MDynamics

Overview

MDynamics is an advanced dynamic processor with clear sound designed for mastering, however due to its high performance and zero latency, it is suitable for any purpose.

With two compressor/expander units and advanced custom processing shape editing, featuring our incredible MeldaProduction Envelope System (MES) technology, gate and side-chain, MDynamics becomes a total dynamic processing solution!

Easy screen vs. Edit screen

The plugin provides 2 user interfaces - an **easy screen** and an **edit screen**. Use the Edit button to switch between the two.

By default most plugins open on the **easy screen** (edit button released). This screen is a simplified view of the plugin which provides just a few controls. On the left hand side of the plugin you can see the list of available **devices / instruments** (previously called 'active presets'), that is, presets with controls. These controls are actually nothing more than multiparameters (single knobs that can control one or more of the plug-in's parameters and sometimes known as Macro controls in other plug-ins) and are described in more detail later. Each device may provide different controls and usually is intended for a specific purpose. The easy screen is designed for you to be able to perform common tasks, quickly and easily, without the need to use the advanced settings (that is, those available on the Edit screen).

In most cases the devices are highlighted using different text colors. In some cases the colors only mark different types of processing, but in most cases the general rule is that **black/white devices** are the essential ones designed for general use. **Green devices** are designed for a specific task or audio materials, e.g. de-essing or processing vocals in a compressor plugin. **Red devices** usually provide some very special processing or some extreme or creative settings. In a distortion plugin, for example, these may produce an extremely distorted output. **Blue devices** require an additional input, a side-chain or MIDI input usually. Without these additional inputs these **Blue** presets usually do not function as intended. Please check your host's documentation about routing side-chain and MIDI into an effect plugin.

To the right of the controls are the meters or time-graphs for the plugin; the standard plugin Toolbar may be to the right of these or at the bottom of the plugin.

By clicking the **Edit button** you can switch the plugin to **edit mode** (edit button pushed). This mode provides all the of the features that the plugin offers. You lose no settings by toggling between edit mode and the easy screen unless you actually change something. This way you can easily check what is "under the hood" for each device, or start with an device and then tweak the plugin settings further.

Devices are factory specified and cannot be modified directly by users, however you can still make your own and store them as normal presets. To do so, configure the plugin as desired, then define each multiparameter and specify its name in its settings. You can then switch to the easy screen and check the user interface that you have created. Once you are satisfied with it, save it as a normal preset while you

are on the easy screen. Although your preset will not be displayed or selected in the list of available devices, the functionality will be exactly the same. For more information about multiparameters and devices please check the **online video tutorials**.

If you are an advanced designer, you can also view both the easy and edit screens at the same time. To do that, hold **Ctrl** key and press the Edit button.

Edit mode
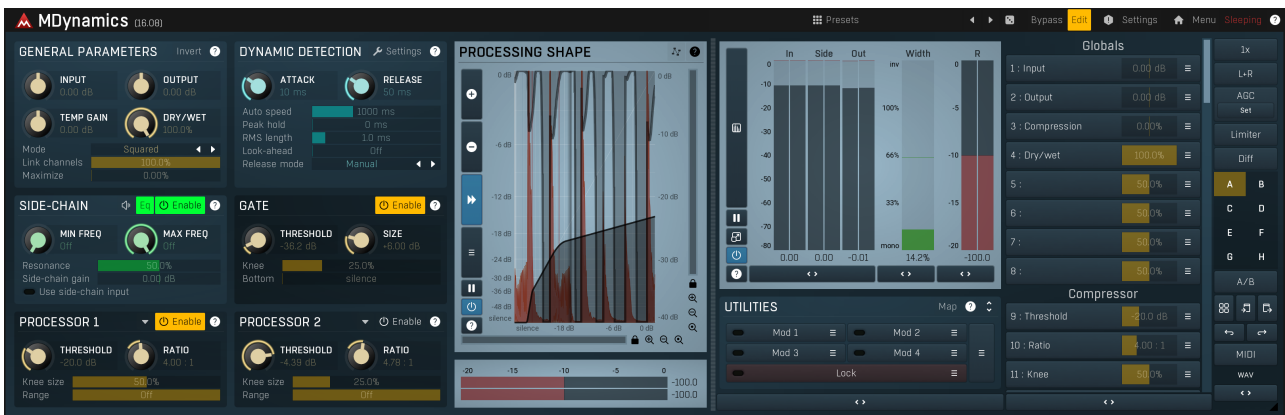
 Presets

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.

 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 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.

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.
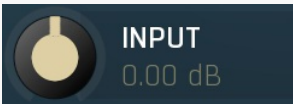
General parameters panel
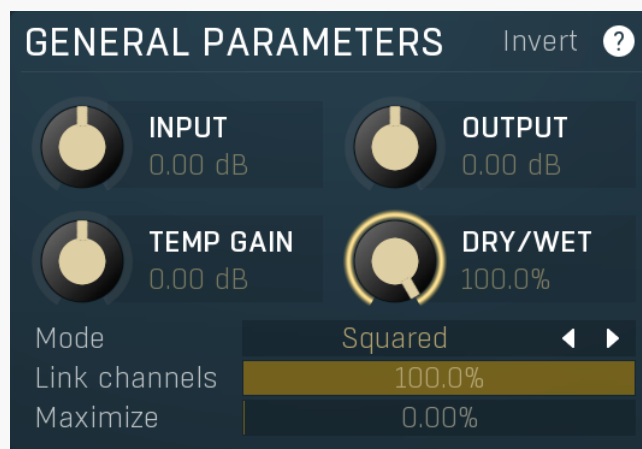
General parameters panel contains the parameters related to dynamic processing.

Invert

Invert reverts the polarity of the compressed signal, so the **Dry/Wet** can then be used to invert the transfer shape. It can be used to convert a gate into a ducker for example, dry/wet can then serve as amount of ducking.

Input gain

Input gain defines gain applied to the incoming signal. If you set ratio to 1:1 and custom shape is disabled, then the plug-in works simply as a fast gain processor.
Range: -24.00 dB to +24.00 dB, default 0.00 dB
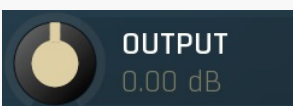
Output gain

Output gain defines the gain applied to the output signal. If you set ratio to 1:1 and custom shape is disabled, then the plug-in works simply as a fast gain processor.
Range: -24.00 dB to +24.00 dB, default 0.00 dB

Temp gain

Temp gain defines the temporary gain applied to the input signal and then reversed on the output. You can achieve the same effect by setting **Input gain** to a value **G** and **Output gain** to value **-G**. Moreover, this plug-in tries to approximate the gain reduction. Absolutely accurate approximation is not possible; however when you set the parameters so that the level is touching the threshold with temporary gain at 0dB, then any change to the temporary gain should change the amount of compression but keep the output level stable. Therefore the temporary gain in fact controls **amount of compression**.
Range: -24.00 dB to +24.00 dB, default 0.00 dB

Dry/Wet

Dry/Wet defines the ratio between dry and wet signals. 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 the dry/wet ratio control to reduce 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 the ratio does NOT have the same effect as lowering dry/wet in most cases.
Range: 0.00% to 100.0%, default 100.0%
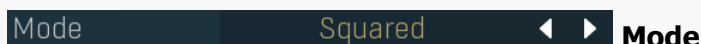
Mode

Mode affects the processing shape. The plug-in features special non-linear transfer shapes which affect the way the signal is processed. **Logarithmic** produces classic dynamic processing where a signal exceeding the threshold by 10dB at a compression ratio of 2 : 1 produces 5dB attenuation in output level. In this same scenario, **Squared** mode produces a slightly greater output attenuation of 6.4dB and **Linear** mode produces a still greater value of 7.5dB. Thus, Squared and Linear modes produce progressively more compression / expansion. There is no compromise in sound quality between the different modes. Comparing the three modes, Linear mode requires the least amount of CPU power, and Logarithmic the most.

