MDynamicEq

Overview

Dynamics processors, such as compressors and expanders, dynamically manipulate the overall level of the audio material. Equalizers change the spectral character of the audio, statically. Multiband processors, such as MDynamicsMB, can do both, but they are often complex tools and can be potentially destructive if not used properly. MDynamicEq and MAutoDynamicEQ plugins represent a revolutionary bridge between both worlds. They manipulate the spectral character, and also react to input levels. Both plugins are very similar, but MAutoDynamicEq has some additional functionality. If a feature is different or not available in MDynamicEq, it will be noted within the description.

At first glance MAutoDynamicEq looks similar to our other parametric equalizers. However each band also has dynamic settings, often named dynamic gain as the main parameter is controlled by the vertical bar. Double-click on the bar to set it to its default 0dB value, which makes the band work as a normal static equalization band. Dragging the bar will change its value and the band will now start changing according to input levels.

By specifying a **negative dynamic gain**, for example -6dB, the band starts moving downwards. The higher the input level is, the lower the dynamic gain will be, but it will never exceed -6dB. Therefore this works like a compressor, but affecting only the part of the spectrum controlled by the band.

By selecting a **positive dynamic gain**, for example +6dB, the band starts moving upwards. The higher the input level is, the higher the dynamic gain will be, but it will never exceed +6dB. Therefore this works like an expander, but affecting only the part of the spectrum controlled by the band.

Another way to think about this is: a very low input level will be equalized by the Gain value for the band; a very high input level will be equalized by the (Gain + Dynamics Gain) value; and all levels in between.

The Band Settings

Each equalization band has a separate level processor, so this plugin is effectively many separate dynamic processors! The frequently-used settings are displayed / changeable in the equalization graph area on in the band list. If you do not see them, expand the Bands panel. All the settings for the band are on the Band Settings window, which is displayed by right clicking the band number on the equalization graph or in the band list.

Now, how is the level measured?

You can select the source to measure, the band signal or a side-chain, it can be filtered by the band filter or a configurable band-pass filter first, the responsiveness can be tailored by an envelope follower and it can be adjusted by a transformation curve.

Looking at the **Dynamics** panel: A side-chain input can be used with the band; just click the side-chain button to enable it.

Each band can react either to the input signal as set, or to a filtered signal (filtered by the same filter and settings as selected for the band). For example, if a low-shelf filter is used, that filter affects the low frequencies. It's a reasonable assumption (and it is set by default), that the measurement signal is first pre-filtered so that the level is measured from the signal that the band is actually affecting, in this case the bass. The low-shelf filter will therefore control the bass spectrum and will also be reacting to that same bass spectrum. The filtering mode is set from the **Mode** drop-down list.

Next there is an envelope follower similar to those found on dynamic processors. In this case it has just a few parameters to make things simple. Attack time and Release time parameters control the speed - how quickly the band reacts to the input. Both are set to **Auto** by default making it simulate an opto circuit. Both attack and release times can be adjusted from the band settings or in MAutoDynamicEq, these can be adjusted from the band list available below the equalization graph.

Each band also has an **Advanced Settings** window. To show it, hold **ctrl** and click a band with the right mouse button, or use the right mouse button in the band list or click the **Advanced** button in the band settings window dynamics panel.

The advanced settings contain a level detector band-pass pre-filter which can be used to make the band react to a particular part of the spectrum, a level transformation graph for special responses and some creative ideas, and a parameter called Shape, which is worth mentioning here. It defines how the band will react to input levels, basically how strong the response will be. Often the dynamic gain may be increased/decreased yet the band hardly moves. This is because the input level isn't high enough. The Level gain parameter (on the band settings window) can be used to increase the input level, but this may just not be enough and actually the response from different shapes is not the same. Selecting the **shape** can help. Typically the **Linear** shape is very weak and is suitable for detailed tweaking. **Squared** shape is usually ideal, and it is a compromise between linear and logarithmic, which is the strongest.

Examples

There are many things that can be done with dynamic equalizers. We will skip the static features that can be performed with normal parametric equalizers and show a few examples demonstrating the dynamic capabilities.

Enhancing low-end

There are many ways to enhance the bass part of the spectrum, using saturation for example, which is also present in both equalizers. But we'll demonstrate a more ambitious way. The bass spectrum is the easiest to overcrowd, because each frequency has a very long period compared to the higher frequencies and the difference between the frequencies of neighbouring semitones is very small. Therefore it should be kept as clean as possible.

First create a low shelf band with quite a low Q, a nice smooth shape with no resonance. Place it at 1kHz and set the dynamic gain to +6dB. This will now work like a smooth low-frequency expander, gently enhancing the bass spectrum. When there is a signal in the bass, it will be amplified even more, and when there's not much bass it will keep it intact, hence providing space in the mix. This can actually be done with any part of the spectrum, not just bass.

De-essing

When singing, "S" sounds usually produce more direct air-flow, which is then captured by the singer's microphone. As their volume is generally much louder, one of the engineer's tasks is to lower/attenuate them. This can be done using a single band compressor with a band-pass in the side-chain, but this often produces unnecessary or unwanted pumping and can even change the character of the singer's voice. Multiband compressors can provide better transparency, but they may be hard to setup. The same thing can be done more easily, with better quality and in no time using dynamic equalizers.

Sibilance is located somewhere above 2kHz depending on the vocalist. The exact location can easily be found using the analyzer, sonogram, or using the auto-listen feature. Place a band at the center of the sibilance, and make it a peak filter. Set gain to 0dB (this can be set lower, but doing so may change the character of the voice). Use dynamic gain and set it to about -12dB. Then when a burst of loud sibilance occurs at that frequency, it gets attenuated. Low levels of sibilance are untouched. Additionally you can set the **Threshold** value in the band setting window such that attenuation only occurs above the specified level.

Avoiding mud between bass and bass drum

Instruments occupying lower frequencies often muddy the mix making both elements hard to recognize. Engineers use equalizers, multiband compressors and sometimes specialized tools to fix this problem. Dynamic equalizers also provide a very effective way to tackle it. The idea is that since both instruments cannot coexist in the same part of the spectrum, we have to remove a part of one of them. Let's say we attenuate everything below 100Hz from bass. This can be done with a standard equalizer, but then the bass will be weak, even when the bass drum is not playing! Here is a better way to resolve this problems.

First send the bass drum into the plugin's side-chain in your host. Then create a low-shelf band, open its band settings and set it to react to the side-chain by pressing the Side-chain button. Set its gain to 0dB as we do not want the signal processing constantly, only if the drum is actually playing. Finally set its dynamic gain to around -24dB. Then when the bass drum starts playing, the band will lower its gain attenuating that part of the bass track and providing space for the bass drum.

Automatic equalization

Please note that this feature is available only in MAutoDynamicEq! If you do not see the buttons, expand the **Automatic equalizer** section.

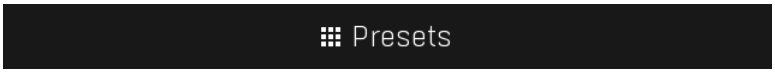
To get the plugin to generate the equalizer settings for you, follow these steps:

1. Analyse your recording - start playback and press Analyse target button. Most of the graphs in the spectral view will disappear and a **green line**, depicting a long term analysis, will be displayed. It will eventually stop moving, which usually means that the analysis is finished.

2. Get a source analysis - you can either load a predefined analysis using the Load button, or analyse another recording using Analyse source button (by the same method used to analyse the target), draw the requested frequency response using Draw button or even analyze an audio file using File button. You can also analyse your whole mix and let the plugin help fit the problem track into it, and make the mix clearer, less muddy. See Separate button for more information.

3. Click Equalize or Separate button and the plugin will adjust the bands. Equalize sets the bands to make the source sound much more like the target; Separate aims to reduce frequency collisions at high levels between the two.

You should notice the Smoothness parameter, which spreads the energy in the spectral view. It makes the analysis easier to understand visually, but it also affects the automatic equalization as well. A higher smoothness setting typically provides more natural results. Adjust the smoothness, click Equalize or Separate again and check if there is an improvement. Additionally, remember that the **Dry/wet** level controls the depth of processing of the audio - those levels are reflected in the equalization graph. You may wish to drop the level towards 0% (unaffected source) and increase it until the audio sounds "enough like" the target.



Presets

Presets button shows a window with all available presets. A preset can be loaded from the preset window by double-clicking on it, using the arrow buttons or by using a combination of the arrow keys and Enter on 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.

Holding **Ctrl** while pressing the button loads an existing preset, selected at random.

Presets can be backed up by using either the Export button, or by saving the actual preset files, which are found in the following directories:
Windows: C:\Users\{username}\AppData\Roaming\MeldaProduction
Mac OS X: ~/Library/Application support/MeldaProduction

Exported preset files can be loaded into the plug-in's preset store using the Import button. Or the preset files themselves can be copied into the directories named above.

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

Left arrow

Left arrow button loads the previous preset.

Right arrow

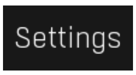
Right arrow button loads the next preset.

Randomize

Randomize button loads a random preset.

Panic

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



Settings

Settings button shows a menu with additional settings of the plugin. Here is a brief description of the separate items.

Activate lets you activate the plugin if the drag & drop activation method does not work in your host. In this case either click this button and browse to the licence file on your computer and select it. Or open the licence file in any text editor, copy its contents to the system clipboard and click this button. The plugin will then perform the activation using the data in the clipboard, if possible.

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.

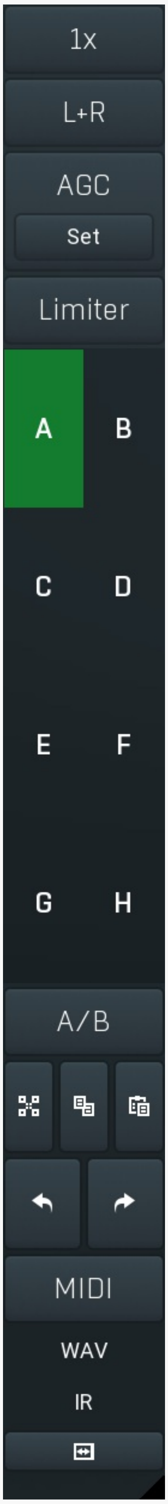
Dry/wet affects determines, for Multiband plug-ins, which multiband parameters are affected by the Global dry/wet control.

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.

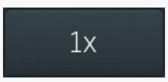
WWW

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.

Plugin toolbar



Plugin toolbar provides some global features, A-H presets and more.



Upsampling

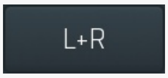
Upsampling can potentially improve sound quality by processing at a higher sample rate. Processors such as compressors, saturators, distortions etc., which employ nonlinear processing generate higher harmonics of the existing frequencies. If these frequencies exceed the Nyquist rate, which equals half 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. Upsampling (or 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 upsampling 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 upsampling even they become highly aliased. Now select 16x upsampling and listen to the difference. If you again select 1x upsampling, 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 upsampling, 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 upsampling. 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 upsampling 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.

Upsampling 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 mode in the plugin settings), which is not very usable in real time applications. Secondly, upsampling also takes much more CPU power, due to both the processing being performed at a higher sampling rate (for 16x upsampling at 44100 Hz, this equates to 706 kHz!), and the complex filtering. Finally, and most importantly, upsampling 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 upsampling 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 upsampling to some heavily distorting processors.



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.

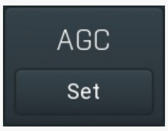
There are also 4 subsidiary modes: **Left & zero Right (L(R0))**, **Right & zero Left (R(L0))**, **Mid & zero Side (M(S0))** and **Side & zero Mid (S(M0))**. Each of these processes one channel and silences the other.

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.



AGC

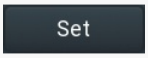
AGC button enables or disables the automatic gain control - the automatic adjustment of the output volume such that it matches the input volume. Human hearing is very adaptable. In fact differences in loudness, for example when loading a preset, may go unnoticed and instead be perceived by the listener as "better sounding", leading to a misjudgement. This feature should prevent this effect, thus allowing the listener to focus on the sonic qualities only.

AGC works by measuring input and output loudness, and then compensating for the difference while also taking into account any induced latency. The loudness measurement follows the ITU and EBU specifications with an RMS of 400ms, meaning that the reaction time is 400ms. This is very important, as you should be aware that AGC needs time to properly adjust after any change of settings. Also note that this is a nonlinear operation. It may cause some distortion due to the long measurement time. It should be negligible though.

AGC makes sense in most applications including reverberation and equalization for example. However, in some cases it can work against the plugin. A simple example of this is a tremolo, where the plugin manipulates output volume. If the tremolo rate is slow enough, say 1Hz, it makes the period longer than the actual AGC measurement time. So whenever the tremolo changes audio level, the AGC starts compensating for it. This can of course be used creatively, since AGC will always be a little "late", but it is definitely not a desired outcome in normal use.

Another example of this is compression. When used with short attack and release times, AGC can effectively compensate for the attenuation of the compressor. However when the attack and release times are higher than 100ms, the compressor's reaction time becomes too slow, and in conjunction with AGC, severe pumping can occur.

As a general rule of thumb as for all audio processing tasks, use it only if you know you need it. AGC is a powerful tool that can make your workflow easier, but it can also be damaging.



Set

Set button uses the AGC (automatic gain compensation) processor to calculate the ideal output gain to ensure that the output audio loudness is equal to the input level. To use it, simply enable playback in your host and click the button. The plugin's output gain will be adjusted to match the input and output levels as closely as possible.
If the AGC is already enabled, the change will be instant and you can disable the AGC afterwards. Typically you will browse presets, generate random settings etc. During the entire time you will have AGC enabled to prevent you from experiencing different output loudness levels. When you find a sonically ideal setup, you simply click the Set button to set the output gain automatically and disable the AGC as you won't need it anymore.
If the AGC is not already enabled, clicking the Set button displays a window with progress bar for a few seconds, while the plugin temporarily enables AGC and analyses input and output of the plugin. After that the AGC is disabled again.

To get the best results, you should feed the plugin with some "universal" signal. If you are processing a specific instrument, play a typical part, a chorus in case of vocals for example. If you are creating presets designed for general use, white/pink noise may be the best signal to use.

Limiter

Limiter button enables or disables the safety limiter. Its purpose is to protect you from peaks above 0dB, which can have damaging effects to your processing chain, your monitors and even your hearing.

It is generally advised to keep your audio below 0dB at all times in all stages of your processing chain. However, several plugins may cause high level outputs with certain settings, often due to unprevented resonances with specific audio materials. The safety limiter prevents that.

Note that it is NOT wise to enable this "just in case". As with any processing, the limiter requires additional processing power and modifies the output signal. It is a transparent single-band brickwall limiter, but you still need to be careful when using it.

A-H presets selector

A-H presets selector controls the current A-H preset. This allows the plugin to store up to 8 sets of settings, including those parameters that cannot be automated or modulated. However it does not include channel mode, upsampling and potentially some other global controls available from the Settings/Settings menu.

For example, this feature can be used to keep multiple settings, when you are not sure about the ideal configuration When you change any parameter, only the currently selected preset is modified.

The four buttons below enable you to switch between the last 2 selected sets using the A/B button, morph between the first 4 sets using the morphing button and copy & paste settings from one preset to another (via the clipboard).

It is also possible to switch between the presets using MIDI program change messages sent from your host. The set selected depends on the Program Change number: 0 selects A, 7 selects H, 8 selects A, 15 selects H and so on.

A/B

A/B button switches between the active and previously active A-H preset (not necessarily the A and B presets themselves). To compare any 2 of the A-H presets, select one and then the other. Clicking this button will then switch between these two. You can do the same thing by clicking on the particular presets, but this makes it easier, letting you close your eyes and just listen.

Morph

Morph button lets you morph between the A, B, C and D settings. Morphing only affects those parameters that can be automated or modulated; that does include most of the parameters however. When you click this button, an X/Y graph is shown allowing you to drag the position indicator to any position between the letters A, B, C and D. The closer you drag the indicator to one of the letters, the closer the actual settings are to that preset.

Please note that this will overwrite and change the preset that is currently selected, so it is best to select a new preset e.g. 'E', then use the morphing method. This way you will define the settings for A, B,C and D, morph between them, and store the result in 'E' without any modification of the original A, B, C and D presets.

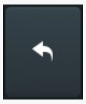
Please note that the ABCD morphing itself cannot be automated and that, while morphing, the changes to the underlying parameters are not notified to the host (there may be hundreds of change events).

Copy

Copy button copies the current settings to the system clipboard. Other presets, upsampling, channel mode and other global settings are not copied.
Hold **Ctrl** to save the settings as a file instead. That may be necessary for complex settings, which may be too long for system clipboard to handle. It may also be advantageous when you want to send the settings via email. You can load the settings by drag & dropping them to a plugin or holding **Ctrl** and clicking **Paste**.

Paste

Paste button pastes settings from the system clipboard into the current preset. Hold **Ctrl** to load the settings from a file instead. Hold **Shift** to paste the settings to all of the A-H slots at once.



