MConvolutionMB

Easy screen vs. Edit screen

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

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

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

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

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

Devices are factory specified and cannot be modified directly by users, however you can still make your own and store them as normal presets. To do so, configure the plugin as desired, then define each multiparameter and specify its name in its settings. You can then switch to the easy screen and check the user interface that you have created. Once you are satisfied with it, save it as a normal preset while you are on the easy screen. Although your preset will not be displayed or selected in the list of available devices, the functionality will be exactly the same. For more information about multiparameters and devices please check the **online video tutorials**.

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

Edit mode

Presets

Presets button shows a window with all available presets. A preset can be loaded from the preset window by double-clicking on it, selecting via the buttons or by using your keyboard. You can also manage the directory structure, store new presets, replace existing ones etc. Presets are global, so a preset saved from one project, can easily be used in another. The arrow buttons next to the preset button can be used to switch between presets easily.

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

Presets can be backed up by 3 different methods:
A) Using "Backup" and "Restore" buttons in each preset window, which produces a single archive of all presets on the computer.
B) Using "Export/Import" buttons, which export a single folder of presets for one plugin.
C) By saving the actual preset files, which are found in the following directories (not recommended):
Windows: C:\Users\{username}\AppData\Roaming\MeldaProduction
Mac OS X: /Library/Application support/MeldaProduction

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

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

Left arrow

Left arrow button loads the previous preset.

Right arrow

Right arrow button loads the next preset.

Randomize

Randomize button loads a random preset.

Panic

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

generating very high values as well and as a result the playback will stop. You can just click this button to reset the plugin and the playback will start again.

Settings Settings

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

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

Sleeping Sleep indicator

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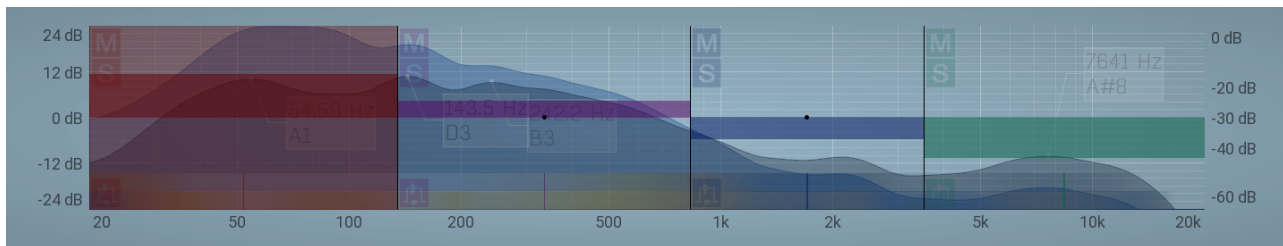
 Band editor

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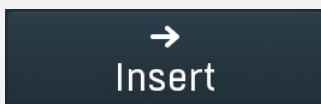
Band management panel
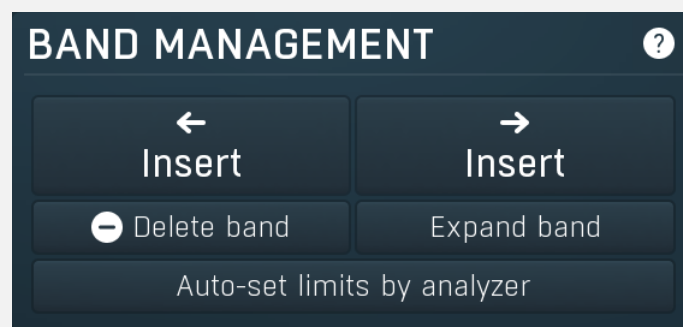
Band management panel contains basic features to create, delete and manipulate bands.
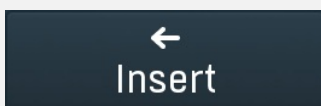
 Insert left

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 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 ?

1   2   3   4   5   6
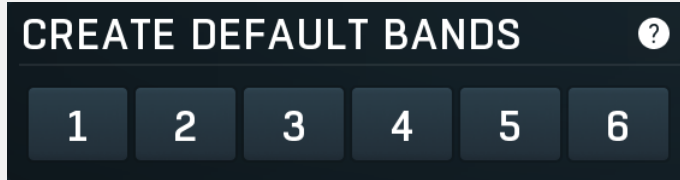
Create default bands panel lets you easily create a predefined set of bands. This is the easiest way to say create default plugin settings with 4 bands.

Clipboard panel

CLIPBOARD ?

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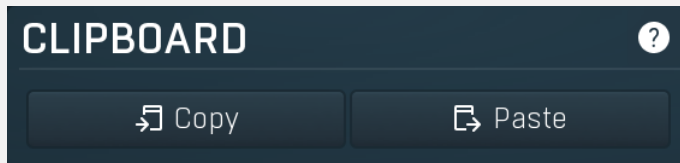
Clipboard panel contains features to transfer band settings via the system clipboard. Note that as always you can paste the settings as text into an email or forum post for example.

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 ?

Gain   Gain (all bands)
Panorama   Panorama (all bands)
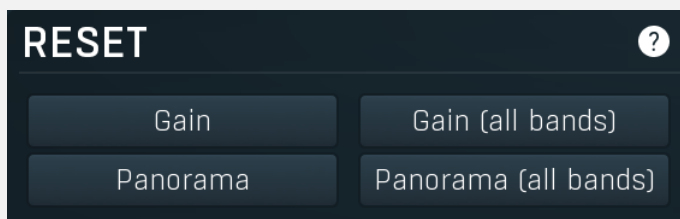
Reset panel lets you reset some band parameters.

