MXXX

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.

 Randomize

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

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

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

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

Panic

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

Settings

Settings 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.
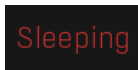
Sleep indicator

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.
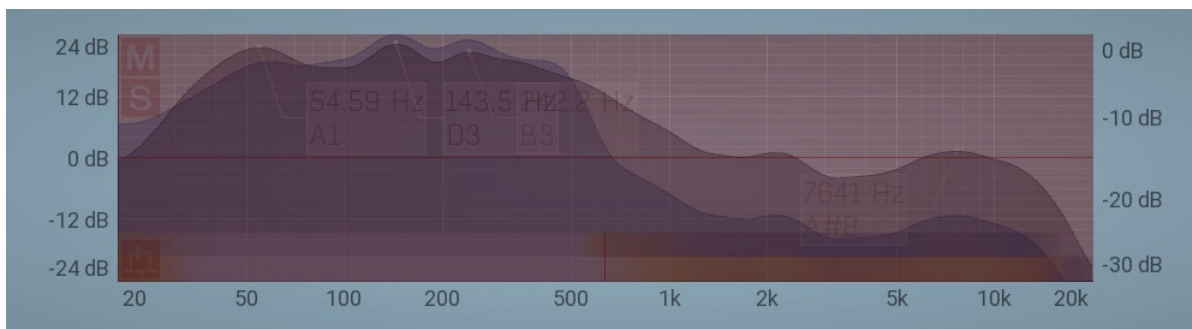
Band editor

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.
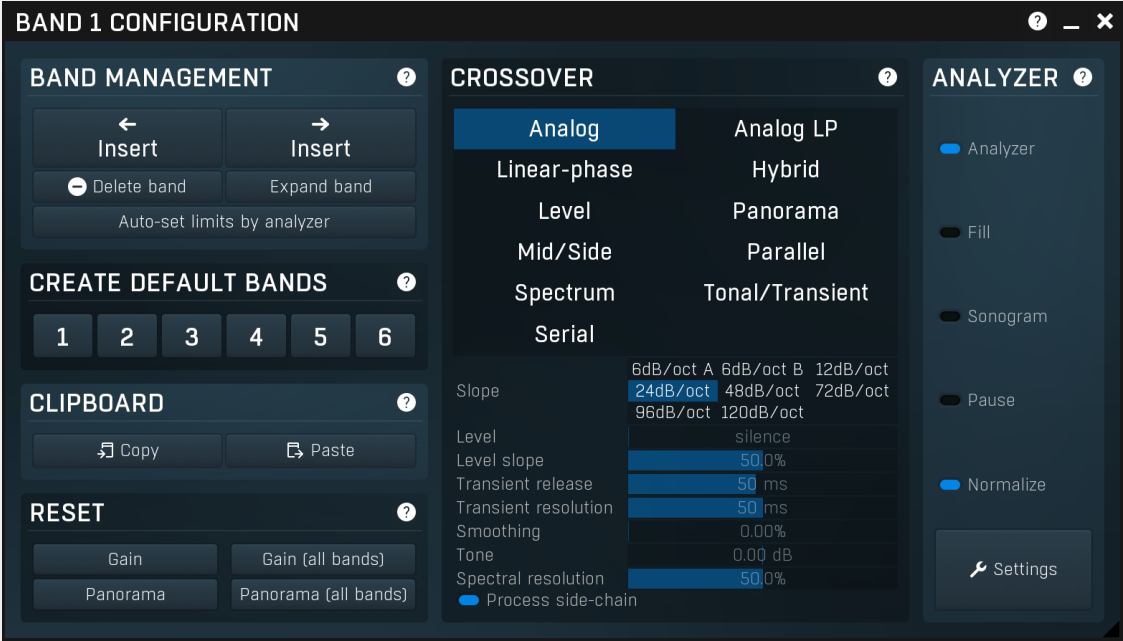

Band menu provides features to control the set of bands and copy & paste band settings (**Band management** section), reset band input gain & panorama (**Band gain & panorama** section), and to select and customize the crossover (**Crossover** section) and analyzer options.

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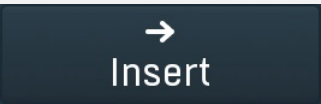
Band management panel
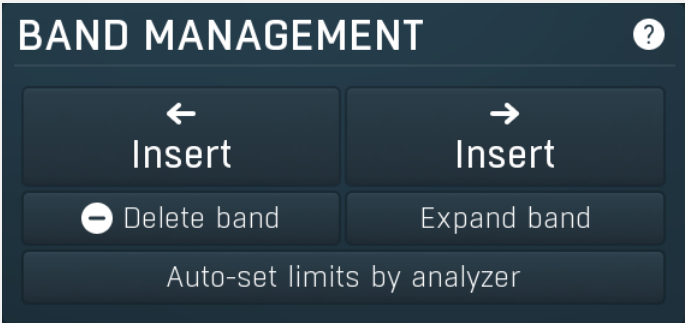
Band management panel contains basic features to create, delete and manipulate bands.

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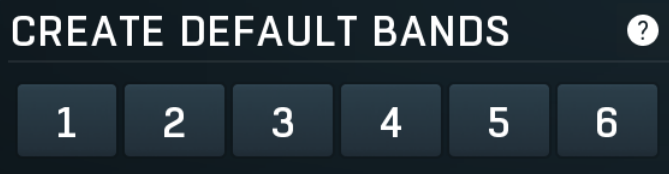
Create default bands panel lets you easily create a predefined set of bands. This is the easiest way to say create default plugin settings with 4 bands.

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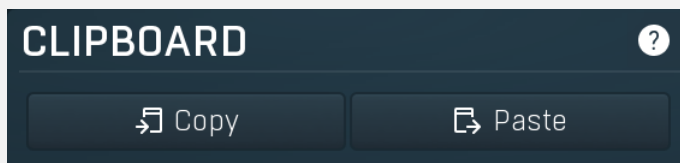
Clipboard panel contains features to transfer band settings via the system clipboard. Note that as always you can paste the settings as text into an email or forum post for example.
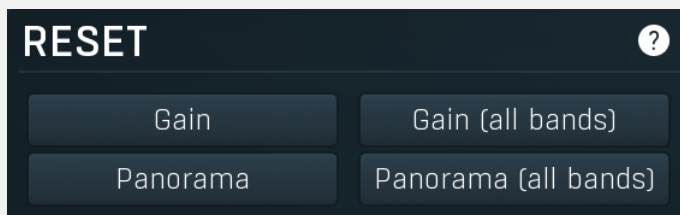
Copy

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

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

Crossover panel

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.

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 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

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

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

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

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 general 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

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 tranparent and unless the bands are actually doing something, you won't be able to hear a difference when changing this parameter.

