

# MPhatik



## Overview

MPhatik is a unique kind of dynamics multi-effect, which lets you process the audio material and then apply the original (or completely different) level envelope to that processed audio. As an example, it measures the level of the input (or side-chain), processes the input signal using the integrated effects, then measures the output level and applies a dynamic gain to make the output level same as the input.

As a typical example - distortion/saturation is generally used to brighten the audio and generate higher harmonics. As a side-effect, the level envelope changes and the audio loses dynamics. MPhatik lets you restore the original dynamics, fully partly, or even creatively.

Note that you couldn't do what the plugin does with traditional tools such as compressors or expanders, because these only have a single level follower, so they can only react to their input, they don't measure difference in level in the way that MPhatik does.

## Case 1 - Phattening an input signal

The first scenario is the most common and is especially useful on drums. Simply put the plugin on, say, the snare drum track. The defaults will be just fine for starters. MPhatik in **Matching** mode calculates the input level, then processes the audio using amp, convolution and compressor (in that order). After that it calculates the output level and applies a gain so that the output level envelope becomes similar to the input's. In other words, it restores the original dynamics.

Besides the parameters of the effects themselves, there are several global parameters, such as **Attack** and **Sustain** that let you creatively shape the sound.

## Case 2 - Restoring original dynamics after processing using your own 3rd party effects

It is simple to do the same thing as Case 1 using other plugins, instead of the integrated amp, convolution and compressor. Simply place MPhatik after your 3rd party effects and send the original (pre-effects) audio to MPhatik's side-chain (the precise method depends on your host). Then enable the Side-chain switch in MPhatik and possibly disable the integrated effects. This way the plugin will act solely as a dynamics processor, restoring the original level envelope.

The insert buss of your track will look like this:

Audio -> 3rd party effects -----> MPhatik \----- Send to side-chain -----/

## Case 3 - Applying an envelope of a different signal to your track

The approach from Case 2 can be taken even further - by sending any signal to MPhatik's side-chain you can make your track have the

same dynamics. This has numerous creative possibilities.

For example, you can apply a drum track's dynamics onto a static pad to make the pad rhythmical. Then just use the **Dynamics** parameter to control how much processing takes place.

Another example - when recording drums (or other instruments) using multiple microphones, it is common that one microphone records the true dynamics better than another one. One may produce very exaggerated sustain for instance. In that case you may simply apply the good dynamics to the track with not-so-good dynamics.

MPhatik can also perform **Ducking** (just select the mode in the Globals panel). This may be useful in any of the 3 mentioned cases but as specific examples - you might want to resolve collisions between multiple drums or make sure a bass drum cuts sharply through the mix, which is very handy when mixing dance music. All you need to do is to apply MPhatik to the signal that you want to duck and send the triggering track (say the bass drum) into its side-chain.

## Presets

### Presets

Presets button shows a window with all available presets. A preset can be loaded from the preset window by double-clicking on it, selecting via the buttons or by using your keyboard. You can also manage the directory structure, store new presets, replace existing ones etc.

Presets are global, so a preset saved from one project, can easily be used in another. The arrow buttons next to the preset button can be used to switch between presets easily.

Holding **Ctrl** while pressing the button loads a random preset. There must be some presets for this feature to work of course.

Presets can be backed up by 3 different methods:

A) Using "Backup" and "Restore" buttons in each preset window, which produces a single archive of all presets on the computer.

B) Using "Export/Import" buttons, which export a single folder of presets for one plugin.

C) By saving the actual preset files, which are found in the following directories (not recommended):

Windows: C:\Users\{username}\AppData\Roaming\MeldaProduction

Mac OS X: /Library/Application support/MeldaProduction

Files are named based on the name of the plugin like this: "{pluginname}.presets", so for example MAutopan.presets or MDynamics.presets. If the directory cannot be found on your computer for some reason, you can just search for the particular file.

Please note that prior to version 16 a different format was used and the naming was "{pluginname}.presets.xml". *The plugin also supports an online preset exchange. If the computer is connected to the internet, the plugin connects to our server once a week, submits your presets and downloads new ones if available. This feature is manually maintained in order to remove generally unusable presets, so it may take some time before any submitted presets become available. This feature relies on each user so we strongly advise that any submitted presets be named and organised in the same way as the factory presets, otherwise they will be removed.*



### Left arrow

Left arrow button loads the previous preset.



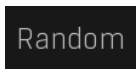
### Right arrow

Right arrow button loads the next preset.



### Randomize

Randomize button loads a random preset.



### Randomize

Randomize button (with the text 'Random') generates random settings. Generally, randomization in plug-ins works by selecting random values for all parameters, but rarely achieves satisfactory results, as the more parameters that change the more likely one will cause an unwanted effect. Our plugins employ a smart randomization engine that learns which settings are suitable for randomization (using the existing presets) and so is much more likely to create successful changes.

In addition, there are some mouse modifiers that assist this process. The smart randomization engine is used by default if no modifier keys are held.

Holding **Ctrl** while clicking the button constrains the randomization engine so that parameters are only modified slightly rather than completely randomized. This is suitable to create small variations of existing interesting settings.

Holding **Alt** while clicking the button will force the engine to use full randomization, which sets random values for all reasonable automatable parameters. This can often result in "extreme" settings. Please note that some parameters cannot be randomized this way.



### Panic

Panic button resets the plugin state. You can use it to force the plugin to report latency to the host again and to avoid any audio problems. For example, some plugins, having a look-ahead feature, report the size of the look-ahead delay as latency, but it is inconvenient to do that every time the look-ahead changes as it usually causes the playback to stop. After you tweak the latency to the correct value, just click this button to sync the track in time with the others, minimizing phasing artifacts caused by the look-ahead delay mixing with undelayed audio signals in your host. It may also be necessary to restart playback in your host.

Another example is if some malfunctioning plugin generates extremely high values for the input of this plugin. A potential filter may start generating very high values as well and as a result the playback will stop. You can just click this button to reset the plugin and the playback

will start again.

## Settings

### Settings

Settings button shows a menu with additional settings of the plugin. Here is a brief description of the separate items.

**Licence manager** lets you activate/deactivate the plugins and manage subscriptions. While you can simply drag & drop a licence file onto the plugin, in some cases there may be a faster way. For instance, you can enter your user account name and password and the plugin will do all the activating for you.

There are 4 groups of settings, each section has its own detailed help information: **GUI & Style** enables you to pick the GUI style for the plug-in and the main colours used for the background, the title bars of the windows and panels, the text and graphs area and the highlighting (used for enabled buttons, sliders, knobs etc).