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)**

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	6dB/oct A	6dB/oct B	12dB/oct
	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 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	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 tranparent 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

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.
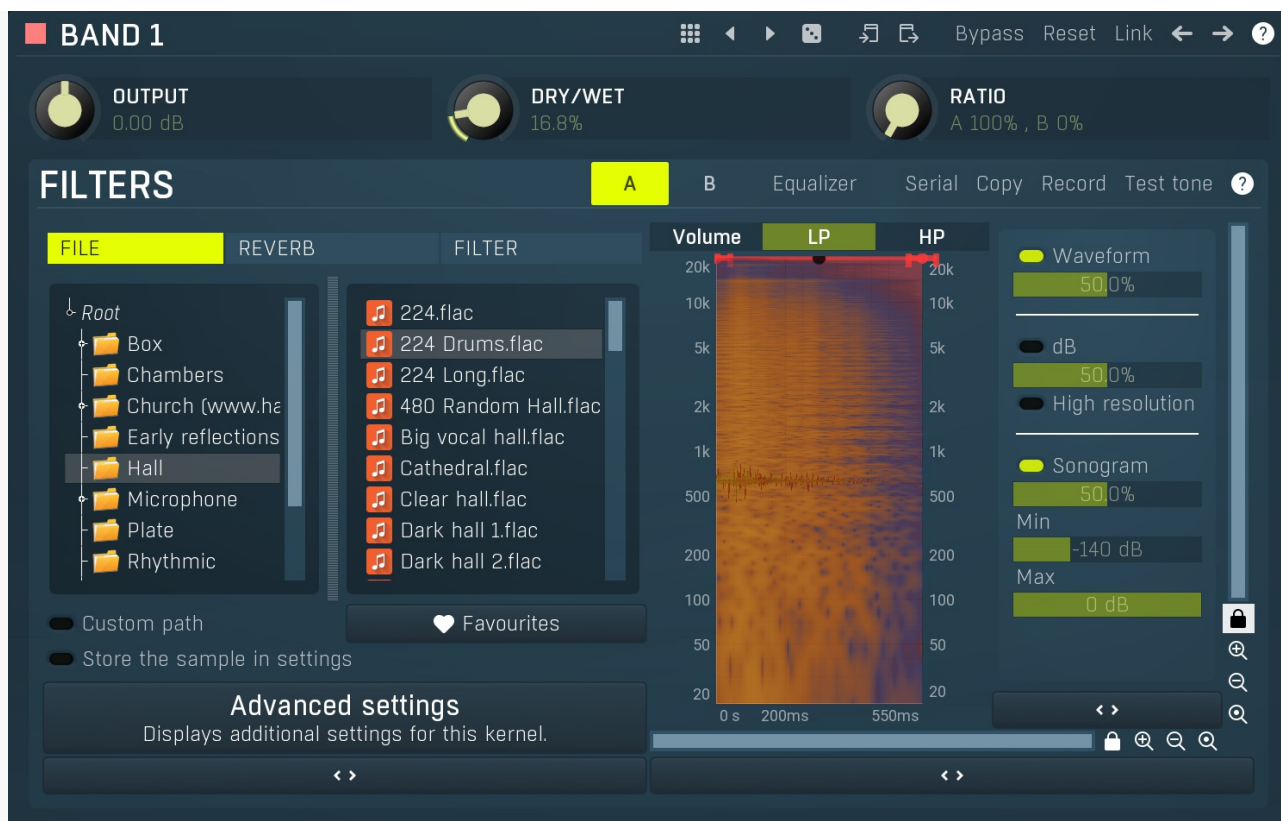
🔧 Settings

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

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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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.

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.

Output gain

Output gain defines the gain applied to the output.
Range: -24.00 dB to +24.00 dB, default 0.00 dB
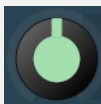
Dry/Wet

Dry/Wet defines ratio between dry and wet signals. 100% means fully processed, 0% means no processing at all.
Range: 0.00% to 100.0%, default 50.0%

Ratio

Ratio defines the relative amounts of the A and B filter results. It can be used to modulate between multiple filters. You can for example load a reverberation impulse response into A filter, and another one into B filter. Then you can move the ratio to increase the volume for one and decrease it for the other. Or you can even attach the ratio to a modulator. This way the sound will be subtly changing over time, which results in less static and boring sounds.
Range: A 100% , B 0% to A 0% , B 100%, default A 100% , B 0%

Filters panel
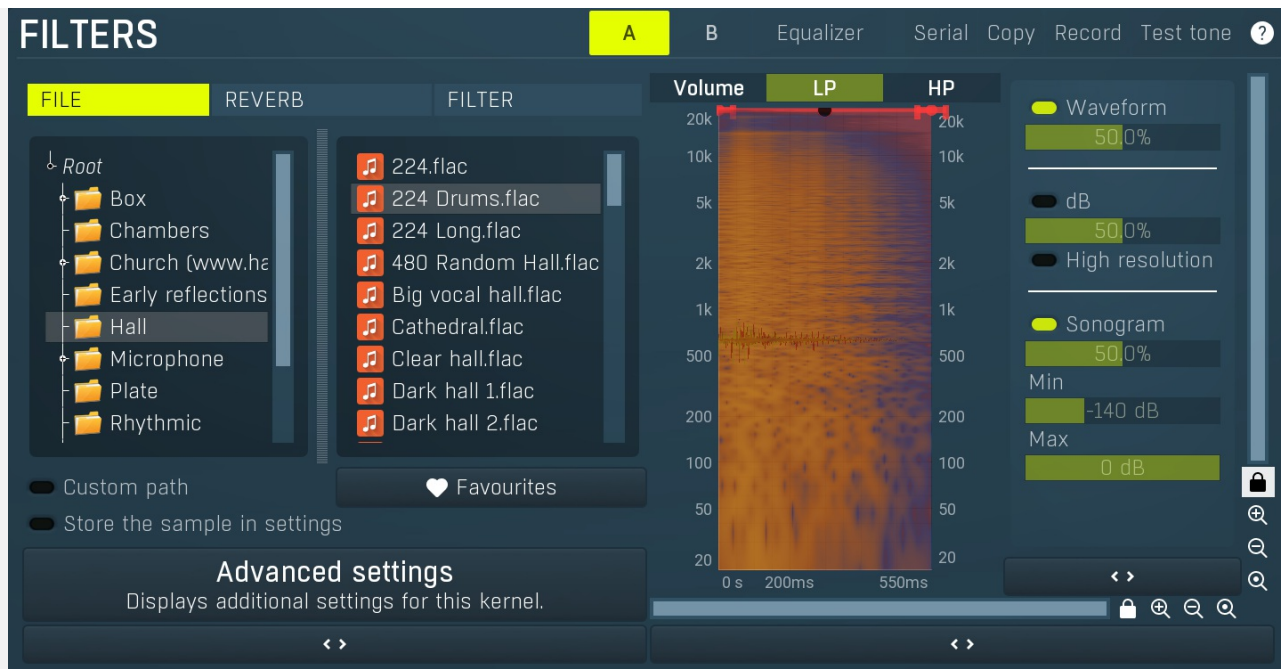
Filters panel contains the actual processing settings. The processor uses 2 kernels, called A and B. Each of them is a fully featured multichannel convolution processor and you can use a file (or more than one) containing an impulse response or the integrated algorithmic reverb engine with them. The plugin processes both kernels in parallel, mixes them according to the **Ratio** parameter, feeds the output to the master equalizer, and finally mixes then results with the dry input signal according to the **Dry/Wet** parameter.