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.
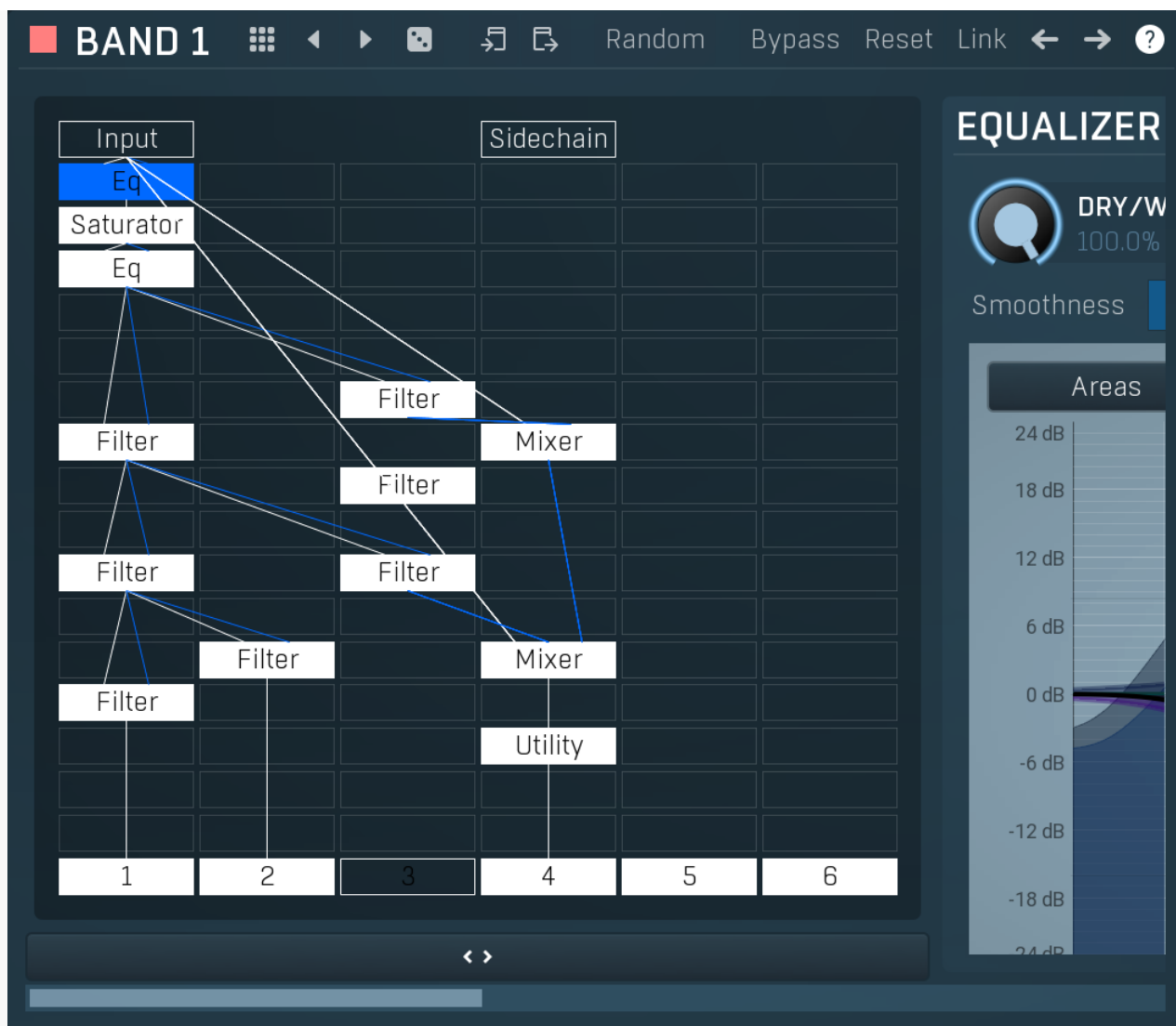
Settings

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.

⠿ Presets

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Randomize

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Copy

Copy button copies the settings onto the system clipboard.

Paste

Paste button loads the settings from the system clipboard.

Randomize

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

Reset

Reset button loads the default settings.

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.

Modular editor
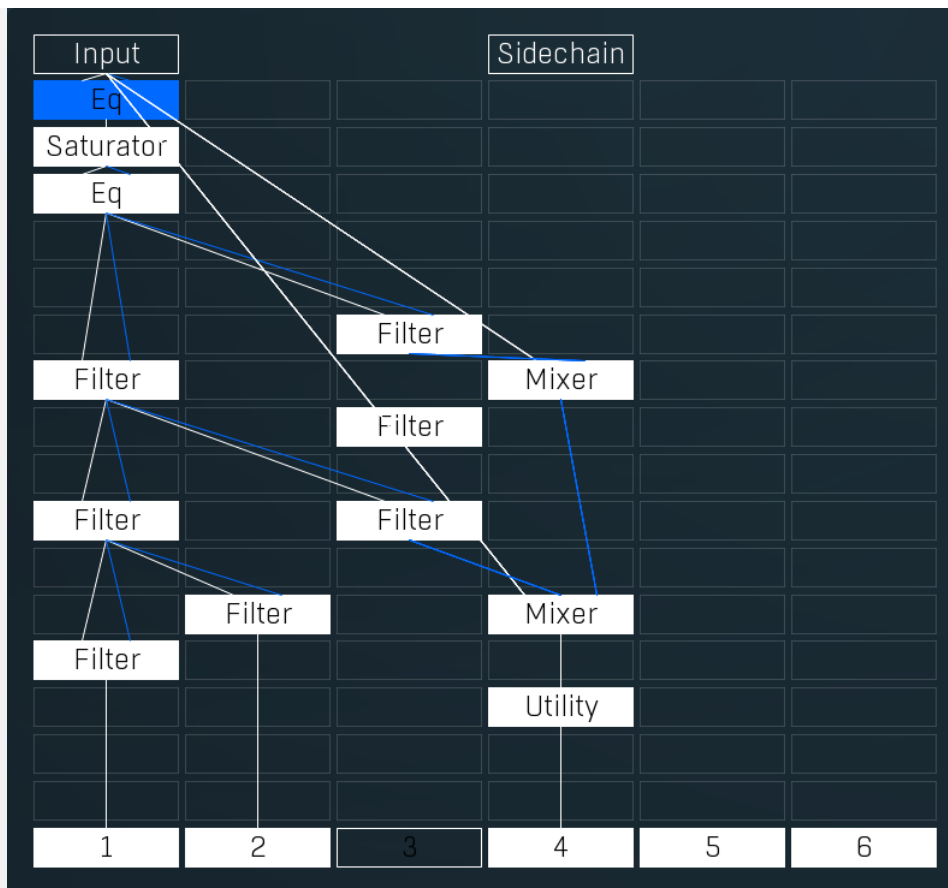
Modular editor lets you edit the processing flow. There is at least one input at the top, several outputs at the bottom and several processing boxes in-between. Processing is always performed from top to bottom and is indicated by a solid line. All outputs except the first one are disabled by default, unless you put a plugin into its chain. Every output can be disabled by clicking on it.

Each processing box can contain one plugin. By default all plugins are taking their inputs directly from above (thus from the same chain that they are in), but if you for example put a plugin in the first row of the second chain, there is no input above, so it is redirected to the first input. Some plugins may have multiple input channels or even a variable number of input channels. Most plugins have just one output channel, in which case they place the output into the same chain that they are in. Some plugins have variable output channels in which case they send them to the same channel, the next one to the right and so on.

If you **click** on an empty box a menu with all available processing plugins will be shown. If you click on a box which already contains a plugin, it will be selected and its editor shown to the right. The currently-selected plugin is highlighted.
If you hold **Ctrl** and click on the plugin or click using **right mouse button**, the advanced settings window will be displayed. You can use it to configure the plugin input and output channels and much more.
Double-click on the plugin to enable or disable it.
Hold **Alt** and click on the plugin to delete it.
Hold **Shift** and click on the plugin to show its editor in a pop-up window.

