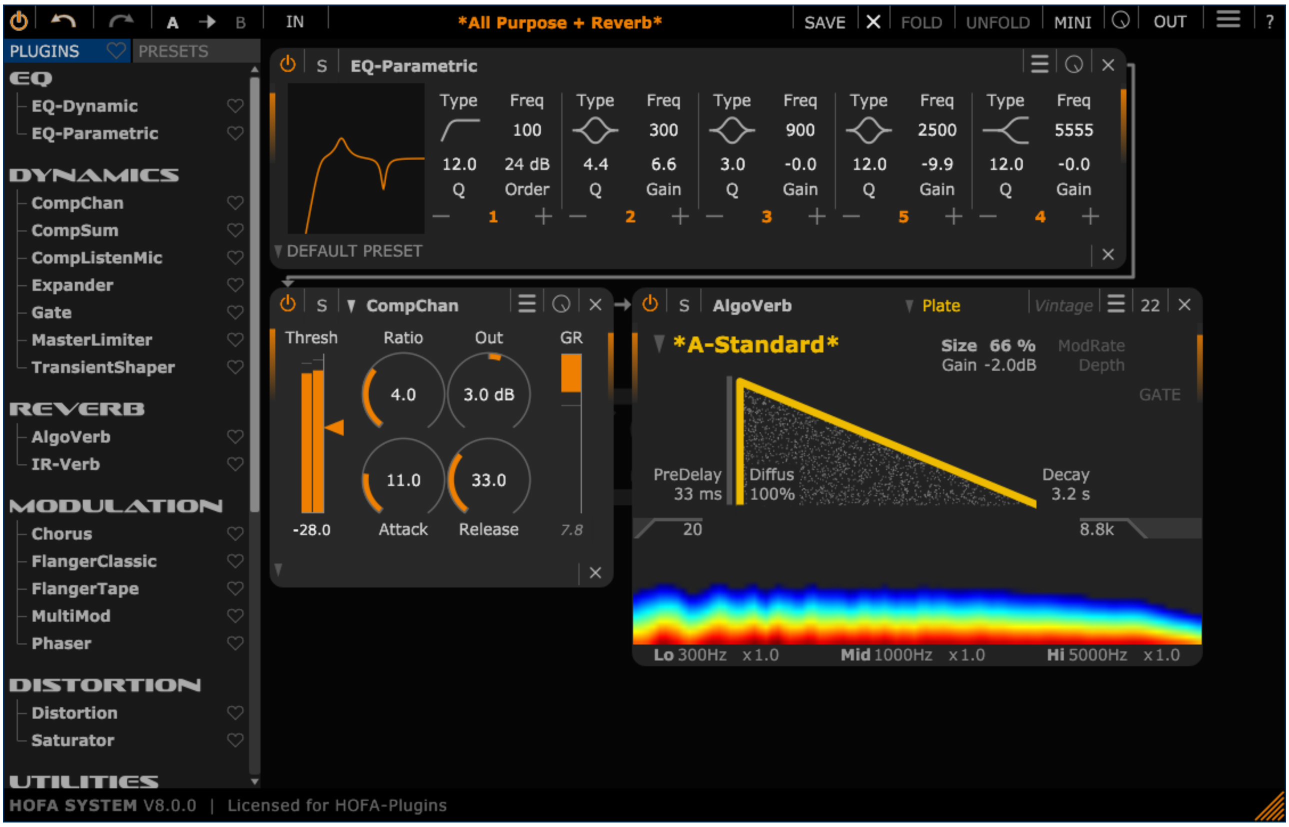


HOFA SYSTEM Plugin Bundle

User Manual

Version 8.0.7

HOFA-Plugins HOFA-College HOFA-Akustik HOFA-Media HOFA-Studios HOFA

Content

1 Introduction

The HOFA SYSTEM Plugin Bundle is a multi-effect plugin with a wide range of possibilities. The high-quality effects and plugins can be combined in any way within the HOFA SYSTEM user interface. In addition, all integrated effects are available as single plugins and can therefore also be used directly in the DAW (Digital Audio Workstation).

HOFA SYSTEM's straightforward workflow allows both serial and parallel signal processing to be carried out in no time at all. With the aid of routing plugins for mid/side, dual-mono, parallel and multi-band processing, also complex effect structures can be realized. HOFA SYSTEM even allows feedback loops. With just a few mouse clicks you can create your own effect chains, channel strips, mix and mastering setups and much more.

To keep a clear overview of extensive plugin arrangements, the folding function allows several SYSTEM plugins to be combined into a so-called folder. Any desired parameters of a folder can be assigned to a single controller, making it possible to control complex effects with just a few controllers.

All plugin parameters can be modulated individually and in many ways. Furthermore, all parameters can be set externally and automated in the DAW.

Another major advantage of HOFA SYSTEM is its flexibility regarding the production environment. Each preset created in HOFA SYSTEM can be loaded and reproduced in another HOFA SYSTEM - no matter what operating system or DAW is used. For example, it is possible to start production on a Windows computer with any DAW and continue it later on a Mac with another DAW - without having to recreate the familiar channel strips, mastering setups and routings for a long time. With HOFA SYSTEM you can simply take your usual workflow with you wherever you go.

2 Licenses

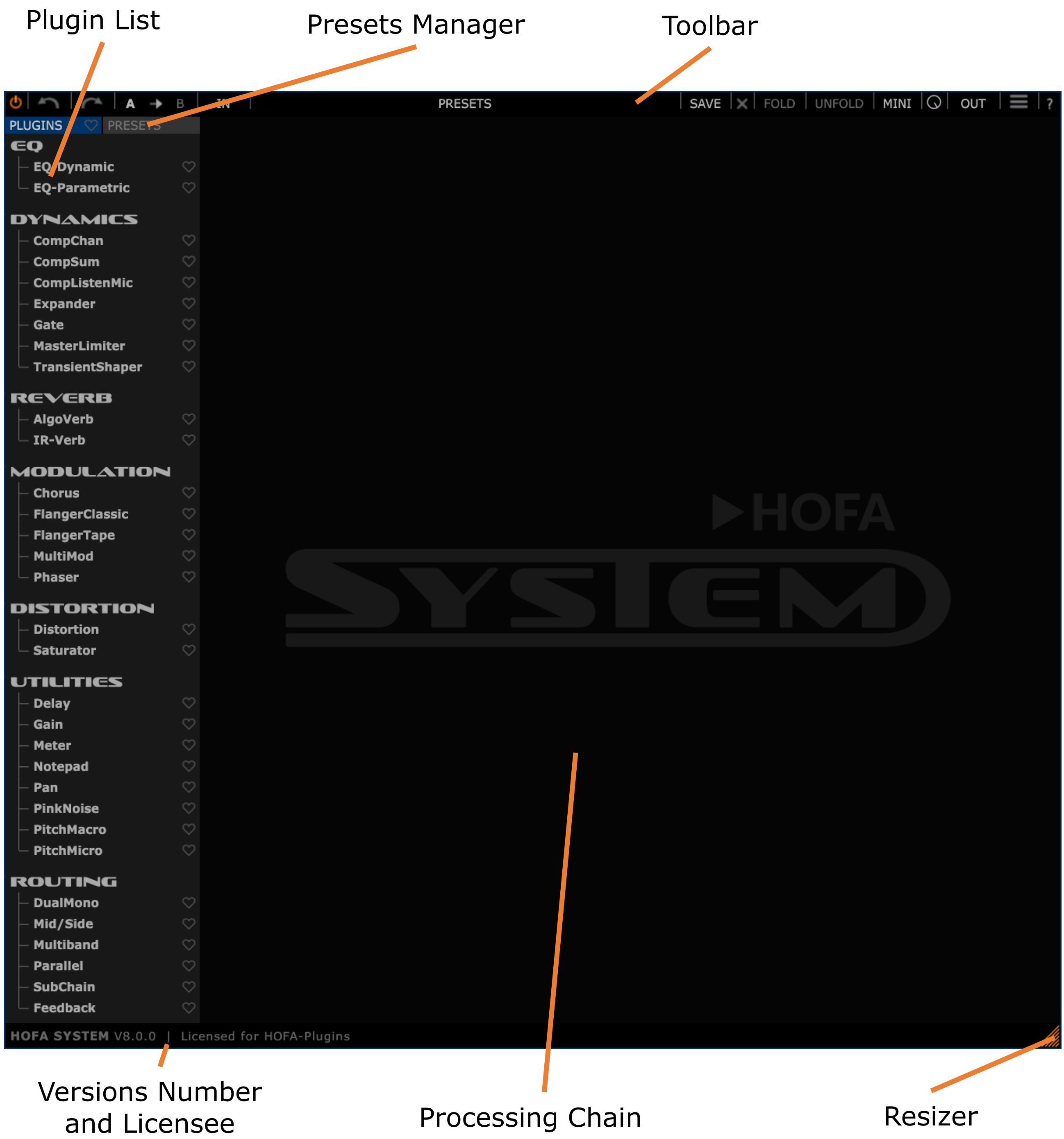
There are different licenses for HOFA SYSTEM. Basically, all integrated plugins are always available, but the range of functions may be limited for non-licensed plugins.

At the moment the following licenses are available, so that the corresponding plugins can be used without restrictions:

License	**Plugin(s)**
HOFA SYSTEM MixBox	CompChan, CompSum, Delay, Expander, EQ-Parametric, Gate, PitchMacro, Pan
HOFA SYSTEM AlgoVerb	AlgoVerb
HOFA SYSTEM PhaserChorusFlanger	Chorus, FlangerClassic, FlangerTape, MultiMod, Phaser
HOFA SYSTEM Saturator	Saturator
HOFA SYSTEM EQ-Dynamic	EQ-Dynamic
HOFA SYSTEM IR-Verb	IR-Verb
HOFA SYSTEM MasterLimiter	MasterLimiter
HOFA SYSTEM TransientShaper	TransientShaper

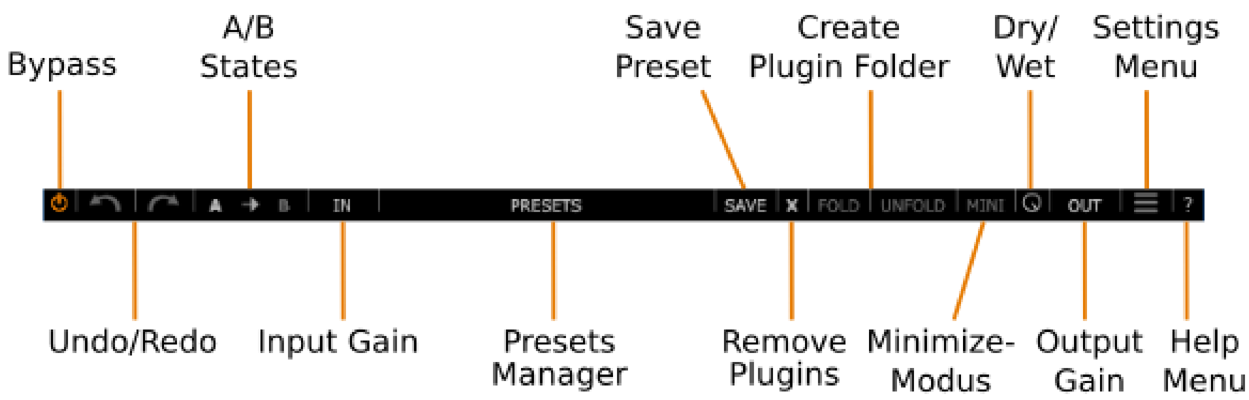
The activation of the licenses can be done in the HOFA Plugins Manager as well as directly in HOFA SYSTEM or in the single plugins.

3 Overview


Plugin List
Presets Manager
Toolbar
A B IN PRESETS SAVE FOLD UNFOLD MINI OUT ?
PLUGINS PRESETS
EQ
EQ-Dynamic
EQ-Parametric
DYNAMICS
CompChan
CompSum
CompListenMic
Expander
Gate
MasterLimiter
TransientShaper
REVERB
AlgoVerb
IR-Verb
MODULATION
Chorus
FlangerClassic
FlangerTape
MultiMod
Phaser
DISTORTION
Distortion
Saturator
UTILITIES
Delay
Gain
Meter
Notepad
Pan
PinkNoise
PitchMacro
PitchMicro
ROUTING
DualMono
Mid/Side
Multiband
Parallel
SubChain
Feedback
HOFA SYSTEM
HOFA SYSTEM V8.0.0 | Licensed for HOFA-Plugins
Versions Number and Licensee
Processing Chain
Resizer

3.1 Toolbar

The toolbar provides frequently used functions, which are explained below (from left to right):



- If the **Bypass** button is switched off, the signal processing in HOFA SYSTEM is bypassed.
- Using **Undo/Redo**, changes can be reverted or reapplied.
- The **A/B States** can be used to quickly compare different settings.
- **IN** is a gain control that can be used to adjust the gain on the input signal of the processing chain.
- If you press the **PRESETS** button, the global **Presets Manager** (see chapter 5) is displayed on the left side instead of the plugin list.
- **SAVE** stores either the currently selected plugins or the entire processing chain as a preset. Saved presets can be loaded in the Global Presets Manager (see chapter 5.1).
- The close button (**X**) removes either only selected plugins or all plugins from the processing chain..
- **FOLD/UNFOLD**: You can combine several plugins into one folder to create your own SYSTEM plugins. This is explained in detail in the chapter 8 .
- The **MINI** button activates or deactivates the Minimize mode. If it is enabled, all plugins are minimized to show the complete effect structure in the processing chain. More info in the chapter 9 .
- **Dry/Wet** offers the possibility to use HOFA SYSTEM parallel to the direct

signal, i.e. to mix unprocessed (dry) and processed (wet) signals.

- The **OUT** control adjusts the volume of the output signal. This controller is located behind the Dry/Wet controller and therefore also changes the volume of the unprocessed dry signal.
- The **Settings menu** contains additional settings. A detailed description can be found in the chapter 10 .
- The **?** button opens the help menu. This allows you to open the user manual, switch tooltips on/off and check for updates. In addition, this menu provides an overview of the various HOFA SYSTEM licenses, in which you can also set activations.