 A

A button switches the editor to the A filter. There are 2 filters available together, A and B, and you can interpolate between them using the **ratio** slider.

 B

B button switches the editor to the B filter. There are 2 filters available together, A and B, and you can interpolate between them using the **ratio** slider.
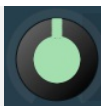
 Equalizer

Equalizer button enables the master equalizer which is applied to the filter output.

Master equalizer
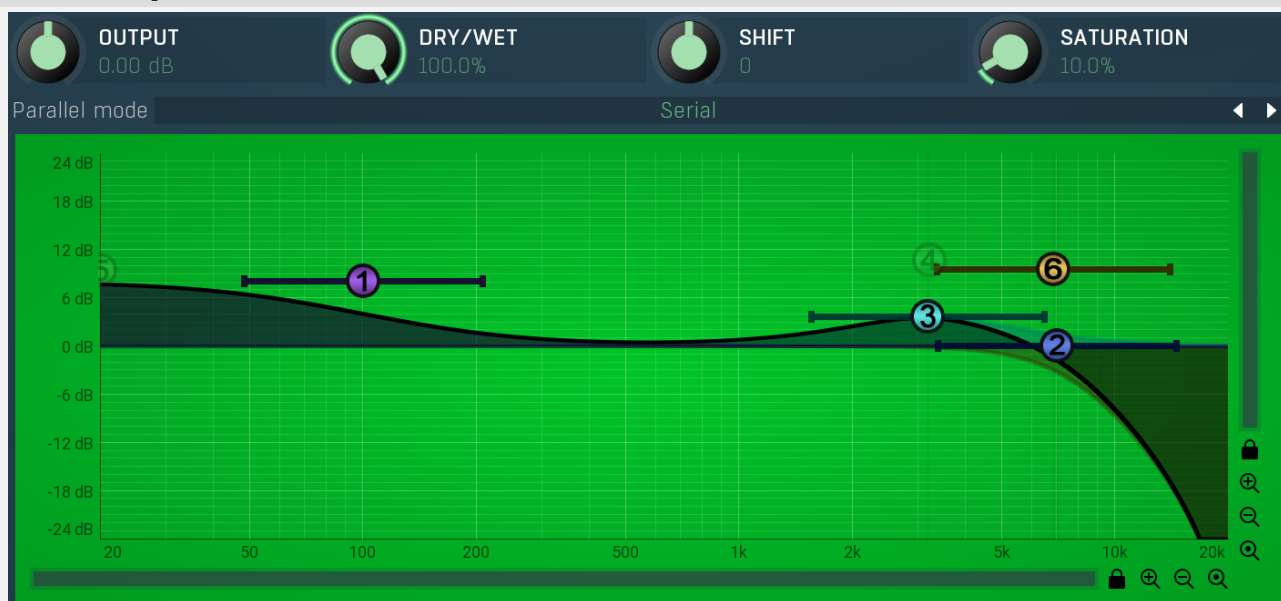
Master equalizer is useful to process the output of the A and B filters. The plugin first performs convolution with the A or B filters or with both of them. It then uses this equalizer before it mixes the processed signal with the dry input, depending on the dry/wet setting. For example, you can use this to make a reverb darker using a low-pass filter. The master equalizer processes the wet signal only, it does not affect the dry signal.

Output gain

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

Dry/Wet

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

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

Soft saturation

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

Parallel mode	Serial	◀ ▶

Parallel mode

Parallel mode defines if the bands are processed serially and parallelly, which may provide interesting phasing effects.
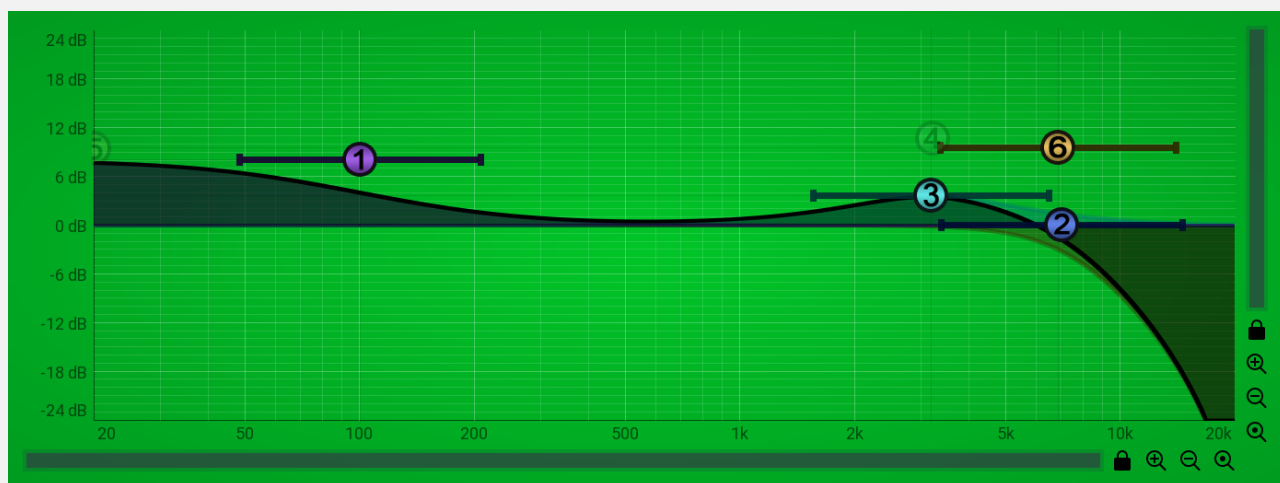
Equalizer shape graph

