KARMA·FX

SYNTH MODULAR 2

User's Manual

Revision 0.33

KarmaFX Synth Modular Version 2.00+
VSTi / VST for PC / Windows™ and Audio Unit / VSTi / VST for Mac / OS X.
Created by Kasper Nielsen,1998-2020.

E-mail: karmafx@karmafx.net
Web: https://karmafx.net

Software License Agreement

1. Grant of License and Copyright: KarmaFX grants you, the original purchaser, the non-exclusive and non-transferable license to use the program and data files constituting this software and the upgrade programs(s) and the data file(s) which may be distributed to you from time to time (herinafter called the "Software") on up to two computers which are under your control. You are allowed to use the Software royalty-free for music and sound production, recordings and/or performances. The ownership of the rights to and the copyright of the Software (whether supplied via a CD, diskette, download via the internet or otherwise) belong to KarmaFX. KarmaFX reserves the right to make modifications to the Software and to any support services without prior notification.

2. Restrictions: The Software contains copyrighted information. For the purpose of protecting such copyrights, you may not sell, sublicense, rent, lease, convey, distribute, copy, modify, translate, reverse engineer, convert to another programming language, decompile, or disassemble the Software for any purpose, or make derivative works based on what is contained in the Software. The Software is Registered in your name and may only be used by you personally. You may make a reasonable number of backup copies of the Software for your own personal use only. You may not transfer the Registered Software to another computer through a network or to another party by any means. As a condition to use the Software, you agree that you must not use the Software to infringe the intellectual property or other rights of others, in any way. The unauthorized reproduction, distribution, modification, public display, communication to the public or public performance of copyrighted works is an infringement of copyright.

3. Termination: This agreement takes effect on the date you have received the Software. The license granted in this agreement will terminate automatically without notice from KarmaFX in the event that you breach any provision of this agreement or in any way infringe KarmaFX's copyright in the Software. In such event you must destroy the Software and all backup copies immediately.

4. Disclaimer: The Software is provided as-is, and KarmaFX disclaims any warranty or liability obligations to you of any kind. KarmaFX disclaims all warranties relating to the Software, whether express or implied, including but not limited to any implied warranties of merchantability and fitness for a particular purpose, and all such warranties are expressly and specifically disclaimed. KarmaFX shall not be liable for any indirect, consequential or incidental damages arising out of the use or inability to use the Software even if KarmaFX has been advised of the possibility of such damages or claims. The user of the Software bears all risk as to the quality and performance of the Software. KarmaFX will not be liable in any way for any allegations or findings of infringement of copyright or other proprietary rights as a result of your use of the Software.

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 VST-compatible host application 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 and VST 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, 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 and Control modules.
- Expression option, so parameters can respond to MIDI Mod.Wheel, Aftertouch and Velocity inputs.
- 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.
- VST 2.4+ compatible host application.
- Low latency sound card (preferably with ASIO driver).

Mac / OS X:

- Mac computer with an Intel processor and 200 MB free disk space.
- Operating system: OS X 10.6 or later.
- Audio Unit (v2) or VST 2.4+ 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 will 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.57) has been added that offers sample-based granular synthesis, with variable grain size and rate and modulation/diffusion options.
- A new **Patch Browser** (p.35) 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.101) allows for patches to exist and work within other patches. This is a powerful feature which 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 for all practical purposes means that the synth per patch now offers an unlimited amount of modules.
- New Effects modules have been added: A **Folder** module (p.90) that performs wavefolding through classic 0Hz Frequency Modulation, a **SoftClip** module (p.99) that offers variable softclipping, a **Repeater** module (p.100): a beat-synced, repeated delay, a **Limiter** module (p.98) that offers soft-knee peak limiting, and the **MultiComp** module (p.92)**:** a 3 band multiband soft-knee compressor.
- New Controller modules have been added: The **Phase** module (p.76) which offers control over the phase control signal, and complements the updated **Frequency** module (p.75), a **Scope** module (p.83): a synced oscilloscope, frequency and phase analyzer of up to two separate input signals, as well as all internal controls signals, and a MIDI **Keyboard** trigger/visualization module (p.84). **Unison** has been reworked to offer more modes as well as **Phase Detune** (p.79) and Pattern module

has been updated with **Steptime**, **Velocity** and **Min**/**Max Note** range-settings (p.80). **Notepitch** & **Pattern** modules can now be detuned from *Concert Pitch* (A=440Hz).

- New Amplifier modules have been added: An **Inverter** module (p.72) for audio and modulation inversion and a **Mid/Side** module (p.73) 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.60), plus all Resonant Multimode filters now have **Saturation** and **Drive** controls.

- An **Exponential Tuning** of all Filter-Cutoffs has been added (p.60), giving better control over both high and low- filter frequencies, which sounds more natural when modulated. A **Parabolic Tuning** option is available for backwards compatability. Similarly, a **Resonance** tuning option has been added (p.60), featuring a flat **Digital** response and a simulated **Analog** response that attenuates resonance at low frequencies. The Filters **Keyboard Tracking Base Key** is now also adjustable.

- New Modulator modules have been added: A **Shaper** module featuring customizable waveshaping of modulation signals (p.114), and a **Control** module that exposes the internal control signals for use as custom modulation sources (p.115). **LFO** now supports optional **Random Poly Phase** Reset 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.77).

- 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, Modulation Index, LFO rate & amount, Amplification, etc. Optionally responds to Mod.Wheel, Aftertouch 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 now also 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.34).

- 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.31).

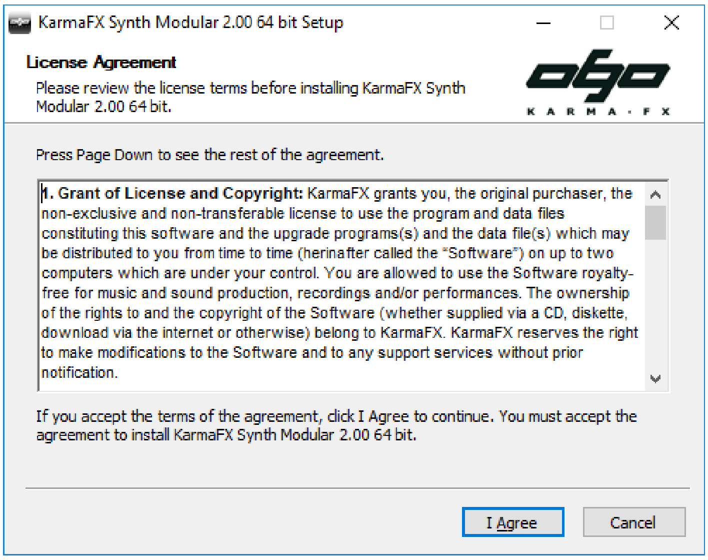
These additions are just the most prominent features. Many more minor improvements, and optimizations have been added in version 2. Please refer to the full version 2 changelog online for more technical details.

3. Installation

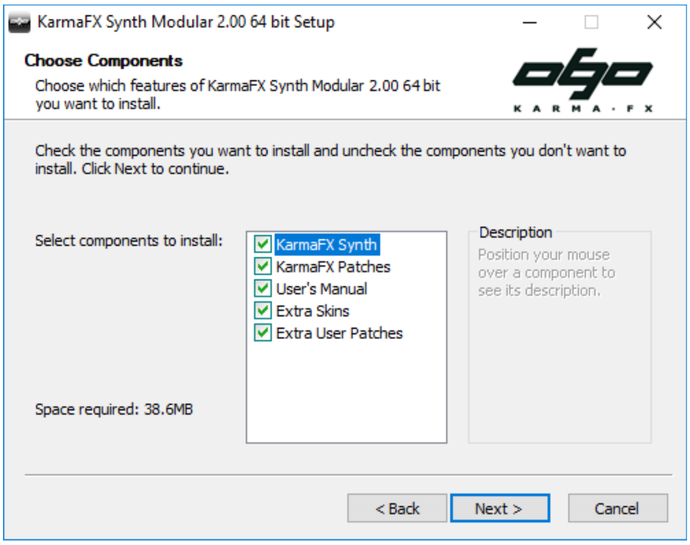
3.1 Installation on PC/Windows

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…

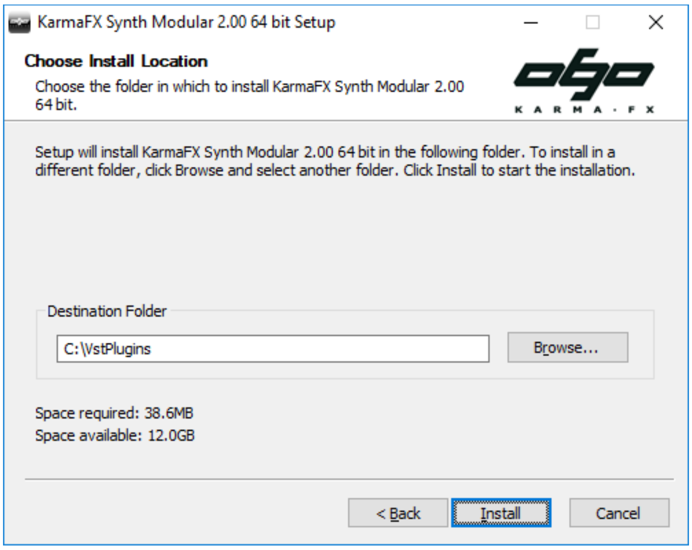
▶ Locate and run the the installation file. Use **KarmaFX Synth Modular 2 32 bit** or **KarmaFX Synth Modular 2 64 bit** for either 32-bit or 64-bit installations respectively.



▶ 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.



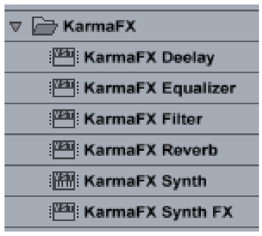
▶ Now select the VST-folder where your host application's VST-plugins are placed. In order to save patches inside the synth, it is important that this folder is user-writeable. We suggest:

```
C:\VstPlugins
```

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, if any.

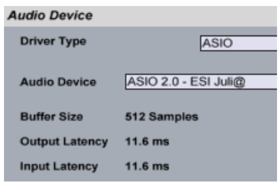
Click **Install** and the Synth will install into that folder.
Afterwards, click **Close** to end the installation program.

▶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:



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

▶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.

