boom LIBRARY

ENRAGE

MANUAL

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1. PREFACE

WHAT IS ENRAGE AND WHAT MAKES IT SPECIAL?
ENRAGE is a modular multi effect plug-in. It offers the possibility to stack up to 6 parallel and (simultaneously) 8 serial processing devices into processing chains. Combined with very precise modulators, flexible routing capabilities and high-quality processing devices you can be very creative in designing highly innovative effects or mastering grade chains.

1.1 Minimum System Requirements

WINDOWS: Windows 8 (64-bit), 8 GB RAM, Intel® Core™ i5 (relatively recent)
MAC: macOS 10.13 (64-bit), 8 GB RAM, Apple Silicon or Intel® Core™ i5 (relatively recent), OpenGL 2.1 capable GPU

1.2 Installation

After downloading and opening the installer of ENRAGE please follow the on-screen instructions to install the application.

THE MANUAL WILL BE COPIED HERE:
Windows: C:\Program Files\BOOM Interactive\Enrage
Mac: /Applications/BOOM Interactive/Enrage

USER PRESET BANKS WILL BE LOCATED IN SUBFOLDERS HERE:
WINDOWS: C:\Users\[user name]\AppData\Roaming\BOOM Interactive\Enrage\Presets
MAC: /Users/[user name]/Library/Application Support/BOOM Interactive/Enrage/Presets

1.3 iLok Registration

During the first start of ENRAGE after installation, the iLok registration window pops up. ENRAGE is licensed using the PACE Licensing Platform. You need to have an iLok account to use it, however setting up an iLok account is free. You will find all necessary information on how to setup your account on www.ilok.com. You can either directly authorize your computer (machine authorization) or use a 2nd or higher generation iLok hardware dongle. A second or third generation iLok (iLok2 or iLok3) is a product of PACE that can be purchased at www.ilok.com or from any participating music retailer.

Next: please download the iLok License Manager at www.ilok.com.

Upon the first launch of the application, you will be asked to register ENRAGE with your iLok account.
A free iLok account can be created under www.ilok.com

After your purchase, you automatically receive an order confirmation from us containing the download link for the installer plus a 30 digits long iLok activation code (e.g. 1234-1234-1234-1234-1234-1234-1234-12).

HOW TO ACTIVATE THE LICENSE

- Open the iLok License Manager application.
- Either select the menu: Licenses -> Redeem Activation Code or click on the small Redeem Activation Code Icon on the upper right of the application.
- You should then copy paste the entire code you received from us into the entry form. Select your iLok as the activation location to immediately activate the license on this iLok and confirm the location.

Now you are ready to go!

2. QUICK START

Once ENRAGE has been instantiated, the most basic setup loads per default. At the top center you will find a grey field labeled **"Untitled"**. Clicking on it opens the preset browser. Browse through the categories to load presets you desire and try the MACROS or MOD SOURCES to dial it in. Done with the Quick Start – now onto the real fun…

Watch a step-by-step video tutorial about how to activate your iLok license:

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The product doesn't show up in your DAW?
Not activating the software when first launching the DAW can lead to DAWs putting the plug-in into a "failed to scan" blocklist or blacklist.

In such cases it is usually sufficient to:
• check the DAW's blacklist/blocklist
• remove the plugin from that list
• activate it in the iLok License Manager
• cause a plug-in rescan or restart your DAW

3. GUI OVERVIEW

ENRAGE is a highly complex, professional tool with a rather complex GUI. However, we tried to keep it as user friendly as possible.

3.1 General Overview

You will find a header and footer as well as four additional, color coded segments: **CONTROL**, **MOD SOURCES**, **DEVICES** (including the Device List, the Rack, the Mod Matrix and the Device Editor) and **MACROS**.

For a better overview of the features and functions, we use the following ENRAGE GUI color code in this manual:

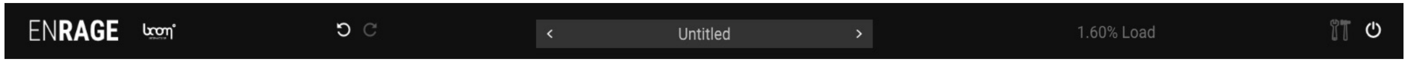
CONTROL

MOD SOURCES

DEVICES

MACROS

3.2 Header



At the very top, going from left to right you will find some general functions.

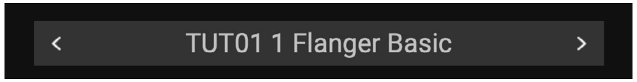
3.2.1 Title and Logo

Clicking on the Title and Logo shows the version of ENRAGE currently running as well as the Credits and contact information.

3.2.2 Undo and Redo

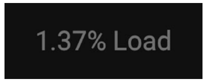
The little circling arrows are undo- (arrow to the left) and redo steps (arrow to the right) for all changes with the exception of macro automation. The maximum number of available undo steps may vary.

3.2.3 Preset



In the center of the header, you will find the name of the currently loaded preset. Clicking on it will bring you to the Preset Browser (3.7)

3.2.4 CPU Load



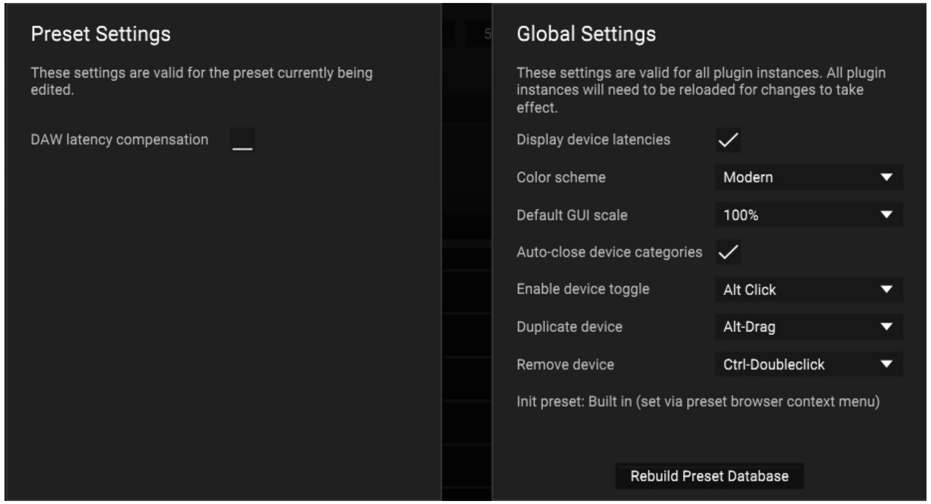
To the right of the preset name, you will find a single CPU load meter to check how intense the preset is for your DAW. Note that this is merely an estimate, it does not take all cores into account.

3.2.5 Manual

The question mark button to the right of the CPU load meter will open the Enrage Manual.pdf file directly from the Plugin UI. This of course only works if the manual has been installed correctly during the Enrage installation process.

3.2.6 Settings

The Settings button is located between the Manual and the Bypass button. Clicking on it will bring up two distinct setting windows.



3.2.6.1 Global Settings

Within this window you can make general adjustments that affect all instances of this plugin such as **Color scheme** (Modern - Classic - Colorblind) and usability optimizations.

By clicking on the **Rebuild Preset Database** button on the bottom, ENRAGE will scan all Preset Directories and update your Preset Browser. **Display device Latency** will reveal the Latency of the currently selected device at the upper right corner of the Device Editor. The **Default GUI scale** can be decreased to 80% to match a rather small screen or increased up to 200% in 25% steps. You can also change the default modifier keys for the following actions: **Enable device toggle**, **Duplicate device** and **Remove device.**

3.2.6.2 Preset Settings

ENRAGE generally does not introduce latency. Only FFT based processing devices such as **Pitch Shift, Vocoder, Splash, Warp** & **Timbre** introduce latency natively deriving from the principles of FFT processing. Due to the complexity of ENRAGE and the possibility to create internal loops and wild combinations of parallel and serial processing, ENRAGE cannot guess what exactly needs to, or should be compensated. But this can manually be dealt with.

The basic idea is, the longest introduced combined latency of one processing chain needs to be compensated.

EXAMPLE: STEREO IN -> PITCH SHIFTER -> STEREO OUT

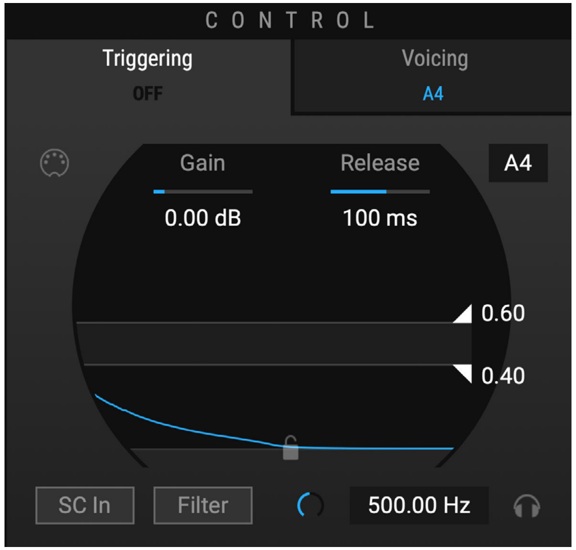
- Depending on the selected Mode in Pitch Shifter the latency for this device will be 24 ms or 48 ms. Now all you need to do is insert this value as your preset bound DAW latency compensation. For six **Pitch Shift** devices in a serial chain there would be an overall latency of 6 x 48 ms for this preset. A parallel setup on the other hand would result in 48 ms only.
- To report this latency to the DAW, activate **DAW latency compensation** and insert the latency of your preset. Head over to the Global Settings to the right and activate **Display device latencies**. Now every device introducing latency will display the exact amount in the top right corner of the Device Editor of the currently selected device.

| **TIP:** This might come in handy if you want to incorporate lookahead mechanics in your designs.

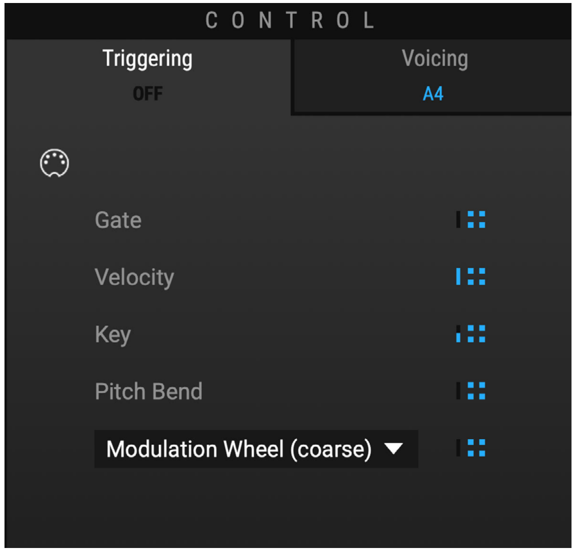
3.2.7 Bypass

The icon on the far right within the header is the global bypass button for ENRAGE which bypasses everything except for the global output gain: “Out Gain”.

