Frohmager
User Manual

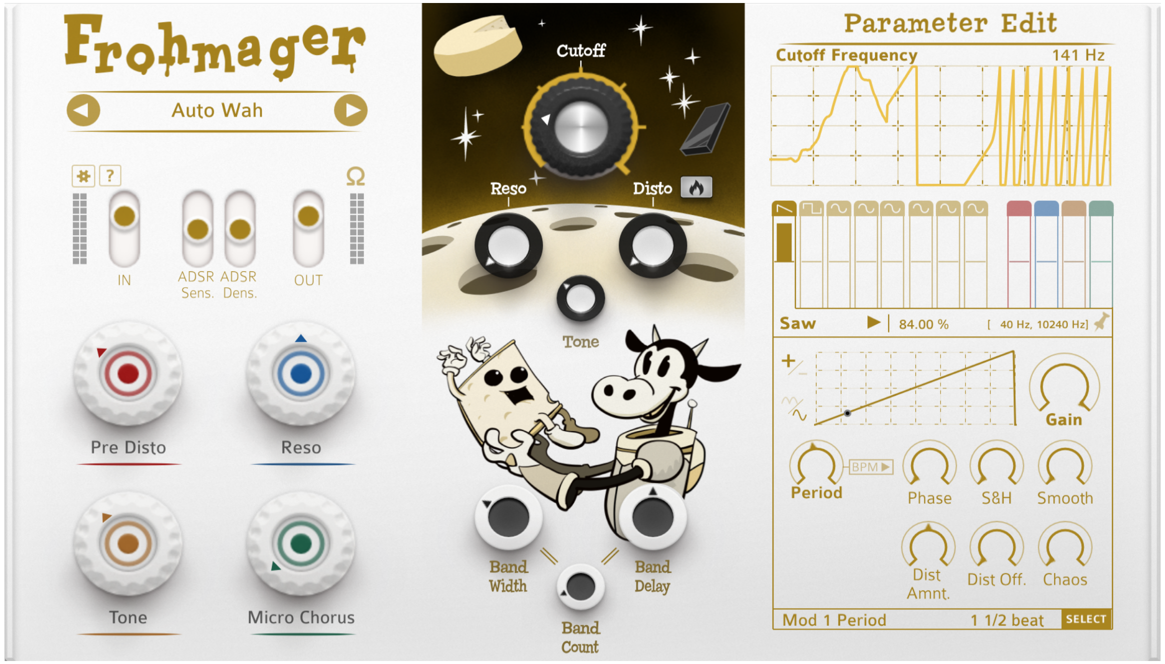


Table of content

The Quick Note (aka TL;DR)

Frohamager is a unique filter based on a Low Pass filter that also lets extra harmonics bands pass through then does all sort of unholy things to them, namely:

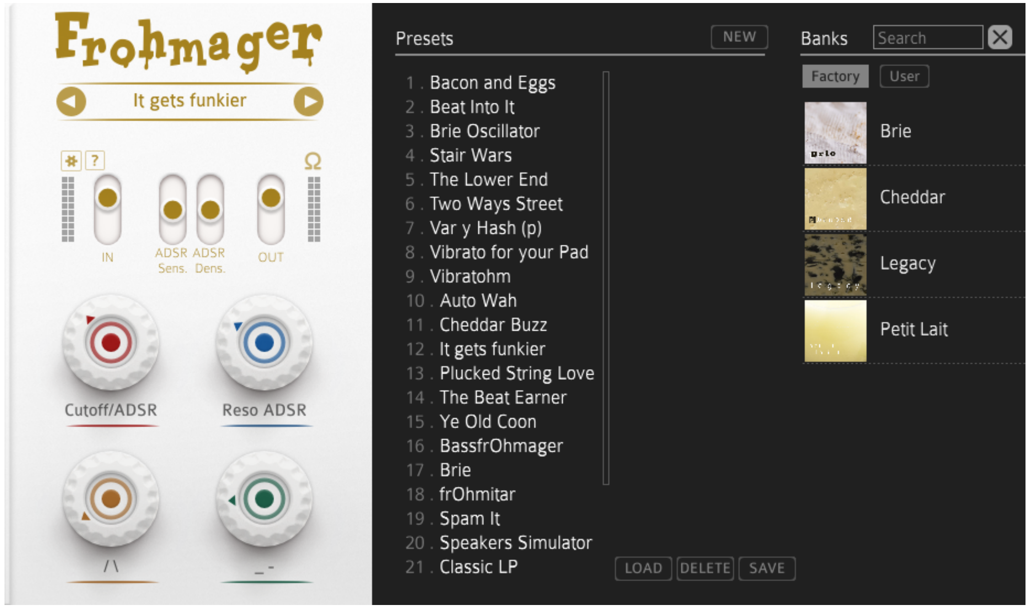
- distort them
- distribute resonance amongst them
- multiply them
- move them
- delay them
- modulate them
- also featuring an engaging preset browser with carefully designed presets

