MLimiterX

Overview

MLimiter is a state-of-the-art mastering brickwall limiter that makes your recordings sound louder with minimum distortion and artifacts and best of all, you do not need to be a scientist to use it!

MLimiter is built on top of our MDynamics kernel, and its highly simplified user interface provides all the features you need to perform high-quality limiting quickly and easily.

Introduction

A brickwall limiter increases loudness by reducing the ratio between the average and peaks in the signal, however the dynamics of the audio material is always sacrificed. MLimiter is simple to use. Watch the **gain reduction meter** ("R") to the right, and manipulate the **Threshold**. Decreasing the threshold increases the output gain and allows limiting up to 0dB. Care is needed as this can cause severe distortions if not used correctly. In most cases you don't need to worry about the edit screen, just simply focus on the easy screen, with its simplified 4-knob interface.

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.

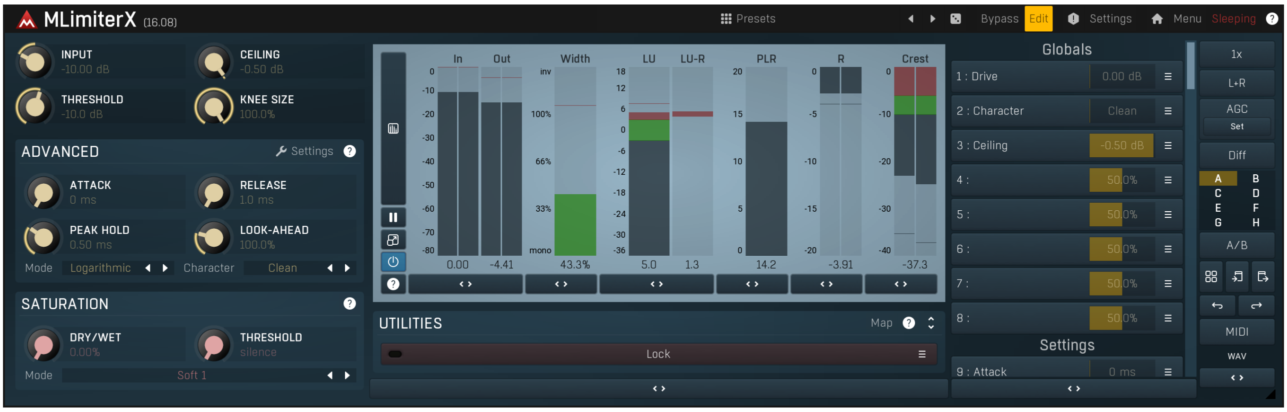
To the right of the controls are the meters or time-graphs for the plugin; the standard plugin Toolbar may be to the right of these or at the bottom of the plugin.

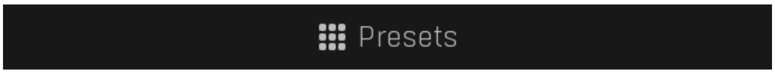
By clicking the **Edit button** you can switch the plugin to **edit mode** (edit button pushed). This mode provides all the of the features that the plugin offers. You lose no settings by toggling between edit mode and the easy screen unless you actually change something. This way you can easily check 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

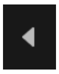
Presets button shows a window with all available presets. A preset can be loaded from the preset window by double-clicking on it, selecting via the buttons or by using your keyboard. You can also manage the directory structure, store new presets, replace existing ones etc. Presets are global, so a preset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

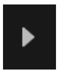
Files are named based on the name of the plugin like this: "{pluginname}.presets", so for example MAutopan.presets or MDynamics.presets. If the directory cannot be found on your computer for some reason, you can just search for the particular file.

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

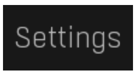
Right arrow button loads the next preset.

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).

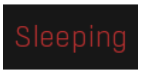
Advanced settings configures several processing options for the plug-in.

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.

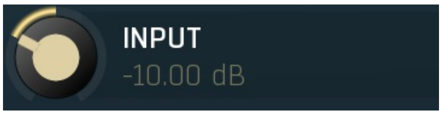
Dry/Wet affects determines, for Multiband plug-ins, which multiband parameters are affected by the Global dry/wet control.