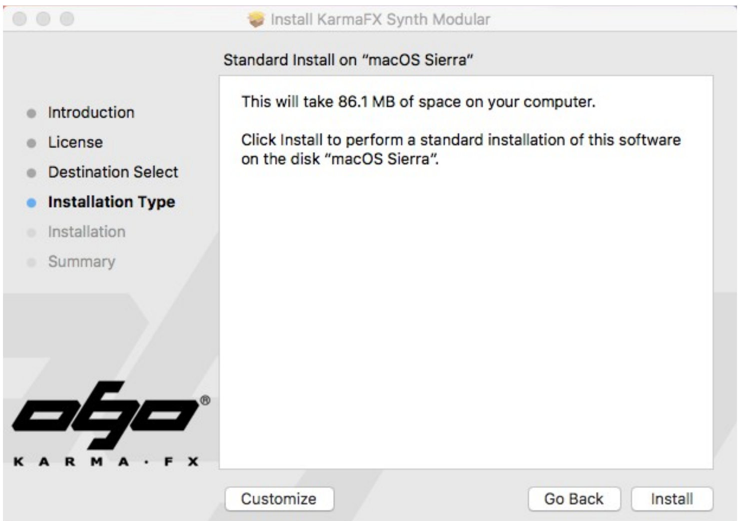
This manual in PDF format is located in:
`<vstfolder>\KarmaFX\KarmaFX_Synth\Manual`

3.2 Installation on Mac/OS X

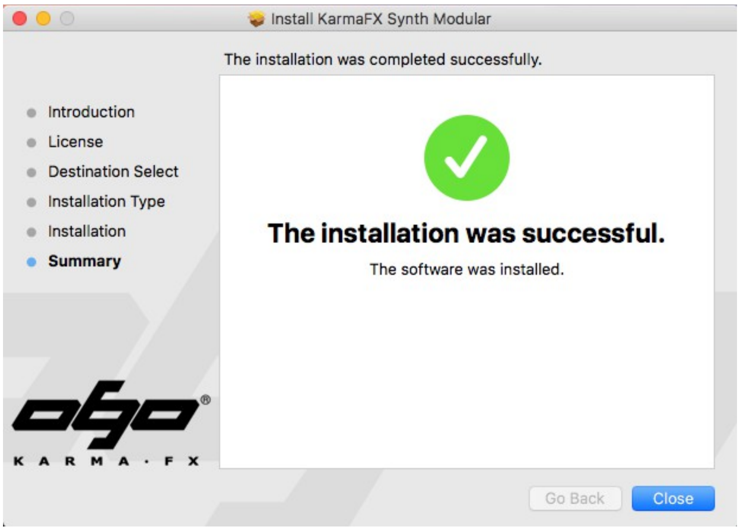
For installation on Mac/OS X, first make sure to close all running sound applications. Then...

▶ 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/`

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`

4. Concepts

This section will give some general background information on modular synthesis and explain the basic concepts used by the Synth. This is important in order to understand how the Synth produces and processes sound internally.

4.1 Digital Simulation of Modular Synthesis

Back in the days when analog synthesis thrived, a hardware modular synthesizer was considered state of the art for sound synthesis – and it still is in many ways.

A modular synthesizer is a system consisting of small devices, called modules, each solving their own specific task. The modules are wired together by cables. The cables carry a signal: an electric current of variable voltage, from a module's output to another's input. The signal can be an audio signal or a control signal.

Control signals are non-audio signals used to control certain variable parameters.

Some standard modules found in modular synths are:

VCO: Voltage Controlled Oscillator, which produces a waveform at a specified pitch.

VCF: Voltage Controlled Filter that cuts away frequencies, changing a sound's timbre. Hence, this is usually called *subtractive synthesis*.

VCA: Voltage Controlled Amplifier, which controls the amplitude or overall volume.

ADSR: Envelope generator (abbreviation for Attack, Decay, Sustain, Release) that simulates the contour of a natural decaying sound like a piano. Can be used to modulate, e.g., the VCA or the VCF to control the amplitude or the timbre of the sound over time.

LFO: Low Frequency Oscillator, which outputs a low frequency waveform, usually a sine or a triangle wave. Normally used as a control signal, e.g., for changing the pitch of the VCO to create a vibrato effect.

VCO/VCF/VCA section in Roland's System 110m

Many different modules exist, but the real strength is that modules can be wired together in completely arbitrary ways, making the sound creation possibilities almost endless.

PREVIEW

PAGE MISSING IN PREVIEW

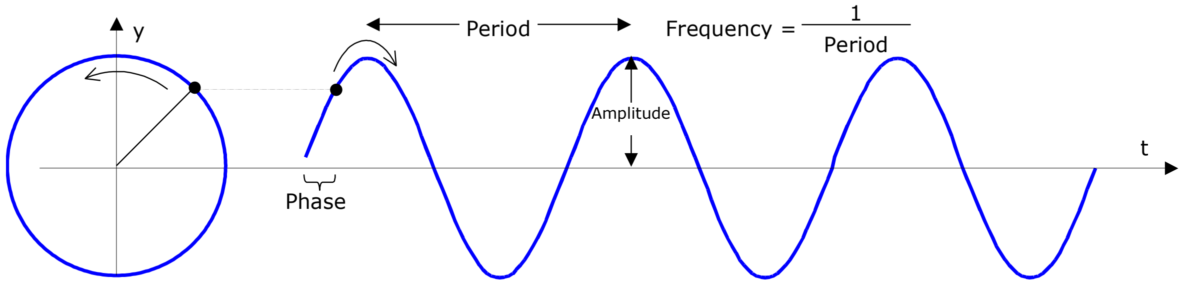
4.2 The Art of Sound Synthesis

This section gives a brief introduction to the fundamental principles of sound synthesis. If you are already familiar with *sine waves, additive* and *subtractive* synthesis, and words like *amplitude*, *frequency* and *phase*, you may skip this section entirely.

The word sine is derived from the Latin word sinus, which means "bay" or "fold".

Sine waves

Sine waves are a fundamental component in synthesis. In fact all sounds can be built up of sine waves at different speeds. The construction of a sine wave can be perceived as a point swirling around on a circle and plotting its *y* value over time:



The time it takes for the point to do a full cycle is called the *Period*. It tells us the speed of the sine wave. Another measure for this is *Frequency,* which is simply *1/Period*. Frequency is in fact equal to the number of cycles the sinusoid does per second.

The size of the sine wave is called the *Amplitude.* It measures its general loudness or air pressure.

Last but not least, where we start our cycle on the circle can be of importance. This is called the *Phase.*

So, a sine wave can be described using the three parameters: *Frequency, Amplitude* and *Phase*. In mathematical terms:

*Sine(t) = Amplitude * sin(2*PI*Frequency*t + Phase)*

PI is the name of the Greek letter π, pronounced "pie". It is a constant approximately equal to 3.14159.

If we were to exchange the *sin* function in the expression above with some other function, this formula can in fact be used to describe any periodic (repeating) signal.

PREVIEW

PAGE MISSING IN PREVIEW

Sample Synthesis

Sample synthesis is a technique that uses recorded sounds stored as digital samples. The idea is that instead of trying to re-create the sound of a real instrument, why not just record the real thing and play it back?

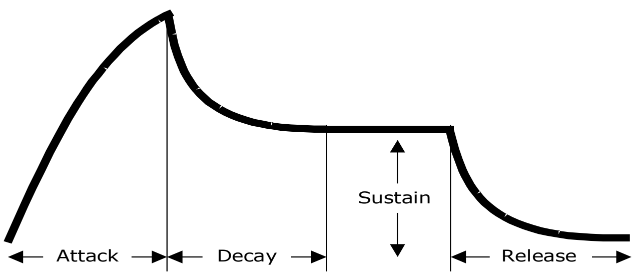
Typically, several sampled sounds have been recorded at pre-selected frequencies. They are then played back at different frequencies using interpolated resampling. Finally, the samples are altered by filter and amplitude changes in order to create a realistic imitation of an instrument. Samples are either single cycle waveforms (Wavetable synthesis) or larger recordings of instruments. The latter is a natural choice, e.g., for percussion.

Granular synthesis is a variation of sample-based synthesis, where the sample is chopped up into tiny sample clips, called grains. These are then re-assembled and played back overlapped using a windowing function. Unlike regular sample playback, granular synthesis allows the frequency to be controlled independently from the sample playback speed.

Synthesis by Modulation

If we look at the synthesis techniques above, it is clear that they all share a common concept, namely *Modulation,* or *changes* in sound. Modulating sound parameters using control signals is important to make sounds come to life, and a key feature in modular synthesis.

Typically, a parameter is modulated by an *envelope* signal. When a key is hit, e.g., on a piano, the sound amplitude rises quickly to a certain level, then fades a bit, until the key is released and the sound dies out. This shape of the sound is called an Envelope. The most common is the ADSR, which is an abbreviation for *Attack, Decay, Sustain, and Release*.



As shown, Attack, Decay, and Release, control the time for rising and falling of the envelope level, while Sustain controls the level when a key is held.

So, if we apply this envelope to a sound, we are in fact modulating the sound's amplitude. This concept can be extended further to modulate all kinds of parameters. Here is a list of common modulation types:

PREVIEW

PAGE MISSING IN PREVIEW

Sound Effects

So far we have looked at additive/subtractive/sample synthesis as a means to generate sounds and modulation synthesis as a means to alter sound characteristics. But sound effects such as delay, reverb, distortion, chorus and pitch-shifting can quite heavily alter sound characteristics too, and are, therefore, commonly used tools in modular synthesis.

Delay (echo) simply copies an incoming signal and outputs it at a later time. While simple in concept it can add depth and space to a sound, especially if the effect is combined with stereo and filter changes.

Reverb can make a sound appear as if it is placed inside a physical room. This is done by introducing several phased and decaying echoes in order to simulate wall reflections.

Distortion takes an incoming signal and modifies it to make it sound harsher. A common distortion technique is *waveshaping*, where an input value is mapped to an output value as dictated by a waveshaping function. The simplest (and crudest) form of distortion simply clamps a signal to a certain minimum and maximum amplitude value.

Wave Folding is a waveshaping effects that is based on frequency modulating a 0Hz carrier waveform. This *folds* the input with the waveform and creates harmonic overtones.

Chorus (& Flange) is an effect that fattens a sound by layering several detuned versions of the same sound. It is usually done using delay with a periodically modulated delay time.

Pitch-shifting changes the pitch of an incoming sound without changing the speed of the sound, again simulated using delay.

Phasing is a type of filtering that alters the phases of the different frequency components in the signal using so called allpass filters. A typical phaser modulates these phase shifts, creating an interesting and characteristic swooshing sound.

Compression / **Limiting** is a family of *dynamic processing* effects that alters the input's dynamic range, typically by attenuating amplitude levels when they cross a certain threshold. For compression, the signal can then be boosted, making details in the sound clearer, but without clipping.

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.

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.

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

NEW IN V.2

tag in the Modules Guide, page 48.

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.

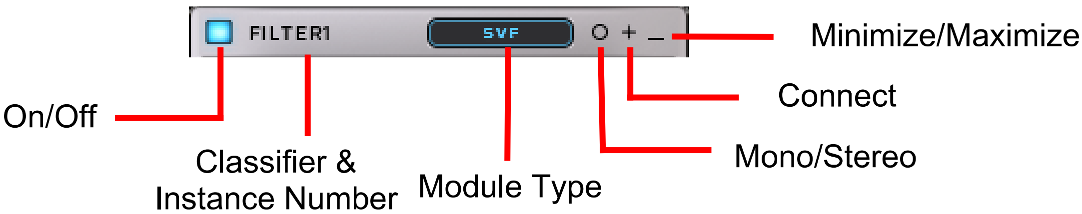
Modules generating signals that modulate parameters (Modulator modules) are directly connected to the particular parameter's knob or slider. In the bottom of the Workspace is a Control Panel, useful for quickly editing common parameters as well as for browsing and managing patches.