Undo

Undo button reverts the last change. Only changes to automatable or modulatable parameters and global settings (load/randomize) are stored.

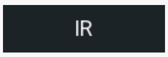
Redo

Redo button reverts the last undo operation.

WAV

WAV button lets you process a file using the plugin with current settings. You can either click the button and select a file, or drag & drop the file (or multiple files) onto the button. If you let the plugin process WAV files, these will be saved with the original settings. If you use a different file type (such as MP3), the plugin will create WAV files with 32-bit bits-per-sample floating point.

Please note that the files will be overwritten, so make a copy first if you want to keep the original.



IR

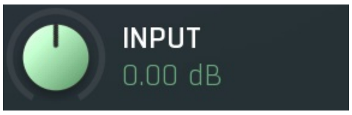
IR button lets you generate impulse response file, which approximates what the plugin does. You can use that in various IR players, including some hardware. Please note that any dynamic/modulated/somehow changing in time behaviour cannot be captured by an IR file.

Collapse

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

Dry/wet

Dry/wet defines ratio between dry and wet signals. 100% means fully processed, 0% means no processing at all. In normal mode only peak and shelf filters are affected correctly, other filters are left at 100% unless the ratio is set to 0%, in which case the equalizer is bypassed.
Range: 0.00% to 100.0%, default 100.0%

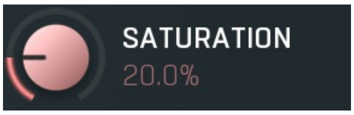


Input gain

Input gain defines input gain applied before the equalization. Therefore this affects all dynamics-based processes.
Range: -24.00 dB to +24.00 dB, default 0.00 dB

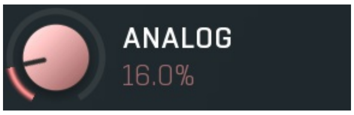
Output gain

Output gain defines output gain applied after the equalization. Please note that the real output gain is affected by dry/wet parameter, as opposed to input gain, which is not, because it affects the resulting sound.
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%

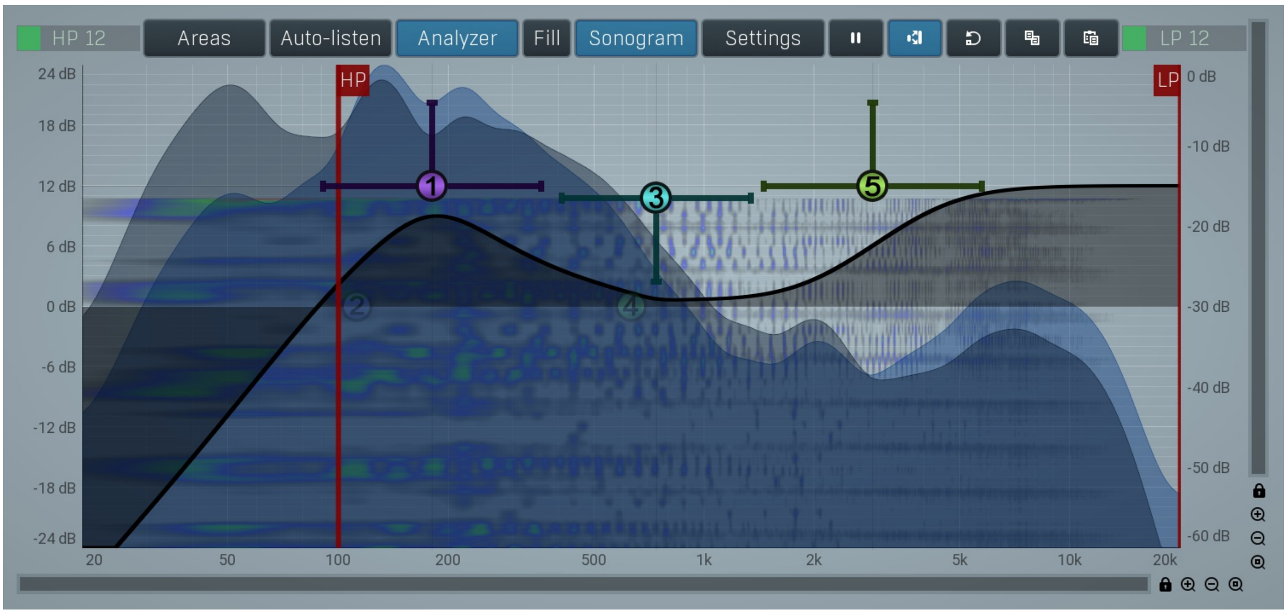


Analog

Analog controls the amount of internal nonlinearities in each filter, typical for analog equalizers. Note that this processing also changes the actual filter shapes, so they won't fully match the displayed graphs anymore when Analog feature is used. It can also require a solid amount of CPU power.
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%



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.

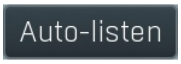
Double-click on a band point to enable or disable a band. Drag it to change its frequency and gain. Drag the horizontal nodes to change its Q. Hold **ctrl** key for fine tuning. Click using the right mouse button on it to open a window with additional settings.

The equalizer graph also contains 2 **red vertical lines** on the right and on the left. These are the high-pass and low-pass filters, conveniently placed so that you can perform this often-used task quickly, efficiently and most of all using the highest quality filters available on the market. The high-pass filter also serves as a DC blocker. Slopes for both filters can be adjusted in the title area of the equalizer graph panel.

Typically you want to remove the low frequencies (high-pass filter via the left line) from just about any audio material except for bass and bass drum. Even if the frequencies are not there and are not shown in the sonogram or analyzer views, you may still want to apply a high-pass filter to let the equalization remove any potential low frequency rumble the track might contain. This is always a good practice to clear the resulting mix.

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.



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.

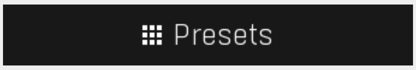
Sonogram

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.

Settings

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

Analyzer settings



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.

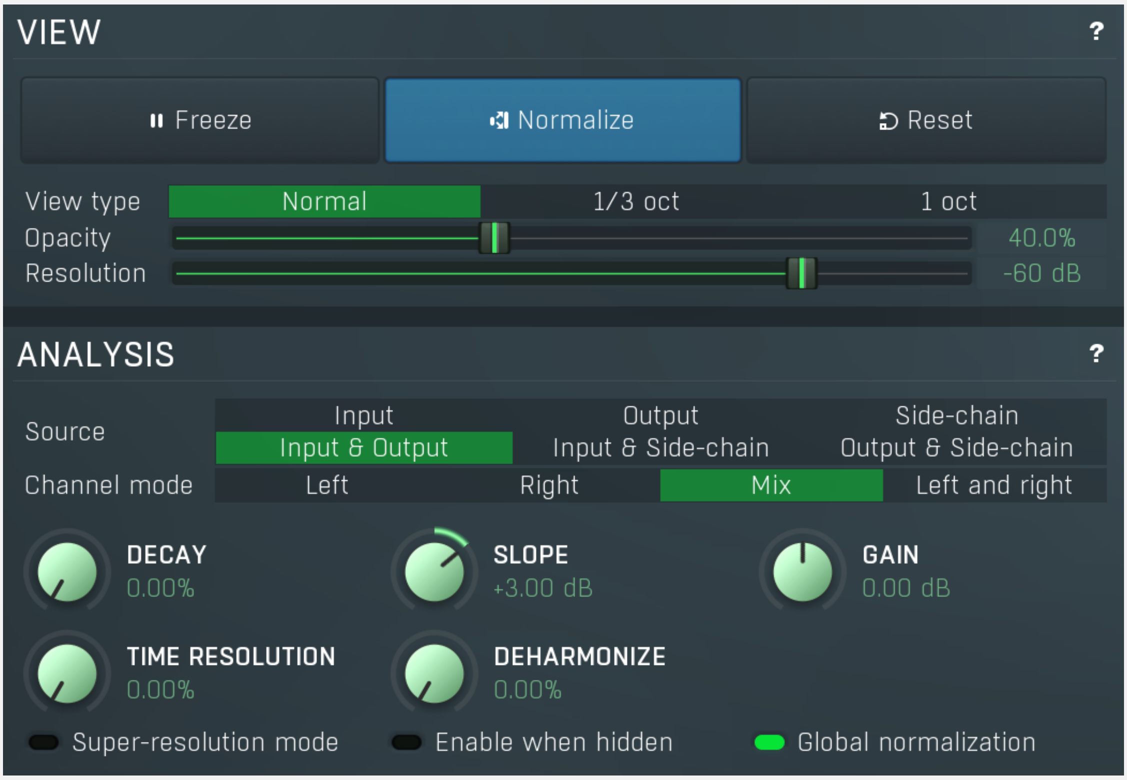
Paste

Paste button loads the settings from the system clipboard.

Tab selector

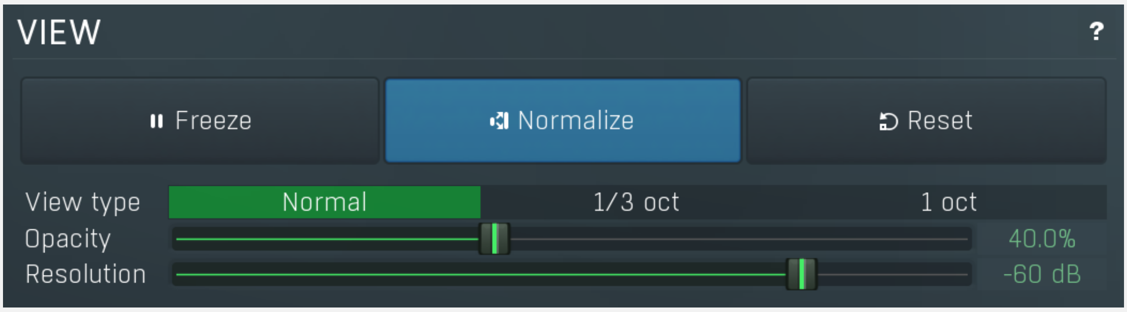
Tab selector switches between subsections.

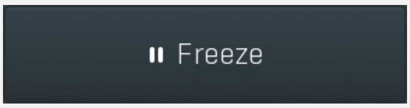
Main settings panel



Main settings panel contains the most useful settings controlling the analyzer behaviour and view.

View





Freeze

Freeze button stops processing 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.

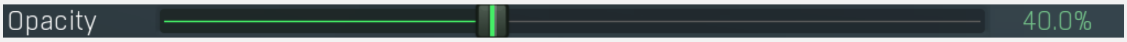
When comparing 2 spectrums you are usually interested mainly in the frequency level differences. In most cases both audio materials will have different overall levels, which would mean that one of the graphs would be "lower" than the other, making the comparison quite difficult. Normalize fixes this 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 the analyzer state. This is particularly useful when analyzing infinite average and maximum values.

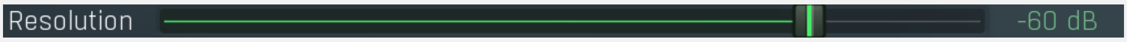
View type

View type controls the way the spectrum is displayed. By default a smooth curve is presented. This view provides the best resolution and detail, but other modes (1/3 octave, 1 octave) may be easier to read.



Opacity

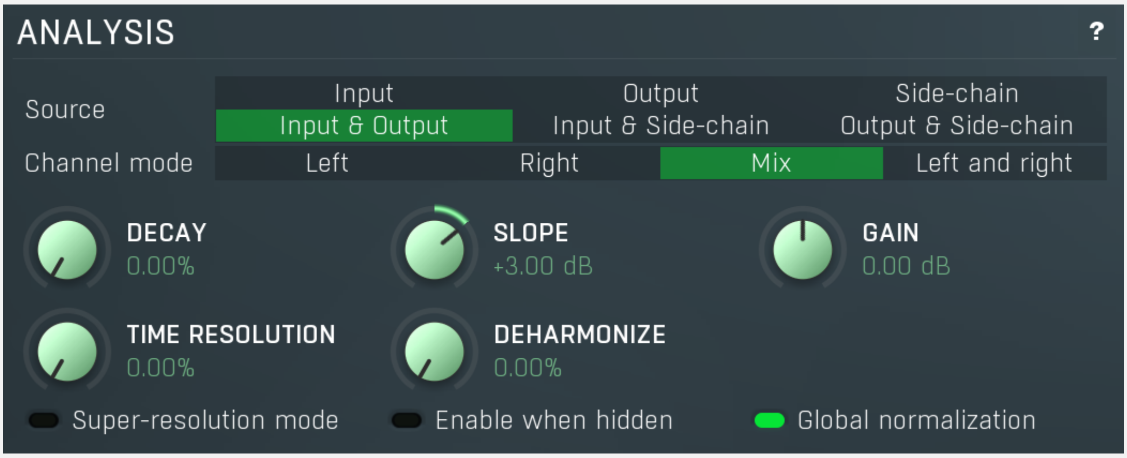
Opacity controls the opacity of all analyzer graphs.



Resolution

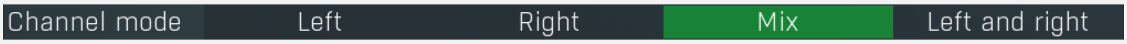
Resolution defines the vertical range on the display. The human auditory system has a resolution of about 90dB and the relevant range is usually less than 60dB. However you may want to use a higher resolution to check for technical problems - aliasing, distortion etc.

Analysis



Source mode

Source mode defines which audio stages are to be analyzed. By default both input & output are selected and analyzed. However you may want to analyze only the input, or the output (or the external side-chain, where available, on its own or with the input or output).

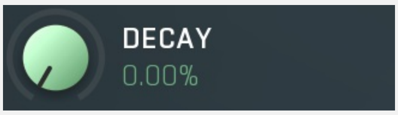


Channel mode

Channel mode defines which channels are to be analyzed. By default all channels are merged into a mono sum (Mix mode), which is then analyzed. However you may want to analyze separate channels or display both the left and right channels separately. Please note that if two channels (for example: input & output, or input & side-chain) are displayed at the same time then mix mode is used instead of left & right mode. Similarly, when the plug-in is in Surround mode then Mix mode is used.

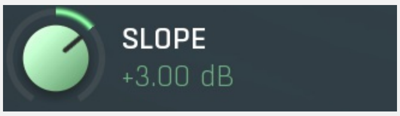
Also please note that when the plug-in is in one of the Mid / Side modes of operation, then you should read 'Left' as 'Mid' and 'Right' as 'Side'.

Different analyser combinations can, of course, be saved as different named presets.



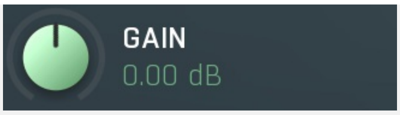
Decay

Decay controls the speed at which the magnitudes return to the minimum value (silence). It is an alternative to averaging, which affects the speed that the frequencies both gain and lose their magnitudes. With a decay of 0% the magnitude goes to the minimum immediately. With 100% it stays the same forever, so it makes it display the maximum.



Slope

Slope makes the analyser increase the magnitude of higher frequencies, since they are typically lower in energy. 3dB per octave is a typical value, which makes pink noise horizontal as pink noise contains equal energy in each octave. Therefore if you set slope to 3dB, the response would be the same for the FFT and 1/3 octave graphs.

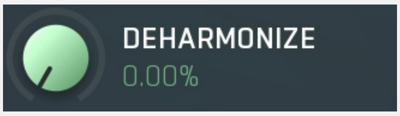


Gain

Gain makes all frequencies change magnitude by the specified amount. This has no meaning when normalization is enabled.

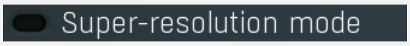
Time resolution

Time resolution improves the time resolution, but lowers the spectral resolution. This is typically useful for more scientific analyses, where the signal is moving quickly and you need to follow its movements quickly. This is often advantageous for sonograms with very high FFT sizes.



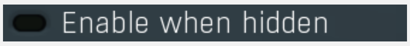
Deharmonize

Deharmonize tries to remove harmonics in the content and leave only fundamentals. This may help you find the dominant frequencies in the signal.



Super-resolution mode

Super-resolution mode activates a special processing algorithm, which provides high resolution even in the low frequency spectrum. Using standard FFT algorithms you can increase the FFT size to get better bass resolution, but this also slows down the response. Super-resolution mode keeps the quick response in high frequencies as they are naturally quicker, but also highly enhances the bass spectrum resolution. It requires additional CPU power.



Enable when hidden

Enable when hidden causes the analysis engine to continue processing the signal even when the GUI is hidden. Otherwise the sonogram is stopped, therefore will not be immediately available when the GUI is shown again.