3.2 Plugin List and Processing Chain

At the center of HOFA SYSTEM's operational concept are the plugin list and the processing chain. In the **plugin list** at the left side you will find all plugins integrated in HOFA SYSTEM. Use the heart symbols in the list to highlight your personal favorites. Clicking on the heart at the top of the list will display only your favorite plugins.

The **processing chain** at the right side represents the signal flow and the user interfaces of the inserted plugins. The plugins are inserted into the signal flow by dragging them from the plugin list into the processing chain. Within the processing chain, the plugins can be arranged at will and so the signal flow can be changed.

3.3 Creating Processing Chains

As already mentioned in the previous section, the plugin list and the processing chain are the main "tools" for creating effect chains. All plugins listed in the plugin list can be drag-and-dropped into the processing chain and are integrated into the signal flow in this way. Depending on the selected width of the HOFA SYSTEM interface, a plugin is inserted into a new "line" to avoid horizontal scrolling.

A newly inserted plugin can be positioned either serially or parallel to an existing plugin. In the processing chain, the insertion point is indicated by an orange line. If the line is vertical, the plugin is inserted serially to an existing plugin. When a horizontal line appears, the insertion is parallel. A parallel arrangement of plugins can also be done using the Parallel or Multiband plugins, see sections 4.2.3 and 4.2.4 .

When parallel inserting plugins via drag-and-drop, please note that the multiband mode is the default setting here. In the resulting Multiband plugin (see section 4.2.3) the input signal is first divided into several frequency bands. Each frequency band can now be edited separately. Then the signals of the individual frequency bands are added together so that a single (total band) signal results "at the end" again. If, however, true parallel processing of the signal is desired, you must switch from "Multiband" to "Parallel" in the header of the Multiband plugin.

Within the processing chain, the plugins can be arranged freely by dragging and the signal flow can thus be changed. To do this, move the mouse over a "free area" in the plugin interface (e.g. on the header). If the mouse cursor is displayed as a hand, the plugin can be "grabbed" and **moved** by clicking on it. While holding down the Ctrl key (Windows) or the Cmd or Alt key (macOS), the plugin is **copied**. In this case, a + sign appears at the insertion point in addition to the vertical or horizontal line.

You can also move or copy more than one plugin by clicking with the mouse on a free space in the processing chain and dragging a frame around the corresponding plugins while keeping the mouse button pressed. The selected plugins are then marked by an orange frame and can now be moved or copied - like one single plugin.

Besides moving and copying plugins, it is also possible to **replace** existing plugins by dragging the new plugin directly onto another one.

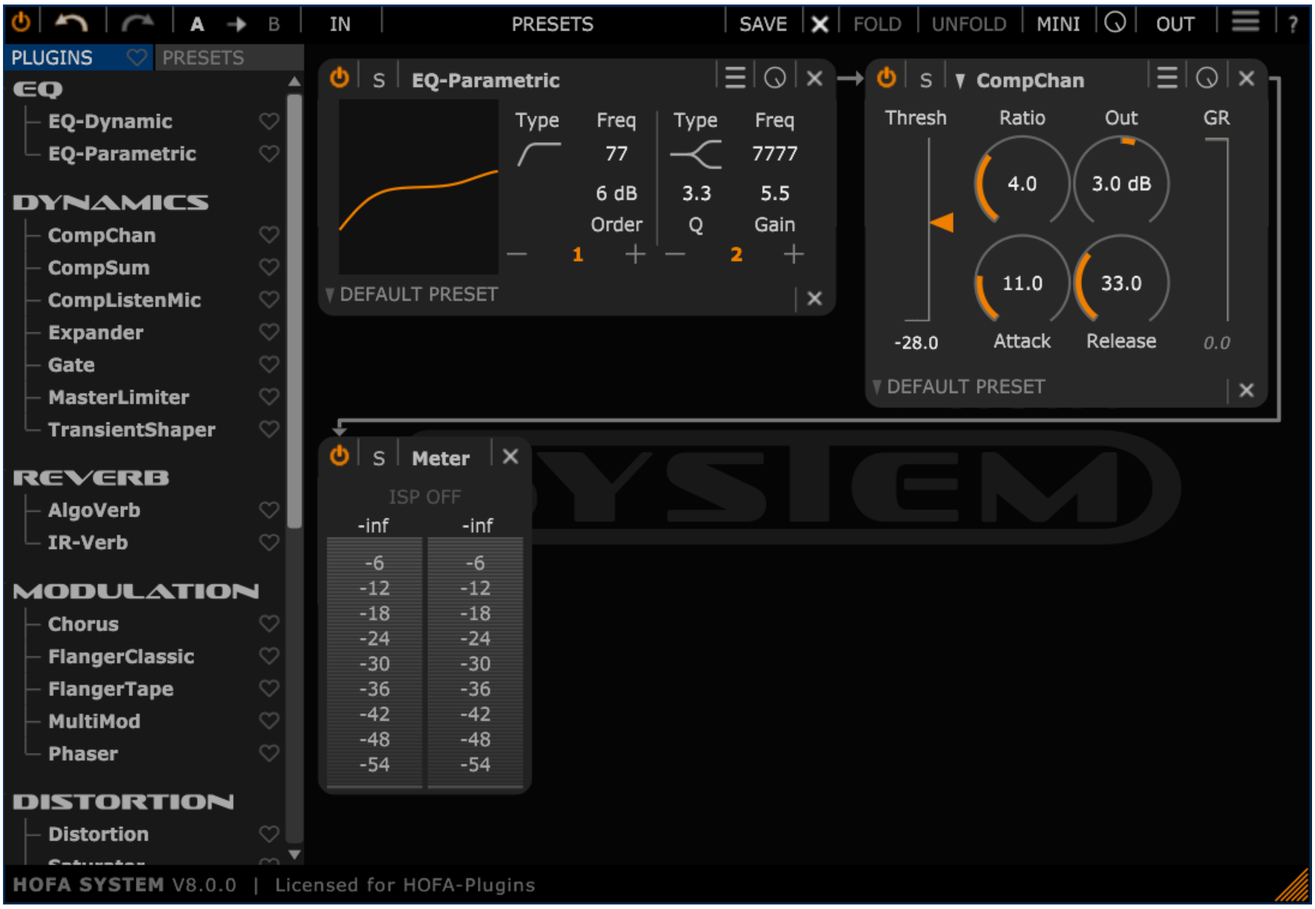
To **remove** plugins from the processing chain, press the Close button (**X**) in the header of the respective plugin. In addition, the plugin can also be selected by clicking and deleted via the global Close button (**X**) in the HOFA SYSTEM toolbar.

Thin arrows between the plugins illustrate the created signal flow. They can be shown or hidden in the Settings menu under "Show Routing Arrows".

The following two sections explain two simple examples of processing chains.

3.3.1 Channelstrip

The following picture shows the serial signal flow of a simple channelstrip - consisting of EQ → compressor (CompChan) → meter:



To create the channelstrip, the individual plugins are simply dragged one after the other from the plugin list into the processing chain and arranged in series. Pay attention to the vertical line at the insertion point.

3.3.2 Multiband Compressor

The following illustration shows a multi-band compressor. In order to create it, the CompChan plugin is first dragged from the plugin list into the processing chain. Then a second CompChan is placed below the first plugin. When inserting the second plugin, pay attention to the horizontal line at the insertion point, which indicates that the insertion is parallel. Because the multiband mode is set by default, you don't need to do anything else.

With the newly created multiband compressor, the input signal is divided into two frequency bands with a crossover frequency of 632 Hz. The upper CompChan plugin compresses the upper frequency band (frequencies above 632 Hz), the lower plugin compresses the lower frequency band (frequencies below 632 Hz). In this example, the crossover has an edge steepness of 12 dB/oct.

3.4 Presets Manager

See chapter5 .

3.5 Version Number and Licenseholder

In the lower left corner of the HOFA SYSTEM user interface you will find the version number and the holder of the license. A click on this section opens another window with detailed information about the licenses of the individual products, in which you can also make activations. In case of problems, the necessary information can be quickly communicated to the support.

3.6 Resizer

The window size of HOFA SYSTEM can be changed. Click on the small orange triangle in the lower right corner of the user interface and set the desired size with the mouse button pressed.

4 HOFA SYSTEM Plugins

The HOFA SYSTEM plugins are divided into effect plugins and routing plugins. The former are audio effects. In contrast, routing plugins can hold such effect plugins and allow different routings. More details can be found in section 4.2 .

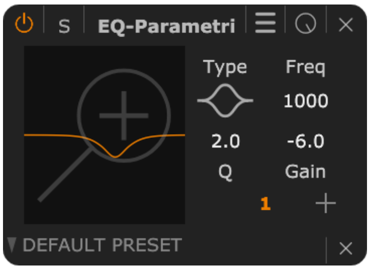
All HOFA SYSTEM plugins have a header with the following controls (from left to right):

- **On/Off-Switch**: If the switch is off, the plugin is not used and signal processing is bypassed.
- **Solo**: All other modules will be switched off.
- **Plugin Name:** In the screenshot above the plugin name is "Delay". Some plugins allow you to switch a mode (e.g. PitchMicro/PitchMacro).
- **Settings**: On the one hand, the default settings for the plugin can be saved and loaded here. On the other hand, both the Dry/Wet control and the Presets button for opening the Presets Manager (see chapter 5 **Fehler! Verweisquelle konnte nicht gefunden werden.**) can be s hown and hidden. Furthermore, the menu contains individual settings for the plugin.
- **Dry/Wet Control**: This control allows fading of unprocessed (dry) and processed (wet) signals. It can be shown or hidden in the Settings menu.
- **Close Button (X)**: Removes the plugin from the processing chain.

The effect plugins are sorted by category and are described in the next section. The subsequent section contains explanations of the routing plugins.

4.1 Effect Plugins

4.1.1 EQ-Dynamic/EQ-Parametric



EQ-Dynamic and EQ-Parametric are equalizers with up to eleven fully parametric filter bands. Besides the dynamic function in the EQ-Dynamic, both plugins essentially work the same.

Only one filter band is available in HOFA SYSTEM FREE. An EQ-Dynamic license (respectively a MixBox license for the EQ-Parametric) is required to use all eleven filter bands.

After insertion, the plugins both show one filter band only. Use the **+** and **–** buttons to insert or remove additional filter bands. The processing order of the filter bands can be changed by drag-and-drop. This is especially relevant, when using the dynamic function. Clicking the band number will activate or deactivate that band.

The **Type** selector allows you to choose one of 9 filter characteristics:

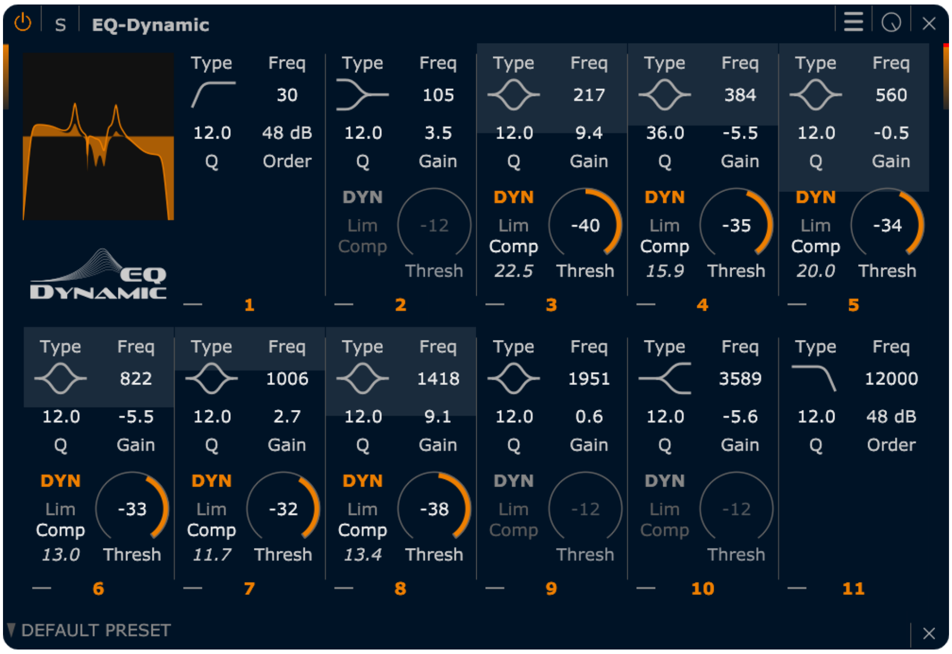
- High-pass
- Low-pass
- Band-pass
- Peak/Notch
- Low-shelf and Low-shelf resonant
- High-shelf and High-shelf resonant
- Flat

Click **Order** to change the slope of high-pass and low-pass filters. You can choose 6, 12, 24 or 48 dB/octave. If you choose 12 dB/octave or higher, the quality **Q** can be adjusted as well.

The dynamic function of EQ-Dynamic is activated by the **DYN** button. The **Thresh** knob is used to adjust the threshold that determines the level at which the dynamic function will start to work.

There are two types of dynamic gain reduction available. **Comp** has a more compressor-like characteristic, while **Lim** can be used for harder limiting in the frequency band.