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

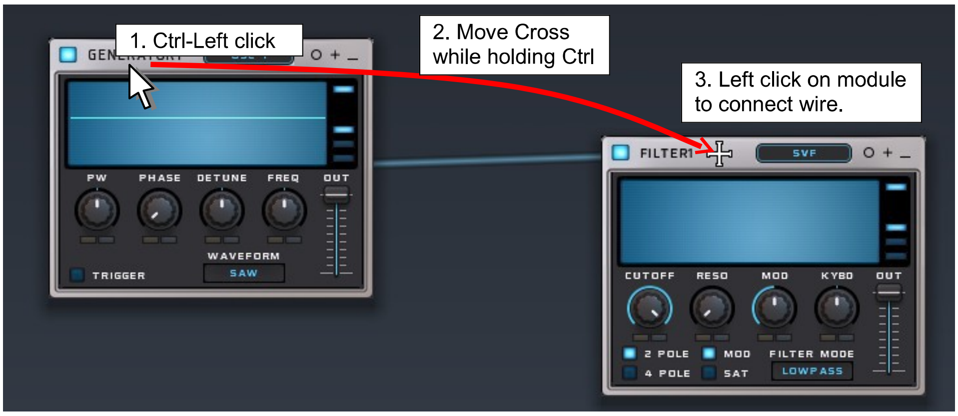
5.1 Modules

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

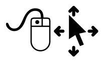
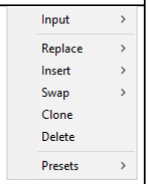
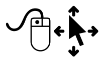
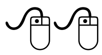


The first thing you notice on the title bar is the *module classifier*, e.g. *Generator, Filter, Modulator*, etc., and its current *instance number*. A dropdown menu chooses the *module type*, i.e., the functionality of the module. If it’s a filter module, the dropdown will show a selection of available 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.

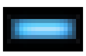
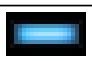
You can have up to 16 simultaneous instances of each module class, i.e., sixteen generators, sixteen filters, sixteen controllers, and so on. The only exception is the Output module: There can only be one Output module.

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.

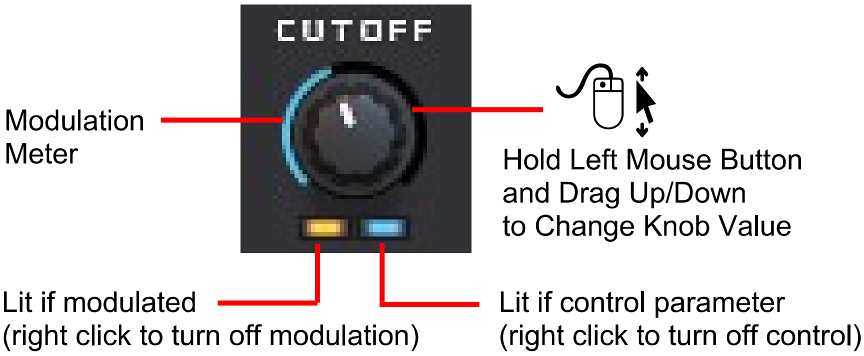
The display is controlled by the four switch buttons to the right of the display. The text area just below the waveform display shows info related to the current module, and parameter values whenever they are changed.

	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.

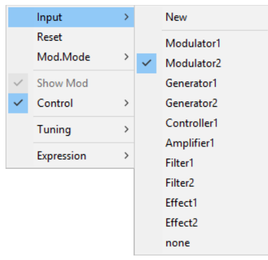


Tip: When changing a parameter, moving the mouse slowly will give you greater precision. Holding the Shift key will also increase precision.

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.

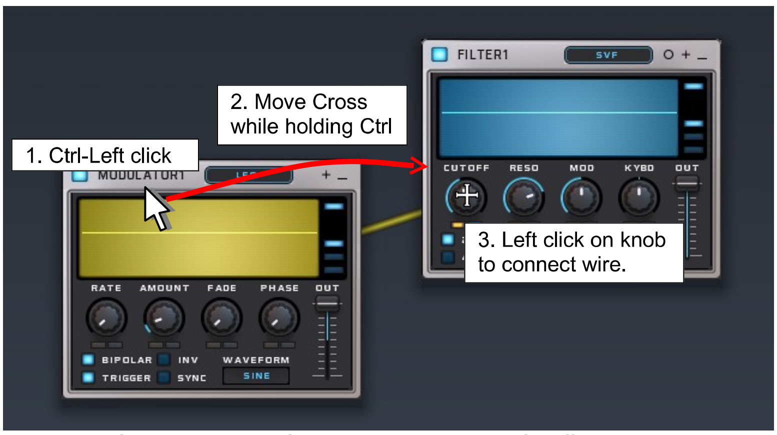
Tip: Holding Ctrl and left clicking on a knob or slider resets its value to center-position.

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.

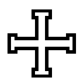


To connect a module to a parameter either right click on the knob/slider and select the source module from the Input menu. Alternatively, either press and hold **Ctrl** and click on the title bar, or on the title bar's Connect button on the source module, and then click on the knob/slider you want to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

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



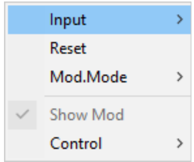
Connecting an LFO modulator to the Filter Cutoff knob



Connection cursor

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

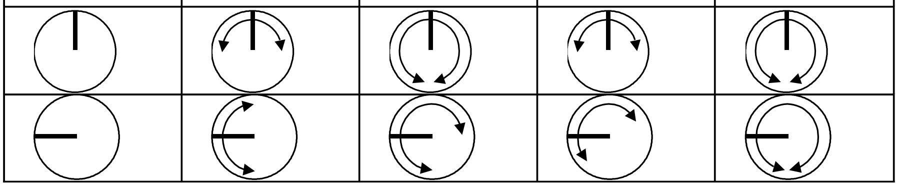
Besides the Input submenu, the right click menu has some interesting options:



Reset: Resets the parameter value to its default value.

Mod. Mode: Choose between *Linear* or *Range scaled* (default) modulation mode. In linear mode the modulated signal is simply added to the initial setting. In range-scaled mode, the signal is still added, but first after it is multiplied by a scale factor. The scale factor is adjusted so a full range modulation (-1 to 1) will always use the full range of the knob that is modulated.

Knob setting	50% Mod. Linear	100% Mod. Linear	50% Mod. Range scaled	100% Mod. Range scaled



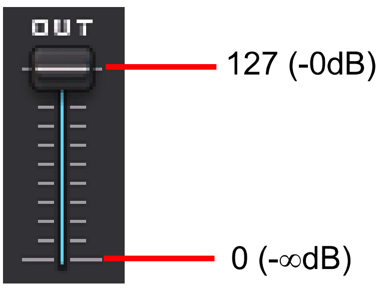
Range scale mode usually sounds "better" because it uses the full range and avoids modulation clamping. Linear mode is best used for parameters that have to be linearly modulated. E.g., range scale modulating a "note" or "octave" parameter can make it go out of tune.

Show Mod.: For Skins that don't have modulation meters, this shows the current modulation of the parameter directly in the interface, by animating the knob or slider. This is very useful to get an idea on what is really going on. However, doing it on many knobs will most likely make your head spin.

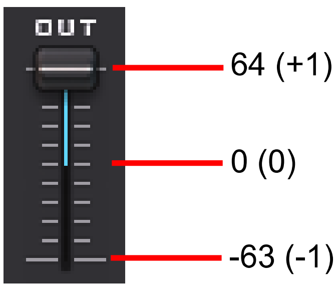
Control: Link the parameter to one of 128 available assignable *control parameters*. A control parameter is simply a parameter seen from the host, meaning that it is possible to change and automate it from within the host application. 32 of the available 128 control parameters appear on the Control Panel in the synth for instant tweaking within the synth (see section 5.4).

Some parameters also have extended options to, e.g., change the parameter's tuning, range or to make it snap to certain intervals.

Finally, all modules have an output slider for controlling the final amplification of the signal:

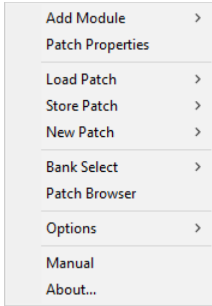


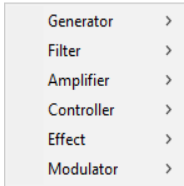
Audio Output slider



Modulator Output slider

5.3 The Right Click Menu

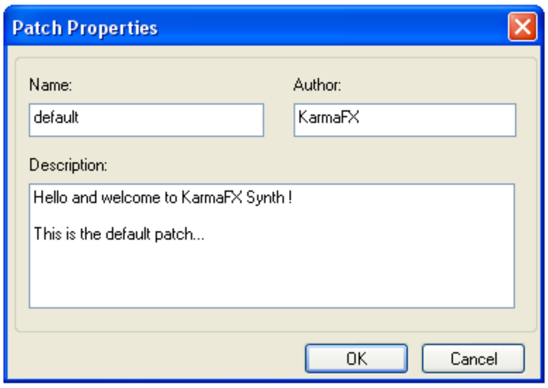




The main right click menu gives you easy access to common patch editing tasks, like adding new modules, loading and storing patches as well as changing general options in the synth. To open it, simply right click on the workspace background.

Add Module: Creates a new module of your choice selected from the submenu. The chosen module is placed at the mouse cursor.

Patch Properties: Shows the patch properties dialog box. Here you can change the name of the patch and the name of the author. If you want you can also add a description of the patch.



Load Patch: Directly choose a patch to load from the current patch bank. A bank is simply a collection of up to 128 patches stored in one folder.

Store Patch: If a bank contains vacant slots, these will also be selectable (unlike the load patch submenu). You can also select an existing patch if you want to replace and overwrite an already existing patch. The synth will ask for a confirmation before overwriting any patches.

Bank Select: Select the patch bank to browse from in this submenu. Note that this selection will not alter the currently loaded or edited patch. This means that it is possible to copy a patch from one bank to another, by first loading the patch from one bank, then selecting another bank, and finally storing the patch in that bank. Selecting a *<New Bank>* item will prompt for a new bank name, and when OK is pressed, creates a new bank folder by that name and switches to that bank. Use this when you want to roll your own patch banks.

Patch Browser: Opens the Patch Browser (section 5.6). Can also be achieved by clicking on the Control Panels display (section 5.5).

Options: Shows the options submenu. Here you can alter various overall settings in the synth (see section 5.4).

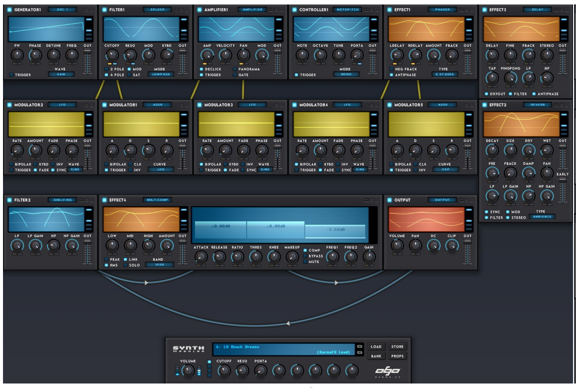
Manual: Opens this manual in PDF format.

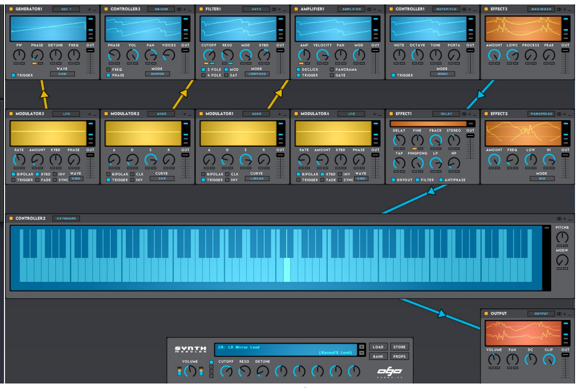
About: Shows the About Dialog for KarmaFX Synth Modular. Here you can see version information, the current build number and date, as well as the name of the registered user (hey, that's you! ☺).

