

MLimiterMB



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

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

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.



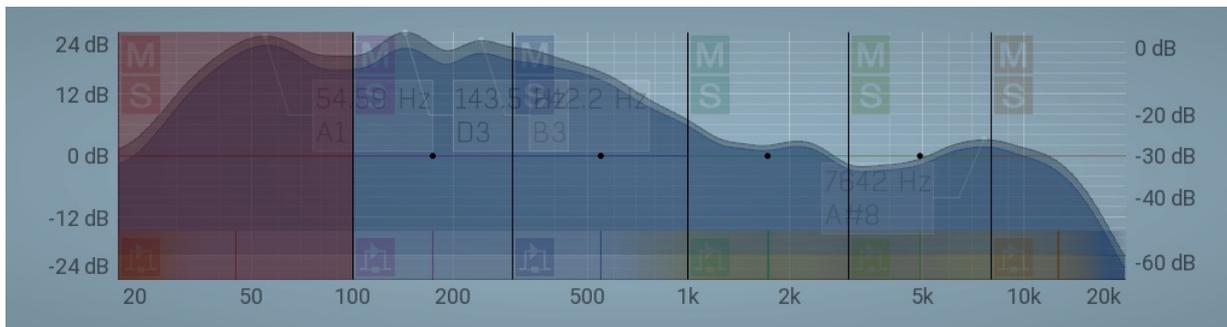
WWW

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.



Band editor

Band editor displays the available frequency bands, the crossover frequencies delimiting them, and the input gains and panoramic positions.

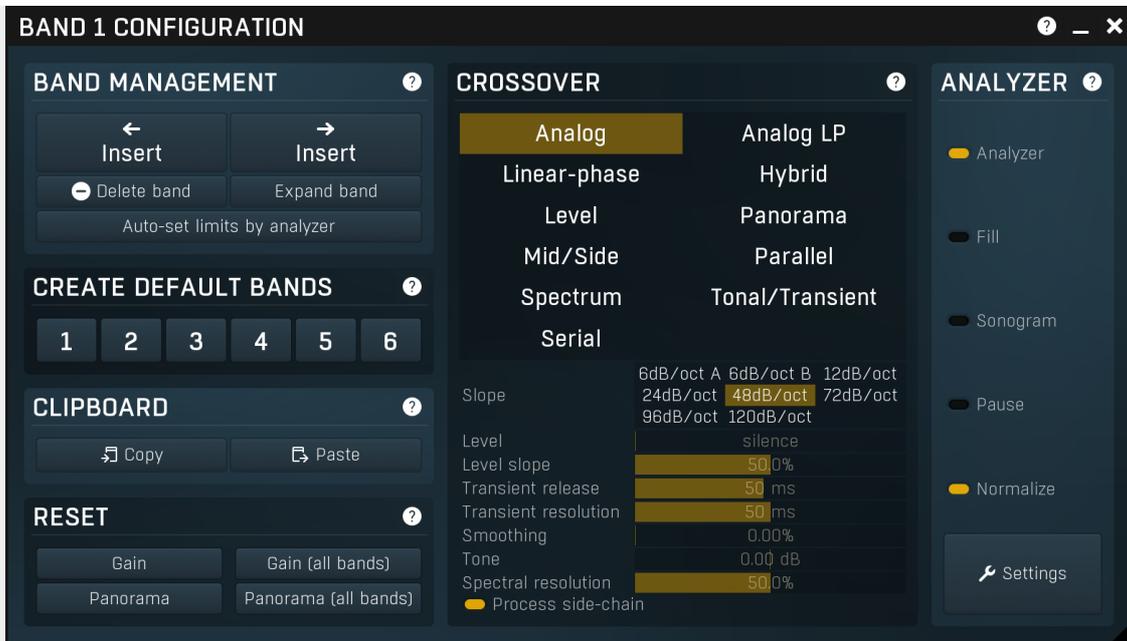
Use the left mouse button to drag the band boundaries (the vertical lines between bands), the band itself (the central dot in each band) and the input gains (the horizontal bars in each band). The short vertical bars in the bottom of each band control the input panoramic positions (when L+R Channel Mode is selected) or the input Widths (when M+S Channel Mode is selected).

Use the right mouse button to open the **Band Configuration** window where you can manage the bands and crossover filters and the appearance of the analyzer waveforms in the band editor.

Buttons to the left-hand side of each band let you mute, solo and bypass the processing in each band. Please note that the **Mute** and **Solo** buttons act on the output for each band, that is after the actual band processing.

The Collapse button to the right of the Band Editor minimises the editor, releasing space for other editors in the plug-in.

Band menu

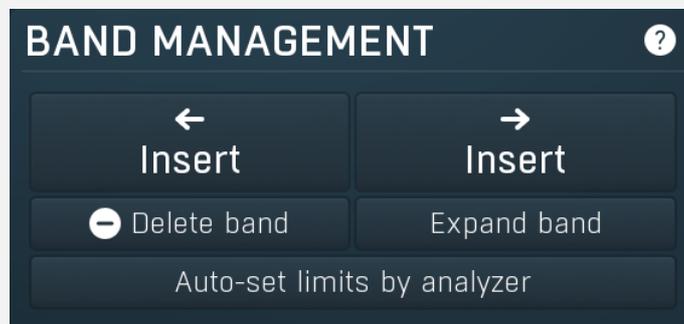


Band menu provides features to control the set of bands and copy & paste band settings (**Band management** section), reset band input gain & panorama (**Band gain & panorama** section), and to select and customize the crossover (**Crossover** section) and analyzer options.