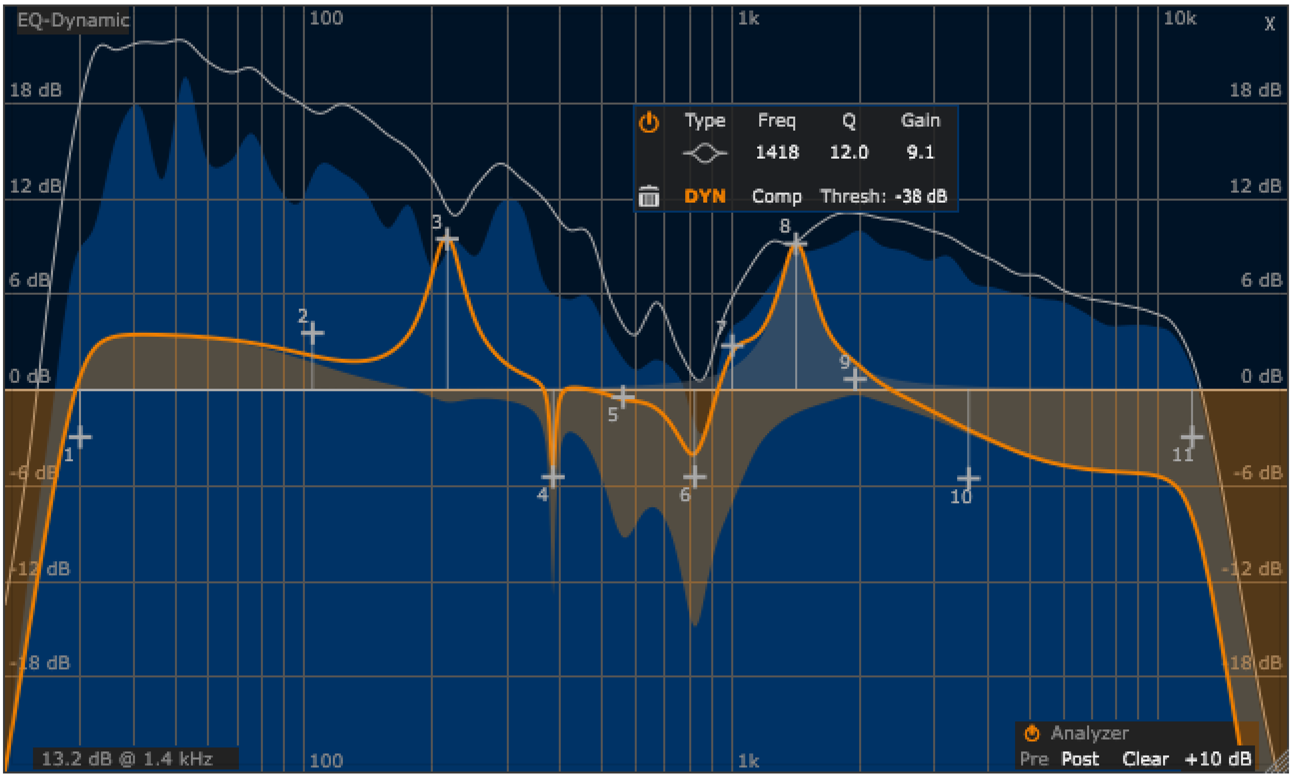
The current gain reduction is shown in each filter band. Its peak value can be seen below the **Lim/Comp** switch. This visualization can be switched off using the menu entry **Show Gain Reduction in Bands**.



The sidechain signal used to control the dynamic function is the input signal of each band, filtered appropriately. Because of this, the processing order of the filter bands is important as mentioned above. The menu entry **Use External Sidechain** allows using the external sidechain signal provided by the DAW alternatively.

The frequency response display at the left shows the curve of the EQ as a line. When using dynamic bands, the resulting frequency response is shown as a coloured plain.

Clicking on the frequency response display (magnifier glass) opens a separate window that also displays the overall frequency curve. Again an orange line represents the static response and the plain represents the dynamic one.

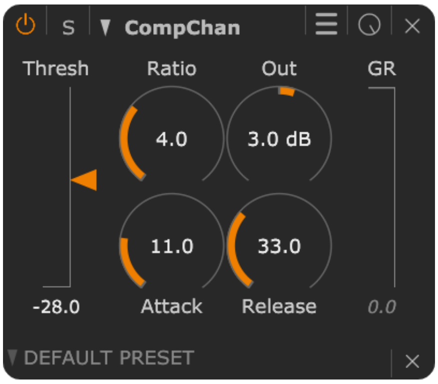


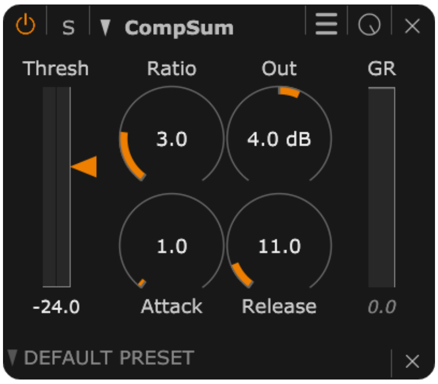
In this view, the individual filters can be set graphically by clicking on the corresponding cross and dragging it to the desired frequency and the desired gain or attenuation. Click on the cross to switch the respective filter on or off. A double-click will add a new band at the current position or delete the one that is already there. When holding the shift key, a vertical movement will change the Q factor instead of the gain. This can also be adjusted using the mouse wheel. When approaching a band with the mouse, a panel will show up that contains all controls for that band and the frequency response of that single band is highlighted.

In addition, this window includes a spectrum analyzer that can be adjusted by the controls at the bottom right. The **Pre** and **Post** buttons allow selecting if the signal is shown before or after it has been processed by the equalizer. By clicking on the **Clear** button, the grey peak line in the signal spectrum can be deleted/reset. The slider to the right of the Clear button allows vertical shifting of the signal spectrum.

Furthermore, the gain and frequency of the current mouse position are displayed in the lower left corner, allowing peaks in the spectrum to be detected precisely.

4.1.2 CompChan/CompSum





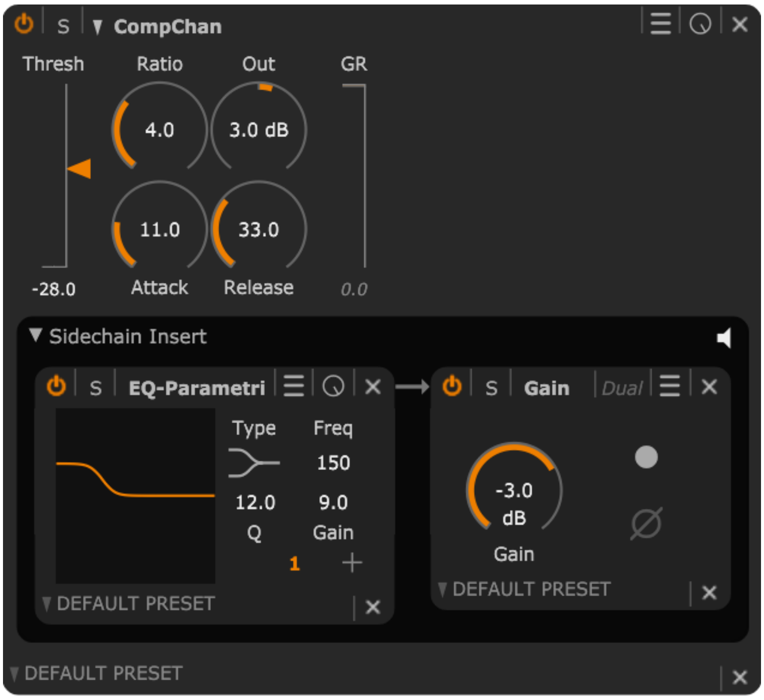
The CompChan/CompSum plugins are typical channel/bus compressors with controls for threshold, ratio, attack, release and output gain.

Both compressors work adaptively due to which the attack and release times aren't given in milliseconds but are relative values.

The input level is shown in the **Threshold** control. On the right side the current gain reduction (**GR**) and its peak value are shown. The peak value can be reset by clicking the gain reduction meter.

By default the sidechain signal used for dynamic processing is the input signal of the CompChan/CompSum. In the Settings menu you will find two entries to change this.

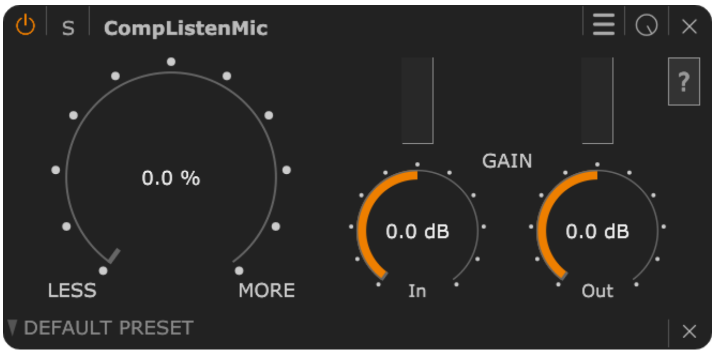
The option **Use External Sidechain** allows using the external sidechain signal provided by the DAW instead of the input signal. In this case the **Threshold** control is displayed in grey instead of orange.



If **Enable Sidechain Insert** is activated, the „Sidechain Insert" area appears at the bottom, which will be empty by default. You can drag any SYSTEM plugin into this area, which will then be used to filter the selected sidechain signal before dynamic processing. Even complex combinations of plugins are possible here. In order to keep the overview, the loudspeaker symbol in the upper right corner allows listening to the filtered sidechain signal directly. The arrow in the upper left corner allows folding the Sidechain Insert area to save screen space while the filtering remains active.

Without a MixBox license, Ratio and Release are fixed and can't be changed.

4.1.3 CompListenMic



The CompListenMic is an emulation of a famous talkback compressor. Besides in- and output gain the degree of compression can be set.

The module is well suited to compress drum room signals.

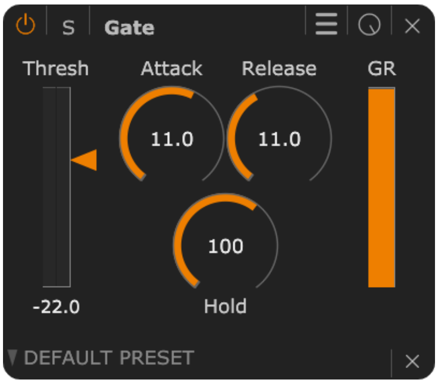
4.1.4 Expander

The Expander plugin provides an upwards expander, i.e. loud parts will be further increased in loudness. This can be used for sound shaping, for example to further enhance transients. Available controls are threshold, ratio, attack and release. As the dynamics are very adaptive, the attack and release times are set relatively instead of milliseconds. On the right side a display shows the currently applied **Gain**. The gain modification is limited to 24 dB.

By default the sidechain signal used for dynamic processing is the input signal of the Expander. In the Settings menu you will find two entries to change this, which are explained in chapter 4.1.2 .

In order to use the Expander to its full extent, a MixBox license is required. Otherwise, only threshold can be set.

4.1.5 Gate



This is a simple gate. The threshold sets the level above which the gate opens. Longer attack times let the gate react slower so short signals will still be blocked even if they exceed the threshold. After the gate opens it will stay open for the hold time while the release time determines the time the gate takes to close again.

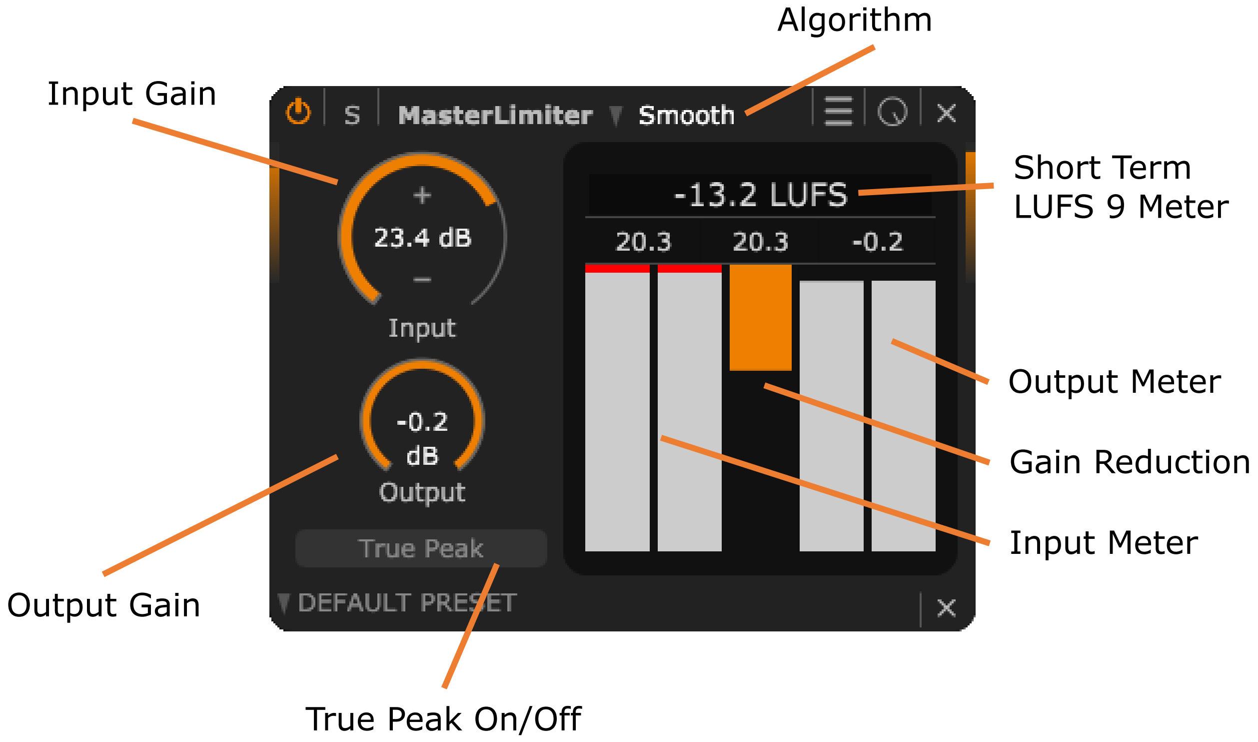
All times are given in milliseconds.

By default the sidechain signal used for dynamic processing is the input signal of the Gate. In the Settings menu you will find two entries to change this, which are explained in chapter 4.1.2 .

Without a MixBox license only threshold can be adjusted.