Link channels

Link channels controls how much the signal level for each channel is controlled by the other channels. With 0% the link is disabled and each channel is not affected by the other channels at all. This is suitable to balance stereo channels, for example. With 100% the link is enabled and all channels are controlled by levels of all channels equally (that is the average level of those channels), therefore the processor will apply the same amount of processing on all channels. This is the default in most cases as it preserves relative levels between the channels.
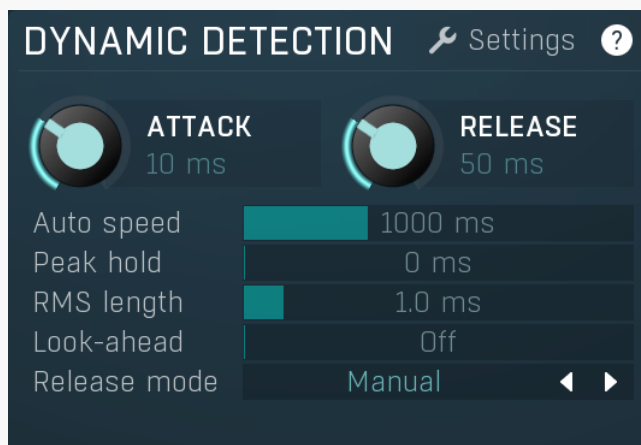Range: 0.00% to 100.0%, default 100.0%

Maximize

Maximize controls the automatic output gain according to current processing shape. In most cases it is better to use the AGC feature and let the processor set the output gain automatically.
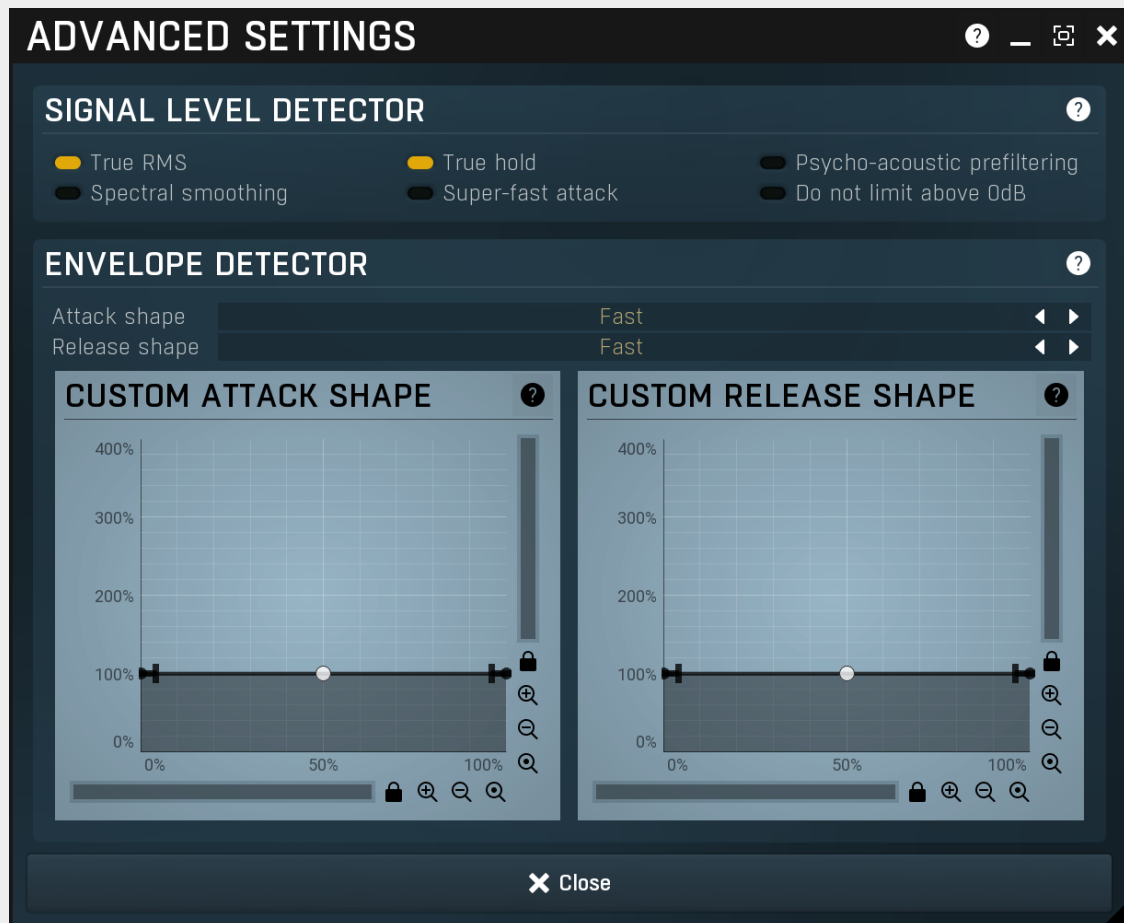Range: 0.00% to 100.0%, default 0.00%

Dynamic detection panel

Dynamic detection panel contains the parameters defining how the plug-in determines the level of the source signal.
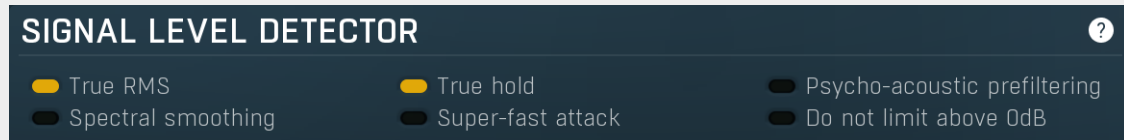
Settings
Settings button shows additional dynamics detector settings.

Advanced settings contains more esoteric and advanced settings of the level detector. These include various kinds of detector signal preprocessing, attack & release responses and custom shapes, etc.

Signal level detector

True RMS

True RMS enables the true RMS calculation instead of the simplified approximation with a slightly different response. When disabled, the calculation is faster and requires almost no memory, however it is also inaccurate. This may not necessarily be a disadvantage, but it may be worth checking the true RMS processor, which provides the standard RMS calculation with the response you would expect. True RMS processing is not much slower than the approximated version, but requires a considerable amount of memory.

True hold

True hold enables the true peak hold algorithm. When disabled, hold is implemented using a special filter which catches peaks and maximizes the level detector signal input by those peaks. In time the peaks decrease in level according to the **hold** parameter. This is effective, requires almost no CPU and memory is required, but it is also inaccurate. For example, since the peaks are not keeping their levels, it cannot be used along with the **look-ahead** feature to avoid distortion in limiters.

True hold, on the other hand, implements the fastest currently-known algorithm to provide the true peak hold response; this does not decay in time and correctly tracks peaks. The typical use in limiters, for example, is to use the same **hold** and **look-ahead** values - the **look-ahead** gives the limiter time and **hold** tracks the highest peaks ahead of the actual dynamic processing. This can highly improve the audio quality by removing unwanted distortion.