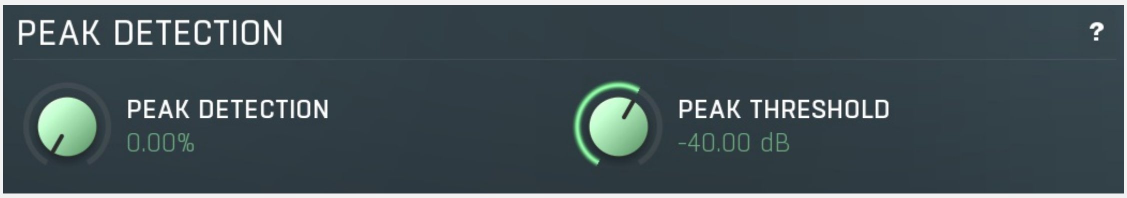
Global normalization

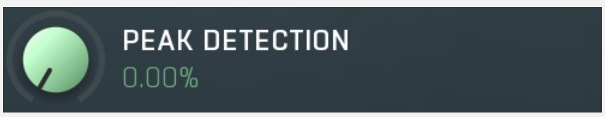
Global normalization makes the **normalization** work based on the maximum of all graphs visible at the time. This means that the levels between the graphs will stay the same, but the maximum level will be 0dB. This is useful for comparing relative levels. If you disable this, all graphs will be normalized separately and will touch 0dB unless they are silent; and this is useful for comparing spectra.

Advanced panel

Advanced panel contains more advanced settings controlling the scientific parameters of the audio analysis.

Peak detection





Peak detection

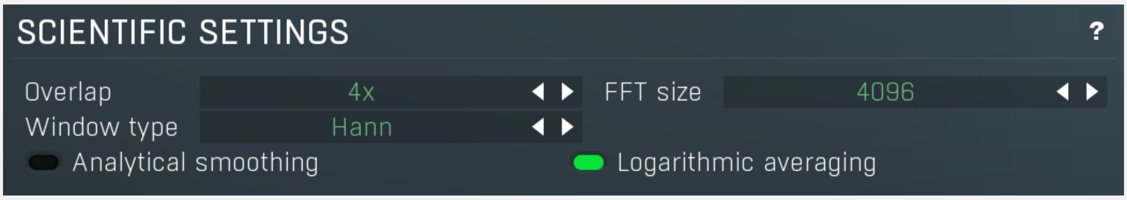
Peak detection tries to the remove skirts of separate sinusoids letting you view the frequencies contained in your audio material.

This may be handy when performing more scientific analyses.

Peak threshold

Peak threshold defines the level below the maximum which is used for peak detection. You can use this to control which peaks get through and to get rid of small insignificant ones.

Scientific settings



Overlapping

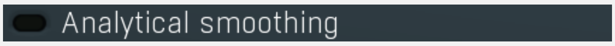
Overlapping makes the analyser perform multiple FFT processing on the same data which results in better precision at the cost of higher CPU impact. With higher overlapping the response also speeds up.

FFT size

FFT size defines FFT processing block size. It basically controls the resolution. However for higher resolution in bass content it is recommended to use super-resolution mode instead as it keeps the quick response in higher frequencies.

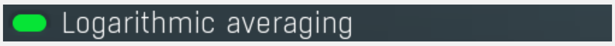
Window type

Window type defines the type of window used to pre-process the source samples. This has several consequences for the frequency response, but it is a little scientific parameter. If you do not have specific requirements you can just leave this set to its default.



Analytical smoothing

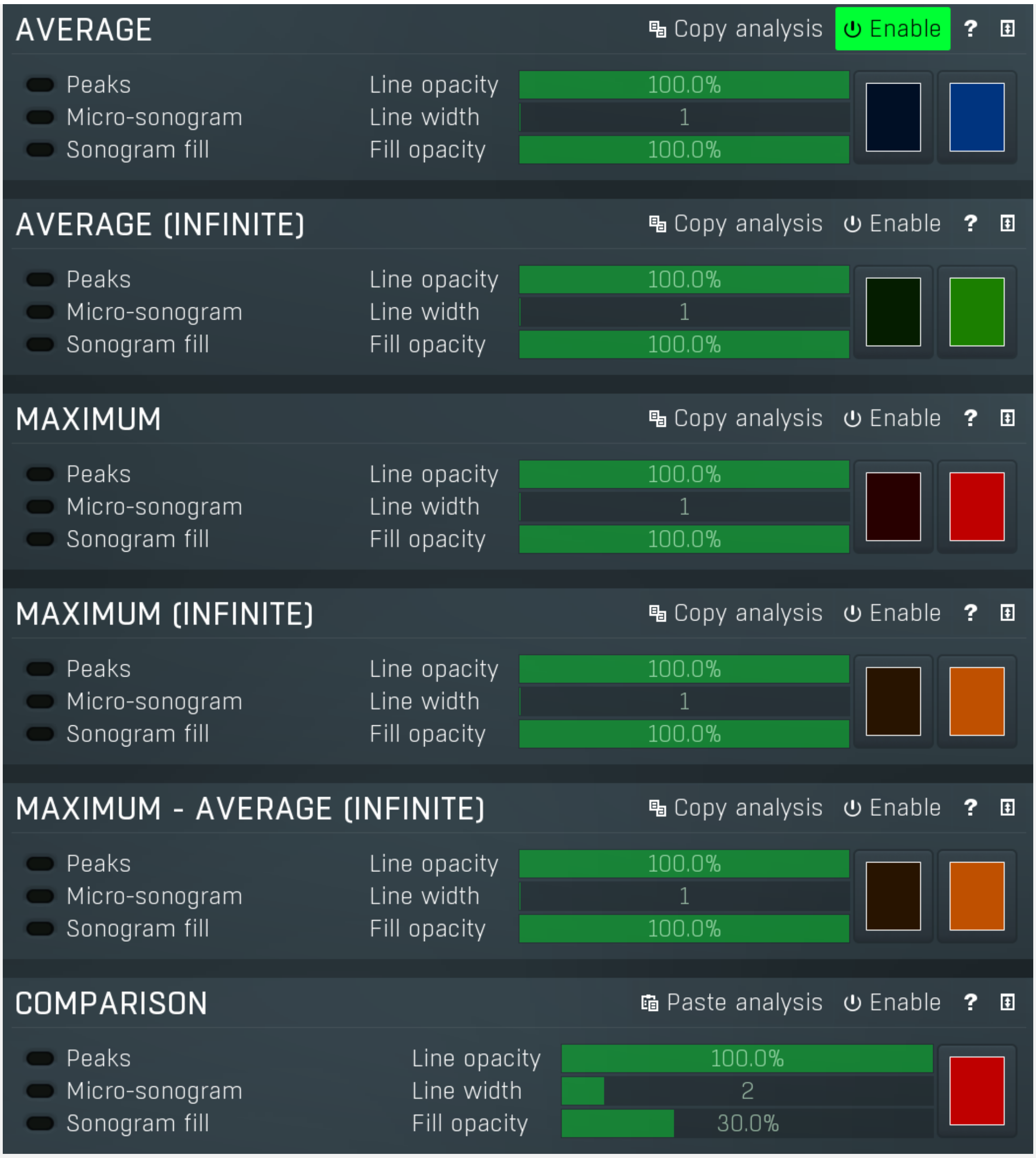
Analytical smoothing switch activates a more complicated smoothing algorithm, which provides more accurate results, however it may require much more CPU power. Unlike normal smoothing this method doesn't change the proportions of frequencies with higher magnitudes. It is useful mostly for technical analysis and for most musical signals it is often better to use the default smoothing method.



Logarithmic averaging

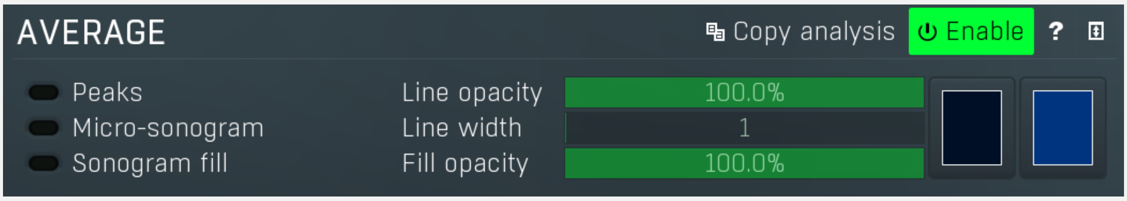
Logarithmic averaging switch activates averaging in logarithmic mode, hence decibels. If you disable it, linear averaging will be used.

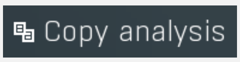
Graphs panel



Graphs panel contains visual settings for the different graphs that you can show in the analyzer.

Average





Copy analysis

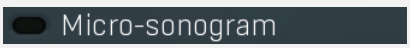
Copy analysis button copies the current state of the analysis into the system clipboard so that you can paste it into another analyzer for comparison. Hold **ctrl** to export the analysis into a CSV file.

Peaks

Peaks enables detection of frequencies with the highest magnitudes. Frequencies which are at most 20dB lower than the maximum are displayed, and there may be at most 8 of them. Please note that this feature requires additional CPU power.

Line opacity

Line opacity controls the opacity of the graph outline.



Micro-sonogram

Micro-sonogram displays a small single-state sonogram at the bottom of the graph. This may help you compare relevant frequencies, because it is usually easier to compare colors than graph values.

Line width

Line width controls the width of the graph online.

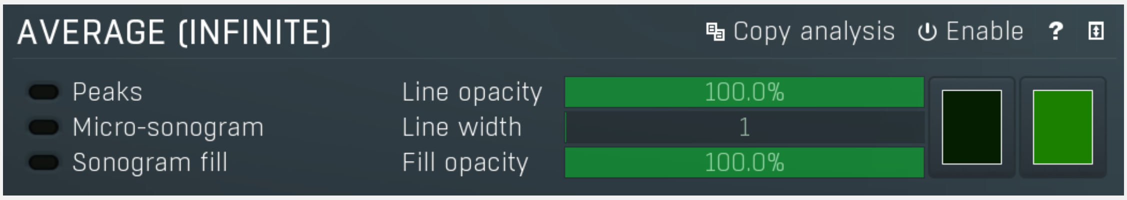
Fill

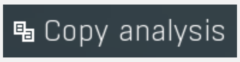
Fill makes the sonogram (enabled by **Show sonogram**) fill the whole area.

Fill opacity

Fill opacity controls the opacity of the graph interior fill.

Average (infinite)

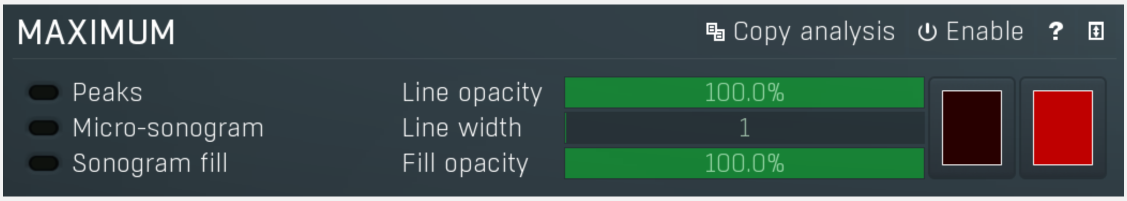


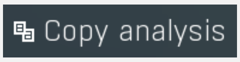


Copy analysis

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.

Maximum

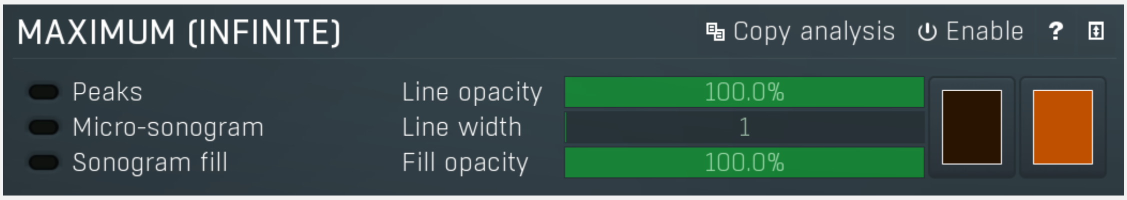


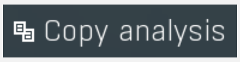


Copy analysis

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.

Maximum (infinite)

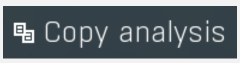




Copy analysis

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.

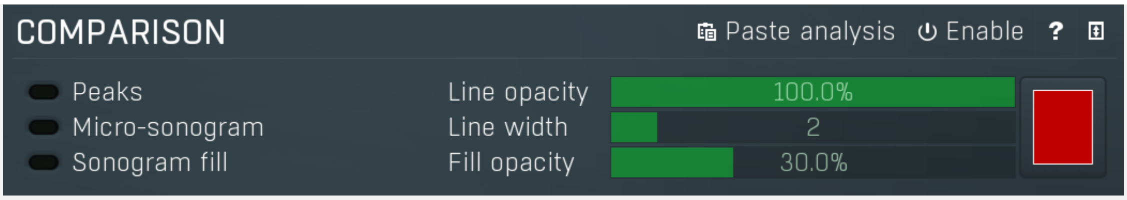
Maximum - Average (infinite)

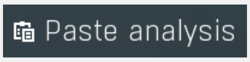


Copy analysis

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.

Comparison



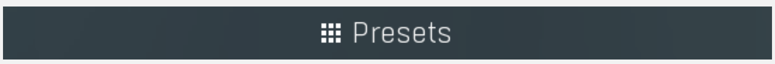


Paste analysis

Paste analysis button pastes an analysis from the system clipboard and displays it as a comparison. This way you can compare your analysis to any other analysis from MeldaProduction plugins.

Sonogram panel

Sonogram panel contains visual settings of the sonogram, mainly the sonogram colors. A sonogram uses a set of colors. When the particular frequency's level is at the minimum, the first color is used. When it is at the maximum, the last color is used. Otherwise it interpolates the colors in-between.



Presets

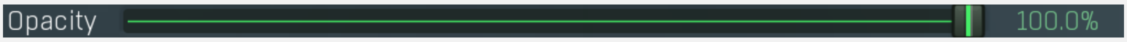
Presets button displays a window where you can load and manage available presets. Hold **Ctrl** when clicking to load a random preset instead.

Left arrow

Left arrow button loads the previous preset.

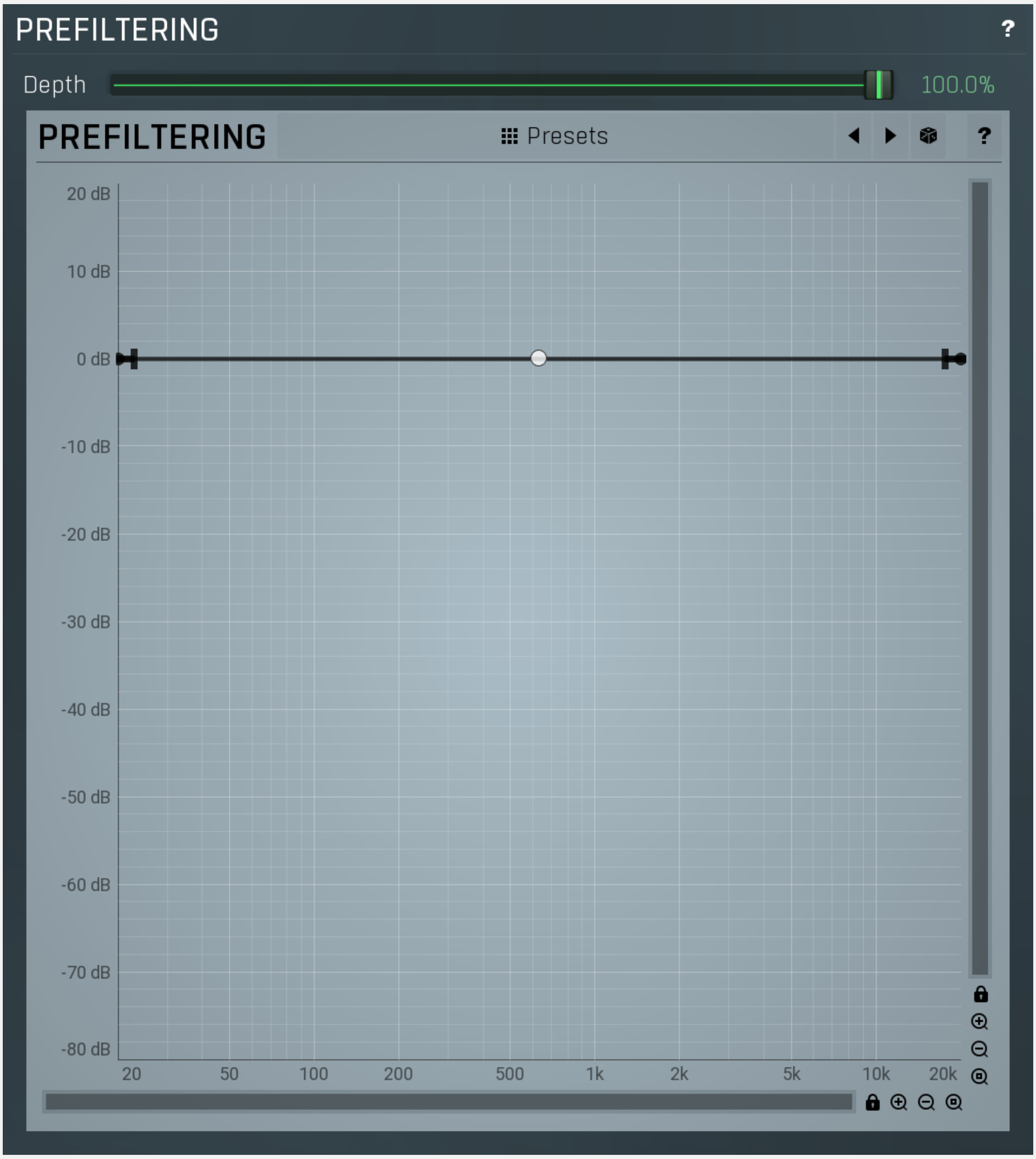
Right arrow
Right arrow button loads the next preset.