Collapse

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

Equalizer 1

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.

Copy

Copy button copies the settings onto the system clipboard.

Paste

Paste button loads the settings from the system clipboard.

Channel mode

Channel mode button shows the current processing channel mode, e.g. **Left+Right (L+R)** indicates the processing of left and right channels. This is the default mode for mono and stereo audio material and effectively processes the incoming signal as expected. However the plugin also provides additional modes, of which you may take advantage as described below. Mastering this feature will give you unbelievable options for controlling the stereo field.

Note that this is not relevant for mono audio tracks, because the host supplies only one input and output channel.

Left (L) mode and Right (R) mode allow the plugin to process just one channel, only the left or only the right. This feature has a number of simple uses. Equalizing only one channel allows you to fix spectral inconsistencies, when mids are lower in one channel for example. A kind of stereo expander can be produced by equalizing each side differently. Stereo expansion could also be produced by using a modulation effect, such as a vibrato or flanger, on one of these channels. Note however that the results would not be fully mono compatible.

Left and right channels can be processed separately with different settings, by creating two instances of the plugin in series, one set to 'L' mode and the other to 'R' mode. The instance in 'L' mode will not touch the right channel and vice versa. This approach is perfectly safe and is even advantageous, as both sides can be configured completely independently with both settings visible next to each other.

Mid (M) mode allows the plugin to process the so-called mid (or mono) signal. Any stereo signal can be transformed from left and right, to mid and side, and back again, with minimal CPU usage and no loss of audio quality. The mid channel contains the mono sum (or centre), which is the signal present in both left and right channels (in phase). The side channel contains the difference between the left and right channels, which is the "stereo" part. In 'M mode' the plugin performs the conversion into mid and side channels, processes mid, leaves side intact and converts the results back into the left and right channels expected by the host.

To understand what a mid signal is, consider using a simple gain feature, available in many plugins. Setting the plugin to M mode and decreasing gain, will actually lower or attenuate the mono content and the signal will appear "wider". There must be some stereo content present, this will not work for monophonic audio material placed in stereo tracks of course. Similarly amplifying the mono content by increasing the gain, will make the mono content dominant and the stereo image will become "narrower".

As well as a simple gain control there are various creative uses for this channel mode.
Using a **compressor** on the mid channel can widen the stereo image, because in louder parts the mid part gets attenuated and the stereo becomes more prominent. This is a good trick to make the listener focus on an instrument whenever it is louder, because a wider stereo image makes the listener feel that the origin of the sound is closer to, or even around them.
A **reverb** on the mid part makes the room appear thin and distant. It is a good way to make the track wide due to the existing stereo content, yet spacey and centered at the same time. Note that since this effect does not occur naturally, the result may sound artificial on its own, however it may help you fit a dominant track into a mix.
An **equalizer** gives many possibilities - for example, the removal of frequencies that are colliding with those on another track. By processing only the mid channel you can keep the problematic frequencies in the stereo channel. This way it is possible to actually fit both tracks into the same part of the spectrum - one occupying the mid (centre) part of the signal, physically appearing further away from the listener, the other occupying the side part of the signal, appearing closer to the listener.
Using various **modulation effects** can vary the mid signal, to make the stereo signal less correlated. This creates a wider stereo image and makes the audio appear closer to the listener.

Side (S) mode is complementary to M mode, and allows processing of only the side (stereo) part of the signal leaving the mid intact. The same techniques as described for M mode can also be applied here, giving the opposite results.
Using a **gain** control with positive gain will increase the width of the stereo image.
A **compressor** can attenuate the side part in louder sections making it more monophonic and centered, placing the origin a little further away and in front of the listener.
A **reverb** may extend the stereo width and provide some natural space without affecting the mid content. This creates an interesting side-effect - the reverb gets completely cancelled out when played on a monophonic device (on a mono radio for example). With stereo processing you have much more space to place different sounds in the mix. However when the audio is played on a monophonic system it becomes too crowded, because what was originally in two channels is now in just one and mono has a very limited capability for 2D placement. Therefore getting rid of the reverb in mono may be advantageous, because it frees some space for other instruments.
An **equalizer** can amplify some frequencies in the stereo content making them more apparent and since they psycho acoustically become closer to the listener, the listener will be focused on them. Conversely, frequencies can be removed to free space for other instruments in stereo.
A **saturator / exciter** may make the stereo richer and more appealing by creating higher harmonics without affecting the mid channel, which could otherwise become crowded.
Modulation effects can achieve the same results as in mid mode, but this will vary a lot depending on the effect and the audio material. It can be used in a wide variety of creative ways.

Mid+Side (M+S) lets the plugin process both mid and side channels together using the same settings. In many cases there is no difference to L+R mode, but there are exceptions.
A **reverb** applied in M+S mode will result in minimal changes to the width of the stereo field (unless it is true-stereo, in which case mid will affect side and vice versa), it can be used therefore, to add depth without altering the width.
A **compressor** in M+S mode can be a little harder to understand. It basically stabilizes the levels of the mid and side channels. When channel linking is disabled in the compressor, you can expect some variations in the sound field, because the compressor will attenuate the louder channel (usually the mid), changing the stereo width depending on the audio level. When channel linking is enabled, a compressor will usually react similarly to the L+R channel mode.
Exciters or saturators are both nonlinear processors, their outputs depend on the level of the input, so the dominant channel (usually mid) will be saturated more. This will usually make the stereo image slightly thinner and can be used as a creative effect.

How to modify mid and side with different settings? The answer is the same as for the L and R channels. Use two instances of the plugin one after another, one in M mode, the other in S mode. The instance in M mode will not change the side channel and vice versa.

Left+Right(neg) (L+R-) mode is the same as L+R mode, but the the right channel's phase will be inverted. This may come in handy if the L and R channels seem out of phase. When used on a normal track, it will force the channels out of phase. This may sound like an extreme stereo expansion, but is usually extremely fatiguing on the ears. It is also not mono compatible - on a mono device the track will probably become almost silent. Therefore be advised to use this only if the channels are actually out of phase

or if you have some creative intent.

There are also 4 subsidiary modes: **Left & zero Right (L(R0))**, **Right & zero Left (R(L0))**, **Mid & zero Side (M(S0))** and **Side & zero Mid (S(M0))**. Each of these processes one channel and silences the other.

Surround mode is not related to stereo processing but lets the plugin process up to 8 channels, depending on how many the host supplies. For VST2 plugins you have to first activate surround processing using the **Activate surround** item in the bottom. This is a global switch for all MeldaProduction plugins, which configures them to report 8in-8out capabilities to the host, on loading. It is disabled by default, because some hosts have trouble dealing with such plugins. After activation, restart your host to start using the surround capabilities of the plugins. Deactivation is done in the same way. Please note that all input and output busses will be multi-channel, that includes side-chain for example. For VST3/AU/AAX plugins the activation is not necessary.