As a vehicle for your inspiration and creativity, it also brings some unique features to the table:

- Macros[2]: four knobs for live fun or quick production adjustment that gives you immediately the relevant controls for a patch with unprecedented accuracy and responsiveness. This is achieved by extensive control of ranges of assignation and response curves
- Modulate the modulation: create rich modulations by using one to modulate another
- Fine control over modulation range through min/max boxes allows you to stick to the beat easily

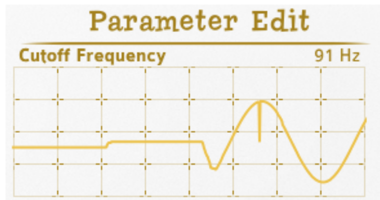
You obviously can fire up Frohmage and use it like any other plugin. That being said, here are a few recommendations that will significantly improve your experience:

- **adjust input and ADSR sensitivity&density once and for all** depending on what's your dry sound. Presets that you load will likely sound better that way. That being said, it's likely to sound fine with default settings.



- **Start by browsing presets through the browser.** They contain descriptions that will help you make the best use of them. **Always tweak the four** macros[2] to fully explore them

- like any other Ohm Force plugin **parameters are selected once you click them**. The selected parameter, as indicated in the main edit, is the one to which modulation in the right column will apply. If you want to select a modulation parameter in the main edit, use the special select push button bottom right.



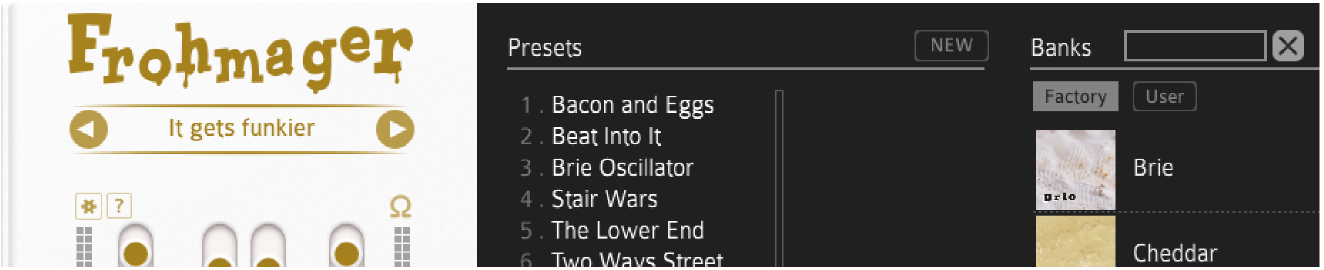
If you're familiar with the Frohmage ancestor or just enjoy discovering by experience you can leave the classroom.. The rest of this manual references the User Interface and basic usage guidelines. Checking it out on a "need-to-know" basis should be enough.

Presets

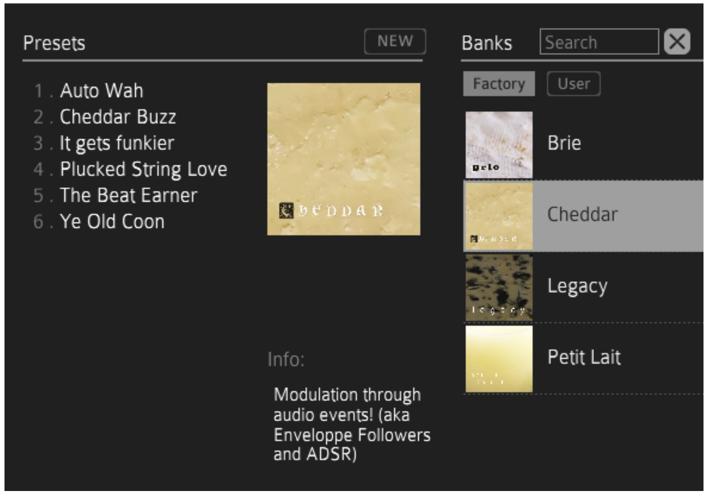
We took great care of the preset experience in Frohmager. In our view creativity dictates that plugins, no matter how complex, have to be able to accelerate your process when composing. To balance the amount of detail control our whole DSP UI strives to provide it was clear to us that presets should be fun and easy to browse, carefully organized, and offer a level of "cut to the chase" controls that would allow users to make them their own through finetuning and live play without having to understand the specific quirks of the patch.