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

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

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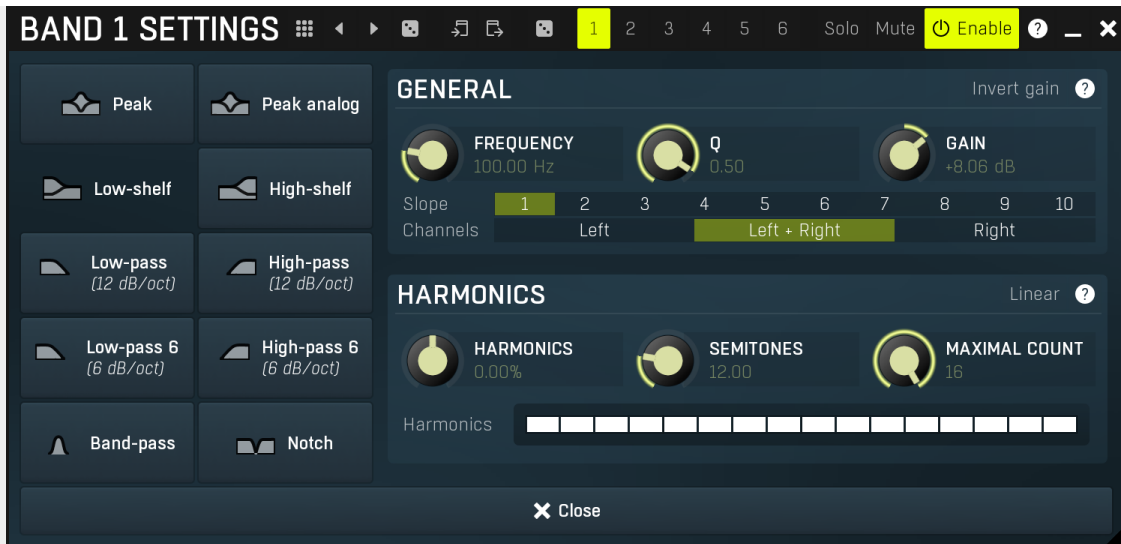
Band settings window contains settings for the particular band and can be displayed by right-clicking on a band or from a band list (if provided). On the left side you can see list of available filters, click on one to select it. On the right side, additional options and features are available.

Presets

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

Left arrow

Left arrow button loads the previous preset.

Right arrow

Right arrow button loads the next preset.

Randomize

Randomize button loads a random preset.

Copy

Copy button copies the settings onto the system clipboard.

Paste

Paste button loads the settings from the system clipboard.

Random

Random button generates random settings using the existing presets.

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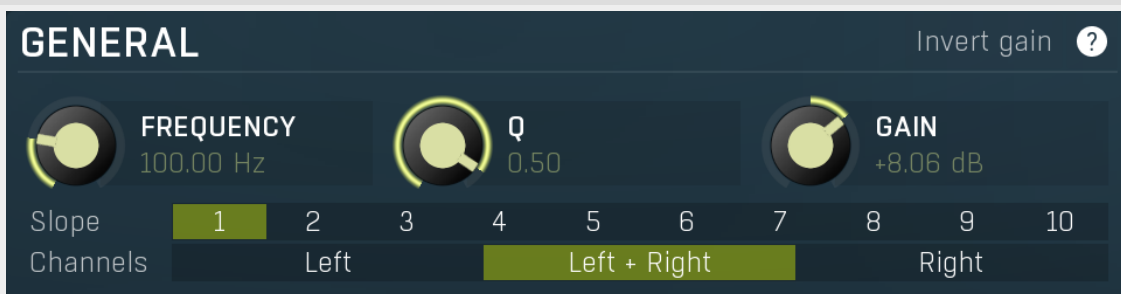
General panel contains standard filter settings such as frequency or Q. Most of these values are available directly from the band graph, but it may be necessary to use these controls for more accurate or textual access.
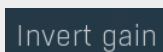
Invert gain
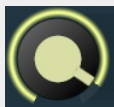
Invert gain inverts the gain of the band, e.g. makes -6dB from +6dB.

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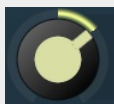
Frequency defines the band's central frequency, which has different meaning depending of filter type.

Q

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

Gain

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

Slope

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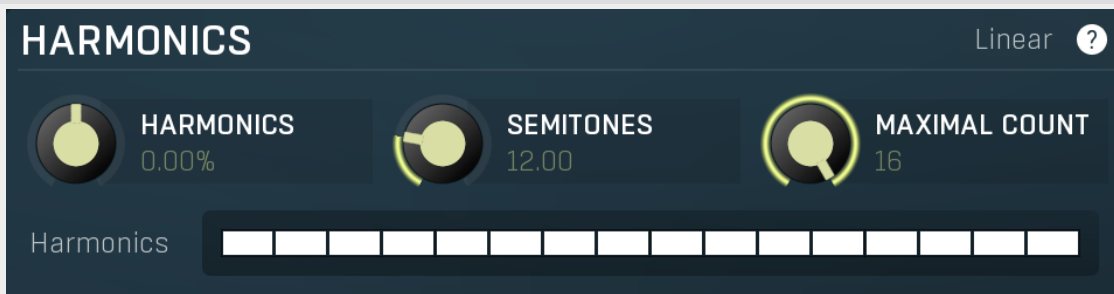
Channels

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

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

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

Harmonics panel

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

Linear

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

Harmonics

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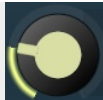
Semitones

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

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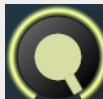
Maximal count

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

Harmonics grid

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

Serial

Serial button activates or deactivates serial mode. In serial mode convolution with A impulse response is performed and the output is processed using B impulse response. In parallel mode (default) both convolutions are performed on the source signal and you can then use **ratio** to mix between them.