First place the plugin on a surround track - a track that has more than 2 channels. Then select **Surround** from the plug-in's Channel Mode menu. The plugins will regard this mode as a natural extension of 2 channel processing. For example, a compressor will process each channel separately or measure the level by combining the levels of all of the inputs provided. Further surround processing properties, to enable/disable each channel or adjust its level, can be accessed via the **Surround settings** in the menu.

Ambisonics mode provides support for the modern 3D systems (mostly cinema and VR) with up to 64 channels (ambisonics 7th order). Support for this is still quite rare among the DAWs, so this needs to be activated in all DAWs using the **Activate ambisonics** item in the bottom. This is a global switch for all MeldaProduction plugins, which configures them to report 64in-64out capabilities to the host, on loading. After activation, restart your host to start using the ambisonics capabilities of the plugins. Deactivation is done in the same way. Please note that all input and output busses will be multi-channel, that includes side-chain for example.

First place the plugin on an ambisonics track, supported are all orders from 1st (4 channels) to 7th (64 channels). Then select **Ambisonics** from the plug-in's Channel Mode menu. Finally select the **Ambisonics settings** in the menu and configure the Ambisonics order and other settings if needed. The plugins will regard this mode as a natural extension of 2 channel processing. For example, a compressor will process each channel separately or measure the level by combining the levels of all of the inputs provided.

A Oversampling

Oversampling can potentially improve sound quality by processing at a higher sample rate. Processors such as compressors, saturators, distortions etc., which employ nonlinear processing generate higher harmonics of the existing frequencies. If these frequencies exceed the Nyquist rate, which equals half of the sampling rate, they get mirrored back under the Nyquist rate. This is known as aliasing and is almost always considered an artifact. This is because the mirrored frequencies are no longer harmonic and sound as digital noise as this effect does not physically occur in nature. Oversampling reduces the problem by temporarily increasing the sampling rate. This moves the Nyquist frequency which in turn, diminishes the level of the aliased harmonics. Note that the point of oversampling is not to remove harmonics, we usually add them intentionally to make the signal richer, but to reduce or attenuate the harmonics with frequencies so high, that they just cannot be represented within the sampling rate.

To understand aliasing, try this experiment: Set the sampling rate in your host to 44100 Hz. Open MOscillator and select a "rectangle" or "full saw" waveform. These simple waveforms have lots of harmonics and without oversampling even they become highly aliased. Now select 16x oversampling and listen to the difference. If you again select 1x oversampling, you can hear that the audio signal gets extensively "dirty". If you use an analyzer (MAnalyzer or MEqualizer for example), you will clearly see how, without oversampling, the plugin generates lots of inharmonic frequencies, some of them which are even below the fundamental frequency. Here is another, very extreme example to demonstrate the result of aliasing. Choose a "sine" shape and activate 16x oversampling. Now use a distortion or some saturation to process the signal. It is very probable that you will be able to hear (or at least see in the analyzer) the aliased frequencies.

The plugin implements a high-quality oversampling algorithm, which essentially works like this: First the audio material is upsampled to a higher sampling rate using a very complicated filter. It is then processed by the plugin. Further filtering is performed in order to remove any frequencies above the Nyquist rate to prevent aliasing from occurring, and then the audio gets downsampled to the original sampling rate.

Oversampling also has several disadvantages of which you should be aware before you start using it. Firstly, upsampled processing induces latency (at least in high-quality mode, although you can select low-quality directly in this popup), which is not very usable in real time applications. Secondly, oversampling also takes much more CPU power, due to both the processing being performed at a higher sampling rate (for 16x oversampling at 44100 Hz, this equates to 706 kHz!), and the complex filtering. Finally, and most importantly, oversampling creates some artifacts of its own and for some algorithms processing at higher sampling rates can actually lower the audio quality, or at least change the sound character. Your ears should always be the final judge.

As always, use this feature ONLY if you can actually hear the difference. It is a common misconception that oversampling is a miraculous cure all that makes your audio sound better. That is absolutely not the case. Ideally, you should work in a higher sampling rate (96kHz is almost always enough), while limiting the use of oversampling to some heavily distorting processors.

Show window

Show window button displays the plugin in a dedicated pop-up window. You can do the same thing by Shift + click on the processor item in the modular grid.
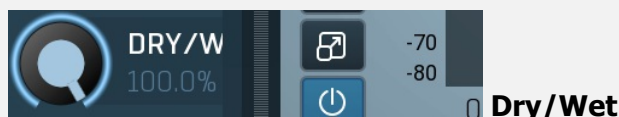
Dry/Wet

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

Note that in the case of minimum-phase (not linear-phase) equalizers this is actually not technically possible, without going back in time. So the plugin simulates it by modifying the actual filters where possible. However the low-pass, high-pass, band-pass and notch filters cannot be simulated. These filters are left with 100% dry/wet unless the ratio is set to 0%, in which case the whole processing is bypassed.
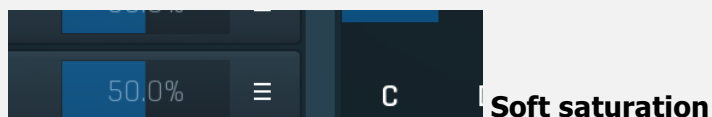Range: 0.00% to 100.0%, default 100.0%
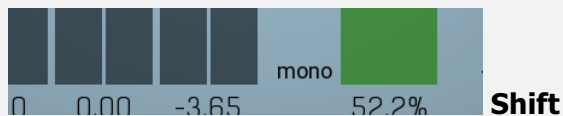
Shift

Shift lets you pitch shift all bands by specified number of semitones. It doesn't change the actual band points, but changes the resulting EQ shape appropriately.
Range: -24.00 to +24.00, default 0
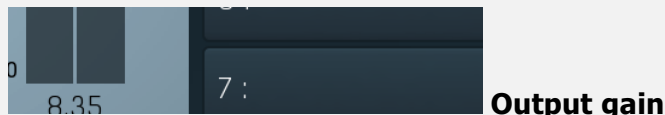
Output gain

Output gain defines output gain applied after the equalization.
Range: -24.00 dB to +24.00 dB, default 0.00 dB

Soft saturation

Soft saturation defines amount of saturation simulating analog equalizers.
Range: 0.00% to 100.0%, default 0.00%

Smoothness

Smoothness makes the analyzer smooth out the curve, so it contains less bumping up and down. It approximates the energy in each frequency and the resulting graph should be easier to understand. Also the smoothness affects the automatic equalization. Usually higher value provides more natural results, however you should verify using your ears.
Range: 0.00% to 20.0%, default 5.0%

Averaging

Averaging makes the analyser show mean values over a specified period of time, which makes the values "jump up and down" less and displays a more user-friendly value, which, of course, is not so accurate in the time-domain.
Range: 0 ms to 5000 ms, default 300 ms

