, a modular synthesizer is controlled by a keyboard. So playing a melody on the keyboard's keys will play the created sound accordingly at varying pitch. This is done by controlling the modules frequency (e.g., VCO) by a connected keyboard using two signals: A *Frequency* or *Keyboard Control Signal* that controls the module's frequency depending on the pressed keys, and a *Trigger Control Signal* that tells when a key is pressed and released.

Moog Modular

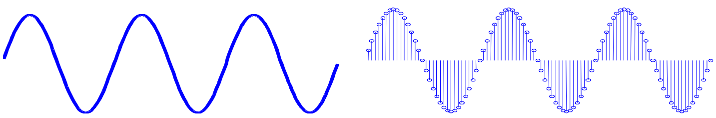
Historically, the first modulars had real physical modules that the musician connected with real cables to construct a sound. One of the first modulars was the *Moog Modular Synthesizer*, designed by Bob Moog back in 1963. Later came modular synthesizers from other manufacturers (*ARP, Serge, Buchla, EMS* and the *Roland System 700*). Although they were expensive at first, musicians and sound designers were awe-struck by the ability of these machines to create new unique sounds as well as imitate existing instruments.

ARP 2600

Modular synthesizers have since had a great impact on music evolution. Especially electronic music has been influenced by their capability to create sounds that do not necessarily sound like ordinary instruments.

KarmaFX Synth simulates the exact same signals as in an analog modular synth, but in a digital system, namely the computer. The difference between an analog and a digital signal is simply that the digital signal is a sampled version of its analog brother, i.e., it is made up of a series of numbers instead of a continuous voltage.

Sample rate is the number of samples per second



continuous analog signal *sampled digital signal*

The sample rate determines the quality of a digital signal, i.e., the higher the sample rate the better the approximation to the analog world. For sound, a frequency of 44100 samples per second (44.1kHz) is normally considered adequate for human hearing. For the best possible quality, KarmaFX Synth internally uses the same high frequency for both sound and control signals.

In fact, the Synth does not distinguish between audio and control signals. They are treated the same and can be used interchangeably like in a true analog modular system.

PREVIEW

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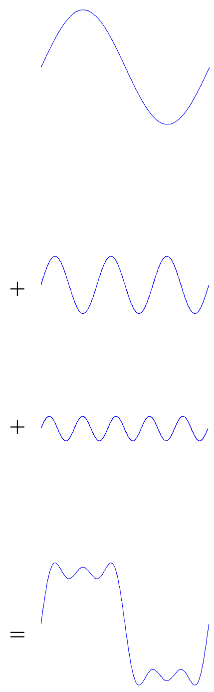
Additive Synthesis

Additive synthesis is a technique that generates sounds by summing several sine waves at different amplitude, frequency, and phase. The idea is that since all sounds can be decomposed into sinusoids, all sounds can (in theory) be synthesized using sinusoids too.

Any sound produced by an instrument typically consists of a periodic waveform starting with a fundamental sine wave. This sinusoid has the lowest frequency of the sound and dictates the resulting sound’s musical pitch. Any additional sine waves have a frequency that is an integer multiple of the fundamental frequency. These sinusoids are called the sound’s *harmonics*.

An additive sound’s harmonics phase, frequency and amplitude can be changed over time to make the sound more interesting. Drifting sounds are often perceived by humans as beautiful, and additive synthesis is therefore good for modeling pads, evolving textures and ambiences.



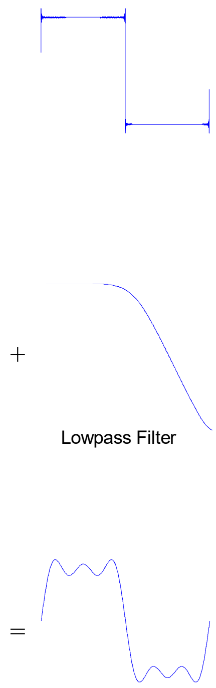
Subtractive Synthesis

Subtractive synthesis is the exact opposite of additive synthesis. Here a sound is created by starting with a complex waveform and then modifying this harmonically rich sound by using filters to attenuate and/or boost frequency content.

The common Lowpass filter has a cutoff frequency and resonance setting. The cutoff frequency determines at which frequency the filter should start to attenuate, while the resonance controls the amount of boost to apply near the cutoff frequency. This is good for simulating the natural timbre of many instruments.

Typically, the filter is varied to change the timbre of sound over time. Likewise the amplitude can be changed to simulate the rise and decay of the sound.

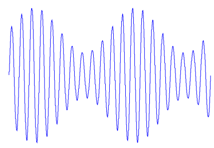
Historically, subtractive synthesis gained popularity due to its use in analog voltage controlled synthesizers. A simple circuitry can be used to generate the common sawtooth, square and triangle waveforms, all rich in harmonic content. It is simple to control and can produce a wide variety of sounds.



Amplitude Modulation (AM): Envelope amplitude modulation is used to shape a sound. Low frequency periodic Amplitude Modulation creates a *tremolo* effect.

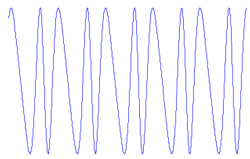
Tremolo is a modulation effect created by varying the amplitude, or volume, of the signal.

Frequency Modulation (FM): The source (*Carrier*) signal's frequency is modulated by a, normally periodic, modulation signal (*Modulator*). This changes the timbre of the sound resulting in a more complex waveform by creating so called *sidebands*. When creating harmonic sounds, the frequency of the Modulator must have a harmonic relationship to the Carrier, i.e., if the frequency of the carrier is n, then the Modulator's frequency could be 2n, 3n, 4n og ½n, etc. However, frequencies that are non-integer multiples of each other (non-harmonic) can also be used, e.g., for creating bells and percussive sounds. Low frequency FM gives the musical impression of *vibrato*.



Amplitude modulated sine

Phase Modulation (PM): At low frequencies, Phase Modulation works like Amplitude Modulation. At high frequencies it gives similar results to Linear Frequency Modulation.



Frequency modulated sine

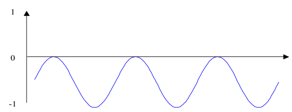
Pulsewidth Modulation (PWM): For periodic waveforms like square and triangle it is possible to control when the waveform changes sign; the waveform's so called *duty-cycle or pulsewidth*. Modulating the pulsewidth, i.e, changing the fraction of time the signal is "on" instead of "off", gives a result similar to adding many identical waveforms running slightly out of phase, resulting in a fatter sound.

Filter modulation: Modulating, e.g., Lowpass, Highpass, Bandpass filter parameters (like the Cutoff frequency) is often used to open or close a sound over time. A filter can also be forced to (more or less) follow the frequency of the frequency control signal (*keyboard tracking)* in order to keep the same timbre regardless of the waveforms pitch.

Ring modulation: Ring Modulation multiplies a source signal with a modulation signal (typically a sine wave). The resulting output is the sum and difference between the two signals; good for creating bell-like and metallic sounds.

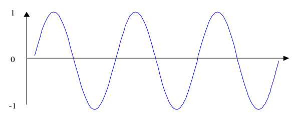
Ring modulation got its name due to ring-shaped diodes used in the analog circuit that originally created the effect.

We normally distinguish between two kinds of control signals: *Unipolar* and *Bipolar*. Unipolar has always all positive or all negative amplitude values.



Unipolar signal

Bipolar has both positive and negative values centered around 0. Typically, modulation signals are unipolar, while sound signals are bipolar.



Bipolar signal

Analog vs. Digital

Digital synthesis is in many ways superior to analog synthesis. It is free from static noise and easier to control. However, digital audio has one drawback called *Aliasing*: The Achilles heel of digital synthesis.

Aliasing is a type of distortion introduced by trying to represent frequencies above or equal to half the sample rate: The so called *Nyquist frequency*. Frequencies above that frequency will wrap around the Nyquist frequency and introduce ordered noise in the high frequencies.

Aliasing is usually unwanted, and KarmaFX Synth goes through great lengths to avoid, remove and reduce aliasing by using techniques such as high quality interpolation, band-limiting and oversampling. Still, in some cases, particularly in sample-based and extreme FM/PM synthesis, the Synth cannot avoid aliasing completely. For instance, if you sample a harmonically rich sound played at a low key and play it back 5 octaves higher, aliasing will be heard. How bad it sounds depends on the sample, the pitch, the interpolation technique, etc.

Standard CD quality runs at 44100 Hz. This gives a Nyquist Frequency of 22050 Hz, which all signal frequencies should be below to avoid aliasing.

To reduce aliasing one option is of course to increase the host's sampling rate (e.g., to 96kHz) at the cost of extra CPU cycles, but in general you should not really worry about aliasing. For the majority of sounds this is usually not a problem. Besides, you can force aliasing out of almost every digital hardware synth on the market, even the really expensive ones!

MIDI

MIDI (Musical Instrument Digital Interface) is a standardized protocol used for communication between keyboards, MIDI controllers, computers, synthesizers, and even application hosts and VSTs.

Typically, MIDI is used to send *note on* and *note off* events when a note key is pressed and released, respectively on a MIDI keyboard. Data, such as what key (*note index*), how hard it is pressed (*velocity*) is also sent.

MIDI Keyboard/Controller
(Novation Remote SL)

Likewise, control data can be transmitted so parameters can be changed during a song or live performance either using a synthesizer or dedicated MIDI controller. Midi control signals are sent as a sequence of events, where each event carries a value between 0 and 127. MIDI events are usually referred to as *MIDI Control Changes* or simply *MIDI CC*'s.

MIDI Controller
(Behringer BCF2000)

Typical MIDI CC's include *Pitch bend*, and *After touch*: Pitch bend events are usually sent from a MIDI keyboard or instrument in response to changes in position of the pitch bend wheel. After touch tells how much pressure is applied to a key while its being held. Many other MIDI CC's exists (referenced by their hex value) and can be mapped to control user-defined parameters.

4.3 Synthesis in KarmaFX Synth

KarmaFX Synth is built on the same principles as Subtractive-, Sample-, and Additive-synthesis, plus of course Modulation synthesis due to its modular nature, making it a capable and flexible synth and effect unit.

Like most other synths, a defined sound in KarmaFX Synth is called a *patch*. A Patch can be played either *monophonically* or *polyphonically*, but only one patch can play at a time in one instance of the Synth. However, the number of simultaneous synth instances is only limited by your systems CPU and memory resources.

A Monophonic patch only plays one voice while a Polyphonic patch can play multiple simultaneous voices.

Each patch consists of a set of connected modules. Just like a real modular, a module is a single component that either generates or processes a signal. A module has user adjustable parameters, i.e., knobs and sliders, that control its functionality. Almost all parameters can be controlled (modulated) by signals from other modules. We will refer to these signals as *modulator signals* or simply *modulation signals*. Each module can also read and react to MIDI events.

Most modules can work in either mono or stereo mode. If a stereo signal is sent to a mono module, only the left channel of the signal is processed. However, if a mono signal is sent to a stereo module, the mono signal is duplicated to both stereo channels.

Because modules solve different tasks, they are placed into one of these basic categories: Generator, Filter, Amplifier, Effect, Controller, Modulator, or Output.

Generator: A generator creates a sound signal at a certain frequency and hence does not take any inputs. A typical generator is an oscillator producing, e.g., a sawtooth or a square wave.

Filter: A filter processes the input, in the spirit of subtractive synthesis, by removing or boosting certain frequencies. A typical filter is the familiar resonant lowpass filter with simple cutoff and resonance control, known from analog synthesizers.

Amplifier: An amplifier module simply changes the volume (loudness) of any incoming signal, usually triggered by MIDI note-on and note-off events. A typical amplifier is a simple *gate* that turns the volume up when a key is pressed and down when released.

Controller: A controllers has three main tasks. First, it creates control signals (frequency/phase/note/trigger) that are sent to every module feeding the controller. Filters can, e.g., use the frequency control signal for keyboard tracking while oscillators use it to change pitch. Secondly, the controller controls the amount of polyphony, i.e., how many voices the patch is capable of playing. All incoming voices are mixed before they are output from a controller. Finally, controllers can write and alter MIDI events. In polyphonic mode, a frequency control signal is generated for each voice and the controller determines which MIDI events should go to which voice.

Effect: Effects simply process the input in some way. A typical effect is, e.g., delay and reverb. Since effects rarely need polyphony (and in some cases are quite CPU heavy), they are most often placed last in the signal chain.

New modules that are introduced in Version 2 are shown with a:

NEW IN V.2

tag in the Modules Guide, page 52.

Modulator: Modulators create or process signals that are meant for controlling parameters, i.e., instead of routing a modulator signal into the sound signal chain, it is wired into a knob to control its value over time. A typical modulator is a Low Frequency Oscillator (LFO) outputting a sine wave.

Output: The output module is a special module that simply sends its given input to the host application. MIDI communication from the host also passes through this module.

The use of modulation signals is what makes modulars so powerful. Any patch cable's voltage can cause changes to one or more parameters of a module. The same basic principle applies in KarmaFX Synth, but some signals are handled somewhat differently: As previously mentioned, the *frequency control signal* is traditionally sent from a keyboard to individual modules (along with a trigger/gate signal to tell when a key is pressed and released).