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.

Spectral smoothing

Spectral smoothing

Spectral smoothing enables special pre-processing of the level detector signal, aiming to further reduce distortion, especially with low attack values. This feature attempts to make the signal smoother by applying a complex filtering, which does not change the frequency levels. By doing so, you may expect a slower detector response. Limiters need to be extremely quick, hence it is not appropriate for them.

Super-fast attack

Super-fast attack

Super-fast attack ensures the level will never go below the threshold, allowing the dynamic processor to react as quickly as possible, even if **attack** time is higher than 0ms. This is specifically designed for compression and is incompatible with gating and any downwards processing. Note that if you use a soft knee, you may expect gain reduction even if the audio level is very low, or even silence for that matter.

Do not limit above 0dB

Do not limit above 0dB

Do not limit above 0dB ensures that if the input level exceeds 0dB, it will not be limited back to 0dB in the way that analog devices do. It is always recommended to keep your audio signal below 0dB in all stages of the processing.

The digital world allows amplifying signals way above 0dB and attenuation back below 0dB without (or with minimum) artifacts. However digital processors may react differently and most of them are designed and tested for signals below this red line. This plugin is not an exception and without this option enabled it will react in the same way that analog devices would - it would not let the input signal level rise above 0dB.

Envelope detector
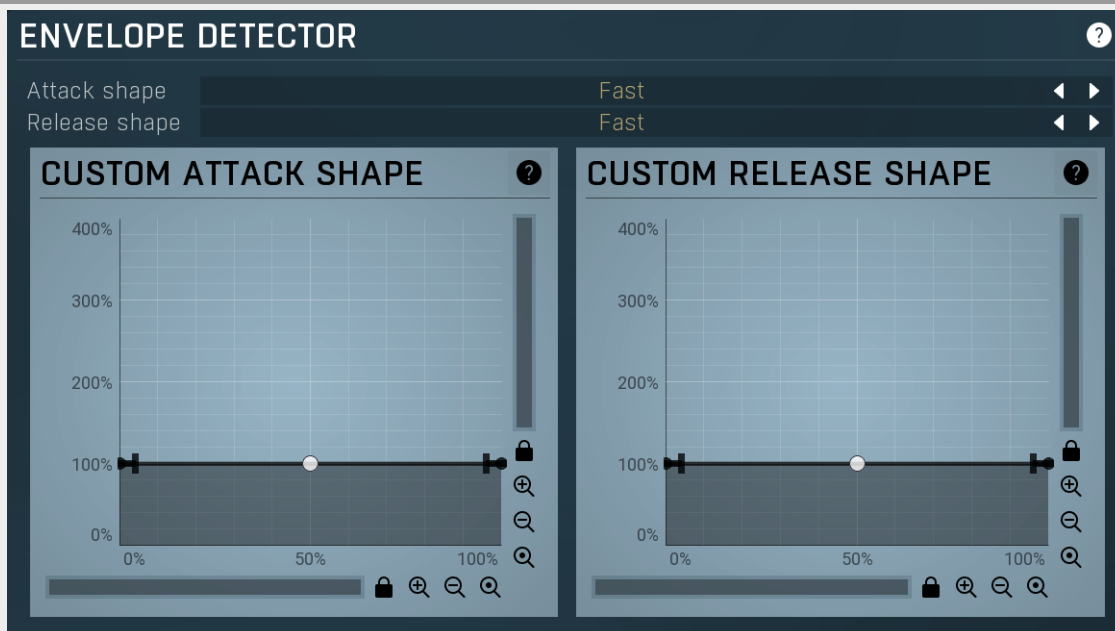
Attack shape

Attack shape controls the shape of the attack stage. The shape mainly affects the **ratio between pumping and distortion**, which simply cannot be avoided. Please note that the attack time parameter is quite dependent upon the mode, so you may expect differences in the actual attack time for different modes of the Attack shape.

Slow modes usually produce more pumping, but less distortion, as the detected level follows the input level more slowly. Conversely **Fast modes** reduce pumping, but cause more distortion. The type of the distortion is different between modes. You may actually profit from the distortion caused by some modes as the generated higher harmonics may enhance the audio. The default **Fast mode** provides a good compromise between distortion and pumping.

There are also **2 custom modes** available. With these modes you can actually draw the shape. Note that what you draw is NOT what you get. The custom shape graph converts the difference between the input level and the current detected level (as represented by the X-axis) into the speed of level detection (as represented by the Y-axis).

*For example, if you set the graph to show 100% across the X axis, then the results will be similar to the **Slow mode**. As the graph is flat, the speed of the detector is the same for all differences between the input and detected levels. If you then move the point on the right upwards to say 400%, it will mean that, if there is a big difference in the levels (a high X value), the detected level will follow the input level 400% faster than it normally would. The closer the detected level gets to the current audio level (a lower X value), the slower the change in the detected level. Similarly, if you take the point on the left and move it downwards to 0%, it will slow down the change to the detected level as it approaches the audio level (a low X value).*
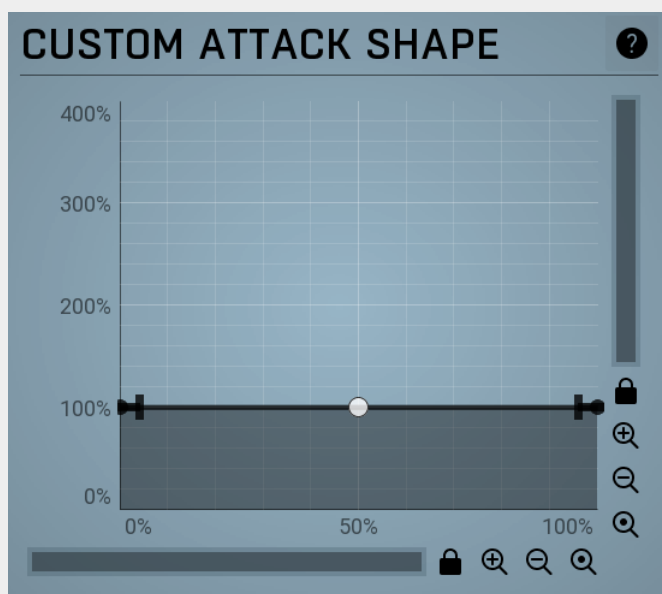
Release

shape

Release shape controls the shape of the release stage. The shape affects the **ratio between pumping and distortion**, which simply cannot be avoided. Please note that the release time parameter is quite dependent on the mode, so you may expect differences in actual release time for different modes of the Release shape.

Slow modes usually produce more pumping, but less distortion, as the detected level follows the input level more slowly. Conversely **Fast modes** reduce pumping, but cause more distortion. The type of the distortion is different between modes. You may actually profit from the distortion caused by some modes as the generated higher harmonics may enhance the audio. The default **Fast mode** provides a good compromise between distortion and pumping.