4.1.6 MasterLimiter

The MasterLimiter is a brickwall limiter optimized for mastering. It prevents your signal from going beyond a certain value, set by the output knob.
The following controls are available to you:



Algorithm: The algorithm control lets you switch between 3 different modes. You can find a detailed description in the usage section of this chapter. When using the SYSTEM FREE version, only the selection "Smooth" is available.

Input Gain: The input gain determines how loud the signal goes into the limiter. You can check the input level by looking at the **input meter** on the left side.

Output Gain: The output gain determines the resulting level after the limiting process. In the **output meter** on the right side, you can see how loud your signal is.

In the middle of the meter section, you can see the **gain reduction**. It tells you how much the limiter turns your signal down, in order to keep the level below the limiting threshold.

A **Short Term LUFS 9 meter** is located at the top. It shows perceived loudness. For details more about LUFS metering see chapter 11.2 .

True Peak On/Off: Here you can set if the MasterLimiter should do True Peak limiting and remove intersample peaks. For details see chapter 11.1 .

Even though all intersample peaks are removed at this stage, any further processing of your signal is able to introduce new ones. To avoid any intersample peaks, the True Peak limitation is done at the very end of the SYSTEM signal chain (even after the final downsampling, if oversampling is active). For this reason, the True Peak mode is only available if the MasterLimiter is the last plugin in your SYSTEM signal chain.

Histogram

To open the histogram view, simply click on the level meters. The histogram shows you a detailed loudness trend of your audio over time. In the upper half you can see the gain reduction in dB. In the lower half you can see your input signal, the output signal and the measured loudness value in LUFS.

Usage

Unfortunately, it's mandatory to distort the signal if you want to limit it. But just how much distortion is too much? This is where the MasterLimiter comes into play! The intelligent plugin seeks out when the track is distorted too much virtually by itself. At a certain point, it won't make your track any louder, to save it from distortion - no matter how much you turn up the input signal. This means you should only put in so much gain that the limiter reaches this point. Otherwise, you will destroy the global dynamics of your track.

How much distortion is tolerated, is depending on the limiter algorithm.

Smooth: Allows almost no distortion at all. Vice versa this algorithm allows the least amount of increased loudness. It's used best on material that's vulnerable to distortion, like choir recordings for example.

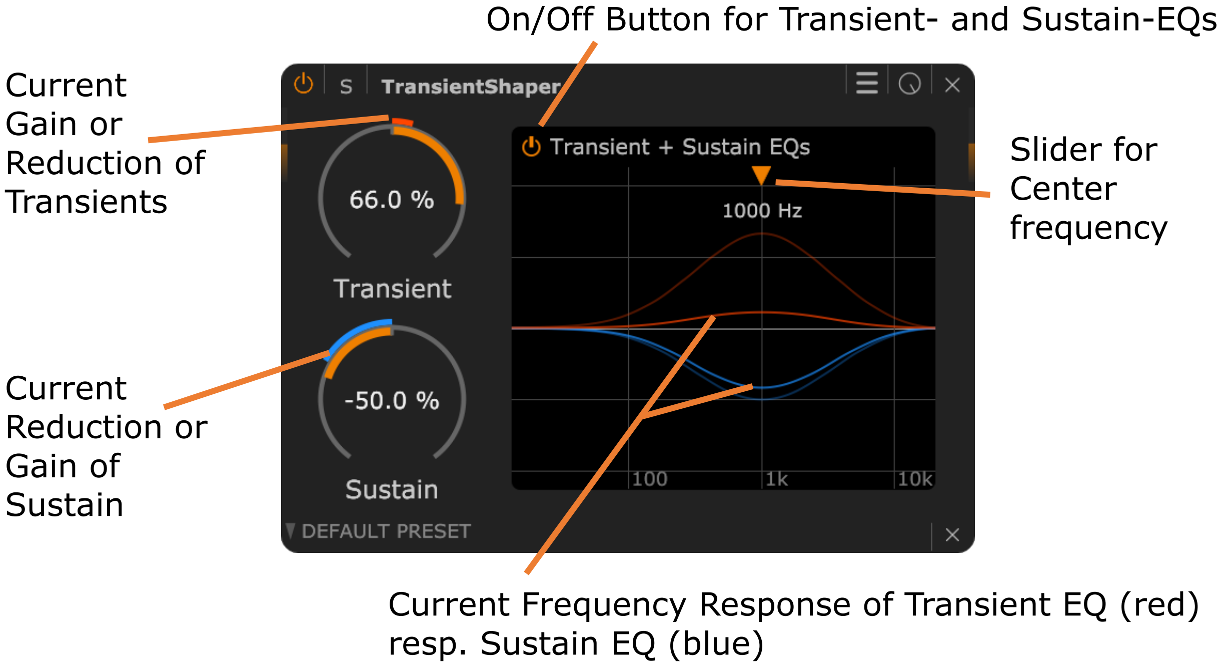
Transparent: This mode gives you the best compromise out of loudness and clean sound. It should work well in 98% of all cases.

Loud: This mode works best if loudness is your primary goal without overexaggerating. But you have to expect some shifts in your global dynamics.

Pump: The Pump mode excels at processing bass heavy music where it´s important to leave the sub pure without distortion and give it a good amount of pump. Use this mode if you need a sound-defining limiter that glues everything together.

Hard: The hardest of our brick wall algorithms! This mode will push every audio material to it's maximum loudness with minimal colouring!

4.1.7 TransientShaper



With the TransientShaper, transients and sustain phases in the audio signal can be amplified or reduced by up to 18 dB. There are two controls for this: **Transient** and **Sustain**. Processing is independent of the level of the input signal.

In general, the following applies:

- Transient boost / sustain cut ⇒ Sound comes to the fore.
- Sustain boost / transient cut ⇒ Sound fades into the background.

In addition, the TransientShaper offers the possibility of frequency-selective processing of the transients and sustain phases. To do this, activate the **Transient + Sustain EQs**. These are dynamic peak filters, the gain of which depends on the detected transients or Sustain phases is regulated. The center frequency of both EQs can be adjusted by dragging the orange triangle in the frequency chart. For example, the kick of a bass drum can be precisely emphasized by adjusting the center frequency to about 4 kHz and **Transient** is set to a positive value. The Transient EQ can also be used to give a snare more "crack" without distorting the rest of the sound as a normal EQ would do.

The areas of application of the TransientShaper are very diverse:

- For example, the transients of drum and percussion sounds can be boosted to increase their presence in the mix. Likewise, the transients can be lowered so that the sounds fade into the background. In this way, percussive instruments can be optimally placed in the mix. Both individual drum tracks and complete drum kits can be edited.
- To bring clarity to a mix, the reverberation of drum tracks (such as snare, toms, overheads) can be reduced by lowering sustain.
- The picking sound of plucked instruments can be emphasized by raising **Transient**. In the same way, you can give a slap bass more punch.
- By editing the transients and sustain phases, you can greatly change the character of a sound. Thus, the TransientShaper is perfectly suited for sound design.
- There are many more possible applications. To get an impression, please refer to the presets.

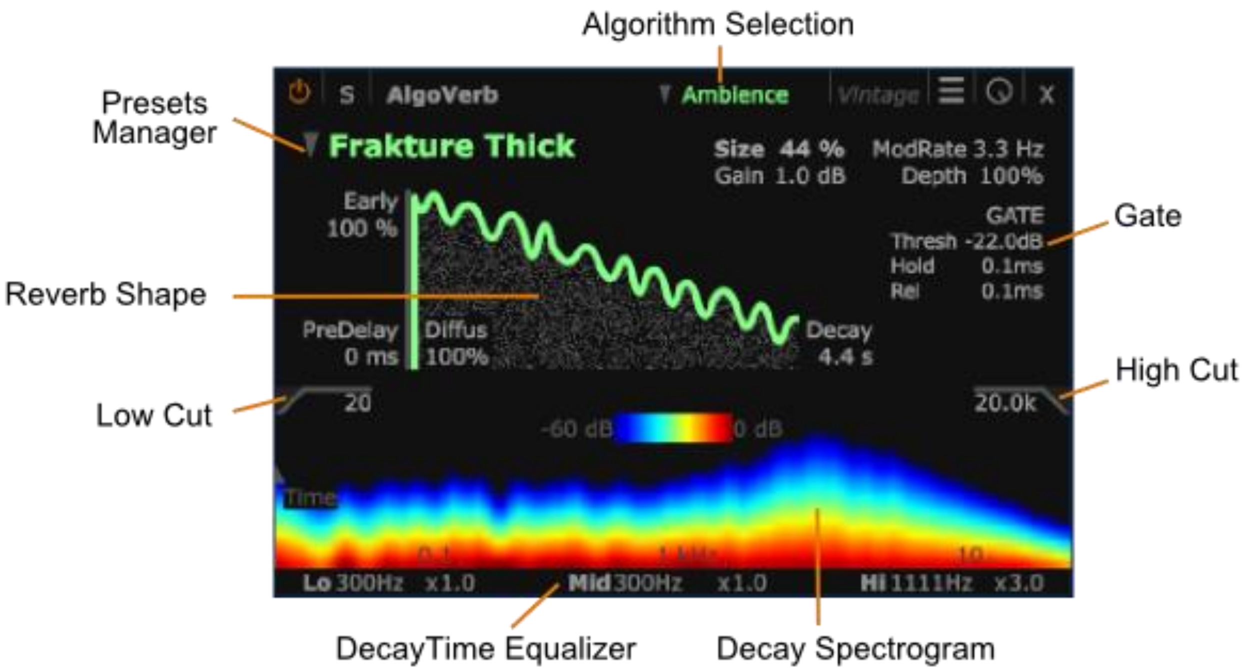
As with the other dynamics plug-ins (CompChan, CompSum, Gate, etc.), the options **Use External Sidechain** and **External Sidechain Insert** are also available for the TransientShaper via the settings menu. The explanation of these options can be found in section 4.1.2 .

In SYSTEM FREE (without a TransientShaper license), only the Transient slider is available.

4.1.8 AlgoVerb

AlgoVerb is an algorithmic reverb with ten unique reverb algorithms. In addition to the usual standard functions, such as pre-delay and decay time, the AlgoVerb plugin has a novel DecayTime Equalizer. This allows the decay time to be designed differently for various frequencies. In addition, there is an integrated gate to generate e.g. the classic drum reverb of the 1980s.

The AlgoVerb plugin is divided into the following controls:



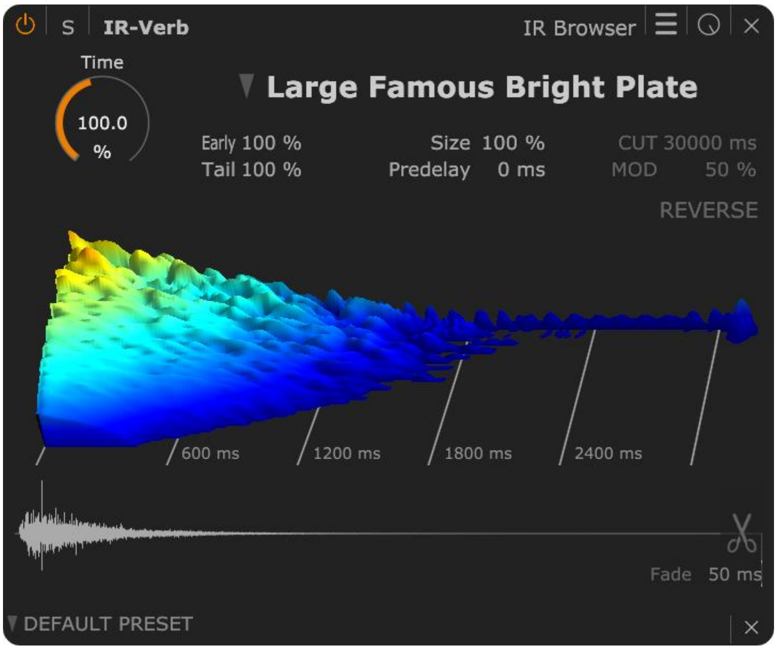
- Ten different reverb algorithms are available in the **Algorithm selection**. Without AlgoVerb license, only the Plate algorithm is available.
- If the **Vintage** mode is active, the signal processing takes place with a sampling frequency that was typical in the 70s. In addition, the resolution of the modulation is reduced.
- **Presets** shows the name of the current preset. By a mouse click the Preset Browser opens (see chapter 5). There are over 150 HOFA presets to choose from, demonstrating AlgoVerb's wide range of capabilities. You can also use this function to manage your own presets.
- The **Reverb Shape** section provides familiar parameters to adjust the

sound of the AlgoVerb. For this purpose there are controls for PreDelay, Early Reflections, Diffusion, Decay, Size, Modulation Rate and Depth. By clicking on **ModRate** or **Depth** the modulation can be switched off or on.

- The volume of the reverberated output signal can be adjusted via the **Gain** control. In the settings you can define whether the amplification takes place before or after the Dry/Wet control.
- The **Hicut** and **Lowcut** controls cut the spectral range in the high and low end.
- The **Decay Spectrum** shows the reverb's spectral decay where the horizontal axis represents the frequency and the vertical axis the time. High levels are shown red and decay to blue.
- The **Gate** can be used to create classical gated reverbs. The familiar gate parameters Threshold, Hold- and Release-Time are available for this. The gate is controlled by the input signal, which is not reverberated.
- Below the Decay Spectrum are the controls of the **DecayTime Equalizer**. It can be used to set frequency-dependent decay times. For treble and bass there is a **Hi**gh and **Lo**w shelf and a peak filter (**Mid**) in the mid range. The decay can be influenced relatively within a range between a quarter and four times the set decay time. For example, it is possible to set the decay time in the mid range only half as long as the total decay time, but twice as long in the low end.

4.1.9 IR-Verb

The IR-Verb is an Impulse Response (IR) player based on the renowned HOFA IQ-Series Reverb. It comes bundled with 135 versatile Reverb-IRs (13 with FREE version) suitable for all your production needs. If IQ-Series Reverb is installed and activated on your system, all its IRs (including any activated IR-Packs) are also available to IR-Verb. Besides that, it's possible to load your own IRs and store them in a custom user library for quick and convenient access.