You can display this menu by **right-clicking** on the band editor.

One of the essential things to control in the band menu is the number of bands. The plugin can either operate as a single bundle plugin. In this case there is no crossover employed of any kind and the first and only band receives all MIDI data if the plugin makes use of it somehow. If there are 2 or more bands however, the plugin somehow produces signals for each band using the crossover, based on the spectrum or level for example, and there's a change in MIDI behaviour as well - 1st band receives only MIDI channel 1, 2nd receives only MIDI channel 2 etc.

Band management panel

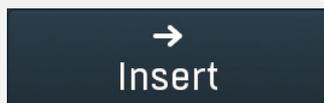


Band management panel contains basic features to create, delete and manipulate bands.



Insert left

Insert left button inserts a new band to the left of the currently-selected band (the last one that you clicked on).



Insert right

Insert right button inserts a new band to the right of the currently-selected band (the last one that you clicked on).



Delete

Delete button deletes the currently-selected band (the last one that you clicked on).



Expand band

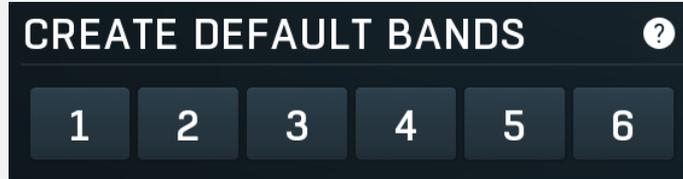
Expand band button solos (or unsolos) the band that you clicked on and disables the crossover temporarily, so that you can audition what the settings of this band would do to the entire signal, without any of the other bands having any affect.

Auto-set limits by analyzer

Auto-set limits by analyzer

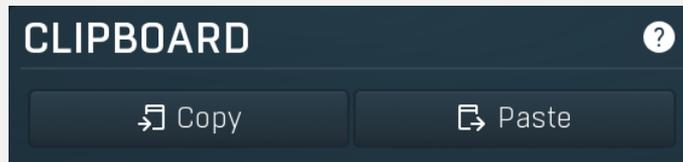
Auto-set limits by analyzer button adjusts the band limits using the current analyzer state, so that there's approximately the same signal level in each band. It is often useful to increase the averaging in the analyzer settings, so that the analysis doesn't 'jump' that quickly.

Create default bands panel



Create default bands panel lets you easily create a predefined set of bands. This is the easiest way to say create default plugin settings with 4 bands.

Clipboard panel



Clipboard panel contains features to transfer band settings via the system clipboard. Note that as always you can paste the settings as text into an email or forum post for example.

 Copy

Copy

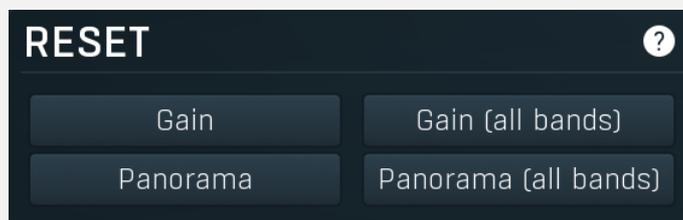
Copy button copies the band settings into the system clipboard. Note that the plugin band parameter settings are not copied; only the band limits, gains and panoramas.

 Paste

Paste

Paste button loads the band settings from the system clipboard. Note that the plugin band parameter settings are not pasted; only band limits, gains and panoramas.

Reset panel



Reset panel lets you reset some band parameters.

Gain

Gain

Gain button resets the input gain of the currently-selected band (the last one that you clicked on) to 0dB.

Gain (all bands)

Gain (all bands)

Gain (all bands) button resets the input gain of all bands to 0dB.

Panorama

Panorama

Panorama button resets the input panorama of the currently-selected band (the last one that you clicked on) to center.

Panorama (all bands)

Panorama (all bands)

Panorama (all bands) button resets the input panorama of all bands to center.

Crossover panel

CROSSOVER ?

Analog	Analog LP
Linear-phase	Hybrid
Level	Panorama
Mid/Side	Parallel
Spectrum	Tonal/Transient
Serial	

Slope: 6dB/oct A, 6dB/oct B, 12dB/oct, 24dB/oct, **48dB/oct**, 72dB/oct, 96dB/oct 120dB/oct

Level: silence

Level slope: 50.0%

Transient release: 50 ms

Transient resolution: 50 ms

Smoothing: 0.00%

Tone: 0.00 dB

Spectral resolution: 50.0%

Process side-chain

Crossover panel contains configuration of the crossover used to separate the signals for each band. **Crossover** is a technical term for an algorithm or device which splits a signal into multiple bands (or signals), which when mixed back together recreate the original signal (meaning that the crossover is transparent). The plugin provides several types of crossover with a flat (or nearly flat) response, which means that whichever crossover you choose and whatever signal you send into the plugin, the output levels of each frequency, after the bands are mixed back together to get the output signal, will be (almost) exactly the same, unless there is some processing applied in the bands themselves. Most of the available crossover types produce bands with different frequency ranges; however there are also a few more creative ones.