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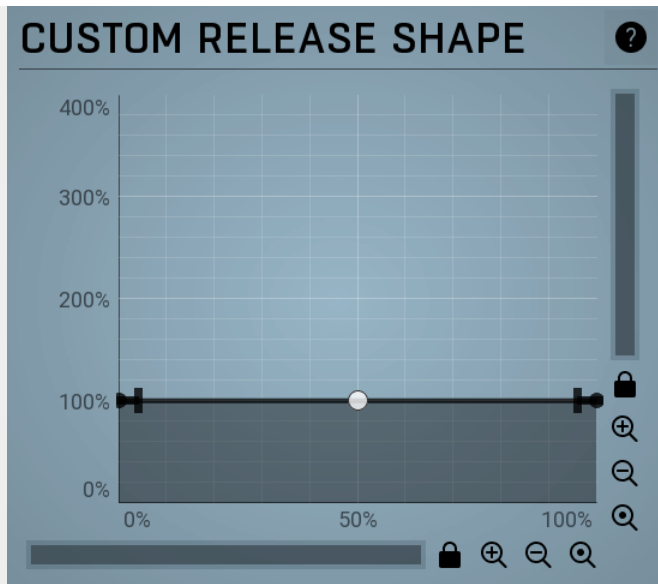

Custom attack shape

Envelope graph

Envelope graph provides an extremely advanced way to edit any kind of shape that you can imagine. An envelope has a potentially unlimited number of points, connected by several types of curves with adjustable curvature (drag the dot in the middle of each arc) and the surroundings of each point can also be automatically smoothed using the smoothness (horizontal pull rod) control. You can also literally draw the shape in drawing mode (available via the main context menu).

- **Left mouse button** can be used to select points. If there is a *point*, you can move it (or the entire selection) by dragging it. If there is a *curvature circle*, you can set up its tension by dragging it. If there is a *line*, you can drag both edge points of it. If there is a *smoothing controller*, you can drag its size. Hold **Shift** to drag more precisely. Hold **Ctrl** to create a new point and to remove any points above or below.

- **Left mouse button double click** can be used to create a new point. If there is a *point*, it will be removed instead. If there is a *curvature circle*, zero tension will be set. If there is a *smoothing controller*, zero size will be set.

- **Right mouse button** shows a context menu relevant to the object under the cursor or to the entire selection. Hold **Ctrl** to create or remove any points above or below.

- **Middle mouse button** drag creates a new point and removes any points above or below. It is the same as holding Ctrl and dragging using left mouse button.

- **Mouse wheel** over a point modifies its smoothing controller. If no point is selected, then all points are modified.

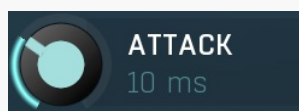- **Ctrl+A** selects all points. **Delete** deletes all selected points.

Custom release shape

Envelope graph

Envelope graph provides an extremely advanced way to edit any kind of shape that you can imagine. An envelope has a potentially unlimited number of points, connected by several types of curves with adjustable curvature (drag the dot in the middle of each arc) and the surroundings of each point can also be automatically smoothed using the smoothness (horizontal pull rod) control. You can also literally draw the shape in drawing mode (available via the main context menu).

- **Left mouse button** can be used to select points. If there is a *point*, you can move it (or the entire selection) by dragging it. If there is a *curvature circle*, you can set up its tension by dragging it. If there is a *line*, you can drag both edge points of it. If there is a *smoothing controller*, you can drag its size. Hold **Shift** to drag more precisely. Hold **Ctrl** to create a new point and to remove any points above or below.

- **Left mouse button double click** can be used to create a new point. If there is a *point*, it will be removed instead. If there is a *curvature circle*, zero tension will be set. If there is a *smoothing controller*, zero size will be set.

- **Right mouse button** shows a context menu relevant to the object under the cursor or to the entire selection. Hold **Ctrl** to create or remove any points above or below.

- **Middle mouse button** drag creates a new point and removes any points above or below. It is the same as holding Ctrl and dragging using left mouse button.

- **Mouse wheel** over a point modifies its smoothing controller. If no point is selected, then all points are modified.

- **Ctrl+A** selects all points. **Delete** deletes all selected points.

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

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 5000 ms, default 100 ms

Auto speed

Auto speed defines how quickly the automatic release works. Specifically how much the release time increases/decreases per second. It is relevant only in **automatic** release modes. *For example if you set it to 5000ms, the release time will be able to increase by 1000ms in 5000ms, when incoming signal exceeds the lowest threshold.*
Range: 1.0 ms to 10000 ms, default 1000 ms

Peak hold

Peak hold defines the time that signal level detector holds its maximum before the release stage is allowed to start. As an example, you can imagine that when an attack stage ends there can be an additional peak hold stage and the level is not yet falling, before the release stage starts. This is true only when **true peak** mode is enabled (check the advanced detector settings if available).

It is often used in **gates** to avoid the gated level falling below the threshold too quickly, while having short release times. If you want the gate to close quickly, you need a short release time. But in that case the ending may be too abrupt and even cause some distortion. So you use the peak hold to delay the release stage.

It is also used along with **look-ahead** to avoid distortion in **limiters and compressors**. If you need a very short attack, the attack stage may be too quick and cause distortions. In limiters this attack time is often 0ms, in which case it becomes a clipper. Setting look-ahead and peak hold to the same value will make the detector move ahead in time, so that it can react to attack stages before they actually occur and yet hold the levels for the actual signal to come.
Range: 0 ms to 1000 ms, default 0 ms

RMS length

RMS length smoothes out the values of the input levels (not the input itself), such that the level detector receives the pre-processed signal without so many fluctuations. When set to its minimum value the detector becomes a so-called "peak detector", otherwise it is an "RMS detector".

When you look at a typical waveform in any editor, you can see that the signal is constantly changing and contains various transient bursts and separate peaks. This is especially noticeable with rhythmical signals, such as drums. Trying to imagine how a typical attack/release detector works with such a wild signal may be complex, at least. RMS essentially takes the surrounding samples and averages them. The result is a much smoother signal with fewer individual peaks and short noise bursts.

RMS length controls how many samples are taken to calculate the average. It stabilizes the levels, but it also causes a slower response time. As such it is great for mastering, when you want to lower the dynamic range in a very subtle way without any instabilities. However, it is not really desirable for processing drums, for example, where the transient bursts may actually be individual drum hits, hence it is usually recommended to use peak detectors for percussive instruments.

Note that the RMS detector has 2 modes - a simplified approximation is used by default, and a true RMS is processor can be enabled

from the advanced settings (if provided). Both respond differently, neither of them is better than the other, they are simply different.
Range: Peak to 100 ms, default 1.0 ms

Look-ahead **Look-ahead**

Look-ahead delays the actual signal being processed, but keeps the detector signal intact. This makes the processor use a signal that has not actually arrived for dynamic calculation. This allows the processor to respond even faster, in fact, ahead of time. This feature is useful for mastering, however it naturally induces latency.

Look-ahead can be available in milliseconds (with obvious meaning) or in percentages. In percentages the look-ahead delay is computed automatically based on the attack and hold times. For example, if look-ahead is 100%, attack time 2ms and peak hold 10ms, then the look-ahead is 10ms; 60% look-ahead would be 7.2ms. If the look-ahead is simply an on/off switch, then it is toggling between 0% and 100% values.

Before using look-ahead, you should understand what such a feature does exactly as the results can potentially be damaging to your audio. Look-ahead basically moves the signal back in time, in other words its signal detector measures the input levels ahead of time. This means that when the detector is in the attack stage, the level is rising, the actual signal is not rising yet, but it will do so soon. However, the same applies to the release stage! When the detector moves to the release stage, the actual signal is not falling yet. This can lead to very strange artifacts (which can be used creatively of course).

The common way to fix this is to set the **release time** considerably higher than the **attack time**. In this way, the level will rise ahead of time in the attack stage, and same will happen for the release stage and the level will go down, however, since the level is falling slowly, the look-ahead will not be that relevant.