3.3 CONTROL



On the upper left you will find the blue area, **CONTROL**. It controls any triggered modulation, such as **Curve**, **ADSR** or the restart of **LFO**. You can trigger via audio (active by default) or MIDI (hit the little icon on the upper left within the **CONTROL** section and potentially set midi voicings in the **Voicing** tab, also within this section). **Voicing** is always set to mono until you manually select **Poly**. In this case ENRAGE will behave like a polyphonic synthesizer and the trigger-envelope or the MIDI signal will not only trigger envelopes but a whole voice. The **Poly** mode also offers a global Voice-Envelope (AHR) and an optional **Auto-Bypass**.

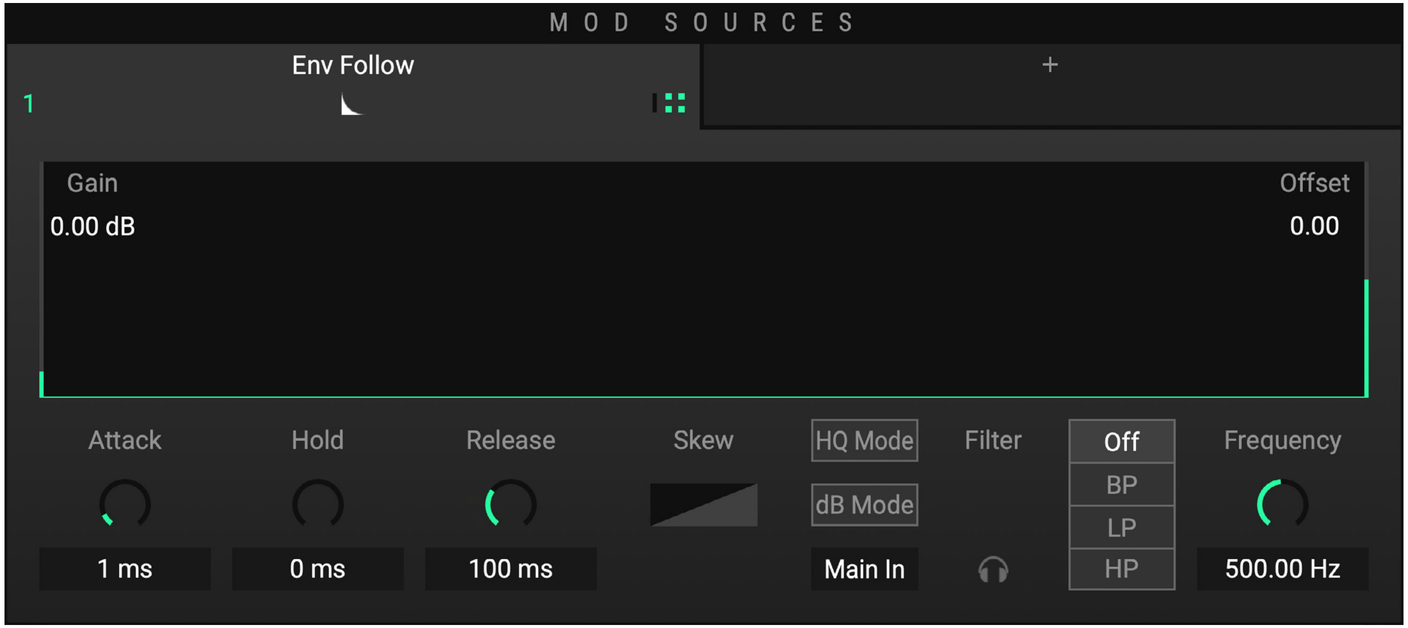


The **CONTROL** section offers several modulation sources as well. You can simply apply any given parameter the same way you would apply a modulation value from the MOD SOURCES section. You will find a detailed explanation on how to do so in the following chapter. Gate will output either 0 or 1 depending on the trigger state. In MIDI mode this will represent the note on/off state. You can also use information deriving from Velocity, Key, Pitch Bend or your Modulation Wheel / Foot Pedal / etc. Once you have applied these modulation values you can still switch back to audio-triggering and make full use of all the options offered within this section.

TIP: Audio Triggering
Audio triggering is configured exactly like it is in Enforcer:

WATCH TUTORIAL

3.4 MOD SOURCES



Right next to **CONTROL** you will find the green section: **MOD SOURCES** (Modulation Sources). They can be routed to alter parameters inside of ENRAGE. An **Envelope follower** will be loaded by default. There are a total of ten different **MOD SOURCES** which are explained in detail in chapter 5. They all work in the same way concerning how to modulate any **DEVICE** parameter. Some of them even have the possibility to be modulated themselves. You can drag the **Env Follow** icon (::) onto an **LFO** **Rate**, just to give one example. This will increase the speed based on the input loudness. There are some advanced and special ones, namely **Pitch Tracker**, **Change** and **Formula**. If you know what they are and how to use them, feel free to do so. If you do not, maybe stay away from them for a start.

HOW TO APPLY A MOD SOURCE TO A PARAMETER

- Select a device in the **rack** to which you want to apply a **MOD SOURCE**.
- Now drag the little green cubic Mod Assignment icon (⠶) on the lower right of the **MOD SOURCE** onto any black circle ● right below the **parameter** you want to modulate.
- Now drag the little green **1** downwards to set the modulation amount. If you do not want the modulation amount to exceed the possible maximum as determined by the knob position, just press and hold shift whilst dragging.

3.5 DEVICES

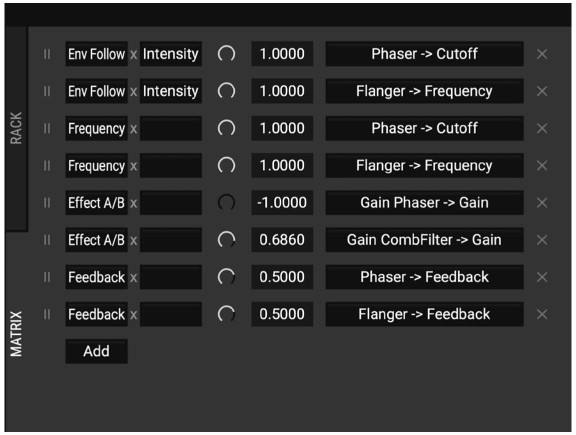
DEVICES are color coded in yellow. The complete lower half of the GUI is dedicated to this section basically. On the far left you will find the **DEVICE Browser**. ENRAGE offers a total of 50 different **DEVICES**. Once you made your choice, just drag it onto the Rack right next to it.

3.5.1 The Rack

The Rack will serve as your signal chain building ground. This is where you place **DEVICES** and organize the routing. You can stretch devices to the left or to the right. This way you can control whether you want your signals to be mixed or processed separately. Combining two parallel streams might result in highly overloaded outputs, so be aware of the signal flow.

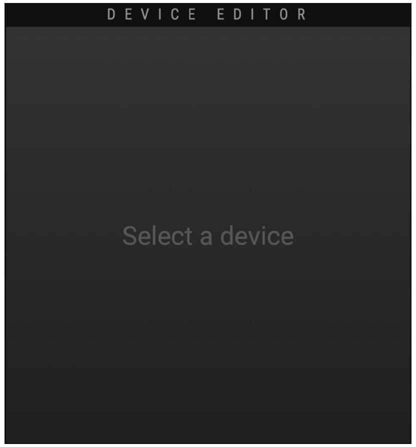
TIP: Routing
You will always need a **Stereo In** and **Stereo Out** in order to actually route audio through the device chain and send it back to your DAW.

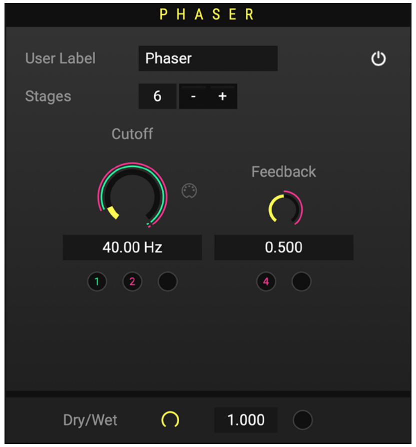
3.5.2 The Matrix

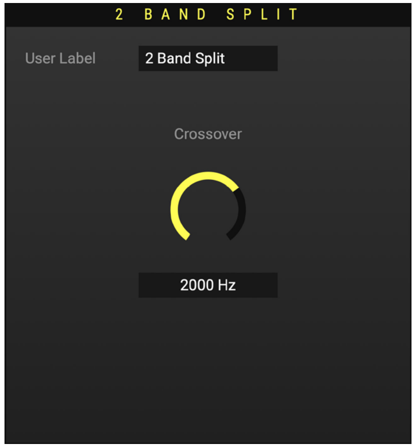


In between the **DEVICE Browser** and the Rack, you can switch to the Modulation Matrix. It shows all assigned modulations. Here you can also add multipliers, change values, remove, or add modulations. To assign modulation values or multipliers you can simply click on the empty button space or directly drag & drop the **MOD SOURCE** or **MACRO** of your choice.