Randomize
Randomize button loads a random preset.

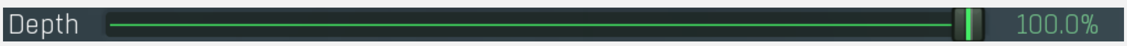


Opacity
Opacity controls the opacity of the sonogram.

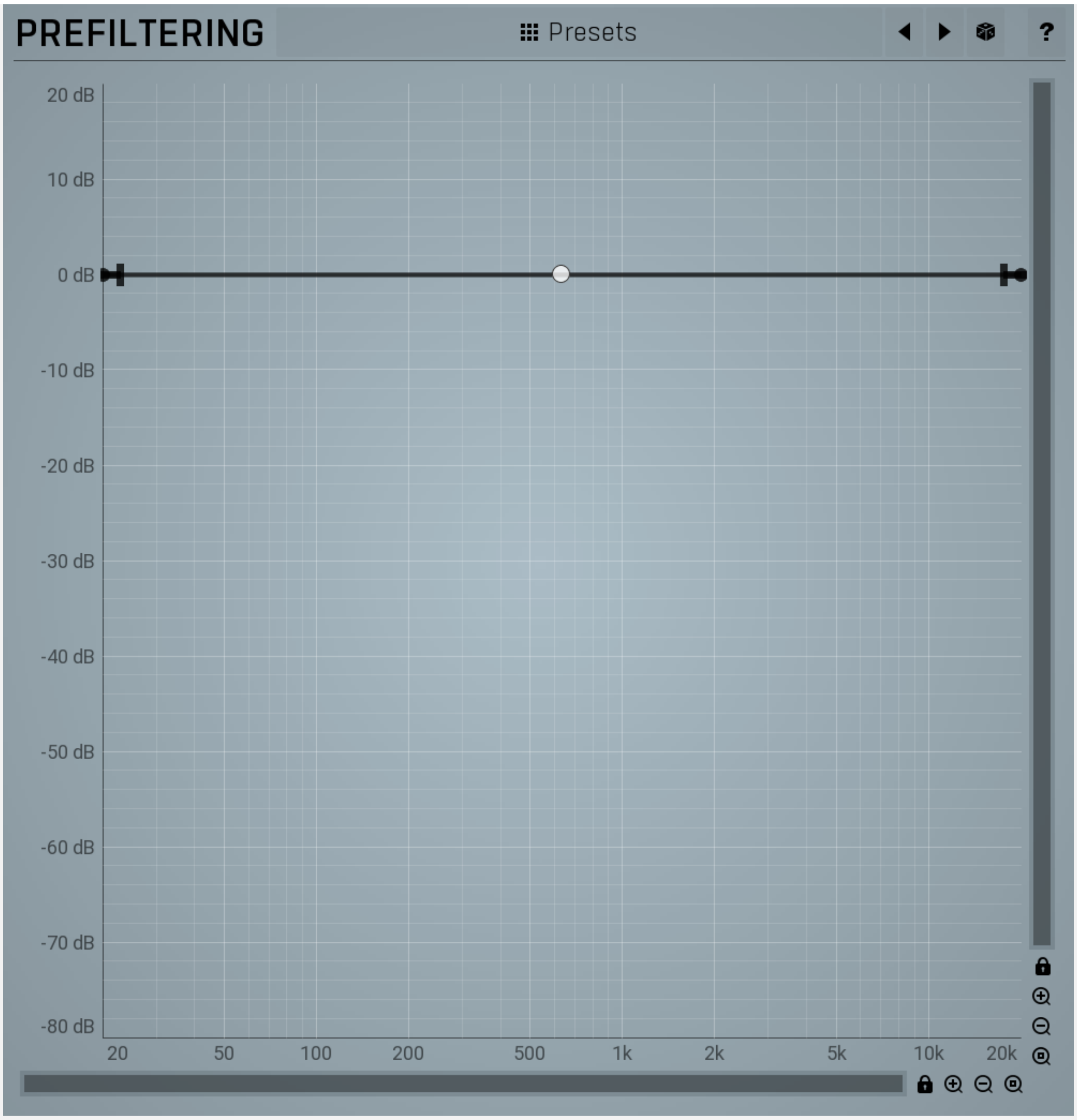
Prefiltering panel



Prefiltering panel provides the optional prefiltering, which means that level of each frequency is either increased or decreased before analysis. Normally the analyzer shows scientific levels of each frequency. However you can for example use the predefined loudness curves, which makes the analyzer show how the human auditory system responds to the frequencies, so it in fact provides more accurate analysis taking into account the fact that human hearing is more complicated than the mathematical model.



Depth
Depth controls the amount of prefiltering. 100% makes the analyzer follow the prefiltering graph precisely, 0% essentially disables this feature.

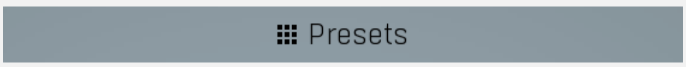


Prefiltering

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.



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.

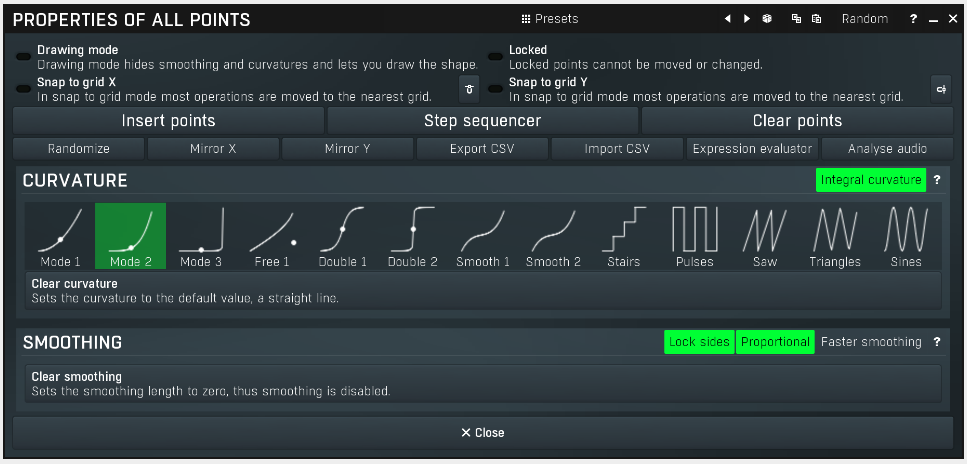
Right arrow

Right arrow button loads the next preset.

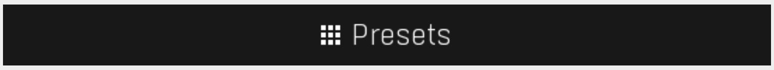
Randomize

Randomize button loads a random preset.

Envelope graph menu



Envelope graph menu provides additional features which are used to edit the graph. Open the menu using right mouse button in the graph. Please note that if you select some points in the graph, or click on a point for example, the menu will be different and will cover only those features related to the selected set of points.



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

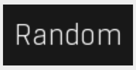
Randomize button loads a random preset.

Copy

Copy button copies the settings onto the system clipboard.

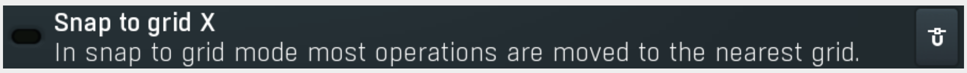
Paste

Paste button loads the settings from the system clipboard.



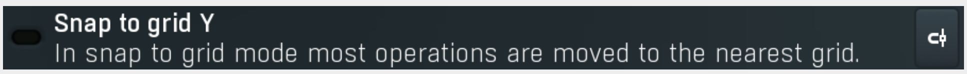
Random

Random button generates random settings using the existing presets.



Snap to grid X

Snap to grid X activates the snap to grid feature. Alternatively you can press **Alt** while dragging a point or selection.



Snap

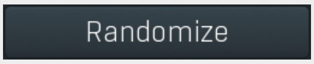
Snap button activates the snap to grid feature. Alternatively you can press **Alt** while dragging a point or selection.

Insert point

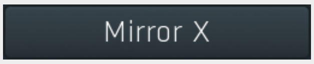
Insert point button creates a point at mouse position.

Step sequencer
Step sequencer button generates the envelope from step sequencer.

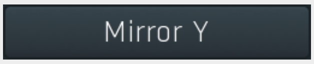
Clear points
Clear points button deletes all points.



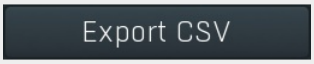
Randomize
Randomize button slightly modifies the Y coordinates.



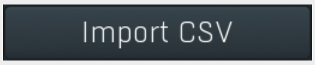
Mirror X
Mirror X button inverts the X coordinates of all points.



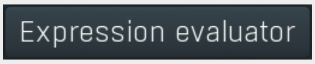
Mirror Y
Mirror Y button inverts the Y coordinates of all points.



Export CSV
Export CSV feature lets you export the graph to a CSV file. CSV file is a simple text format, which has multiple lines with X and Y coordinates delimited by ';'. For example:
0.275;0.2
0.438;0.5
0.775;0.67



Import CSV
Import CSV feature lets you select a CSV file and imports the graph points from it. CSV file is a simple text format, which has multiple lines with X and Y coordinates delimited by ';'. For example:
0.275;0.2
0.438;0.5
0.775;0.67

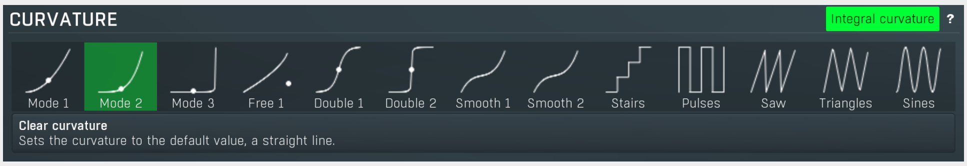


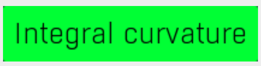
Expression evaluator
Expression evaluator lets you generate points based on a mathematic formula. The only input variable is 'x', so as an example you may write 'ln(x^3 + 1) - sin(x*x)'.

Expression evaluator uses traditional C/C++ style formating, which is natural for most people. It provides arithmetics, logical and conditional operators. Following terms are supported:
Unary operators: **-** (negative sign), **!** (logical negation)
Binary arithmetic operators: **+**, **-**, *****, **/**, **^**, **%**, **max**, **min**
Binary logical operators: **==**, **!=**, **<**, **<=**, **>**, **>=**, **&&**, **||**, **^^**
Unary functions: **log**, **sqrt**, **sqr**, **abs**, **exp**, **sin**, **cos**, **tan**, **sinh**, **cosh**, **tanh**, **inv**, **asin**, **acos**, **atan**, **ln**, **log10**, **sgn**, **floor**, **ceil**, **round**, **rand**, **f01** (frequency from 20...20000 into log scale 0..1)
Ternary logical operator: **a ? b : c** (if a is true, then the result is b, otherwise it is c)
Constants: **pi**, **e**, **sqrt2**, **ln2**

Analyse audio
Analyse audio lets you analyse a portion of an audio file at specified intervals, extract its level envelope and use those levels to construct the graph's curve.

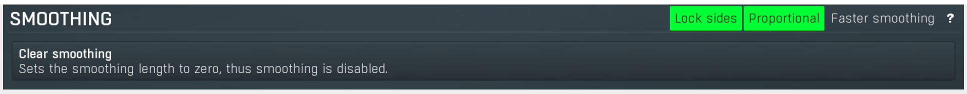
Curvature

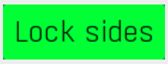




Integral curvature
Integral curvature makes the multi-curvature modes such as rectangles always have an integral number of items, e.g. 1, 2, 3, ... rectangles. If you disable this, it will be also possible to have for example 2.3 rectangles, which will however cause a discontinuity.

Smoothing



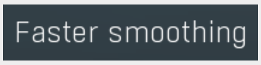


Lock sides

Lock sides makes the smoothing factor equal on both sides.

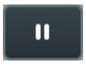
Proportional

Proportional makes the smoothing area size defined by the smaller side.



Faster smoothing

Faster smoothing enables slightly faster algorithm, which can however often cause unnecessary curving.



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

Copy button copies the current analysis to the system clipboard. Then you can use the paste button to show the analysis as a comparison in any of analyzer instanced.

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.

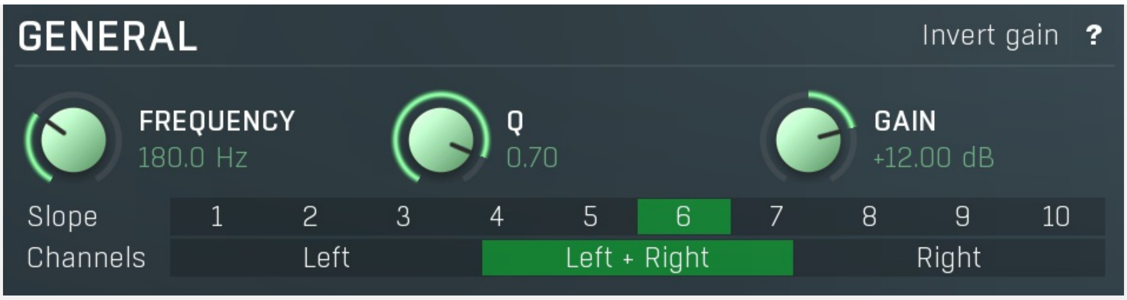
Paste

Paste button loads the settings from the system clipboard.

Random

Random button generates random settings using the existing presets.

General panel



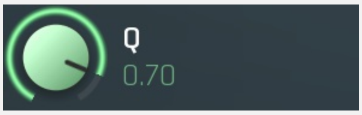
General panel contains standard filter settings such as frequency or Q. Most of these values are available directly from the band graph, but it may be necessary to use these controls for more accurate or textual access.

Invert gain

Invert gain 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.

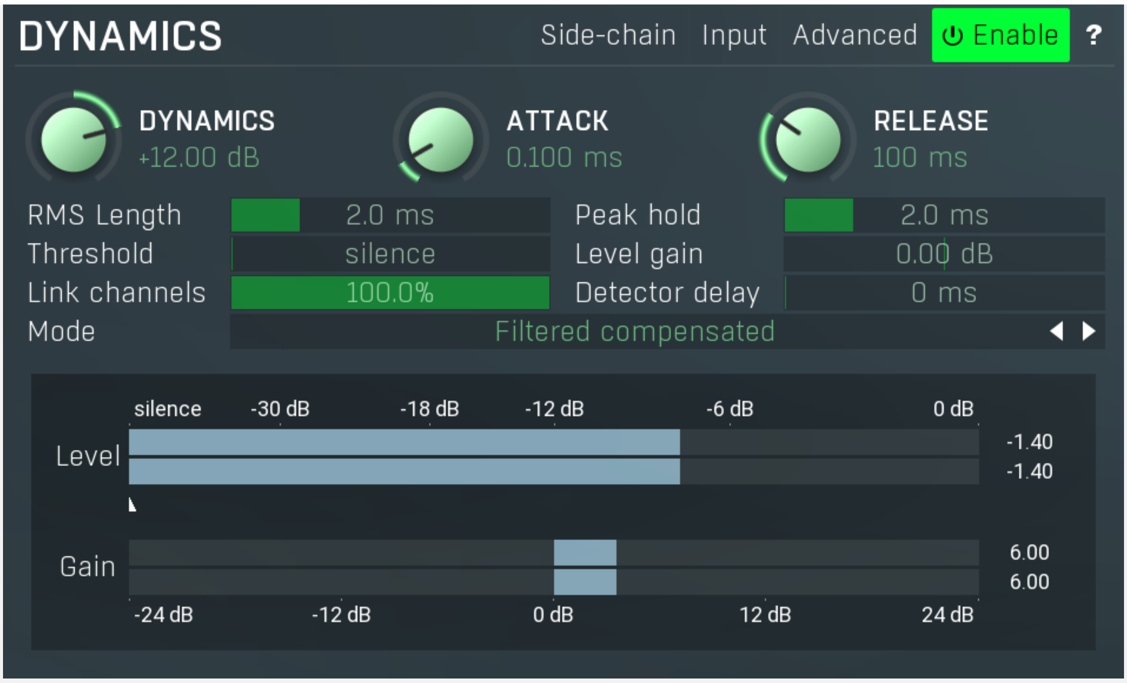
Channels

Channels controls which channels the band processes. If the input is stereo (left and right channels, L+R selected on the main **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.