Cue in the **macros**[2] & Preset browser team. While you can browser preset with the arrows, clicking the preset name will open the browser (and close it afterward, or you can use the X button)

Overview



By default presets will open with the factory list in alphabetical order. While this is fine to browse randomly or use the search box, browsing by banks will be more convenient once you're not just toying with Frohmager but actually using it in your projects.

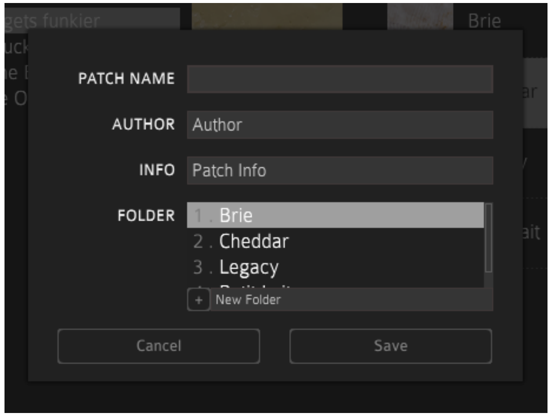


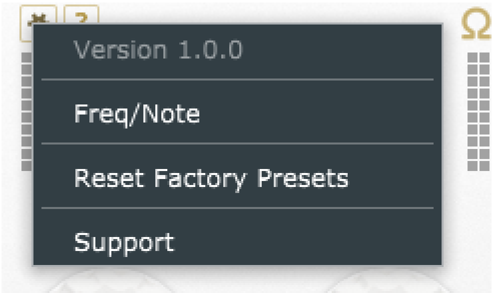
Banks are organized by type of need. They have an iconic picture to help you remember which is for what but also a description to tell you their role unambiguously. Some are focused on event-based modulations, other LFOs, etc.

While most of those can find a use outside of their original purpose it will save you time to know at least what to expect before browsing.

When loading a preset (double click or load) note that the **macros²** label and position change. It's time to play with those big knobs. Those are not just shortcuts to some smaller knobs in the DSP area. They're painstakingly programmed in fine detail and offer considerably more control over what a preset does than any single control in the DSP area. (see **macros²** programming to see how)

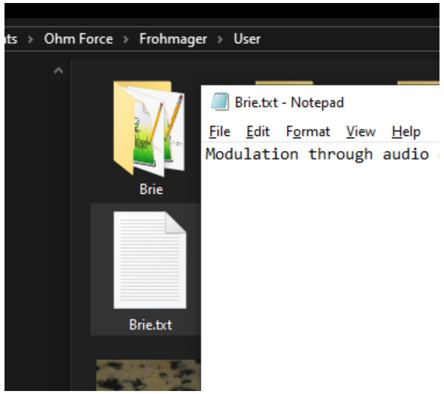
Creating your own presets





When saving your preset you'll have the opportunity to document it similarly via the info box. Note that you can also create your own bank. All of these will appear in the User tab on the right with a default picture

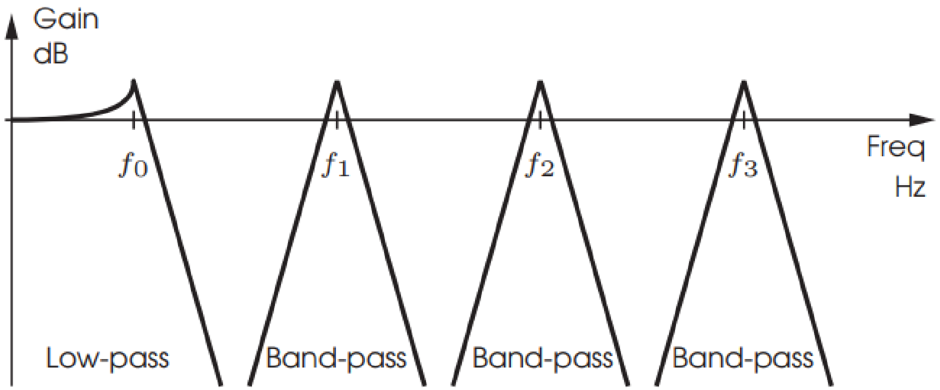
It will never be written over the factory presets. If those get damaged in any way you can always reset them through the option menu



You can also put your own image and description for the bank.
On Windows: Documents\Ohm Force\Frohmager\User
On MacOS: Library\audio\presets\Ohm Force\Frohmager\User

You'll find the presets (fxp files) in the folders of the banks they're filed in. If you create a .txt file with the same name as a folder, it will be displayed as the description for it. Similarly, if you put a 512*512 .png image with the same name it will become it will be displayed in the browser for that bank.