**Advanced settings** configures several processing options for the plug-in.

**Global system settings** contains some settings for all MeldaProduction plugins. Once you change any of them, restart your DAW if needed, and it will affect all MeldaProduction plugins.

**Dry/Wet affects** determines, for Multiband plug-ins, which multiband parameters are affected by the Global dry/wet control.

**Smart interpolation** adjusts the interpolation algorithm used when changing parameter values; the higher the setting the higher the audio quality and the lower the chance of zippering noise, but more CPU will be used.



## WWW

WWW button shows a menu with additional information about the plugin. You can check for updates, get easy access to support, MeldaProduction web page, video tutorials, Facebook/Twitter/YouTube channels and more.

## Sleeping

### Sleep indicator

Sleep indicator informs whether the plugin is currently active or in sleep mode. The plugin can automatically switch itself off to save CPU, when there is no input signal and the plugin knows it cannot produce any signal on its own and it generally makes sense. You can disable this in Settings / **Intelligent sleep on silence** both for individual instances and globally for all plugins on the system.

## Globals panel

Globals panel controls the main parameters of the plugin.

## Side-chain

### Side-chain

Side-chain button selects the side-chain input of the plugin for processing. By default it is disabled and the original input (before processing by amp, convolution and compressor) is used. Therefore the plugin makes the processed signal dynamics match the original. However you may use the side-chain input to make the processed signal match the dynamics of any other signal.

## Psycho-acoustic prefiltering

### Psycho-acoustic prefiltering

Psycho-acoustic prefiltering enables the loudness estimation pre-filtering processor. When disabled, the level detector reacts to the input level of the incoming signal. This is the traditional way, but it has nothing to do with human hearing, which reacts differently to different frequencies - our ears hear the different frequencies of equal loudness at different levels, being most sensitive to sounds between 2 and 5 kHz, (see the Fletcher-Munson curves, which are one of many sets of equal-loudness contours for the human ear) Psycho-acoustic pre-filtering pre-processes the level detection signal in a similar way to human hearing - it attenuates those frequencies we do not hear well and amplifies frequencies that we do. That way the level detector starts responding to what we actually hear, not to some sort of scientific signal as it usually does.

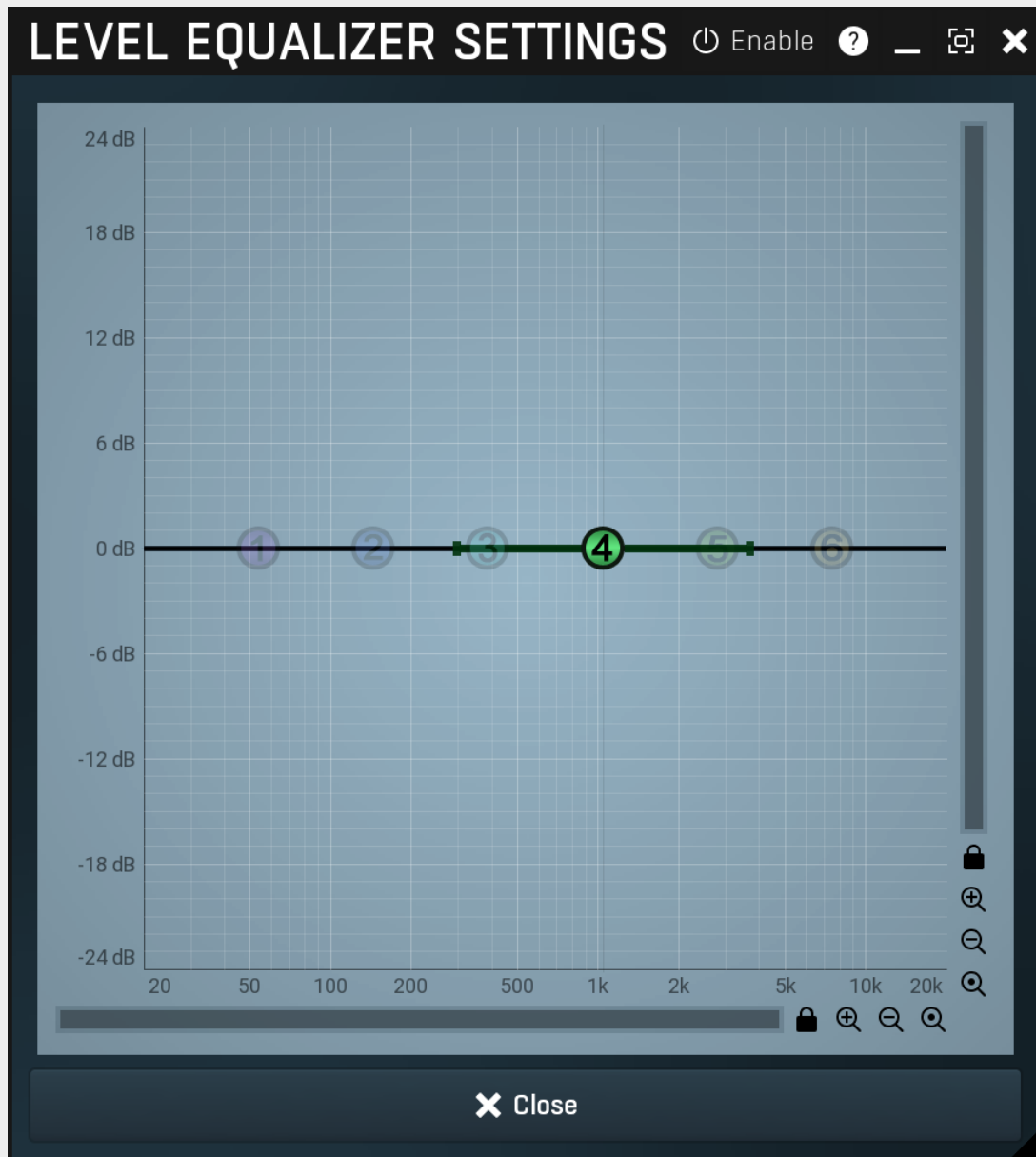
This feature is disabled by default simply because most users are not used to working with this feature, but it is perfectly safe to use it.

However, do not use it with limiters, where you want to remove the peaks, hence you are not focussed on human hearing, but rather are dealing with the technological problems in digital and analog audio.