In KarmaFX Synth this signal still exists, but is created by a Controller module based on received MIDI events. The frequency control signal is then passed on automatically to all modules that transmit their output to that Controller. Triggering is handled by passing on events to busses of MIDI events, which internally generates a *trigger control signal* and *note control signal*. All modules can read and react to the flow of MIDI events on these busses. Aside from the frequency control signal, Controllers can also generate a *phase control signal*.

The neat thing about this design is that you have less control signals to worry about. For instance, typically a keyboard *control voltage* (CV) would have to be wired into many different modules. In KarmaFX Synth this is done for you, by means of the internal frequency control signal, "behind the scenes" so to speak.

5. User Interface

In order to operate KarmaFX Synth, it is essential to understand its interface. When the synth loads up you'll be presented with a modular environment built up of small, boxed, modules placed on a Workspace and connected by wires.

Modules can be dragged around and connected arbitrarily using the mouse. Each module carries out a task, either generating or processing a sound signal that ultimately is sent to an output module and received by the host.

Each connection is shown as a wire with a small arrow indicating which direction the signal flows. A module has user adjustable parameters, i.e., knobs and sliders, that control its functionality. Almost all parameters can be controlled (modulated) by signals from other modules.

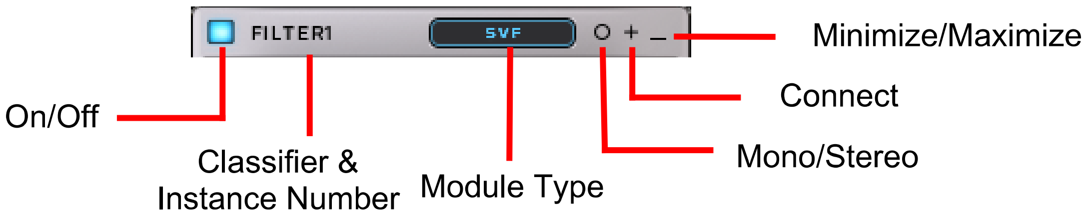
Modules generating signals that modulate parameters (Modulator modules) are directly connected to the particular parameter's knob or slider. In the bottom of the Workspace is a Control Panel, useful for quickly editing common parameters as well as for browsing and managing patches.

The following sections, describe how to use the modules, the wires, the Control Panel and the menus.

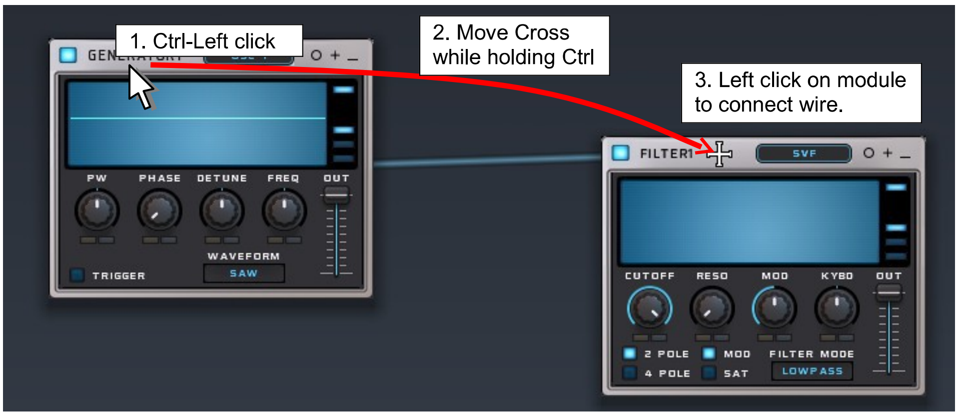
5.1 Modules

A module consists of a *title bar*, a *display area* and a *parameter area*. Some modules also have a special user-interaction display for, e.g., editing envelopes or to display samples. The details of how to operate these is covered in section 7.

Title bar

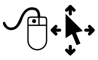
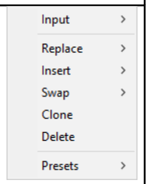
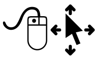
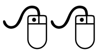


The first thing you notice on the title bar is the *module classifier*, e.g. *Generator, Filter, Modulator*, etc., and its current *instance number*. A dropdown menu chooses the *module type*, i.e., the functionality of the module. If it's a filter module, the dropdown will show a selection of available filters. You can use this selection to quickly try out different types, without breaking the wiring. Audio modules capable of stereo processing have a *Mono/Stereo* switch. A *Connect* button lets you connect one module to another module or to any knob or slider for modulation. A *Minimize* button collapses the module so it only takes up a small part of the screen. Finally, an on/off LED switch can be used to switch a module on or off. When the LED it lit, the module is on. When a module is off it uses no CPU and has no output. Connecting modules is shown below:



Connecting two modules using the mouse: Either hold Ctrl and click the titlebar <u>or</u> click the *Connect* button. Then click on the target module.

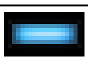
You can have up to 16 simultaneous instances of each module class, i.e., sixteen generators, sixteen filters, sixteen controllers, and so on. The only exception is the Output module: There can only be one Output module.

Title bar - Actions	
	**Pressing left mouse on title bar + Drag** Lets you move the module around on the screen. A module automatically snaps to an invisible grid to make it easier to align with other modules. The actual position of the module itself doesn't affect the sound or routing in any way. You should simply try to set it up so it is pleasant to work with.
	**Right click on title bar** Brings up a menu allowing you to select the active *Input* to this module, *Replace* the module with a different kind, *Insert* another module in-front of this, *Swap* the module backwards or forwards in the signal chain, *Clone* the module (and its parameters), *Delete* the module, or save the modules parameters as *Presets* for later retrieval. 
Ctrl 	**Hold Ctrl, click title bar and then on the target module** Press and hold **Ctrl** and click once on the title bar. A small cross appears. Then, while still holding **Ctrl**, click on the title bar or the display of the module you wish to connect to. A wire will appear between the modules to show connection. Alternatively, instead of holding Ctrl, simply click the Connect button.
	**Double click on title bar** Minimize or maximize module (same as clicking on the minimize/maximize icon).

Display

The displays main purpose is to show the final waveform produced by the module. Have you ever sat in front of a silent hardware or software synth and wondered where the sound went, just to realize that the amplifier or the filter was switched off? Then you already know that problems like these are both annoying and time consuming. Moreover, they are even more likely to happen in a modular environment when a lot of stuff is going on. Because of this, KarmaFX Synth offers plenty of visual feedback to keep you in control at all times.

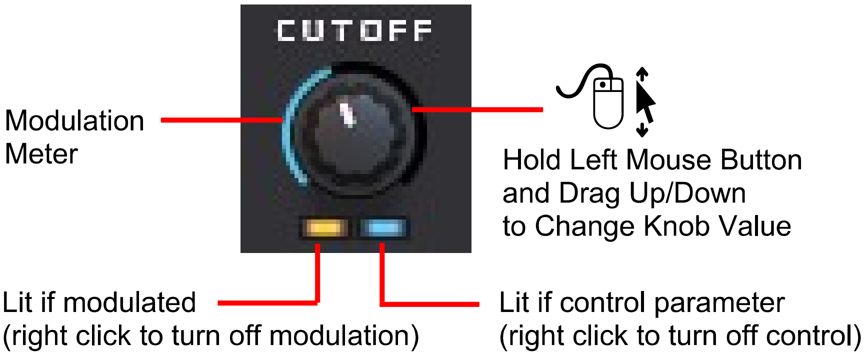
The display is controlled by the four switch buttons to the right of the display. The text area just below the waveform display shows info related to the current module, and parameter values whenever they are changed.

	The first button turns the display on or off.
	Second button switches to time domain display, showing the waveform like an oscilloscope. (This is the default)
	Third button switches to show the frequency spectrum.
	Fourth button turns the display into a special modulation mode, showing the current modulation of the last edited parameter in the module. In this mode, clicking on any of the modules knobs/sliders will show the modulation of that parameter.

Modules running in polyphonic mode will only display the first generated voice in the display. This means that you may hear sound that you cannot see. To alleviate this you can temporarily switch polyphonic controllers to mono mode, observe the generated sound, and then switch it back when done tweaking.

Parameters: Knobs, Sliders and LEDs

Parameters are the most important interaction components in the synth. As on hardware synthesizers they allow you to change and fine-tune a module's settings – thus controlling the final sound of the patch. To change a knob or slider, simply press the mouse button and move the mouse up or down; or use the mouse' scroll wheel for fine tuning.

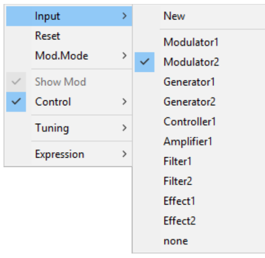


Tip: When changing a parameter, moving the mouse slowly will give you greater precision. Holding the Shift key will also increase precision.

As a special feature, parameters with knobs have two indicator LEDs: The left LED is lit if the parameter is modulated. The right LED is lit if it's a control parameter.

Tip: Holding Ctrl and left clicking on a knob or slider resets its value to center-position.

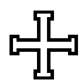
Almost all parameters in the synth can be modulated. In this context "modulation" simply means controlling a parameter by a signal from another module. To modulate a parameter, simply route the modulator module directly to the knob or slider you wish to modulate. The *Modulation Meter* shows the current modulation value.



To connect a module to a parameter either right click on the knob/slider and select the source module from the Input menu. Alternatively, either press and hold **Ctrl** and click on the title bar, or on the title bar's Connect button on the source module, and then click on the knob/slider you want to modulate. In any case, a modulation wire will appear showing the connection.

Connecting an LFO modulator to the Filter Cutoff knob

Tip: Choosing *New* in the Input menus adds a new module and automatically patches it to the respective module or knob/slider.



Connection cursor

If more than one module is connected to a parameter, their signals are internally mixed before modulating its value.

PREVIEW

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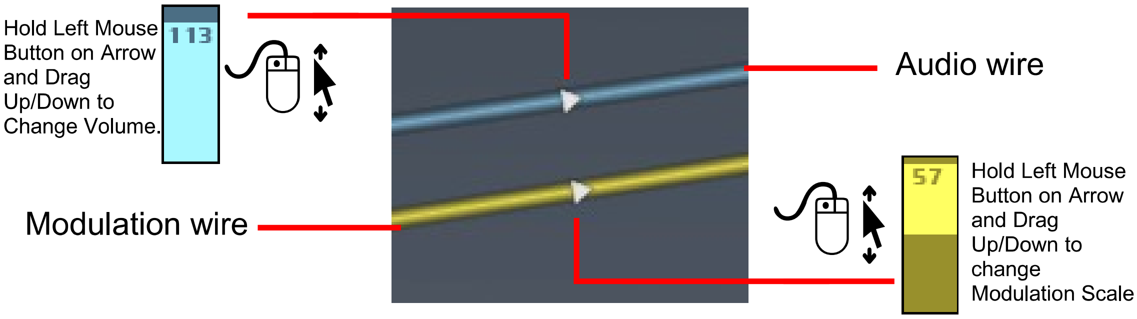
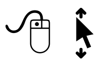
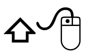
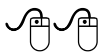
5.2 Wires

The wires connect modules to modules (audio wire) and modules to parameters (modulation wire). Thus, there are two kinds of wires, shown in different colors. A small arrow indicates the direction of the signal flow.



As the above figure shows you can alter the amplification of the signal sent through the wire using the mouse on the arrow head. Right clicking on the arrow cuts the wire and removes the connection.

Wire Arrow - Actions	
	**Pressing left mouse on wire arrow + Drag up/down** Change the amplification of the signal. For audio wires the amplification range is 0 to 127 [0..1]. For modulation wires it is –64 to 64 [-1..1], making it possible to invert the signal.
	**Right click on wire arrow** Cuts the wire and removes the connection.
 or 	**Shift Left Click or Double click on wire arrow** Insert module on wire. First, the *Add Module* menu is shown. The selected module is then inserted on this wire, preserving all connections.

The synth does not distinguish between audio and modulation signals. They are treated exactly the same and can be used interchangeably like in a true analog modular system. The wires are only colored differently to make it easier for you to see what's going on.

This means that you can easily modulate a knob or slider using another audio signal, or - if you choose - send a modulation signal (e.g., from an LFO) into an audio module.

PREVIEW

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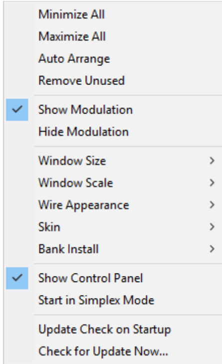
5.4 The Options Menu



The Options menu gives access to overall settings in the synth:

Minimize/Maximize All: Minimizes (collapses) or Maximizes (expands) all modules in a patch. The minimize mode is useful for getting a quick overview of the wiring in the patch, since everything but the title bar of the modules is hidden, leaving good room for the wires. Modules are by default always maximized.

Auto Arrange: Relocates all connected modules into a grid starting from top left corner and moving across the screen. This can sometimes help clean-up a messy patch where modules have been moved so they either overlap or aren't aligned with the grid.

Remove Unused: Deletes modules that are either disabled or not connected, while retaining all existing wiring. This is a quick way to clean-up a patch that contains many inactive modules, while ensuring that it sounds exactly the same.