Nonlinear gain

Nonlinear gain changes the way the band gain is interpreted. Normally the gain is assumed to be exactly as you set it. But in some cases, e.g. during mastering, you may focus on very small gains and that's exactly what this feature does.
Range: 0.00% to 100.0%, default 0.00%
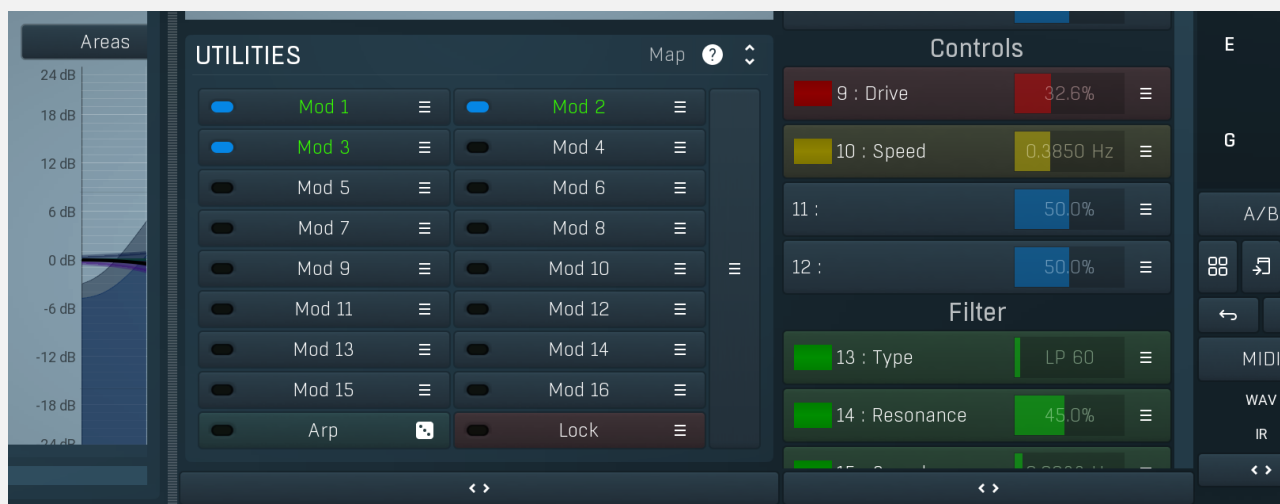
Equalizer shape graph

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

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

Areas

Areas button displays settings for the visual areas, which are useful for better visual orientation in the frequency spectrum. These areas are customisable guidelines displayed in the equalizer editor and may contain different octave bands or typical drum

frequencies for example. Note that these areas are always only guides, so your particular snare drum may not fit exactly in the very well with the example. In that case it is highly advantageous to use the sonogram or analyzer. Or you can edit your own areas.

UTILITIES | Auto-listen

Auto-listen button enables the auto-listen feature, which temporarily changes the equalizer shape when dragging a band to let you see and hear what that particular band is actually doing. For example, when dragging a peak filter, the equalizer disables the other bands and changes this one to a band pass filter, so that you can focus on the frequencies that the peak filter is modifying.

Also, when this is enabled, you can click anywhere in the band's area (shaded) and the equalizer will let you listen to the frequencies at that position using a band-pass filter. This is great for searching for problematic frequencies for example. Vertical position controls the band-width. You can also hold **shift** to get this feature if auto-listen is not enabled.

Analyzer

Analyzer button enables or disables the spectrum analyzer, which shows the levels of individual frequencies. In most practical cases it is more convenient to use the sonogram, which shows the frequencies in time, but provides a lower level resolution as the levels are differentiated by color. The spectrum analyzer also provides a micro-sonogram (shown in the bottom of the panel) which uses the same color-based view as the sonogram.

Fill

Fill button enables or disables the full-sized analyzer micro-sonogram. This means that the micro-sonogram at the bottom of the equalizer graph will fill the whole analyzer view. Color differentiation is often easier to understand than the classical spectrum analyzer, so this might help you better understand the spectrum of your audio material.

An alternative is to use the spectrum sonogram.

Analyzer Rainbow Colors

Analyzer Rainbow Colors lets you see the analyzed sound spectrum in beautiful colors, following the same style as visible light. It ranges from infra-red colors for the lowest frequencies to ultra-violet colors for the highest frequencies in the analyzed audio. If rainbow colors are disabled, the analyzer and graph will be single-colored, following the setup from Settings/Graphs.

Map | Sonogram

Sonogram button enables or disables the spectrum sonogram, which shows levels of individual frequencies in time. Levels are differentiated by color, so the accuracy is not as good as when using the spectrum analyzer. However, the time axis improves the visual orientation in the spectrum for typical audio signals. In contrast, the spectrum analyzer is more of a scientific tool.

Con | Settings

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

trols | Pause

Pause button stops the analyzer temporarily.

Normalize

Normalize button enables or disables the visual normalization, which makes the loudest frequency be displayed at the top of the analyser area (0dB); it does not normalise the sound. This is very useful for comparing frequency levels, however it does hide the actual level.
When comparing 2 spectrums you are usually interested mainly in the frequency level differences. In most cases both audio materials will have different overall levels, which would mean that one of the graphs would be "lower" than the other, making the comparison quite difficult. Normalize fixes this and makes the most prominent frequencies of the spectrum reach the top of the analyzer area (or have the most highlighted color in case of sonogram).

Reset

Reset button resets analyzer graphs. This is particularly useful when analyzing infinite average and maximum values.
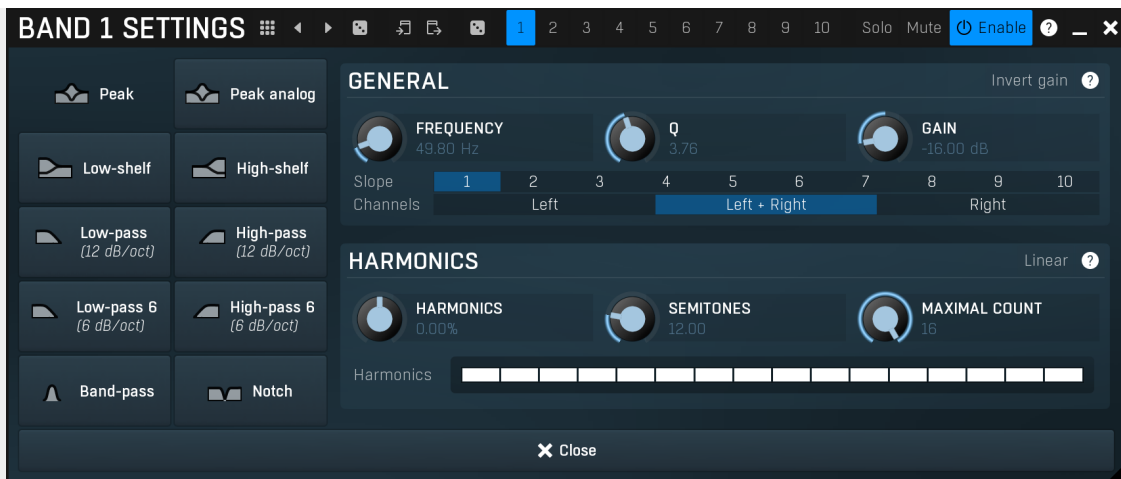
Copy analysis

Copy analysis button copies the current state of the analysis into the system clipboard so that you can paste it into another analyzer for comparison. Hold **ctrl** to export the analysis into a CSV file.

E | Paste

Paste button pastes the analysis from the system clipboard and displays it as the comparison in the graph.

Band settings window

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

Presets

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

Left arrow

Left arrow button loads the previous preset.

Right arrow

Right arrow button loads the next preset.

Randomize

Randomize button loads a random preset.