Another option is to use the **peak hold** feature. It is highly recommended to enable **true hold** in the advanced detector settings if available. Essentially this feature maximizes the input level over a certain period of time. *So for example, if you set look-ahead to 5ms and peak hold to 5ms as well, the actual signal will arrive 5ms later than the detector signal, however the peak hold feature will ensure that the detector holds the highest peaks for 5ms, so the attack stage will be ahead of time, but the release will not! You can consider it a form of latency compensation for the release stage.*

Look-ahead is commonly used in **limiters** *along with very low (often 0ms) attack times to avoid distortion. With 0ms attack time the limiter is immediately following the input and when the level gets above 0dB, it turns it down to 0dB, so the attack stage is effectively being clipped. To avoid distortion produced by this effect, you can increase look-ahead and peak hold to the same value, say 1ms. As a result the attack stage occurs before it actually occurs, so the distortion is still present, but in much lower levels and usually is masked by the forthcoming transient.*
Range: Off to 1000 ms, default Off

Release mode **Manual** ◀ ▶ **Release mode**

Release mode defines how the plug-in performs when decreasing level. In **manual mode** this is based only on the **release time**, which is suitable for most cases when the signal has constant characteristics. Automatic release modes can adapt to signals with unstable characteristics.

Automatic and **Automatic fast** modes: the longer the level stays above the threshold, the longer the release time will be and thus, the longer it will take to move below the threshold and end the release stage. The idea is that if the input is loud for some time, it will most likely stay that way for some more time, hence it should be stabilized to avoid unnecessary temporary fluctuations, which could result in pumping.
Both automatic modes increase the release time when the input signal is above the threshold and vice versa. The speed of the increase depends on the **Auto speed** parameter. Automatic fast mode uses full speed immediately after crossing the threshold, automatic mode varies the speed according to the current signal level.

For example, when a guitarist plays softly, the level is low and fluctuates around the threshold and the release time gets slower. So the processor quickly responds to sudden changes. However, when the guitarist starts playing a solo, the level rises and, the longer the solo is, the longer the release time becomes, hence the response becomes slower avoiding unnecessary fluctuations (pumping) when the solo contains small silent sections.

Linear 1 and **Linear 2** modes: the higher the level is, the longer the release. The idea is that if the input is very loud, it will probably stay that way for some time, so it is wise to keep the levels up too. This is similar to the automatic modes, however the main factor is not how long the level is high, but how high it is.
Below the threshold the release time is the same as the attack time, above the threshold the release time rises from the attack time up to the specified release time parameter. Linear 1 mode usually provides higher release times than does Linear 2.

Opto mode: the higher the level is, the shorter the release. So this is kind of the opposite of linear modes. The idea is, that you are expecting short transients, which you wish to deal with. Normally the higher the level would get in such a transient, the longer it would take to get the level below the threshold, so, when used in a compressor for example, these transients would cause unnecessary compression in the sustain stage. The opto detector lowers the level quickly, minimizing the amount of compression in the sustain stage.

For example, let's say you are compressing a full drumset, but there is a very dominant sharp and short hi-hat sound, so it is appropriate to have short release times. You would use **Opto** *mode. But the rest of the drumset deserves a softer treatment, so you want to keep longer release times. Use one of the other modes.*

Side-chain panel

Side-chain panel lets you do additional filtering of the level detector input. Please note that its name does not mean it is related to the processor's secondary input (if it has one). Its main purpose is to filter the signal fed to the level detector. This is useful when, for example, you want to remove high volume peaks of a particular frequency.

For example, de-essing typically reduces "s" sound contained at about 2.5kHz and above, which is often far louder than the rest of the recorded voice signal. As another example, you may be processing a drumset, where the hi-hat is too prominent. You would like to compress the hi-hat, but keep the rest of the drumset intact. So you filter out everything except for the hi-hat in the level detector's side-chain and the compressor will listen to the hi-hat only. It is rarely possible to filter out everything except for the requested signal, so compromises need to be made.

Note that the filtered signal is typically lower in amplitude, but volume maximization is not recommended, since it may increase the volume to dangerous levels.

Audition
The Audition button toggles playback of the filtered signal instead of the actual effect output. When enabled, you will hear the actual filtered level detector signal. This may be processed in various ways, but in most cases you will be interested in setting up the side-chain filter.

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.
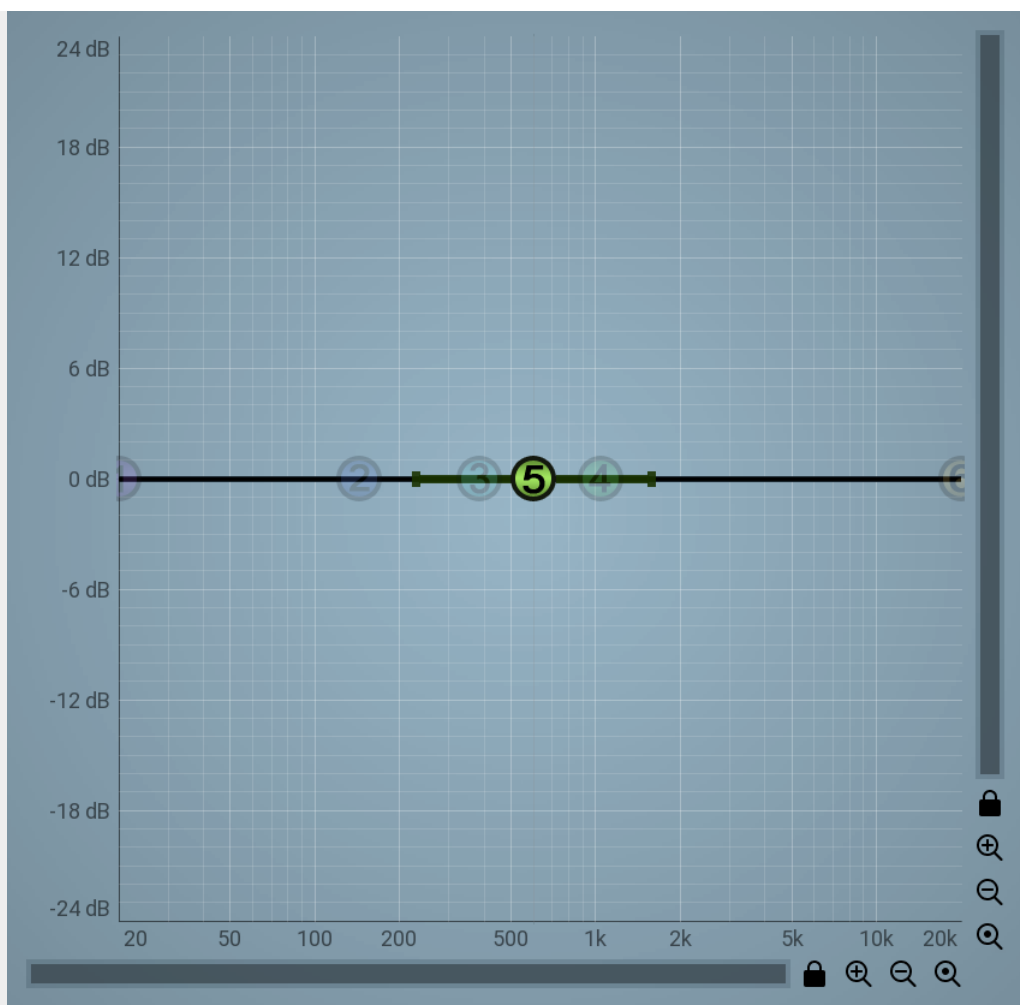
Level equalizer settings