Main component of the IR-Verb's user interface is the large waterfall diagram in the center. It visualizes the frequency spectrum over decay time of the current IR. Time is plotted to the right, amplitude to the top and frequency out of the display plane. You can hide the waterfall diagram by clicking **HIDE 3D** and thereby save some space in the user interface. Below the waterfall diagram there is an additional two-dimensional representation of the waveform which will remain visible also if the 3D diagram is not shown.

Controls

The following controls can be used to manipulate the currently loaded IR:

- **Time** controls the reverberation time by applying gain or damping to the IR over the decay time. This is equivalent to changing the damping of a real room e.g. by adding or removing furniture.

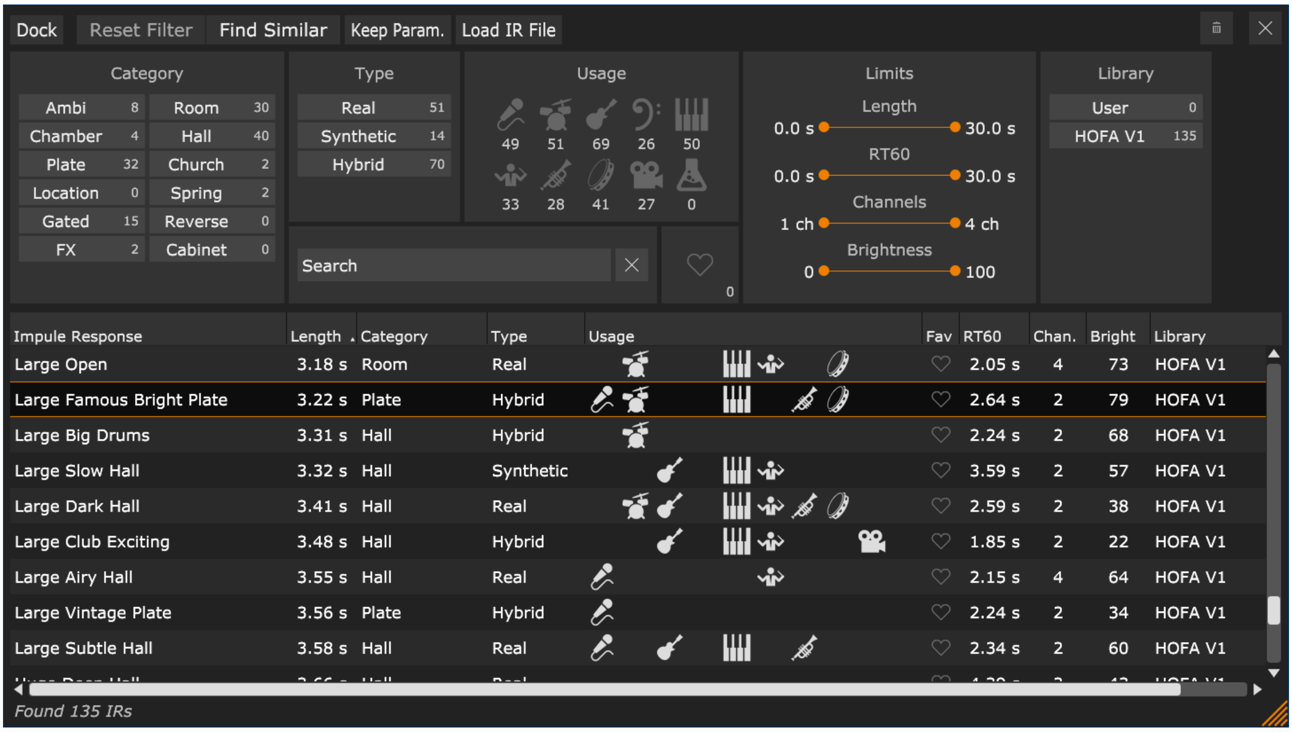
Early Reflections/Reverb Tail
The decay time of a reverb can be divided into two different stages: Early reflections and a diffuse reverb tail. Early Reflections are the parts which are reflected only once or a few times. The human brain uses these to extract important information about the directions of the sound. The diffuse reverb tail is the sum of all superimposed reflections over a long period of time.

The **Early** and **Tail** controls regulate the level of early reflections and reverb tail independently. If the value for the early reflections is set to **Off**, they will be removed from the IR entirely, moving the reverb tail to the start of the IR.

- **Predelay** inserts an additional delay before the early reflections.
- The **Size** parameter changes the length of the decay time by stretching the IR. This is equivalent to “changing” the size of a real room.
- The **CUT** control allows cutting the IR to a certain length, e.g. to create a classic gated reverb sound. Clicking on CUT bypasses it without changing length value.
- **MOD** enables a subtle modulation of the processed signal. Clicking MOD disables the modulation without changing the intensity value.
- The **REVERSE** button allows for creative effects by loading the IR backwards.

Loading an IR from the browser resets all these parameters to their defaults. To prevent this behavior, just enable **Keep Parameters** in the IR-Browser.

IR Browser



The IR-Browser makes it easy to find the perfect IR for any situation by providing several filter and sort options. It can be opened by clicking **IR Browser** in the toolbar, by clicking on the name of the currently loaded IR or by clicking into the waterfall diagram. The browser will also open automatically when a new IR-Verb instance is being loaded. This can be disabled in the menu.

When opening the browser in a new instance for the first time, its position will be linked to the plugin's position: If the plugin is moved, the IR browser will follow. By moving the browser manually, the linkage will be broken up and the browser's position and size will be handled independent from the plugin position. To dock the browser to the plugin again, click **Dock** in the IR browser.

Depending on the available space, the filter options will be displayed at the top or left hand side of the browser window. Filters can be applied for category, type, library, suggested usage and several numeric parameters. You can also search for a name or parts of it using the full-text search. All active filters can be cleared by clicking **Reset**. Opposed to that, to find similar IRs like the one currently loaded, click **Similar**.

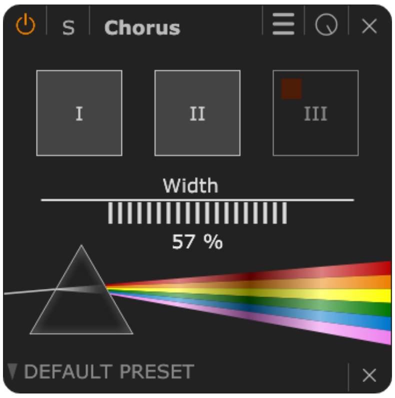
Using your own IRs

Besides using the IR libraries curated by HOFA, you can also provide your own impulse responses. In the IR browser, use the button **Load IR File** to open a file dialog in which you can choose an arbitrary audio file. It will be loaded directly into the IR-Verb. By default, an IR loaded this way will also be copied into the DAW project and hence stay available even when moving the project across different computers. The drawback of this is, that the DAW project file can grow quite huge. Copying the IR into the project file can therefore be disabled in the menu on a per-instance basis. This will force the plugin to load the IR directly from the filesystem.

You can also Drag’n’Drop audio files into the IR browser window to either load them directly or add them to the User Library. The attributes of an imported IR can then be edited within the browser in order to benefit from the browser’s filter and sorting features while searching for IRs.

To get rid of an imported IR, just hit the trash can icon when it is selected in the browser.

4.1.10 Chorus

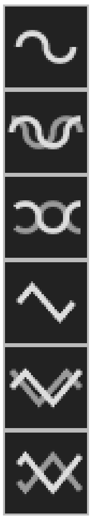


The Chorus-Plugin is a two-voice chorus. The second voice can be switched on by pressing the **I** button. If you click on button **II**, a part of the modulated signal is fed back to the input. Button **III** changes the modulation frequency. Use the **Width** control to adjust the stereo width.

4.1.11 FlangerClassic

This is a classic flanger effect. Basically, the wet signal is delayed by a modulated delay time. The mix of wet and dry signal results in the typical flanger sound.

The Delay control sets the center delay time in milliseconds. The modulation amount is then controlled by the Depth control in percent of the delay time.

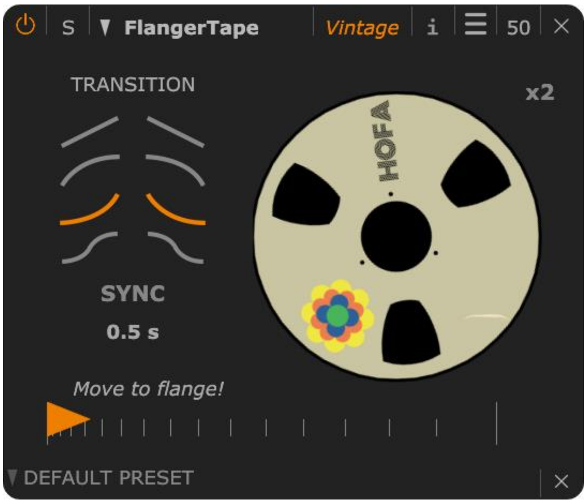


The different modulation types are shown on the right. Besides the selection between sine and triangular shape it is also possible to introduce a phase shift of 0°, 90° or 180° between the modulators for left and right channel.

The Feedback control sets the amount of output that is fed back to the input. In case of a phase-shifted modulation the channel's output is sent to the other channels input if the **CROSS** button is active.

The modulation frequency in Hz is set by the **Rate** control. If **SYNC** is active, the tempo is synced to the song tempo of the DAW. In this case, instead of a frequency a note length is set, for example 1/4D for a dotted quarter note.

4.1.12 FlangerTape



The FlangerTape plugin produces the classic tape-flanger sound of the 1960/70s. This was created with the help of two tape machines that played identical recordings at the same time. The speeds were varied by pressing the reels of both tape machines alternately with the finger. This resulted in a changing delay between the two signals, which caused the flanger effect.

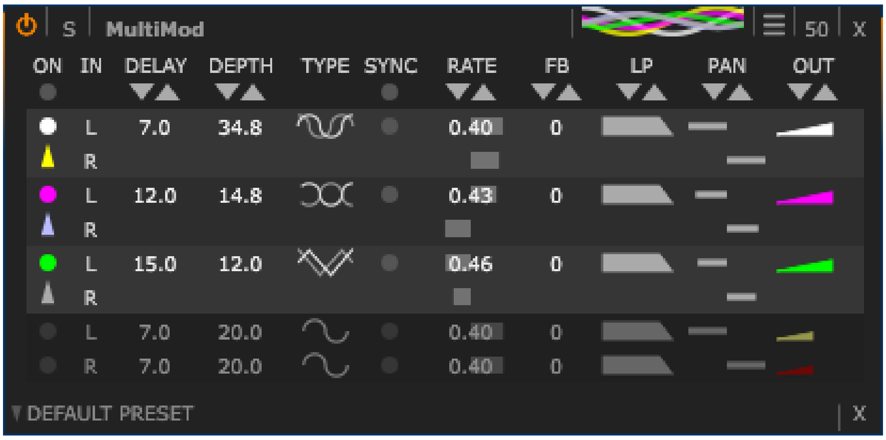
This delay can now be easily adjusted with the FlangerTape plugin using the **Move to flange!** control by moving the orange triangle. The triangle determines the scheduled value and the grey bar shows the current value of the delay. The type of transition from scheduled to actual value can be set using the **Transition** curves. For this purpose, the buttons for the attack curves are located on the left side and the buttons for the release curves on the right side. The transition duration is defined by the value below the **SYNC** button.

By activating **SYNC**, it is possible to synchronize the transition time to the DAW's song tempo. In this mode, the note duration can be selected (instead of seconds), e.g. 1/2D for a dotted half note.

If you press the **x2** button in the upper right corner, the tape-flanger sound becomes even more intense.

The **Vintage** button is located in the top bar of the module. If this is activated, a tape emulation is activated. This emulates not only the frequency response of a tape machine but also wow and flutter and tape saturation.

4.1.13 MultiMod



The MultiMod plugin is a versatile tool to create flanger and chorus effects. It provides eight mono voices whose delay times can be modulated. Furthermore, feedback, lowpass, panorama and gain can be set per voice.

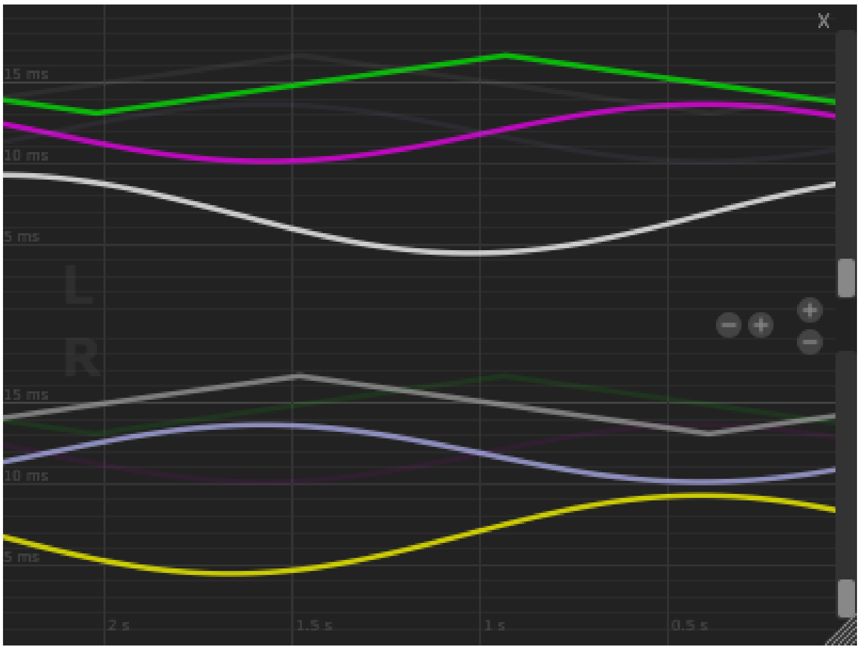
Below the column header are controls to globally change parameters for all voices at the same time.