Audio Processing



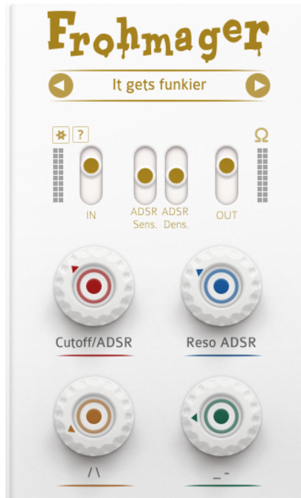
Frohmager is a multi-band resonant filter: it is built using a resonant low-pass filter running in parallel with multiple band-pass filters. These are then followed by a distortion stage. The low-pass filter (LPF) passes only those frequencies below a certain threshold point, known as the cutoff frequency. At the cutoff point there is a peak in the frequency response, known as the resonance. This emphasizes a very small frequency range, coloring the sound. Bandpass filters only allow small frequency ranges above the cutoff frequency to pass. All the bands are equally frequency-spaced, which increases the sound coloration. It is also possible to delay the sound of each band.

Time Units

A lot of audio processing in modulation is time-dependent and often perfect synchronization is needed. Frohmager lets you choose each time-related parameter if you want it to stay in sync with your project tempo and bars through the BPM mode. BPM mode expressed time in beats and fractions of beats. BPM mode includes all the musically relevant fractions in one go. You can try quickly a ⅓ of a beat delay compared to ½ beat one by just moving the knob.

If you rather want your value to be independent of the tempo and fixed, pick the ms (for millisecond) ones. That unit will be familiar if you've ever worked with guitar delays, chorus, etc.

Interface



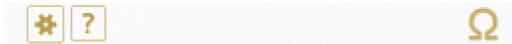
META COLUMN

This column regroups parameters that apply at a global level. They're typically adjusted depending on the incoming sound rather than what happens within the effect. For this reason, while their state is saved in your project, it is not saved in presets (except for **macros**[2]). These parameters are the only ones that can't be modulated.

Quick Preset Browser

This is where you browse your preset. Click the arrows to go to previous/next preset. Use save button to quickly save a preset in your custom ones (it will overwrite previous save with a similar name).

Preferences



This cog menu allows you to display cutoff frequency in Hz or note, Reset Factory Presets, or contact support.

Help

Open this very documentation

Update/Feedback

If your plugin needs an update, this button will display “Update” and allow you to directly download the latest version.

Ohm Force links

This menu allows you to quickly access product information

Meta settings

Left VU meter monitors the input signal, right VU meter the output. You typically want to have both using half or more but not all of their vertical space. The IN and OUT faders are the ones you need to use to that end.

ADSR Sensitivity and Density
ADSR is a modulation that will trigger when the plugin “thinks” the audio input had an even in it - the attack of a note or drum hit, typically. If it seems to you it triggers too much - or not enough, those are the two parameters to adjust.

ADSR Sens. lets you define the level of input volume at which you want ADSR to be triggered. Use the input VU meter on the left to adjust where you want it to kick in. Each time the VU meter goes above the ADSR Sens fader, it will trigger ducking. A classic setting is to have it happen each time something happens on the track. It can be very low on virtual instrument tracks, and pretty high on an acoustic Hi Hat track with a lot of background noise.

ADSR Density limits how much busy the detection can get. If you want the ADSR to chill a bit while you’re playing that heavy percussive track, this is the fader to lower.

Note that if you have just touched those two parameters, you still have no ADSR happening at all in your effect unless you have an ADSR modulating a parameter (see Modulation).

Macros^2

As you can guess from their size those buttons are designed to be a large part of your experience with Frohmager

Macros are “alias” controls used in many plugins and DAW to enable you to control one (or sometimes several) parameters within one single controller.

Macros^2 is our gimmick to describe the fact that we added an unprecedented level of control on your plugin, making **macros^2** that much more useful up - to the point they reach a deeper purpose.

- First, you control what range of your **macros^2** knob course affects what range of each of the mapped parameter(s). This allows you for instance to have the first quarter of a **macros^2** knob control the gain of the first tap from -∞ to 0dB (and not more), then the second quarter of the knob’s course, the gain of the second tap similarly and so on,... to end up with one simple, intuitive knob that adds up taps along its course without ever having one too loud.

- Second, each of those mappings can have its response curve. While the linear one is your go-to option, some situation calls for a curve that will make the parameter change slowly and then accelerate, or the opposite.