Analog crossovers have no latency, but they exhibit a phase-shift. That is usually irrelevant unless you are going to mix the output with the input later on. Analog crossovers are based on the classic analog components that you can find in speaker systems for example, however they are perfectly accurate and their slope (band separation) ranges from 6dB/octave to a very steep 120dB/octave. The higher the slope is, the more separated is each band (that is, there is less overlap between bands), but also the bigger is the phase shift. That can reach such an extent that some bassy materials become severely phasey, which may or may not be a good thing. An exception to the rule is the 6dB/oct crossover, which is zero-phase naturally. Its disadvantage is that the separation between bands is rather low, 6dB/oct is often not enough.

Analog LP crossover is a linear-phase equivalent to the **Analog** crossover. It introduces latency as does any linear-phase filter, but it does not cause a phase-shift. This may be especially advantageous for higher filter slopes, which, with classic analog crossovers, would cause severe transient smearing. Please note that the crossover type may not be 100% transparent, especially with small bands in bass spectrum and high slopes.

Linear-phase crossover is a fully digital crossover with a high slope (frequency-dependent), which introduces latency, but exhibits no phase-shift. This crossover mode is designed specifically for mastering.

Hybrid crossover is linear-phase as well, hence it introduces latency, but no phase-shift. However, its slope is more similar to the slopes of the analog crossovers.

Level crossover is a very specialized tool, which doesn't filter the input signal at all (hence it is not only linear-phase, but also zero-phase). Instead of filtering, it simply performs a gain on each band in such a way that when all the bands are mixed back

together, the output is the original signal again. When you select this crossover, the spectrum analyzer graph disappears and the X axis in the band editor changes from frequencies to dB levels. So the band limits are not frequencies anymore, but rather sound levels.

The current level displayed in the graph area is controlled by the **Level** value below and you are likely to use a modulator, most likely in **Follower** mode, to control this latter value. The crossover then applies gain to each band depending on how much the current level fits into the band. The **Slope** parameter controls how quickly each band fades into the adjacent one. This crossover effectively turns the plugin into a very advanced dynamics processor; using a Follower Modulator the band used to process the input audio depends on the audio level.

The are many possibilities for this crossover. But the basic principle is to select a spare Modulator, configure it as a Follower and select the Global parameter "Crossover Level value" as its target, with a "Full range" range mode. After configuring the Modulator, you will be able to see the detected value curve in the Modulator's Level graph. Then if the input signal is strongest, the right most band is processed etc. So if you for example use a delay with 2 bands and set the band limit high enough, the 2nd band will be processing only the loud parts of the signal and vice versa.

Panorama crossover is another specialized tool, similar to the level crossover; it splits the signal into bands according to the panorama. If, for example, you create 3 evenly spaced bands (100%L to 33%L, 33%L to 33%R, 33%R to 100%R), then the leftmost band will contain mainly the signals located in the left speaker, the rightmost band will contain mainly signals from the right speaker and the middle band will contain centred signals. Please note that this doesn't mean the crossover attempts to analyze the space the signals are coming from and send them to the respective bands, which is probably what your brain would attempt.

This crossover is useful only when processing stereophonic (or surround, in which case the channels from 3 upwards are kept intact) signals and can be used for all kinds of mixing and creative processing. For example, using a multiband compressor with this crossover can be used to effectively control the stereo image as each band would be processing a different part of the stereo image. To mention another example, a multiband delay or reverb can be used to produce a different ambience for different parts of the stereo image.

Mid/side crossover is similar to panorama crossover, but it splits the signal according to their position in mid/side location. In other words, the more to the left a band, the more centred is the signal in it. Similarly the more to the right a band, the more "to the side" is the signal in it. You can think of it as the panorama view folded back on itself, around the center position. If, for example, you create 3 evenly spaced bands (centre to 33% L or R, 33% L or R to 67% L or R, 67% L or R to 100% L or R), then the leftmost band will contain the centred signal, the rightmost band will contain the signals to the extreme left or right and the middle band will contain signals in between. It can be used for similar tasks as the panorama crossover.

Parallel crossover is not a crossover actually, it simply disables the crossover and as a result each band processes the full input signal. In practice this "not really crossover" mode lets you process multiple streams of the input audio signal in parallel. As a consequence there is likely to be an increase in output level, so take care and turn down the output level first. For example, if you use a compressor, this in effect produces an extreme parallel compression. As another example, you can use a reverb to produce several rooms in parallel, potentially leading to a fuller space for example.

Spectrum crossover is the first of the spectral crossovers. It splits the signal into individual frequencies, analyzes their levels and sends the frequencies with the highest level into the highest band etc. It marks each frequency with its level (as you can see on the dB scale on the X axis in the crossover band editor) and puts it into the appropriate band. The crossover is linear-phase and fully transparent.

It provides a huge (not only) creative potential as it lets you process the dominant and weak parts of the signal individually. For instance, by compressing the dominant frequencies using MDynamicsMB you can bring more attention to the unsubstantial frequencies in the signal and in a way stabilize it without disrupting the silent parts of it. Note that this is NOT the same thing as using a normal compressor, because this way it treats only the loud frequencies even if the weak frequencies are present at the same time. Another example could be using MDelayMB to generate echoes only to the dominant parts of the signal, such as snare and bass drums in a drum loop.