## Eq

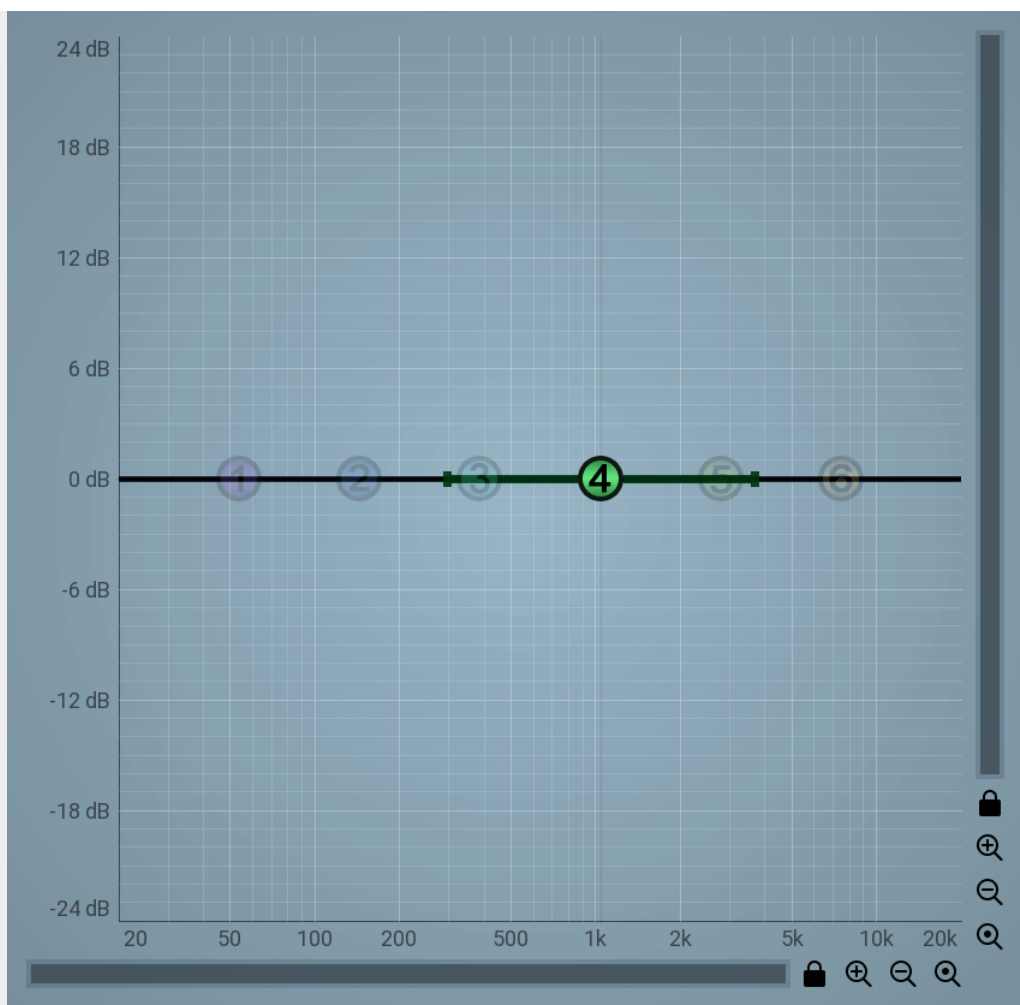
Eq button shows the settings of the side-chain equalizer. This equalizer does not affect the outgoing signal, but processes the signal entering the level detector. You can use it to target those frequencies to which you want the processor to react. In most cases you will be using low/high/band-pass filters to remove those parts of the spectrum that you are not interested in utilizing. For example, to make the detector react to a bass drum, you may use a low-pass filter with a frequency of say 100 Hz. Additionally, the equalizer lets you perform more complicated processing. For example, you may want the detector to react to the whole spectrum, but especially the high end of the spectrum, in which case a high-shelf filter may be the appropriate one to choose.

### Level equalizer settings



## Enable

Enable button enables or disables the level equalizer. It is disabled by default to lower CPU consumption.



**Equalizer shape**

## graph

Equalizer shape graph controls and displays the frequency response. There are several bands available, each of them can be enabled/disabled, can be set to a different filter, can have different frequency, Q and other parameters.

Double-click on a band point to enable or disable a band. Drag it to change its frequency and gain. Drag the horizontal nodes to change its Q. Hold **ctrl** key for fine tuning. Click using the right mouse button on it to open a window with additional settings.

## Band settings window



Band settings window contains settings for the particular band and can be displayed by right-clicking on a band or from a band list (if provided). On the left side you can see list of available filters, click on one to select it. On the right side, additional options and features are available.



## Presets

Presets button displays a window where you can load and manage available presets. Hold **Ctrl** when clicking to load a random preset instead.



## Left arrow

Left arrow button loads the previous preset.



### Right arrow

Right arrow button loads the next preset.



### Randomize

Randomize button loads a random preset.



### Copy

Copy button copies the settings onto the system clipboard.



### Paste

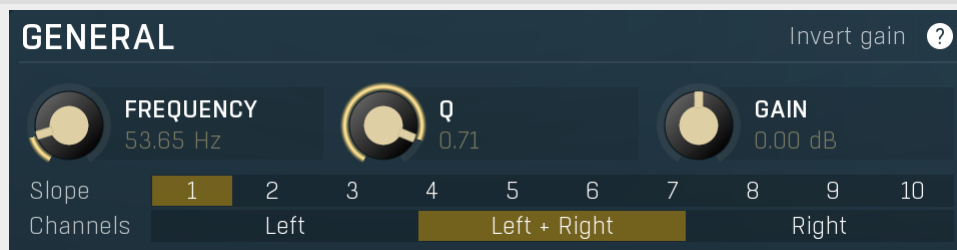
Paste button loads the settings from the system clipboard.



### Random

Random button generates random settings using the existing presets.

## General panel



General panel contains standard filter settings such as frequency or Q. Most of these values are available directly from the band graph, but it may be necessary to use these controls for more accurate or textual access.

Invert gain

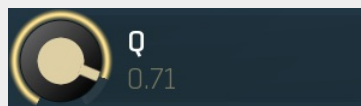
### Invert gain

Invert gain inverts the gain of the band, e.g. makes -6dB from +6dB.



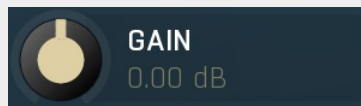
### Frequency

Frequency defines the band's central frequency, which has different meaning depending of filter type.