Copy

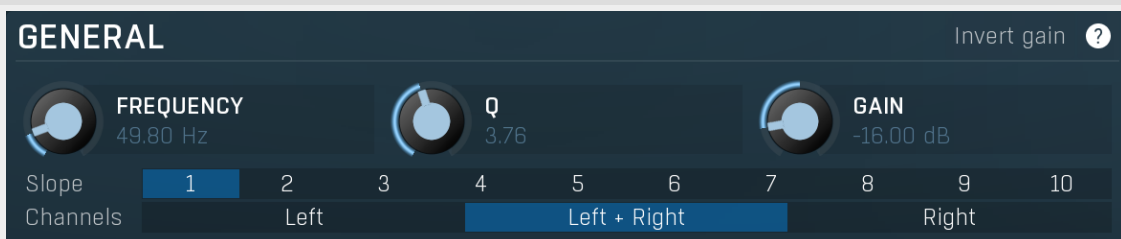Copy button copies the settings onto the system clipboard.

Paste

Paste button loads the settings from the system clipboard.

Random

Random button generates random settings using the existing presets.

General panel

General panel contains standard filter settings such as frequency or Q. Most of these values are available directly from the band graph, but it may be necessary to use these controls for more accurate or textual access.
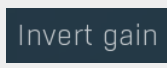
Invert gain Invert gain
Invert gain inverts the gain of the band, e.g. makes -6dB from +6dB.

Frequency

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

Q

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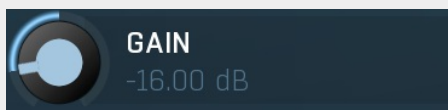

Gain

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

Slope	1	2	3	4	5	6	7	8	9	10

Slope

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

| Channels	Left	Left + Right	Right
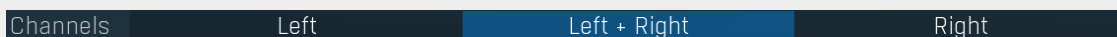
Channels

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

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

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

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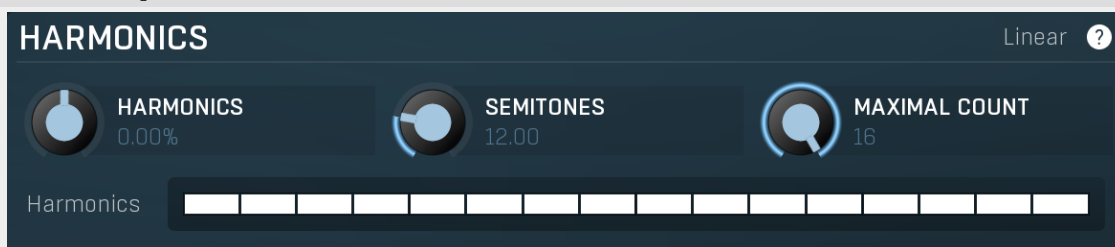
Harmonics panel contains parameters of the harmonics - clones of the main band created at higher frequencies derived from the frequency of the main band. This is often useful for removing natural noises, which usually bring some harmonics with them etc.

Linear

Linear button enables the linear harmonics spacing. When the main band frequency is say 100Hz and the **Semitones** value is 12, then in the default logarithmic mode the harmonics are 200Hz, 400Hz, 800Hz etc., increasing by 12 semitones (1 octave) each time. This is suitable because the filters themselves are logarithmic.
However harmonics generated by physical instruments are not spaced in this way. Rather, for a **Semitones** value of 12, they increase by a multiple of 12/12 of the main frequency each time. For example, for a base frequency of 100Hz, they will be at 200Hz, 300Hz, 400Hz, 500Hz etc. In linear mode the harmonics work in this way, but please note that then there is only a limited set of harmonics and Q is modified to approximate a reasonable behaviour, which is not always possible.

Harmonics

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

Semitones

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

that is, increasing by a multiple of 12/12 of the main frequency each time.

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

MAXIMAL COUNT
16

Maximal count

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

Harmonics grid

Harmonics

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

Bands panel

Bands panel contains the list of available bands along with their basic parameters. You can use it to enable/disable a band, change the parameters and show the band settings window if you do not wish to edit the bands within the equalizer graph panel or if you need to set some values by numeric text entry. The panel is collapsed by default, as it can take a lot of space.

Reset

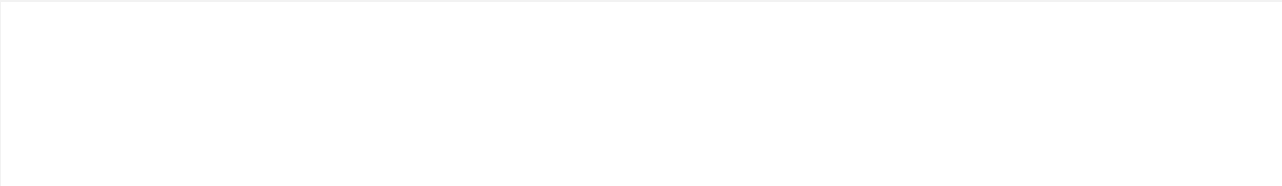Reset button restores the original equalizer settings.

Invert

Invert button inverts the gains of all bands.

Band graphs

Band graphs button enables/disables display of band graphs.

Automatic equalizer panel

Automatic equalizer panel contains long-term analysis and automatic equalization functions.

Analyze source

Analyze source button starts or stops the source analysis, source defines how you want your audio to sound. In your host, route the source audio only to the plug-in and start playback, then press this button to start the analysis. When the graph (shown as a **red line**) stops moving the analysis is finished and you can press the button again to complete the process.

Alternatively, you can analyze an audio file (WAV, MP3, FLAC etc.) offline by clicking the **File** button and browsing to the file or by dragging & dropping the file from your host or Explorer / Finder onto the **Analyze source** button (this latter method may not work in all hosts, especially on Mac). Three other 3 buttons let you save and load an analysis or even draw the desired response manually.

Analyze target

Analyze target button starts or stops the target analysis, target refers to the audio that you want to process. In your host, route the target audio only to the plug-in and start playback, then press this button to start the analysis. When the graph (shown as a **green line**) stops moving the analysis is finished and you can press the button again to complete the process.

Alternatively, you can analyze an audio file (WAV, MP3, FLAC etc.) offline by dragging & dropping the file from your host or Explorer / Finder onto the **Analyze target** button (this may not work in all hosts, especially on Mac).

This tip may come in handy - instead of playing the whole song back to get an accurate analysis you can render the song and analyse the output file, which will probably be faster.

Equalize

Equalize button performs automatic equalization - it adjusts the bands to match the source and target analyses as closely as possible. To do that, you need to have the analyses of both the source and target audio first. When you have both analyses ready, you will see the **red** (source) and **green** (target) analyses in the graph area and this button becomes available.

First get the source analysis using the **Analyse source** button. The Source is the reference audio material that you want your track to sound like.
This is most likely a different track, so you have several options to get this analysis. You can move the plugin to the reference track, perform the analysis then move the plugin onto the track you want to process. Or, open another instance of the plugin on the reference track, perform the analysis and copy the analysis (using the copy/paste buttons, below the A-H preset selectors and A|B comparison button) to the plugin on your track. Or, by saving and loading the analysis (using the **Save** and **Load** buttons). Alternatively, you can draw the desired spectrum or analyse an WAV/MP3/FLAC file by dragging & dropping the file onto the **Analyze source** button.