PREVIEW

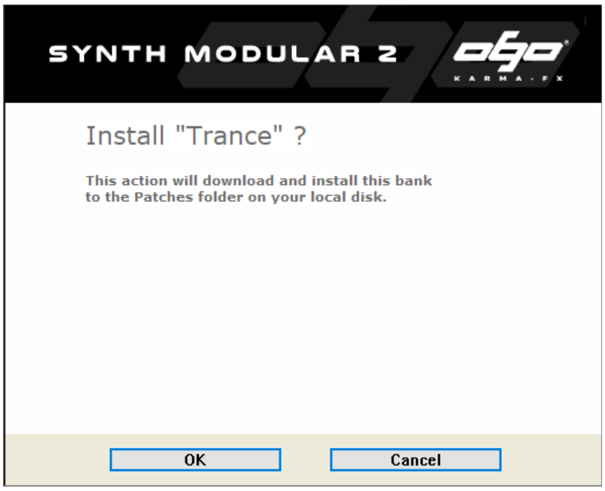
PAGE MISSING IN PREVIEW

Skin: Changes the skin, i.e., the graphics used to draw the user interface (modules, knobs, slider, wires, etc.). Again a matter of taste. The "Ivory" skin is used for the screenshots in this manual. Here are some alternative skin examples:

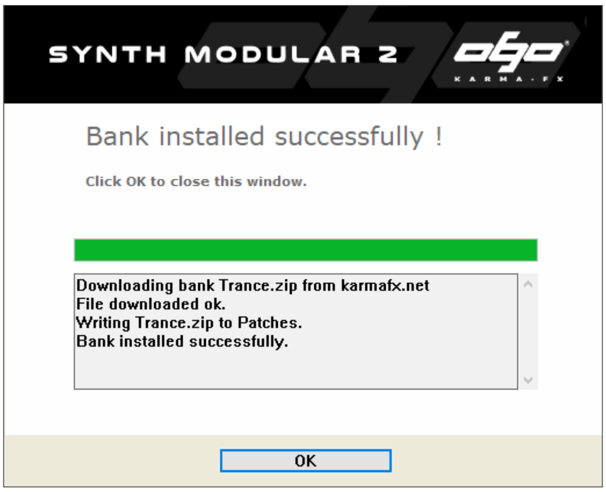

"Dark" Skin


"Neo" Skin

Bank Install: This menu lists the patch banks that are available online. Since KarmaFX Synth is created to work offline as well as online, you need to select the **Update Bank List** menu item, for the synth to contact the server and download the latest patch bank list. Patch banks that have already been installed are shown grayed out. Clicking on a bank will ask if you wish to install this bank to your local disk.



Click **OK**, and the synth will proceed to download and save the chosen patch bank in the Patches folder. Hence, the downloaded patch bank is made available instantly for you to browse and use.

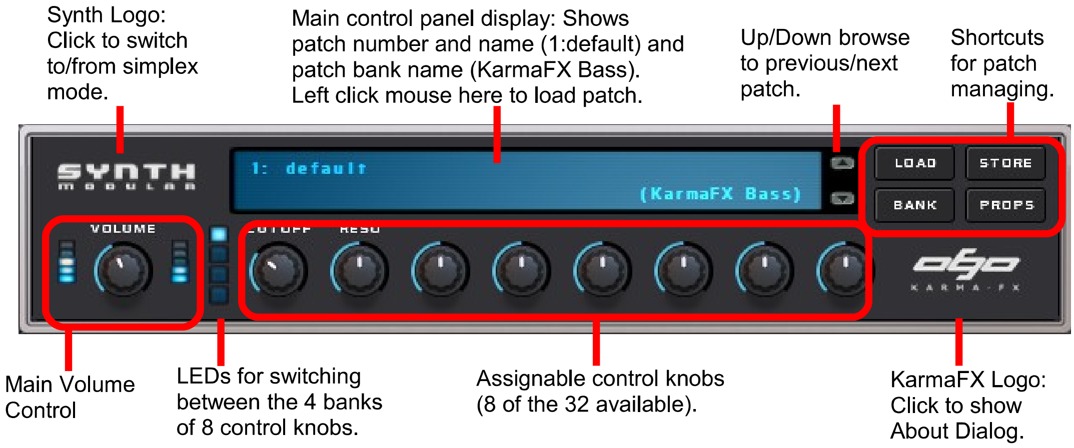


PREVIEW

PAGE MISSING IN PREVIEW

5.5 The Control Panel

The Synth's Control Panel located in the bottom of the workspace is useful for editing and browsing patches. The main display shows the current patch name and bank name. The arrow-buttons to the right of the display are used for quickly browsing to the previous/next patch.



The four rightmost buttons are shortcuts to **Load** and **Store** patches, change **Bank** or properties (**Props**). The Load Patch button brings up the Patch Browser (See Section 5.6.), while the rest work exactly as when choosing the same options from within the right-click menu,

Load Patch

Store Patch

Bank Select

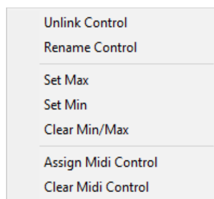
Patch Properties

The leftmost knob controls the **Volume** for the patch. This knob is simply linked to the volume knob on the Output module. The LED's to both sides of the knob indicate the current output level in the left and right channels respectively. It is good practice to make sure that it doesn't reach the red level, but the synth will not clip the signal (unless you actively turn on clipping in the output module). So even if it's in the red, you can still pull down the volume in your host application a get a clean (unclipped) sound.

The row of eight knobs below the main display, give direct access the user-linkable controls, in 4 sets of 8 knobs. The LED's to the left, switch between the 4 sets of knobs. Whenever a knob or slider is linked to a control it will automatically appear here, as well as inside the host application. This means that you can tweak it and even record it in the host for automating parameters in a song. If you want, several different parameters can be linked to the same control knob.

Control knobs should generally be assigned to parameters that you would most likely want to change in a song.

Right clicking on a Control Knob will bring up a menu, giving you the option to **Rename Control** to a different name or **Unlink Control**. The Control Knob's ranges can be reprogrammed by using the **Set Max** and **Set Min** options, and reset again using **Clear Min/Max**. You also have the option to assign a knob to a specific MIDI Control (MIDI CC) for easy editing of parameters from e.g. an external MIDI device. Click **Assign Midi Control** to start *MIDI-Learn* mode. Then change a knob or slider on your MIDI device to automatically assign that MIDI CC to the control knob. For this to work, make sure the host transmits all MIDI events to the plugin. Use the **Clear Midi Control** option to disconnect the MIDI Control again.



PREVIEW

PAGE MISSING IN PREVIEW

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).

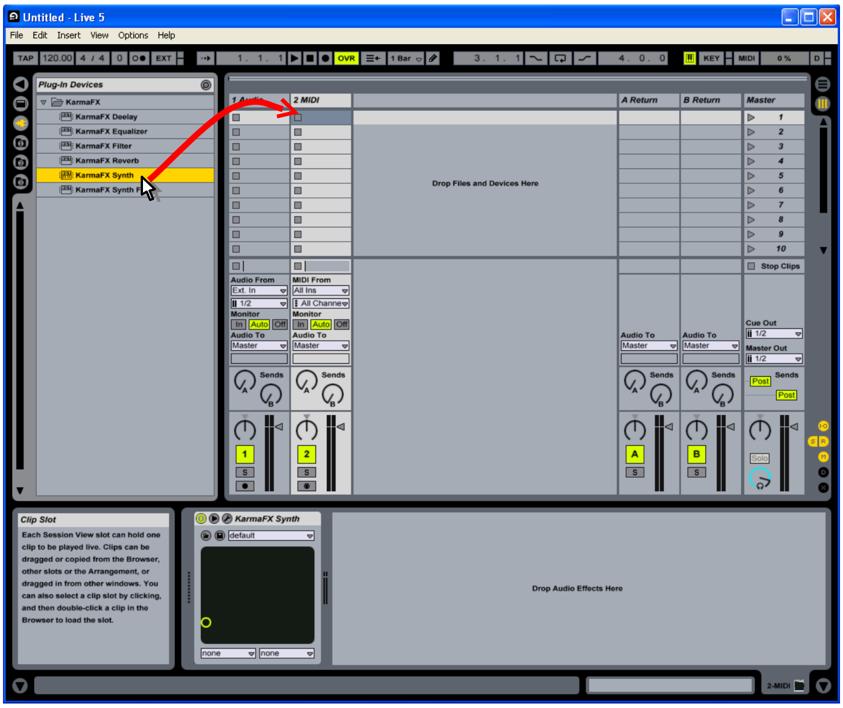
If the synth should generate sound based on the played nodes, it should be used as an instrument. The instrument generates audio based on MIDI events, so it has to be placed in MIDI track.

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.

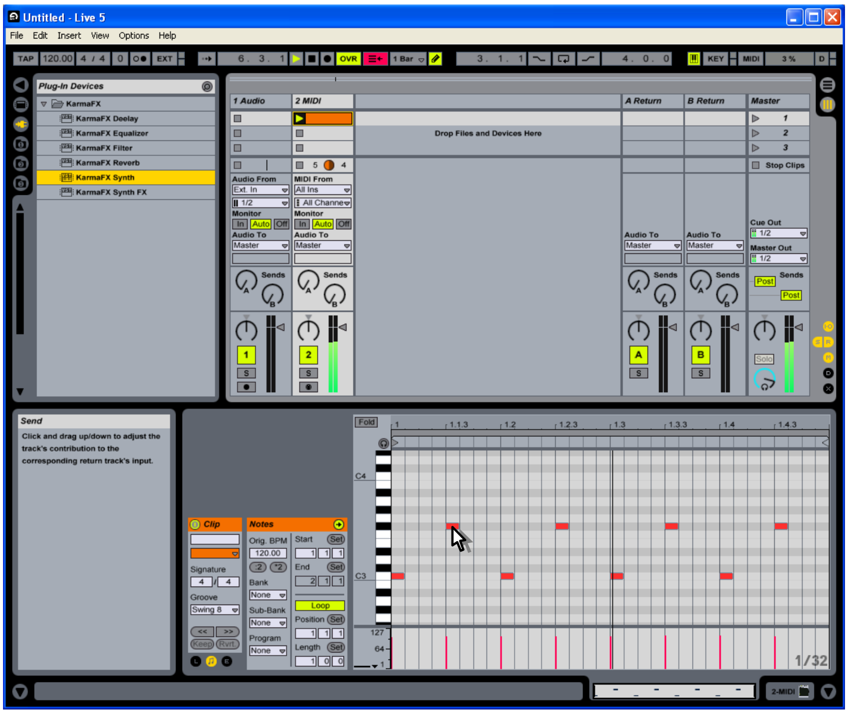
The following illustrates how to use KarmaFX Synth in the host Ableton Live. You host of choice might work differently. If in doubt, please refer to your host applications users manual to learn how to use VST plugins.

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.