LEVEL EQUALIZER SETTINGS
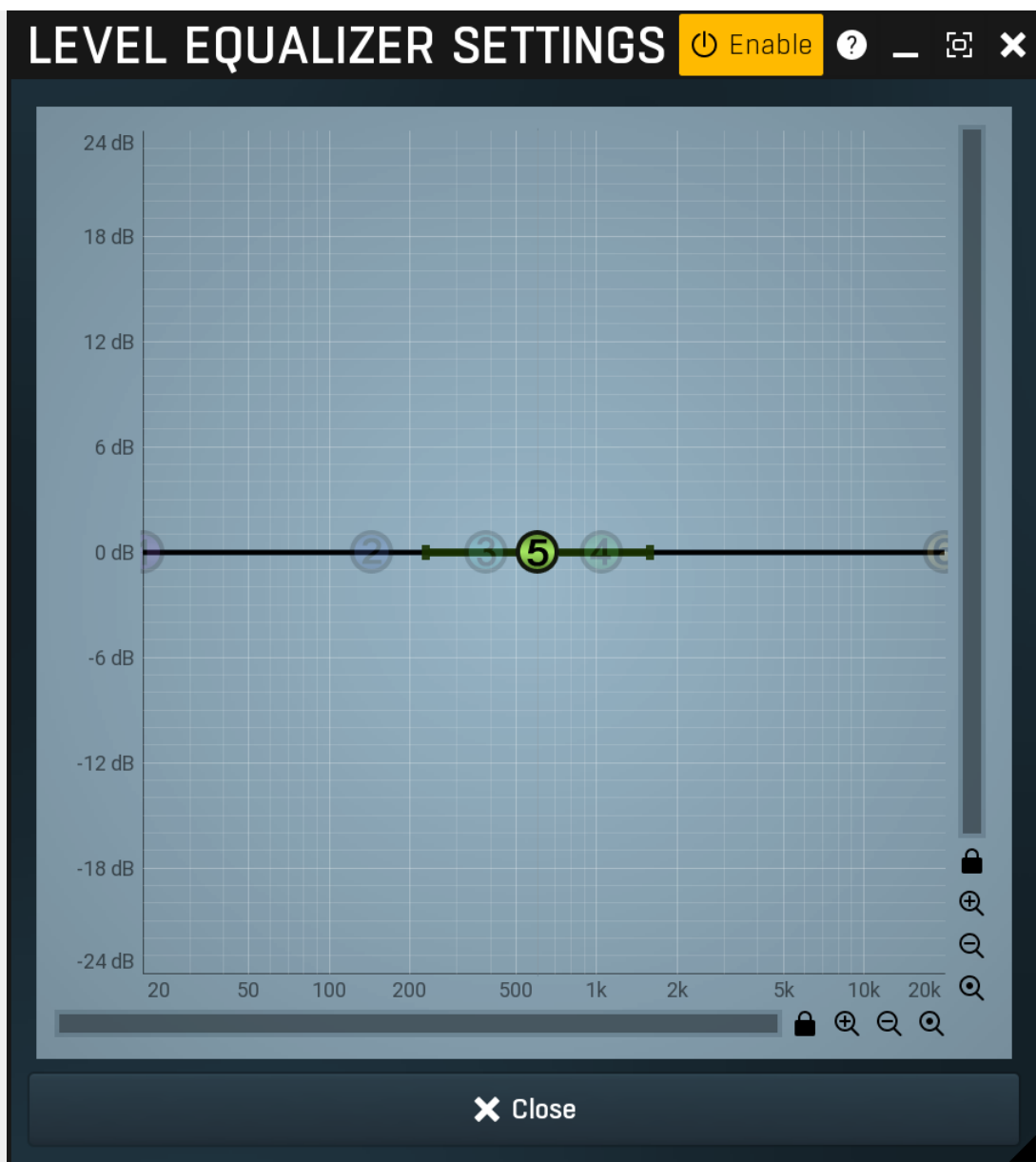
⏻ **Enable** **Enable**

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

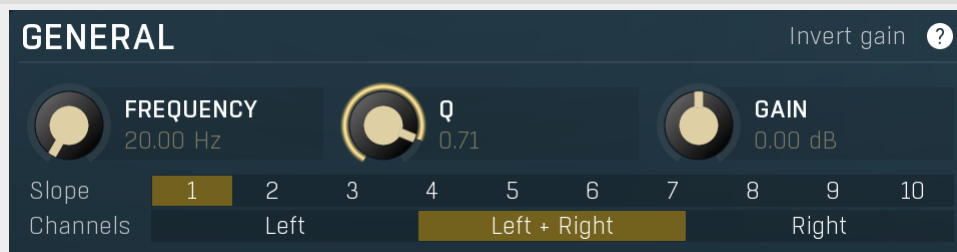Copy button copies the settings onto the system clipboard.

⇤ Paste

Paste button loads the settings from the system clipboard.

🎲 Random

Random button generates random settings using the existing presets.

General panel

GENERAL   Invert gain ❓

FREQUENCY   Q   GAIN
20.00 Hz   0.71   0.00 dB

Slope	1	2	3	4	5	6	7	8	9	10

Channels	Left	Left + Right	Right

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.
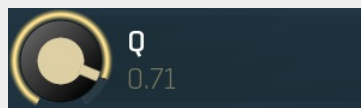
Invert gain   **Invert gain**

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

FREQUENCY
20.00 Hz   **Frequency**

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

Q
0.71   **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.
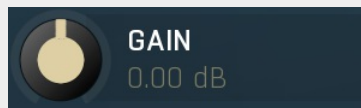
GAIN
0.00 dB   **Gain**

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

Slope	1	2	3	4	5	6	7	8	9	10	**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.
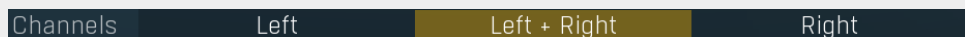
Channels	Left	Left + Right	Right	**Channels**

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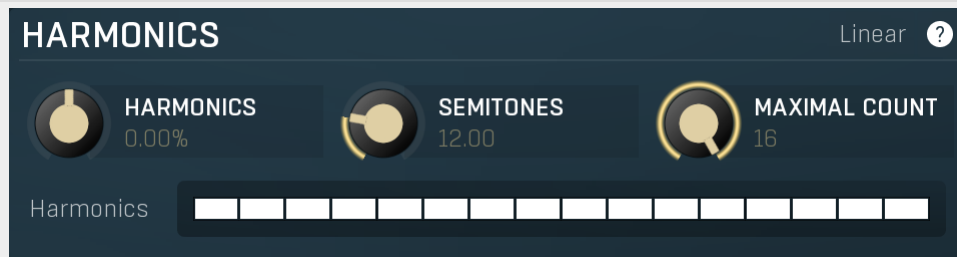
Harmonics panel contains parameters of the harmonics - clones of the main band created at higher frequencies derived from the frequency of the main band. This is often useful for removing natural noises, which usually bring some harmonics with them etc.