3.5.3 Device Editor

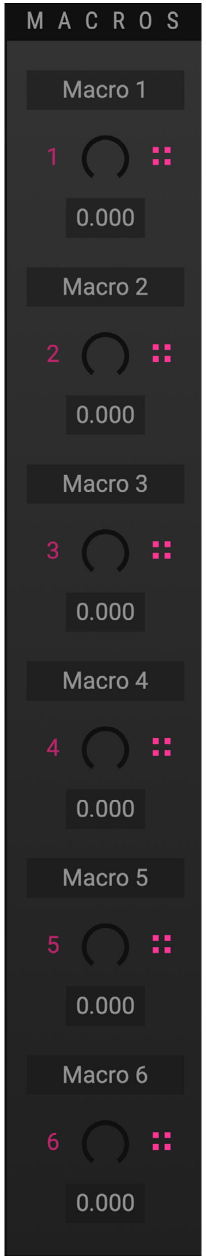






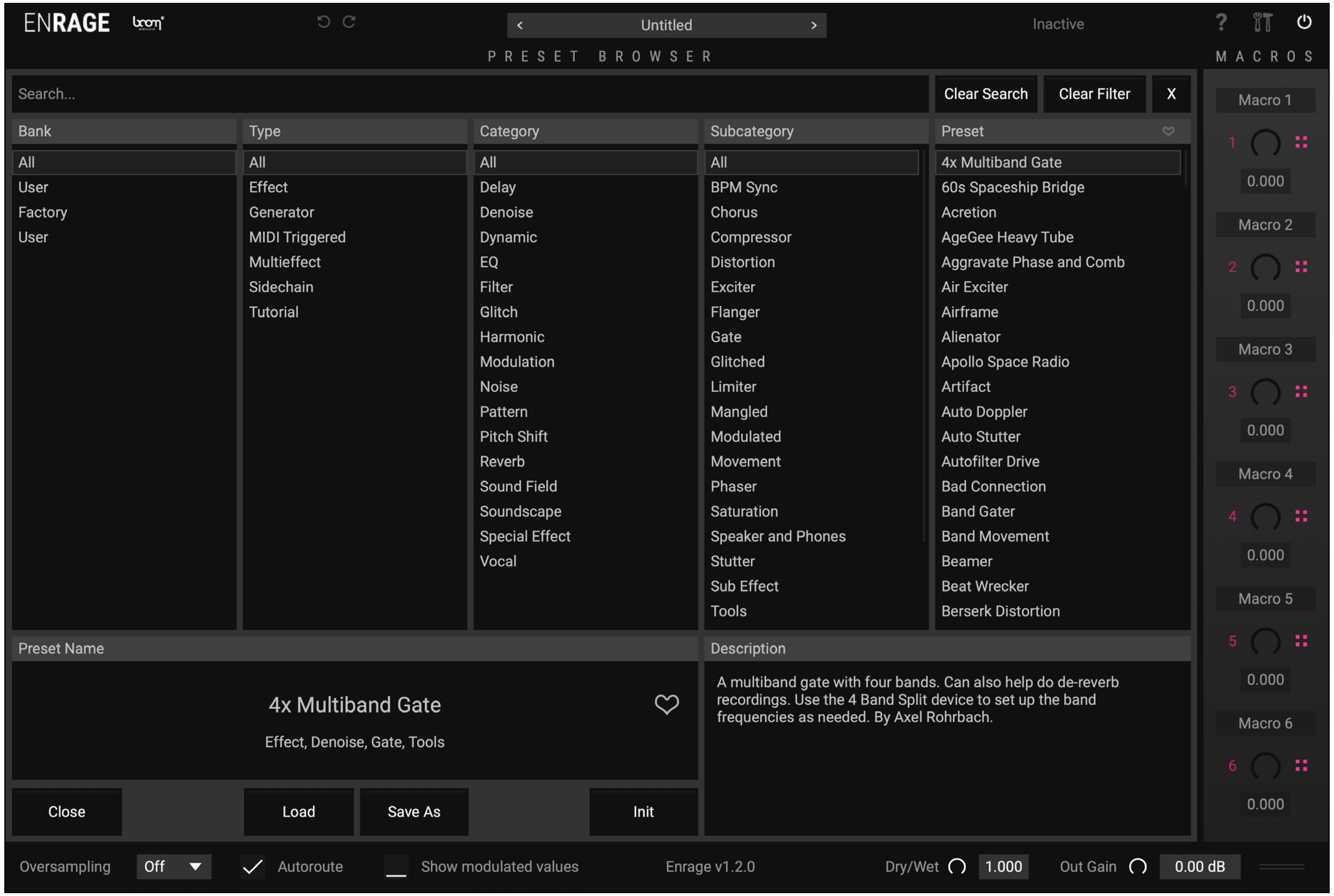
Right next to the Rack you will find the parameters and settings of the currently selected **DEVICE**. There might be drop down menus, knobs, curves, numbers and or buttons, depending on the functionality of the **DEVICE** itself.

3.6 MACROS



One of the most important parts are the **MACROS**, colored red. Whenever you see the little black circles you can connect a **MOD SOURCE** to, you can also apply a **MACRO**. They can be automated by your DAW, everything else within ENRAGE cannot be automated. Their purpose is to offer you a fast and simple way to control the overall template setup.

3.7 PRESET BROWSER



On the very top of the GUI you will find a field labelled “Untitled”. If you click on it, the Preset Browser will open. It should rather be self-explanatory. First you will find a search bar right below the PRESET BROWSER title. All presets are then filtered in five different columns: **Bank, Type, Category, Subcategory and Preset**.

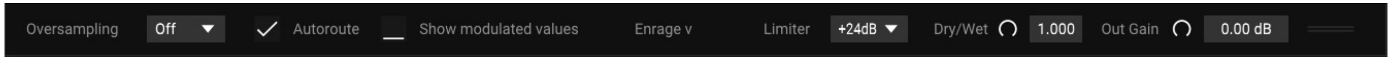
When clicking on a preset of your choice, it will show the **Preset Name** on the bottom left corner as well as **Category** and **Subcategory** classifications. You then have 4 buttons to proceed with. **Close** will minimize the browser and bring you back to the main GUI. **Load** will load up the selected preset and by clicking **Save As** you can store your own presets in the User Bank. Above the **Init** button you will find a small heart icon. By clicking on it you will mark the selected preset which will help you to quickly remember your favorite templates. At the bottom right of the browser you will find a short **Description** on how to use certain presets paired with general information and potential use cases.

TIP: Initial preset
The Init button will load the initial preset labeled „untitled“. You can specify your own init preset that will load in new Enrage instances by right clicking on a preset name and clicking **Set as init preset**. Revert to the built in init preset in the Global Settings.

TIP: Updating your preset database
If you wish to update your database, head over to the global settings window by clicking the settings icon on the top right corner of ENRAGE.

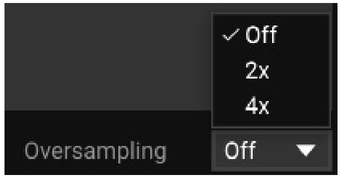
TIP: Sorting by favorites
You can then sort the preset list to show favourited presets first by clicking the "heart" toggle at the top right of the name column.

3.8 Footer



The Footer is located at the bottom of the GUI. Here you will find additional Options affecting a variety of sections. This chapter is structured according to the Footer from left to right.

3.8.1 Oversampling



On the very left you can switch on internal **Oversampling** at a rate of **2x** or **4x**.

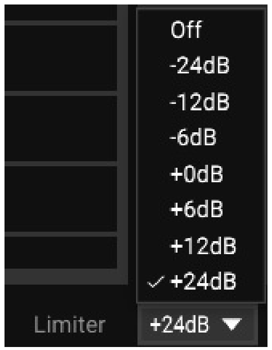
3.8.2 Autoroute

Next you will find an option affecting the Rack. By enabling **Autoroute**, ENRAGE will try to arrange **DEVICES** you place or move within the Rack automatically.

3.8.3 Show modulated values

When enabled, ENRAGE will display the modulated values in real-time, based on active modulation. When disabled, all value labels will display the knob positions before modulation gets applied instead.

3.8.4 Limiter



To the right of the ENRAGE Version currently running, you will find a global output **Limiter** that ranges from -24dB to +24dB and will be applied after the global **Out Gain**. The **Limiter** will be highlighted every time it gets triggered. This will help you keep an eye on the extent of your output that runs into the **Limiter**.

TIP: Protect your ears & speakers
The output limiter can be a handy tool if you experiment with constructions that could temporarily result in high output gain. For example, when setting up and configuring audio feedback paths. To protect your speakers and your ears in such cases, enabling the output limiter temporarily can be helpful.

3.8.5 Dry/Wet

Next you have a global **Dry/Wet** knob. Be aware that it can be automated within your DAW, but it cannot be modulated within ENRAGE. You can of course achieve something alike by building it yourself in the Rack. If a preset has DAW latency compensation enabled in the preset settings, the dry signal will be delayed by the configured amount.

3.8.6 Out Gain

The global output gain is located at the very right of the Footer. Just like the **DRY/WET** the Out Gain can also be automated within your DAW but not be modulated within ENRAGE. To do so, simply modulate a **Gain DEVICE** instead.

4. DEVICE LIST

4.1 IN/OUT:

4.1.1 Stereo In

Routes the global ENRAGE input into the Rack.

4.1.2 Stereo Out

Routes the Rack audio to the global output of ENRAGE.

4.1.3 Sidechain In

If you have a sidechain setup in your DAW going into ENRAGE, you can route this Sidechain track into the Rack. This might be useful for feeding ENRAGE with different carrier and modulator audio streams in case you want to build a Vocoder for example.

4.1.4 Tap Send

Grabs the audio stream at the inserted position to feed it back to the Mod Sources that process audio input. You can also reinject the signal within the rack at another position. Enrage has four internal buses, called Tap 1 .. 4, which you can select in the device editor.

4.1.5 Tap Receive

Reinsert the signal sent by a **Tap Send DEVICE** back into the Rack. This allows you to build longer processing chains and construct feedback loops. Be careful when doing that!

4.2 Gain

4.2.1 Gain Lin

A simple gain device that lowers or increases gain. You can choose between **Attenuate** or **Boost** in the dropdown menu. The **Turbo Modulation** button removes smoothing. Sometimes the **Turbo Modulation** is needed but be aware that this might introduce clicks depending on the situation. Finally, you can switch the phase of the input signal inside the device via the **Phase Invert** button. Modulation will be applied in a linear fashion.

4.2.2 Gain dB

The same as **Gain Lin**, but modulation will be applied exponentially (Decibels). If you work with the **Env Follower** in dB mode, **Gain dB** is the right device to use.

4.3 Delay

4.3.1 Tape Delay

A standard delay. Can be switched to **Ping Pong** mode if needed. Set a **Min** and **Max Delay** in milliseconds or note length. Via **Delay Mod** you can blend from **Min** to **Max Delay**. **Feedback** feeds the delayed audio back into the delay line. Via the **Filter** dropdown menu, you can choose between different filters for the feedback path of the delay. Via the **Edit** button you get all the parameters you find in the **Filter** -> **Multifilter** device.

4.3.2 Shift Delay

With this device, you can apply a frequency shift in the feedback path of the delay. It comes with a High- and a Low-Cut function that can both be modulated, as well as the **Shift** value itself. The feedback function works just like the one previously explained.

4.3.3 Multitap Delay

This delay has up to 8 single delays which can each have different **Time**, **Gain**, **Pan** and **Feedback** settings. Feedback will combine as "cross-feedback" into all taps, which allows for intricate and dense patterns. You can also switch between three different **Modes** that affect the stereo spread of the delay taps depending on their pan setting in different ways. With **Randomize** you can randomly create short, medium and long early reflection patterns. Changes will be displayed visually underneath these parameters. At the bottom of this **DEVICE**, you will find an additional **Damping** control that will apply a smooth 6dB low pass filter.

4.3.4 Grain Delay

A granular delay which repeats audio snippets of the incoming audio. Change the length of those grains via the **Rate** parameter. Those grains can be pitched down or up and via **Feedback** be routed back into the delay. **Min** and **Max Delay** can be set in milliseconds or note length and the **Delay Mod** lets you blend between those two delay times. When modulating the delay time, **Grain Delay** tends to generate dense, diffuse delay patterns.

4.3.5 Repeater

A powerful looper device which once triggered, samples a specific length set with **Min** and **Max Time**. The **Trigger** parameter is an on / off parameter that can for instance be modulated with an **LFO**, via the **Transient Mod Source** or by a **Macro**. Creates stunning stutter effects. With the **Speed** knob can also control and modulate the playback speed. By setting the Mode to **Frequency** instead of **Duration** you will be able to create tonal effects with this device as well. The **Frequency** knob can be modulated and set to either hertz or semitones. Try applying a **Pitch Tracker** as your modulation source.

4.3.6 Fixed Delay

This delay device cannot be modulated (use the **Tape Delay** for modulation instead) but allows you to set a very precise **Delay Time** in fractional milliseconds. This is particularly useful if you want to create effects incorporating some sort of lookahead or compensate latency in parallel signal chains.