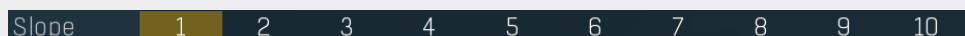
### Q

Q defines bandwidth. Please note that Q is an engineering term and the higher it is, the lower the bandwidth. Our implementation is trying to be more user-friendly, and by increasing the value (thus to the right), the bandwidth is increased as well. The editor still displays the Q value correctly.



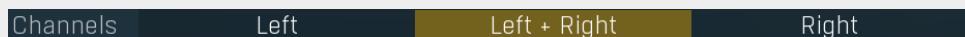
### Gain

Gain defines how the particular frequencies are amplified or attenuated. This parameter is used only by peak and shelf filters.



### Slope

Slope can potentially duplicate some of the filters creating steeper ones. By default, the slope is 1 and this usually means 2-pole 12 dB/octave filters. By specifying 2 you can make the plugin uses 4-pole 24 dB/octave filters instead etc. To see the actual slope of each filter look into the filter type list on the left.



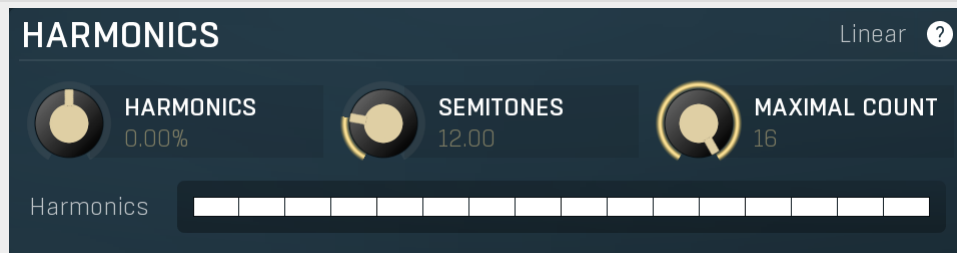
### Channels

Channels controls which channels the band processes. If the input is stereo (left and right channels, L+R, selected on the toolbar **Channel mode** button), then you can make a band process only the left, only the right, or both channels. Similarly when the plugin is set to M/S channel mode, you can choose between mid, side or both channels.

When one of more bands are set to process a single channel, then 2 EQ curves are displayed, in red for the Left or Mid and in green for the Right or Side. If these are not distinct, then we recommend using a style with a light background for these graphs.

You cannot process left with one band and side with the other, because these are working in different encoding modes. In this case you can easily use 2 instances of the plugin in series, one in L/R mode and the other in M/S.

## Harmonics panel



Harmonics panel contains parameters of the harmonics - clones of the main band created at higher frequencies derived from the frequency of the main band. This is often useful for removing natural noises, which usually bring some harmonics with them etc.

Linear

### Linear

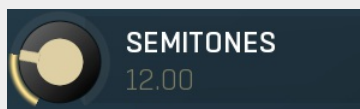
Linear button enables the linear harmonics spacing. When the main band frequency is say 100Hz and the **Semitones** value is 12, then in the default logarithmic mode the harmonics are 200Hz, 400Hz, 800Hz etc., increasing by 12 semitones (1 octave) each time. This is suitable because the filters themselves are logarithmic.

However harmonics generated by physical instruments are not spaced in this way. Rather, for a **Semitones** value of 12, they increase by a multiple of 12/12 of the main frequency each time. For example, for a base frequency of 100Hz, they will be at 200Hz, 300Hz, 400Hz, 500Hz etc. In linear mode the harmonics work in this way, but please note that then there is only a limited set of harmonics and Q is modified to approximate a reasonable behaviour, which is not always possible.



### Harmonics

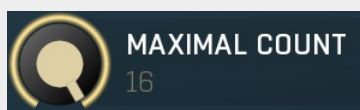
Harmonics defines the gain of the created harmonics. With maximum value (+/- 100%), all harmonics will have the same gain as the main band. A lower value makes the higher harmonics have lower gain. A negative depth will make alternate harmonics have positive and negative gains and is particularly useful for creative effects.



### Semitones

Semitones defines the frequency interval of the harmonics. For example, if the band is at 100Hz and the number of semitones is 12 (default), then the first harmonic will be at 200Hz (12 semitones higher), second at 400Hz etc., increasing by 12 semitones (1 octave) each time. Thus they are logarithmically-spaced harmonics. When linearly-spaced harmonics are enabled, this merely changes the ratio between them. In this mode, 100Hz is followed by 200Hz, 300Hz, 400Hz, 500Hz etc, that is, increasing by a multiple of 12/12 of the main frequency each time.

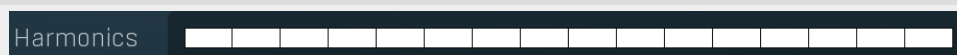
For a value of 7 (a perfect fifth), the logarithmic harmonics would be at 150Hz, 225Hz, 337.5Hz, 506.25Hz etc, increasing by 7 semitones (= 50%, as  $1.05946^7 = 1.498$ ) each time and the linear harmonics would be at 158Hz, 251Hz, 397Hz, 628Hz etc, increasing by 7/12 each time.



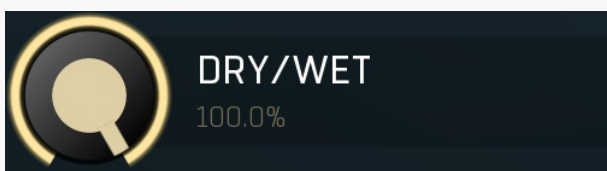
### Maximal count

Maximal count defines the maximum number of harmonics that could be created. The harmonics that are created depends on them being activated in the **Harmonics grid**.

## Harmonics grid



Harmonics grid is useful to turn on/off particular harmonics manually. Click any one to enable / disable it.

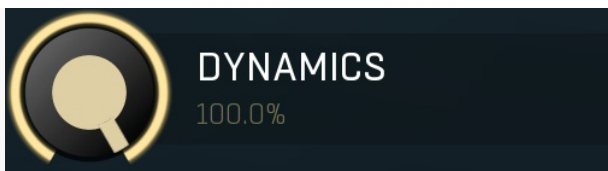


### Dry/Wet

Dry/Wet defines ratio between dry and wet signals. 100% means fully processed, 0% means no processing at all.

Range: 0.00% to 100.0%, default 100.0%





## Dynamics

Dynamics controls the amount of dynamics processing. 100% means the dynamics processor fully matches (depending on the **Mode**) the dynamics to the original, 0% means there will be no dynamics matching at all and the plugin will serve just as an amp, convolution and compression multi-effect. If you don't use the integrated effects and use the side-chain feature, then this parameter serves the same purpose as **Dry/Wet**.