- Each factory preset comes with its own set of highly customized **macros^2** and you don’t know half of what a preset does if you don’t play with those four knobs after loading it. Their general purpose is documented in the preset description.

Double-click on a **macros^2** name to change it.

Some **macros^2** work with a neutral point at 12 AM, some at 8 AM, and some have no neutral point anyway. Again see presets descriptions for more clues.

Macros^2 values are stored in presets.

CENTRAL COLUMN: DIGITAL SIGNAL PROCESSING

Cutoff is the frequency at which your filter operates. This is the expression parameter by definition, and electronic music's most signature move is built upon tweaking or modulating that parameter.

Reso(nance) impacts the intensity of what happens at the cutoff value. A typical setting is a small, positive resonance but negative resonance can yield interesting results as well.
Resonance also impacts greatly how distortion will play out.
Resonance's typical use of modulation would be to create evolving textures in the sound.

Disto(rtion) controls the distortion amount. Distortion is created by softly clipping the signal.

Routing: the distortion will process the whole signal if this button is set to the left position, or the LPF output only if set to the right position.

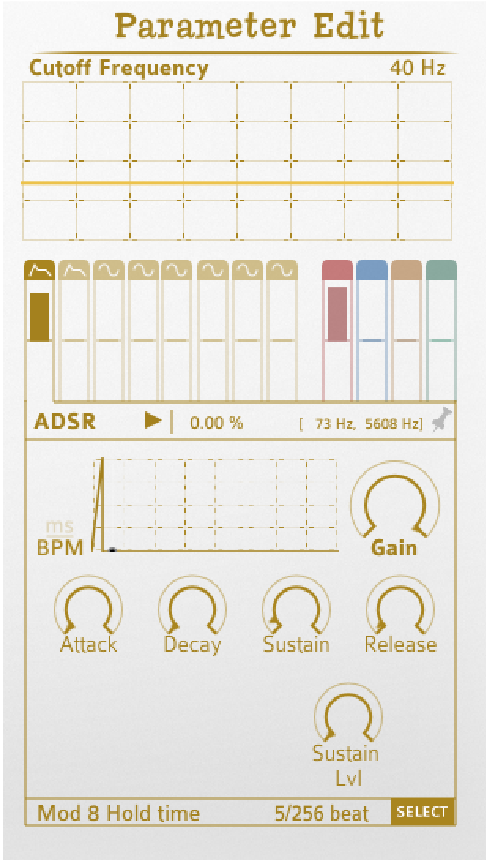
Tone: The filter output signal is divided into two parts: the LPF output and the BPF output. With the tone parameter, you can change the balance between the LPF and BPF sections. When the fader is set to the left, only the LPF output can be heard. When it is set to the right, the balance is reversed and only the BPF output can be heard.

Band Width defines the width of the gap between adjacent bands.

Band Count changes the number of bands, between 1 and 16. 1 means only the LPF is active.

Band Delay affects the delay times of all the bands. When it is in the middle position, there is no delay. When the knob is turned to the left, the lower bands are more delayed, and when the Knob is turned to the right, the higher bands are more delayed.

RIGHT COLUMN – PARAMETER EDIT AND MODULATION



Each time you click a parameter in the central column it gets selected (blue circle) which means the parameter edited and modulated in that column changes to that one.

This is where you can enter exact values for parameters by double-clicking that value in the first line.

The graph below shows you both the setting value of the parameter and its actual value in real time as affected by modulations and **macro²**. This is the same as the ring meter appearing in the knobs but in a way that's easier to monitor over time.

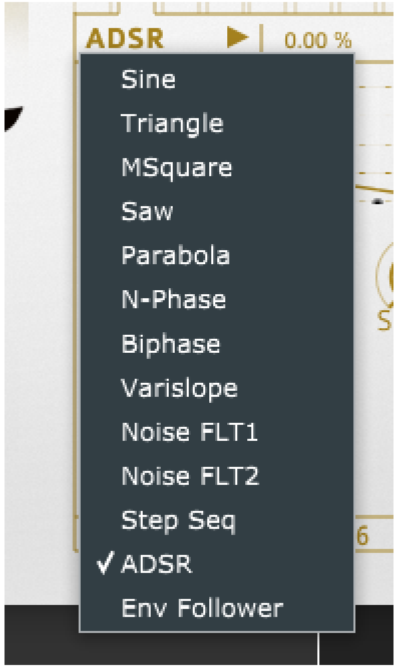
Below is the modulation matrix for that parameter. That's where you add modulations and **macro²** modifiers to that parameter. Each fader also works as a tab selector for that modulator.