Each voice has the following parameters:

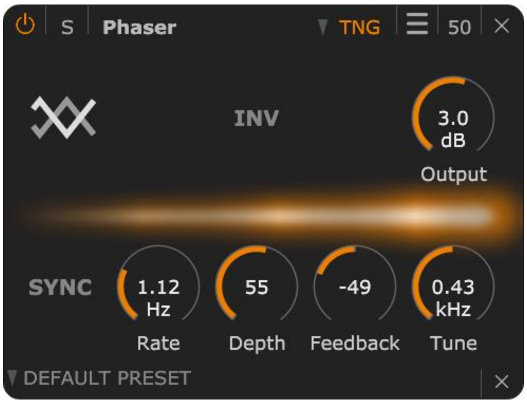
ON	Switches a voice on or off. If two voices are linked (by selecting a double-waveform modulation type), this is indicated in the lower voice by an arrow on the upper voice.
IN	Chooses the input signal. This can be either left (L), right (R) or mid (M).
DELAY	The voice's delay time in milliseconds.
DEPTH	The modulation depth in percent of the delay time.
TYPE	Determines the waveform of the modulation LFO. For double waveforms, a phase shift is generated between the left and right channels. This allows stereo effects to be created. In this case, two voices are connected, whereby the controls of the upper (master) voice then also apply to the lower (slave) voice.
SYNC	When SYNC is enabled, the modulation frequency (RATE) is synchronized with the tempo of the DAW and can be specified as a note value.
RATE	Sets the modulation speed: either the frequency in Hertz or a note value when SYNC is activated.
FB	The voice's feedback from its output to input.

LP	The voice's lowpass filter frequency.
PAN	The voice's stereo panorama.
OUT	This controls the output level of the voice.

The module's toolbar contains a small window which shows the LFO's waveforms. By clicking this window a larger window appears which can also be scaled:.



4.1.14 Phaser



The phaser comes with five styles whose sound characters vary greatly. The styles can be changed by clicking on the plugin name. Additionally, the different phasers are listed in the plugin list.

The modulation type is set via the waveform selection. For example, phase shifts of 0°, 90° or 180° between left and right channel can be adjusted. This allows stereo effects to be created.

The **INV** button inverts the wet signal's phase. This results in a different sound, often with a reduced bass.

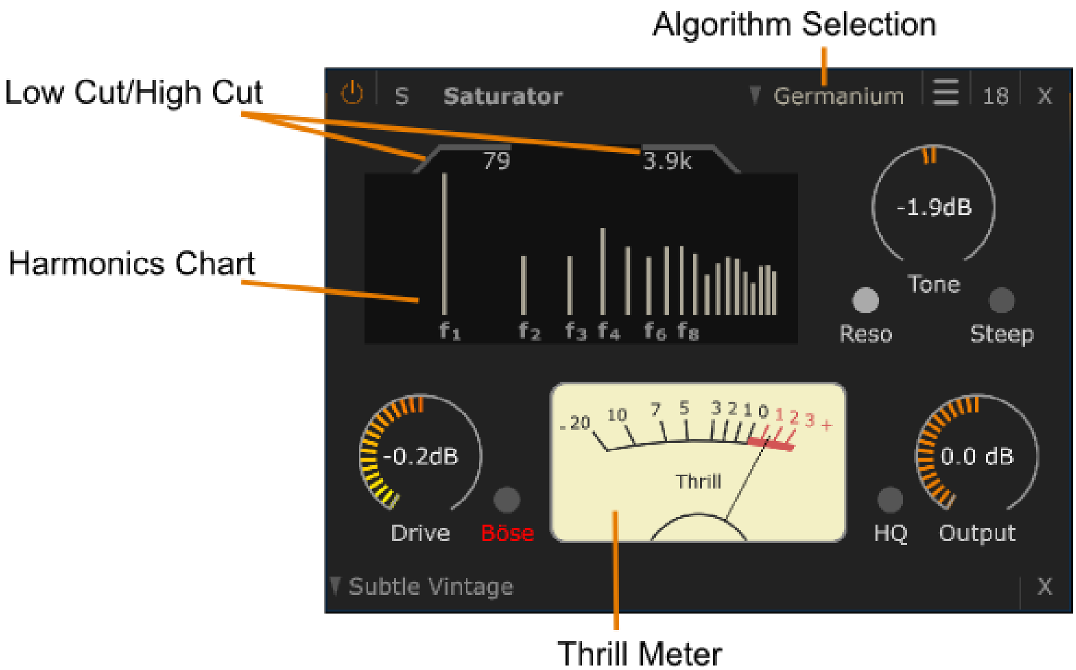
Output controls the output gain of the wet signal or output signal. The settings menu contains an entry ("Process Output Gain after Dry/Wet") where this can be set.

Rate sets the modulation frequency. If **SYNC** is active, the tempo is synced to the DAW's tempo. In this case, instead of a frequency a note length is set, for example 1/4D for a dotted quarter note.

4.1.15 Distortion

The distortion module can be used for simple distortion applications. **Drive** controls the amount of distortion while **Output** allows to adjust for level changes. In the Settings menu you can choose whether output gain should be processed before or after Dry/Wet.

4.1.16 Saturator



The Saturator emulates the "warm" sound of analog audio devices that are driven in the saturation range. In principle, the Saturator generates non-linear distortions, adding harmonics to the audio signal. These – sometimes only subtle – changes in the overtone spectrum can underline the character of individual instruments and make mixes appear lively. With high drive settings, however, the Saturator can also produce "heavy" distortion.

Five different saturation **Algorithms** are available, which can be set using the algorithm selection:

- **1950s**: This algorithm is based on a preamp that was used in tape machines in the 1950s. It produces a very soft tube sound.
- **Germanium**: This algorithm emulates the sound of a preamp built with

germanium transistors. With medium and high drive settings, the spectrum is very rich in overtones.

- **Thermionic**: The algorithm is based on a 19" full-tube distortion unit operating in pentode mode. Both even and odd harmonics are produced, making the algorithm sound aggressive yet warm.
- **Triode**: This algorithm models the behavior of a simple triode circuit. This adds just one harmonic to the signal, highlighting the upper octave range of the signal.
- **Tape**: The tape algorithm is based on the sound of a 1/4" tape machine from an American manufacturer from the 1960s. At low drive values, the sound is rather neutral. At higher drive values, the algorithm smoothly changes to saturation.

The **Harmonics Chart** shows the overtone spectrum of a sine signal. This way you can get an idea how the signal is colored or distorted by the algorithm. The harmonics are marked with fx, whereby the fundamental frequency is marked with f1. The first harmonic is f2 - which equals twice the fundamental frequency.

Above the harmonics chart you can find the **Low Cut/High Cut** controls. They can be used to limit the signal in the bass and treble ranges. The **Tone** slider can also be used to adjust the balance between bass and treble - from "dark" to " bright". If the **Reso** button is activated, the cutoff frequency range of the low-cut filter is boosted by a few dB. By switching on the **Steep** button, the slope of the highcut filter is increased from 6 dB/octave to 12 dB/octave.

The **Drive** knob controls the degree of saturation/distortion. Activating the **Böse** button raises Drive by an additional 20 dB. **Attention: This can make the output signal very loud!**

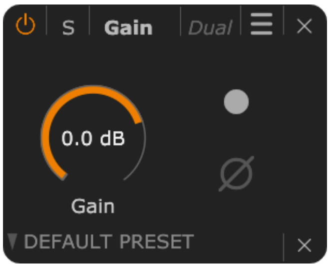
The signal level can be adjusted using the **Output** control. In the Settings menu you can set whether the amplification should take place before or after the Dry/Wet controller.

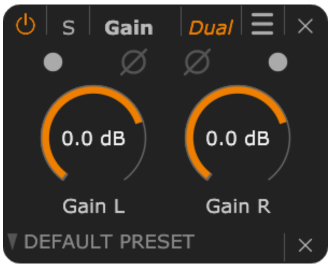
The **Thrill** meter shows an average value of the distorted signal. If the **HQ** button is pressed, the saturator works with internal oversampling (in addition to the globally set oversampling) and thereby reduces artifacts caused by non-linear processing.

4.1.17 Delay

This is a simple delay plugin. Its delay time can be set both in milliseconds and synchronously with the DAW tempo. In the latter case, the sync mode must be switched on via the **Sync** button. Sync mode is disabled by default. It can only be activated if a MixBox license exists.

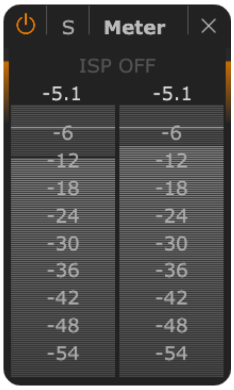
4.1.18 Gain





The Gain plugin allows level adjustments as well as inverting the phase. When used on stereo tracks, it also provides a dual mode where gain and phase can be set separately for left and right.

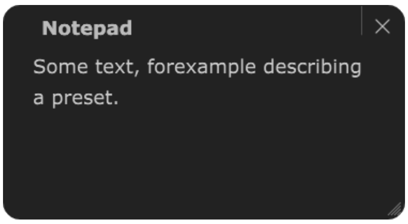
4.1.19 Meter



This is a simple peak/hold meter. By clicking onto one of the hold values, these will be reset.

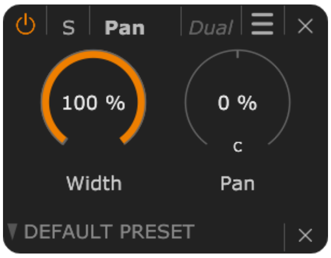
If the **ISP** switch is activated, it also detects intersample peaks.

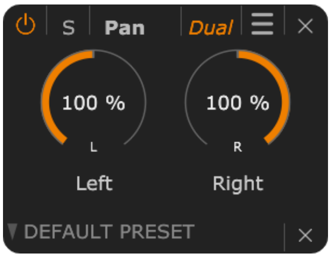
4.1.20 Notepad



The Notepad plugin allows to add text notes. This can for example be useful to describe a preset or a complex routing.

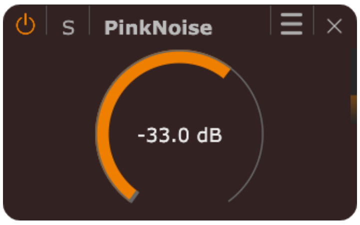
4.1.21 Pan





The panning module can either be used as normal or as dual pan. The normal pan also provides a **Width** control while in dual mode the panning for left and right channel can be set independently. The dual pan is only available with a MixBox license.

4.1.22 PinkNoise



This plugin generates a pink noise that is added to the signal. It can be used, for example, to emulate analog sound behavior or it simply serves as a signal generator.

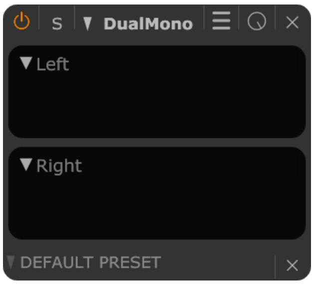
4.1.23 PitchMicro/PitchMacro

The PitchMicro and PitchMacro plugins enable easy pitch shifting. The setting ranges are between -100 and +100 cents (PitchMicro) and -12 and +12 semitones (PitchMacro). PitchMacro is only available with a MixBox license.

4.2 Routing Plugins

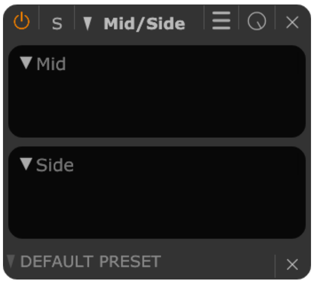
Routing plugins can contain HOFA SYSTEM effect plugins and allow numerous different signal routings, such as mid/side splitting, parallel processing and much more.

4.2.1 DualMono



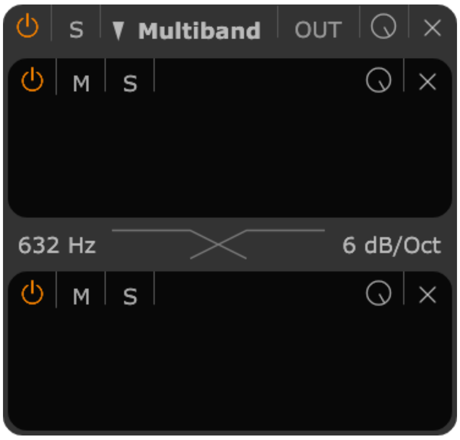
The DualMono plugin allows processing left and right channel independently. The upper signal path (**Left**) represents the left channel. The lower path (**Right**) represents the right channel. To modify the signals, effect plugins can be dragged into the two paths.

4.2.2 Mid/Side



The Mid/Side plugin allows the signal to be divided into middle and side signals. The center signal is transmitted in the upper signal path (**Mid**) and the side signal in the lower path (**Side**). To modify the signals, effect plugins can be dragged into the two paths.

4.2.3 Multiband



The Multiband plugin allows the signal to be divided into several frequency bands. These can be processed separately before they are summed to a single (full band) signal at the end.

With the help of the multiband plugin, e.g. a multiband compressor can easily be implemented by inserting a CompChan plug-in into the upper and lower signal paths, see also Section 3.3.2 .

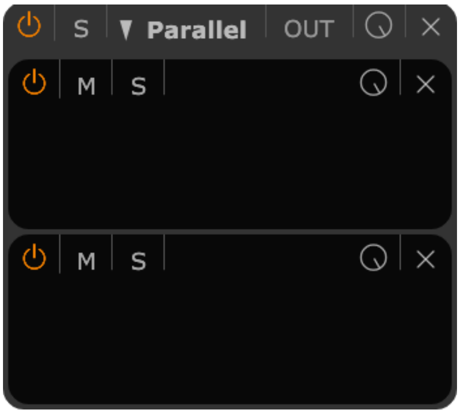
Between the two signal paths you can find the parameters for the crossover. On the left side its cut-off frequency is adjustable (632 Hz in the illustration) and on the right side its slope. The slope is the same for all frequency bands and can be set to 6, 12 or 24 dB/octave.