4.4 Distortion

4.4.1 Distortion

A distortion device with four different distortion types: **Soft**, **Medium** and **Hard** clipping plus **Fuzzy** for some extra grunge. **Symmetry** majorly alters the distortion, however if **Symmetry** is used with values other than zero, DC offset might be introduced. Simply activate the **DC Filter** in these cases to prevent this from happening. This device also features a **Pre Gain** and **Post Gain** for dialing in the amount of distortion.

4.4.2 Decimator

Bitcrushing and Samplerate mangling as you know it.

4.4.3 VariDrive

A highly musical and dynamic saturation / distortion device. All parameters do interact with each other and the two **Even** knobs have the biggest impact on even harmonics, whereas the two **Odd** knobs have the most impact on odd harmonics.

4.4.4 Waveshaper

Create custom distortion with the Waveshaper. Double click into the curve to add main points and drag the hollow points to alter the curve. Double clicking on points deletes them, double clicking on hollow points, however, resets the curve to a linear line. Change the grid from 2x2 to whatever you need. Shift-drag points and hollow points to only move them on the given grid. You can also alt-drag curve shapes (hollow points) to help with creating symmetrical shapes. The segment to the right of the dragged segment will be automatically adjusted accordingly.

4.5 Dynamics

4.5.1 Compressor

A versatile Compressor / Limiter that comes with all the settings you expect. Can be used for parallel compression via the **Dry/Wet** mix as well.

4.5.2 Gate

Simple and functional Gate with similar settings as the Compressor but with an additional **Hold** parameter instead of Make Up Gain.

4.6 Filter

4.6.1 Multi Filter

A filter with different selectable types: Low Pass 12dB (LP12), Low Pass 24dB (LP24), Band Pass 12dB (BP12), Band Pass 24dB (BP24), High Pass 12dB (HP12), High Pass 24dB (HP24), Notch 12 dB (Notch 12), Notch 24 dB (Notch 24), Allpass 12 dB (Allpass 12), Allpass 24 dB (Allpass 24), as well as Vintage Band, Low and High Pass filters in 12 and 24 dB, which also feature additional saturation controllable via the **Drive** parameter. In addition to the **Cutoff** frequency you can also alter the **Q** value.

TIP: Phaser-Effects
Since allpass filters have a flat magnitude response (they don't have any effect on amplitude), try mixing those with the **dry/wet** control. By combining multiple allpass filters, you can construct your own Phaser-style effects.

4.6.2 Param Eq

A one band parametric EQ with **Frequency**, **Gain** and **Q** parameters. They can be altered simultaneously via mouse controls by simply moving around the point located on the curve in the graphic interface of this device. You can also switch between Peak, **Lo Shelf** and **Hi Shelf**.

4.6.3 Splash

A special frequency mangling device. Quite similar to the parametric Eq, it simulates the behavior of a water surface to define the magnitude response over the whole frequency spectrum. You can alter the characteristics of the physics simulation with the parameters Tension, **Damping** and **Spread**. Changes in **Gain** or **Frequency** ripple through the frequency spectrum based on a fluid surface simulation.

4.6.4 Vowel Filter

A filter effect that creates vowel characteristics, blending through vowels. Higher **Q** settings increase the effect. It has two different modes: **Talkbox** and **Vowelize**.

4.6.5 2 Band Split

A two-row wide device that splits the incoming audio at a set frequency. Allows multiband processing.

4.6.6 3, 4, 6 Band Split

Same as two band split but occupies more rows to split even more bands. Be aware that all crossover frequencies can be modulated!

4.7 Generator

4.7.1 Sine

A sine tone generator with adjustable **Frequency** or **MIDI** key tracking and **Gain**.

4.7.2 Saw / Square

Just like the sine tone generator but with **Color** you can add harmonics, with the **Harmonics** parameter on the other hand you can morph between even (saw) harmonics and odd (square) harmonics.

4.7.3 Noise

A noise generator. You can switch between **White** and **Pink** noise and set the output gain accordingly. **Retro** is an emulation of a noise generator technique used in popular home computers of the early 80s (Linear Feedback Shift Register). Setting the device to this particular type you will find an additional, modulatable **RATE** parameter.

4.7.4 Crackle

A special device that introduces randomly distributed single impulses. Set the amount of crackles with the **Min Delay** and **Max Delay** parameters. The distribution of cracks can be altered with three parameters: **Delay Distribution** lets you weight the amount of crackles towards **Min Delay** or **Max Delay**, **Gain Distribution** alters the randomness of the crackle gain and **Pan Distribution** changes the stereo image behavior.

TIP: Crackle + IR-Verb
Put an "IR Verb" device after the crackle device, set it to full wet, and experiment with different samples that you can load as impulse responses.

4.8 Imaging

4.8.1 Stereo Split

A single input / two output device wide device that splits the left stereo channel to the left side and the right channel to the right side of the device, so you can then process them separately in two different chains.

4.8.2 Stereo Merge

A two input / single output device. Audio going into the left side is routed to the left stereo track and audio going into the right side of it is routed to the right stereo track.

4.8.3 Panning

This device allows you pan the input to the left or to the right and offers the possibility to narrow the stereo width.

4.8.4 M/S Decode

Decodes an incoming Mid Side (MS) signal into stereo. Make sure the left channel is the Mid signal and the right channel is the Side signal.

4.8.5 M/S Encode

Encodes stereo input into Mid Side (MS) signals. The Mid signal will be on the left channel, the Side signal will be on the right. channel.

4.8.6 M/S Split

A single input / two output device, that splits a stereo signal into Mid and Side. Mid goes to the left and Side goes to the right channel for dedicated MS processing.

TIP: Internal MS prcessing
Split stereo input to MS for internal MS processing. This way you can process the Mid Signal and / or the Side signal of a stereo source.

4.8.7 M/S Merge

A two input / single output device which expects the Mid content on the left input, Side content on the right input. Converts those to stereo and outputs a single stereo signal.

4.9 Modulation

4.9.1 Chorus

The chorus device is a very short, modulatable delay that is ideal for constructing chorus style effects. It offers the possibility to change the **Pre Delay** time in milliseconds (the higher the value, the lower the frequency) and the **Max Depth** surrounding the **Pre Delay** in milliseconds also. **Mod** uses these parameters to create the typical chorus effect. With **Feedback** you can strengthen the effect if suiting. Modulating the time with a **Triangle LFO** creates a variety of familiar sounding chorus effects.

4.9.2 Flanger

A high quality flanger effect with **Damp**, **Frequency** and **Feedback**. To create a basic flanger, modulate the **Frequency** parameter with a **Triangle LFO**, potentially in **DAW sync**.

4.9.3 Phaser

A classical phaser effect with up to six selectable stages (you won't find this very often). Features a **Cutoff** frequency and **Feedback** parameter. Modulated by an **LFO**, it creates the typical setup you know from various phaser plug-ins.

4.9.4 Vocoder

Create your own vocoder with tons of options in the Rack. Being a two-input device, the Carrier input is on the left and the Modulator input on the right channel. You can switch between two different **Modes**. It will by default be set to **Adaptive**. **Resolution** configures how precisely the vocoder will transfer spectrum of the modulator to the carrier. Low values will smooth the frequency spectrum strongly, in which case small details in the timbre of the modulator will have less effect on the carrier. **Shift** will affect the formants of the modulator. If your modulator or carrier are quiet, you can make up the result with **Boost**. To further improve the responsiveness of this **DEVICE**, you can control the **Attack** and **Release** in percentage. All parameters mentioned can be modulated. Changing the **Mode** to **Direct** will deactivate **Resolution**, **Attack** and **Release** to present a simplified and more direct approach.

TIP: Basic Vocoder Setup
Basic setup would be inputting a vocal sample on the right channel as the Modulator (**Stereo In**). Setting the **CONTROL** to MIDI, Voicing to Poly, inserting a **Saw / Square Generator** device as a Carrier on the left channel with MIDI controlled Frequency and most likely an **ADSR** (Saw / Square) with some Color mixed in and you are set with the most basic vocoder you can think of. Much for fun though obviously when you experiment with it and for instance use the **Sidechain In** device as Carrier while putting ENRAGE on a vocal track.
Try a **Noise** generator as carrier, set mode to **Adaptive** and use a high **Release** setting. This will result in almost "reverb" like output.

4.9.5 Ring Mod

A Ring Modulator multiplies two audio signals in the time domain where the left input gets multiplied by the right input. Either input can be the **Stereo In** or a **Generator** device or go ahead and experiment with a **Sidechain In** device and feed other audio material from your session into it. A common effect is applying ring modulation with a sine oscillator.

4.9.6 Timbre

Allows formant shifting, brightness adjustment and other timbral transformations of your sound. The **Warp** and the **Resolution** knob can both be modulated and offer intuitive access to this device. You can also choose between four distinct **Modes** to further alter the behavior. Works on polyphonic material as well.

4.10 Pitch

4.10.1 Pitch Shift

A general-purpose pitch shifter device which offers the opportunity to shift in semitones. The overall behavior can be altered firstly by the **Algorithm**. Here you can choose between **Granular** and **Spectral**. In most cases **Spectral** will perform better, yet it will smear transients whereas **Granular** does try to preserve them which in some cases may result in „machine gun“ like repetition effects. Just try them both and see what suits your audio source. Now you can further adjust the behavior by setting the **Response** type. Whilst working with percussive audio or the like it is recommended to select **Snappy**, whereas **Smooth** would be the choice for tonal audio material.

4.10.2 Freq Shift

A frequency shifter that adds or subtracts the set frequency from the input. A perfect pitch shifter for monophonic, single tone audio input with only one dominant fundamental such as kickdrums for instance. Gets creative when used on harmonically rich material or audio that mainly contains noise. Leaves all transients perfectly in shape and is highly artifact free. Because of these attributes it should be first choice when retuning drums is what you aim for.

TIP: Add unique texture to your sound
Activate **Stereo Spread** and experiment with the **Feedback** amount to add unique texture to your sound. Make sure to pull back the **Dry/Wet** ratio for a more subtle effect.

4.10.3 Voice Shift

A pitch shifter dedicated to monophonic, tonal audio and more specifically: vocals. This pitch shifter also provides control over the formants which can and should be used creatively as well.

4.10.4 Warp

A frequency mapping device. Map the incoming frequency to a different frequency. Via **Amount** you can blend from a straight and normal mapping towards the curve you selected.

4.10.5 Sub Octaver

A special bass enhancer that in phase adds one octave and / or two octaves below the input signal. Works best on low, tonal input.

4.11 Reverb

4.11.1 Basic Reverb

Basic and CPU friendly room simulation with the possibility to modulate **Width**, **Size** and **Damp**. Set the **Room Scale** and **Modulation** to taste.

4.11.2 IR VERB

A **Convolution Reverb** that precisely simulates the reverberation of a physical or virtual space. Import pre-recorded impulse responses of real locations by right-clicking on the small screen and hovering your cursor on **Built in IRs**. You can simply drag audio files from a file browser into the sample window as well. It will also visually display the audio sample. By left clicking on the display window, you can adjust the sample start. This is specifically important whilst working with reversed IR's.

TIP: Experiment with IR Verb
You can get very creative with this device and import all sorts of audio samples to design unique texture for your reverb. The **Normalize** button will then normalize gain and the **Reverse** button will reverse the reverberation tail. Right underneath these buttons you will find the **Pre Delay** section, where you can set your Pre Delay either in milliseconds or in notes.