Secondly, use **Analyse target** button to analyse the audio that you are processing (the **green** line in the graph).

Finally press the **Equalize** button to perform the equalization.

Automatic equalizer tries to match the spectral content of the source analysis graph to the target analysis, producing an equalization curve that aims to make the target audio sound tonally more like the source audio. In most cases the result will be too strong so it is worth considering lowering the **Dry/Wet** parameter to say 30-40% to get a more natural output. You can also make use of the **Smoothness** parameter, above the graph area. Increasing the smoothness before pressing **Equalize** will create a less-pronounced equalization curve.

Separate

Separate button performs automatic separation - adjusting the bands so that the target does not contain those frequencies that are prominent in the source. This is useful, for example, during mixing to avoid collisions between multiple tracks. To do that, you need to have analyses of both the source and target audio first. When you have both analyses ready, you will see **red** (source) and **green** (target) analyses and this button becomes available.

*For example, say you want to avoid collisions between bass and bass drum. One of them will have to be sacrificed and processed by the equalizer, let's choose the bass. In that case you would analyse the bass drum as the source (the **red** line) and put the equalizer into the bass track afterwards and analyse that as the target (the **green** line). **Separate** would then produce an equalization curve that reduces, in the bass track, those frequencies that are prominent in the bass drum track.*

Another example is typical when your mix is already busy, but you need to put one more track to it. In this case you analyse the whole mix as the source and your new track as the target. The separation will then allow only those frequencies from your new track that are not prominent in the whole mix; in other words, frequencies that are already prominent in the mix are not affected by those same frequencies in the new track. If this separation were not done, then those frequencies would start colliding with the rest of your mix and that could make it sound muddy and crowded.

When you have both analyses ready, click this button to perform the separation. In most cases the result will be too strong so it is worth considering lowering the **Dry/Wet** parameter to say 30-40% to get more natural output. You can also make use of the **Smoothness** parameter, above the graph area. Increasing the smoothness before pressing **Equalize** will create a less-pronounced equalization curve.

Reset

Reset button clears both the source and target analyses.

Remove resonances

Remove resonances button takes the target analysis, tracks peaks in it and creates notch filters to remove them. It is especially useful with various audio materials, such as drums, where resonances of the instruments are too prominent. You simply analyze the part of the audio material, where the resonances are mostly audible, and click this button. Then you can just disable the bands that
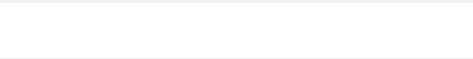
remove the frequencies that you actually want to keep. It may also be worth trying to play with the Q values of each band to make sure the resonances are completely gone.
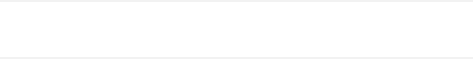
Swap source and target
Swap source and target button swaps the analyses of source and target and can be helpful when you want to try equalizing 'the other way around'.

Settings
Settings button shows additional settings of the automatic equalizer algorithm.

Auto-equalizer settings

Auto-equalizer settings provides additional settings for the automatic equalization algorithm.

Bottom frequency
Bottom frequency defines lowest frequency taken into account when performing automatic equalization. Use it to avoid unnecessary processing of the bottom-end.

Top frequency

Top frequency defines highest frequency taken into account when performing automatic equalization. Use it to avoid unnecessary processing of the high-end.

Maximum bands

Maximum bands defines maximal number of bands that will be configured during the automatic equalization. Generally more bands provide more accurate match to the source analysis, however this may not always be desired. You can also make use of the **Smoothness** parameter, above the graph area. Increasing the smoothness before pressing **Equalize** will create a less-pronounced equalization curve.

Space in mix limit

Space in mix limit controls the minimal level below which a particular frequency is considered silent. It is used for the **Separate** feature to determine which frequencies in the source (full mix for example) are candidates for separation and which are not.

For example, if this value is -20dB and the level of the frequency 1000Hz in the source analysis is -20dB, then the separation engine will consider 1000Hz to be "available" in the target audio and will not try to remove this frequency from the target. However if the level in the source were -10dB, then the engine would consider this frequency 50% occupied and may try to use a filter to remove some of the 1000Hz from the target if it contains this frequency as well.

In other words, frequencies in the source with levels louder than the mix limit will be those that would be reduced in the target.

Algorithm

Algorithm selector lets you choose which method is used to convert the frequency response into a configuration of the equalizer bands. (This algorithm itself is actually very complex and requires lots of CPU when adjusting the bands during the equalization/separation processes.) Multiple algorithms are available, each of them with a slightly different output.
Default provides the most accurate results, but it takes lots of CPU. Note, as mentioned above, that this CPU usage is required only for the pre-computing, the actual audio processing requirements are the same for all algorithms.
Optimized algorithm is much faster, however it can produce slightly less accurate results. It may be more suitable because it smoothes out the peaks in the frequency response.
Super-fast mode is the fastest of all of them, but it trades speed for accuracy in its results. You may find it useful in specific cases.

Never amplify

Never amplify ensures that the automatic equalization only attenuates frequencies so that no amplification occurs. This may be desired, since amplification often increases noise level. On the other hand, due to its nature, it will probably lower the output level. It may be worth considering using AGC to set the output gain afterwards.

Disable pass/notch

Disable pass/notch makes automatic equalization use only peak and shelf filters. This way you can use the **Dry/Wet** parameter even in the non-linear-phase mode, because peak and shelf filters can be affected by the ratio in this mode too.

Use peak filters for resonance removal

Use peak filters for resonance removal option makes the **Remove resonances** feature use peak filters instead of the default notch filters, which are more effective, but **Dry/Wet** doesn't affect them.
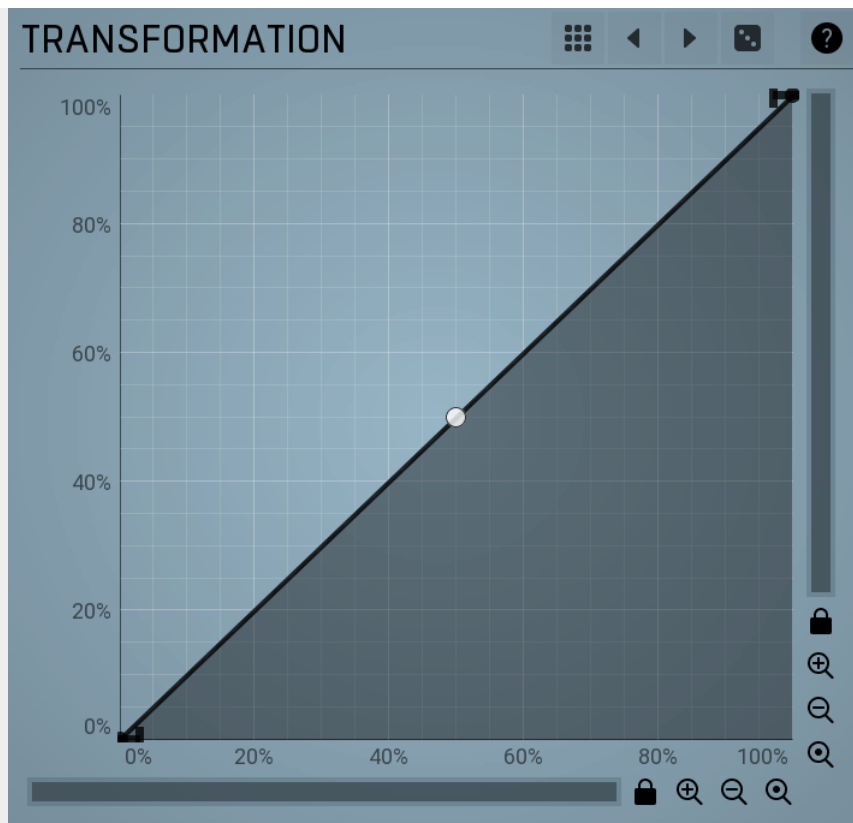
TRANSFORMATION