Copy

Copy button copies settings of the A or B kernel into the other kernel.

Record

Record button is useful to record the signal from a hardware unit or even another plugin. You simply put one instance of this plugin in front of the black-box that you want to simulate, whether it is a hardware or a plugin and another behind it. Click **test tone** in the first instance to generate an impulse and start recording in the second. When the output fades out, click the record button again to stop recording. The plugin will analyze the recorded impulse response and save it as an IR sample. The recorded IR simulates the device. Please note that only linear devices can be simulated. For example any device containing some type of modulation, e.g. chorus or flanger, cannot be simulated. IRs are mostly used to simulate reverberation of real spaces and responses of amps and microphones.

To record another plugin, use MConvolver (A)->BlackBox->MConvolver (B). Instance A will generate the test tone and instance B will record it.

To record a piece of hardware, such as an amp, use MConvolver (A)->Output->Amp input, and Amp output (or microphone)->Input->MConvolver (B). Again, instance A will generate the test tone that goes into the amp, which processes it in some way and is routed back into your audio device and to instance B.

Test tone

Test tone button generates a broad-band pulse used for recording impulse responses. See **record button** for more information.

File/reverb/filter header

File/reverb/filter header selects particular generator for selected filter A or B.

File mode

File mode is the most common generator; it uses recorded reverberation impulse responses. You can use any audio file (WAV, MP3, FLAC etc.) on your file system. You can either search for the files in the supplied file browser or drag & drop the file from your system explorer/finder onto the file browser (this may not work in some hosts on Mac OS X though).

If the file contains multiple channels, these will be used according to current **topology** (selected in the **Advanced settings** window). Additionally the plugin tries to find further channels from similarly named files in the same directory according to this scheme:
select **xxx1.yyy** -> tries to load **xxx2.yyy**, **xxx3.yyy**...
select **xxxL.yyy** -> tries to load **xxxR.yyy**
select **xxx.1.1.yyy** -> tries to load **xxx.1.2.yyy**, **xxx.2.1.yyy**, **xxx.2.2.yyy**

This is useful when you have several monophonic samples and you want to combine them into a single filter. For example you may have churchL.wav and churchR.wav containing recordings of 2 impulse response from a church used for left and right channel.

The plugin also provides **true-stereo** processing mode (as one of the topologies available) using 4 IRs, in which the first 2 IRs are used for left output and next 2 IRs for right output. Several variations are available.

Finally the plugin provides **surround** processing with up to 8 channels and since it is very impractical to have 8 channel IR wave files, it's likely you will want them to be combined from multiple mono or stereo samples such as church1.wav, church2.wav, church3.wav etc.
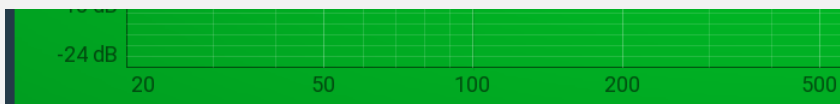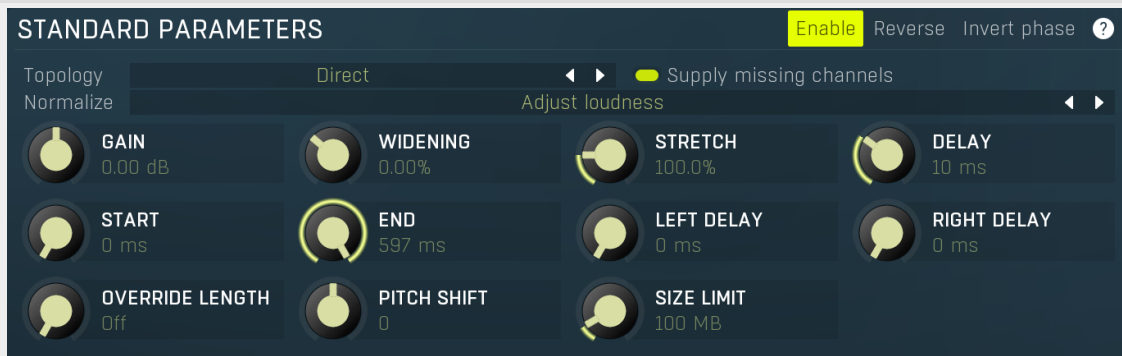
 Advanced settings

Advanced settings button displays additional settings for this kernel.

Kernel settings A

KERNEL SETTINGS A

Standard parameters

Enable **Enable**

Enable button enables the processing by this kernel. It may be useful to bypass the kernel when not needed in order to save CPU.

Reverse **Reverse**

Reverse button mirrors the IR providing reversed reverberation.

Invert phase **Invert phase**

Invert phase inverts the phase of each sample of the IR. This can be helpful when mixing the original with the IR causes some phasing issues.

Topology **Direct** ◄ ► **Topology**

Topology lets you choose between processing topologies.

Direct is the default value, where the first channel is convolved with the first impulse response channel, the second channel with the second IR channel etc. When you have for example 4 channels and just 2 IR channels, then channel 3 uses IR 1 and channel 4 uses IR 2.

True-stereo uses 4 IRs to process stereo input into stereo output. With direct topology left output depends on the left input

only. With true-stereo however, the left output depends on both input channels as it does in the real world. LLRR configuration convolves IRs 1 and 2 with both channels to produce the left output and IRs 3 and 4 to produce the right output. LRLR configuration uses IRs 1 and 3 for the left output and IRs 2 and 4 for the right output. Since you will probably use 2 stereo impulse responses, this will actually behave like adding both IRs together, thus like 2 reverbs together.