4.12 Analyzer

4.12.1 Analyzer

A visual analyzer with three different display types and an **RMS** / **Peak** metering section.

4.12.1.1 Spectrum

This setting will help you to gain insight of the overall frequency spectrum with frequency on the horizontal axis and gain in -dB on the vertical axis.

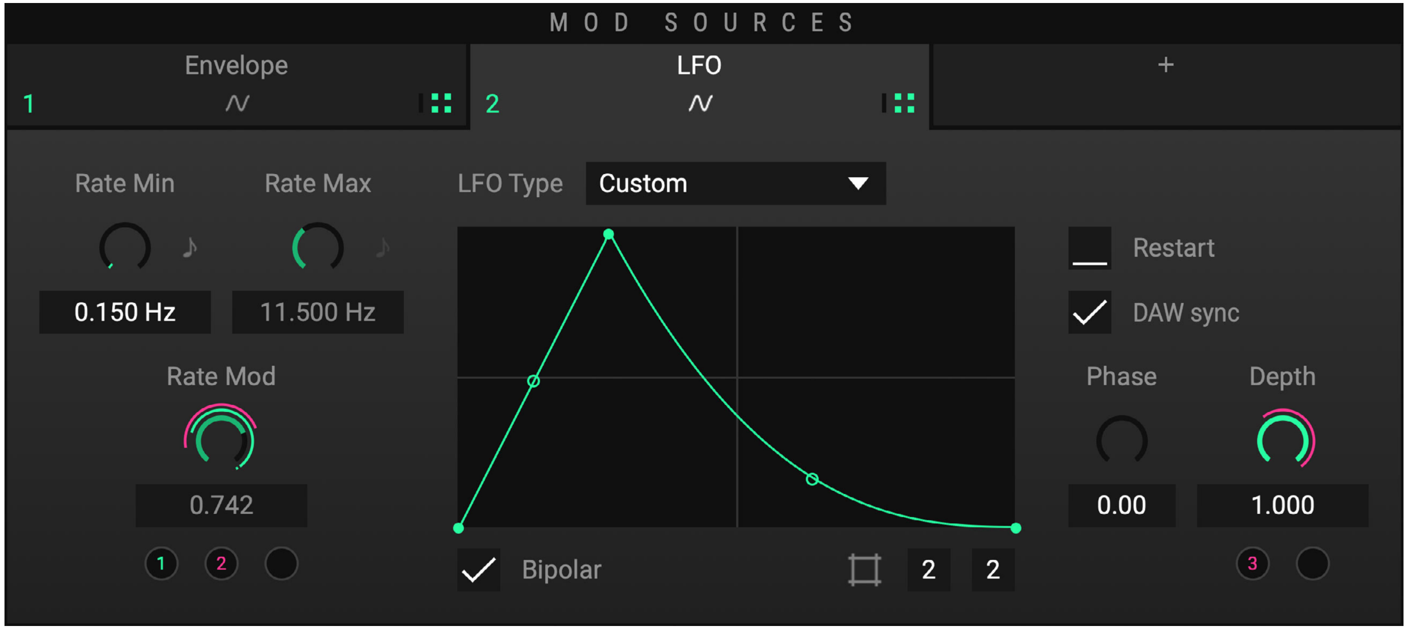
4.12.1.2 Scope

Here you find an **Oscilloscope** that lets you observe the waveform of incoming audio signals. Comes with left and right channel separation and a **Gain**, as well as a **ZOOM** knob to optimize the visualization according to your audio material.

4.12.1.3 XY

In this setting you can precisely monitor the panorama of your input signal through a **Vectorscope**. This presentation layer will help you to keep an eye on stereo width and mono compatibility of your incoming audio signal.

5. MOD SOURCES



Everything that relates to **MOD SOURCES** is color coded green. Thus, whenever you apply one to a **DEVICE** or to another Mod Source you will see a little green number which represents which Mod Source modifies the given parameter.

5.1 Env Follow

The **Envelope Follower** does what you would expect with numerous different ways to tweak it. Being one of the core **MOD SOURCES**, you can change it to whatever you need.

5.1.1 Display

The **Envelope Follower Display** visualizes the input signal altered by the settings. On the bottom you have zero modulation, on the top maximum modulation.

5.1.1.1 Gain

With **Gain** you can visually raise or lower the input gain for example if the input is simply not loud enough and you want to modulate higher values, or if the input gets quieter because the **Envelope Follower Filter** is in use.

5.1.1.2 Offset

Offset lets you alter the analysis even more by shifting the starting point upwards or downwards. Shifting it upwards means the minimum modulation is not zero anymore. Shifting it downwards means that quieter input signals will not be analyzed at all.

5.1.1.3 dB Mode: dB From

In **dB Mode** you have two different settings. **dB From** is the lowest analyzed negative dB value. Likewise, **dB To** on the right side of the visualizer is the ceiling at 0 dB.

5.1.1.4 dB Mode: dB To

You can limit the analyses on higher gains, similar to visually clipping the input signal.

5.1.2 Attack, Hold and Release

Attack, **Hold** and **Release** are set to moderate values by default which should give usable results most of the time. However, if you need to grab fast moving gain changes or want slower and smoother modulation you can alter this behavior.

5.1.3 Skew

Skew basically bends the output curve of the input analysis towards quieter or more energetic signals. Click and hold in the middle of the little graph and drag it up or down to change the curve. Right click on it for a drop-down selection with different **Skew** curves.

5.1.4 HQ Mode

High quality mode will lead to less "ripple" with short attack and release times. It is slightly more CPU heavy and rather dedicated to compressor / limiter creation. To wrap it up, it’s always a good idea to switch it on when working with rather fast attack and release times.

5.1.5 dB Mode

In **dB Mode** the **Envelope Follower** analyzes the signal in decibel, thus exponentially, rather than linear. Be aware that the options for the visualizer change from **Gain** and **Offset** to **dB From** and **dB To** (see above).

5.1.6 Input Selector

By default set to **Main In**, which analyzes the incoming audio stream. Click on it to change the **Envelope Follower** input to sidechain or a tap source input from within the Rack.

5.1.7 Filter Section

To be able to focus the analyzer on specific frequency ranges, there is a built-in filter section prior to the analysis. You can select filter types (**BP** = Bandpass, **LP** = Lowpass, **HP** = Highpass) and select the frequency you wish to use. The little headphone icon lets you audition this filter with a bypassed devices Rack to dial it in very precisely.

5.2 LFO

The **LFO** is straight forward, yet it comes with some interesting options.

5.2.1 Rate Min, Rate Max and Rate Mod

Starting with **Rate Mod**, this parameter blends between **Rate Min** and **Rate Max**. Set all the way down to 0.000 means that the LFO speed is the same as **Rate Min**. Turned all the way up to 1.000 on the contrary, the LFO speed is the same as **Rate Max**. So set the rates according to your needs. The little note icon lets you put the rate into note length grid. **Rate Mod** again can be modulated by other **MOD SOURCES** (like **Envelope Follower** or another **LFO**) or via the **MACROS**.

5.2.2 LFO Type

Select between basic curves like **Sine**, **Triangle**, **Saw** and **Square**, set it to **Random Sample & Hold**, which creates a random value that holds for the length of the LFO rate, or **Random S&H ramped**, which smoothly moves from one random value to another over the length of the **LFO** Rate or even use **Custom** to draw your own **LFO** shapes.

5.2.3 Restart

Restart can be used when you want to trigger restarts of the **LFO** via the **CONTROL** section. That can be either via audio or midi, see **CONTROL** section (chapter 3.3)

5.2.4 DAW sync

Syncs the note length to the DAW bpm. Note: when **DAW sync** is active you cannot modulate between **Rate Min** and **Rate Max** anymore because that behavior is taken over by the tempo of your DAW. You can then do tempo changes in your DAW instead of course.

5.2.5 Bipolar

Turned on by default, bipolar makes the **LFO** go up and downwards. Deselect it to get a range from 0.000 from the bottom to 1.000 at the top of the **LFO** curve.

5.2.6 Gain

Gain lets you adjust the amount of the signal that will be affected and works like a dry/wet knob in this regard.

5.2.7 Phase

Phase lets you add an offset to the LFO position.

5.2.8 Depth

With Depth you can take control over the intensity of this **MOD SOURCE**. It is important to note that you can modulate this parameter as well.

5.3 CURVE

Curve is a simple yet powerful **MOD SOURCE** which lets you draw custom curves which get triggered via **CONTROL** (3.3) either by audio or midi. Double click in the **Curve** window to create a new anchor point. Double click on that point again to remove it. You can alter the grid and via shift-click and drag you are able to move points only on the selected grid. Set the length, the curve should take from start to end. The **Duration** can be measured in milliseconds or note length. **Gain** lets you adjust the amount of the signal that will be affected. **Depth** will allow you to play around with the intensity of **Curve**.

5.4 ADSR

An envelope **MOD SOURCE** describing a change over time. **Attack** displays the time taken to run up from zero to max. **Decay** sets the time to run down onto the sustain level. **Sustain**, the only value not describing time, sets the output level of **ADSR** during the **Sustain** period. **Release** describes the time to run down from the sustain level back to zero. **ADSR** needs to be

TIP: Create pumping ducking effects
Set rate to 1/4, draw a custom shape, set to unipolar, modulate gain -> easily create pumping ducking effects.

triggered either via audio triggering or MIDI triggering (see 3.3: **CONTROL**). As opposed to **CURVE**, which simply plays back the value it is set to and upon retriggering starts back from the beginning, **ADSR** will rise to max from the current value within the **Attack** period. Respectively, when releasing before **Attack** has finished, the **Release** will start from the current value.

5.5 MAPPING

Another mighty **MOD SOURCE** which allows you to map other mod sources to a new value with a custom curve. However, instead of being triggered, it must be modulated by at least one other **MOD SOURCE** or one (or several) of the **MACROS**. This way you can create an inverted version of an existing **LFO** for instance.

5.6 TRANSIENT

Transient analyzes incoming audio for sudden increases in volume and outputs those changes in dynamic as an envelope. A sudden increase in volume will result in a spike, which you could apply to a **Gain** device in order to implement a transient shaper for example.

5.6.1 Gain and Offset

The stronger / more sudden the increase, the higher this spike will be. Adjust the **Range** according to your input material using **Gain** and **Offset** to prevent overshoots.

5.6.2 Time

The **Time** setting specifies a release time for this envelope, you can think of it as „snappiness“.

5.6.3 Skew

With **Skew**, you can further change the shape of the detected envelope.

5.6.4 Source

Source lets you specify the input, **Transient** should analyze. This by default is the **Main In**put. You can alternatively use a Sidechain (**SC In**) or **Tap** as input sources.

5.6.5 Filter

An additional filter similar to the one of the **Envelope Follower** and **Trigger Detector** in the **CONTROL** section, that helps focusing the analysis on a particular frequency range.