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.

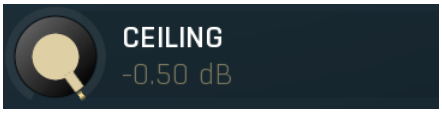
WWW
WWW button shows a menu with additional information about the plugin. You can check for updates, get easy access to support, MeldaProduction web page, video tutorials, Facebook/Twitter/YouTube channels and more.



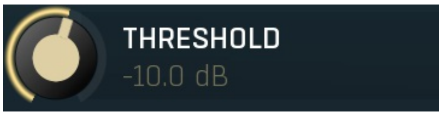
Sleep indicator
Sleep indicator informs whether the plugin is currently active or in sleep mode. The plugin can automatically switch itself off to save CPU, when there is no input signal and the plugin knows it cannot produce any signal on its own and it generally makes sense. You can disable this in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



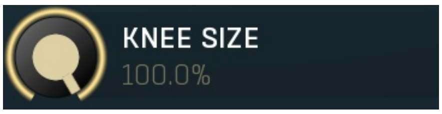
Input gain
Input gain defines gain applied to the incoming signal. If you set the ratio to 1:1 and custom shape is disabled, then the plug-in works simply as a fast gain processor.
Range: -24.00 dB to +24.00 dB, default 0.00 dB



Ceiling
Ceiling defines gain applied to the output signal after the saturation. Therefore this defines the maximum level that the output can reach.
Range: -24.00 dB to 0.00 dB, default 0.00 dB

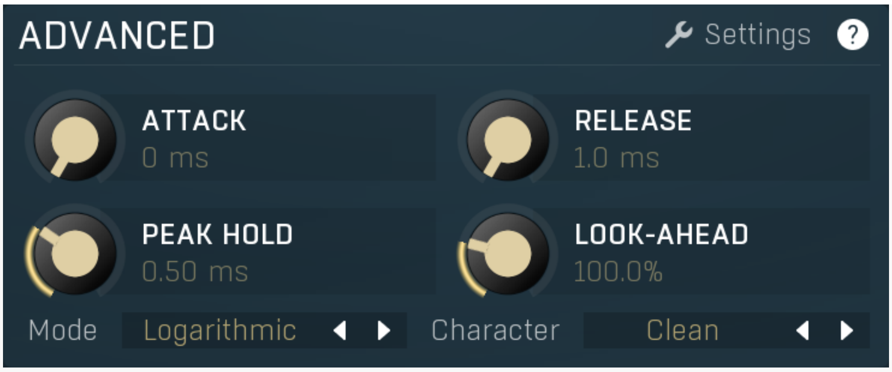


Threshold
Threshold determines the minimum signal level above which the compression effect starts to apply.
Range: -80.0 dB to 0.00 dB, default 0.00 dB

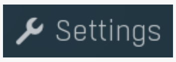


Knee size
Knee size defines size of the knee.
Range: 0.00% to 100.0%, default 100.0%