Show/Hide Modulation: Shows or hides modulation of all modulated parameters. When *Show* is enabled, knobs and sliders are animated to show what effect the modulation has. This is the same as right clicking on each parameter and selecting “Show Mod.”. It is useful when creating new patches, to verify that parameters behave as expected when modulated. Default mode is to hide all modulation. This option is now superseded by the built-in knob metering in the version 2 skins, but is still available for backwards compatibility with the classic skins.

Window Size: Choose between 1500x1000 (default) or custom resolutions (up to 4096x2048; minimum size is 400x400). Low resolutions are useful when running on laptops with limited screen size. The window size should change instantly. If not, please close and reopen the plugin. Patches saved in custom resolutions higher than the default might appear differently when reloaded in the default resolution, as modules may have to be relocated to make them fit on screen.

Window Scale: Scale the UI by 100% (default) or 200%. On skins that support it, extra high-quality scales are available: 125%, 150%, 175% 200%. This is useful when running on a system with a high resolution screen setting, such as Retina or similar high DPI resolutions. The window size should change instantly (and if not, on reopen). On Mac/OSX Retina scaling should be manually activated using the HiDPI/Retina setting.

Wire Appearance: This changes the way the wires are drawn. Choose between Linear (default), Curved or Dangling. This is basically just eye-candy and a matter of taste. Just choose what suits you best.

Linear Curved Dangling

PREVIEW

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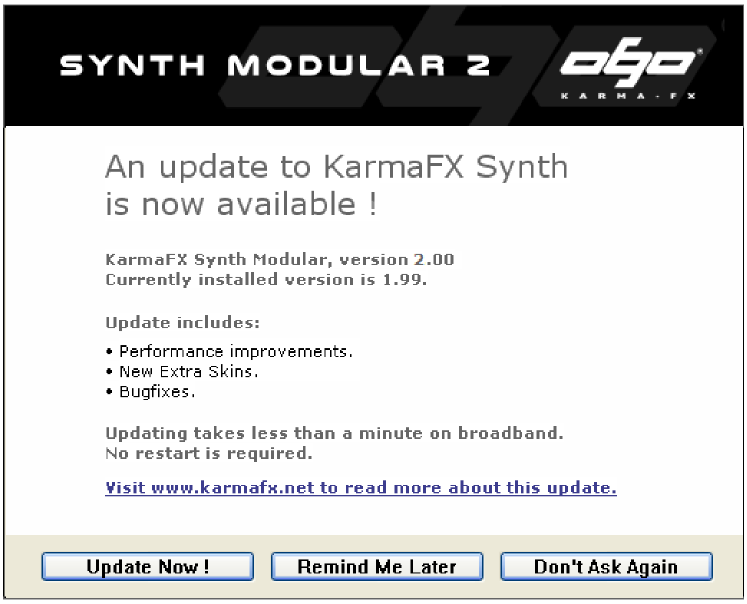
Show Control Panel: When enabled (default and recommended setting) a Control Panel is shown in the bottom of the screen. How to operate the Control Panel is covered in section 5.5.

Start in Simplex Mode: When enabled, new instances of the synth will start in *simplex mode*. Simplex mode is a special patch editing mode where the interface is shrunk so only the Control Panel is shown. This of course takes up a lot less screen space but also limits patch interaction considerably. You can toggle simplex mode on and off manually by clicking on the Synth modular logo in the Control Panel.

Synth running in simplex mode

Update Check on Startup: When enabled and internet access is available, the synth will check for new updates on startup.

Check for Update Now: Use this to check for updates manually. When a new update is found, the version information is shown and the updates main new features are listed, and you'll be prompted whether to install it or not. Updates can require the system to download and launch the full installer, so the host application will have to be closed down and restarted once the installation completes.



5.6 The Patch Browser

The Patch Browser offers a way to quickly browse patches and patch banks. The left side of the Patch Browser window is the *bank view* that shows the available banks, while the *patch view* on the right side shows all the patches in the currently selected bank.

You can switch bank by simply left-clicking on a bank name in the bank view. To scroll the bank view, you can click on the arrows next to the view or use the mouse-wheel while hovering.

To select a patch, you can left-click on the listed patches in the patch view, and it will load instantly.

On hosts that support plugin keyboard-input (and while the Patch Browser has focus), you can also use the keyboard's arrow-keys to navigate both patches and banks, or even switch between the bank and patch views. This is useful if you want to quickly browse the patches, while trying to find a particular sound.

Using the keyboard, it is also possible to filter search the patch view. To enable keyboard filtering, click the panel button on the far right. Then type the name of a patch to instantly filter the patch view according to the typed search text. Only the patches that match the text or part of the text will be shown.

The search text is shown in the bottom right of the patch view. To clear the filter, either repeatedly press backspace to remove the text, hit the Escape key, or simply click the panel button again. Alternatively, closing the Patch Browser and opening it again will also clear the filter.

To close the Patch Browser, either click the Load button again, or hit the Escape key. Clicking on the Workspace background will also close the Patch Browser window.

6. Basic Operation

So far we have looked at the Synth's interface and its main features. Now, it's time to start using those features! This section shows how to setup KarmaFX Synth in your host application (either as instrument or effect) and start browsing the built in patches. Once familiar with the basics, we will use our newly gained knowledge from the previous sections to build a simple patch and save it for later retrieval.

6.1 Using the Synth as Instrument and Effect

Before you can start using the Synth, you need to know how to insert and use VST/Audio Unit plugins inside your host application. The way to do this differs from host to host, but normally you'll have the option to drag or select a VST/Audio Unit from a plugin-list and place it in an audio and/or MIDI track.

The Synth can operate in two modes: As instrument (VSTi/Audio Unit Instrument) or effect (VST/Audio Unit Effect).

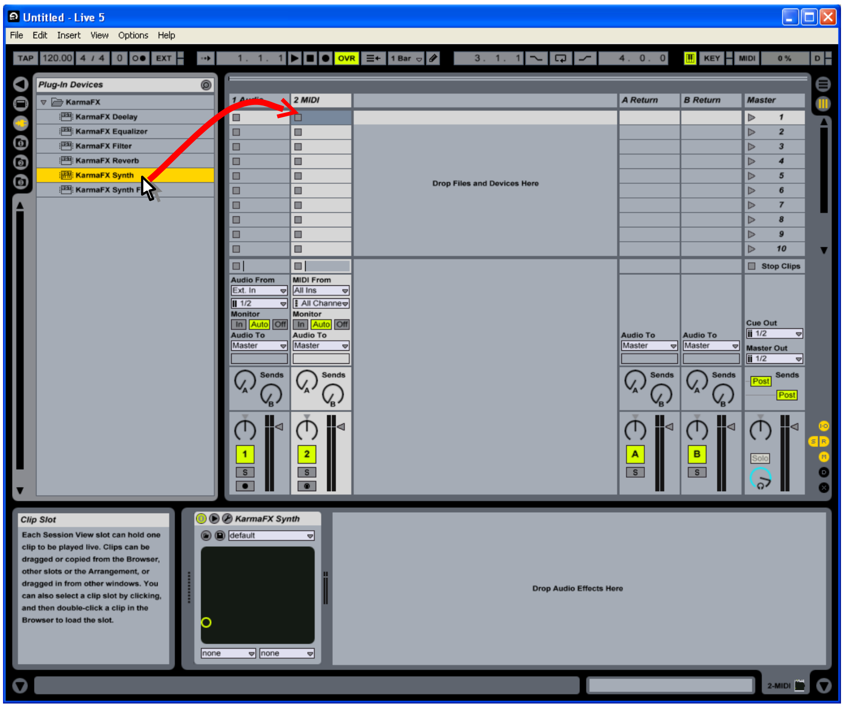
If the synth should generate sound based on the played nodes, it should be used as an instrument. The instrument generates audio based on MIDI events, so it has to be placed in MIDI track.

If the synth should alter the sound output of a track, it should be used as an effect. The effect needs to be put after any audio generators, and can be placed in an audio-, aux- or even master track.

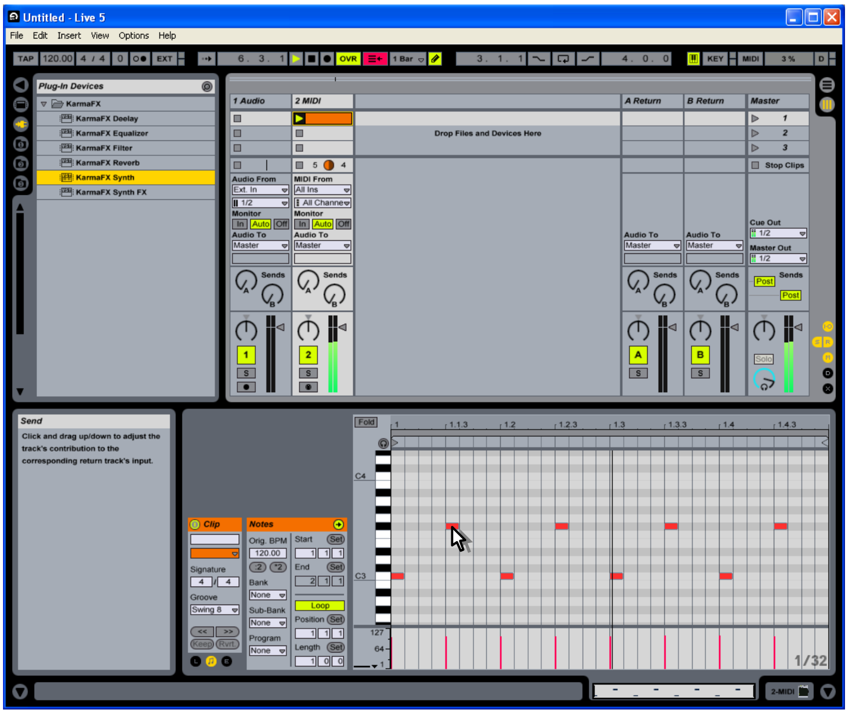
The following illustrates how to use KarmaFX Synth in the host Ableton Live. You host of choice might work differently. If in doubt, please refer to your host applications users manual to learn how to use VST plugins.

Using KarmaFX Synth as Instrument in Ableton Live

► Locate the KarmaFX plugin folder in the host's list of VST plugins. Then drag KarmaFX Synth to a vacant MIDI track.



► Enter a sequence pattern and hit play to hear the synth in action. By default the synth starts up with a default bass patch.

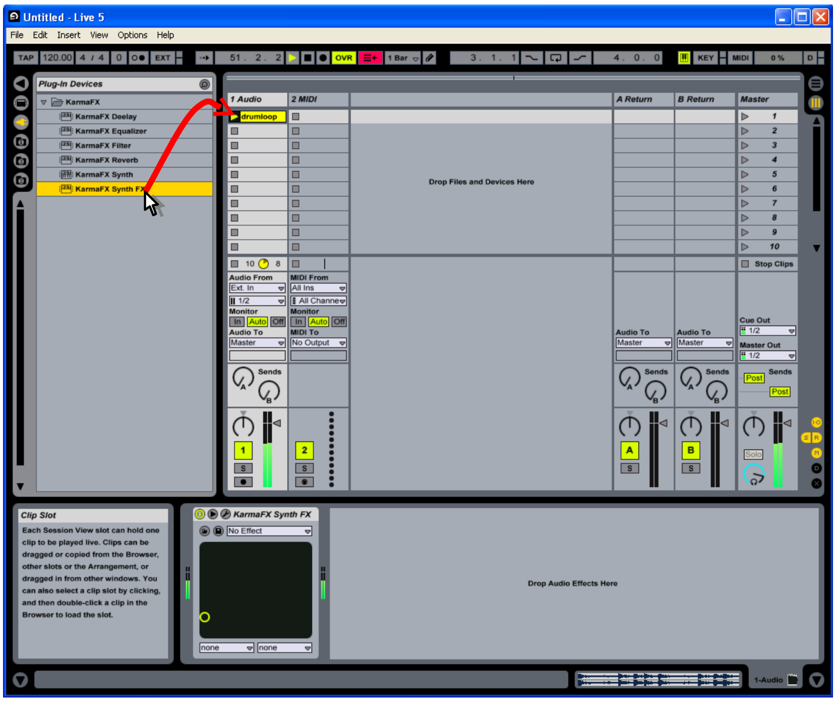


► Alternatively, if you have a MIDI keyboard connected, click on MIDI input monitoring, and play a few notes on your keyboard.

Using KarmaFX Synth as Effect in Ableton Live

▶ First, click on the audio track you wish to add the effect to (preferably one that generates sound).

▶ Now locate the KarmaFX Synth FX in the VST plugin list and drag it to the effect area in the bottom of the screen.



▶ By default KarmaFX Synth FX start up with the KarmaFX Effect bank, so you can start using it as an effect immediately. The default effect patch does exactly nothing, but browse down a few patches and you'll find a simple delay. Then tweak a few knobs and hear how it changes the decaying echo.

6.2 Browsing and Handling the Built-In Patches

The synth has more than 1000+ different patches to choose from. Patches are placed in *banks*, where each bank contains up to 128 patches. The banks are arranged and named according to their category. For instance, there is a "Bass" bank, a "Pad" bank a "Drum" bank, and so on.

By default the synth starts up with the "KarmaFX Bass" bank (or "KarmaFX Effect" bank when using the effect plugin). To browse/use and listen to the many patches in a bank there are several options:

▶ Use the Patch Browser to choose bank and patch directly (See section 5.6). To open the Patch Browser, use the Load button in the Control Panel or simply click the Control Panels display.