5.7 CHANGE

Change analyzes the change of any input data. If the input goes up, the change mod source will output how fast it goes up - vice versa, if the input goes down, the change mod source will output how fast it does that. Wiggle the input knob and you'll get the idea. Select which input or even combined inputs it should analyze. Use the **Envelope Follower** for instance, to create your own transient detector. With **Smoothing** you can control abrupt or too quick behavior to create more useful output values. **Skew** again controls the sensitivity and with **Bipolar** you can set output values around 0.500 or create a range from 0.000 to 1.000.

5.8 PITCH TRACKER

The **Pitch Tracker** analyzes the pitch of an incoming, monophonic signal.

5.8.1 Input

Select the input source that you wish to be analyzed. This by default is the **Main In**put. You can alternatively use **SC In** or **Tap**.

5.8.2 Root Note

You also must provide a range, which is defined by setting a root note and a range of semitones. E.g. if you set **Root Note** to C2 and **Range** to 24 semitones, a detected pitch of C2 will result in output value of 0, C3 outputs value 0.5, up to C4 with the output value of 1.

5.8.3 Qual. Thresh

Because the **Pitch Tracker** is very sensitive and might behave erratically if the signal is too noisy or if it is not composed of one clearly identifiable fundamental frequency, you can define a quality threshold. The **Pitch Tracker** will only update its output if the detection quality is above this threshold. Experiment with this value in combination with **Smoothing** to get optimal results for your input material.

5.8.4 Output

Alternative to the detected pitch, you can set the **Pitch Tracker** to output **Clarity** instead. In this case, the output will be the „Pureness" of the detected signal. The more tonal and clean it is, the higher the output.

5.9 FORMULA

A **MOD SOURCE** that lets you generate modulation values by using mathematical expressions. If you are one of the craftier types, you will surely love this.

5.9.1 Input

You can use 2 separate input modulation values simultaneously and combine them in whatever way you want. To get started just grab a modulation value and pin it to one of the inputs the same way you would set up a **MACRO**.

5.9.2 Eval Rate

Right next to the inputs you will find an evaluation rate parameter with which you can set the rate in which the output value gets reevaluated and updated. You set the rate in frequency or note length by clicking on the note icon.

5.9.3 Output

The small window next to **Eval Rate** will visualize the modulated value. Right next to it you will find **Interpolation** where you can either set the drawn line to **Step** or **Linear**. Additional settings are a **Bipolar** display set by default and optional **DAW sync**.

5.9.4 Out= / Mathematical functions

Think of the **Out=** tab as it were a calculator. Type in how you wish to modulate the input values Mod.in1 and Mod.in2. You can also use the previously generated output value Mod.out. Find all supported functions in chapter 6.2 down below.

5.9.5 Example Use cases

SAMPLE AND HOLD

- Assign any other **MOD SOURCE** to **Input 1**, type "Mod.out1" into the expression text editor. Now, Formula will output the value of **Input 1** directly. To turn this into a "**sample and hold**", set **Interpolation** to **Step** and reduce the **Eval Rate**. This way you can turn any **MOD SOURCE** into a steppy one.

SMOOTH

- You can use **Formula** to "smooth" another **MOD SOURCE**, by programming a primitive filter: Assign any other **MOD SOURCE** to **Input 1**, then use the expression: "0.01 * (Mod.in1-Mod.out) + Mod.out"
- Now, if **Input 1** changes, **Formula**'s output will rise very slowly up or down. The factor 0.01 affects how fast it changes. Congratulations, you have programmed a low pass filter!

5.10 DELAY

Delay allows you to delay other MOD SOURCES (input) by setting the Delay Time either in hertz or in notes synched to your host. Input represents the value that is going to be delayed. This parameter can be modulated as well. Assign any other MOD SOURCE to Input to delay it by a specific amount of time.

6. TIPS AND TRICKS

6.1 General Usage

ENRAGE will enable you to approach situations where you find yourself stuck. In other words, if you don't know how to solve an issue (whether it be mixing or sound design), try ENRAGE before you end up enraging.

6.1.1 Know what you aim for

Not having an aim will certainly lead to confusion. Since this Plugin is offering a huge variety of creative possibilities, it is even more important to set yourself a goal. Try think of it like you would paint a picture. Once you know what you want to paint, you've got a frame that you now just need to fill with paint. Not knowing what to paint on the other hand is this invisible barrier that keeps most of us from even making the first step. This sounds kind of philosophical, but it helps to make this point clear. If you go ahead and try to build a specific effect chain, ENRAGE will most definitely help you achieve that goal. As soon as you have become an expert painter you can of course start with a blank piece of paper and see where your pen will lead you. But until then, try sticking to the script.

6.1.2 Countermovement of Device parameters

While dialing in the parameters it can be very helpful to mirror your settings to reach the perfect tuning. You can do this by countermoveing them. If you feel like there is too much low end for instance, try boosting the higher frequency spectrum instead of cutting out too much bass. Or do a bit of both. ENRAGE is offering loads of possibilities in this regard as you can easily set up macros to automate several device parameter movements simultaneously.

6.1.3 Do not use what you don't know!

You won't eat a plant that you aren't sure whether it is safe to eat. Luckily, we have the undo function but to avoid nasty feedback loops or digital clipping, it is recommended not to use certain functions of ENRAGE when you are not sure what they are supposed to do. That is why you have this manual anyways.

6.2 Quality of life - Shortcuts

Regarding the modulation assignment of **MOD SOURCES** and **MACROS**, ENRAGE offers some neat shortcuts to optimize your workflow.

6.2.1 Hover Highlight

Hovering over a modulation assignment icon will highlight every target it is already assigned to. If the targeted **MOD SOURCE** or **DEVICE** is selected, it will also highlight the specific target value.

6.2.2 Drag on Target

Dragging the modulation assignment icon on a **MOD SOURCE** or **DEVICE** will open its editor after a short delay to further assign it to the target value of your choice.

6.2.3 Double-click on Icon

Double clicking on the modulation assignment icon will open already assigned targets one after another. This will come in handy when working with rather complex templates.

6.2.4 Reset Knob Values

Reset knobs to their default values by simply double clicking on them.

6.2.5 Curve Editing

• Shift-dragging a curve point will snap it to the grid.

• Shift-dragging a curvature modifier handle (empty circle) will snap it to a fixed step size

• Alt-dragging a curve point will symmetrically move the opposite point in the opposite direction

• Alt-dragging a curvature modifier handle will set the following curvature handle to the opposite position

• ctrl/cmd-alt-dragging a curvature modifier handle will set the following curvature handle to the same position

Note that these actions apply to all curve editors.

6.3 FORMULA Functions

For every Function insert the prefix: “Math.” Here is one example: “Math.abs(x)”

FUNCTION	DESCRIPTION
abs(x)	Returns the absolute value of x
acos(x)	Returns the arccosine of x, in radians
acosh(x)	Returns the hyperbolic arccosine of x
asin(x)	Returns the arcsine of x, in radians
asinh(x)	Returns the hyperbolic arcsine of x
atan(x)	Returns the arctangent of x as a numeric value between -PI/2 and PI/2 radians
atanh(x)	Returns the hyperbolic arctangent of x
ceil(x)	Returns x, rounded upwards to the nearest integer
cos(x)	Returns the cosine of x (x is in radians)
cosh(x)	Returns the hyperbolic cosine of x
exp(x)	Returns the value of Ex
floor(x)	Returns x, rounded downwards to the nearest integer
log(x)	Returns the natural logarithm (base E) of x
max(x, y, z, ..., n)	Returns the number with the highest value
min(x, y, z, ..., n)	Returns the number with the lowest value
pow(x, y)	Returns the value of x to the power of y
random()	Returns a random number between 0 and 1
round(x)	Rounds x to the nearest integer
sin(x)	Returns the sine of x (x is in radians)
sinh(x)	Returns the hyperbolic sine of x
sqrt(x)	Returns the square root of x
tan(x)	Returns the tangent of an angle
tanh(x)	Returns the hyperbolic tangent of a number
E	Returns Euler's number (approx. 2.718)
LN2	Returns the natural logarithm of 2 (approx. 0.693)
LN10	Returns the natural logarithm of 10 (approx. 2.302)
LOG2E	Returns the base-2 logarithm of E (approx. 1.442)
LOG10E	Returns the base-10 logarithm of E (approx. 0.434)

FUNCTION	DESCRIPTION
PI	Returns PI (approx. 3.14)
SQRT1_2	Returns the square root of 1/2 (approx. 0.707)
SQRT2	Returns the square root of 2 (approx. 1.414)

7. TUTORIALS

In this section we will create some fun presets together which will reveal the general workflow as well as some tricks and usage tips. We want to equip you with the basic principles of this mighty tool, so you get an idea of what ENRAGE is truly capable of.

7.1 Flanger

7.1.1 Flanger Basic

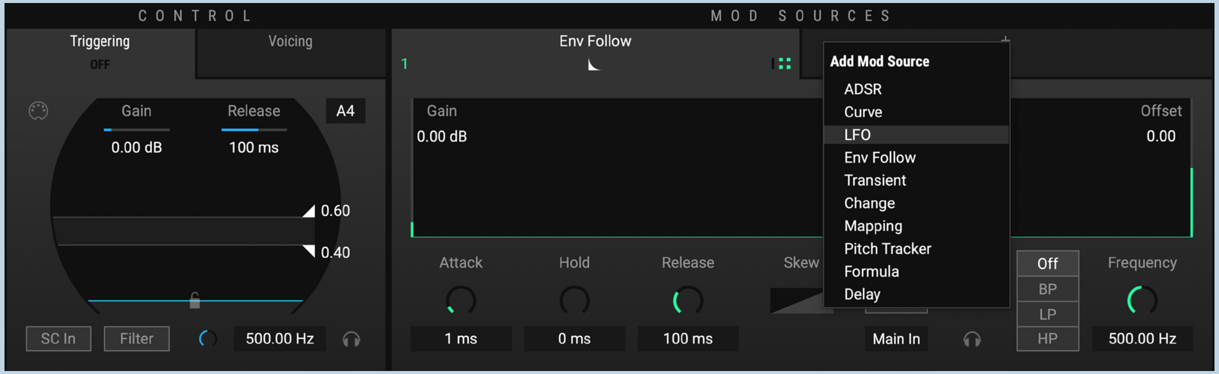
We do have an amazing sounding Flanger device on board. Let's see how this goes! First, please select the „Untitled“ Preset.