Transient crossover is also a spectral crossover. It splits the signal into individual frequencies and sends the transient parts for each of them into the highest band etc. It marks each frequency with its "current transientness" (defined by the percentage scale that you can see on the X axis in the crossover band editor) and puts it into the appropriate band. The crossover is linear-phase and fully transparent.

It provides a huge (not only) creative potential as it lets you process split the signal into tonal and transient parts (and anything in between) and treat each individually. For instance, by compressing the transients using MDynamicsMB you can easily control the attack of drums. Note that this is NOT the same thing as using a normal compressor, because this way you can treat only the attacks in an already mixed signal without affecting the remaining part of the signal. Another example could be using MDelayMB to generate echoes only for the attacks of each drum.

Serial crossover is not a crossover actually, it simply disables the crossover and processes all bands in series. For instance a multiband compressor can be exploited to perform multiple compressions in series, which is often considered better sounding compared to a single compressor driven hard. Please note that if each band has a latency, the latencies will add up.

	6dB/oct A	6dB/oct B	12dB/oct	
Slope	24dB/oct	48dB/oct	72dB/oct	
	96dB/oct	120dB/oct		Slope

Slope defines the slope of each band transition and is used only by analog crossovers (including the linear-phase versions). It essentially controls the separation between the bands - the higher the slope, the lower the overlap between bands. Higher slopes require more CPU power and exhibit higher phase shift, which may be a problem especially when percussive materials. In these cases it may be necessary to switch to a linear-phase version.

Interesting exception to the classic rule are the 6dB/oct crossovers, which are linear-phase by nature (while still being zero latency), because the bands compensate for each other's phase shift. A side-effect of this is that the signal level in each band is much higher than using other crossovers, so you may expect these crossovers sound considerably different to the other modes.

Level silence **Level value**

Level value is used only with **Level crossover** and controls the level at which the signal is split into each band. You will probably want to attach this parameter to a modulator in Follower mode for instance.

Level slope 50.0% **Level slope**

Level slope is used only with some crossover modes (Level, Spectrum and Tonal/Transient) and controls how quickly each band fades into the next one. It's similar to the **Slope** parameter used with analog crossovers.

Transient release 50 ms **Transient release**

Transient release is only used by the **Tonal/Transient** crossover and controls the release time of each transient. The transients detected by the crossover are naturally very short, so this provides a way to make them longer, hence send more signal to the higher bands of the crossover (receiving transients) and less to the lower bands (receiving the remaining part of the signal).

Transient resolution 50 ms **Transient resolution**

Transient resolution is only used by the **Tonal/Transient** crossover and controls the behaviour of the spectral transient detector. You can use it to adjust the crossover to your audio material and we would recommend a simple trial-and-error approach.

Smoothing 0.00% **Smoothing**

Smoothing is only used by spectral crossovers and controls how frequencies affect their surroundings. Without smoothing the individual bands may sound a bit artificial, because human brain generally dislikes separated frequencies. It usually doesn't matter unless you audition the bands separately, but sometimes when more "brutal" processing is used on each band, it may become audible, which is where the smoothing can provide a solution at the cost of additional CPU and lower separation between bands, because it naturally makes the frequencies "more alike".

Tone 0.00 dB **Tone**

Tone is only used by spectral crossovers and controls the spectral slope applied by the detector. It is exactly the same feature as the **Slope** in analyzers and the crossover uses it to determine how to spread the frequencies between the bands. Higher slope gives more energy to higher frequencies and vice versa. Note that whatever the settings are, the crossover still produces signals that perfectly sum to the original input signal, meaning that it is perfectly transparent and unless the bands are actually doing something, you won't be able to hear a difference when changing this parameter.

Spectral resolution 50.0% **Spectral resolution**

Spectral resolution is only used by spectral crossovers and controls the spectral transformation settings. The higher the value is, the higher FFT size and overlap size is used, and therefore more CPU is usually required as well. Whether higher/lower value is good or not depends on the actual signal, the default 50% should work well with most audio materials. Higher values will generally provide better frequency resolution (usually good for less percussive sounds), lower values will provide better time resolution (usually good for more percussive sounds), eventually it is always about a compromise.

Process side-chain

Process side-chain option makes sure the side-chain is processed by the crossover as well as the main input. If you disable this option, main input will be processed of course, but side-chain will not. This may be handy e.g. in a multiband dynamics processor, which should react to the entire signal, but process each bands individually.

Analyzer panel

ANALYZER

Analyzer

Fill

Sonogram

Pause

Normalize

 Settings

Analyzer panel lets you configure the fully featured integrated analyzer and sonogram.