▶ Use the Control Panel's up/down arrows to go to the previous/next patch.

▶ Right click on the background and select "Load Patch…". Then select a patch from the submenu that appears. Note that, when this is invoked from within a subpatch, the patch is loaded into the subpatch.

▶ Select a patch directly from within your host as a VST preset (The Synth's GUI does not have to be open to change patches this way).

Once a patch is selected, play a few notes on your MIDI keyboard or start to loop a sequence in your host to hear it.

To switch bank choose the "Bank Select" menu item from the right click menu or click the bank button on the Control Panel or use the Patch Browser. Selecting a new bank will not alter the currently loaded or edited patch, so to actually load a patch from the newly chosen bank you have to actually load a patch. Besides not losing any unsaved changes to a patch, this has the advantage that you can copy a patch from one bank to another (see section 5.3).

Once a bank has been selected you can also browse the patches from within your VST host application. The patches will appear as *program presets* in the host. You can even export the selected patch as an .FXP file in most host applications, if you would like to store a patch outside the Synth, e.g., to send a single patch to a friend. You cannot however export an entire bank as .FXB as some host applications permit. To take a backup or share an entire bank with a friend, you should locate the KarmaFX_Synth/Patches folder in the installation folder, and copy the folder with you bank's name.

On disk, a bank consists of a collection of patch files (.kfx) and an index file (index.txt) located in a single folder. Normally you wouldn't have to mess with these files. Still it is nice to know that if a bank's index file is missing, the Synth will attempt to recreate a new index file from the files residing in the folder so that it will load correctly.

Always be sure to backup your homemade patches stored within the built-in banks before installing newer versions of the Synth. Otherwise these patches might be overwritten on installation. To be on the safe side, it is recommended that create a new bank with a unique name and store your own patches there. Then, when re-installing the Synth, your patches should not be overwritten. But it is of course always a good idea make a backup copy - just in case.

6.3 Creating Your First Patch

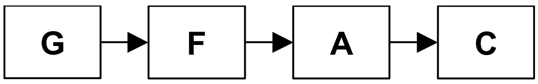
Let’s build a patch from scratch! We’ll start by making a patch template schematically to make this section as general as possible. In fact we can use this template as a basis to construct almost any future patch.

▶ First, we place a generator (denoted **G**). What kind of generator is not so important. It simply creates a sound at a certain frequency:

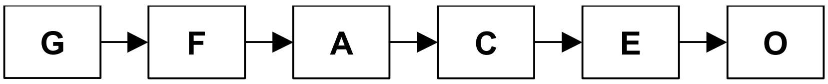
▶ To make it interesting we feed its output into a filter **F**, and send the result into an Amplifier (**A**). For now, let’s just say that this is a simple gate that passes sound when a key is pressed:

Notice how this corresponds to the vintage analog VCO→VCF→VCA patching**.**

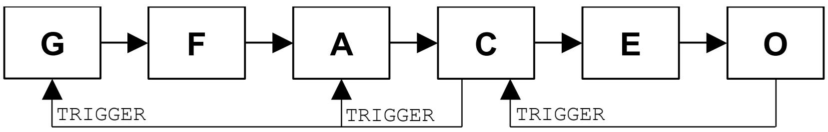
▶ We are now ready to send the results into a controller. The controller will handle all MIDI-triggering of the Generator, Filter and Amplifier as well as send a frequency control signal to these modules:



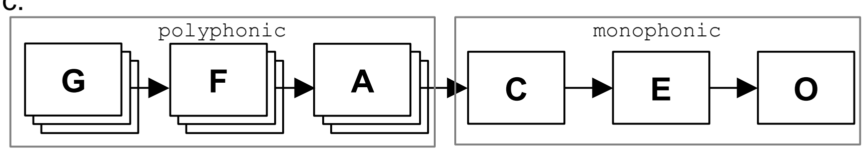
▶ Finally, we send the result into an Effect (**E**) and out to the Output module **O**:



Our simple patch template is basically done. So what would happen if we press a key? Some modules have a Trigger option that tells whether they listen and respond to MIDI events. In our schematic this is true for the Generator, the Amplifier and the Controller. MIDI events sent from the host (through output) trigger the controller, that subsequently passes these events on to the Generator and Amplifier in order to activate them:



If we set the Controller to polyphonic mode, the modules placed before the Controller in the signal chain will behave polyphonically, while the ones after will still be monophonic:

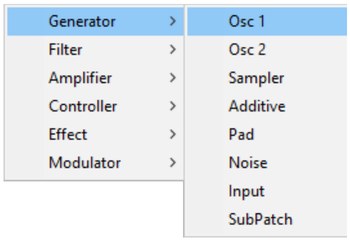


This happens automatically and ensures that only modules that really need multiple voices will get it.

Now let's implement the patch-schematic in the Synth!

▶ Startup an instance of the Synth in your host application. The default bass patch is loaded on startup. Right click on the background and select **New Patch➝Empty Patch**. This clears the workspace, leaving only an output module.

▶ Double click on the background to create a module. We start by creating a simple sawtooth oscillator: **Generator➝Osc1**.



▶ Repeat the same process to create **Filter➝SVF**, **Amplifier➝Amplifier**, **Controller➝NotePitch** and **Effect➝Delay**. Rearrange them as shown by pressing the mouse button on a modules title bars and dragging them.

▶ Set the LFO's amount to around 50 and its rate to 42 (or 0.5 Hz). This modulates the pulsewith of the squarewave over time to fatten up the sound.

▶ Now create an ADSR modulator and route that into the Amplifier's Amp knob. Turn off **Gate** mode to let the ADSR fully control the sounds amplitude. Now try to hit a key and listen to how you can shape the sound using the Attack, Decay, Sustain and Release knobs.

As you have hopefully experienced it's really easy to start making some basic patches and experiment by adding and connecting different modules. Please refer to the module guide in the back of this manual for explanations of more modules. And feel free to go crazy and experiment!

PREVIEW

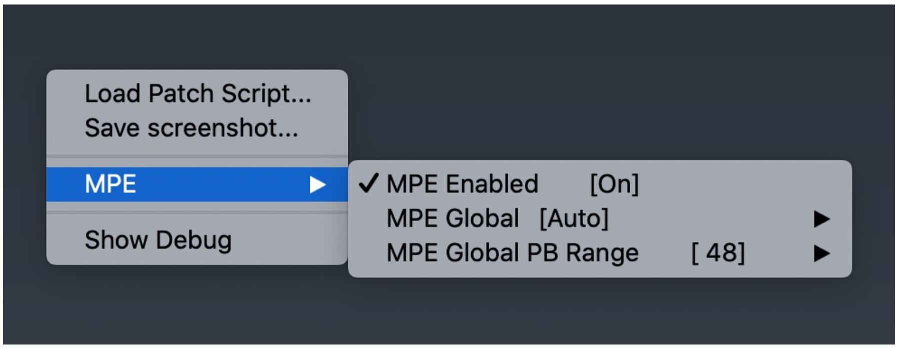
PAGE MISSING IN PREVIEW

6.7 MIDI Polyphonic Expression (MPE) Setup

MIDI Polyphonic Expression (MPE) is a MIDI mode that enables adjustment of pitchbend, and other types of expressive control, continuously for each individual note. It works by assigning each note to its own MIDI channel, so MIDI messages can be applied to each note individually. This section is a quick guide on how to setup and troubleshoot MPE inside the synth. More info and the full MPE Specification can be downloaded at: https://roli.com/mpe

1) Making sure MPE is enabled

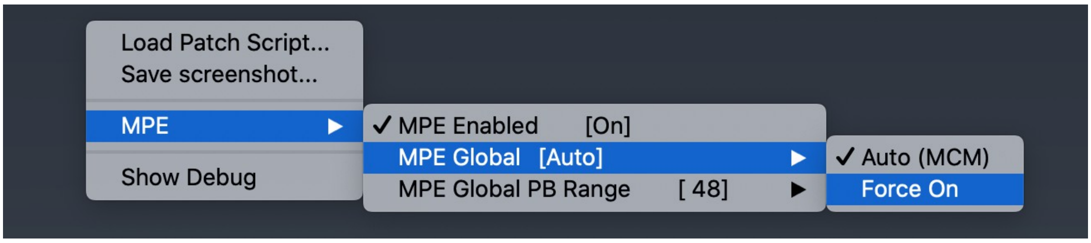
To confirm MPE is enabled inside the synth, first start up the synth. Then open the *debug menu* by holding **Ctrl** + **Shift** and **right click** on the workspace background. If MPE is enabled, there will be a **checkmark** next to the **MPE Enabled** item and the setting will indicate that it is **[On]**.



In modern hosts, MPE is automatically set-up by the host using an *MIDI MPE Configuration Message* (MCM). This is the case in, e.g., *Ableton Live* and *Bitwiq Studio*. In that case, typically, MPE will "just work" out-of-the-box.

If MPE isn't enabled, but you are using and MCM capable host, chances are that MPE is simply disabled on the plugin from within the host. Please consult your DAWs manual and check how to enable MPE on a VST/AU.

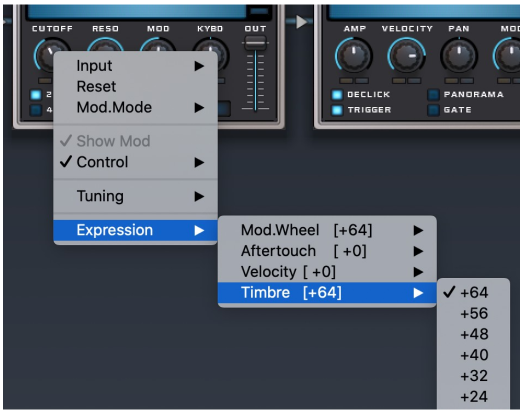
MPE will be disabled by default when running under hosts that don't support MCM. In that case, it is still possible to force enable MPE from inside the synth. Select **MPE Global** and choose **Force On** to force it on for all instances of the synth. This is not recommended, but can be useful, if, e.g, the host does not support MCM or the host does not support MPE and you are using MPE capable MIDI hardware.



3) Working with MPE Timbre / Slide and Aftertouch

Timbre (or Slide) is a new feature of MPE that makes it possible to modify the timbre of the sound on a per-note basis using **MIDI CC74**. As such, it is a form of **Expression**. By default Timbre is not hooked up in every patch, but it is quite easy to enable for typical parameters.

Right click on an Expression capable parameter, and locate the **Expression** sub-menu. A perfect parameter to start with is the **Cutoff** knob in the **Filter** module. Then, to enable Timbre Expression, set the **Timbre** value to something other than 0, e.g., **+64**. This means that Timbre is mapped +100% to the the Cutoff value.



Using Expression menu

Using MidiData module

Typical Expression-capable parameters are **Cutoff, Resonance** and **Mod** in the Filter module, **Sustain** in the ADSR module, **Rate** & **Amount** in the LFO module, and **ModIndex** in the FM module.

An alternative option that also works for parameters that don't support Expression, is to use the **MidiData** module. Create a new MidiData module, and select **Timbre** in the **Type** dropdown-menu. Then connect it to your parameter of choice, e.g., such as Cutoff.

This has the added advantage of making it possible to **Bias** and **Smooth** the incoming MIDI data, as well as enable the **Delta** option. Enabling Delta means that controller movement is only passed through when a key is held (i.e., between Note-On and Note-Off events) and only the delta relative to its value on Note-On is outputted, which is very useful for expressive MPE playing.

For working with **Aftertouch**, simply repeat the above steps, but choose Aftertouch in the Expression menu and Aftertouch in the MidiData **Type** dropdown-menu instead.

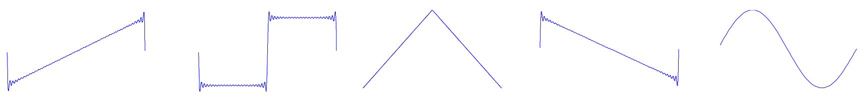
PREVIEW

PAGE MISSING IN PREVIEW

Osc1

Generator

Osc1 is a simple oscillator capable of simulating analog waveforms: Saw (sawtooth), Square, Triangle, Ramp (inverted saw) and Sine.

Parameters	
PW	**Pulsewidth** Square and Triangle are capable of pulse-width changes. For all other waveforms this knob has no effect.
PHASE	**Phase** Controls the **Phase** of the oscillators waveform (i.e., where in the waveform are we?), useful for Phase modulation. When Osc1 is triggered, one can optionally have the waveform restart its phase cycle (See Trigger). **Phase** also adjusts the start offset into this cycle. A **Phase Init Only** option is available in the knobs right click menu, that when enabled means that **Phase** only controls the **Phase Init**. A **Random Poly Phase** option is also available, adding a randomized Phase Init offset when Osc1 is triggered polyphonically. Useful for super saws and strings.
DETUNE	**Detune** Adjust the frequency up or down steplessly one semitone.
FREQ	**Frequency** Adjusts the frequency up or down continuously one or more octaves. Octave range and optional snap can be set through the right-click menu.
TRIGGER	**Trigger** When enabled the signal will phase restart on note-on events.
WAVE	**Waveform** Select the waveform to output: Saw, Square, Triangle, Ramp and Sine:  Saw Square Triangle Ramp Sine

PREVIEW

PAGE MISSING IN PREVIEW

Sampler

Generator

The Sampler module is a device capable of playing arbitrary waveforms stored as 16, 24 or 32 bit samples. One can control the playing frequency as well as start/end position and optional looping of the sample. To play a sample at the correct pitch the module must know its "key" (frequency of the fundamental harmonic). This is by default set to C2 (65.40 Hz), but can be changed to any valid semitone.