You can easily add additional paths to the two existing signal paths by simply inserting another plugin above or below the existing paths or plugins. The insertion point is marked by a horizontal, orange line. Inserting increases the number of bands, with the top paths processing the upper frequency bands and the bottom paths corresponding to the lower frequency bands. Furthermore, the crossover frequencies also change during insertion. They may then have to be adjusted.

If you move the mouse over a signal path, its header appears. Here you can switch the corresponding signal path on, off, mute or solo. In this way, for example, only the bass section can be listened to or the midrange band can be muted. Furthermore, the proportion of the total signal can be adjusted with the controller on the right-hand side. With the X-button you can remove the signal path from the plugin. For reasons of simplicity, the control elements of the signal paths are only visible as long as the mouse is over them or a switch is activated.

A Multiband plugin can be switched to a Parallel plugin (see section 4.2.4) in the header.

4.2.4 Parallel



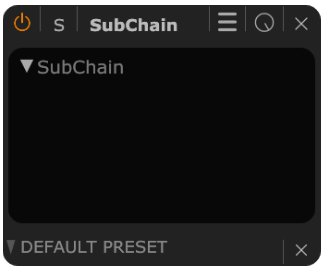
Parallel signal processing is possible with the Parallel plugin. This means that all signal paths in the Parallel plugin receive the same input signal. The individual paths can be edited separately by dragging in the desired plugins. At the end, the signals of all paths are summed.

You can easily add additional paths to the two existing signal paths by simply inserting another plugin above or below the existing paths or plugins. The insertion point is then marked by a horizontal, orange line.

If you move the mouse over a signal path, its header appears. Here you can switch the corresponding signal path on, off, mute or solo. Furthermore, the proportion of the total signal can be adjusted with the controller on the right-hand side. With the X-button you can remove the signal path from the Parallel plugin. For reasons of simplicity, the control elements of the signal paths are only visible as long as the mouse is over them or a switch is activated.

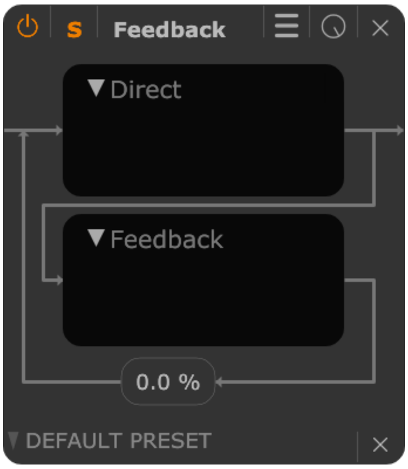
A Parallel plugin can be switched into a Multiband plugin (see section 4.2.3) in the header.

4.2.5 SubChain



Several plugins can be combined in a SubChain. This makes it possible, for example, to provide an effect chain with a shared Dry/Wet control.

4.2.6 Feedback



The Feedback plugin can be used to create feedback loops. It consists of the signal paths DIRECT and FEEDBACK. The input signal is normally processed in the DIRECT path and sent to the output.

The FEEDBACK path is fed back from the output of the DIRECT path to its input. In the FEEDBACK path there is a control that determines the returned part of the output signal.

To avoid instabilities, a soft clipper is inserted in the internal signal processing behind the feedback controller. This allows oscillation, but prevents an uncontrolled increase in the output signal.

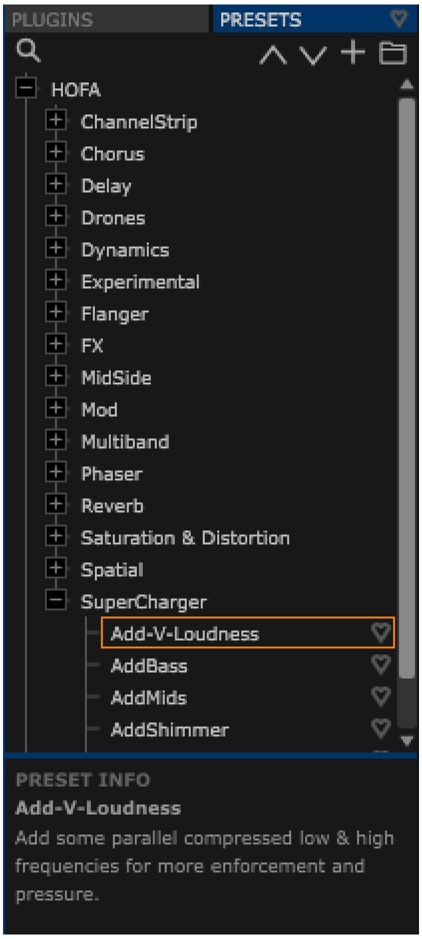
5 Presets Manager

Presets are managed in HOFA SYSTEM using the Presets Manager. This exists on the one hand as a global preset manager in the form of the PRESETS button on the left side of the user interface, to the right of the plugins list. This is where HOFA presets (cannot be overwritten) as well as user presets for the complete HOFA SYSTEM are organized. On the other hand, each SYSTEM plugin (except Gate, Notepad, #Multiband, #Parallel) has its own preset manager for managing user presets. This separation allows you to handle either global HOFA SYSTEM presets or presets for a single plug-in.

To open the Presets Manager of a plugin, the entry **Show Presets** must be checked in the Settings menu of the plugin. A small triangle appears in the footer of the plugin followed by the current preset name, e.g.:

Clicking on this opens the plugin Presets Manager. The close button **X** can be used to hide the footer.

The functions of the global Presets Manager are explained below. The function of the Presets Manager of a plugin and the global Presets Manager is more or less identical.



All presets are displayed in a list. The user presets folder is only displayed if you have already created your own presets.

The toolbar provides the following functions:

Search presets by name and/or info text to quickly find matching presets.

The up/down buttons allow quick **navigation** within the presets. This is also possible in most DAWs using the up/down arrow keys on the keyboard.

Allows you to add or **save** a preset. It is possible to add an info text to the preset. This is displayed in the PRESET INFO window below the list.

For a better overview, the user presets can be sorted in folders. Click on this icon to create a **new folder**.

Use the heart symbols in the list to highlight your personal favorites. Clicking on the heart at the top of the list will display only your favorite presets.

Loading a preset is very easy: you only have to click on a preset in the list. The current processing chain is then deleted and the preset is loaded. Done. It is also possible to drag the preset from the list into the processing chain. This procedure makes it possible to keep the existing processing chain.

It is also possible to **replace** plugins in the processing chain with a preset. For this purpose, the chosen plugins are selected by clicking on a free position in the processing chain and dragging the cursor over the corresponding plugins with the mouse button pressed. Then click on the desired preset in the preset manager. Another way to replace plugins is to drag the preset directly onto a plugin.

When the mouse is positioned over a preset, the **PRESET INFO** created when the preset was saved appears in the lower window. In addition, two icons appear next to the preset name:

Use the pen icon to change the name and info of the preset. If you click on the bin, the preset will be deleted. Folders can also be renamed or deleted, but only empty folders can be deleted.

For a better orientation, user presets can be dragged with the mouse into the various self-created folders and thus be archived.

5.1 Export and Import of User Presets

The export and import of user presets is extremely efficient within HOFA SYSTEM.

For quick export, user presets and entire folders can be dragged from the global Presets Manager to either the desktop or any Explorer folder (PC) or Finder window (macOS). A single preset is saved as a file with the extension "systempreset" while a Zip folder is created from a Presets folder.

The created preset files or folders can now be archived or shared with other users. This also allows you to quickly transfer the presets to another computer.

Importing presets is just as easy. A preset file or a Zip folder with SYSTEM presets can be simply dragged with the mouse into the list of the global Presets Manager and the presets are already available in HOFA SYSTEM.

Ex- or import of user presets is not possible with the Preset Manager of a single plugin.

It is also possible to drag selected plugins from the processing chain onto the desktop or into any Explorer/Finder folder to create a preset. Conversely, a preset file can also be dragged directly into the processing chain. In this way, presets can be exchanged quickly and easily.

6 Single Plugins

All effects included in HOFA SYSTEM are available as single plugins. This makes it possible to use the plugins outside HOFA SYSTEM directly in the DAW. This is useful if you do not need complex effect structures and only want to use a reverb or a compressor, for example. The various routing possibilities (parallel, multiband, feedback etc.) of HOFA SYSTEM are not available when using the single plugins.

The single plugins can be found under these names:

- SYS AlgoVerb
- SYS Comp (contains CompChan und CompSum)
- SYS CompListenMic
- SYS Delay
- SYS Distortion
- SYS EQ (EQParametric)
- SYS EQ-Dynamic
- SYS Expander
- SYS Flanger (contains FlangerClassic und FlangerTape)
- SYS Gain
- SYS Gate
- SYS IR-Verb
- SYS MasterLimiter
- SYS Meter
- SYS MultiMod
- SYS Notepad
- SYS Pan
- SYS Phaser
- SYS PinkNoise
- SYS Pitch (contains PitchMicro und PitchMacro)
- SYS Saturator
- SYS TransientShaper

Depending on your DAW’s settings, they may be not listed alphabetically, but sorted into categories.

The scaling and thus the size of the single plugins is independent of the values for "Plugin Scale" and "Global Scale" in the SYSTEM settings menu. You can save the current size as the "Default Scale" for the respective plugin or for all single plugins at the same time in the plugin menu.

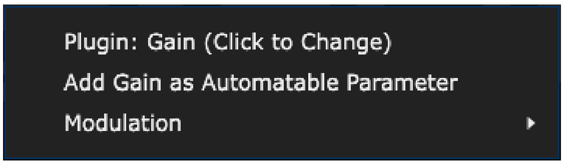
7 Automation and Modulation of Parameters

Almost all parameters in HOFA SYSTEM can be automated in the DAW as well as internally modulated. It is easy to report the parameters you want to automate to the DAW. Furthermore, it offers a wide range of parameter modulation options.

To automate or modulate a parameter, you must first open the parameter menu, which is explained in the following section.

7.1 Parameter Menu

Right-clicking on an operational element (e.g. a controller or a button) displays its parameter menu:



In this example, the gain control of the gain plugin was clicked. The first menu item shows the name of the plugin. This name will be displayed later in the DAW. It can be changed here to distinguish it from the parameters of similar plugins.

The menu items "Add... as Automatable Parameter" and "Modulation" are explained in the following two sections on and .

When using folders, the parameter menu has two additional entries, which are described in more detail in the chapter 8 .

7.2 Automation

As already explained in the section 3.3 HOFA SYSTEM plugins can be added and deleted within the processing chain. For this reason, a fixed assignment of HOFA SYSTEM parameters in the DAW is not possible. Instead, the parameters to be automated must be communicated to the DAW. This is done by opening the parameter menu (by right-clicking on the control element) and then selecting the second menu item "Add... as Automatable Parameter". Now the plugin name followed by the parameter name should appear in the DAW. In the same way, an already assigned parameter can be logged off from the DAW. The second menu item is then "Remove .... as Automatable Parameter".

Unfortunately, some DAWs cannot change the parameter names of a loaded plugin. In this case, the assignment of a parameter is not directly visible in the DAW. If this happens, you can record an automation of the relevant controller in order to find the assigned parameter.

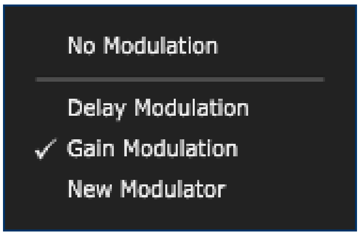
Note that when disconnecting an automated parameter, the automation track remains in the DAW and will apply to newly assigned parameters in the future. To disconnect, either use "Remove .... as Automatable Parameter" in the Parameter Menu or remove the plugin from the processing chain, see section 3.3 .

When using single plugins, all parameters can be directly automated. Assigning parameters is not necessary and consequently the parameter menu has only an entry for modulation, which is explained in the following section.

7.3 Modulation

HOFA SYSTEM provides a powerful parameter modulation. Almost all parameters within HOFA SYSTEM can be modulated.

If you want to modulate a parameter, you must first open its parameter menu by right-clicking on the control element. The third menu item is "Modulation". If you hover over it (without clicking), the following submenu opens:



Here you can either add a new modulator (last menu item) or select an existing modulator (in the example here "Delay Modulation" and "Gain Modulation"). In addition, the parameter modulation can be removed via "No Modulation" (first menu item). A parameter can only have one modulator.

If you go to the "New Modulator" menu item or select an existing modulator, the window for parameter modulation opens:

This overview lists all modulators. The header of a modulator has a black background. On the left is the name of the modulator. It can be renamed. Furthermore, the modulation type (sine, square, sawtooth, triangle and random) can be selected in the header under **Type** and the modulation frequency can be set under **Freq**. When **Sync** is activated, modulation is synchronized to the tempo of the DAW.

All parameters linked to a modulator are listed below the header line. On the left side you can find the plugin- and parameter name. The values in the middle indicate the modulation range. Between them a grey bar shows the current modulation.

The value to the right of the upper modulation limit is the phase shift with which the parameter is modulated. A phase shift can be useful for the generation of stereo effects, among other things. For example, to create a flanger, the delay modulation of the right channel could be phase-shifted to the left channel.

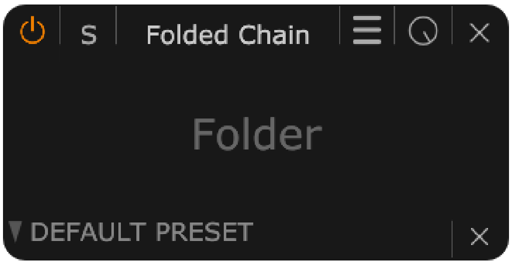
The modulation depth can be set at the second to last position under **Depth** and on the right side the modulation for the corresponding parameter can be switched on and off with the **On** button.

The parameter modulation table can be opened directly from the parameter menu by clicking on the third menu item "Modulation".