Range: 0.00% to 100.0%, default 100.0%

Mode	Matching	Matching (no boost)	Ducking
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### Mode

Mode defines what kind of dynamics processing the plugin performs. Essentially the plugin measures the input level (either unprocessed input after the input gain, or the sidechain) and the output level processed by the amp, convolution and compressor and performs some operation on these levels.

**Matching** makes the output dynamics match the input dynamics. That's the most typical operation. All the included processors - amp, convolution and compressor - generally lower the output dynamics and lengthen the signal envelopes. Matching then lets you restore some of the original dynamics back.

**Matching (no boost)** works in the same way, but it never amplifies the output signal. This mode is the default.

**Ducking** does the opposite - it lowers the output level if the original level increases. While this might be useful for classic side-chain ducking, it seems very useful with integrated effects as well since it produces a sort of pumping, highlighting the prolonged signal envelopes. Ducking generally requires some sort of threshold, which is defined by the **Range** parameter, it is '-range' to be exact.

Drive 0.00 dB **Drive**

Drive defines the input gain, the power modification applied to the input signal. Please note that the input signal level (if the side-chain is not used) is measured after this input gain, hence it also affects the integrated processors, especially amp.

Range: -24.00 dB to +24.00 dB, default 0.00 dB

Range +40.00 dB **Range**

Range controls the maximum gain that the plugin can perform. Please note that its effect is not the same as dry/wet nor as the dynamics parameters, since it only ensures the gain won't exceed a particular level, but otherwise there is no change in the dynamics processing.

Range: 0.00 dB to +60.00 dB, default +40.00 dB

Output 0.00 dB **Output gain**

Output gain defines the power modification applied to the output signal. It is performed after the dynamics processing, hence its main purpose is to fix the output level so that the **Dry/Wet** parameter is easier to use.

Range: -24.00 dB to +24.00 dB, default 0.00 dB

Speed 40 ms **Speed**

Speed defines the time response of the plugin. It essentially controls the attack, release and other characteristics of both level followers - the first one used on the original input / side-chain and the second used on the processed signal.

Range: 1.0 ms to 1000 ms, default 40 ms

Attack 0.00% **Attack**

Attack controls the time response of the plugin in addition to the **Speed** parameters and its effect usually sounds as changing the level of the initial transient, the attack. Therefore it is a very effective way to manipulate the envelope of the sound, especially useful on drums.

Range: -100.0% to +100.0%, default 0.00%

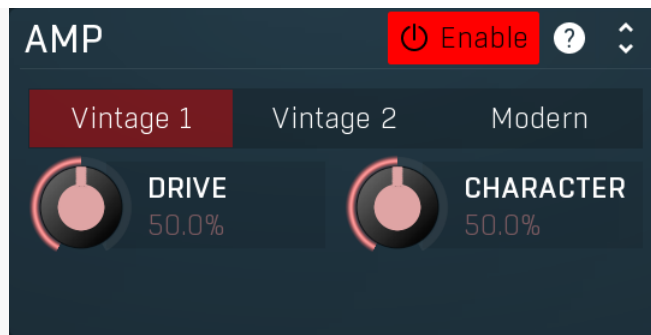
Sustain 0.00% **Sustain**

Sustain controls the time response of the plugin in addition to the **Speed** parameters and its effect usually sounds as changing the length of the sustain. While the amp, convolution and compressor usually lower the dynamics eventually making the signal envelope longer, the processor is designed to restore the original dynamics, but using this sustain parameter you can actually make the envelope longer intentionally. That's very useful for phattening drums for example.

Range: -100.0% to +100.0%, default 0.00%

## Amp panel





Amp panel controls the integrated vintage amp simulator. The effect order is amp -> convolution -> compressor.



### Type

Type defines the sound character, which basically controls the amount and dirtiness of the distortion.



### Drive

Drive controls the input gain resulting in the amount of distortion.

Range: 0.00% to 100.0%, default 50.0%



### Character

Character controls the distribution of higher harmonics, hence, again, sound character.

Range: 0.00% to 100.0%, default 50.0%

## Compressor panel



Compressor panel controls the integrated compressor. The effect order is amp -> convolution -> compressor.



### Dry/Wet

Dry/Wet defines the ratio between dry and wet signals for the compressor. 100% means fully processed, 0% means no processing at all.

This feature essentially provides a modern way to do so-called parallel (or 'New York') compression. Essentially there are main 2 approaches to compression - A) set the threshold high, so that it affects everything above it, B) set the threshold low and use dry/wet to actually lower the effect of compression, which provides an easy way to control the amount of compression without too much editing of the more advanced parameters. Please note that lowering ratio does NOT have the same effect as lowering dry/wet in most cases.

Range: 0.00% to 100.0%, default 50.0%



### Output gain

Output gain defines the power modification applied to the output signal. Since the compressor output is followed by the main dynamics processing, this can have a big impact on the output.

Range: 0.00 dB to +40.00 dB, default +12.00 dB

## Threshold -40.0 dB **Threshold**

Threshold determines the minimum signal level above which the compression effect starts to apply.

Range: -80.0 dB to 0.00 dB, default -40.0 dB

## Ratio Infinity **Ratio**

Ratio defines the compression ratio of the input signal above the threshold. The higher the ratio, the more compression you get.

Range: 1.00 : 1 to Infinity, default Infinity

## Attack 10 ms **Attack**

Attack defines the attack time, that is how quickly the level detector increases the measured input level. When the input peak level is higher than the current level measured by the detector, the detector moves into the attack mode, in which the measured level is increased depending on the input signal. The higher the input signal, or the shorter the attack time, the faster the measured level rises. Once the measured level exceeds the **Threshold** then the dynamics processing (compression, limiting, gating) will start.

There must be a reasonable balance between attack and **release** times. If the attack is too long compared to the release, the detector will tend to keep the measured level low, because the release would cause that level to fall too quickly. In most cases you may expect the attack time to be shorter than the release time.