Linear

Linear 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.
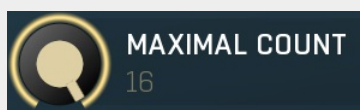
Harmonics

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.
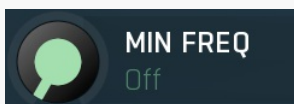
Maximal count

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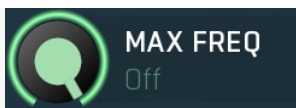
Harmonics grid
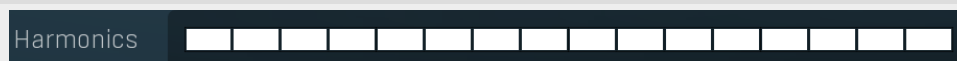
Harmonics grid is useful to turn on/off particular harmonics manually. Click any one to enable / disable it.

Minimal frequency

Minimal frequency defines the side-chain cut-off frequency for the high pass filter - minimal frequency.
Range: Off to 20.0 kHz, default Off

MAX FREQ
Off

Maximal frequency

Maximal frequency defines the side-chain cut-off frequency for the low pass filter - maximal frequency.
Range: 20.00 Hz to Off, default Off

Resonance | 50.0%

Resonance

Resonance defines the resonance of side-chain lowpass filter. Higher resonance makes the filter response steeper, therefore removes more of the frequency content outside of desired range. The resonance of the high-pass (bottom of the frequency range) filter is kept to 0.5 for convenience.
Range: 0.00% to 100.0%, default 50.0%

Side-chain gain | 0.00 dB

Side-chain gain

Side-chain gain defines gain applied to the metering signal, whether it is prefiltered or not, and whichever input channel it comes from.
Range: -24.00 dB to +24.00 dB, default 0.00 dB

Use side-chain input

Use side-chain input

Use side-chain input enables the secondary input channel to be used for the level detector. If you want to drive the processor with a signal other than the main input, you need to route it to the plugin and enable this switch. If disabled the first channel is used as an audio input and its filtered version as the side-chain.
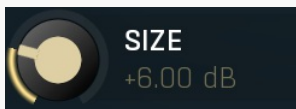
Gate panel

GATE   ⏻ Enable ?

THRESHOLD   SIZE
-36.2 dB   +6.00 dB

Knee   25.0%
Bottom   silence
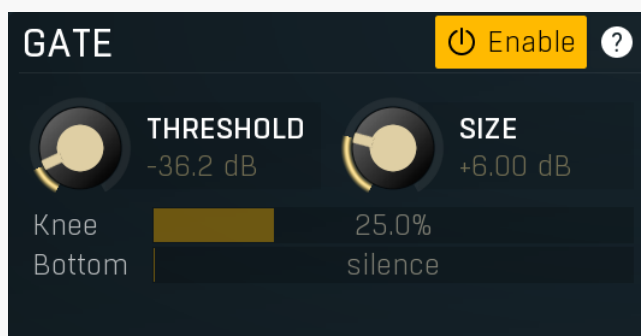
Gate panel contains parameters for the noise-gate.
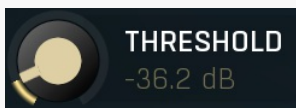
THRESHOLD
-36.2 dB

Threshold

Threshold determines the maximum signal level below which the effect starts to apply.
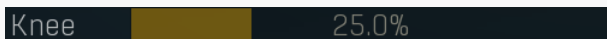Range: -80.0 dB to 0.00 dB, default -28.0 dB

SIZE
+6.00 dB

Size

Size defines size of the interval between the gate threshold and point when the output signal level reaches zero.
Range: 0.00 dB to +24.00 dB, default +6.00 dB

Knee | 25.0%

Knee

Knee defines the size of the smoothing knee.
Range: 0.00% to 100.0%, default 25.0%

Bottom | silence

Bottom

Bottom defines the volume reached when the gate is fully closed, hence the Threshold minus Size. In most cases you can leave this set to **silence**.
Range: silence to -80.0 dB, default silence

Processor 1 panel

Processor 1 panel contains parameters of the primary processor, which can behave like a compressor or expander.

▼ Downwards

Downwards button switches the processor into a downward. expander. In this mode the processor reacts to levels below the threshold, instead of above the threshold as in normal mode. Despite this, upwards compression can be done too. This mode is particularly useful for expansion, since upward expansion is somewhat dangerous as it can significantly amplify the audio way above 0dB.

Compression reduces the dynamic range of sounds above the threshold level, reducing the level over the threshold by the ratio. For a ratio of 1.50:1, 9 dB over the threshold will be reduced to 6 dB over. Levels below the threshold are not changed.
Downward expansion increases the dynamic range of sounds below the threshold level, reducing the level under the threshold by the ratio. For a ratio of 1.50:1, 9 dB under the threshold will be reduced to 13.5 dB under. Levels above the threshold are not changed.

 Threshold

Threshold determines the minimum signal level above which the compression effect starts to apply.
Range: -80.0 dB to 0.00 dB, default -12.0 dB

 Ratio

Ratio defines the compression ratio of the input signal above the threshold. The higher the ratio, the more compression you get.
Range: 1 : 3.00 to Infinity, default 1.00 : 1

 Knee size

Knee size defines size of the knee.
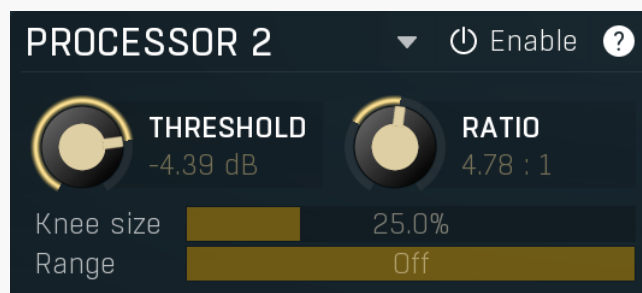Range: 0.00% to 100.0%, default 25.0%

 Range

Range defines size of the interval above the threshold after which the original signal ratio is restored.
Range: +1.00 dB to Off, default Off

Processor 2 panel

Processor 2 panel contains parameters of the secondary processor, which can behave like a compressor or expander.

▼ Downwards

Downwards button switches the processor into a downward. expander. In this mode the processor reacts to levels below the threshold, instead of above the threshold as in normal mode. Despite this, upwards compression can be done too. This mode is particularly useful for expansion, since upward expansion is somewhat dangerous as it can significantly amplify the audio way above 0dB.

Compression reduces the dynamic range of sounds above the threshold level, reducing the level over the threshold by the ratio. For a

ratio of 1.50:1, 9 dB over the threshold will be reduced to 6 dB over. Levels below the threshold are not changed.
Downward expansion increases the dynamic range of sounds below the threshold level, reducing the level under the threshold by the ratio. For a ratio of 1.50:1, 9 dB under the threshold will be reduced to 13.5 dB under. Levels above the threshold are not changed.