 Settings

Settings

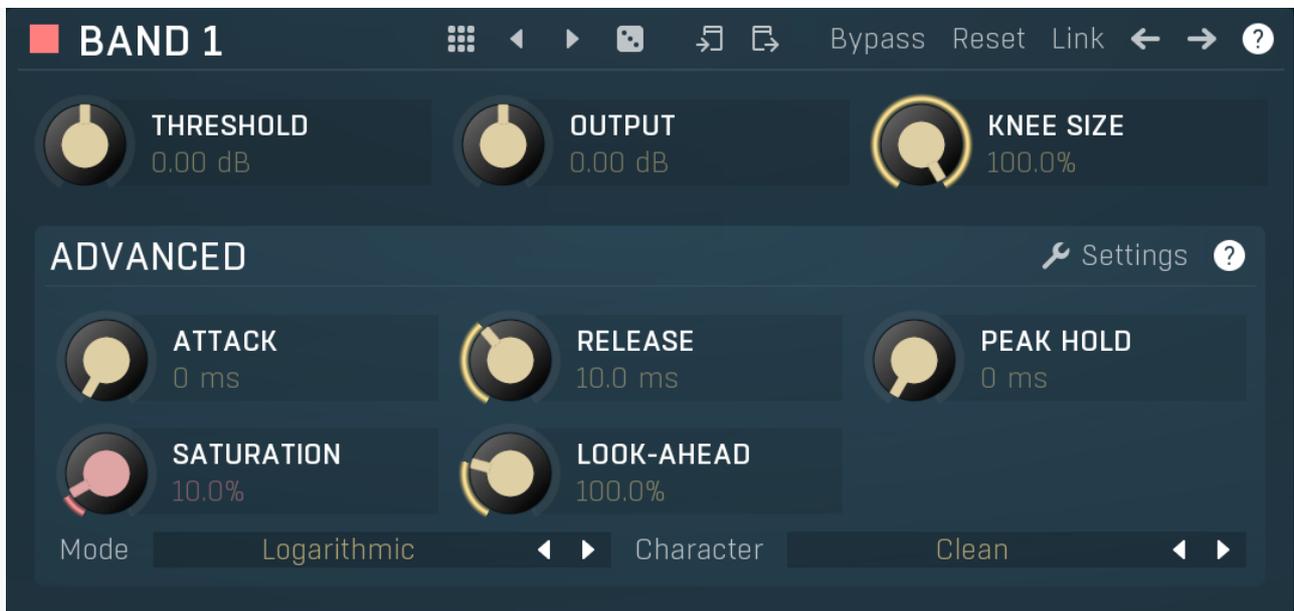
Settings button shows the settings of the spectrum analyzer and the spectrum sonogram.



Collapse

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

Band panel



Band panel contains parameters of a particular band. You can select a band using the band editor above, just click on the band in the graph. Processing is performed on separate bands first and the master (the combined audio from each band) is processed afterwards.



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.

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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.



Reset

Reset button loads the default settings.

Link **Link**

Link button enables parameter linking between bands. Every parameter change performed with this enabled changes that parameter in all bands. Please note that some more rare parameters, which are not available for assignment and automation, may not be changed. But **Pasting** settings from the system clipboard does not change the other bands.

← **Left**

Left button selects the previous band. If this is the first band, it selects the last one instead. This way you can easily cycle between the bands if selecting them in the band editor is hard because they are modulated for example.

→ **Right**

Right button selects the next band. If this is the last band, it selects the first one instead. This way you can easily cycle between the bands if selecting them in the band editor is hard because they are modulated for example.



Threshold

Threshold determines the relative threshold of the band. Increase the value to limit the band less and vice versa. Please note that the processor performs additional processing and the real threshold value may be different.

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



Output gain

Output gain defines the gain applied to the output 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



Knee size

Knee size defines size of the knee.