To understand the working of a level detector, it is best to cover the typical cases:

*In a **compressor** the attack time controls how quickly the measured level moves above the threshold and the processor begins compressing. As a result, a very short attack time will compress even the beginning transient of a snare drum for example, hence it would remove the punch. With a very long attack time the measured level may not even reach the threshold, so the compressor may not do anything.*

*In a **limiter** the attack becomes a very sensitive control, defining how much of the signal is limited and how much of it becomes saturated/clipped. If the attack time is very short, limiting starts very quickly and the limiter catches most peaks itself and reduces them, providing lower distortion, but can cause pumping. On the other hand, a higher attack setting (typically above 1ms) will let most peaks through the limiter to the subsequent in-built clipper or saturator, which causes more distortion of the initial transient, but less pumping.*

*In a **gate** the situation is similar to a compressor - the attack time controls how quickly the measured level can rise above the threshold at which point the gate opens. In this case you will usually need very low attack times, so that the gate reacts quickly enough. The inevitable distortion can then be avoided using look-ahead and hold parameters.*

*In a modulator, the detector is driving other parameters, a filter cut-off frequency for example, and the situation really depends on the target. If you want the detector to react quickly on the input level rising, use a shorter attack time; if you want it to follow the flow of the input signal slowly, use longer attack and release times.*

Range: 0 ms to 1000 ms, default 10 ms

## Release 100 ms **Release**

Release defines the release time, that is how quickly the level detector decreases the measured input level. The shorter the release time, the faster the response is. Once the attack stage has been completed, when the input peak level is lower than the current level measured by the detector, the detector moves into the release mode, in which the measured level is decreased depending on the input signal. The lower the input signal, or the shorter the release time, the faster the measured level drops. Once the measured level falls under the **Threshold** then the dynamics processing (compression, limiting, gating) will stop.

There must be a reasonable balance between **attack** and release times. If the attack is too long compared to release, the detector would tend to keep the level low, because release would cause the level to fall too quickly. Hence in most cases you may expect the attack time to be shorter than the release time.

To understand the working of a level detector, it is best to cover the typical cases:

*In a **compressor** the release time controls how quickly the measured level falls below the threshold and the compression stops. As a result a very short release time makes the compressor stop quickly, for example, leaving the sustain of a snare drum intact. On the other hand, a very long release keeps the compression working longer, hence it is useful to stabilize the levels.*

*In a **limiter** the release time keeps the measured level above the limiter threshold causing the gain reduction. Having a very long release time in this case doesn't make sense as the limiter would be working continuously and the effect would be more or less the same as simply decreasing the input gain manually. However too short a release time lets the limiter stop too quickly, which usually causes distortion as the peaks through the limiter to the subsequent in-built clipper or saturator. Hence release time is used to avoid distortion at the expense of decreasing the output level.*

*In a **gate** the situation is similar to a compressor - the release time controls how quickly the measured level can fall below the threshold at which point the gate closes. Having a longer release time in a gate is a perfectly acceptable option. The release time will basically control how much of the sound's sustain will pass.*

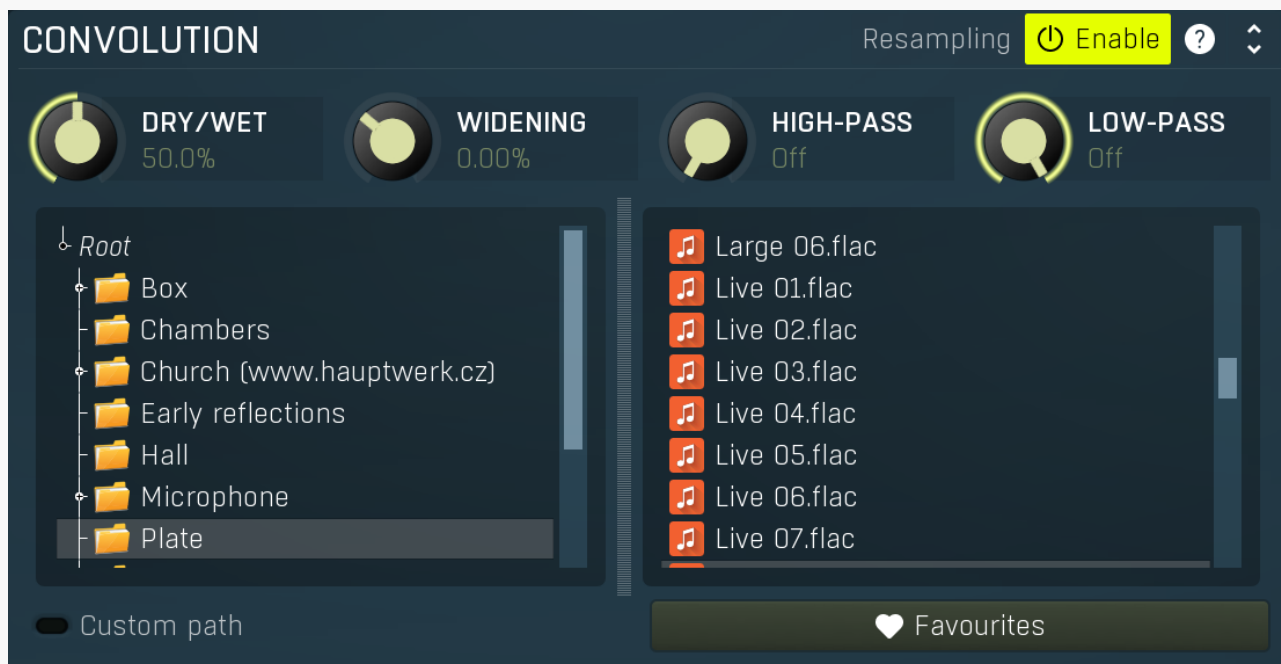
*In a modulator, the detector is driving other parameters, a filter cut-off frequency for example, and the situation really depends on the target. If you want the detector to react quickly on the input level falling, use a shorter release time; if you want it to follow the flow of the input signal slowly, use longer attack and release times.*

Range: 1.0 ms to 10000 ms, default 100 ms

## Measure input **Measure input**

Measure input switch makes the compressor measure the level of the original unprocessed input. That may be handy if you use heavy distortion in amp for example, so that the compressor input already doesn't have much dynamics.

# Convolution panel



Convolution panel controls the integrated convolution reverb. The effect order is amp -> convolution -> compressor.

## Resampling

### Resampling

Resampling activates automatic resampling to current sampling rate. It is disabled by default for sake of backwards compatibility.



### Dry/Wet

Dry/Wet defines ratio between dry and wet signals. 100% means fully processed, 0% means no processing at all.

Range: 0.00% to 100.0%, default 50.0%

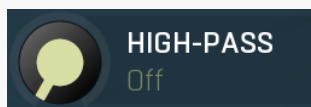


### Widening

Widening defines the broad-band stereo field widening depth. The algorithm is fully mono-compatible as it only extends the existing stereo field and no new signal is added. This parameter should only be used to control the existing stereo field.