Mono to stereo merges the input channels into mono and then performs the convolution. This may be handy if the input is already panned somewhere, for example if the left channel is much lower in volume, which would make the left output also very low in volume. It can also be useful for surround processing - if you have a stereo/mono input and you want to generate the surround channels as well, the plugin can do this by merging the first 2 channels to mono and then generating the whole surround set from it using convolution.

Expand xx -> surround modes are useful for surround processing. If you have a stereo input and you want to generate the surround channels, you could use the **Mono to stereo** topology; this however basically takes the input as mono. In contrast, the expansion topologies assume a stereo input and generate the surround channels from them. LR & RL cases generate each channel from just one of the input channels, which makes it similar to Direct topology. LLRR and LRLR on the other hand generate each surround channel from both input channels, hence they work the same way as true-stereo topologies, but expanded to surround.

Swapped is useful to simulate true-stereo in surround systems. Let's say you have 8 input channels and 8 IR channels. Direct topology would convolve input 1 with IR1, input 2 with IR2 etc. Swapped topology however swaps every pair of channels - In1 * IR2, In2 * IR1, In3 * IR4, In4 * IR3 etc. You can use filter A in direct topology and filter B in swapped topology and set ratio to 50% in which case B will be complementary to A and the result will be the same as true-stereo, but for up to 8 channels.

Supply missing channels

Supply missing channels makes the engine use the same IR channel twice if not enough channels are available. By default it is enabled and makes sure even a monophonic IR file makes all channels processed no matter how many there are. However it is mainly useful to disable this for surround/ambisonics processing, where your IR contains only as many channels as you actually want to process and processing more would produce unwanted spatial shift.

Normalize	Adjust loudness	◀ ▶

Automatic gain

Automatic gain detects the gain caused by the IR itself and compensates for it. Most recorded IRs are normalized or somehow modified so they do not generate the same output level. The plugin can compensate for that. **Adjust level** is an older method, which usually works well with longer reverberation impulse responses. **Adjust loudness** is the new method, which results in approximately same output loudness, unless your audio or impulse response is highly resonant. This makes it ideal. The disadvantage is that it usually requires more CPU when loading or modifying the impulse response.

Gain

Gain provides the input gain for the filter kernel itself.
Range: -48.00 dB to +48.00 dB, default 0.00 dB

Widening

Widening is useful only if you have 2 channel IRs and performs the stereo-widening algorithm, which results in wider output as well.
Range: Mono to 200.0%, default 0.00%

Stretch

Stretch resizes the impulse response. As a result you may observe not only a change of length, but also a change of pitch.
Range: 25.0% to 400.0%, default 100.0%

Delay

Delay provides a predelay in milliseconds, which defines the initial delay of the processed signal and is often used to simulate the distance of the sound source.
Range: 0 ms to 1000 ms, default 0 ms

Left delay

Left delay provides a predelay in milliseconds that is applied only to the left channel. This is useful for additional control over the stereo placement of the resulting audio signal.
Range: 0 ms to 1000 ms, default 0 ms

Right delay

Right delay provides a predelay in milliseconds that is applied only to the right channel. This is useful for additional control over the stereo placement of the resulting audio signal.
Range: 0 ms to 1000 ms, default 0 ms

Override length

Override length changes the length of the IR to the specified length. It can be used for various creative effects, but the most typical use is to synchronize an IR consisting of a loop to the current tempo. It is ignored when the length is minimum, hence 0ms, or OFF. Please note that since its operation is the same as the **Pitch shift** parameter, it is strongly advised not to use both parameters at once.
Range: Off to 60000 ms, default Off

Pitch shift

Pitch shift changes the length of the IR and as a result causes a pitch shift specified in semitones. Please note that since its operation is the same as **Override length** parameter, it is strongly adviced not to use both parameters at once.
Range: -24.00 to +24.00, default 0

Size limit

Size limit restricts the size of the files to load and protects yourself from accidentally choosing files that are too big, which could exceed the amount of available RAM eventually leading to a crash. The size is defined in megabytes, for MP3 files this value is divided by 10.
Range: 0 MB to 1000 MB, default 100 MB

Modulation

Modulation depth

Modulation depth controls the depth of the input vibrato useful to further improve the realisticity of the produced reverberation. It can sound very artificial when overused however. It also requires additional CPU. Set this to 0% to disable the vibrato.
Range: 0.00% to 100.0%, default 0.00%

Modulation rate

Modulation rate controls the speed of the input vibrato oscillator. The faster it is, the more pronounced the effect is.
Range: 0.0100 Hz to 10.00 Hz, default 0.2000 Hz

Equalizer

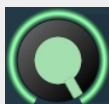## Output gain

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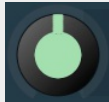Range: -24.00 dB to +24.00 dB, default 0.00 dB

Dry/Wet

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

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

Soft saturation

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

Parallel mode

Parallel mode defines if the bands are processed serially and parallely, which may provide interesting phasing effects.


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.
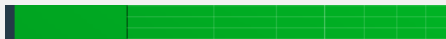
File

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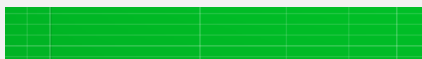
 Favourites

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

 Store the sample in settings

Store the sample in settings is useful when you want to move a project onto a different computer without your specific impulse response files installed. You can just enable this switch and the sample will be stored in the plugin settings. Please note that this may make the project file very big.