A single parameter can have up to eight modulators at a time plus four macros (although you're very unlikely to need nearly as much). **Each modulator can be applied with a factor from 100% to -100%, with 100% being the whole range of the parameter**. So keep in mind a 50% modulation of a pan that's set as centered will modulate it exactly from full left to full right. Higher values will “saturate” the modulation resulting in a good deal of its course capped full left then full right, with no movement while there.

Modulation area

Modulation Type

This menu lets you pick what type of modulator is operating. Some share similar parameters, some are unique.



Most of those are Low-Frequency Oscillators (LFO) - curve-based movements applied to the parameter that while sometimes fast for human ears remains slow compared to the same pattern used as oscillators for synths.

Sine and triangle are typically used for regular back-and-forth movements.

Saw induces a more aggressive pattern moving progressively and then instantly coming back.

Square is typically used a a switcher for two values for a parameter.

Parabola is also a back-and-forth movement where the movement around peak value is very progressive while the movement around minimal value is very fast. Were you to stand next to a paddle ball player with a noisy ball, the ball’s noise pan would follow such a modulation.

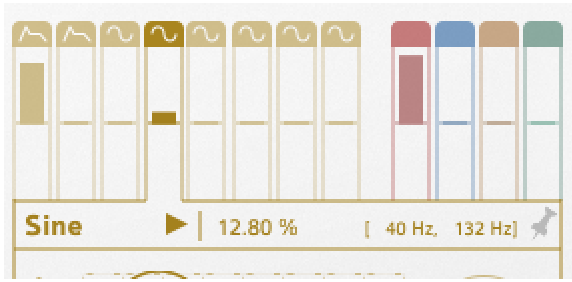
Bi-phase and N-Phase create a movement based on the constant switching between two or more sine. It creates a smooth move animated with regular jumps that can be synched to the rhythm or not.

Varislope can morph between square, triangle, and saw. Modulating this one's parameter can result in very rich evolving modulation.

Noise FLT1&2 are two noise generators that will create two very distinct types of random moves to your signal. FLT1 has the wildest move while FLT2 is always progressive, like a random trajectory. The period for those works differently from other modulators as they don't repeat a pattern and change constantly, but you're guaranteed to go back to the initial value on the period. The period will also affect how quickly their variations happen.

In addition to LFOs, there are also Step Sequencer, ADSR, and Envelope Follower

Values



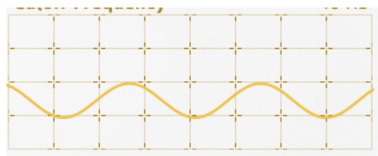
The first number is how much modulation is applied from -100 to +100, with 0 being an inactive modulator attribution. You can change that value with the modulation vertical fader or directly with your numeric keys by double-clicking that figure.
The min and max figures indicate what minimal and maximal value the resulting modulation of the parameter will reach, depending on the above modulation amount but also the current value of the parameter. A handy option is to edit those values by double-clicking them. Frohmager will automatically change the modulation amount and parameter value so that it can stay true to those minimal and maximal values. It's especially useful when you want a parameter to jump between two specific time or pitch-related values.

Sign

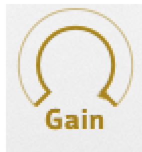
Switch between a positive and negative polarity, effectively inverting the curve. Useful to decide whether the lowest part starts or ends the cycle

Polarity
Switch between unipolar where all the modulation happens above (or below, depending on the sign) the modulated parameter, and bipolar, where the parameter is at the middle value of the modulation.

Modulator Shape
This graph simulates the LFO loop depending on the settings. The moving dot follows what part of the LFO is currently happening.



Gain
While this may look similar to what modulation amount does, changing the amount of modulation here applies that change to the modulator itself, hence impacting all of the parameters modulated at once. Moreover, this one can be selected and modulated.

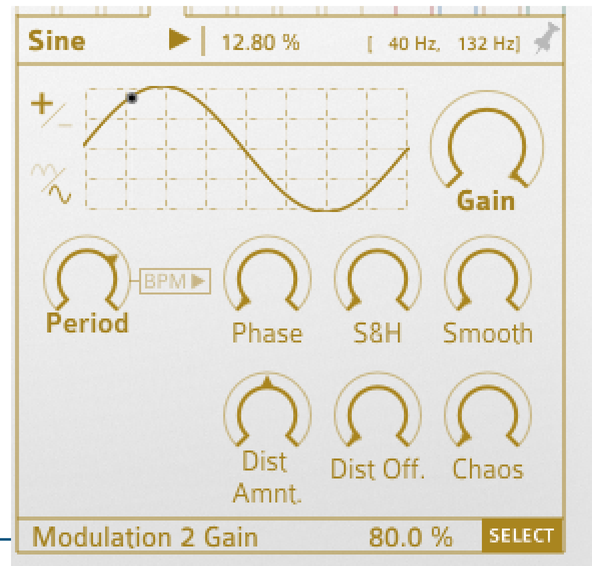


Period defines the period of the modulation. Typically set in BPM. A classic target for modulations.