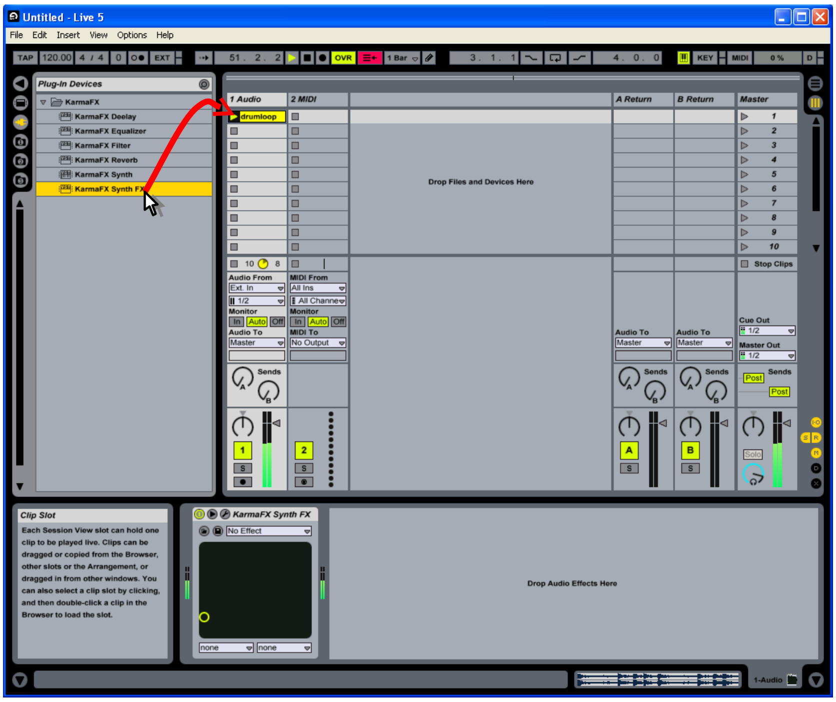


▶ Alternatively, if you have a MIDI keyboard connected, click on MIDI input monitoring, and play a few notes on your keyboard.

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.

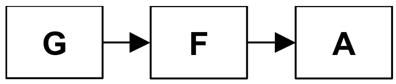
Always be sure to backup your homemade patches stored within the built-in banks before installing newer versions of the Synth. Otherwise these patches might be overwritten on installation. To be on the safe side, it is recommended that create a new bank with a unique name and store your own patches there. 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.

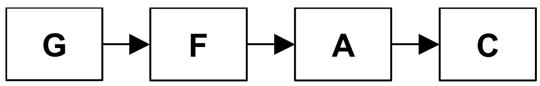
▸ First, we place a generator (denoted **G**). What kind of generator is not so important. It simply creates a sound at a certain frequency:

▸ 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:

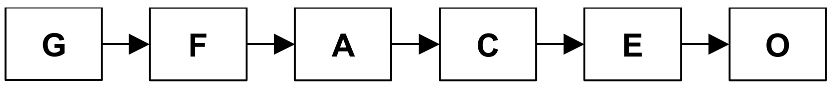


Notice how this corresponds to the vintage analog VCO→VCF→VCA patching**.**

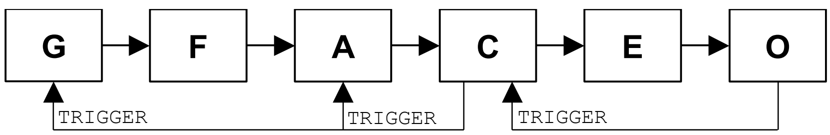
▸ We are now ready to send the results into a controller. The controller will handle all MIDI-triggering of the Generator, Filter and Amplifier as well as send a frequency control signal to these modules:



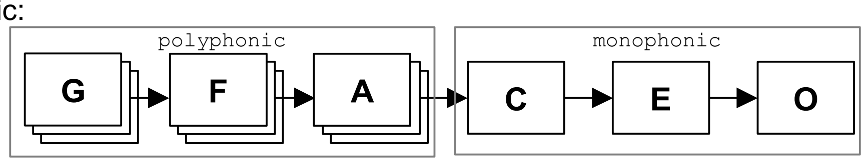
▸ Finally, we send the result into an Effect (**E**) and out to the Output module **O**:



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:

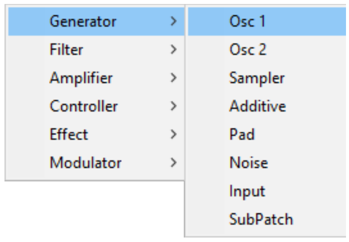


This happens automatically and ensures that only modules that really need multiple voices will get it.

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.

▶ Now we have to connect the modules. Start by connecting Osc1 to SVF. Either hold the **Ctrl** key, click on Osc1's title bar and then click anywhere on the SVF module (with the exception of any knobs & sliders) - or - click the **Connect** button on Osc1 (the small "cross" icon) and then click on the SVF module.

▶ Do the same for the other modules. You can wire several modules in a row by continuing to hold the **Ctrl** key.

We're done! First, check that the wiring and the arrows on the wires are as shown. Then try to play a note in the host's sequencer or on your MIDI keyboard. You should hear a bright sawtooth buzzing sound with a bit of echo. Not very impressive, but it's a start.

Let's dig in to the details of how and why it works: By default Osc1 generates a sawtooth wave at a pitch chosen by the controller. The trigger option on Osc1 selects whether the waveform should phase restart on a note-on MIDI event.

The Amplifier is by default set to *Gate* mode, meaning that it only lets the incoming signal pass between a note on and note off MIDI event pairs. The Controller is set to mono mode, i.e., only one voice is allowed to play at a time.

Try turning the cutoff knob on the SVF module. Since it is set to lowpass by default you should hear the familiar attenuation of high frequencies/harmonics.

▶ We can easily make this patch more interesting. Let's try setting Osc1's waveform to a square wave. Now create an LFO modulator and route that into Osc1's pulsewidth knob. You do this much like when routing a module to another: Either hold the **Ctrl** key, click on the LFO's title bar, and then, while still holding **Ctrl**, click directly on the Pulsewith knob - or - click the **Connect** button on LFO's title bar and then click the Puslewidth knob.

PREVIEW

PAGE MISSING IN PREVIEW

6.4 Saving Patches

By default patches are saved within the song in the host application. This means that if you load up a patch, make some changes and then save the song. The patch is stored too and will reappear exactly as when you saved it the next time you load the same song.

You may also choose to save a patch in a bank. This is useful when using the same patch in different songs, or when rolling your own patch banks. To save a patch, right click on the background and select "Store Patch" or click on the "Store" button on the Control Panel. Then select the patch slot you wish to store your patch in. If the selected slot already contains a patch, the system will ask if you wish to overwrite it.

Some Patch Banks may be *read-only*, which means that you can load but not save patches to the bank. To save a patch you instead have to create a new bank or select a patch bank that is not read-only. Internally a read-only bank is simply a ZIP compressed folder, so another way to make it writable is simply to unzip it to the patch bank folder.

Creating you own patch bank from within the synth is easy too: Simply choose "Create bank…" in the main right click menu.

There are several reasons to why you might want to make you own patch bank. First it enables you to store your work in one place, so you don't have to worry about remembering in which song you used your patch. It is also easier to distribute patches, if you should feel generous and wish to share your work with others. Finally, your work can easily be backed up and restored if you have to reinstall the synth or move the installation to a new machine.

Once you are done with your own patch bank, you can compress the entire patch folder using ZIP compression, making sure to name the compressed file the same as the original patch bank folder - just with an added ".zip" extension. The synth will then read patches directly from a compressed bank, but - as mentioned - it cannot write to it. So aside from saving disk space this effectively makes your patch bank read-only as well as easy to distribute.

PREVIEW

PAGE MISSING IN PREVIEW

PREVIEW

PAGE MISSING IN PREVIEW

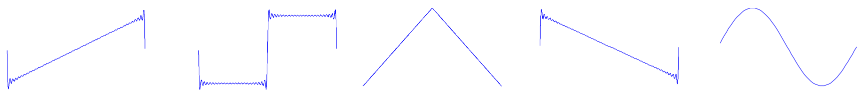
PREVIEW

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** <u>only</u> 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

Sampler

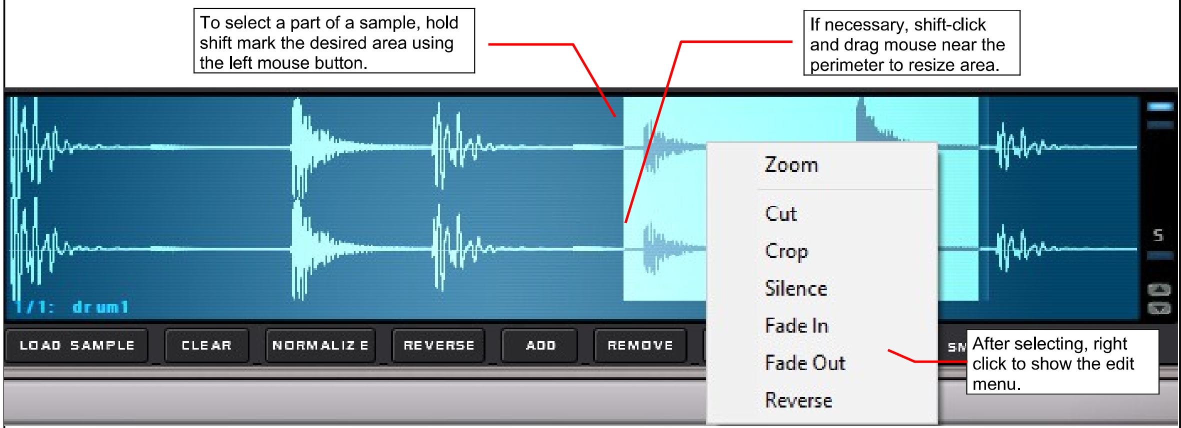
Generator

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.

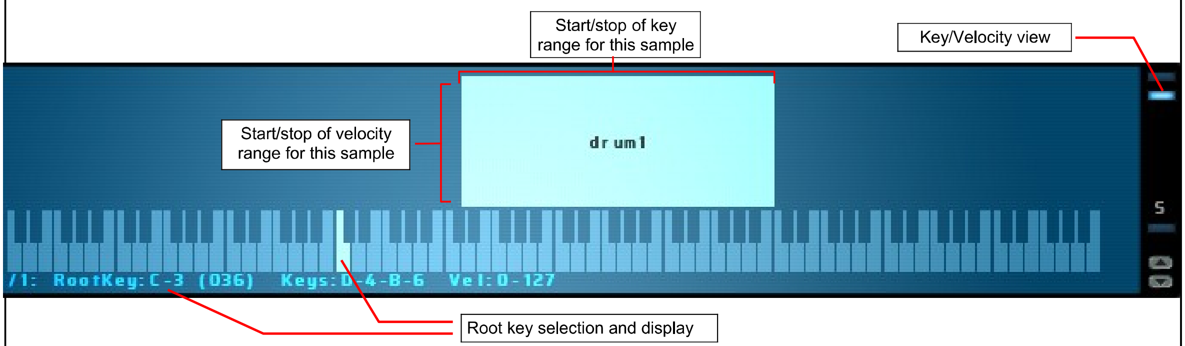
Parameters	
START	**Start Position** Sets the relative start position, where the sample will start to play when triggered and loop to when looped.
LENGTH	**Length** The relative length of the sample range to play and/or loop.
DELTA	**Delta** Adjusts the relative playing speed. Similar to slowing down 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 this knob has no effect.