STEP 1: ADDING A FLANGER DEVICE

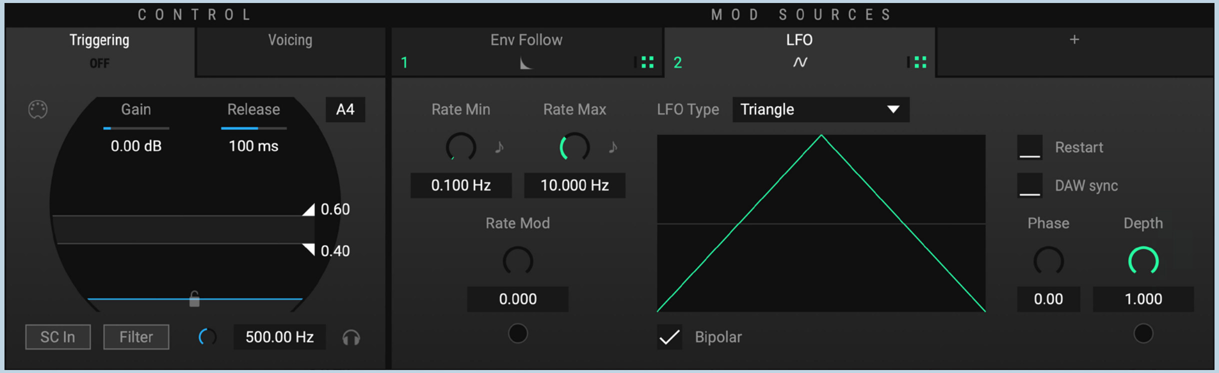
- Open ENRAGE. On the lower left in the devices list select the **Modulation** Category.
- Drag and Drop a **Flanger** onto the **Gain** device already located on the grid to your right.
- The signal chain should look like this: **Stereo In – Gain – Flanger – Stereo Out**

STEP 2: ADDING AN LFO TO MAKE IT FLANGE

- At the top of the GUI, go to the green colored **MOD SOURCES** section.
- Click the (+) icon to the right of **Env Follow** and select **LFO**.

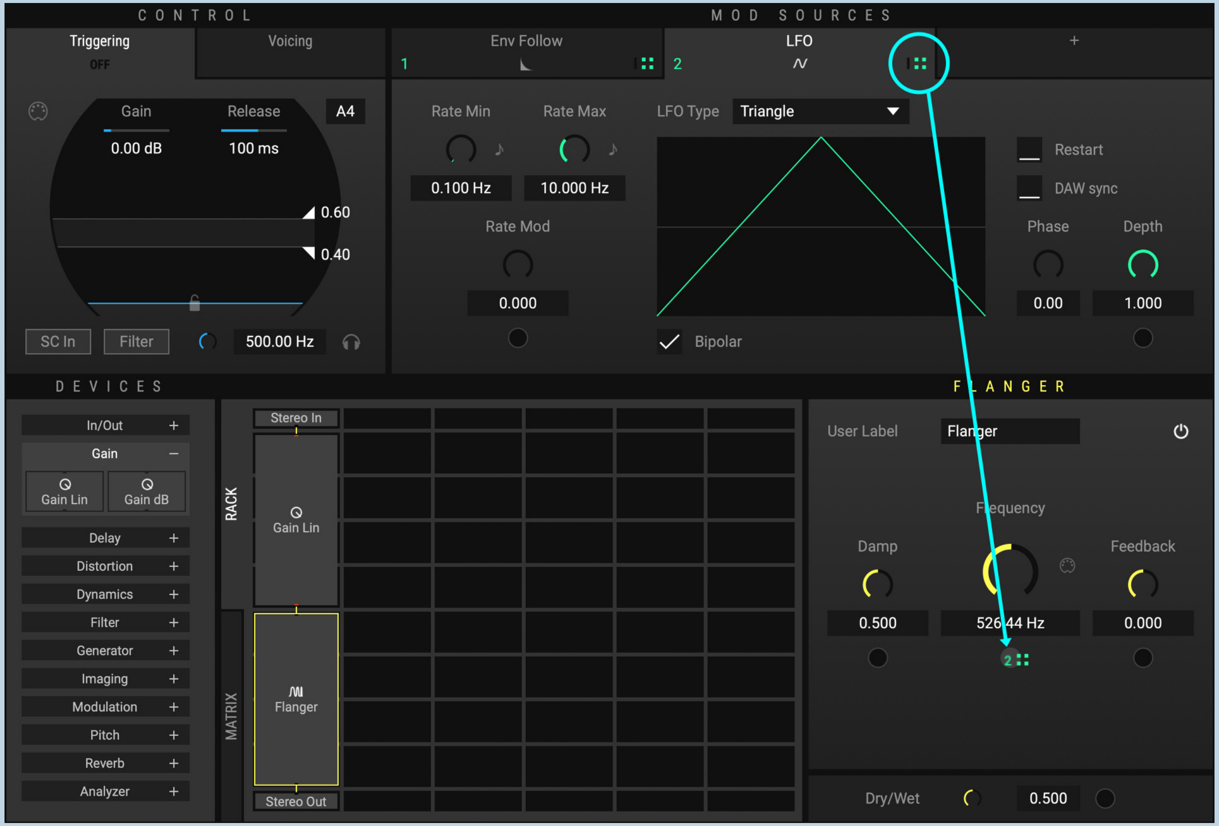


- Most Flangers have a lower rate so we need to make a few adjustments here.
- Select **Triangle** in the **LFO Type** dropdown menu.
- Set the **Rate Min** to **0.100 Hz**.
- Now your **LFO** setup should look like this

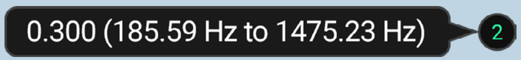


STEP 3: ASSIGNING THE LFO MODULATION TO THE FLANGER DEVICE

- Drag the Mod Assignment icon (::) of the **LFO** onto the Mod Target circle ● located under the Frequency Knob of the **Flanger Device**.



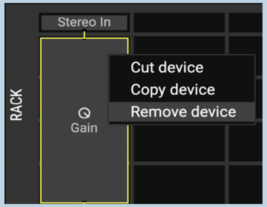
- Klick and pull the little green number up and set it to about 0.300.



TIP: Fine tune values
Hold ctrl + drag to fine tune all values.

STEP 4: LET'S KEEP THE RACK AND MOD SOURCES TAB NEAT AND TIDY

- Right-click on **Gain** located above the **Flanger** device and select **Remove device**. Right click on the **Env Follow** icon and click **Remove**.



- If you are looking at something like this now, your very own Flanger is ready to use.

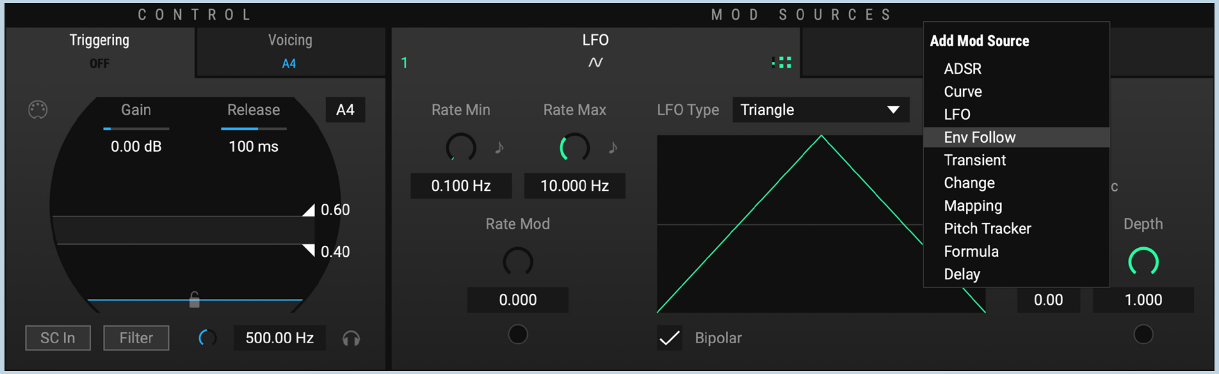
TIP: Don't stop here
You can of course tweak whatever you like from here. Try out different settings and add more devices to learn more about ENRAGE's modular setup or read the next chapter to dive in a bit deeper.

7.1.2 Dynamic Flanger

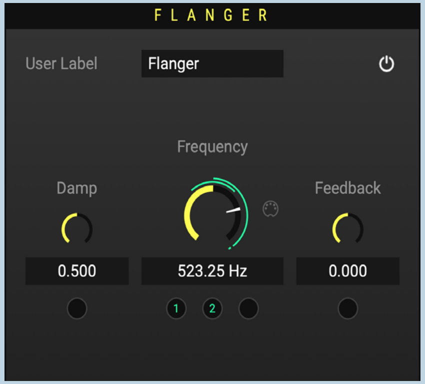
Now let's make it a bit more interesting. Thousands of ways to go from here alone. We want to present one. This is based on the basic Flanger Tutorial above. In ENRAGE, you can also load the preset "Tut01 1 Flanger" which is the result of the previous chapter.

STEP 1: ADD AN ENVELOPE FOLLOWER TO FURTHER MODULATE THE FLANGER FREQUENCY

- Next to the **LFO**, use the plus icon to add an Envelope Follower (**Env Follow**).



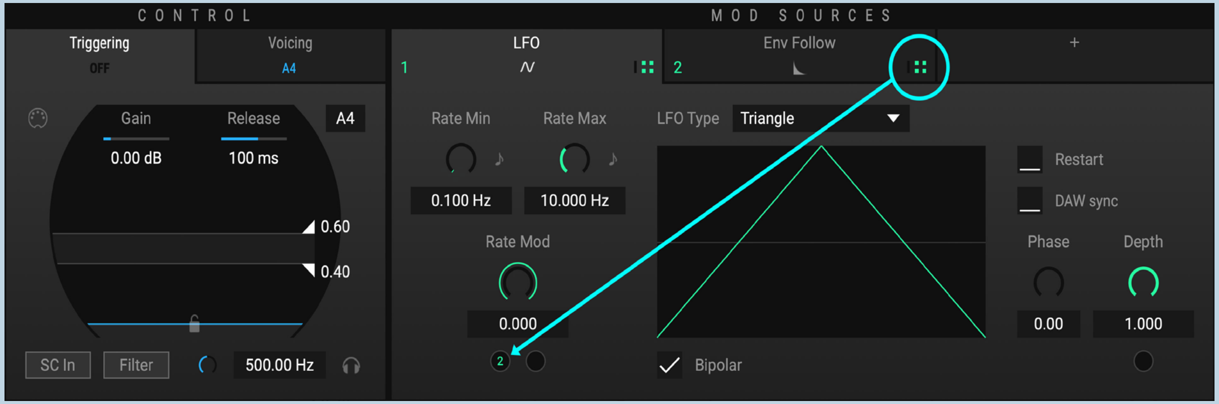
- Use the **Envelope Follower**'s green Mod Assignment icon (::) and connect it to the Frequency Mod Target circle ● of the **Flanger** device right next to the green **1** of the **LFO** which has already been assigned. Now push up the **2** by taste.



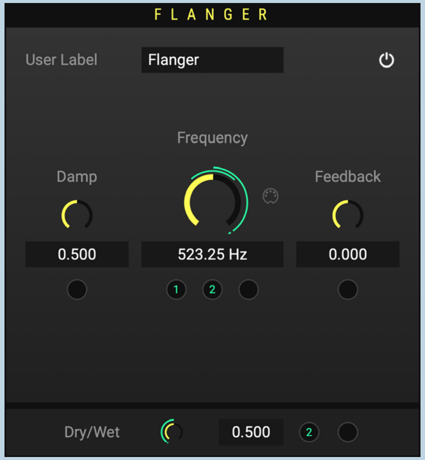
- What is happening now? The **LFO** still shifts the frequency by the given rate up and down based on a center frequency, the **Env Follow** on the other hand shifts this center frequency up, whenever the incoming signal increases in volume.