Advanced panel



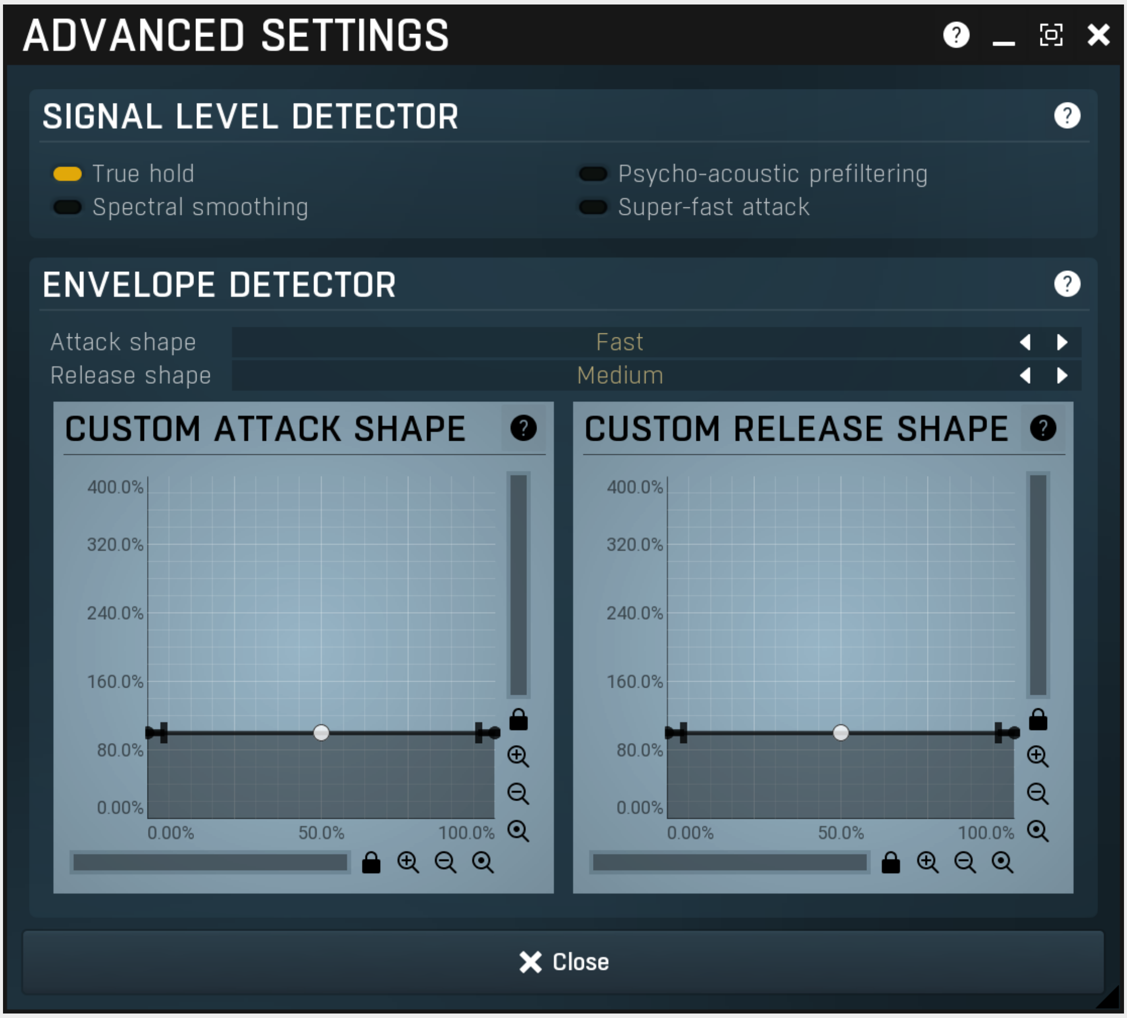
Advanced panel contains more advanced parameters defining how the plug-in determines the level of the source signal, dynamic shape transformation etc.



Settings

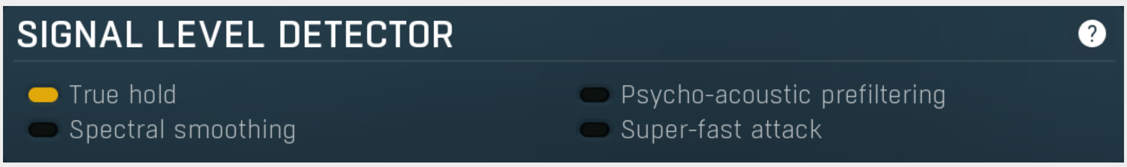
Settings button shows additional dynamics detector settings.

Advanced settings



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

Signal level detector



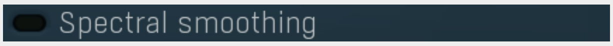
True hold

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.