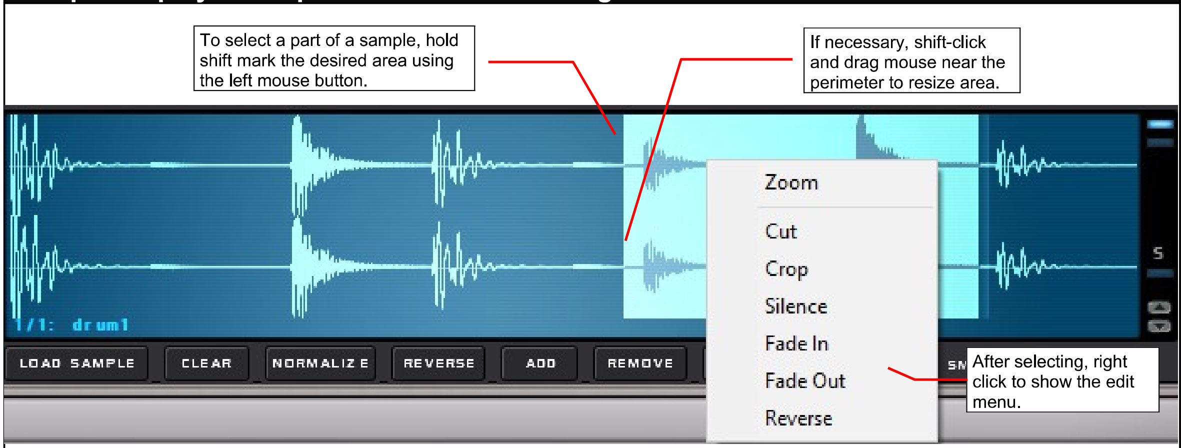
Multiple samples can be loaded into one Sampler, but only one sample can play at a time. Sample switching can be done manually; or automatically by setting key/velocity ranges for each sample (Multi mode). Two types of looping are supported: Sustained looping that is only active when a key is held (set for each individual sample); or global looping between the positions dictated by the start and length parameters. In order to avoid file and disk dependencies, all samples are stored within the patch.

Parameters	
START	**Start Position** Sets the relative start position, where the sample will start to play when triggered and loop to when looped.
LENGTH	**Length** The relative length of the sample range to play and/or loop.
DELTA	**Delta** Adjusts the relative playing speed. Similar to slowing down or speeding up a record on a turntable. A *Tempo Sync* option is available in the right-click menu. When enabled (default is disabled) the playback rate will sync to the current tempo. Useful for, e.g., drum loops.
FREQ	**Frequency** Adjusts both frequencies up or down continuously one or more octaves. Octave range and optional snap can be set through the right-click menu.
TRIGGER	**Trigger** When enabled the sample will start to play on note-on events. When off the sampler will run continuously (free running).
MULTI	**Multi** Turns on multi sampling mode. In this mode the module uses the assigned key/velocity ranges to determine which sample to trigger.
HQ	**High Quality** Turns on high quality mode, that aliases less when resampling at high and low frequencies. If off the sampler uses less CPU heavy interpolation.
LOOP	**Loop** Selects global loop type. None, Forward, Backward and Pingpong. Loops between the start and start + length positions.
POS	**Position** Instantly changes the playing position when playing a sample.
SAMPLE	**Sample** Changes the currently played sample. When only one sample is loaded or Sampler is in Multi-mode, this knob has no effect.

PREVIEW

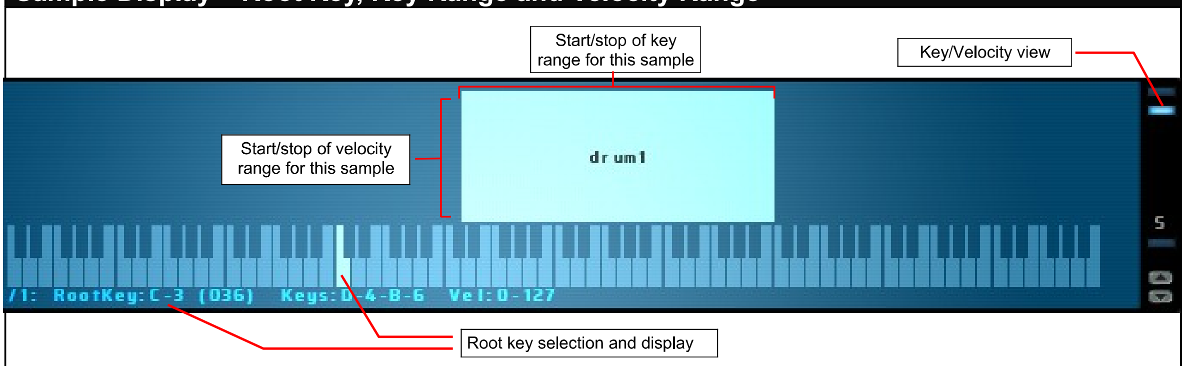
PAGE MISSING IN PREVIEW

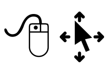
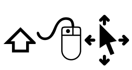
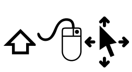
Sample Display – Sample Selection and Editing



ZOOM	**Zoom** Zooms to show only the selected part of the sample.
CUT	**Cut** Removes the selected part of the sample.
CROP	**Crop** Removes everything but the selected part of the sample.
SILENCE	**Silence** Replaces the selection with silence.
FADE IN	**Fade In** Fades the selection from zero to full volume.
FADE OUT	**Fade Out** Fades the selection from full volume to zero.
REVERSE	**Reverse** Reverses the selection so it plays backwards.

Sample Display – Root Key, Key Range and Velocity Range

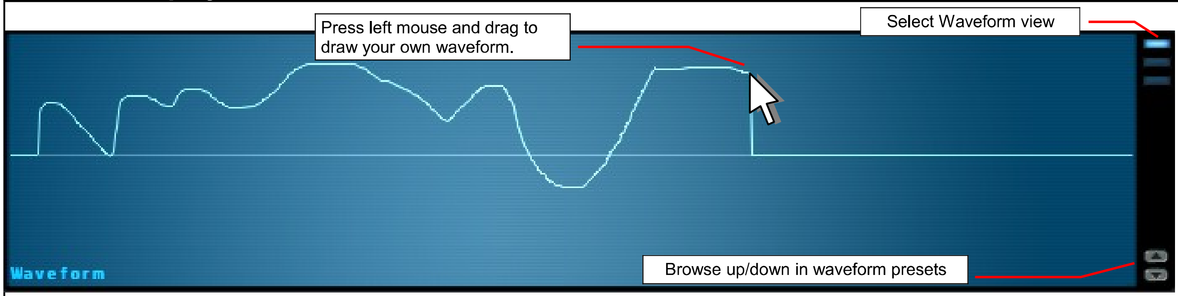


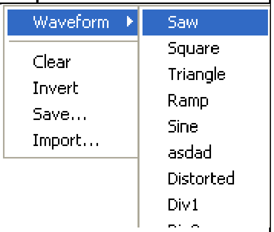
	**Left click on the drawn keyboard** Selects new root key. The root key selects the pitch at which the sample is played "as is", i.e., without any stretching or resampling. All resampling is done according to the chosen root key.
	**Shift + Left click in key/velocity field and drag** Sets start of key range and maximum velocity. In multi mode a sample will only be activated when hitting keys inside the key/velocity range. Velocity ranges cannot overlap.
	**Shift + Right click in key/velocity field and drag** Sets end of key range and minimum velocity. In multi mode a sample will only be activated when hitting keys inside the key/velocity range. Velocity ranges cannot overlap.

PREVIEW

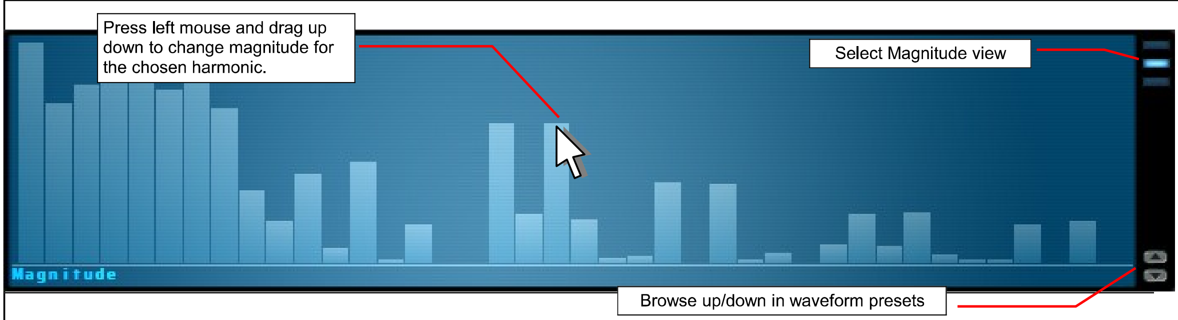
PAGE MISSING IN PREVIEW

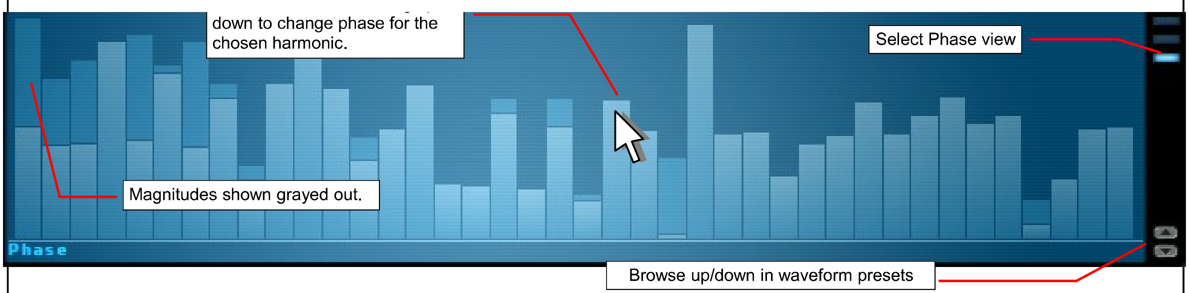
Additive Display – Waveform

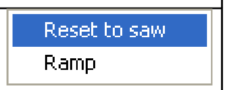


	Left click + Drag Draw your own waveform. When the mouse button is released the drawn curve is converted into the chosen number of harmonic magnitudes and phases.
	Right Click Shows the right click menu: WAVEFORM: Directly select a built-in/user waveform preset. CLEAR: Resets waveform to silence. INVERT: Flips waveform upside down. SAVE...: Save drawn waveform to disk as a user-preset. IMPORT...: Imports a .WAV file and converts it into a waveform. Alternatively drag&drop a file onto the module. 
	Mouse Wheel + Optional Drag Zoom and pan horizontally.
	Double Click Mouse Wheel Reset Zoom.

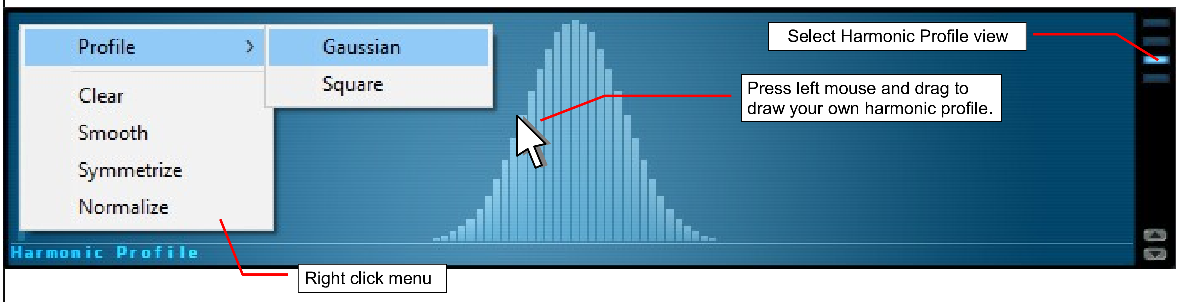
Additive Display – Harmonic Magnitude and Phase





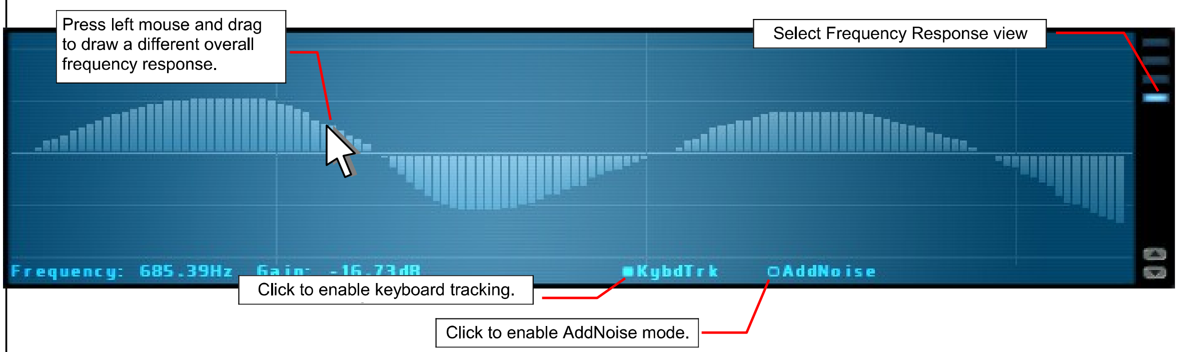
	Left click + Drag Change the magnitude or phase of one or more harmonics.
	Right Click Shows the right click menu: RESET TO SAW: Set magnitude or phases to saw profile. RAMP: Fades magnitudes or phases to zero. 