Widening converts the audio into its mid (mono) and side channels, leaving the mid intact and applying a gain to the side channel, then converts the signal back to left and right channels. As a result the stereo image becomes wider (for widening above 0%) or narrower (for widening below 0%). This method of widening the stereo image may initially sound pleasing, however it can quickly become fatiguing on the ear and often sounds unnatural, especially for larger amounts of widening. Use this parameter to control the existing stereo field and as a special effect. Use it to increase width only with caution.

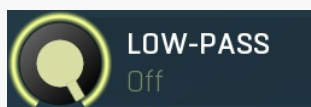
Range: Mono to 200.0%, default 0.00%



### High-pass

High-pass defines the high-pass filter cut-off frequency which may be used to remove part of the convolution spectrum.

Range: Off to 20.0 kHz, default Off



### Low-pass

Low-pass defines the low-pass filter cut-off frequency which may be used to remove part of the convolution spectrum.

Range: 20.00 Hz to Off, default Off

## File

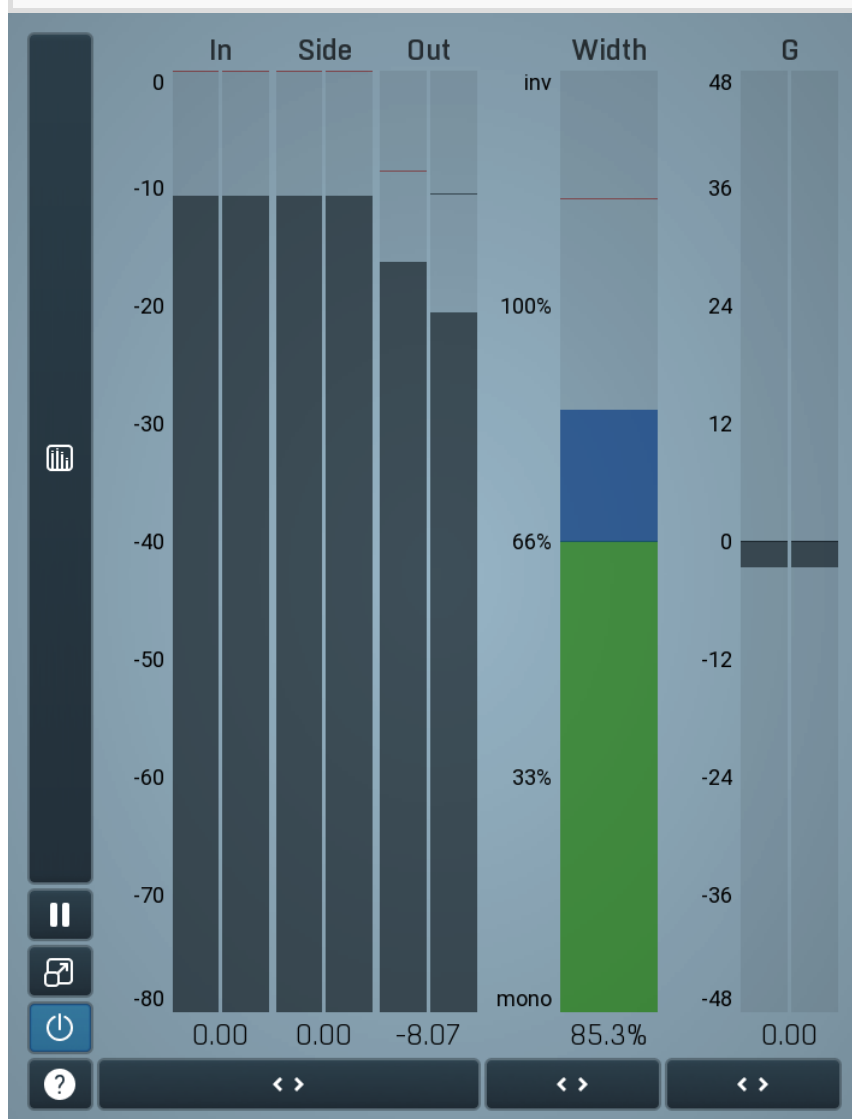


File lets you choose an impulse response file to process. The Factory library installed with the plugin comes with lots of IRs from plates to rooms, halls, box simulations etc. Plates are often a good choice to start with.

### ♥ Favourites

### Favourites

Favourites button lets you manage your personal favourites. Since you may have several file libraries installed on your system, it may be clumsy to search for them every time. Instead you can simply store them in favourites and then quickly go to the particular location without exploring the whole system.



### Global meter view

Global meter view provides a powerful metering system. If you do not see it in the plug-in, click the **Meters** or **Meters & Utilities** button to the right of the main controls. The display can work as either a classical level indicator or, in time graph mode, show one or more values in time. Use the first button to the left of the display to switch between the 2 modes and to control additional settings, including pause, disable

and pop up the display into a floating window. The meter always shows the actual channels being processed, thus in M/S mode, it shows mid and side channels.

In the classical level indicators mode each of the meters also shows the recent maximum value. Click on any one of these values boxes to reset them all.

**In meter** indicates the total input level. The input meter shows the audio level before any specific processing (except potential oversampling and other pre-processing). It is always recommended to keep the input level under 0dB. You may need to adjust the previous processing plugins, track levels or gain stages to ensure that it is achieved.

As the levels approach 0dB, that part of the meters is displayed with **red** bars. And recent peak levels are indicated by single bars.

**Out meter** indicates the total output level. The output meter is the last item in the processing chain (except potential downsampling and other post-processing). It is always recommended to keep the output under 0dB.

As the levels approach 0dB, that part of the meters is displayed with **red** bars. And recent peak levels are indicated by single bars.

**G meter** shows the gain value for each channel. For maximum transparency, you should set the gain as close to 0dB as possible.

**Width meter** shows the stereo width at the output stage. This meter requires at least 2 channels and therefore does not work in mono mode. Stereo width meter basically shows the difference between the mid and side channels.

When the value is **0%**, the output is monophonic. From 0% to 66% there is a green range, where most audio materials should remain.

**From 66% to 100%** the audio is very stereophonic and the phase coherence may start causing problems. This range is colored blue. You may still want to use this range for wide materials, such as background pads. It is pretty common for mastered tracks to lie on the edge of green and blue zones.