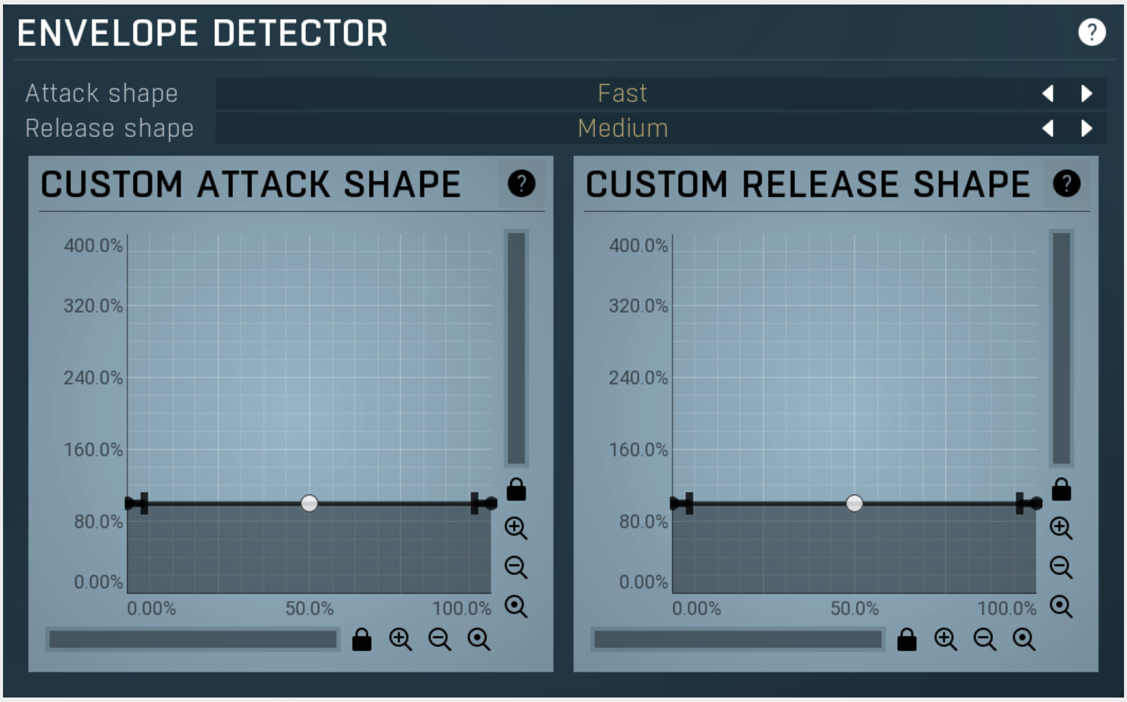
Spectral smoothing

Spectral smoothing 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 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.

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 producemore 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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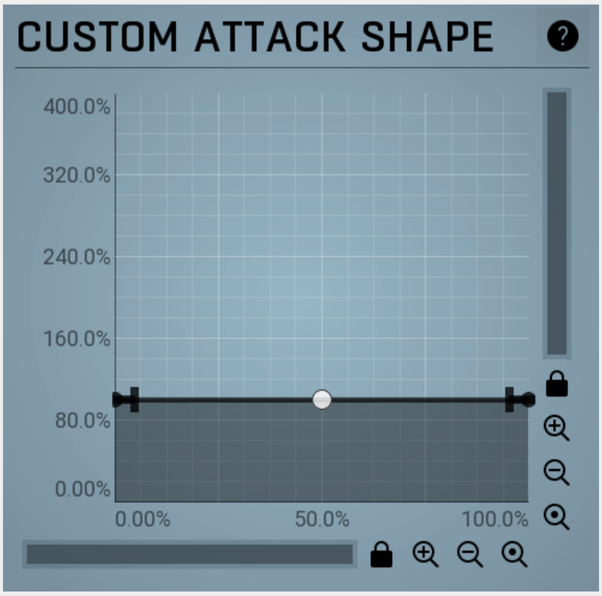
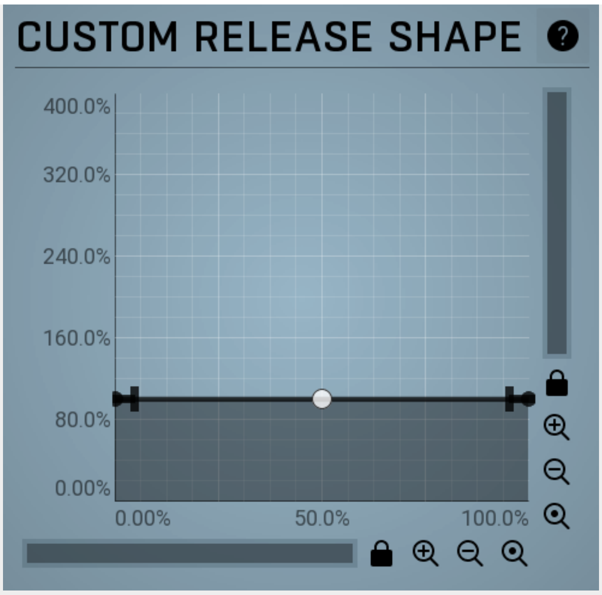