Reverb mode

Reverb mode provides a proprietary impulse response generator, which creates very realistic impulse responses based on several parameters that you can see below. Please note that since this is still a convolution reverb, it will probably require much more CPU than an algorithmic reverb.
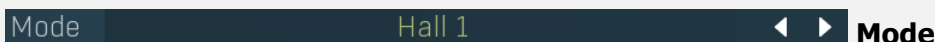
 Mode

Mode controls the type of the reverbation impulse response.
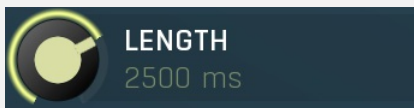
 Length

Length defines total length of the response.
Range: 10 ms to 10000 ms, default 2500 ms

 Predelay

Predelay defines the initial delay before the actual response, which simulates the space between the sound source and the listener. The longer the predelay, the further away the source seems.
Range: 0 ms to 250 ms, default 0 ms
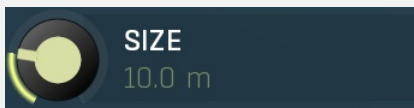
 Size

Size defines the size of the room or hall, which affects several parameters, such as density of the echoes.
Range: 1.0 m to 40.0 m, default 10.0 m
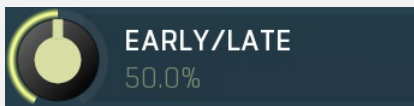
 Early/late

Early/late controls the amounts of early and late reflections.
Range: 0.00% to 100.0%, default 50.0%

DENSITY
50.0%

Density

Density defines the density of the generated echoes. You may need to set density to make your audio material sound more natural.
Range: 0.00% to 100.0%, default 50.0%

DECAY
50.0%

Decay

Decay controls shape of the decay. The higher the decay, the faster the output fades out.
Range: 0.00% to 100.0%, default 50.0%

DAMPENING
50.0%

Dampening

Dampening defines the amount of dampening by surrounding objects, the walls, floor etc., which absorb particular frequencies.
Range: 0.00% to 100.0%, default 50.0%

Enable decay

Enable decay

Enable decay switch lets you disable the default decay processing controlled by the **Decay** parameter, so that you can draw your own decay using the volume graph.
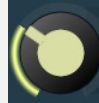
Advanced settings
Displays additional settings for this kernel.
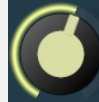
Advanced settings

Advanced settings button displays additional settings for this kernel.

Filter mode

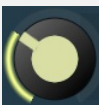Type   LowPass   ◀ ▶

LENGTH
1000 ms

FREQUENCY
1000 Hz

Filter mode provides a special filter generator, such as an extremely steep low-pass filter. It is mostly used by engineers to remove unwanted noises etc. Please note that in order for this to work, all **dry/wet** controllers must be set to 100%.

Type   LowPass   ◀ ▶

Type

Type defines the type of filter to generate, such as low-pass or high-pass. Hilbert filter rotates phase of all frequencies by 90 degrees and can be used for several scientific purposes. Please note that all these filters induce latency equal to half of the filter length and the plugin cannot compensate for it. It can only compensate the dry signal, which means that the dry/wet feature does not work in this case.

LENGTH
1000 ms

Length

Length defines the length of the filter kernel. The longer the kernel, the steeper and more accurate the filter; however the higher the latency is and the more CPU it consumes.
Range: 1.0 ms to 10000 ms, default 1000 ms

FREQUENCY
1000 Hz

Frequency

Frequency defines the limit frequency used for the low-pass and high-pass filters.
Range: 20.00 Hz to 20.0 kHz, default 1000 Hz

‹ ›

Collapse

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

Sample graph

Sample graph contains the image of the impulse response and lets you edit the volume envelope. If you want to shorten the sample, just make the end of the sample silent and it will be shortened automatically, or you can use the **advanced settings**.

Envelope graph
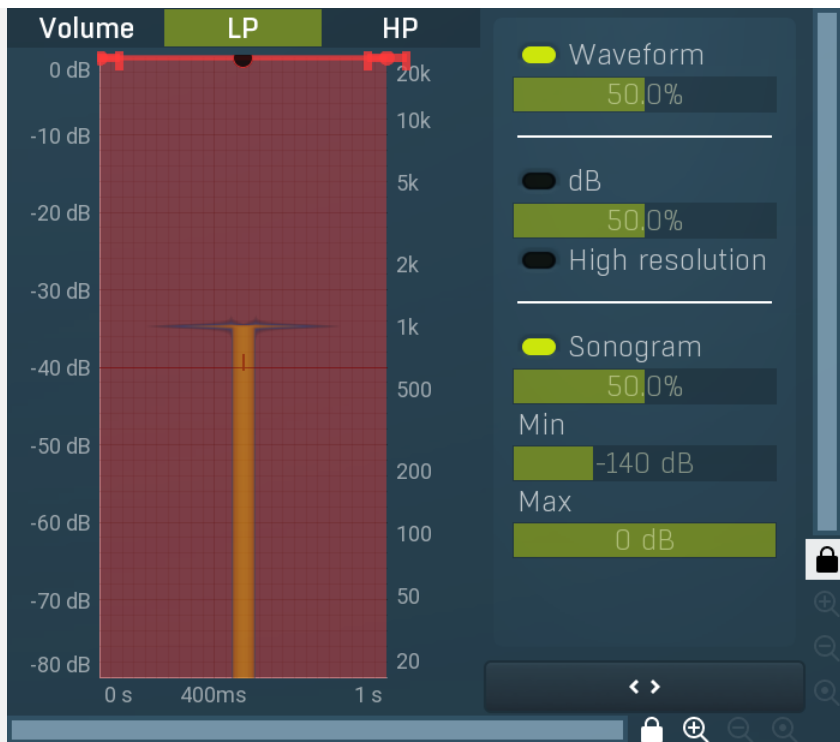
Envelope graph provides an extremely advanced way to edit any kind of shape that you can imagine. An envelope has a potentially unlimited number of points, connected by several types of curves with adjustable curvature (drag the dot in the middle of each arc) and the surroundings of each point can also 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.

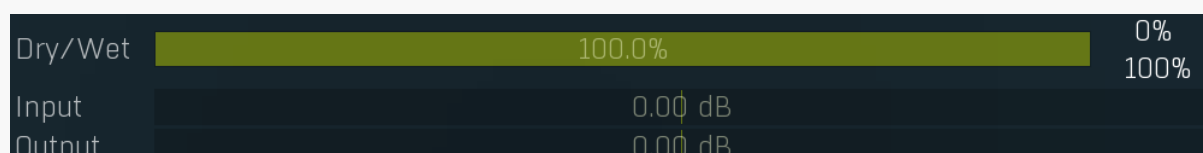- **Ctrl+A** selects all points. **Delete** deletes all selected points.