Phase lets you define the origin of the oscillator's loop. For instance, if you have a sine you may want to have it start at the maximum peak, the minimum one, or any of the two points where it crosses zero. That is where you'll do that.
Also great potential for modulation

S&H (sample and hold) transforms your base curve into a "pixelated" one, which turns any smooth movement into a series of jumpy ones.

Smooth does the opposite, softening every move. Both are often used together.



Distortion Amount controls the symmetry of your modulation pattern. At default value (50%) your curve is split between equal positive and negative phases. At 25% the first is now three times shorter than the later (25 to 75 ratio).

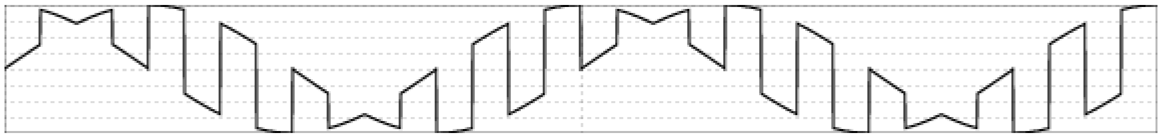
Distortion Phase changes the location of the short and the long phase during the loop. Does nothing if the Distortion amount is at 50%

Chaos adds more randomness to the play head movement on the curve. It auto-compensates so that the time parameter for the loop is always respected but adds some random accelerations and decelerations at a rate that depends on how much chaos you ask for.

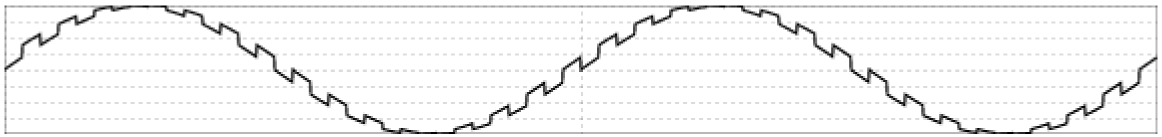
Time (Bi and Nphase)
Control the period of a back and forth between the phase composing the final LFO

Shape (Bi and Nphase)
Control how much delay between the different curves composing the final LFO

Some examples:
Biphase:

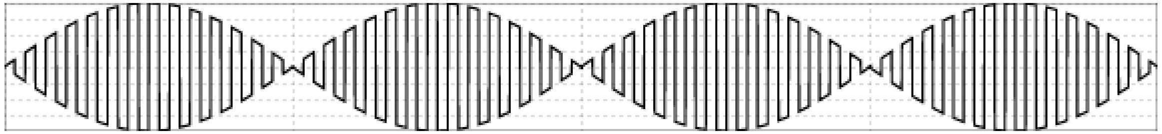


Time: 50% Shape: 25%

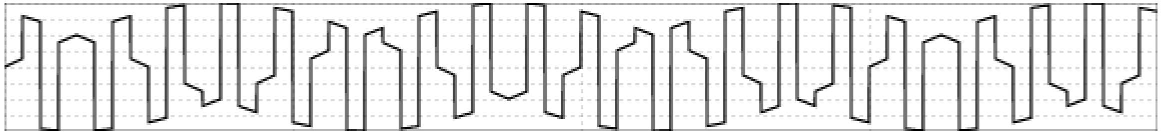


Time: 75% Shape: 3%

N-Phase

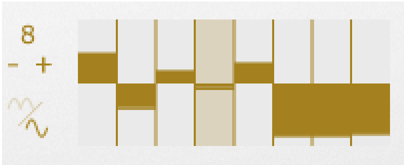


Time: 62.5% Shape : 0%



Time: 50% Shape: 12.5%

Step Sequencer



Divisions let you pick how much division you want on your sequence, which factored with your time setting will define the duration of each of those divisions. For instance, a time value of 16 beats with 32 divisions will mean that each division in the graph lasts half a beat.