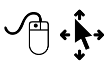
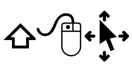
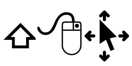
Sample Display – Sample Selection and Editing



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

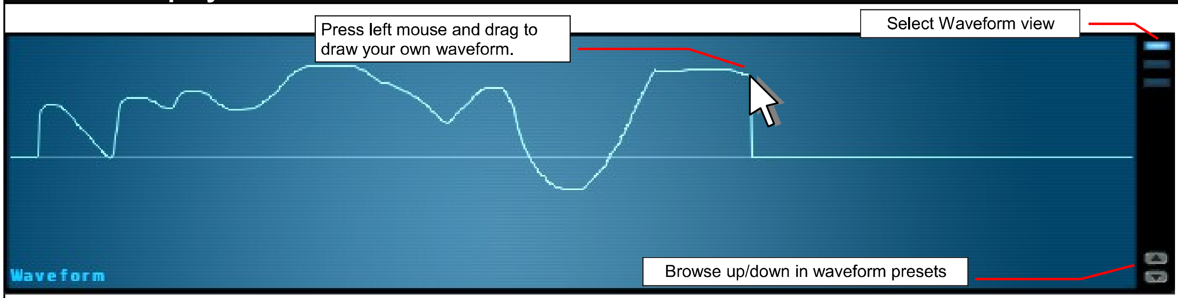


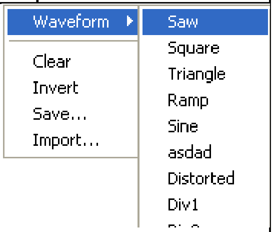
	**Left click on the drawn keyboard** Selects new root key. The root key selects the pitch at which the sample is played "as is", i.e., without any stretching or resampling. All resampling is done according to the chosen root key.
	**Shift + Left click in key/velocity field and drag** Sets start of key 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

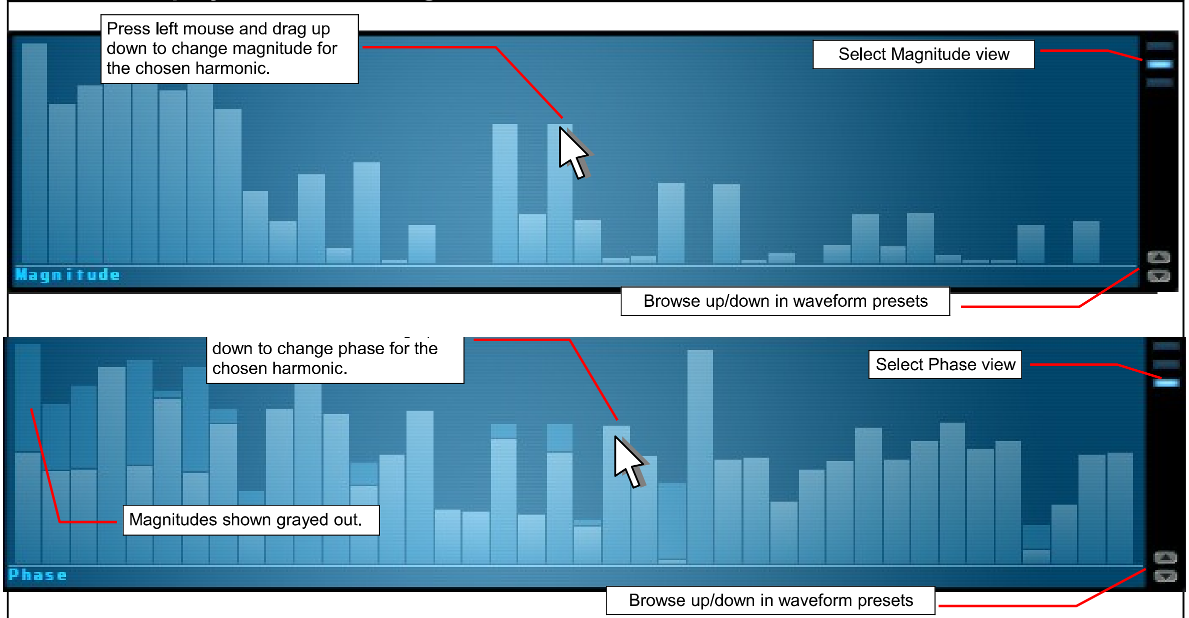
PAGE MISSING IN PREVIEW

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

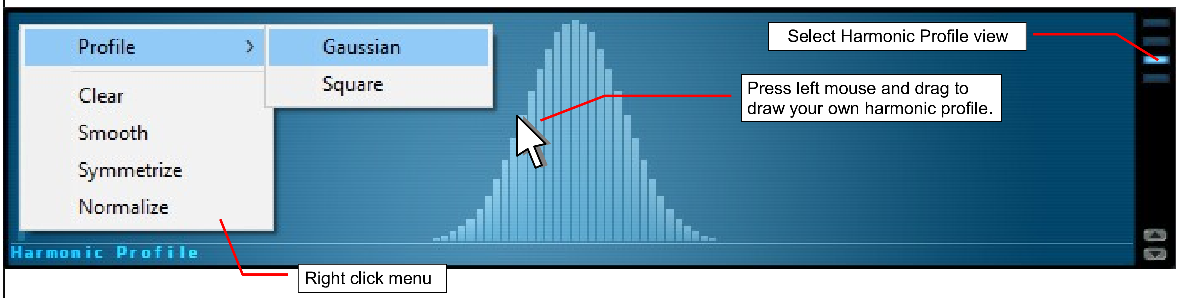


	Left click + Drag Change the magnitude or phase of one or more harmonics.
	Right Click Shows the right click menu: RESET TO SAW: Set magnitude or phases to saw profile. RAMP: Fades magnitudes or phases to zero. 

PREVIEW

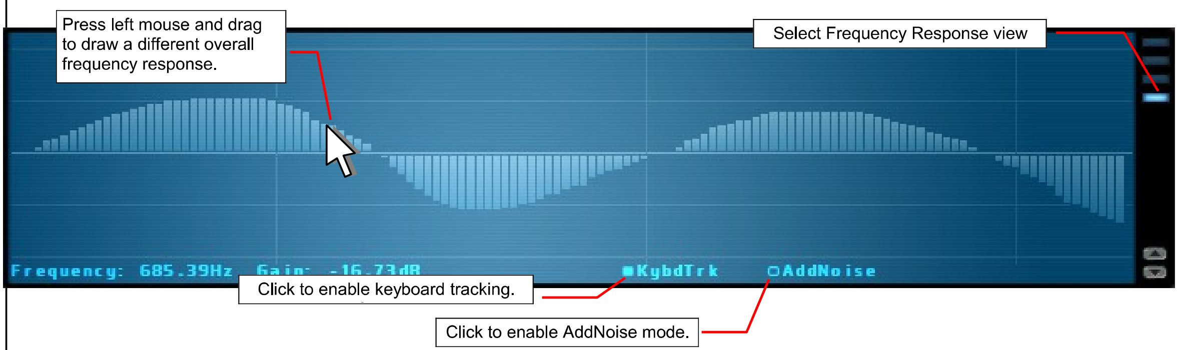
PAGE MISSING IN PREVIEW

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.

PREVIEW

PAGE MISSING IN PREVIEW

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

SVF/SVF2/SKF/Zolzer/Moog/Moog2/Acid/MS20

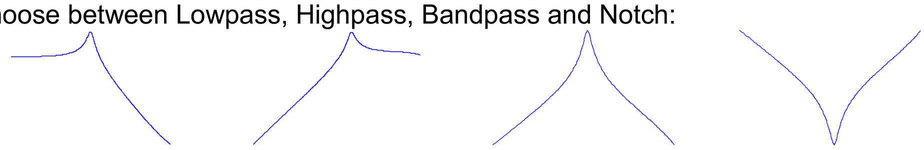
Filter

The SVF and SVF2 (State Variable Filter), SKF (Sallen-Key Filter), Zolzer, Moog, Moog2, Acid and MS20 modules are simulated analog resonant multi-mode filters (i.e., Voltage Controlled Filters, VCFs). They share the same parameters: Cutoff, Resonance, Mod and Kybd (Keyboard Tracking), Saturation and Drive, and have a selection of filter types: Lowpass, Highpass, Bandpass and Notch (Bandreject). The filters have subtle but different sound characteristics: The SVF and SKF are the "standard" synth filters you can turn to whenever general filtering is needed. Zolzer can sometimes sound more round and analog, while Moog has a *Moog*'ish sound to it, meaning that it has a steep cutoff and is remarkably self-resonant. Finally, Acid emulates the sound characteristics of Roland's famous TB303 bass unit, and MS20 the analog sound of the Korg MS20 Filter. Because the filter modules share the same parameters, parameter settings are retained when switching between them.

Parameters	
CUTOFF	**Cutoff Frequency** Selects the cutoff point, which is the frequency in Hertz where the filter will start to attenuate. The right click menu contains an Expression option and a Cutoff Tuning setting (see below).
RESO	**Resonance** Selects the amount of boost to apply at the cutoff (at around –3dB). The right click menu contains an Expression option and a Resonance Tuning setting (see below).
MOD	**Modulation Amount** Adjusts the amount of modulation to apply to cutoff. If cutoff isn't modulated this knob has no effect. Optional modulation factors are 1(default), 2, 4 and 8.
KYBD	**Keyboard Tracking** Sets the amount of keyboard tracking from –200% to 200%. This offsets the cutoff frequency with respect to the incoming frequency control signal. Setting this to 100%, means that the outgoing sound will have the same timbre regardless of the played pitch. Without a frequency control signal, the **Kybd** knob will have no effect. Hence, a filter should be placed before any controllers in the signal chain in order to perform keyboard tracking. The **Base Key** (default:C1) from which the Keyboard Tracking is determined can be setup through the right click menu.
SATURATION	**Saturation** (Knob appears when the *SAT* LED is selected) Controls the amount of saturation of the internal feedback inside the filter. Saturation sets the mix ratio between the unsaturated and the saturated signal. A parameter setting of 0 means no saturation (default).
DRIVE	**Drive** (Knob appears when the *SAT* LED is selected) Controls the drive of the internal saturation of the filter. Drive sets the prescaling of the unsaturated signal feeding into the internal soft saturator inside the filter, such that a higher value means more distortion. For high values, the distortion can cause the filter to overdrive and self-resonate.