Custom attack shape

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

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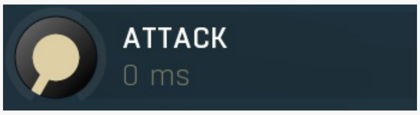


Attack

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:

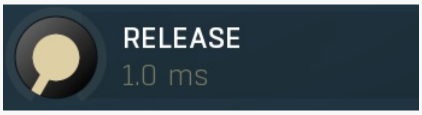
*In a **compressor** the attack time controls how quickly the measured level moves above the threshold and the processor begins compressing. As a result, a very short attack time will compress even the beginning transient of a snare drum for example, hence it would remove the punch. With a very long 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 100 ms, default 0 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.

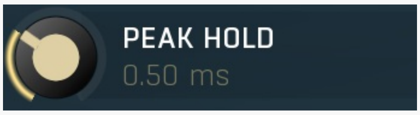
To understand the working of a level detector, it is best to cover the typical cases:

*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 500 ms, default 1.0 ms

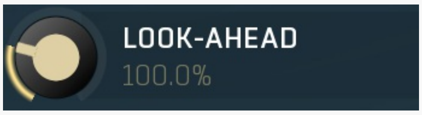


Peak hold

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 50 ms, default 0 ms



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.

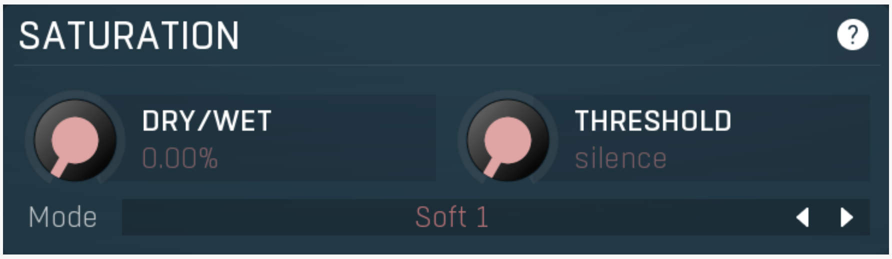
Another option is to use the **peak hold** feature. It is highly recommended to enable **true hold** in the advanced detector settings if available. Essentially this feature maximizes the input level over a certain period of time. *So for example, if you set look-ahead to 5ms and peak hold to 5ms as 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: 0.00% to 400.0%, default 100.0%

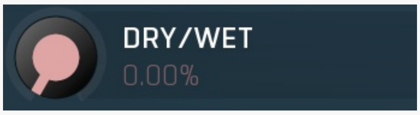
Mode
Mode affects the processing shape. The plug-in features special non-linear transfer shapes which affect the way the signal is processed. **Logarithmic** produces classic dynamic processing where a signal exceeding the threshold by 10dB at a compression ratio of 2 : 1 produces 5dB attenuation in 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.

Character
Character defines the sound character. **Crisp** provides sharper results with some amount of saturation. **Clean** usually provides more transparent results.