Pad Display – Harmonic Profile



	Left click + Drag Draw your own harmonic profile. When the mouse button is released the profile is applied.
	Right Click Shows the right click menu: PROFILE: Select one of the built-in harmonic profile presets. CLEAR: Resets profile to silence. SMOOTH: Lowpass filters the profile a bit to make the curve extra smooth. SYMMETRIZE: Force profile to be symmetric by mirroring its left side. NORMALIZE: Maximizes the profile so the highest peak uses all the available space.
	Mouse Wheel + Optional Drag Zoom and pan horizontally.
	Double Click Mouse Wheel Reset Zoom.

Pad Display – Frequency Response



	Left click + Drag Draw a different overall frequency response curve.
	Right Click + Drag Reset one or more frequency bands to their default (0).
	Mouse Wheel + Optional Drag Zoom and pan horizontally.
	Double Click Mouse Wheel Reset Zoom.
KYBDTRK	**Keyboard Tracking** When enabled the frequency response curve is applied in such a way that the lowest editable frequency corresponds to the lowest played frequency. When disabled the curve is applied regardless of the played key.
ADDNOISE	**Add Noise** When enabled the frequency response curve is used to add noise to the generated waveform instead of modifying its frequency content.

Granular

Generator
NEW IN V.2

The Granular module offers built-in *granular synthesis*. Granular synthesis is a unique form for sample-based synthesis that takes a sample and chops it up into small clips, called *grains*. These grains can are then played back at a defined rate, but can be altered individually with respect to frequency, and shape, and - most importantly – independently from the samples playback speed. This means that it is possible to slow down the sample, without changing its pitch, as well as changing its pitch without changing the playback speed. A whole range of interesting possibilities arise when each grains frequency, position, size and panning parameters are also modulated, resulting in a completely different sounds than the source sample.

Parameters	
SIZE	**Grain Size** Sets the size of each individual grain.
RATE	**Grain Rate** Controls the grain spawn rate.
SHAPE	**Grain Shape** Controls the shape of each grain, from rectangular (0) to triangular (127). For overlapping grains, triangular will give most seamless result.
SPEED	**Speed** Controls the playback speed of the sample in Auto mode
TRIGGER	**Trigger** When enabled the sample will restart on note-on events. When off the sample will play continuously (free running).
MULTI	**Multi** Turns on multi sampling mode. In this mode the module uses the assigned key/velocity ranges to determine which sample to play from.
SYNC	**Tempo Sync** Syncs the rate frequency to the host tempo.
HQ	**High Quality** Turns on high quality antialiasing.
SAMPLE	**Sample Index** Choose playback sample index in non-multi mode.
MODE	**Playback Mode** [Auto/Manual] In **Auto** mode the sample position is advanced automatically, based on the speed parameter. In **Manual** mode the playback position is completely controlled by the position knob, and the speed parameter has no effect.
POS	**Grain Position** Controls the grain playback position within the sample.
FREQ	**Grain Frequency** Controls the grain playback frequency.
PAN	**Pan** Controls stereo panning of each grain in stereo mode.
STEREO	**Stereo** Controls stereo spread by offsetting the left and right channels in stereo mode.
DIFFUSION	**Diffusion** Enables diffusion (low amplitude random modulation) for **Pos**, **Freq**, **Size**, and **Pan**.

Noise

Generator

The Noise module generates Pink, White or Brown noise. Noise is especially useful for synthesis of ocean waves and wind, for percussion or simply to add some extra "bite" or analogueness to leads and basses. By definition noise is (digitally speaking) a series of random numbers containing almost all frequencies. The different flavors determine how the noise rolls-off over the frequency spectrum. White noise is the purest (and harshest) type of noise without any filtering. Pink noise rolls off linearly towards the high end at 3dB/Octave. Brown noise rolls-off at 6dB/Octave. One can optionally enable cycle mode to force the generated noise to repeat at an interval, thus forming a waveform similar to an oscillator, but randomized.

Parameters	
AMP	**Amplifier** Sets the overall volume.
SEED	**Seed** To control the randomness, this knob can be used to set the initial seed sent to the (pseudo) random algorithm.
LOWPASS	**Lowpass** Controls one pole (12dB/Octave) lowpass filtering of the noise. When set to 127, all frequencies pass.
HIGHPASS	**Highpass** Controls one pole (12dB/Octave) highpass filtering of the noise. When set to 0, all frequencies pass.
TRIGGER	**Trigger** When enabled the signal will phase restart on note-on events.
CYCLE	**Cycle** Enables cycle mode, where the generated waveform is reset at an interval corresponding to pitch set by the controller.
NOISE TYPE	**Noise Type** Choose between: **White:** Clean noise without filtering (even distribution of energy). **Pink:** Filtered white noise with 3dB roll off per octave. **Brown:** Filtered white noise with 6dB roll off per octave.

PREVIEW

PAGE MISSING IN PREVIEW

PREVIEW

PAGE MISSING IN PREVIEW

PREVIEW

PAGE MISSING IN PREVIEW

EQ3

Filter

EQ3 is a basic 3 band equalizer with adjustable crossover frequencies, and 2 and 4 Pole modes. Like any equalizer it can be used to to boost or attenuate certain frequencies in order to shape a sounds timbre.

Parameters	
LOW	**Low Gain** Boost or cut the Low-frequency range. Turning the knob all the way up boosts by +6dB. Turning it all the way down removes that range completely (-∞dB).
MID	**Mid Gain** Boost or cut the Mid-frequency range. Turning the knob all the way up boosts by +6dB. Turning it all the way down removes that range completely (-∞dB).
HI	**Hi Gain** Boost or cut the High-frequency range. Turning the knob all the way up boosts by +6dB. Turning it all the way down removes that range completely (-∞dB).
AMOUNT	**Amount** Controls the mixed amount of EQ. All the way up means 100% (Full EQ), while center means 50% EQ and 0 means no EQ (Bypass).
2/4 POLE	**2/4 Pole** Switch between 2 Pole (12dB per octave) and 4 Pole (24dB per octave) filters.
FREQ1	**Freq1** (Knob appears when the *FREQS* LED is selected) Controls the crossover frequency (100Hz to 2kHz) between Low and Mid Bands.
FREQ2	**Freq2** (Knob appears when the *FREQS* LED is selected) Controls the crossover frequency (2kHz to 16kHz) between Mid and Hi Bands.

PREVIEW

PAGE MISSING IN PREVIEW

Shelving

Filter

Shelving is a simple filter module containing two 2 pole shelving filters: Lowpass (High shelf) and Highpass (Low shelf) for cutting the high end and the low end of a signal respectively.

Parameters	
LP	**Lowpass Frequency** Sets the lowpass frequency, where the high cut should start.
LP GAIN	**Lowpass Gain** Sets how much of the lowpass signal to blend with the input.
HP	**Highpass Frequency** Sets the highpass frequency, where the low cut should start.
HP GAIN	**Highpass Gain** Sets how much of the highpass signal to blend with the input.

PREVIEW

PAGE MISSING IN PREVIEW

PREVIEW

PAGE MISSING IN PREVIEW

Mixer

Amplifier

Mixer is a special module for mixing two (and only two) incoming signals. By default the synth automatically mixes all signals sent to a module, but in this case you can control the type of mixing to use.

Parameters	
MIX	**Mix** Changes the mix ratio between the incoming signals. Centered equals standard 50/50 mixing ratio.
AMP	**Amplification** Changes the volume of the incoming signal. Centered means no change in volume. A higher setting boosts, while a lower setting reduces the volume.
PAN	**Pan** When in stereo mode this knob pans the signal left or right.
MOD	**Mix Modulation Amount** Controls how much modulation to apply to the Mix knob. If Mix isn't modulated this knob has no effect.
TYPE	**Mix Type** Chooses the Mixing type to use to combine the two signals: **Add**: Linearly Add the two signals. This standard mixing is the default. **Sub**: Subtract the two signals. **Ringmod**: Multiply the two signals. **2xMono**: Take two mono signals and put them in the left and right channel. **Xor**: Perform a per-sample binary Xor operation on the two signals. **Greater**: Perform per-sample comparison the two signals, and choose the largest.

PREVIEW

PAGE MISSING IN PREVIEW

NotePitch

Controller

The NotePitch controller is an essential and useful module for controlling generator pitch, note triggering and polyphony. Moreover it has simple transpose, finetune, and portamento (glide) functionality. A controller's job is to construct a frequency signal that is sent to all modules feeding their output to that controller. The knobs on the NotePitch module affect this internal control signal.

Parameters	
NOTE	**Note** Transposes the internal signal whole semitones up or down. Base-note is shown in the display's text area. The Right click menu offers a **Tuning** option to detune the root key of the scale (default is A=440Hz), and a **Note-Min/Max** option that allows for setting the range of notes that the module responds to. Default is full MIDI range (C-0 to G-10).
OCTAVE	**Octave** Transposes the internal signal whole octaves up or down. Base-note is shown in the display's text area.
TUNE	**Tune** Finetunes the internal signal steplessly one or more semitone up or down. Choose between 2, 12 or 24 semitone range by right clicking on the knob. The right click menu also offers an *Enable Pitchbend* option (on by default). When on the module will handle MIDI pitchbend events automatically using the chosen semitone range. Finally, the right click menu offers an *MPE PB Range*: The MIDI Polyphonic Expression Pitchbend Range controls the amount of Pitchbend for MPE controlled notes when MPE is enabled by the host (±48 semitones is default).
PORTA	**Portamento/Glide** Changes how fast the internal frequency signal changes. Turned full left means no portamento (or glide). The right click menu offers range scales and two portamento modes: *Fixed* or *Auto*: In Fixed mode, glide always takes place. In Auto mode (default), only overlapping notes will glide.
TRIGGER	**Trigger** When ON the module listens and reacts to note-on MIDI events. It changes the frequency of the internal signal based on the currently played MIDI note.
MODE	**Mode** Chooses Polyphony mode. Choose between: **Mono**: 1 voice only. **Poly** 2-16: Polyphonic with 2 to 16 voices. **Legato**: 1 voice, without re-trigger as long as at least one key is held.

PREVIEW

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FM Modes	
FMLIN	**Linear FM** Linear FM is a classic synthesis technique where the frequency is swept up and down by a frequency deviation defined by the Modulation Index. Regardless of modulation source, Linear FM always sweeps up the same amount of Hz as it sweeps down. This keeps the output in-tune. Yet, because of its linear nature, large frequency deviations can make the frequency cross 0Hz. This calls for so called *Through-Zero* oscillators that support negative frequencies which output the same signal but in reverse phase. All generators in the synth support Through-Zero, so the module has a **TZ** option to turn Through-Zero and negative frequencies off. With TZ enabled, Linear FM produces output almost identical to Phase Modulation.
FMEXP	**Exponential FM** Exponential FM is a form of frequency modulation where the modulation range follows the musical spacing of notes and octaves. For example, going up one octave means doubling the frequency, and going down one octave means halving the frequency. While this is asymmetrical compared to Linear FM, Exponential FM is in a sense simpler, since this control over tuning frequency is available in most analog synthesizers. At low rates, Exponential FM is simply vibrato, and sounds like Linear FM, while it produces more overtones at high rates. Standard Exponential FM does not have Through-Zero behavior, but instead stops at 0Hz, as we cannot halve a frequency to cross the 0Hz line. However, due to its asymmetrical nature, Exponential FM produces a DC offset that detunes the resulting sound. Therefore, the module features an Exponential Sync (EXS) option to correct for this waveform-dependent detuning (requires Through-Zero (TZ)). DC Block (DC) is recommended for other modulation sources.
PM	**Phase Modulation** Phase Modulation is the original classic FM technique, as invented by John Chowning. Frequency and Phase Modulation are in fact closely related, as modulating a signals phase will also modulate its frequency, simply because frequency is the derivative of phase. PM it therefore also known as *Chowning-style FM* or *indirect FM*, and produces results almost identical to Linear FM, without the need for Through-Zero (TZ), since it only modulates the phase.
MOD LIN/EXP/PM	**Modulation Linear/Exponential/PM** These modes do not actually perform FM/PM, but instead outputs the generated internal modulation signal directly for all modes.

Advanced Parameters	
EXTMOD	**External Mod** Controls the external modulation directly. Modulate this knob for FM/PM modulation with an external signal.
EXTMIX	**External Mix** Controls the mix ratio between the external modulation and the internal.
POLAR	**(Uni-)Polarity** Linearly alters the modulation signal from bipolar to positive unipolar.
FEEDBACK	**Feedback** Sets the amount of feedback. Both FM and PM feedback (PMF) is supported.
TZ	**Through-Zero** Enables Through-Zero FM , meaning that negative frequencies may be generated (default). When OFF, frequencies are clamped to 0Hz. Does not affect PM.
INI	**Phase Init** Enables phase init of the modulation signal, on note trigger. Default is off.
DC	**DC Block** Enables a DC blocking filter on the internal modulation signal.
PMF	**Phase Modulation Feedback** When ON, feedback is PM, when off, FM (default).
EXS	**Exponential FM Sync** When ON, corrects for the detuning of Exponential FM. May require TZ.