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.

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 | 100.0% | |

Dry/Wet

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

| 0% |

0%

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

| 100% |

100%

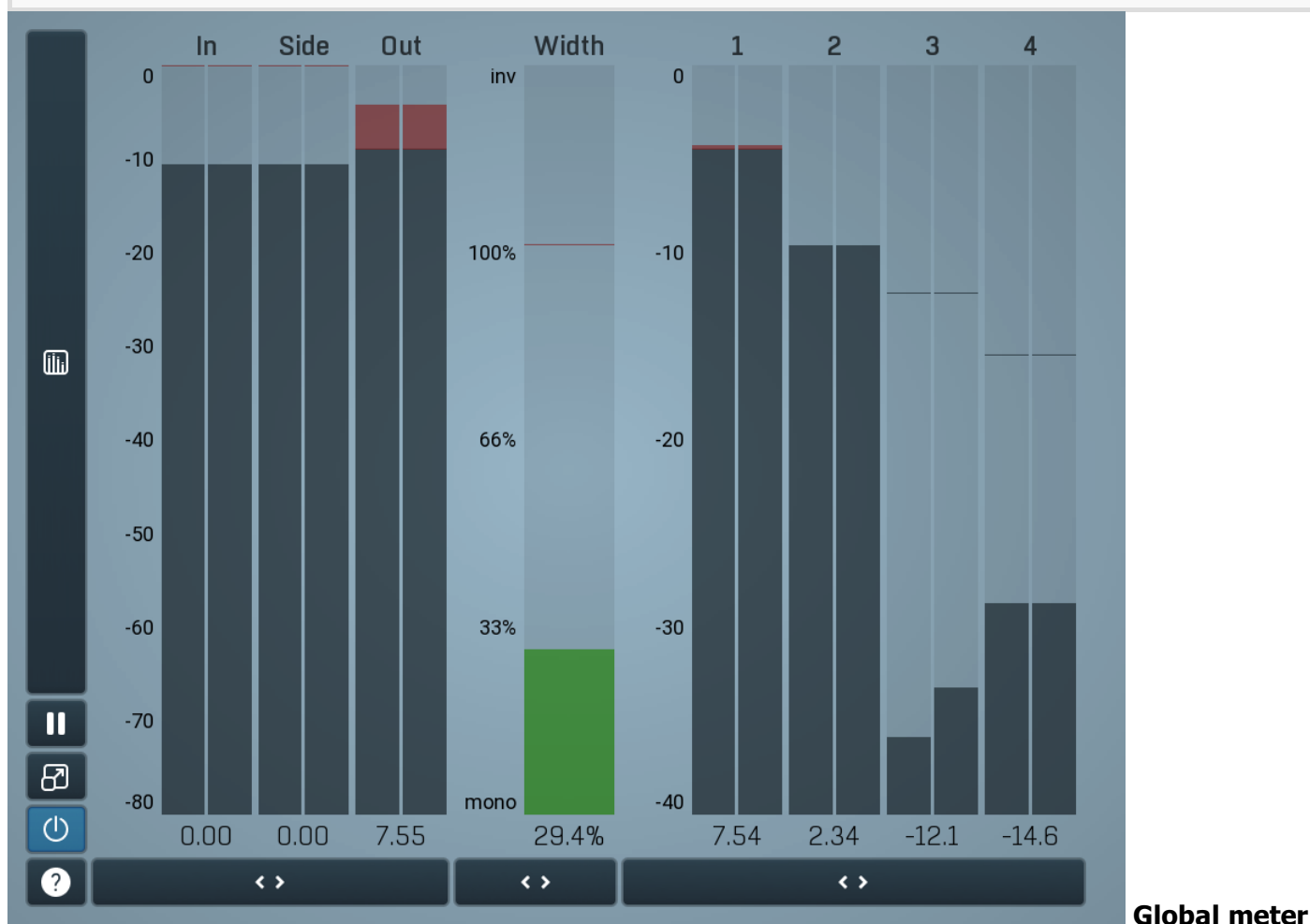100% button makes the **Dry/Wet** virtually 100%. You can use it for comparison.

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

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

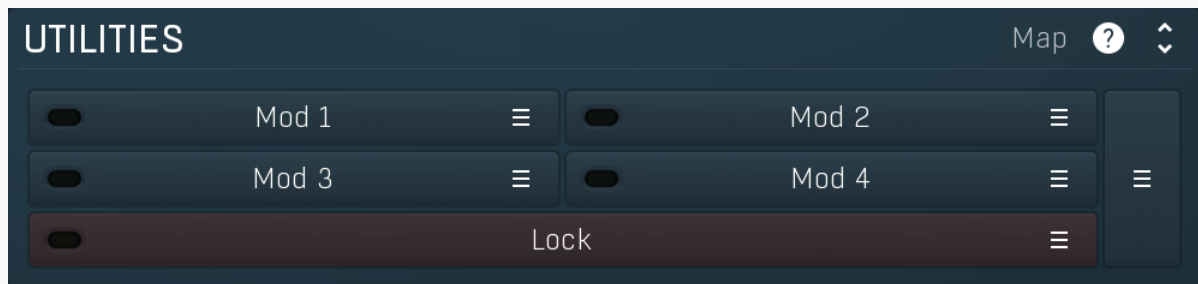Collapse button minimizes or enlarges the panel to release space for other editors.

Collapse

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

Utilities

Map ?

Mod 1 ☰   Mod 2 ☰

Mod 3 ☰   Mod 4 ☰ ☰

Lock ☰

Map Map

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

Modulator

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

☰ Menu

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

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

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

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

☰ Menu

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

Lock

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

Click on it using your right mouse button or use the **menu button** to display the menu with learning capabilities - **Learn** activates the learning mode, **Clear & Learn** deletes all currently-lockable parameters and then activates the learning mode. After that, every parameter you touch will be added to the lock. Learning mode is ended by clicking the button again.

The On/Off button built into the Lock button enables or disables the active locks.

Collapse

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

1 : Dry/wet   16.8% ☰

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.