Saturation panel



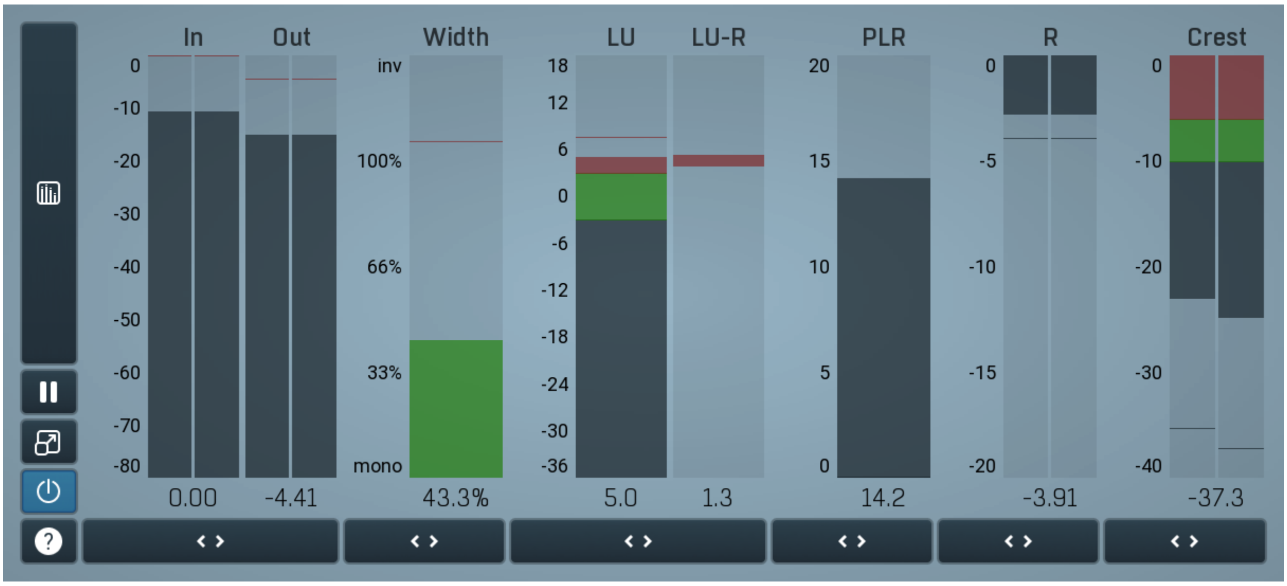
Saturation panel contains parameters of the saturation processor performed after the limiting.Saturation can smoothen the result adding some higher harmonics and generally increase the overall loudness even more, however it distorts the signal too.



Dry/Wet
Dry/Wet defines the ratio between dry and saturated signal.
Range: 0.00% to 100.0%, default 0.00%

Threshold
Threshold determines the minimal signal level above which the saturation starts to apply.
Range: silence to 0.00 dB, default silence

Mode
Mode defines the saturation shape. Generally the later modes are more "crunchy".



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.

LU meter shows the output loudness in EBU-18 scale. The loudness metering follows the ITU-R BS.1770-3 and EBU 3341 specifications. The metering units used are LU (Loudness Units) with 0 LU defined as -23 LUFS (LU Full Scale) and you should consider the LU values to be relative - using them to compare the loudness values between different signals. If the difference in loudness between 2 signals is 10 LU, it is approximately 10 dB as well.

Please note that you should still use your ears to judge loudness properly as there is still no accurate model of human loudness perception and every measurement is only an approximation. Loudness perception is also individual.

If you right click on the meter, additional settings will be displayed. Maximum value displays the maximum since the analysis started, rather than the recent maximum. Loudness pre-filtering uses EBU standard filters to simulate human perception. However, you may want to disable this to get more technical measurements.

There are 3 types of loudness measurements, all following the EBU specifications.
Momentary loudness uses an RMS sliding analysis window of 400 milliseconds; therefore it shows quick fluctuations in loudness.
Short-term loudness works in the same way, but uses a window of 3 seconds, therefore it provides more stable loudness measurements.
Integrated loudness shows the overall loudness, hence it is affected by the whole track from the beginning of the playback until you reset

it by clicking on the value field. The host may reset it too; it depends on your host.