PREVIEW

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PREVIEW

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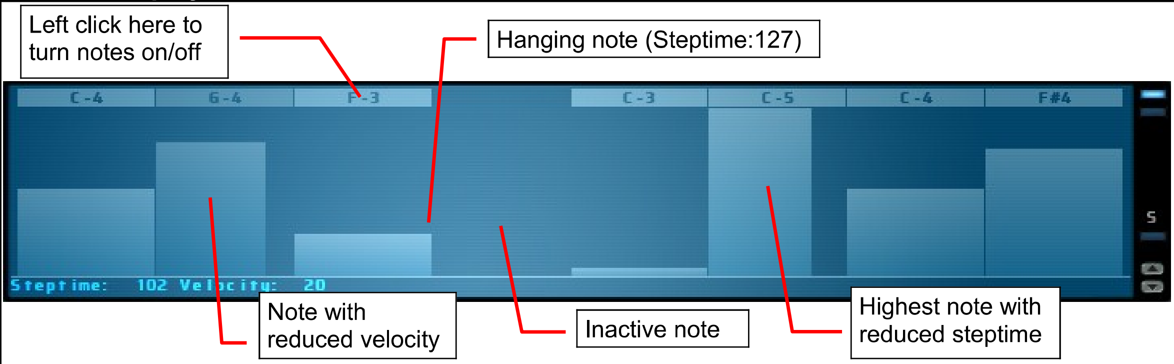
Pattern Display



	Left Click on Note Handles Turn notes on or off.
	Left Click on Display Area Adjusts the individual note levels.
	Shift Left Click + Drag Left/Right or Up/Down Left/Right changes Steptime of individual notes. A Steptime of 127 leaves the note sustained, meaning that it does not send note-off event until a new note begins. Up/Down changes Velocity level of individual notes.
	Right Click Shows Pattern Preset Menu.
	Mouse Wheel + Optional Drag Zoom and pan horizontally.
	Shift + Mouse Wheel + Optional Drag Zoom and pan vertically.
	Double Click Mouse Wheel Reset Zoom.

Pattern Buttons

RESET	**Reset** Resets all notes to their default state (note, velocity, and steptime).
RANDOM	**Random** Sets notes to random values.
LEFT	**Left** Shift note pattern left.
RIGHT	**Right** Shift note pattern right.
MIRROR	**Mirror** Flips note pattern horizontally, so the sequence runs backwards.
FLIP	**Flip** Flips note levels (vertically).
SHUFFLE	**Shuffle** Mixes the current notes in a random order without affecting their amplitude.
FILL	**Fill** Copies all the used notes to all the unused steps as a repeated pattern.
TAP	**Tap** Awaits MIDI input, and records each pressed key's note and velocity in the pattern, one step at a time. Tap-recording automatically ends when the pattern is full. To cancel tap-recording, de-click the tap button again.
COPY	**Copy** Copies the current pattern into a copy-buffer. Useful for duplicating patterns.
PASTE	**Paste** Pastes the content of the copy-buffer, overwriting the current pattern.
RECORD	**Record** Similar to Tap, but records incoming MIDI note and velocity with timing data.
TEMPO	**Tempo** Selects the active tempo with respect to the current song tempo (BPM)

PREVIEW

PAGE MISSING IN PREVIEW

PREVIEW

PAGE MISSING IN PREVIEW

Delay

Effect

The Delay module is a tempo controlled stereo delay, featuring two independent delay lines. The delay time can be set separately for the left and right channels using the Stereo knob. A coarse delay parameter that sets the delay time in eights of a beats (0-16 beats/8) and a fine delay parameter that adjusts the delay steplessly. How much of the delayed signal that is sent back to the delay is controlled by the Feedback knob, and can optionally be low/high pass-filtered. A Tap parameter controls the volume of the first delay tap for both channels. A built-in adjustable cross delay feedback mixer makes it possible to achieve the popular stereo effect known as *ping pong*, where a part of the feedback from the left channel is sent to the right channels input and vice versa.

Parameters	
DELAY	**Delay** (Beats) Sets the delay in beat, snapping to one eighth (1/8) of a beat.
FINE	**Fine** (ms) Finetunes the delay steplessly up or down one beat. A **Range** option allows the fine range to be tuned to beat-fractions of 1/8 (default), 1/4, 1/2 and 1.
FBACK	**Feedback** Sets the amount of feedback, i.e., how much of the already delayed signal to send back into the delay.
STEREO	**Stereo** Offsets the delay in the right channel by plus/minus one beat compared to the left.
TAP	**Tap** Controls the volume of the input signal to feed to the delay.
PINGPONG	**Pingpong** When in stereo, the feedback can work in two modes. Either the signal is sent back to the sample channel (i.e., left to left and right to right) or alternatively to the other channel (i.e., left to right and right to left). When the channel delays are offset the latter can give an interesting "ping pong" effect between the left and right speaker. This knob controls how much ping pong you want. A setting of 0 means no ping pong. 127 means full ping pong, and centered means 50% of both.
LP	**Lowpass** (Hz) Lowpass cutoff knob for filtering the feedback. At 127 all frequencies pass.
HP	**Highpass** (Hz) Highpass cutoff knob for filtering the feedback. At 0 all frequencies pass.
DRYOUT	**Dry Out** Enables dry output, meaning that the original signal is mixed with the delayed signal. When disabled only the delayed signal is output.
FILTER	**Filter** Enables feedback filtering. This has to be enabled for LP and HP to work.
ANTIPHASE	**Antiphase** When ON, the fine delay is added in reverse phase to left and right channels.

Distortion

Effect

The Distortion module does Waveshaping, Resampling and Bit Reduction. Waveshaping changes the shape of the incoming waveform using a nonlinear transfer function. Resampling is a digital effect that takes an incoming signal and picks out samples at a specified interval in order to simulate an (in this case) lower sampling rate. Similarly, bit reduction (bit crushing), squashes the incoming samples dynamic range to simulate less bit precision.

Parameters	
AMOUNT	**Amount** Sets how much of the distorted signal you wish to hear by adjusting the mix ratio between the incoming- and the distorted-signal. 127 means full distortion.
DRIVE	**Drive** Controls the internal variable(s) in the waveshaping algorithm. Cranking it up yields a more distorted sound.
BITS	**Bits** Lowering this reduces the bit precision. When set to max no reduction is performed. The right-click menu offers a **Stereo** option that makes the Bit reduction run in antiphase (off by default).
RATE	**Rate** Controls the sample rate conversion from (resampling). When set to max no resampling is performed. The right-click menu offers a **Stereo** option that makes the Rate reduction run in antiphase (off by default).
HQ	**High Quality** Toggles oversampling to reduce high frequency aliasing.
TYPE	**Distortion Type** Chooses the Waveshaping algorithm: Soft, Hard, Variable, Soft-fuzz, Hard-fuzz, Soft-tube, Hard-tube, Warmer.

PREVIEW

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Chorus / Flange

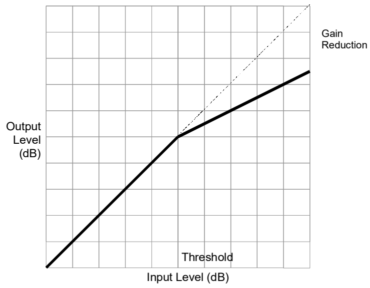
Effect

Chorus is a stereo effect for simulating several simultaneous voices. It does this by introducing a delay (up to 200ms) and varying this delay over time using a modulator. The modulation results in a slightly detuned signal that combined with the input signal sounds like an extra voice. Filtered feedback can be used to intensify this effect. If the delay is less than around 20ms, then the effect is called *flange*: A comb filter effect, where notches appear in the frequency spectrum. Varying the time delay causes these to sweep up and down the frequency spectrum, resulting in a swooping sound.

Parameters	
LDELAY	**Left Delay** Sets the relative delay for the left channel.
RDELAY	**Right Delay** Sets the relative delay for the right channel.
AMOUNT	**Amount** Adjusts the mix ratio between the input and the chorus-signal. Right-click-menu offers a **Constant Power** option (default on), that ensures roughly equal volume regardless of Type setting.
FEEDBACK	**Feedback** Controls how much of the output signal to send back into the chorus.
DEPTH	**Depth** Controls the overall depth of the delay.
SPREAD	**Spread** Offsets the delays and phases between the different chorus stages.
RATE	**LFO Rate (0-100Hz)** Sets the internal LFO rate in Hz.
AMP	**LFO Amp** Set the internal LFO amplitude.
XFEED	**Stereo Crossfeed** Sets the amount of feedback crossfeed from left to right, and right to left channel.
WIDTH	**Stereo Width** Expands or contracts the stereo width of the Chorus signal.
LP	**Lowpass** Lowpass cutoff knob for filtering the feedback. At 127 all frequencies pass.
HP	**Highpass** Highpass cutoff knob for filtering the feedback. At 0 all frequencies pass.
SERIES	**Series** Run the chorus stages in series, or in parallel (default).

Compressor

Effect



The Compressor module is a soft-knee compressor with Peak/RMS detection and optional sidechain. This may sound intimidating, but a compressor is really quite simply a time varying amplifier that makes a signal appear louder by changing the output volume based on the input volume. By using an envelope follower it reduces the gain of an incoming signal if its amplitude exceeds a certain threshold, where the amount of gain reduction is determined by a ratio control. Typically the result is then amplified again (*makeup gain'ed*) in order to fully utilize the available dynamic range. The final result thus appears louder since the low amplitude part is expanded while the top part is compressed.

Parameters	
ATTACK	**Attack** (2-500ms) The attack time for the envelope follower used to determine the amplitude level.
RELEASE	**Release** (2-1000ms) The release time for the envelope follower used to determine the amplitude level.
RATIO	**Ratio** (1.05:1 – 20:1) Controls how much compression should be applied when the signal level exceeds the threshold. A gain reduction ratio of N:1 means that a level increase in the input signal of N dB results in level increase in the output signal of 1 dB.
THRESHOLD	**Threshold Level** (dB) The threshold is a value in dB where the compressor will kick in. The signal part below the threshold will not be compressed.
KNEE	**Knee** (dB) Instead of the kinked compression curve shown above (hard knee), the module can soften the curve (soft knee). *Knee* sets the distance from the threshold value where the knee curve starts. Lowest value means hard knee compression. The higher the value the larger and softer the knee curve will be.
GAIN	**(Makeup) Gain** (dB) The make up gain raises the entire signal after compression to make use of the extra headroom / dynamic range given by the compression.
SIDECHAIN	**Sidechain** Inputs another signal or offset to the envelope detector, meaning that you can controls the gain reduction of one signal from the amplitude of another signal (the sidechain). When centered the sidechain signals equals zero.
MIX	**Mix** Controls how to mix the sidechain with the module's input. A setting of 0, means 100% input. A setting of 127 means 100% sidechain.
PEAK / RMS	**Peak / RMS (Root Mean Square)** Selects whether the envelope follower should be based on Peak or RMS detection. *Peak* is usually best suited for heavy, fast compression (percussion), while *RMS* is good for subtle, musical compression.
MONITOR	**Monitor** Turns ON gain reduction monitoring for this module instead of the standard waveform and frequency display. Useful for verifying that the module actually does compress the sound

MultiComp

Effect
NEW IN V.2

The MultiComp module is a multiband soft-knee compressor width adjustable crossover frequencies and Peak/RMS detection. You can think of this module as three instances of the Compressor module, each operating on a single band of a three band EQ with Low, Medium, and High outputs. For a smooth overall frequency and phase response, the band separation is done using State Variable Filters with a Linkwitz-Riley crossover design. Each compressor have their own separate options, and each band can either run through its dedicated compressor (default), be bypassed (no compression) or muted. The attenuation in dB is shown graphically for each band. Each band can also be temporarily solo'ed. For stereo input, the left and right channel-compression can either be Linked or run in stereo (unlinked).

Parameters	
LOW	**Low** (-12dB to +12dB) Boost or cut the Low-band frequency range.
MID	**Mid** (-12dB to +12dB) Boost or cut the Mid-band frequency range.
HIGH	**High** (-12dB to +12dB) Boost or cut the High-band frequency range.
AMOUNT	**Amount** Sets the mix ratio between the unprocessed (dry) and the compressed signal (wet). Useful for parallel compression. A setting of 127 means full compression.
PEAK / RMS	**Peak / RMS (Root Mean Square)** Selects whether the envelope follower should be based on Peak or RMS detection. *Peak* is usually best suited for heavy, fast compression (percussion), while *RMS* is good for subtle, musical compression.
LINK	**Link L/R** Stereo link left and right channels. When on, the compression in both channels is exactly the same, and based on the maximum of the Left and Right channels. When off the compression is done separately for each channel.
SOLO	**Solo** Temporarily solo the currently selected band. The Solo LED will blink to indicate that the current compressor/band is soloed. Click again to un-solo.
BAND	**Band** Switch between Low, Mid and High band selection. Since each band has separate compression parameters, this alters the compression knobs. The selection can also be done from the band display.
FREQ1	**Freq1** (50Hz to 550Hz) Sets the crossover frequency between Low and Mid bands.
FREQ2	**Freq2** (600Hz to 5kHz) Sets the crossover frequency between the Mid and High bands.
GAIN	**Final Gain** (-12dB to +12dB) Controls the final gain compensation after compression but before dry/wet mixing.