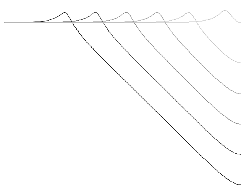
Parameters (Continued)

2/4 POLE 4/8 POLE	**Filter Steepness** The number of poles determine the steepness of the filter, A 2 pole filter attenuates 12dB/Octave, 4 pole: 24dB/Octave and 8 pole: 48dB/Octave.
TYPE	**Filter Type** Choose between Lowpass, Highpass, Bandpass and Notch:

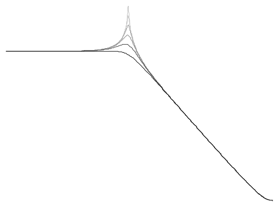


Lowpass Highpass Bandpass Notch

<u>SVF Lowpass Example Frequency Responses:</u>

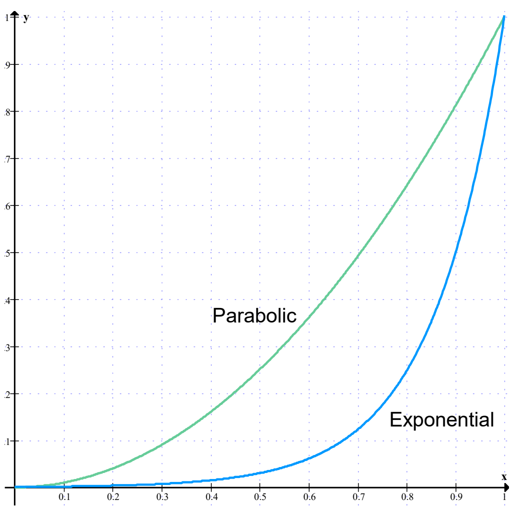


Lowpass Cutoff Sweep

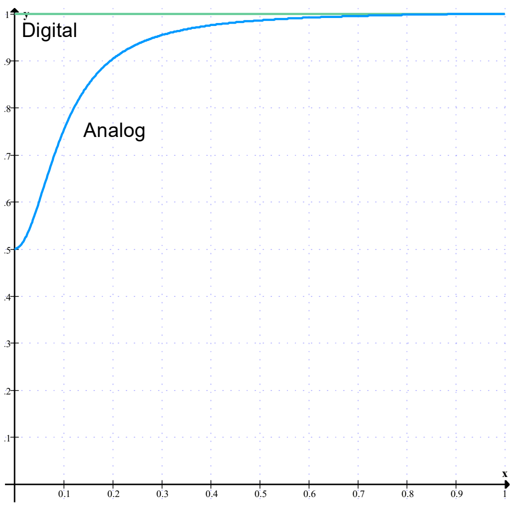


Lowpass Resonance Sweep

Cutoff & Resonance Tuning Curves



Cutoff Tuning



Resonance Tuning

CUTOFF TUNING	All Cutoff parameters in the synth have a Tuning option, that maps the parameters linear setting to a frequency through a Tuning curve. The **Exponential** curve (default) maps the frequency setting *exponentially* similar to how humans naturally perceive pitch. This gives equal amount of control in the high-end of the spectrum as in the low-end, but it is never able to reach 0Hz. The **Parabolic** curve is a simpler legacy option, that gives less low-end control, but if needed does reach 0Hz.
RESONANCE TUNING	The Resonance parameter has a Resonance Tuning setting that optionally tones down the Resonance at low-frequencies, as defined by the **Analog** Resonance tuning curve. This mimics the response of old analog filters. Default is a flat **Digital** resonance response.

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

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

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.
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 has an *Enable Pitchbend* option (on by default). When on the module will handle MIDI pitchbend events automatically using the chosen semitone range.
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.

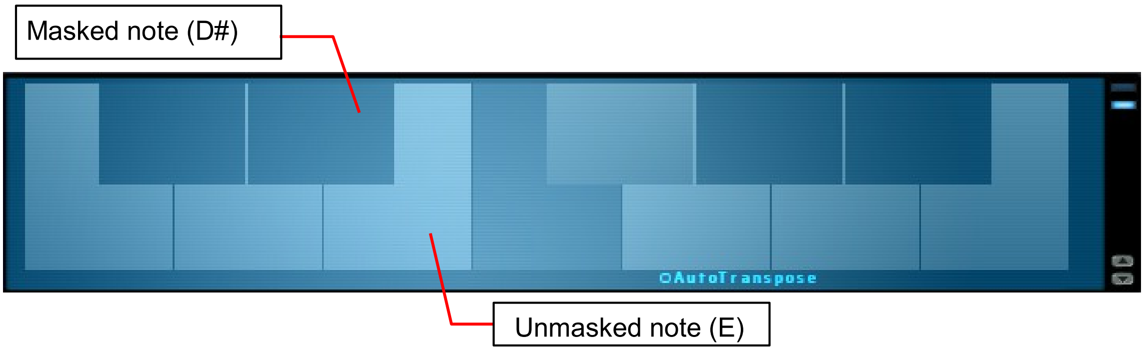
FM Modes	
FMLIN	**Linear FM** Linear FM is a classic synthesis technique where the frequency is swept up and down by a frequency deviation defined by the Modulation Index. Regardless of modulation source, Linear FM always sweeps up the same amount of Hz as it sweeps down. This keeps the output in-tune. 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.
AA	**Antialias** Enables modulation index limiting to reduce sideband aliasing. See also HQ.
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

PAGE MISSING IN PREVIEW

Note Mask Display



	Left Click on Note Handles Turn notes on or off. Masking a note, means that it will never be played and won't be part of the musical-scale. Hence, when a note is triggered that is masked, it is snapped to the nearest non-masked note. Unmasking a note means that it will play normally, and hence be part of the musical-scale.
	Right Click – Show Right Click Menu Semi +1: Shift masking/scale one semitone to the right. Semi -1: Shift masking/scale one semitone to the left. Scale: Choose between 31 pre-programmed musical-scale presets: *Chromatic, Ionian, Dorian, Phrygian, Lydian, Mixolydian, Aeolian, Locrain, Major Blues, Minor Blues, Diminish, Combination Diminish, Major Pentatonic, Minor Pentatonic, Raga Bhairav, Raga Gamanasrama, Raga Todi, Spanish Scale, Gypsy Scale, Arabian Scale, Egyptian Scale, Hawaiian Scale, Japanese Miyakobushi, Ryukyu Scale, Wholetone, Minor 3rd Interval, 3rd Interval, 4th Interval, 5th Interval, Octave Interval*
AUTO-TRANSPOSE	**AutoTranspose** When on, automatically transposes the scale with respect to the *Note* knob.

Range Parameters

STEP	**Steptime** Sets the steptime limit for notes. A setting of 127 means no limit (default), where keys are allowed to be sustained, without any note-off.
VELOCITY	**Velocity** Sets the velocity scale (default is 127: no scale).
MIN	**Min Note** Sets the minimum allowed MIDI note to be played from 0 to 255. The right-click-menu offers a Limit setting on how to handle notes outside range: **Clamp** (clamps to last allowed note), **Wrap** (wraps within the last octave), **Mirror** (mirrors note within the last octave), **Mute** (mutes the note). Default is *clamp*.
MAX	**Max Note** Set the maximum allowed MIDI note to be played from 0 to 255. **Limit** menu is available. See *Min*.

PREVIEW

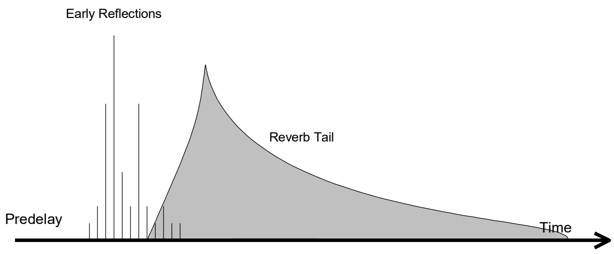
PAGE MISSING IN PREVIEW

PREVIEW

PAGE MISSING IN PREVIEW

Reverb

Effect



The Reverb module adds reverberation to the incoming signal, giving the impression of sound being played inside a room or other confined space.

Parameters	
DECAY	**Decay** (ms) Controls how fast the reverb tail dies out (fades) from 0 to 10 seconds.
SIZE	**Size (Diffusion)** Controls the virtual size of simulated room from small, with few and fast wall-to-wall reflections, to large, with many diffusely spaced out reflections.
DRY	**Dry** (dB) Sets the Dry amount, i.e., how much of the original signal to mix with the output.
WET	**Wet** (dB) Sets the Wet amount, i.e., how much of the reverb signal to mix with the output.
PRE	**Predelay** (ms) Sets the pre-delay: The time delay before the reverb kicks in (0 to 500ms).
FBACK	**Feedback** Sets how much of the reverb signal to feed back into the reverb.
DAMP	**Damp** Controls the attenuation (damping) of the high frequencies in the reverb tail.
PAN	**Pan** Pans the reverb signal left or right.
LP	**Lowpass** (Hz) Lowpass cutoff knob for filtering the reverb tail. At 127 all frequencies pass.
LP GAIN	**Lowpass Gain** (dB) Sets how much of the lowpass signal to blend in.
HP	**Highpass** (Hz) Highpass cutoff knob for filtering the reverb tail. At 0 all frequencies pass.
HP GAIN	**Highpass Gain** (dB) Sets how much of the highpass signal to blend in.
BEAT SYNC	**Beat Sync** When on predelay is forced to be in eighth beats (1/8). When off it is stepless.
FILTER	**Filter** Enables reverb tail filtering. This has to be enabled for LP and HP to work.
MOD	**Modulation** Switches modulation on or off. This is very slow time varying delay modulation that smears out the reverb, making it softer.
STEREO	**Stereo** When on, two separate stereo tails are mixed, one for the left and the right input. In mono mode, only one stereo tail is generated using a mono version of the input.
REVERB TYPE	**Reverb Type** Chooses the algorithm to use for reverb simulation, from simple to complex. The reverbs are named according to they sound and properties: Hall, SmallRoom, Ambience, BigHall, Live, DrumBox, Cathedral and Megaverb.
ER	**Early Reflection** (dB) Sets the level in dB of the early reflections. These are very short spaced out echoes that appear immediately after the predelay, but before the reverb tail.

PREVIEW

PAGE MISSING IN PREVIEW

Folder

Effect
NEW IN V.2

The Folder is a classic Sine based waveshaper module. Technically, the Folder linearly frequency modulates a 0Hz Sine carrier oscillator with the input as modulation signal. Due to the curved nature of the sine wave, this means that the input is folded based on its amplitude. The *Folds* knob controls the drive of the folding, where one stage of folding ranges from almost no folding [-½π;+½π] to full folding [-π;+π]. Several folding stages can be further processed in series. The *Symmetry* knob adds or subtracts a DC offset to the input signal before folding, giving control over the symmetry of the folding process. Finally, the *Feedback* knob allows for both positive and negative feedback.

Parameters	
AMOUNT	**Amount** Sets how much of the folding signal you wish to output by adjusting the mix ratio between the incoming- and the outgoing-signal. 127 means full folding.
FOLDS	**Folds** Sets the amount of folding drive, ranging from almost no folding [-½π;+½π] to full folding [-π;+π].
SYMMETRY	**Symmetry** Adds a positive or negative DC offset to the input signal before folding.
FBACK	**Feedback** The output is fed back into the input, multiplied by this feedback factor.
SOFT / HARD	**Soft / Hard** Controls the overall drive of the folding. **Soft** scales the input by ½ before folding, and scales the output by 2 after folding. **Hard** does not rescale the input, but still scales the output by 2.
HQ	**High Quality** Enables high quality oversampling.
STAGES	**Stages** Sets the amount of folding stages in series (1-4). 1 is default.

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.

PREVIEW

PAGE MISSING IN PREVIEW

PitchShift

Effect

The PitchShift module changes the pitch of an incoming signal steplessly up or down one octave. This is an effect and thus not related to pitch changes in any of the generators or controllers, but works on all incoming signals. It does its magic by using two delay lines with time varying delays. The delays are filled at a fixed rate and read back at a variable rate, causing the pitch shift. To avoid discontinuities the two delay lines work out of sync, so when one increases its delay the other decreases its delay. The two delays are then amplitude modulated and mixed in order to sound like a continuous pitch shift.

Parameters	
SHIFT	**Shift** The amount of pitch shifting to apply. Centered (64) means no pitch shifting. 127 means up one octave, while 0 means down one octave.
DEPTH	**Depth** The amplitude modulation time, or how long to use the result from one delay line.
AMOUNT	**Amount** Sets how much of the pitchshifted signal you wish to hear. 0 means no effect. 127 means full pitchshifting.
FEEDBACK	**Feedback** Since this is a delay based effect, feedback can be applied. *Feedback* controls how much of the output signal to send back in to the pitchshifter.