Dynamics panel



Dynamics panel contains settings of the dynamics processing which control how the filter behaves depending on input signal. Normal filters are static, meaning they don't change any features depending on the input signal. If you enable dynamic properties, by making the **dynamic gain** nonzero, the filter will start listening to the level of the input signal. This requires more CPU of course, as such a band is essentially an extremely complex generalized compressor, but the algorithms used are as efficient as it is technically possible.
A dynamic band varies the gain according to the input level. It can listen to the whole spectrum or to just part of it. By default it is driven by the partial spectrum, which it modifies itself, so, for example, when you have a high shelf, it is essentially listening to a high part of the spectrum. You can do many things with such a dynamic processor, but essentially it can work as a compressor or expander. There are many more advanced ideas that you can do and the full power hasn't really been explored yet.

Sidechain

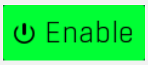
Sidechain switch makes the band measure the input level from the sidechain instead of the input it is processing. This can be used for various techniques, such as avoiding conflicts between bass and bass drum.

Input

Input switch makes the band measure the input level instead of current level in the chain of bands. When this is disabled (default) and the equalizer is processing the bands serially, which means that each band is processing the output from the previous stage, including level measurement. If you enable this switch however, the dynamic processing will be driven by the original input signal instead.
Please note that when **Side-chain** is on, this switch has no meaning, since side-chain has priority.

Advanced

Advanced button displays additional settings for this band. These contain some more esoteric features, such as a dynamic transformation shape.

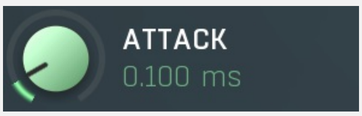


Enable

Enable button enables the dynamic processing. You can use it to switch between enabled and disabled dynamic processing to check the differences.

Dynamics

Dynamics defines the maximum gain of the filter that could be caused by the input signal. For example, if you set it to -24dB and the input signal contained in the band were very strong, the band will be set to an additional -24dB. This would work similarly to a compressor in that band.



Attack

Attack defines the attack time, that is how quickly the level detector increases the measured input level. When the input peak level is higher than the current level measured by the detector, the detector moves into the attack mode, in which the measured level is increased depending on the input signal. The higher the input signal, or the shorter the attack time, the faster the measured level rises. Once the measured level exceeds the **Threshold** then the dynamics processing (compression, limiting, gating) will start.

There must be a reasonable balance between attack and **release** times. If the attack is too long compared to the release, the detector will tend to keep the measured level low, because the release would cause that level to fall too quickly. In most cases you may expect the attack time to be shorter than the release time.

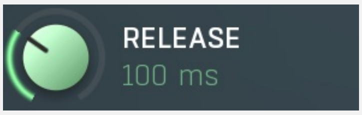
To understand the working of a level detector, it is best to cover the typical cases:

*In a **compressor** the attack time controls how quickly the measured level moves above the threshold and the processor begins compressing. As a result, a very short attack time will compress even the beginning transient of a snare drum for example, hence it would remove the punch. With a very long attack time the measured level may not even reach the threshold, so the compressor may not do anything.*

*In a **limiter** the attack becomes a very sensitive control, defining how much of the signal is limited and how much of it becomes saturated/clipped. If the attack time is very short, limiting starts very quickly and the limiter catches most peaks itself and reduces them, providing lower distortion, but can cause pumping. On the other hand, a higher attack setting (typically above 1ms) will let most peaks through the limiter to the subsequent in-built clipper or saturator, which causes more distortion of the initial transient, but less pumping.*

*In a **gate** the situation is similar to a compressor - the attack time controls how quickly the measured level can rise above the threshold at which point the gate opens. In this case you will usually need very low attack times, so that the gate reacts quickly enough. The inevitable distortion can then be avoided using look-ahead and hold parameters.*

In a modulator, the detector is driving other parameters, a filter cut-off frequency for example, and the situation really depends on the target. If you want the detector to react quickly on the input level rising, use a shorter attack time; if you want it to follow the flow of the input signal slowly, use longer attack and release times.



Release

Release defines the release time, that is how quickly the level detector decreases the measured input level. The shorter the release time, the faster the response is. Once the attack stage has been completed, when the input peak level is lower than the current level measured by the detector, the detector moves into the release mode, in which the measured level is decreased depending on the input signal. The lower the input signal, or the shorter the release time, the faster the measured level drops. Once the measured level falls under the **Threshold** then the dynamics processing (compression, limiting, gating) will stop.

There must be a reasonable balance between **attack** and release times. If the attack is too long compared to release, the detector would tend to keep the level low, because release would cause the level to fall too quickly. Hence in most cases you may expect the attack time to be shorter than the release time.

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

*In a **compressor** the release time controls how quickly the measured level falls below the threshold and the compression stops. As a result a very short release time makes the compressor stop quickly, for example, leaving the sustain of a snare drum intact. On the other hand, a very long release keeps the compression working longer, hence it is useful to stabilize the levels.*

*In a **limiter** the release time keeps the measured level above the limiter threshold causing the gain reduction. Having a very long release time in this case doesn't make sense as the limiter would be working continuously and the effect would be more or less the same as simply decreasing the input gain manually. However too short a release time lets the limiter stop too quickly, which usually causes distortion as the peaks through the limiter to the subsequent in-built clipper or saturator. Hence release time is used to avoid distortion at the expense of decreasing the output level.*

*In a **gate** the situation is similar to a compressor - the release time controls how quickly the measured level can fall below the threshold at which point the gate closes. Having a longer release time in a gate is a perfectly acceptable option. The release time will basically control how much of the sound's sustain will pass.*

In a modulator, the detector is driving other parameters, a filter cut-off frequency for example, and the situation really depends on the target. If you want the detector to react quickly on the input level falling, use a shorter release time; if you want it to follow the flow of the input signal slowly, use longer attack and release times.

RMS length

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

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

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

Note that the RMS detector has 2 modes - a simplified approximation is used by default, and a true RMS is processor can be enabled from the advanced settings (if provided). Both respond differently, neither of them is better than the other, they are simply different.

Peak hold

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

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

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

Threshold

Threshold controls the minimum level above which the dynamic gain actually starts working.

Level gain

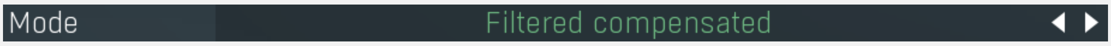
Level gain controls the gain applied to the detector, which can be used for example when the input level is too low, so that dynamic processing would be negligible, unless the level is boosted.

Link channels

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

Detector delay

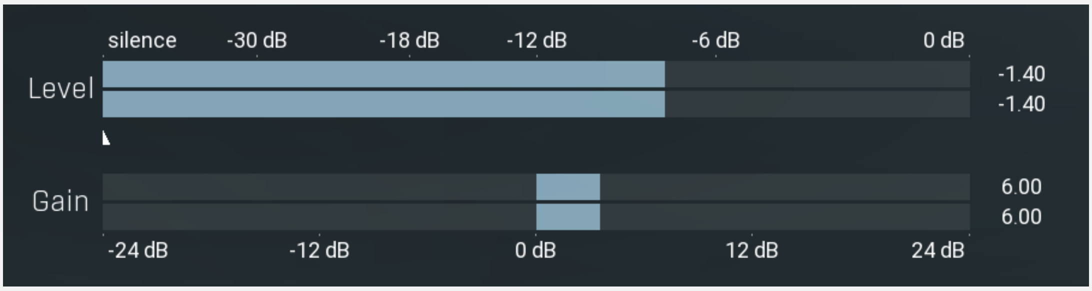
Detector delay lets you delay the detector input, hence the band will react later than the actual input signal.



Mode

Mode controls the way the band reacts to the input signal. It has no meaning if the dynamic gain is 0dB.
Filtered compensated mode is default and it means that the source for measuring input level is a filtered signal with additional compensation. For example, when using a low-shelf filter, the signal is low-passed with a filter with the same settings as the low-shelf, therefore the low-shelf filter is affected only by the signal the low-shelf is actually amplifying or attenuating. Since a low-passed signal with cut-off at 100Hz has usually a much lower level than the one filtered with cut-off at 10 kHz, additional compensation is performed to diminish these differences.
Filtered mode is similar, but the compensation is not performed. This may be advantageous for audio materials that do not contain the full spectrum, e.g. a bass line, where the compensation may make things complicated.
Entire spectrum mode is the simplest - it simply takes the input signal without any further processing. This may be useful for example to attenuate selected frequencies when the input level gets too high.

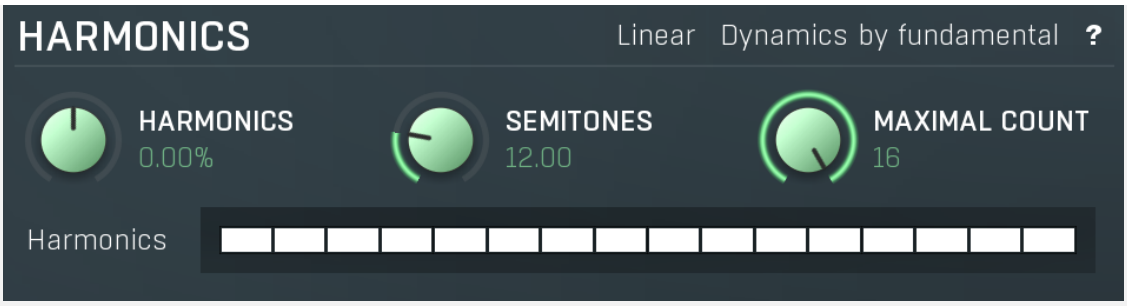
meters



Threshold

Threshold controls minimum level at which the dynamic gain actually starts working.

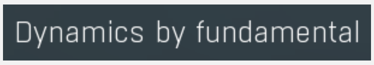
Harmonics panel



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.

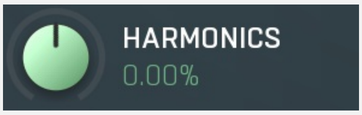
Linear

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.



Dynamics by fundamental

Dynamics by fundamental switch causes each harmonic to be driven by the same detector settings as set for the main band. It is disabled by default, which means that each harmonic is literally a clone of the original filter and has its own dynamics detector depending on its own frequency.
Please note that if you want each harmonic to behave in exactly the same way as the main band, you also need to switch on the Input (at the top of the Dynamics panel), otherwise the harmonics would be measuring the signal processed by the main band.

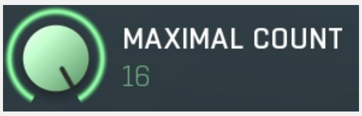


Harmonics

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.



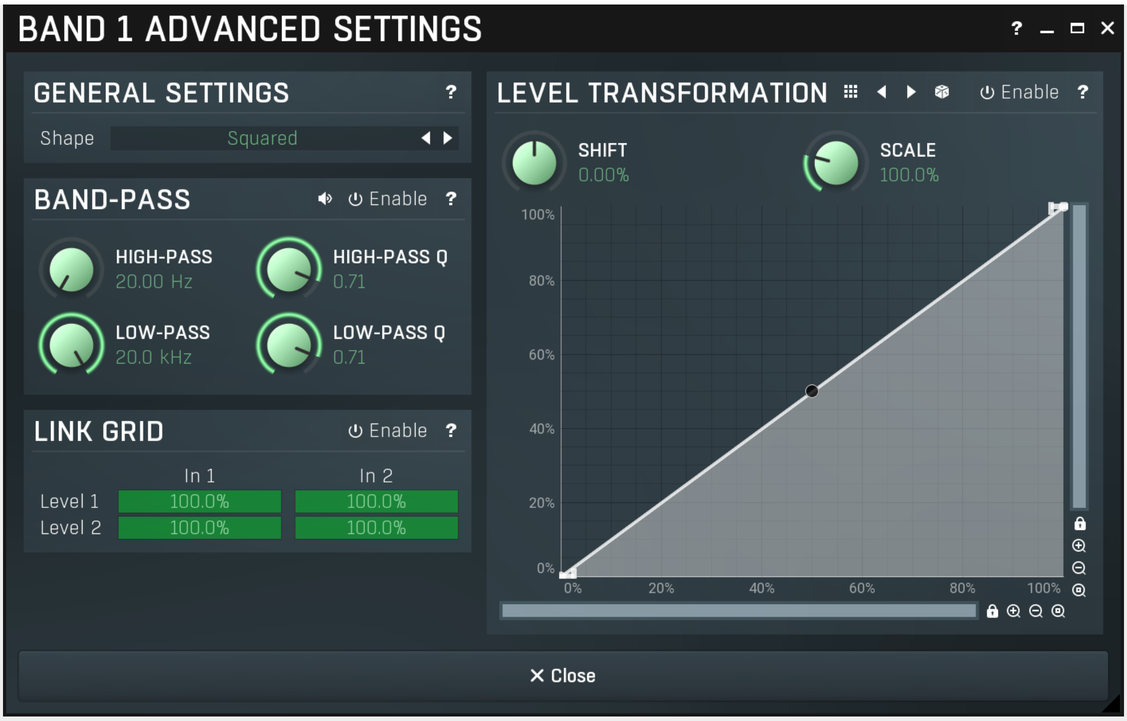
Maximal count

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

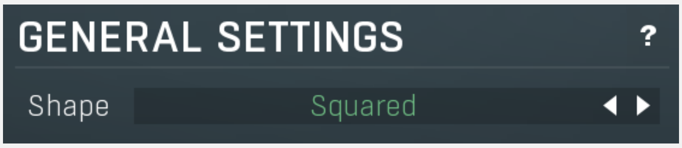
Harmonics grid is useful to turn on/off particular harmonics manually. Click any one to enable / disable it.

Band advanced settings

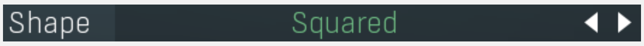


Band advanced settings contains additional settings for the band. These contain some more esoteric features, such as a dynamic transformation shape. It can be displayed by clicking the right mouse button on a band while holding **Ctrl**, from the basic band settings window, or from the band list if provided.

General settings panel



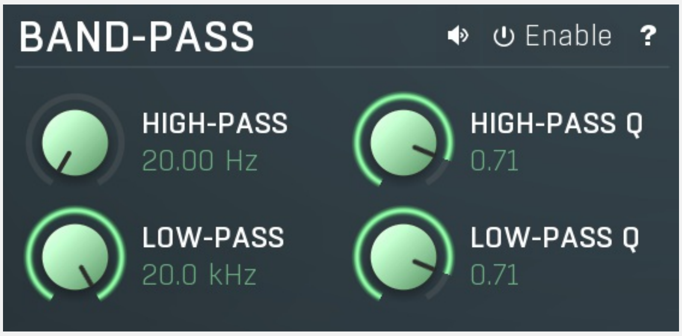
General settings panel contains additional parameters, which are too scientific to be available from the main band settings.



Shape

Shape affects the processing shape. The plug-in features specific non-linear transfer shapes which affect the way the level are interpreted. **Logarithmic** mode is the most physical one, increase from, say, -90dB to -80dB and from -10dB to 0dB produces the same difference in the output dynamic gain. However from the nature of it is tends to generate high gains and usually setting a threshold is needed. **Linear** mode on the other hand tends to stay near minimum gains and usually is the most aggressive. **Squared** mode is a compromise between these two. Comparing the three modes, Linear mode requires the least amount of CPU power and Logarithmic requires the most.

Band-pass panel



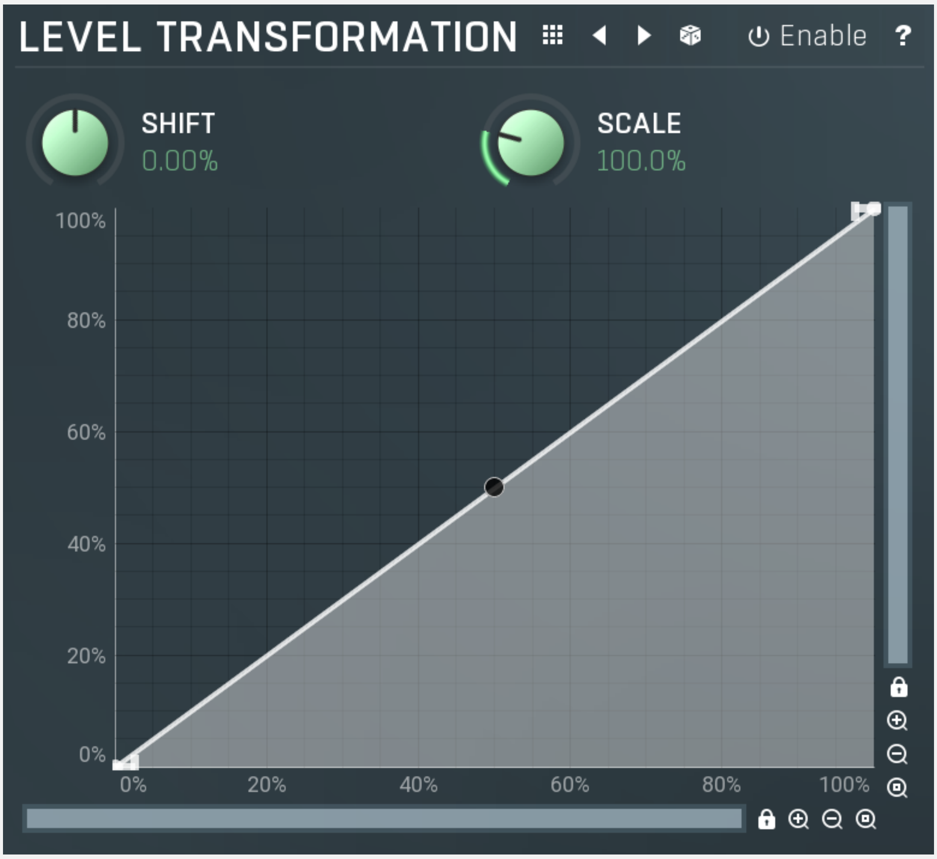
Band-pass panel contains parameters of the band pass, which you can use to process the signal that is used measure level of the band additionally. For example, you may want a band at high frequencies to react to bass content; you can do this by placing the band anywhere on the high frequencies and set the low-pass at say 200Hz.