Transformation

Transformation changes the requested frequency response. The engine first determines the optimum frequency response (the algorithms for equalization and separation are different). Then it computes the optimal response of the equalizer. Before it approximates this response using the equalizer filters, that response can be transformed. The transformation graph shows the original requested response on the X-axis and the new one on the Y-axis. For example you can exaggerate or sharpen the results just using the single point curvature.

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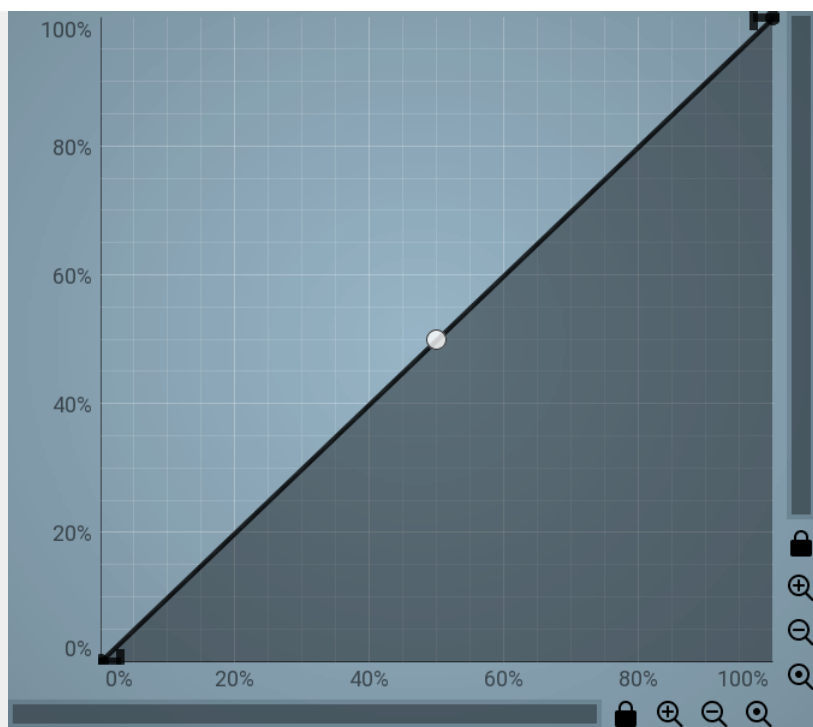
Left arrow
Left arrow button loads the previous preset.

Right arrow
Right arrow button loads the next preset.

Randomize
Randomize button loads a random preset.

EnvelopeEditorGraph

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.

Global parameters panel

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.

 Dry/Wet

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

 0%

0% button makes the **Dry/Wet** virtually 0%. You can use it for comparison.

100%

100% button makes the **Dry/Wet** virtually 100%. You can use it for comparison.

Input gain

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

Output gain

Output gain defines the power modification applied to the output signal, right after it is summed from the bands.

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**.
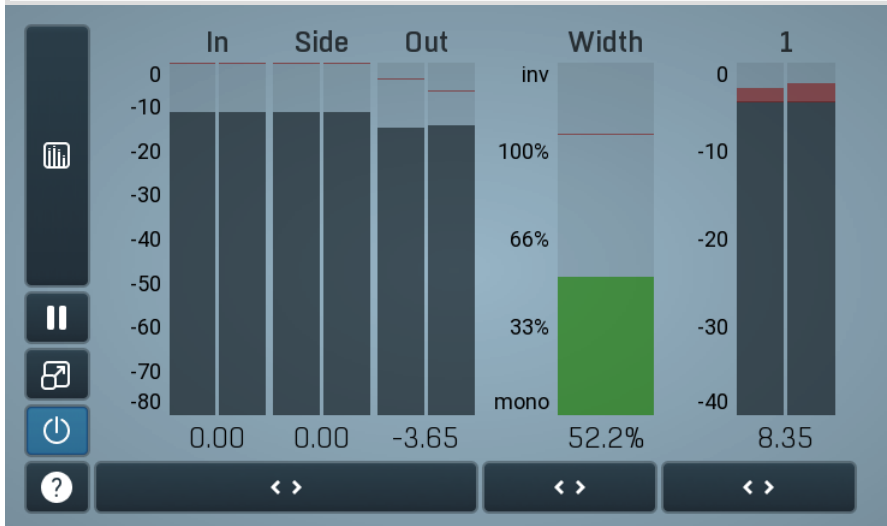
Global meter view

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.

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.

Width meter shows the stereo width at the output stage. This meter requires at least 2 channels and therefore does not work in mono mode. Stereo width meter basically shows the difference between the mid and side channels.
When the value is **0%**, the output is monophonic. From 0% to 66% there is a green range, where most audio materials should remain.
From 66% to 100% the audio is very stereophonic and the phase coherence may start causing problems. This range is colored blue. You may still want to use this range for wide materials, such as background pads. It is pretty common for mastered tracks to lie on the edge of green and blue zones.
Above 100% the side signal exceeds the mid signal, therefore it is too monophonic or the signal is out of phase. This is marked using red color. In this case you should consider rotating the phase of the left or right channels or lowering the side signal, otherwise the audio will be highly mono-incompatible and can cause fatigue even when played back in stereo.
For most audio sources the width is fluctuating quickly, so the meter shows a 400ms average. It also shows the temporary maximum above it as a single coloured bar.
If you right click on the meter, you can enable/disable loudness pre-filtering, which uses EBU standard filters to simulate human perception. This may be useful to get a more realistic idea about stereo width. However, since humans perceive the bass spectrum as lower than the treble, this may hide phase **problems** in that bass spectrum.

Time graph

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

Pause

Pause button pauses the processing.

Popup

Popup button shows a pop-up window and moves the whole metering / time-graph system into it. This is especially useful in cases where you cannot enlarge the meters within the main window or such a task is too complicated. The pop-up window can be arbitrarily resized. In metering mode it is useful for easier reading from a distance for example. In time-graph mode it is useful for getting higher accuracy and a longer time perspective.

Enable

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

Collapse

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

Collapse

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

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Utilities

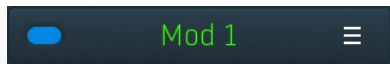| UTILITIES	Map	
Mod 1	Mod 2	
Mod 3	Mod 4	
Mod 5	Mod 6	
Mod 7	Mod 8	
Mod 9	Mod 10	
Mod 11	Mod 12	
Mod 13	Mod 14	
Mod 15	Mod 16	
Arp	Lock	

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.

Randomize

Randomize button generates random arpeggiator settings.

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.