PREVIEW

PAGE MISSING IN PREVIEW

PREVIEW

PAGE MISSING IN PREVIEW

Maximizer

Effect

The Maximizer module is a Sonic Maximizer emulation with Low Contour, Process and Peak Detector. The Sonic Maximizer is a classic sound enhancer effect, that does its magic by splitting the sound into multiple bands and phase offsets each band individually. This typically results in a boost in treble and bass. To counter the harsh nature of high frequency transients in the treble, the peak detector can attenuate peaks.

Parameters	
AMOUNT	**Amount** Sets how much of the maximizer signal you wish to output by adjusting the mix ratio between the incoming- and the outgoing-signal. 127 means full maximizer.
LOWC	**Low Contour** Sets the amount of phase corrected bass frequencies.
PROCESS	**Process** Sets the amount of phase corrected treble frequencies.
PEAK	**Peak Detector** Sets the amount of peak detection, to keep peaks in check. Higher values mean more peak detection and attenuation. A value of 0 means no peak detection.

Limiter

Effect
NEW IN V.2

The Limiter module is a simple soft-knee, peak limiter with optional stereo link option. A limiter is in many ways similar to a compressor, except that it offers no ratio and makeup controls. It is especially useful for ducking peaks that cross the limiting threshold, thus keeping them in check and making sure they don't clip. However, due to the synth's strict low latency requirements the Limiter offers no look-ahead option, which means that the output may still exceed the threshold limit, depending on the attack and release settings.

Parameters	
ATTACK	**Attack** (ms) The attack time for the envelope follower used to determine the amplitude level.
RELEASE	**Release** (ms) The release time for the envelope follower used to determine the amplitude level.
THRES	**Threshold** (-20dB to 0dB) Sets the threshold in dB where the limiter kicks in.
KNEE	**Knee** Sets the softness of the limiting. 0 means hard limiting, higher values soft limiting.
LINK	**Link L/R** Stereo link left and right channels. When on, the limiting in both channels is exactly the same, and based on the maximum peak of the left and right channels. When off the limiting is done separately for each channel.
MONITOR	**Monitor** Turns on gain reduction monitoring for this module instead of the standard waveform and frequency display.

PREVIEW

PAGE MISSING IN PREVIEW

PREVIEW

PAGE MISSING IN PREVIEW

LFO

Modulator

The LFO module is a simulated analog *Low Frequency Oscillator* that generates Sine, Triangle, Square, Saw and Random waveforms at frequencies up to 100Hz. A LFO is typically used for modulating other parameters, but in some cases can also be used as replacement for a standard low frequency generator.

Parameters	
RATE	**Rate** (0-100Hz) Sets the frequency of the oscillator.
AMOUNT	**Amount** Sets the maximum amplitude of the waveform, i.e., how "large" the waveform is.
FADE	**Fade** (0-100ms) Sets an optional fade-in time, in which the oscillator will slowly fade the amplitude from zero to max on trigger events.
PHASE	**Phase** Controls the phase of the oscillator, i.e., where in the waveform cycle are we ?.An optional **Random Poly Phase** option is available through the knob's right click menu, adding a randomized Phase Init offset when triggered polyphonically.
KYBD	**Keyboard Tracking** Sets the amount of keyboard tracking from –200% to 200%. This offsets the LFO frequency with respect to the incoming frequency control signal. Setting this to 100%, means that the LFO frequency will track the played note frequencies exactly with respect to the selected **Keyboard Tracking Base Key**. The **Base Key** can be setup through the knobs right click menu (default:C1).
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 oscillator signal. Useful for say outputting in the [0;1] range in unipolar mode, or to simply generate inverted waveforms.
TRIGGER	**Trigger** When ON the module restarts the waveform on MIDI Note-on events. The waveform is restarted from the currently set Phase value. Fade is also activated if Fade is set to anything above 0.
SYNC	**Sync** Synchronizes the oscillator rate to a whole number of beats. Useful for running modulation in sync to a songs tempo.
KYBD/FADE	**Kybd/Fade Toggle** Toggles between showing the **Kybd** and **Fade** parameter knobs.
WAVEFORM	**Waveform** Selects the desired oscillator waveform. Choose between: Sine, Triangle, Square, Saw, Random and Random16. Random16 has 16 step changes per period, unlike Random which only has one.

PREVIEW

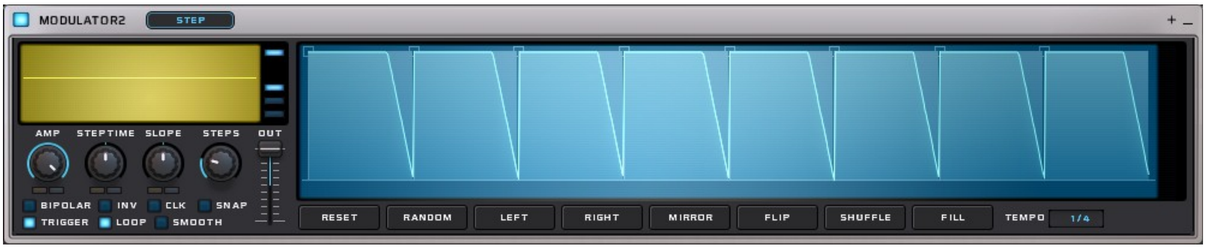
PAGE MISSING IN PREVIEW

PREVIEW

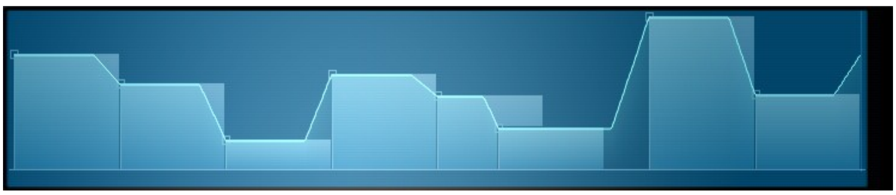
PAGE MISSING IN PREVIEW

Step

Modulator



The Step module is a programmable step sequencer, with 1 to 32 steps. Each step's length and decay slope can be adjusted individually. Each step's start position can also be adjusted in order to have steps start or stop off-beat. The global step-time as well as the global slope can be tweaked for all steps simultaneously. Finally the module features a tempo control and a number of useful buttons for editing and shuffling the step-sequence.

Parameters	
AMP	**Amplitude** (dB) Sets the maximum amplitude for the step sequence.
STEPTIME	**Steptime** (ms) Sets the steptime offset globally for all steps in the step-sequence.
SLOPE	**Slope** Offsets the decay slope settings globally for all steps in the step-sequence.
STEPS	**Steps** Adjusts the total number of steps in the step-sequence, from 1 to 32.
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 step output. Useful for outputting in the [0;1] range in unipolar mode, or to simply generate an inverted step-sequence.
CLK	**Click** Clicks are spikes caused by quick changes in amplitude. OFF means clicks will be ramped (default). ON disables ramping, leaving any clicks as they were.
SNAP	**Snap** If the step module is used to modulate a step based parameter (e.g. notepitch note and octave), enabling Snap will force the step output to snap to the nearest meaningful value for this parameter.
TRIGGER	**Trigger** When ON the step sequence is triggered on MIDI note-on events. When OFF the step sequencer runs continuously (free running).
LOOP	**Loop** When ON the step sequence will loop continuously. When OFF the step sequence will stop playing after a single run.
SMOOTH	**Smooth** When ON the step sequence will function more like a step based envelope, drawing a smooth curve instead of decaying steps: 

PREVIEW

PAGE MISSING IN PREVIEW

PREVIEW

PAGE MISSING IN PREVIEW

PREVIEW

PAGE MISSING IN PREVIEW

MidiTrig

Modulator

MidiTrig is a modulation module for receiving the common note and velocity MIDI triggers as a control signal.

Parameters	
NOTE	**Note Scale** Scales the incoming note value (0-127) by a common scale-factor.
VELOCITY	**Velocity Scale** Scales the incoming velocity value (0-127) by a common scale-factor.
MIX	**Mix Note/Velocity** Mixes the note and velocity signals. *Mix* sets the mix ratio. 127 means full *Velocity* signal. 0 means full *Note* signal.
AMOUNT	**Amount** Scales the resulting mixed signal.
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.
TRIGGER	**Trigger** When ON the module reads MIDI note/velocity data. When OFF the module does nothing.
RAMP	**Ramp** When ON the module attempts to reduce spikes in the signal by ramping them.

Shaper

Modulator
NEW IN V.2

Shaper takes an incoming signal and alters it by doing a simple lookup into a custom waveshaping function. The result is a wave-shaped modulation signal. The shaper-function is edited exactly like an envelope, using a set control points and curves to define the output signal. When using a linear ramp curve (default), the output is exactly the same as the input (disregarding any clipping at the ends of the curve due to overloaded input). Shaper is designed for low-frequency signals like LFO's, and does not perform any band-limiting. However, a smoothing parameter is available for lowpass filtering the resulting shaped output. The rightmost panel-switch enables a mode that shows the instantaneous lookup into the shaper-function.

Parameters	
SLOPE	**Slope** Offsets the slope settings globally for all curves in the waveshaping function.
BIAS	**Bias** Offsets the input [-1:+1] before look-up into the waveshaping function.
SMOOTH	**Smooth** Smooths the output using a first order lowpass filter. 0 means no smoothing/lowpass filtering.
AMOUNT	**Amount** Controls the dry/wet mix of the waveshaped signal with the input signal. 0 means 100% dry (input), while 127 means 100% wet (waveshaped).
BIPOLAR	**Bipolar** When ON (default) the input range of -1 to 1 is mapped to the lookup range. When OFF, the negative unipolar input range of -1 to 0 is mapped to the lookup range. This is useful for ADSR input for example.
CURVE	**Curve** Choose between **Linear**, **Exp(onential)**, **Cubic** or **Hermite**. All curves, except Linear, have adjustable slopes. Default curve is Cubic.

PREVIEW

PAGE MISSING IN PREVIEW

Output

The Output module is in charge of sending the resulting audio signal to the host application (or in case of subpatches, to the parent patch). However, all incoming data from the host (MIDI events) also passes through the Output module. This means that modules only receive this data if they are connected, at some point in the signal chain, to the output module. Only one instance of the Output module is allowed and it is always there (it cannot be removed). The module has Volume, Panning, DC Removal and Clipping parameters for tweaking the output before it is sent to the host.

Parameters	
VOLUME	**Volume** Amplifies or Reduces the volume of the incoming signal. Centered means no change in volume. The Volume knob is always directly linked to the output knob on the Control Panel. The **Volume Range** [+6dB (default), +12dB or +24dB] can be set through the right-click menu. There is also a **Headroom** setting available, where you can choose the synth's output's desired headroom. Choose between the default 85% (-16.5dB), 50% (-12dB), 25% (-6dB) and 0% (0dB). When the synth is used as an instrument it is recommended that this is left at the default setting to avoid channel clipping in the host, especially for polyphonic patches. When the synth is used as an effect however, the Headroom defaults to 0%. In this case, the input module will not have to take the headroom into account, meaning that levels / thresholds etc. inside the synth, match the actual input signal. Headroom is always 0% and hence disabled in subpatches.
PAN	**Pan** Pans a stereo signal left or right.
DC	**DC Removal** (0Hz[off] – 60Hz) Optionally removes very low frequencies from the output signal.
CLIP	**Clip** Optionally clips the signal to a specified number of dB before outputting. A setting of 127 means no clipping (OFF).

Index