THRESHOLD
-4.39 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 -24.1 dB

RATIO
4.78 : 1

Ratio
Ratio defines the compression ratio of the input signal above the threshold. The higher the ratio, the more compression you get.
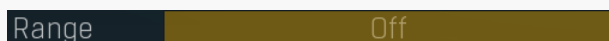Range: 1 : 3.00 to Infinity, default 1.00 : 1

Knee size	25.0%

Knee size
Knee size defines size of the knee.
Range: 0.00% to 100.0%, default 25.0%

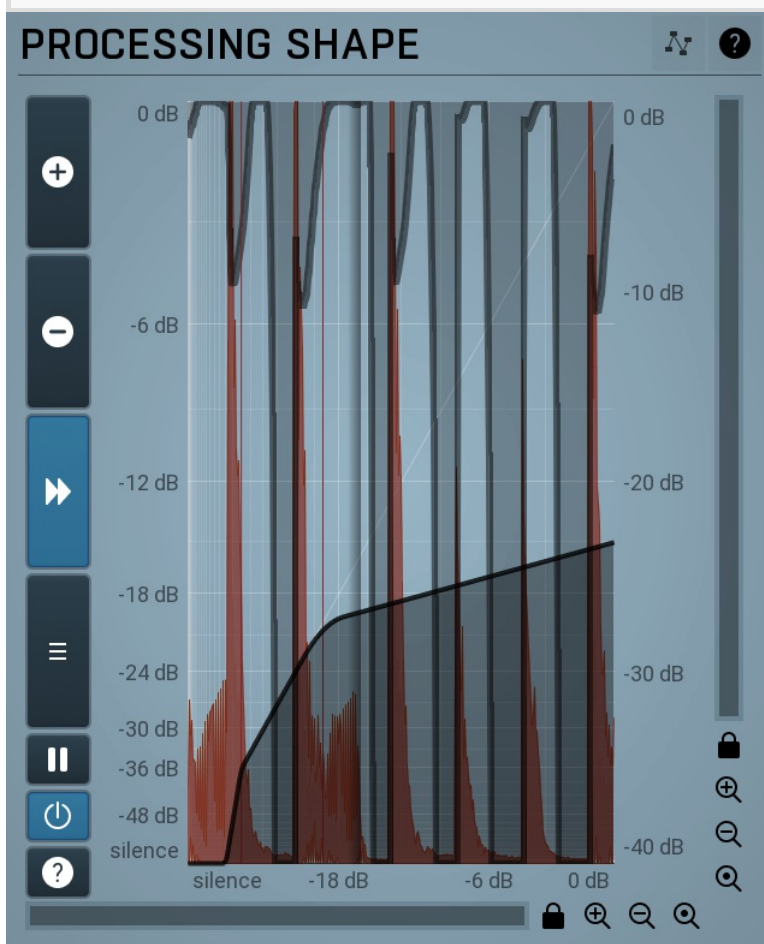| Range	Off

Range
Range defines size of the interval above the threshold after which the original signal ratio is restored.
Range: +1.00 dB to Off, default Off

Level shape graph
Level shape graph displays the dynamic processing transformation shape. The X axis represents the input signal level, Y axis defines the output level.

Please note that this display is not logarithmic. This can lead to confusion, as, for example, a moving expander's threshold changes the graph's slope while the ratio stays the same. This is however necessary, because a logarithmic display can never contain silence, as it is minus infinity decibels, and the silence point is essential for gates for example. The display is therefore a compromise between usability and accuracy.

The moving vertical line shows the current detected level. It may be moving extremely quickly depending on the settings. It may also be invisible if the input level is silence or above 0dB (which is not recommended unless you are using the processor as a limiter). There may be other graphs available, such as input & output waveform and gain reduction time graphs.

Graph
Graph button enables or disables the custom level shape. When enabled, it inherits the automatic settings and you can draw any processing

shape you want.

Plus

Plus button increases the time-graph speed (reduces the period that is displayed).

Minus

Minus button decreases the time-graph speed (increases the period that is displayed).

Rewind

Rewind button enables or disables the time-graph static mode. In static mode the graphs are fixed and the current position cycles from left to right; otherwise the graphs move from right to left and the current position is fixed (at the right-hand side).

Menu

Menu button displays the time-graph settings. In this window you can control which graphs are displayed, the speed and other relevant parameters.
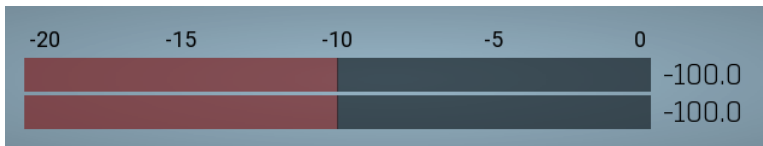
Pause

Pause button pauses the processing.

Enable

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

-20	-15	-10	-5	0

-100.0
-100.0

Meters

Meters display gain-reduction for each channel being processed. Also it contains controls to manipulate time-graphs shown in the transformation shape graph above.
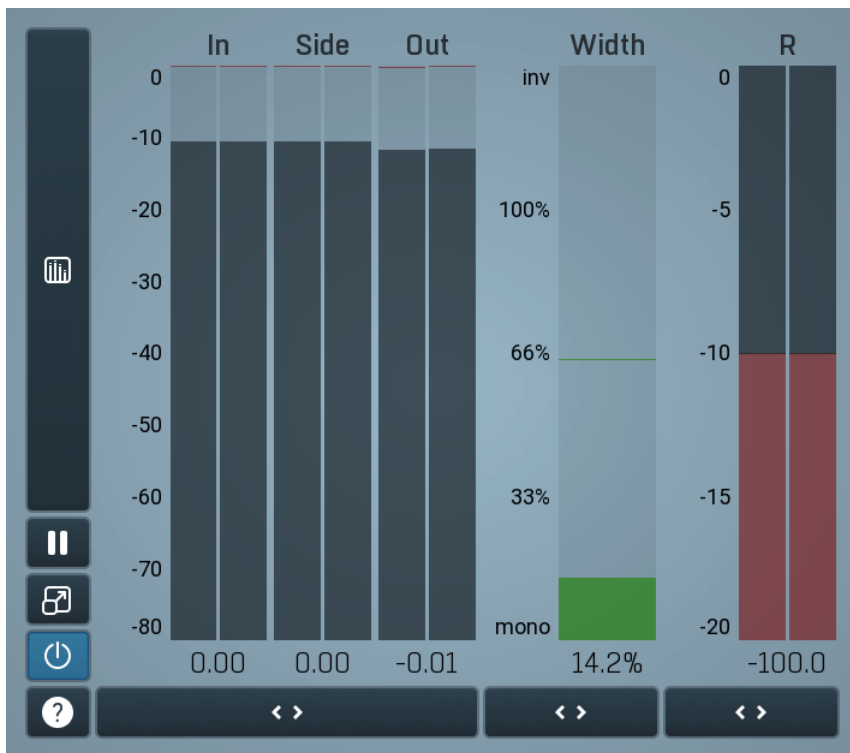
Global meter view

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.

R meter shows gain reduction for each channel. Negative values, running down from the top, mean that compression or limiting is occurring. The lower the value, the stronger the effect. For maximum transparency you should try to achieve the least amount of gain reduction. Expansion is not indicated in this meter.

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.
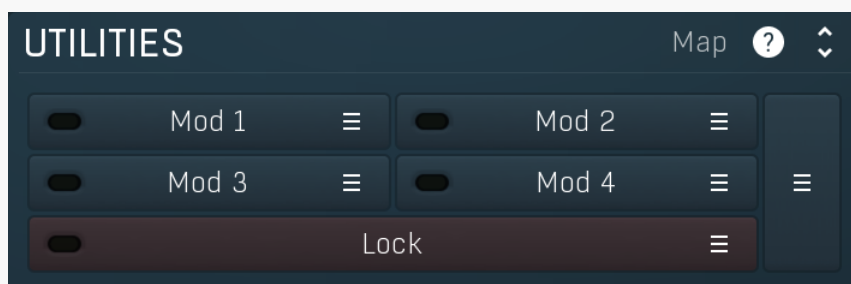
Collapse

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.

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.