Range: 0.00% to 100.0%, default 100.0%

Advanced panel

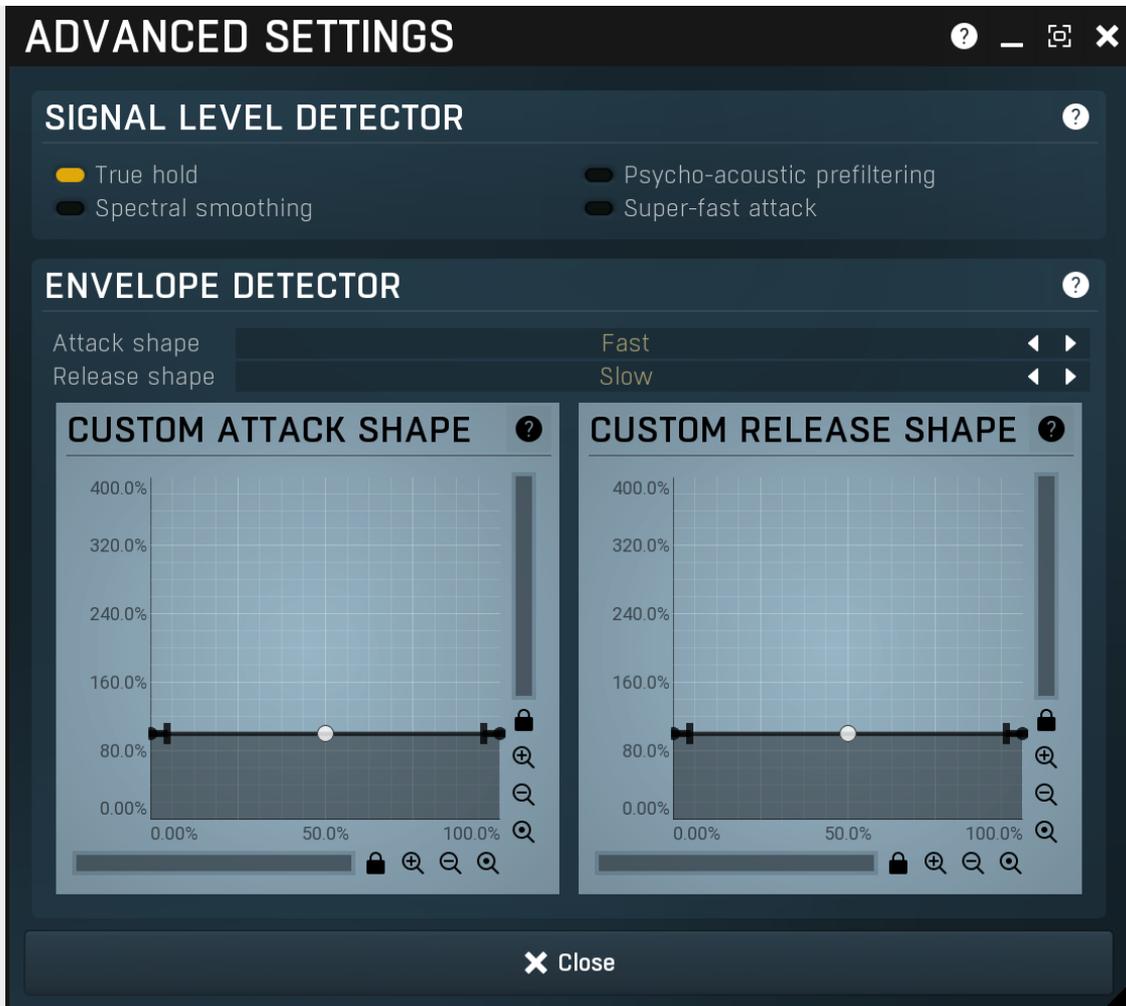


Advanced panel contains more advanced parameters defining how the plug-in determines the level of the source signal, dynamic shape transformation etc.

⚙️ Settings **Settings**

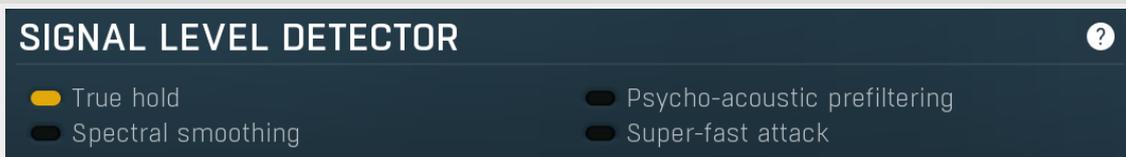
Settings button shows additional dynamics detector settings.

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

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

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

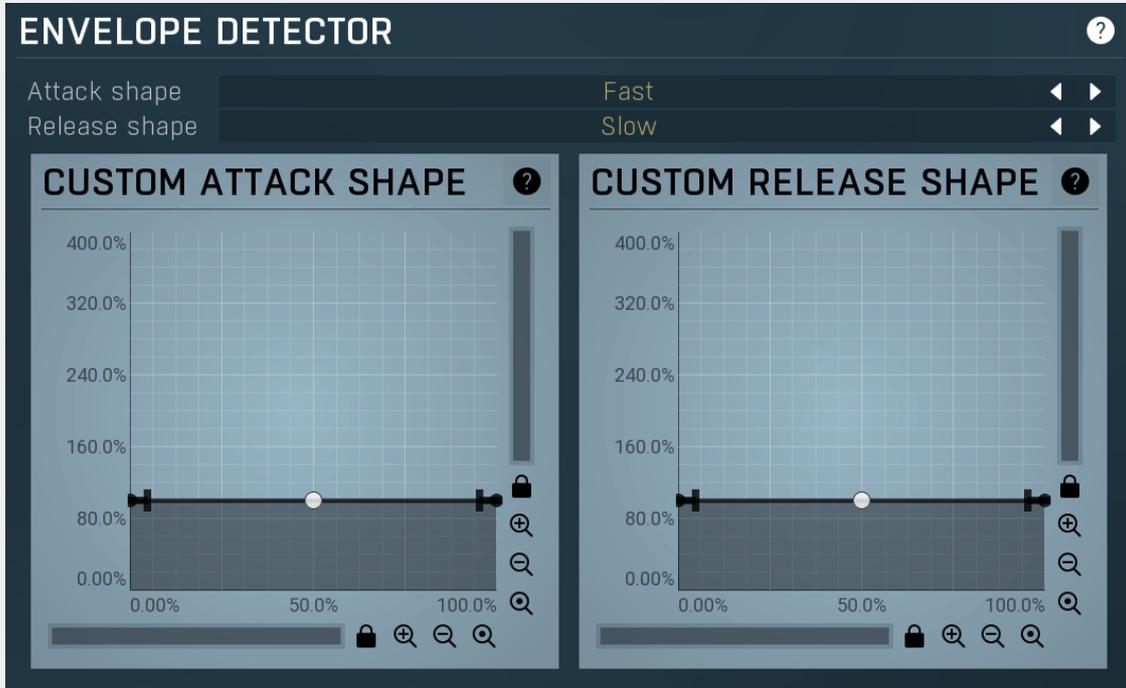
with low attack values. This feature attempts to make the signal smoother by applying a complex filtering, which does not change the frequency levels. By doing so, you may expect a slower detector response. Limiters need to be extremely quick, hence it is not appropriate for them.

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.