Play

Play button enables the band-pass monitoring and hence could be useful to tweak the band pass.

⏻ Enable **Enable**
Enable button enables the band-pass module. It is off by default to save CPU resources.

Level transformation



Level transformation graph lets you transform the dynamic gain according to the input level. The X axis contains the input level; the Y axis controls the output level, which is then used to set the dynamic gain.

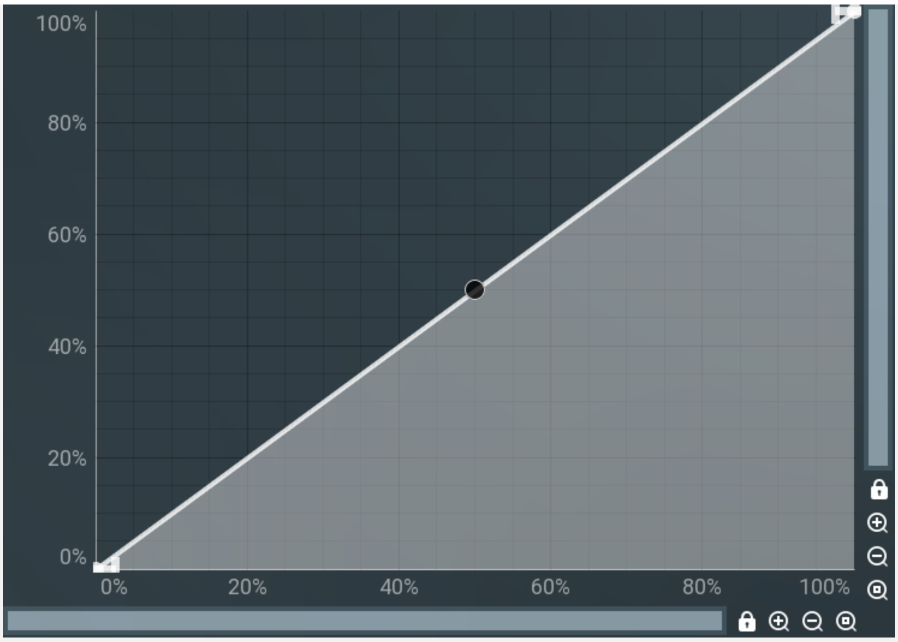
Presets
Presets button displays a window where you can load and manage available presets. Hold **Ctrl** when clicking to load a random preset instead.

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.

⏻ Enable **Enable**
Enable button enables the level transformation module. It is off by default to save CPU resources.



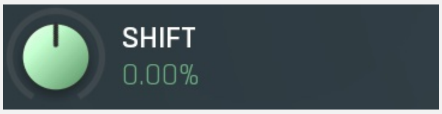
Graph editor

Graph editor lets you edit the envelope graph.

Envelope graph

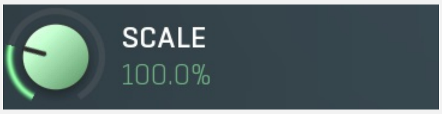
Envelope graph provides an extremely advanced way to edit any kind of shape that you can imagine. An envelope has a potentially unlimited number of points, connected by several types of curves with adjustable curvature (drag the dot in the middle of each arc) and the surroundings of each point can also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



Shift

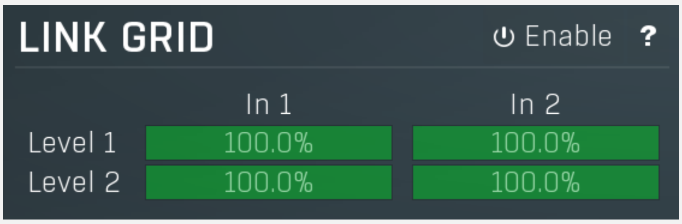
Shift lets you virtually shift the whole graph vertically. This basically shifts the dynamic gain.



Scale

Scale lets you virtually scale the whole graph vertically. This basically scales the dynamic gain.

Link grid panel



Link grid panel controls the linking between the channels; that is. how the input level in each channel affects the levels in the other channels. By default the way channels affect processing in other channels depends solely on the **Link channels**

parameter.
Here you can set up a more complicated relationship. For example, you can make the left channel (1) respond to the right channel (2) only and vice versa. Each column in the grid is an input and each row is an output. Each output level is a mix of the factored input levels. For that example above, the values for "Level 1" would be 0% and 100%, and for "Level 2" they would be 100% and 0%.

⏻ Enable **Enable**

Enable button enables the link-grid module. It is off by default to save CPU resources.

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**

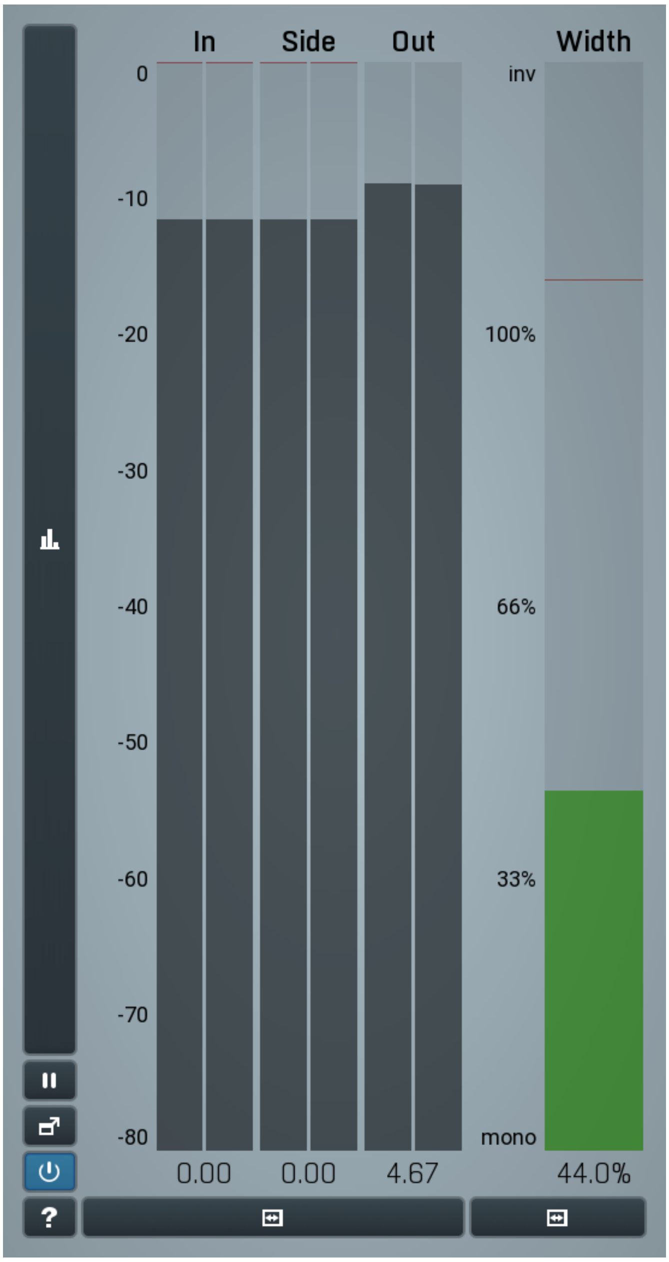
Reset button restores the original equalizer settings.

Invert **Invert**

Invert button inverts the gains of all bands.

Swap gains **Swap gains**

Swap gains button swaps values between gain and dynamics gain.



Global meter view

Global meter view provides a powerful metering system. If you do not see it in the plug-in, click the **Meters** or **Meters & Utilities** button to the right of the main controls. The display can work as either a classical level indicator or, in time graph mode, show one or more values in time. Use the first button to the left of the display to switch between the 2 modes and to control additional settings, including pause, disable and pop up the display into a floating window. The meter always shows the actual channels being processed, thus in M/S mode, it shows mid and side channels.
In the classical level indicators mode each of the meters also shows the recent maximum value. Click on any one of these values boxes to reset them all.

In meter indicates the total input level. The input meter shows the audio level before any specific processing (except potential upsampling 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.

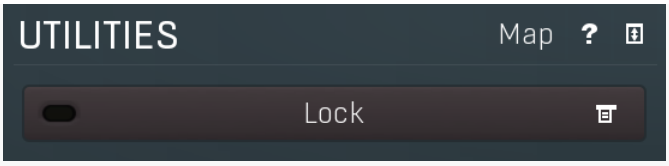
Collapse

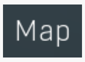
Collapse button minimizes or enlarges the panel to release space for other editors.

Collapse

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

Utilities





Map

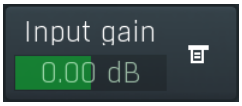
Map button displays all current mappings of modulators, multiparameters and MIDI (whichever subsystems the plugin provides).

Lock

Lock button displays the settings of the global parameter lock. Click on it using your left mouse button to open the Global Parameter Lock window, listing all those parameters that are currently able to be locked.
Click on it using your right mouse button or use the **menu button** to display the menu with learning capabilities - **Learn** activates the learning mode, **Clear & Learn** deletes all currently-lockable parameters and then activates the learning mode. After that, every parameter you touch will be added to the lock. Learning mode is ended by clicking the button again.
The On/Off button built into the Lock button enables or disables the active locks.

Collapse

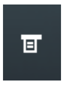
Collapse button minimizes or enlarges the panel to release space for other editors.



Multiparameter

Multiparameter button displays settings of the multiparameter. The multiparameter value can be adjusted by dragging it or by pressing Shift and clicking it to enter a new value from the virtual keyboard or from your computer keyboard.

Click on the button using your left mouse button to open the **Multiparameter** window where all the details of the multiparameter can be set. Click on it using your right mouse button or click on the **menu button** to the right to display an additional menu with learning capabilities - as described below.



Menu

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

Learn attaches any parameters, including ranges. Click this, then move any parameters through the ranges that you want and click the multiparameter button again to finish. While learning is active, "REC" is displayed on the multiparameter button and learning mode is ended by clicking the button again.

Clear & Learn clears any parameters currently in the list then attaches any parameters, including ranges. Click this, then move any parameters through the ranges that you want and click the multiparameter button again to finish. While learning is active, "REC" is displayed on the multiparameter button and learning mode is ended by clicking the button again.

Reset resets all multiparameter settings to defaults.

Quick Learn clears any parameters currently in the list, attaches one parameter, including its range and assigns its name to the multiparameter. Click this, then move one parameter through the range that you want.

Attach MIDI Controller opens the MIDI Settings window, selects a unused parameter and activates MIDI learn. Click this then move the MIDI controller that you want to assign.

Reorder to ... lets you change the order of the multiparameters. This can be useful when creating active-presets. Please note that this feature can cause problems when one multiparameter controls other multiparameters, as these associations will not be preserved and they will need to be rebuilt.

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

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

Collapse

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

Preset selector

Preset management window provides management for your presets.

Backup

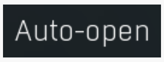
Backup button lets you backup presets for all MeldaProduction software into a single file, so you can transfer it to a different machine and restore the presets there for example.

Restore from backup

Restore from backup button lets you restore presets for all MeldaProduction software from a single file created by the **Backup** button.

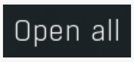
Folders tree

Folders tree lets you organize your presets into any number of folders. Use the buttons at the bottom of the window to create, rename or delete sub-folders. Note that these are not actual files & folders on disk, but are records in the preset database.



Auto-open

Auto-open switch makes the tree automatically open selected items, so that all sub-folders are visible, whenever you select one. This makes it easier to browse through large structures containing many folders. The switch also makes the browser show all presets available in the selected folder including all sub-folders (except when you select the root folder).

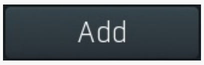


Open all

Open all button expands the whole tree, so you can see all of the folders. This may be handy when editing large preset structures.

Close all

Close all button collapses the whole tree except for the root folder. This may be handy when editing large preset structures.

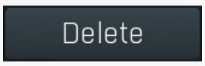


Add

Add button creates a new folder in the tree

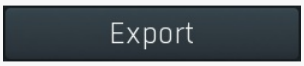
Rename

Rename button lets you rename the selected folder.



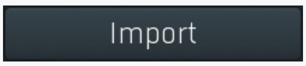
Delete

Delete button deletes the folder including all the presets and subfolders in it.



Export

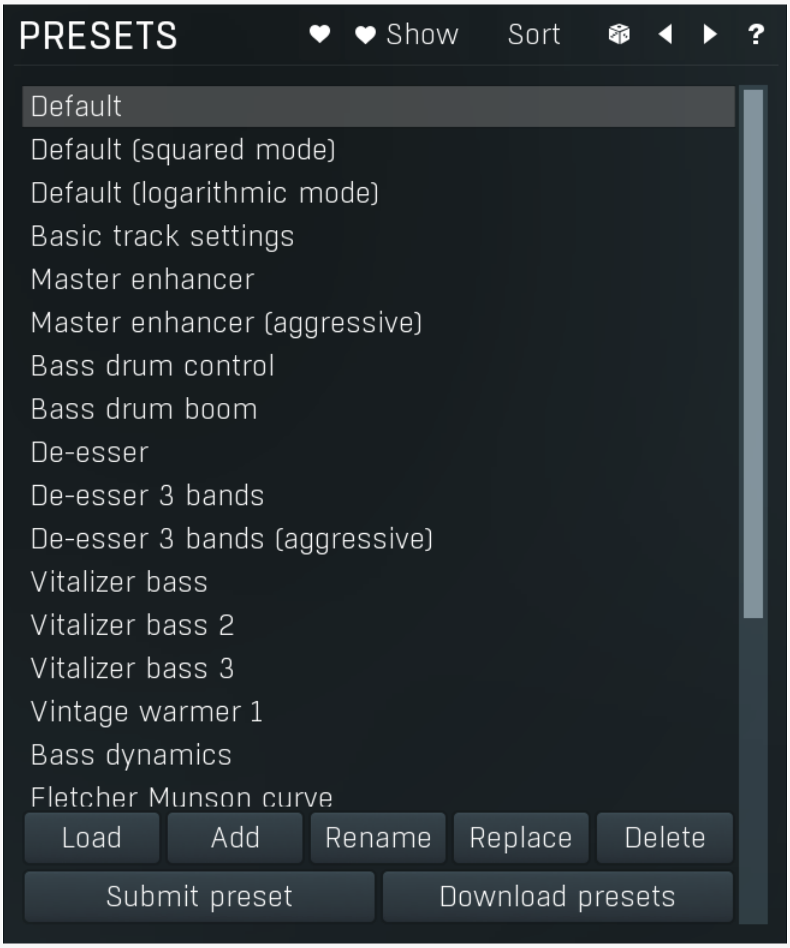
Export button lets you export the selected folder including all presets and sub-folders into a file, which you can then transfer to any computer. Or just use as a back-up.



Import

Import button lets you import a file containing presets and sub-folders and add it to the selected folder. The importer will ask you whether to destroy the original contents, so that the new presets replace previous ones, or to keep both.

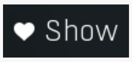
Presets list



Presets list contains all presets available in the selected folder. **Double-click** on a preset or use **Load** button to load a preset. Use the buttons at the bottom of the list to perform additional changes. Please note that these are not actual files & folders on disk, but are records in the preset database.

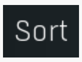
Favourite

Favourite button toggles the 'favourite' indicator for the selected preset.



Show

Show button shows only the favourite presets and hides the others.



Sort

Sort button shows the presets sorted alphabetically.

Random

Random button selects and loads a random preset from the current folder. This way you can quickly browse the presets in the folder in a completely random order.

Previous

Previous button selects and loads the previous preset from the current folder.

Next

Next button selects and loads the next preset from the current folder.

Submit preset

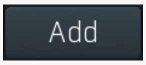
Submit preset button submits the selected preset to the online exchange servers and retrieves all the presets currently in the database. This feature serves as an online database of presets available for all the user community. Please do not submit garbage presets.