Please note that the **Integrated loudness** is NOT the same as an averaged loudness, as it ignores quiet passages. Imagine a track which is generally quiet but has a few loud sections. The averaged loudness will be less than the Integrated loudness. Its calculation uses gating to ignore those quiet passages (levels less than 10 LU less than the current ungated level) of the track. Essentially, **Integrated loudness** is a measure of the loudest sections of the track.

LU-R meter shows the output loudness range in EBU-18 scale. The loudness meter, **LU**, follows the ITU-R BS.1770-3 and EBU 3342 specifications, so 0 LU (Loudness Units) represents -23 LUFS. The loudness range meter, **LU-R**, essentially describes how dynamic the signal is, measured over a sliding analysis window of 3 seconds. Too low a dynamic range usually means the signal is over-compressed. You should compare the dynamic range of your material with other mastered recordings in the same style.

Green bars indicate that the loudness or range is in the preferred regions, red bars suggest that the levels or range need attention.

PLR meter displays the true-peak-to-loudness ratio. It's similar to the crest meters, but based on the modern EBU loudness standard. It ranges from 0 to 20 and the higher the value, the less compressed (more dynamic) is the audio material. It tells you the difference between the maximum peak level and the **Integrated** loudness.

This is very useful when submitting audio tracks to services such as Spotify or iTunes, which normalize the loudness of the audio. Let's show this with an example: You are going to submit a song which has the true peak level of -1 dBTP and integrated loudness of -16 LUFS (which is the usual recommended loudness), so the PLR is -1 - (-16) = 15. This means that the song requires 15 dB of headroom above the normalization level to play without clipping. The song will play without clipping in iTunes (normalization level -16 LUFS, maximum true-peak level -1 dBTP, hence 15dB headroom), but the song might be turned down (by e.g. Tidal and YouTube), limited (by e.g. Spotify) or clipped when played on a platform with less headroom.

As a different example, you may be working on a cinematic score, with target loudness of -26 LU, and you might have used all the headroom for the requested maximum true peak level of -1 dBTP. The PLR is therefore 25. If you try to submit such a file to iTunes, it will normalize it to -16 LUFS, which means increasing the loudness, but your song requires 25 dB headroom and iTunes provides only 15, therefore it will either be rejected or processed in some way (clipped, limited...). Neither of these options are "good", therefore you will need to provide a different master for this platform.

Crest meter shows the output crest factor (calculated as RMS divided by peak level), which essentially indicates how extreme the peaks are in the output waveform. The lower the value is, the more peaks there are in the output, the more dynamic it is. If the value reaches 0dB, then the output is over-compressed and flattened and you should consider going easier on the compressor and limiter.

The range from 0dB to -6dB is red and you should prevent your master output from remaining in this range as that would mean it is extremely over-processed. The green range from -6dB to -10dB is the range most that recordings are in, usually jumping below the -10dB.

Please note that others calculate the crest factor as the peak level of the waveform divided by the RMS value of the waveform, and the higher the value, the more peaks there are in the output, the more dynamic it is.

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.

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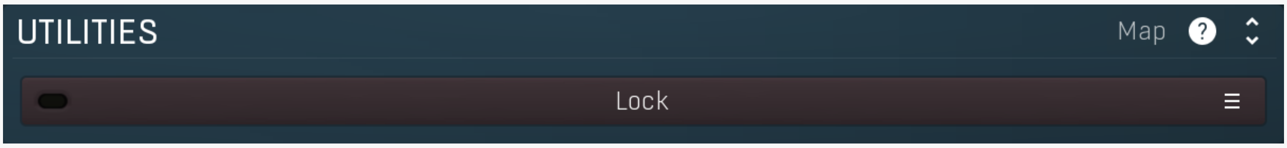
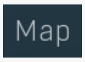
Utilities





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

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.