STEP 2: USE THE ENVELOPE FOLLOWER TO MODULATE THE LFO RATE MOD

- Apply **Env Follow** to the **LFO Rate Mod** and push it all the way up. In addition to modulating the center Frequency of the **Flanger**, you now also modulate the **LFO** speed which is responsible for the Flanger-effect.



- Interesting, but probably too cartoony. No problem. Use the same **Env Follow** again and assign it to the **Flanger Dry/Wet**. Pull it all the way down so the behavior from above still applies, but the louder the input, less of the effect is mixed in.



TIP: These Tutorials only show some workflows. Try to apply these patterns to other topics. For example, exchange Flanger with the Basic Verb, instead of Frequency modulate Dry/Wet and instead of the Dry/Wet of the Flanger modulate the Size of the Reverb. Or simply use Gain to create a stutter effect which reacts faster to higher input loudness. You get the idea

7.2 Vocal Doubler

7.2.1 Vocal Doubler Basic

CREATING A BASIC VOCAL DOUBLER IS SUPER EASY

- Open ENRAGE. From the Device list on the left, select the **Delay** section and drag and drop the **Tape Delay** onto **Gain**.
- Set **Mode** to **Ping Pong** and **Max Delay** to **75ms**. Done. It should look like this:

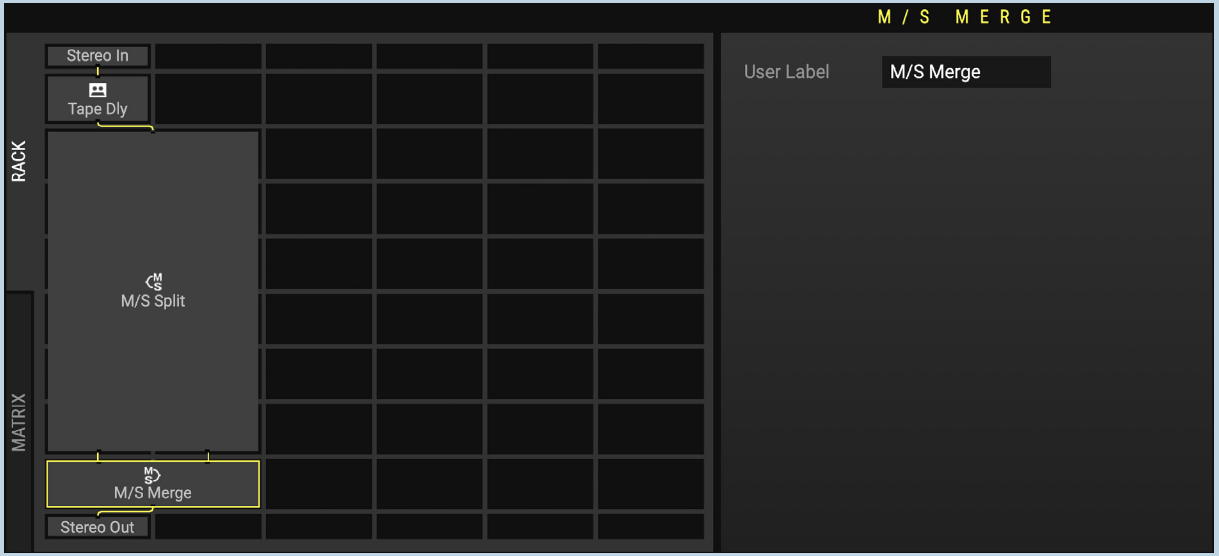
- To change the timing of the doubled voices, you can either pull down **Delay Mod** or push it up. It changes the value from **Min Delay** (in this case 0ms or no delay) to **Max Delay**, (in this case 75ms).

7.2.2 Vocal Doubler Advanced

The vocal doubler above already spreads out the signal in the stereo field. But is there a way to make it more interesting, shiny, smooth? Of course there is! A million ways. Here is one.

STEP 1: SPLIT UP THE SIGNAL TO M/S

- Go to the device list, click on **Imaging** and drag and drop **M/S Split** below the **Tape Delay**. When hovering over the top corner of the new **M/S Split** device in the Rack, you can pull it all the way up so that **Tape Delay** only takes up the space of one row.
- Now place an **M/S Merge** below the M/S Split. It does what it says: it splits the Mid from the Side signal and merges the two signal streams again to regular stereo. For now, you will not hear any change, but you can process the Mid signal on the left column and the Side signal on the right column separately. This is what it should look like:



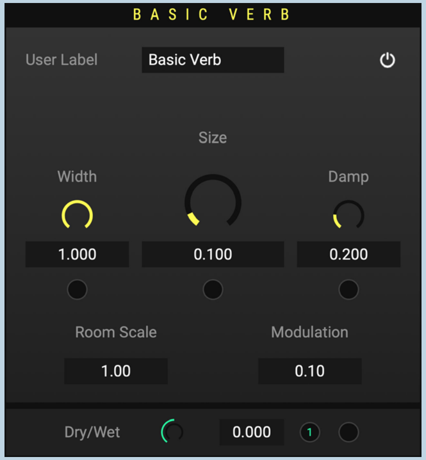
STEP 2: SETTING UP THE DEVICES FOR THE MID SIGNAL AND SIDE SIGNAL STREAMS

- Add one **Gain Lin** device to the Mid signal stream and one **Gain Lin** to the Side signal stream.
- Add a **Chorus** to the Side signal stream only.
- Add a **Basic Verb** to the Side signal stream only.

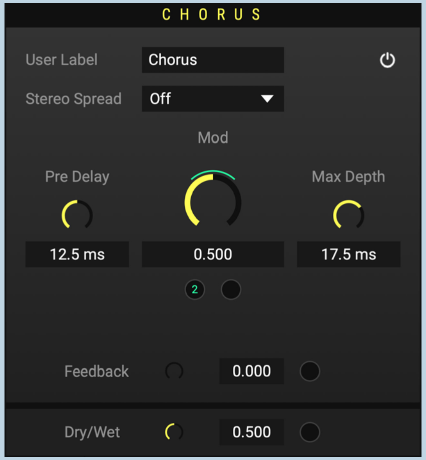
- Will sound terrible though, we need to adjust a few things.

STEP 3: ADJUSTING THE SETTINGS TO GET A DYNAMIC REVERB

- Let's start with the reverb. Set the **Size** to **0.100**.
- Assign the **Env Follow** to the **Dry/Wet** of the reverb by pulling its green Mod Assignment icon (::) onto the **Dry/Wet** Mod Target circle ● of the **Basic Verb**.
- Pull **Dry/Wet** all the way down to zero.
- Pull up the green **1** up to the center of the knob, at about **0.500**. This should sound way better. We created a dynamic reverb that gets mixed in depending on the input loudness.



- On to the **Chorus**. Add an **LFO** by clicking the + icon to the right of the **Env Follow**.
- For the **LFO Type** pick **Triangle**.
- Apply the **LFO** to the **Chorus Mod** by dragging the Mod Assignment icon (::) onto the Mod Target circle ● below **Mod** in the **Chorus** device.
- Pull up the green number **2** to roughly **0.300**.
- Change the **Max Depth** value to **17.5 ms**.
- Some tricky stuff going on now! To wrap it up, we want to create an easy knob to dial in the whole effect, no? Read on how to assign a **MACRO** in the next chapter.

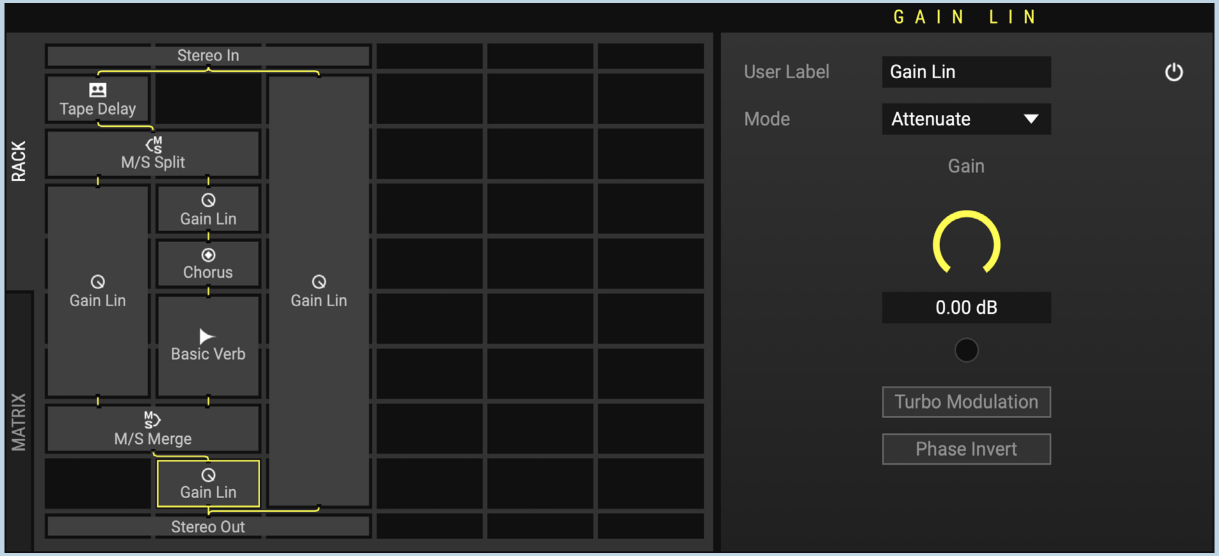


7.2.3 Vocal Doubler Advanced Macros

Again, more than one way to gain control over the above preset, but we now want to give you an idea on how to use **MACROS**. We want one for controlling the dry signal only and one to dial in the effect only.

STEP 1: ADD GAIN DEVICES TO CONTROL ORIGINAL SIGNAL AND EFFECT SIGNAL

- First, select the **Tape Dly** device and set the **Dry/Wet** to **1.000** (100% wet)
- Increase the width of **Stereo In** and **Stereo out** to three columns.
- Add one **Gain Lin** to the third column.
- Additionally, pull up **M/S Merge** one row to make space for yet another **Gain Lin** just below **M/S Merge**.
- It should result in this:



- Pull down both new **Gain Lin** devices all the way down to minus infinite. Now when you play back, you should not be able to hear any signal coming through.

STEP 2: ASSIGN THE MACROS

- Drag and drop the magenta colored, cubic Mod Assignment icon (**::**) of **Macro 1** onto the big **Gain Lin** in the third column and assign it to the **Gain**. Pull up the magenta **1** all the way. Do the same with **Macro 2** but to the **Gain Lin** in column two, the one just below **M/S Merge**. You should see something like this:

- Now you can use Macro 1 to mix in the original dry signal and Macro 2 to dial in the effect as needed. These macros can even be automated. You can for example only add a stereo effect on certain words, phrases, or parts of your project.

TIP: Add labels to recap your idea
If you add some useful labels to the **Devices**, **Mod Sources**, and the **Macros**, you will be able to recap exactly what you did. ou can of course tweak whatever you like from here. Try out different settings and add more devices to learn more about ENRAGE's modular setup or read the next chapter to dive in a bit deeper.

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