Envelope detector



Attack shape

Fast

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

Slow

Release

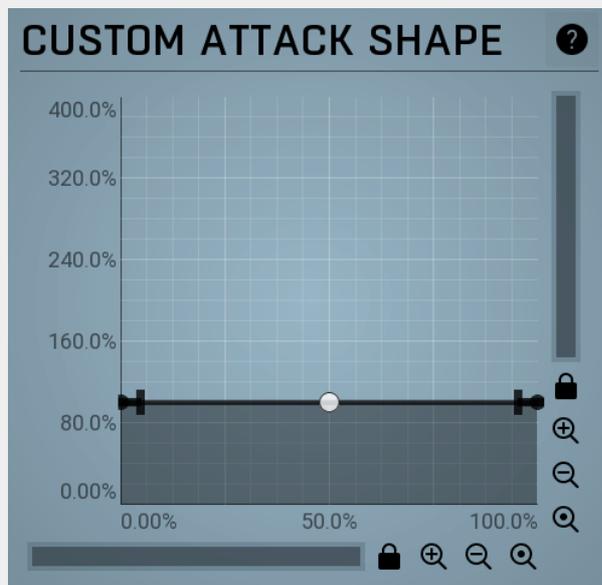
shape

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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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).

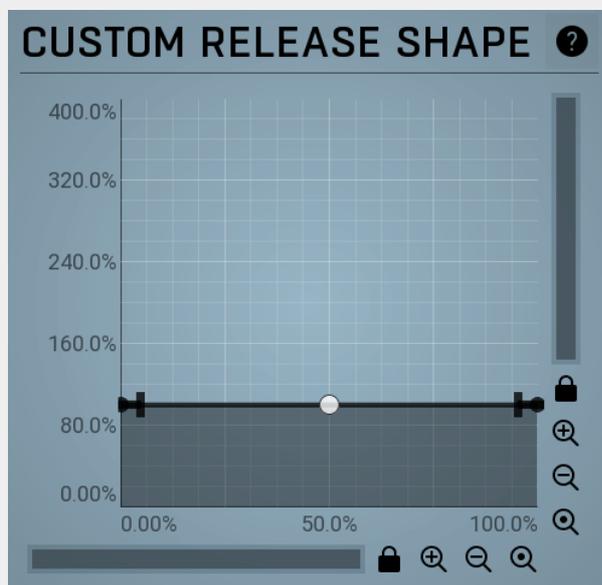


Custom attack shape

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.
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- **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 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



Saturation

Saturation controls the amount of saturation performed after the limiting. Saturation can smooth out the result adding some higher harmonics and can generally increase the overall loudness even more, however it distorts the signal too.

Range: 0.00% to 100.0%, default 0.00%



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%

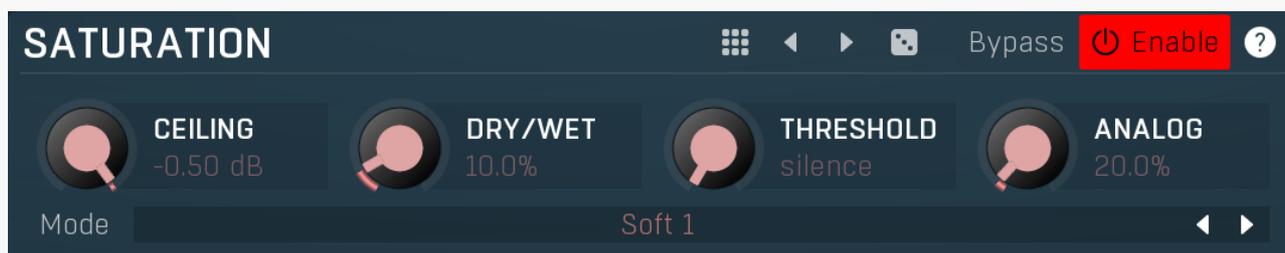
Mode Logarithmic ◀ ▶ Mode

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 Clean ◀ ▶ 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 master effect that is processed after the individual bands are processed and the signal is mixed back together.



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:

- Using "Backup" and "Restore" buttons in each preset window, which produces a single archive of all presets on the computer.
- Using "Export/Import" buttons, which export a single folder of presets for one plugin.
- 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.



Enable

Enable button enables or disables the effect.



CEILING
-0.50 dB

Ceiling

Ceiling defines the gain applied to the output after the saturation, hence this is the maximal level that the output can reach.

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



DRY/WET
10.0%

Dry/Wet

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

Range: 0.00% to 100.0%, default 20.0%



THRESHOLD
silence

Threshold

Threshold determines the minimal signal level above which the effect starts to apply. By lowering the threshold you increase loudness and also distortion.

Range: silence to 0.00 dB, default silence



ANALOG
20.0%

Analog

Analog determines the amount of even harmonics added to the signal in addition to the main saturation, which is typical for analogue saturation. The amount of the harmonics is dependent on the saturation signal, unlike the full harmonic control in the **Harmonics** panel, which is completely independent of the actual saturation processing.

Range: 0.00% to 500.0%, default 50.0%

Mode Soft 1 ◀ ▶ **Mode**

Mode defines the limiting shape.

Global parameters panel

Input	0.00 dB
Output	0.00 dB
Temporary gain	0.00 dB
Slope	0.00 dB
Mode	Manual

Global parameters panel contains global controls, which are usually relevant to global processing performed either before the signal reaches the crossover and gets split into bands, or after the signals are processed and summed back to the master signal.