Download presets

Download presets button retrieves all the presets currently in the database. This feature serves as an online database of presets available for all the user community. Please consider participating by submitting your presets as well.

Load

Load button loads the specified preset. Please note that you can do the same thing by double-clicking the preset itself or pressing the Enter key.

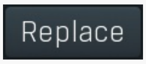


Add

Add button creates a new preset using the current settings.

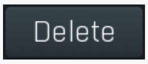
Rename

Rename button lets you rename the selected preset.



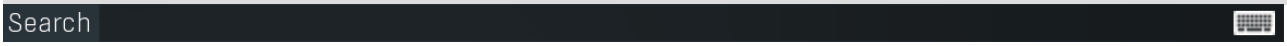
Replace

Replace button replaces the selected preset by one with current settings.



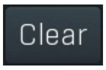
Delete

Delete button deletes the selected preset.



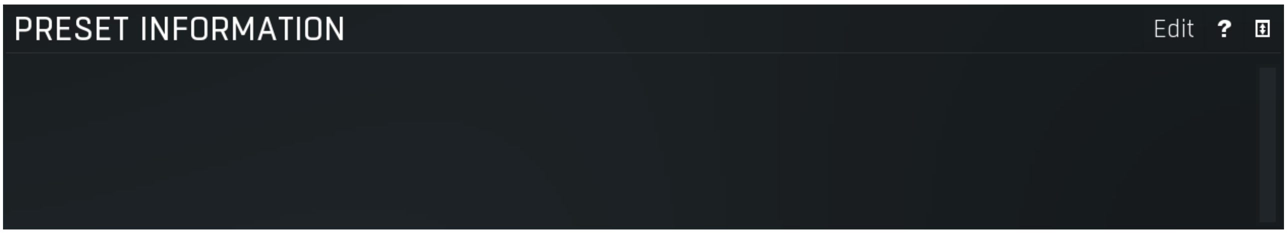
Search

Search filters the list of available presets to those containing the keywords in name or information.



Clear

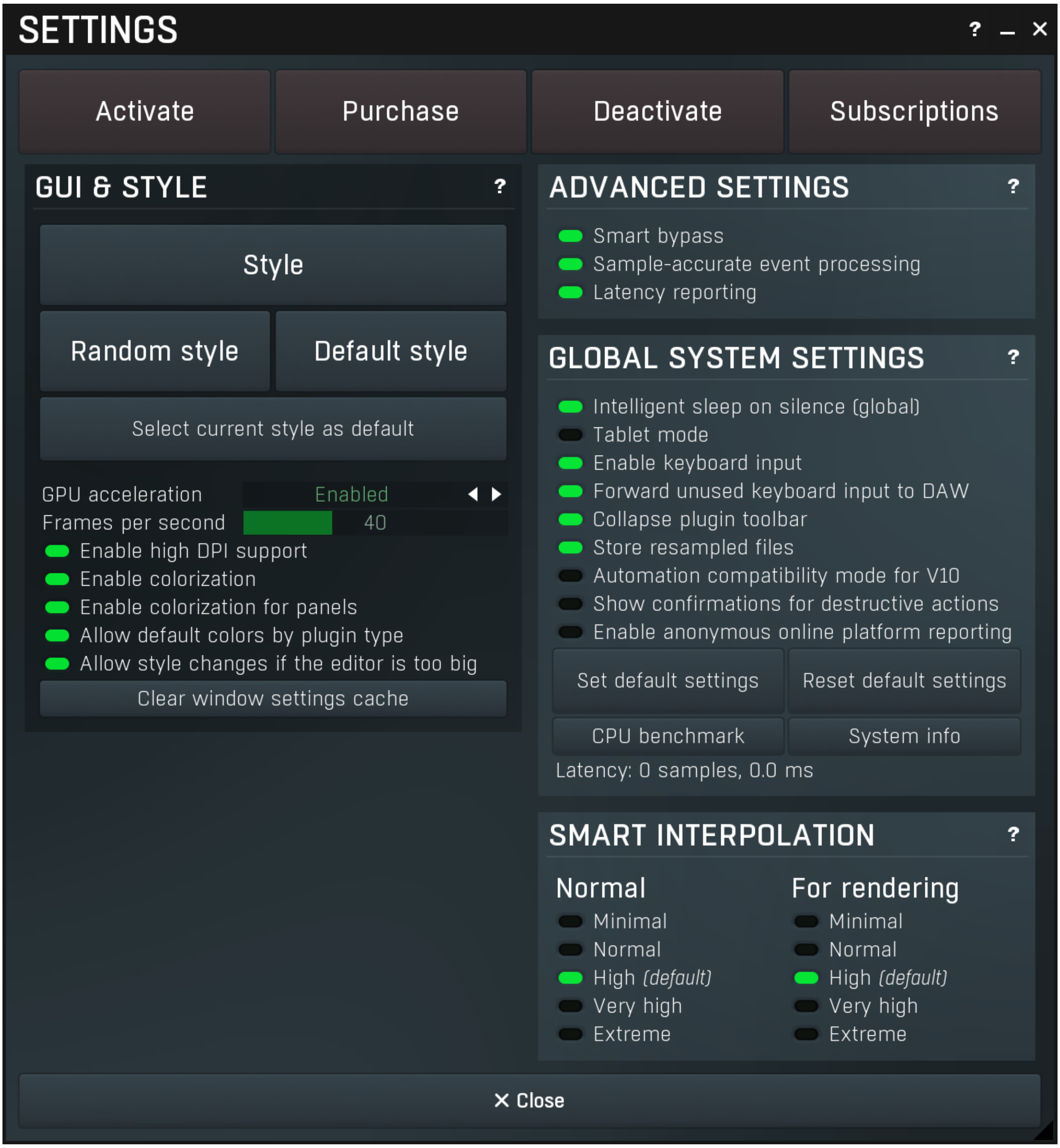
Clear button deletes all text in the search field.



Preset information

Preset information field contains optional information about the preset, which you can edit when creating or renaming the preset.

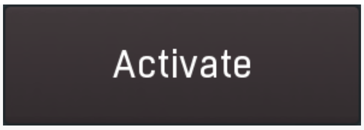
Plugin settings



Plugin settings window offers more advanced settings and is available via the Settings button.

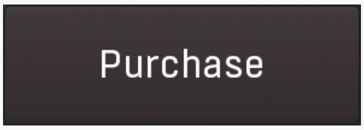
Licence panel

Licence panel lets you manage licences on this computer.



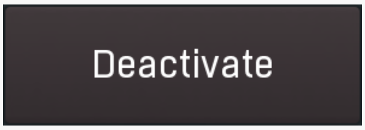
Activate

Activate button lets you activate your licence for the plugin on this computer.



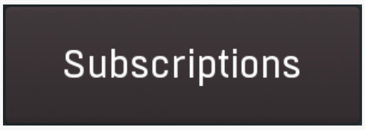
Purchase

Purchase button navigates to the plugin's website, from which you can purchase a licence for the plugin.



Deactivate

Deactivate button lets you deactivate any licences on this computer. It can be useful when you need to work on a public computer or if you sell your licence.

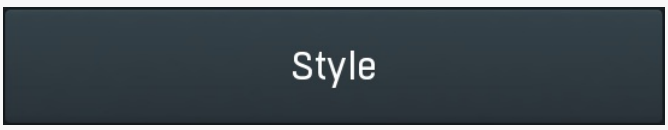


Subscriptions

Subscriptions button lets you manage the subscription based licencing.

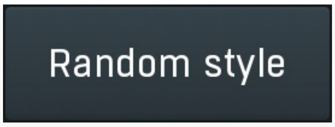
GUI & Style panel

GUI & Style panel lets you configure the plugin's style (and potentially styles of other plugins) and other GUI properties.

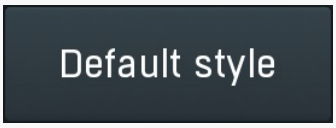


Style

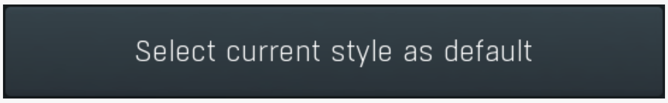
Style button lets you change the style for this particular plugin.



Random style
Random style button selects a random style with random editor mode.



Default style
Default style button reverts to the default style and default size of the GUI. Hold the Ctrl key while clicking to revert all MeldaProduction software products, not just the current plugin.



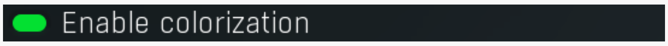
Select current style as default
Select current style as default button stores the current style as the default for all MeldaProduction software. This is used for the other plugins that are currently using the default style; that is, those plugins for which you have NOT selected a specific style. Please note that if you have already selected a specific style for a particular plugin, then it won't be changed until you use the Default style button.

GPU acceleration
GPU acceleration controls how much the GPU is used for visual rendering to save CPU power.
Enabled mode provides maximum speed and lets the GPU perform as many drawing operations as possible.
Compatibility mode uses the GPU for drawing, but doesn't use modern technologies for maximum performance. Use it if you experience occasional problems with drawing, the usual case for older ATI graphics cards. With Pro Tools on OSX this mode is always used instead of Enabled mode due to compatibility problems with this host.
Disabled mode disables GPU acceleration completely, drawing is then performed by the CPU. Use only if you experience technical difficulties.
A known problem may occur when using multiple displays with multiple graphical interfaces. When moving the plugin window from one display to another, it may stop displaying correctly until you move it back to the original display.

Frames per second
Frames per second controls the refresh rate of the visual engine. The higher the number is the smoother everything is, but the more CPU it requires. You might want to lower this value if your computer is running out of CPU power.

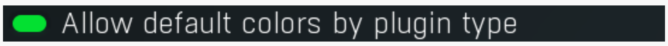
Enable high DPI / retina support
Enable high DPI / retina support enables the plugin to use the high resolution on high DPI (Windows) and retina (OSX) devices. It is enabled by default and detected automatically, if the host allows it. If you run into any problems, you can disable it using this option. It may be desired if you use multiple displays where only some of them feature the high resolution making the image on the low resolution ones look ugly.

If you disable this option, on Windows the high DPI device detection will be ignored and the plugin will probably appear very small. You can manually compensate for it by using a bigger style. On OSX disabling this option will disable the high DPI rendering, resulting in the classic blurry look of non-compliant applications. Changes take effect after you restart the host.



Enable colorization
Enable colorization enables the plugin to change the colors of certain elements overriding your style settings. Plugins use that to highlight different parts of the graphics interface for easier workflow. You may want to disable it if you just feel it's not for you. This particular option is relevant only for controls - knobs, sliders, checkboxes etc.

Enable colorization for panels
Enable colorization for panels enables the plugin to change the colors of certain elements overriding your style settings. Plugins use that to highlight different parts of the graphics interface for easier workflow. You may want to disable it if you just feel it's not for you. This particular option is relevant only for containers - panels, graphs etc.

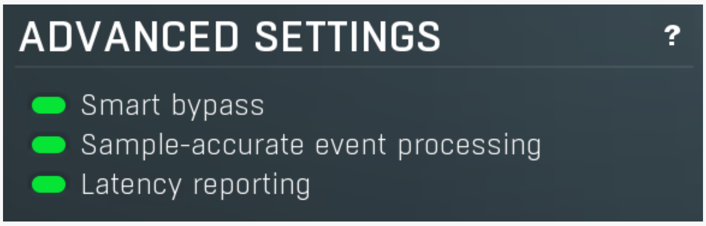


Allow default colors by plugin type
Allow default colors by plugin type is on by default and makes the plugin select its default colors depending on the type of the plugin. Hence for instance equalizer will always be green. This is done by selecting one of the first 8 color presets for the current style, so the actual colors depend on selected style and its presets. You may want to disable this if you for example want all plugins to look the same including the style and colors. It is necessary to restart your host for a change to this option to take effect.

Allow style changes if the editor is too big
Allow style changes if the editor is too big is on by default and makes the plugin change its style, editor mode and other settings if it finds out it is too big to fit the current screen resolution.

Clear window settings cache
Clear window settings cache button deletes stored states of all popup windows on all MeldaProduction software. The window settings mostly contain positions and sizes, but in some cases also the data inside the popup windows. You can use this feature if something goes wrong, a window doesn't appear at all, problems like that. While this shouldn't happen and it's generally better to contract our support, this button provides a potential quick fix.

Advanced settings panel



Advanced settings panel contains settings that control the behaviour of this instance. These are properties that rarely need to be changed, so they have been moved here.

Smart bypass

Smart bypass enables the high quality crossfading bypass system, which ensures a smooth transition between the processed and dry signals. You may want to disable it if you are using settings with latency on a plugin, which demands lots of CPU power, which would otherwise need to perform processing even when bypassed, which is pretty much the only downside of the smart bypassing algorithm.

Sample-accurate event processing

Sample-accurate event processing makes the plugin schedule every event such as MIDI or automation to their accurate locations with sample accuracy, if the host allows it.

For example, if the block size in your host's audio settings is 1024 samples, this means the plugin is probably processing blocks of 1024 samples, in 44100 Hz sampling rate it is about 23ms. If this setting is disabled, any change in automation, MIDI, modulation etc. may then be granularized to 23ms (once per block), which means that you will not be able to recognize events that occur say 10ms apart from each other. When this setting is enabled however, the plugin divides processing blocks to sub-blocks and processes the events at their correct positions. This may, of course, require more CPU power.

Latency reporting

Latency reporting makes the plugin report latency to the DAW, if any. Normally this is enabled, but in certain live situations you may want to disable this, so that the DAW stops compensating the latency on other tracks. It has no effect if the plugin is placed on master track.

Global system settings panel

Global system settings panel contains advanced settings which are applied to all plugins on this computer.

Intelligent sleep on silence (global)

Intelligent sleep on silence (global) is a global switch, which disables the **Auto disable on silence** feature in all plugins on the system. It is provided "just in case" something goes wrong.

Tablet mode
Tablet mode enables better support for tablets at the expense of the mouse. Enable this if you are using a tablet to control the plugins and it is behaving incorrectly.

Enable keyboard input
Enable keyboard input enables the keyboard input for the main plugin window. You may want to disable if the plugin intercepts spacebar key (often used by the host for playback enable/disable and your host doesn't allow for the problem itself.

Forward unused keyboard input to DAW
Forward unused keyboard input to DAW makes the plugin forward unused keyboard events to the DAW from its popups. If this is disabled, pressing say spacebar commonly used to start/stop playback won't work if a popup window is active. Enabling this makes this work and it is optional just in case your DAW does something unexpected.

Collapse plugin toolbar
Collapse plugin toolbar makes all plugins collapse the plugin toolbar containing more advanced features such as channel modes, A-H presets, upsampling, safety limiter etc. It is enabled by default to make the user interfaces cleaner and easier to grasp for beginners.

Store resampled files
Store resampled files allows the plugins create audio files for sampling rates being used if they differ from the original file sampling rate. It is used only by a few plugins, but it can improve the loading performance a lot at the cost of some additional storage on the hard drive. Disable this option if you are short on free space.

Automation compatibility mode for V10
Automation compatibility mode for V10 reverts the set of automation parameters back to version 10 and earlier. Use this if you need the plugins to work with projects, which contain autmation, made using version 10 or older. In version 11 the list of automatable parameters have been highly simplified and reorganized and multiparameters are provided for the vast number of hidden parameters. This should speed up loading, improve workflow with the plugins and improve compatibility with various hosts.

Show confirmations for destructive actions
Show confirmations for destructive actions makes the plugin display a confirmation window whenever you are going to change the plugin settings irreversibly when using a feature, for example: when resetting your settings.

Enable anonymous online platform reporting
Enable anonymous online platform reporting helps us maximize compatibility with your operating system and host. If enabled, our plugins will send information about the system and host that you are using. We can use this information to find out which plugins and platforms are used the most and maximize testing and support there. Platform reporting is completely anonymous and requires only minimal internet connection time (a few kB once a week).

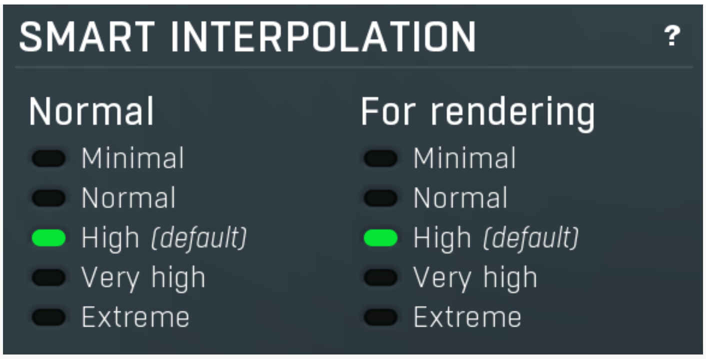
Set default settings
Set default settings button stores the current plugin settings as the defaults, so that when you open a new instance of the plugin, these settings will be loaded automatically.