8 Folding Plugins

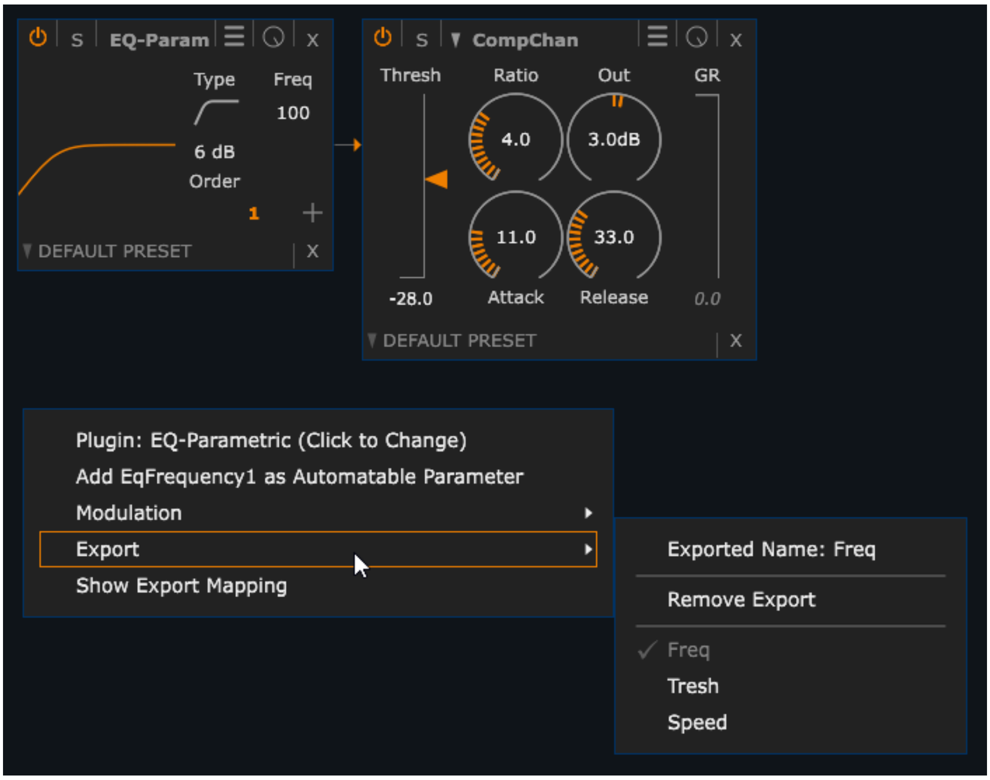
With complex plugin setups, it may happen that you lose the general survey. To deal with this, there is the so-called folding function. With its help it is possible to combine several HOFA SYSTEM plugins into a compact folder. In addition, it is possible to set and control the parameters of the folder plug-ins "from the outside".

To create a folder or a folded chain, the relevant plug-ins must be selected by clicking on a free position in the processing chain and dragging the cursor over the desired plug-ins with the mouse button pressed. The selected plugins have an orange frame. Then click on the **FOLD** button in the toolbar. The resulting folder is initially called "Folded Chain":



The name can be changed simply by clicking on it. The folder does not contain any control elements after its creation. By double-clicking on the folder, with the mouse cursor displayed as a hand, you can "look inside". All included plugins will be displayed. In addition, the toolbar displays "Main | Folder Name" to show that you are inside a folder. By double-clicking on the main window again or outside a plug-in, you can "exit" from the folder and switch to the "normal" processing chain.

To have access to a control or display element within the folder from the outside, the respective element must be exported. To do this, open the folder by double-clicking on it.



Then right-click on the control or display element you want to export. As a result, the parameter menu opens which now has two further entries "Export" and "Show Export Mapping". Select "New Export" from the "Export" submenu. After you have assigned a name for the parameter in the following dialog window, the element appears on the user interface of the folder and can be controlled from there. When you right-click on display elements, only the "Export ..." and "Show Export Mapping" menu items are visible. Names cannot be changed.

In the same way, an element can be removed from the folder interface. To do this, select "Remove Export" from the export submenu, see the figure above. For display elements, the check mark must be removed accordingly.

In the Export submenu all exported parameters are additionally listed (in the above example these are Freq, Thresh and Speed). Furthermore, the **EXPORT MAPPING** with the exported parameters opens under the menu item "Show Export Mapping".

The following example shows the creation of a channelstrip folder consisting of the two plugins EQ-Parametric and CompChan. The EQ is a high pass, and its frequency is exported. The control elements Threshold, Attack and Release of the CompChan should also be controllable from the folder interface, whereby there should be a common control "Speed" for Attack and Release. The initial situation is shown in the figure above. The export procedure is as follows:

1. export frequency from EQ → "New Export" and name it as "Freq".
2. export Threshold from CompChan → "New Export" and name it as "Thresh".
3. export Attack from CompChan → "New Export" and name it as "Attack".
4. export Release of CompChan → Select "Attack" and name it as "Release".
5. export Gain Reduction GR from CompChan → "Export Meter".
6. rename the control for attack and release to "Speed", see below.

The result is shown in the following illustration:

In addition to the "Channelstrip" folder (above), the figure shows an overview of the export mapping (below). The operating and display elements are highlighted. The corresponding parameters are listed below.

The column **Exported Parameters** lists the names of the elements and parameters. The names can be changed by clicking on them. The names of the parameters are preceded by an X. If you click on this, the parameter is disconnected from the control and deleted from the list. The column **Value** shows the values of the elements and parameters. Under **Prec** the number of decimal places for the respective control element can be set. To individualize the elements and the surface of a folder, their color can be changed via the color selection.

The control ranges of the parameters and elements can be defined via the values in the **Min** and **Max** columns. If the **Log** button on the right side is activated, the value change of the corresponding control element is logarithmic (i.e. fine resolution for small values, large resolution for large values).

The EXPORT MAPPING overview can also be accessed by right-clicking on the elements of a folder, so that you do not have to open the folder if you want to change the settings.

The control and display elements of a folder can be arranged in any order. To do this, activate the "Item Positioning Mode" item in the header menu of a folder, after which the elements can be moved.

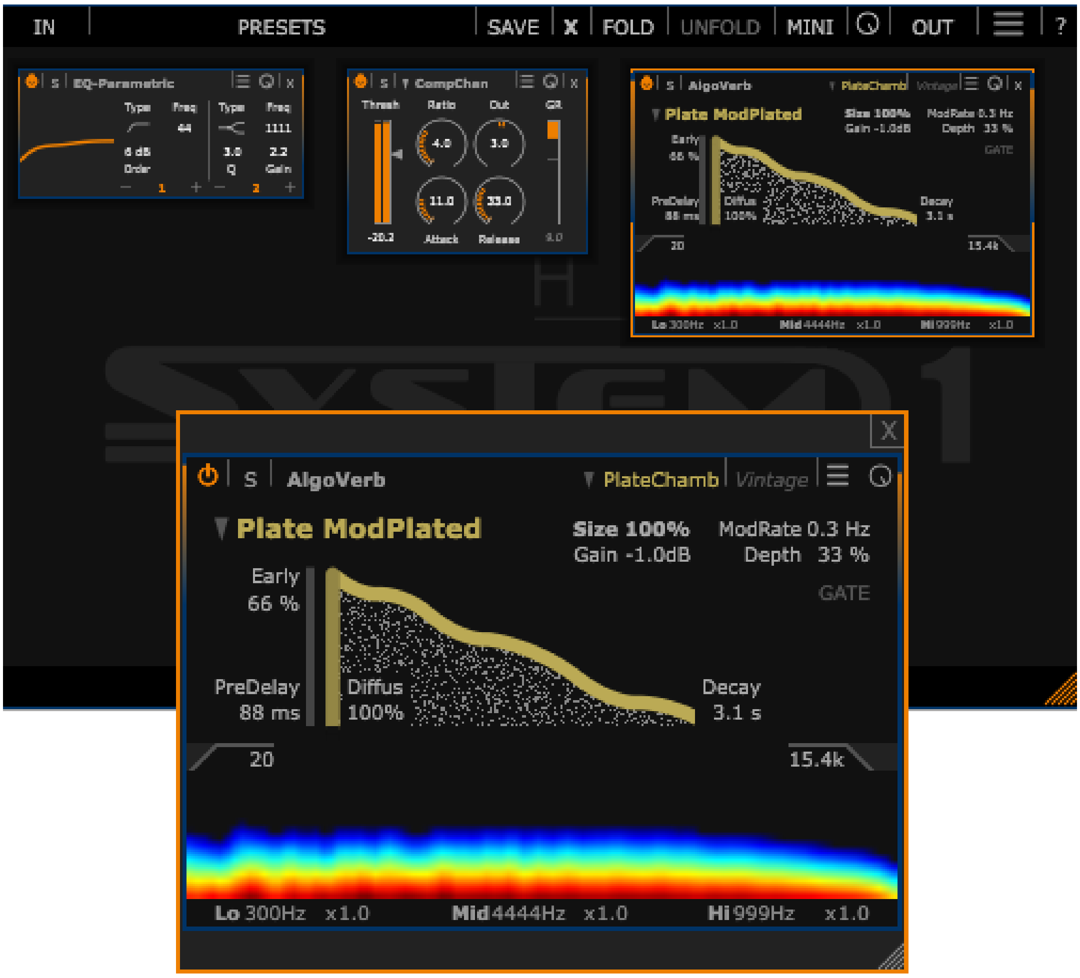
A folder can be dissolved by selecting it and clicking on **UNFOLD** in the toolbar.

9 Minimize Mode

We already mentioned In the previous chapter that you can lose the general survey when using large effect routings, because not all plugins are directly visible. This is where the Minimize mode can help. It allows all plugins to be made visible "in one go".

Minimize mode can be activated or deactivated in the toolbar by clicking on **MINI**. After its activation, all plugins are displayed in a reduced size. Depending on the screen and personal preference, the degree of reduction can be set in the global settings menu under **Minimize Min Scale.**

It is not possible to use the plugins directly in Minimize mode. Instead, an extra window is available in which the user interface of a selected plugin is displayed in normal size:



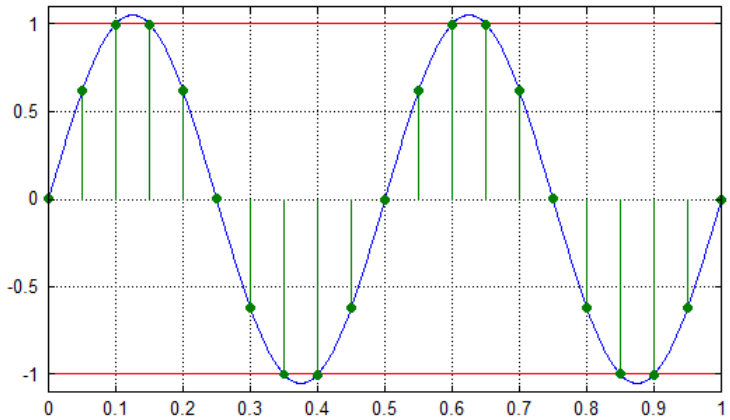
10 Settings Menu

Default Settings	Saves or loads the current plugin's state (including all modules and the plugin's size) as default. All new instances will be loaded with these settings.
Global Scale	The global scale determines the total size of HOFA SYSTEM inside the DAW and also the font size.
Scale for MINI View	This determines the size that is used when MINI mode is active.
Live Mode	In live mode, there is no internal latency compensation - latency-free plug-ins are "immediately" audible. Furthermore, oversampling is not possible in live mode.
Clear Peaks when Starting Playback	If active, all peak hold displays will be reset if playback is started.
Clear Peaks when Controls are changed	If active, all peak hold displays will be reset if a control is changed.
Show Warning when Replacing Plugins with Chain Presets	If active, a warning is shown when modules are replaced by a chain preset.
Show Warning when Latency Changes	If active, a warning is shown when the latency of HOFA SYSTEM is changed.
Show Routing Arrows	Displays the routing arrows between the plugins in the processing chain.
Online Oversampling	Sets the oversampling during playback. Available values are 1x (no oversampling), 2x, 4x, 8x, 16x and 32x.
Offline Oversampling	Sets the oversampling during offline rendering. Available values are 1x (no oversampling), 2x, 4x, 8x, 16x and 32x.

11 Interesting facts

11.1 Intersample peaks

In the digital domain, signals are represented by samples taken at discrete time intervals. This can lead to a signal that exceeds the digitally available magnitude range when converted to an analog signal while the digital signal still lies inside the range. The following figure demonstrates this:



While the (green) digital signal lies in between ±1, the (blue) analog signal exceeds this range. This wouldn't be a problem if the analog signal had enough headroom. Unfortunately, most DA converters run on low supply voltages and therefore lack the needed headroom. Therefore, a signal might clip even if a sample peak meter doesn't show this.
The True Peak mode

11.2 EBU R 128

EBU R 128 is an EBU (European Broadcasting Union) standard defining guide values for loudness in broadcasting. The origin of this standard lies in the increasing loudness in broadcasting and the dynamic variations resulting from that (e.g. ads are often louder than regular program). To counteract this, loudness normalization was introduced, aside with a referenced target level.

With EBU R 128 an open standard for loudness measurement was created. Also new units LU and LUFS (Loudness Unit / Full Scale) were introduced. One LU is basically the same as one dB but describes a loudness measured according to EBU R 128. While LU is a relative level, LUFS is an absolute level, the relationship being 0 LU = -23 LUFS.

To evaluate a loudness, three different ballistics are introduced. These are the momentary loudness, the short-term loudness and the integral loudness. The first and second ballistic show the actual loudness with the second reacting slower. The last one measures the loudness over a longer time period, resulting in a loudness for that whole time. This measurement is gated, so silent parts don't influence the measurement. Also, this loudness is used for a loudness normalization according to EBU R 128.

For more details have a look at:
https://en.wikipedia.org/wiki/EBU_R_128

The official EBU recommendation can be found here:
https://tech.ebu.ch/docs/r/r128.pdf

12 Contact

HOFA GmbH

Lusshardtstraße 1-3

D-76689 Karlsdorf

Germany

E-Mail: plugins@hofa.de

Phone: 0049 7251 3472 444

www.hofa-plugins.de