Input 0.00 dB

Input gain

Input gain defines the power modification applied to the incoming signal, before it is split into bands.

Output 0.00 dB

Output gain

Output gain defines the gain applied to the output signal, right after it is summed from the bands, but before the signal is passed into the saturator. By increasing this value you can get more saturation, hence the sound will be louder but still won't get above the 0dB limit.

Temporary gain 0.00 dB

Temp gain

Temp gain defines a temporary power modification 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**.

Slope 0.00 dB

Slope

Slope defines the balance between the band thresholds therefore it basically defines the resulting sound character. A negative value provides more energy for bass frequencies, which is often desirable.

Mode Manual ◀ ▶

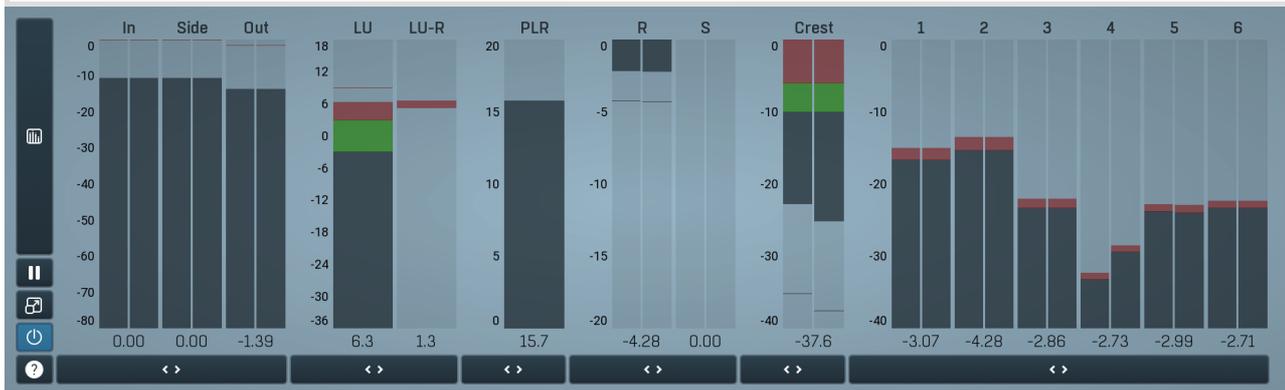
Mode

Mode defines how the limiter responds to changes in band thresholds and the resulting gain reduction. **Input gain** is increased to a point where loudness is at a high enough level, then the **Threshold** of each band is used to ensure none of the bands are over processed. During this processing some dynamics are lost and there is a change in gain. The Mode parameter defines how the limiter reacts to this change.

Manual mode is often found on limiters and means that no automatic compensation for the gain reduction is performed. In manual mode the **Output gain** can be used to bring the output up to 0dB. The disadvantage is that by lowering a band threshold the output level is decreased, so the output gain has to be used again to perform manual compensation, otherwise not all of the dynamic range will be used. Additionally, increasing all thresholds can easily push the limiter past the 0dB level forcing the saturator that follows it, to overdrive the signal.

Automatic output gain compensation makes the limiter adjust the output gain automatically according to the current thresholds set for each band. When the threshold of a band is changed, the limiter will ensure that the output will still reach exactly 0dB plus the specified **Output gain**, which is then useful to provide some additional drive before the saturation is performed. The advantage of this is that once the output gain is set, the difference in the output level should be minimized when the thresholds are changed. This allows for the use of the band thresholds to control the output loudness.

Automatic threshold compensation causes the thresholds to be changed in relation to the others. For example, increasing one threshold causes the others to decrease. As a result a stable output level can be expected and the thresholds are used to control the amount of gain reduction in all bands. This mode is designed to minimize any loss of dynamics, however it is probably harder to use.



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.

Numbered band meters display the input levels for each band, but also indicate the gain reduction by the colored top part. If the top part is red, it is indicating a gain reduction. If it is green, the band is actually increasing the gain, which usually means expansion.

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.

S meter shows the saturation level for each channel. This basically shows how much signal over 0dB has passed through the limiting section. A limiter can remove all of the peaks above 0dB only if it is in single-band mode and attack is set to 0ms or using the look-ahead + true hold trick. In all other cases, there can and will be peaks above 0dB passing through the main dynamics unit. These peaks are then saturated or clipped and are indicated using this saturation meter. Saturation provides a crisper signal, in higher levels it can cause audible distortion, which may be desired as it brightens the signal. Less saturation provides a cleaner signal. In higher levels you may hear severe pumping however.

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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Collapse

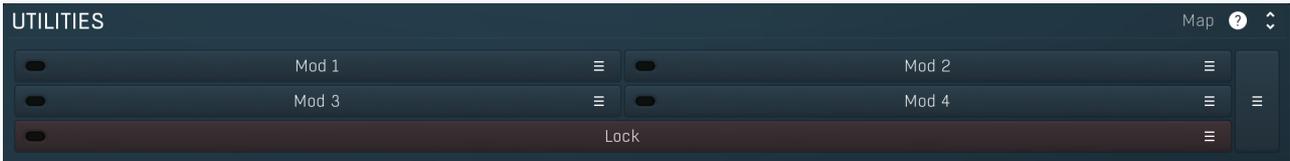
Collapse button minimizes or enlarges the panel to release space for other editors.



Collapse

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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

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1 : Drive

0.00 dB



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

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