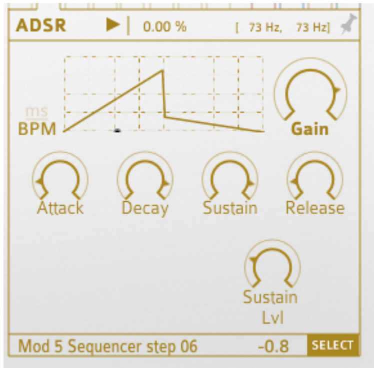
Simply hold-click on the graph to set each division's value. Believe it or not but each of those can be modulated as well…

Polarity
Switch between unipolar where all the steps add to the modulated parameter, and bipolar, where the parameter is at the middle value and steps can add or subtract from it.

ADSR

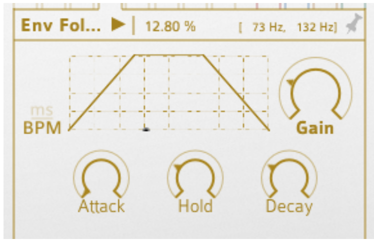
ADSR is an envelope that works like sounds work in reality - an Attack and a Decay describing the start of the sound - percussive or instead eerie like a flute. Sustain, Sustain Level and Release then define how the sound lasts and how it ends.

ADSR is commonly used on filter cutoff in synth but here you can apply it on all parameters.



Envelope Follower

That modulator follows the envelope of the input signal. If Attack, Hold, and Decay are at zero, it follows it exactly which in any eventful sound will result in jumpy modulation. Attack, Hold and Decay will make for a more musical/pleasant result by delaying and smoothing the modulation. A typical use of EF can be seen in Auto Wahs.



Modulator edit

This area behaves like the parameter edit except only for modulation parameters. Select those by clicking them then edit their value by double-clicking it in the box. Additionally, the select button will select the parameter in the main edit which will allow in turn to attribute it its own modulation sources.

Macro² settings

The last four colored faders/tabs of the matrix let you map the selected parameter to the **macro²** of your liking. One parameter can be used in several **macro²** at once. This can have some use although most of the time you'll want them on one controller only. More commonly, one **macro²** can be mapped to as many parameters as you want. As you hopefully guessed the color here matches the one of the **macro²** controler affected in the left column.

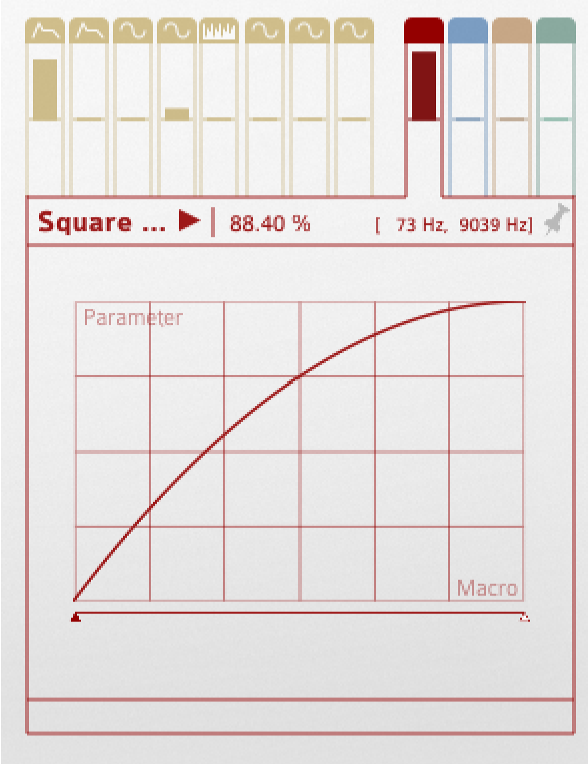
The mapping amount is defined exactly like the modulators - from -100 to 100% of the parameter's full range, with the same caveat that you can hit the parameter's range cap easily. Negative mapping will actually lower the parameter as you increase its range. The min/max value works the same way too.

The shape menu lets you use several response curves for the mapping.

Linear is the most common use and will have the same response on the **macro²** knob as on the mapped parameter

All the others are parabolics. **Square** (and its amplified version, **Cubic**) will move the parameter subtly first and dramatically at the end of the mapped area. **Square Inverse** and its amplified version, **Cubic Inverse**, do the opposite. The extra layer of control those response curves provide comes in handy when dealing with sensitive parameters regarding gain like resonance, distortion amount, or feedback, or when you want some transformation to become very quickly noticeable while retaining some room for accentuation.

Use the two triangles around the red knob scheme to define the range of the knob that will modify the selected parameter value. **macro²** knob at min range value or below will let the



parameter be unmodified. Max range and above will stick on the parameter setting multiplied by affectation.

If you click the parameter name in this **macro²** area a menu will let you select any other parameter already affected by that **macro²**.