**Above 100%** the side signal exceeds the mid signal, therefore it is too monophonic or the signal is out of phase. This is marked using red color. In this case you should consider rotating the phase of the left or right channels or lowering the side signal, otherwise the audio will be highly mono-incompatible and can cause fatigue even when played back in stereo.

For most audio sources the width is fluctuating quickly, so the meter shows a 400ms average. It also shows the temporary maximum above it as a single coloured bar.

If you right click on the meter, you can enable/disable loudness pre-filtering, which uses EBU standard filters to simulate human perception. This may be useful to get a more realistic idea about stereo width. However, since humans perceive the bass spectrum as lower than the treble, this may hide phase problems in that bass spectrum.



### Time graph

Time graph button switches between the metering view and the time-graphs. The metering view provides an immediate view of the current values including a text representation. The time-graphs provide the same information over a period of time. Since different time-graphs often need different units, only the most important units are provided.



### Pause

Pause button pauses the processing.



### Popup

Popup button shows a pop-up window and moves the whole metering / time-graph system into it. This is especially useful in cases where you cannot enlarge the meters within the main window or such a task is too complicated. The pop-up window can be arbitrarily resized. In

metering mode it is useful for easier reading from a distance for example. In time-graph mode it is useful for getting higher accuracy and a longer time perspective.



### Enable

Enable button enables or disables the metering system. You can disable it to save system resources.



### Collapse

Collapse button minimizes or enlarges the panel to release space for other editors.



### Collapse

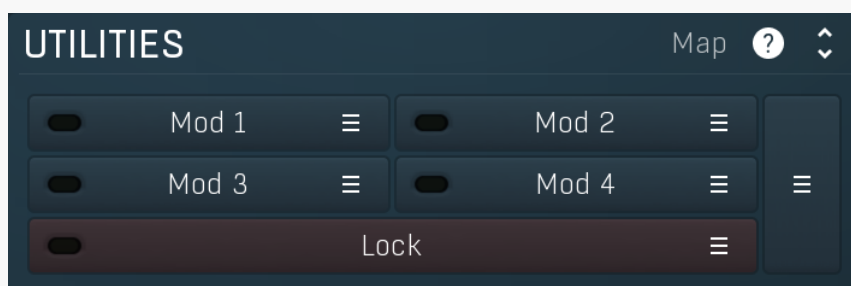
Collapse button minimizes or enlarges the panel to release space for other editors.



### Collapse

Collapse button minimizes or enlarges the panel to release space for other editors.

## Utilities



### Map

Map button displays all current mappings of modulators, multiparameters and MIDI (whichever subsystems the plugin provides).



### Modulator

Modulator button displays settings of the modulator. It also contains a checkbox, to the left, which you can use to enable or disable the modulator. Click on it using your right mouse button or use the **menu button** to display an additional menu with learning capabilities - as described below.



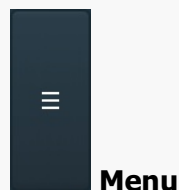
### Menu

Menu button shows the **smart learn** menu. You can also use the right mouse button anywhere on the modulator button.

**Learn** activates the learning mode and displays "REC" on the button as a reminder, **Clear & Learn** deletes all parameters currently associated with the modulator, then activates the learning mode as above. After that every parameter you touch will be associated to the modulator along with the range that the parameter was changed. Learning mode is ended by clicking the button again.

In smart learn mode the modulator does not operate but rather records your actions. You can still adjust every automatable parameter and use it normally. When you change a parameter, the plugin associates that parameter with the modulator and also records the range of values that you set.

*For example, to associate a frequency slider and make a modulator control it from 100Hz to 1KHz, just enable the smart learn mode, click the slider then move it from 100Hz to 1KHz (you can also edit the range later in the modulator window too). Then disable the learning mode by clicking on the button.*



### Menu

Menu button displays additional menu containing features for modulator presets and randomization.



### Lock

Lock button displays the settings of the global parameter lock. Click on it using your left mouse button to open the Global Parameter Lock window, listing all those parameters that are currently able to be locked.

Click on it using your right mouse button or use the **menu button** to display the menu with learning capabilities - **Learn** activates the learning mode, **Clear & Learn** deletes all currently-lockable parameters and then activates the learning mode. After that, every

parameter you touch will be added to the lock. Learning mode is ended by clicking the button again. The On/Off button built into the Lock button enables or disables the active locks.



## Collapse

Collapse button minimizes or enlarges the panel to release space for other editors.

1 :

50.0%



## Multiparameter

Multiparameter button displays settings of the multiparameter. The multiparameter value can be adjusted by dragging it or by pressing Shift and clicking it to enter a new value from the virtual keyboard or from your computer keyboard.

Click on the button using your left mouse button to open the **Multiparameter** window where all the details of the multiparameter can be set. Click on it using your right mouse button or click on the **menu button** to the right to display an additional menu with learning capabilities - as described below.



## Menu

Menu button shows the **smart learn** menu. You can also use the right mouse button anywhere on the multiparameter button.

**Learn** attaches any parameters, including ranges. Click this, then move any parameters through the ranges that you want and click the multiparameter button again to finish. While learning is active, "REC" is displayed on the multiparameter button and learning mode is ended by clicking the button again.

**Clear & Learn** clears any parameters currently in the list then attaches any parameters, including ranges. Click this, then move any parameters through the ranges that you want and click the multiparameter button again to finish. While learning is active, "REC" is displayed on the multiparameter button and learning mode is ended by clicking the button again.

**Reset** resets all multiparameter settings to defaults.

**Quick Learn** clears any parameters currently in the list, attaches one parameter, including its range and assigns its name to the multiparameter. Click this, then move one parameter through the range that you want.

**Attach MIDI Controller** opens the MIDI Settings window, selects a unused parameter and activates MIDI learn. Click this then move the MIDI controller that you want to assign.

**Reorder to ...** lets you change the order of the multiparameters. This can be useful when creating active-presets. Please note that this feature can cause problems when one multiparameter controls other multiparameters, as these associations will not be preserved and they will need to be rebuilt.

In learning mode the multiparameter does not operate but rather records your actions. You can still adjust every automatable parameter and use it normally. When you change a parameter, the plugin associates that parameter with the multiparameter and also records the range of values that you set.

*For example, to associate a frequency slider and make a multiparameter control it from 100Hz to 1KHz, just enable the smart learn mode, click the slider then move it from 100Hz to 1KHz (you can also edit the range later in the Multiparameter window too). Then disable the learning mode by clicking on the button.*



## Collapse

Collapse button minimizes or enlarges the panel to release space for other editors.