BitShuffle

Effect

The BitShuffle module is a digital distortion effect, inspired by a common hardware circuit-bending technique. It reduces an incoming signal to between 1 and 16 bits, and allows you to shuffle and choose from the available bits to reconstruct the output. The module also does simple dithering (1 bit noise) to reduce bit reduction artifacts. It can be used to do distortion ranging from very subtle (many bits, little shuffle) to extremely hard distortion (few bits, lots of shuffle).

Parameters	
BITS	**Bits (1-16)** Choose the number of bits for the internal digital representation. 1 bit being the lowest quality (sound can only be on or off) to 16 bits (CD quality).
AMOUNT	**Amount** Sets how much of the bitshuffled signal you wish to hear. 0 means no effect. 127 means full bitshuffle.
GAIN	**Gain** Since bit-shuffling can affect the overall volume level, this parameter enables you to change the volume up or down after bitshuffle is applied.
DITHER	**Dither** Controls the amount of dither to add to the signal. Dither is simply one bit noise added to the signal before the bit reduction quantization. 0 means no dither, 127 means full dither.

BitShuffle Buttons	
	
RESET	**Reset** Resets all bits to their default ordering (decreasing).
RANDOM	**Shuffle** Shuffles the selected bits in a random order.
LEFT	**Left** Shift/Rotate bits left.
RIGHT	**Right** Shift/Rotate bits right.
MIRROR	**Mirror** Flips bit-pattern in reverse order.
RANDOM	**Random** Set bit values to random numbers (between 1 and the selected number of bits)

PanSpread

Effect

The PanSpread module is a stereo expander with individual low and high spread, plus crossover frequency adjustment. It is useful for widening the stereo image of a stereo input signal. The module has a built-in filter that separates low frequencies from high, as you normally want different stereo responses at low and high frequencies. The Low and Hi parameters offers extreme widening up to 400%. Yet, care should be taken not to over-do stereo widening, as this can lead to stereo phase cancellations in the final mix. If the input signal is mono, the module has no effect. If you want to introduce stereo to a mono-only signal, one trick is instead to use a Delay module with a fine delays of <1ms to offset the left and right channels slightly.

Parameters	
AMOUNT	**Amount** Sets how much of the panspread signal you wish to output by adjusting the mix ratio between the incoming- and the outgoing-signal. 127 means full panspread.
FREQ	**Frequency** (20Hz-20kHz) Sets the crossover frequency of the internal lowpass and highpass filter.
LOW	**Low** (0%-400%) Widens the Low frequency signal by 0% (mono) to 400%. Center means 100% (no change).
HI	**High** (0%-400%) Widens the High frequency signal by 0% (mono) to 400%. Center means 100% (no change).
MODE	**Mode** Choose between modes: **Mix** (default), **Low Only** (only output low part of the filtered signal) and **Hi Only** (only output high part of the filtered signal).

PREVIEW

PAGE MISSING IN PREVIEW

PREVIEW

PAGE MISSING IN PREVIEW

SoftClip

Effect
NEW IN V.2

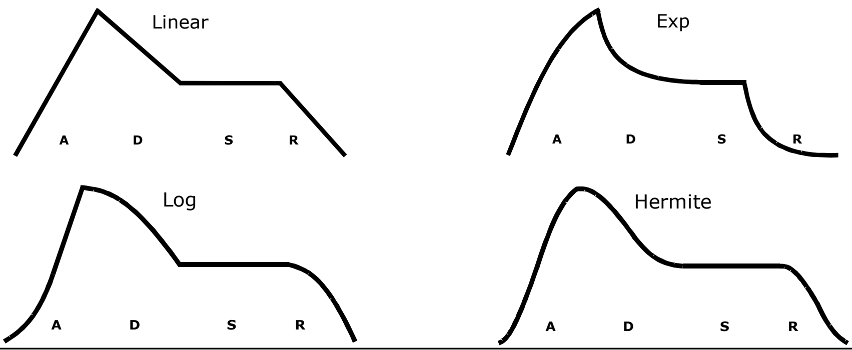
The SoftClip module is a soft clipping distortion effect. It works by soft clipping the peak parts of the amplitude signal above a desired threshold. The *Soft* knob can be used to control the level of softness, ranging from hard clipping to -6dB below-threshold soft-clipping. The module can automatically account for the loss in gain based on the threshold setting, and can optionally add digital distortion to the soft clipped signal, featuring adjustable stereo bit-reduction and noise.

Parameters	
THRES	**Threshold** Sets the threshold level in dB for clipping to occur.
SOFT	**Soft** Sets the amount of soft clipping, ranging from 0dB (hard clipping) to -6dB (soft-clipping).
DRIVE	**Drive** (-20dB to +6dB) Optionally scales the input signal before clipping.
AMOUNT	**Amount** Adjusts the mix ratio between the incoming- and the soft-clipped-signal. A value of 0 means no-clipping,while a value of 127, means 100% soft-clipping.
DIST	**Distortion Amount** Sets the amount of digital distortion to add to the soft-clipped signal.
BITS	**Bits** Adjusts the digital bit-reduction of the clipped signal. When set to max no reduction is performed. The right-click menu offers an **Antiphase** option that makes the Bit reduction run in stereo / antiphase (on by default).
BIAS	**Bias** Controls the negative and positive stereo-offset bias of the Bit reduction.
NOISE	**Noise** Amount of bit-scaled noise to add after bit reduction.
AUTOGAIN	**Auto Gain** When enabled, automatically sets the gain based on the clip threshold level.
DIST	**Digital Distortion** This enables digital-distortion (bit reduction) on the soft-clipped part of the signal, controlled by the Dist, Bits, Bias and Noise parameters.
HQ	**High Quality** Enables high quality oversampling.
DIFF	**Difference** Outputs the difference between the clipped an unclipped signal.

ADSR

Modulator

ADSR module is a four-stage envelope consisting of *Attack, Decay, Sustain* and *Release*. The ADSR module is stereo-capable.

Parameters	
A	**Attack** (0-10000ms) Attack is the time taken for the initial level to go from from 0 to 100%.The module defaults to a forced reset to 0 on each retrigger. An alternative **Analog** attack option is available through the right click menu, where a retrigger instead rises from the currently reached envelope level. Attack also offers **Declick Ramp** option that controls the declick ramp timing. Both **A**, **D**, **S** and **R** have **Range** options of: 10ms, 50ms, 100ms, 200ms, 500ms, 1000ms, 2000ms, 5000ms, 10000ms (default).
D	**Decay** (0-10000ms) Decay sets the transition time from 100% until reaching the *Sustain* level.
S	**Sustain** (dB) Sets the constant amplitude that is produced when a key is held. Note that when *Sustain* is set to maximum, the amplitude cannot drop during the *Decay* stage. The right click menu offers Expression option.
R	**Release** (0-10000ms) Sets the time for the sound to fade from *Sustain* level to 0 when key is released.
BIPOLAR	**Bipolar** When ON the module produces a signal in the amplitude range [–1;1]. When OFF it only uses the amplitude range [-1;0] (Unipolar).
CLK	**Click** Clicks are spikes in the envelope caused by quick changes in amplitude, e.g., when receiving many triggers in a sequence. When ON the module doesn't handle clicks in any special way, it just let's them pass (useful for percussion). When OFF the module ramps off clicks (default). See **Attack** knob for additional settings.
TRIGGER	**Trigger** When ON the module restarts the envelope on MIDI note-on events.
INV	**Invert** Inverts (or flips) the ADSR signal. Useful for say outputting in the [0;1] range in unipolar mode, or to simply generate an inverted ADSR envelope.
CURVE	**Curve** Selects the active envelope curve. Choose between **Linear**, **Exp** (Exponential), **Log** (Logarithmic) or **Hermite** (Smooth cubic curve): 

PREVIEW

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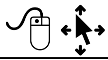
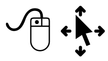
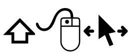
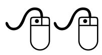
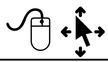
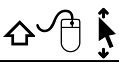
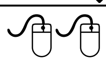
Step Display

	**Left Click on Display Area** Adjusts the individual amplitudes of the steps.
	**Left Click on Control Point + Drag** Moves Control Point (step start) to new amplitude and a potential off beat position.
	**Shift + Left Click on Control Point + Drag** Moves a step's Control Point (step start) without changing its amplitude.
	**Double Click on Control Point** Forces selected step to start on a beat position-beat.
	**Mouse Wheel + Optional Drag** Zoom and pan horizontally.
	**Shift + Mouse Wheel + Optional Drag** Zoom and pan vertically.
	**Double Click Mouse Wheel** Reset Zoom.
	**Right Click** Shows Step Preset Menu.

Step Buttons

RESET	**Reset** Resets all steps to their standard amplitude and position.
RANDOM	**Random** Sets random amplitudes for all steps.
LEFT	**Left** Shift step pattern left.
RIGHT	**Right** Shift step pattern right.
MIRROR	**Mirror** Flips step pattern horizontally, so the sequence runs backwards.
FLIP	**Flip** Flips amplitude levels (vertically).
SHUFFLE	**Shuffle** Mixes the current steps in a different order without affecting their amplitude.
FILL	**Fill** Copies all the used steps to all the unused steps as a repeated pattern. Useful when, e.g., going from 8 steps to say 16 steps.
TEMPO	**Tempo** Selects the active tempo with respect to the current song tempo (BPM).

PREVIEW

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PREVIEW

PAGE MISSING IN PREVIEW

PREVIEW

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MidiData

Modulator

MidiData is a modulation module for receiving general MIDI data like Pitchbend, Aftertouch, and data from arbitrary MIDI controllers. You can also use the module in Manual mode to simply "draw" you own modulation using the value knob.

Parameters	
VALUE	**Value** Sets the value of the modulation signal in *Manual* mode. In any other mode, this knob has no effect.
CONTROL	**Control** Selects the MIDI controller id when in *Controller* mode. In any other mode, this knob has no effect.
BIAS	**Bias** Offsets the signal [-1:+1] in bipolar mode and [-0.5:+0.5] in unipolar mode. Default value is 0, no bias.
SMOOTH	**Smooth** Smooths the signal using a first order lowpass filter. 0 means no smoothing/lowpass filtering (default).
MIDILEARN	**Midi Learn** When enabled the module will wait for any MID controller to change and assign the *Control* knob to that controller. Typically, you turn on *MidiLearn*, change a MIDI controller on your MIDI device, and then turn *MidiLearn* back off again. You can now use you MIDI device as modulation source.
DELTA	**Delta** When OFF (default) controller movement is simply passed through. When ON controller movement is only passed through when a key is held (i.e., between note-on and note-off events) and only the *delta* relative to its value on note-on is outputted.
BIPOLAR	**Bipolar** When ON the module produces a signal in the amplitude range [–1;1]. When OFF it only uses the amplitude range [-1;0] (Unipolar).
INV	**Invert** Inverts (or flips) the signal.
TYPE	**Type** Choose between: *Pitchbend, Controller, Aftertouch, Timbre* and *Manual* mode.

PREVIEW

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PREVIEW

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Control

Modulator
NEW IN V.2

Control is a modulator module that exposes the control signals that the synth uses internally. These signals travel backwards through the signal chain, starting at the output module and/or get altered or (re)created by any of the Controller modules that they pass through. Having the signals exposed in a separate modulator module makes it possible to route these signals into any parameter. The module can also be used for a quick inspection of control signals at a specific place in the signal chain, although the Scope module is recommended for this. The module allows for offsetting, smoothing, flipping and stretching the chosen control signal using the *Bias*, *Smooth*, *Polarity*, and *Scale* knobs. This should be sufficient for most use cases. If not, the signal can of course also be altered by routing it through other modules.

Parameters	
SCALE	**Scale** Scales the control signal from 0 to 2. Default is 1 (no scale).
POLARITY	**Polarity** Controls the (bipolar) amplitude polarity. Full right means positive (+1) and full left means negative (-1) polarity (inverted signal).
BIAS	**Bias** Introduces a DC Bias that offsets the signal (post scale and polarity) from -1 to +1.
SMOOTH	**Smooth** Filters the signal with a one pole lowpass smoothing filter. 0 means no filter.
BIPOLAR	**Bipolar** When ON the control signal is scaled by 2 and centered, normally resulting is a signal in the amplitude range [–1;1]. When OFF, the control signal is passed *as-is.*
SIGNAL	**Signal** Chooses between the 4 internal control signals: **Frequency** is the *frequency control signal* that is used by oscillators for tone generation and by filters for e.g. keyboard tracking. This is traditionally a positive unipolar signal. However, negative frequencies can make the output both bipolar and negative unipolar. **Phase** is the bipolar *phase control signal* that is used by oscillators for e.g. phase modulation. **Trigger** is the positive unipolar *trigger/gate control signal*, used for (re)triggering and gating in e.g. the Amplifier module. 1 means note-on and 0 note-off. **Note** is the positive unipolar *note control signal*, which essentially is equivalent to the incoming stream of MIDI notes. This is the signal that is usually converted into the frequency control signal using a Controller module.

PREVIEW

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Index