Reset default settings
Reset default settings button removes the defaults that you set using **Set default settings** button, so that when you open a new instance of the plugin, the factory defaults will be loaded.

CPU benchmark
CPU benchmark button calculates the performance of the plugin with the current settings.

System info
System info button displays some technical information about the build and the machine.

Smart interpolation panel



Smart interpolation panel controls the depth of the smart interpolation algorithm, which controls the parameters in order to provide maximum audio quality and lower the chance of zipper noise. Smart interpolation is engaged whenever you change any parameter via the GUI, modulators, multiparameters, MIDI or automation.

Many parameters can be automated easily and the plugin responds with sample-accurate results. However, several parameters need exhaustive pre-processing when changed. In these cases, the parameters are not updated every sample, but, for example, once every 32 samples. This highly reduces CPU usage, but affects the output quality.

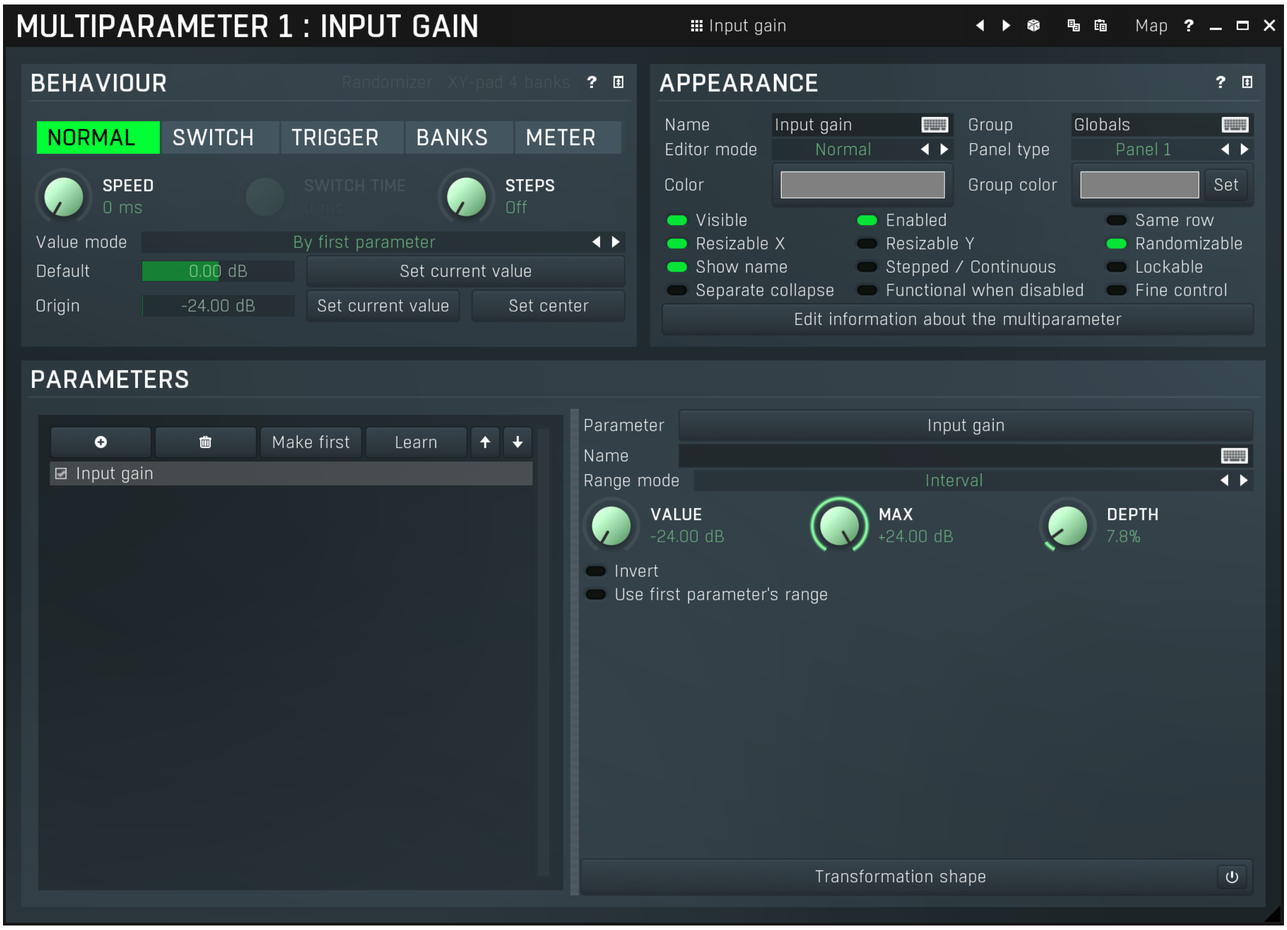
With modulators the situation is more complicated. Besides the updating issue, the modulator itself can perform some pretty advanced processing, hence it is better to perform the processing in blocks. However, the bigger the block, the less often the modulator updates those parameters associated with it and the resulting modulation is less accurate. In a way you can say that the modulator is slower and lazier. This may actually be wanted, so when it comes to modulators it is not true that a better mode always means better output quality.

The smart interpolation mode controls the maximum number of samples being processed before the parameters are updated. **Minimal mode** uses 2048 samples and rarely will do anything unless processing offline. **Normal mode** uses 256 samples and usually is enough to achieve good quality results. **High mode** uses 32 samples and provides perfect quality for most cases. It is also a good compromise between CPU usage and audio quality, so it is the default. **Very high mode** uses 4 samples and you will rarely need it. **Extreme mode** uses 1 sample, which means that everything is updated after every single sample. This provides the highest possible accuracy and quality you can ever achieve, however it requires lots of CPU and it is very unlikely that you will ever need it. If you use this mode and still hear audio artifacts, then either what you are hearing is actually CPU overload, or you are doing something that is not physically possible.

The higher the mode, the quicker the parameter updates, but the more the CPU load.

Please note that modulating certain parameters without artifacts is impossible. For example, when modulating a delay very quickly, the physics of such a process just cannot occur in the natural world and the results are appropriately unnatural. These physically impossible processes usually manifest themselves as distortion or zipper noise.

MultiParameter editor



Multiparameter is a powerful structure, which can speed up your workflow significantly and even perform automatic tasks, often useful when performing in real-time for example. Essentially a multiparameter is a controller which controls other parameters, in fact, an unlimited number of them. Each parameter has limits and potentially a transformation curve for more advanced processing. By manually moving the multiparameter (or automating/modulating it) you can control all of the associated parameters at once.

*This is just the beginning, but it is worth demonstrating how it could be used. We will show it on a vibrato effect. MVibratoMB (and partly MVibrato) is very good at simulating rotary speakers. A rotary speaker traditionally contains a speed switch, or in our case we will think of it as a speed knob - a control that alters the spin speed of the rotary. This would normally be the **Rate** parameter of the vibrato. However, when the rate is increased, the vibrato starts changing the pitch too much, sounding a little too "honky-tonk". We can compensate for this by lowering the **Depth** parameter. As it is not very convenient to control 2 parameters at once, we use a multiparameter to control both parameters with appropriate ranges (ascending for the **Rate** and descending for the **Depth**).*

Besides this basic usage, multiparameters can also work as triggers and switches. Set a multiparameter's mode to **Trigger** or **Switch** and it stops being a slider and becomes a button. When you click the button, the multiparameter starts moving on its own - over the dialled-in switch time it will increase its value (and also the values of any associated parameters) to a maximum and, in the case of trigger mode, then decrease it back to a minimum. In switch mode clicking the button again, the multiparameter decreases back to the minimum value. To make the multiparameter into a simple switch, we can set the switch time to minimum, but in this case we want to extend the functionality in our rotary example.

As mentioned, rotary speakers often have a speed switch. Once switched on, the speed starts increasing until it reaches the "fast" setting, and when switched off, the speed starts decreasing to the original "slow" rate. All we need to do to replicate this functionality is to set the multiparameter's mode to 'switch'.

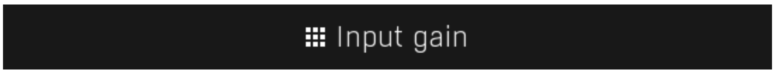
A real rotary actually has 2 speakers, one for low frequencies and the other for the higher ones. As you might expect, these do not have the same spin rate nor do they speed up or slow down equally either. Here is where we can start showing the true potential of multiparameters.

To simulate this, we have to use two bands of MVibratoMB, the first one will simulate the lower reproductor, and the second will be the higher. We use the first multiparameter to control the first band's rate in the same way as described in the example above. Similarly, we use the second multiparameter to control the second band's rate. Now we have 2 switches and can make each band speed-up or slow-down separately, but we want just one switch for both bands. To do this, we use a third multiparameter to control the first and second multiparameters, in switch mode again but with a 0ms switch time. Pressing the button of the 3rd multiparameter instantly activates the other 2 multiparameters, they both start speeding-up, over a different time period as we requested. Pressing the button again, releases it which also instantly releases the first 2 multiparameters and they start slowing down. Just like the real thing.

Now that we have shown you what is possible with multiparameters, it is worth mentioning that they are used extensively for building devices on the easy screens of most Melda plugins. Every multiparameter given a name in the **Information** panel will be shown on the

Easy screen (if the plugin has one). Check our online video tutorials to get more information about **multiparameters and building devices**.

It is also worth mentioning that you can access the multiparameter settings directly from easy screen by holding Ctrl+Alt and clicking on the target control. It may simplify building devices. Note that this may not work for some editor modes such as meters or bar graphs.



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

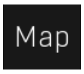
Left arrow
Left arrow button loads the previous preset.

Right arrow
Right arrow button loads the next preset.

Randomize
Randomize button loads a random preset.

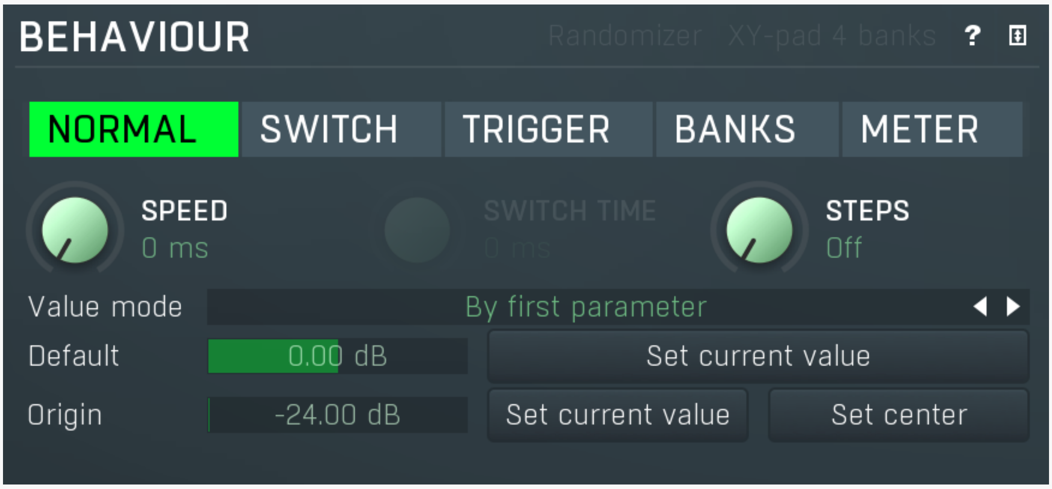
Copy
Copy button copies the settings onto the system clipboard.

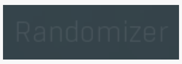
Paste
Paste button loads the settings from the system clipboard.



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

Behaviour





Randomizer
Randomizer switch is available only for **Trigger** mode and it makes the multiparameter produce random values for each associated parameters. This is useful to implement some sort of randomization feature, which covers a set of parameters. You usually want to set the **Switch time** to 0, so that the randomization is instant, but longer values may be useful for some creative effects.

XY-pad 4 banks
XY-pad 4 banks switch is available only for **Banks** mode and it lets you create XY pads, that would interpolate between 4 banks you specify. 1st bank belongs to the left top corner, 2nd to the right top, 3rd to left bottom and 4th to the right bottom. Note that in order for this to work, the multiparameter must NOT be the last one and it occupies the next multiparameter as well, so you need to name the next multiparameter and associate it to some parameters, ideally the same ones.

Mode

Mode controls the behaviour of the multiparameter.

Normal mode makes the multiparameter work like any other control.

Switch mode hides the slider and shows a button instead. The button has 2 states. By pushing the button, the multiparameter value starts rising from 0% to 100% over a specified time interval. By pushing it again the value starts falling back to 0%. You could do the same thing having the multiparameter in normal mode and moving the slider from left to right and then back, but mode this performs that automatically and maintains a constant time period.

Trigger mode is similar to switch mode, but the button has only a single state and when you push it, the value automatically goes from 0% to 100% and then back without any need to push the button again.

Banks mode is very different. A multiparameter in banks mode keeps several states (called banks) for all of the parameters, much like A-H presets, but only with a limited set of parameters. The multiparameter then morphs between the banks or can be set to switch directly between them (no interpolated values). This is a marvellous way to control many parameters with complex settings by using a single multiparameter.

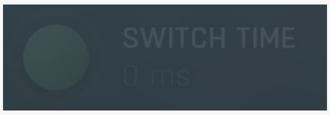
Let's explain the banks mode in more detail. Say you switch a multiparameter to banks mode, learn a few parameters and set the number of banks to 4. Then bank 1 contains a value for all of the parameters. Similarly bank 2 contains a different value for each of them. And so on. If you set the multiparameter slider to 0%, the associated parameters will be set to values in bank 1. If you set the slider to 100%, bank 4 will be used. If you set the slider to 33.3%, bank 2 will be used. And what if you select 50%? Then it will be halfway between bank 2 and bank 3.

You can have many banks, you can edit each of them, generate random settings etc. So let's say you want to create some complex movement. You use a multiparameter in banks mode, select a reasonable number of banks. You can edit each of them, but it is easier to use the randomization button to generate random settings for each of them. Then every time you move the multiparameter, all of the associated parameters will move, somewhere between the banks. You can then use a modulator or automation to slowly adjust the multiparameter.

Meter mode makes the multiparameter work as a meter. Instead of controlling other parameters it starts following the value of them. You can then use that to implement a simple meter on the easy screen (if the plugin has one).

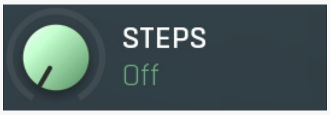
Speed

Speed controls the interpolation time. When it is zero and you change the multiparameter value, the associated parameters are adjusted immediately. If this is non-zero however, the actual parameters won't change immediately but will interpolate over time. The speed value is actually the time needed to go from minimum to maximum or vice versa. So if this is 1 second and the current value is say 0% and you click 100%, it will take 1 second for the multiparameter to get there.
This feature is provided mainly because changing some parameter via MIDI or mouse may cause unnecessary zipper noise or inaccuracies due to low MIDI precision. Using the interpolation you can somewhat slow everything down, so that the artifacts become negligible. It can also be used creatively. The default value has been experimentally tested to avoid all artifacts for most parameters.



Switch time

Switch time defines the time needed to switch from the minimum value to the maximum one, or conversely. It is used only in **switch** and **trigger** modes.



Steps

Steps lets you create an arbitrary number of equi-distant steps for the multiparameter values. While this technically limits the possibilities of the multiparameter by limiting the number of accessible values, it is sometimes easier to choose from a predefined number of options than from the full range. If you want to use different ranges between the steps, use the Banks mode with Interpolate values disabled.

Value mode

Value mode defines the units displayed on the multiparameter.
Percents mode lets the multiparameter display percentages between 0% to 100%.
Percents (-100% to 100%) displays percentages between -100% to 100%.
By first parameter mode uses the current value of the first parameter that is controlled by the multiparameter. For example, if you want to control a plugin gain, but also in addition to the changed gain control other parameters, you may still want to call the multiparameter "gain" and the units should be decibels as usual, not percentages which do not make much sense for such a multiparameter.
By bank name displays the name of the nearest bank.
By bank name interpolated considers name of all banks numbers. It then interpolates between them and displays the result as a number.
By bank name interpolated log is similar, but interpolates the values in logarithmic domain.considers name of all banks numbers. It's useful for units, which are naturally logarithmics, such as frequency.
By bank number shows the index of the nearest bank.

Default

Default controls the default value of the multiparameter. You can edit it directly or just set the MP into its reasonable default and click the **Set current value**.Most GUI components created for the multiparameter respond to right-click by setting the default value in the same way that other parameters do. It is essential for user experience when building your own devices.

Set current value

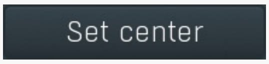
Set current value stores the current value as the default one for the multiparameter.

Origin

Origin informs the GUI engine of the origin of the value. For instance, a default value for panorama is in the center and it is logical that visual elements controlling panorama should somehow highlight the center position. If, for example, you are using a value button to edit the panorama, by default it displays the current value using a bar starting from the left side (being the origin defined as minimum) towards the actual value, but here it is better to display the bar from the center towards the current value, whether it is on the left or right of the center. Therefore the center should be the origin.

Set current value

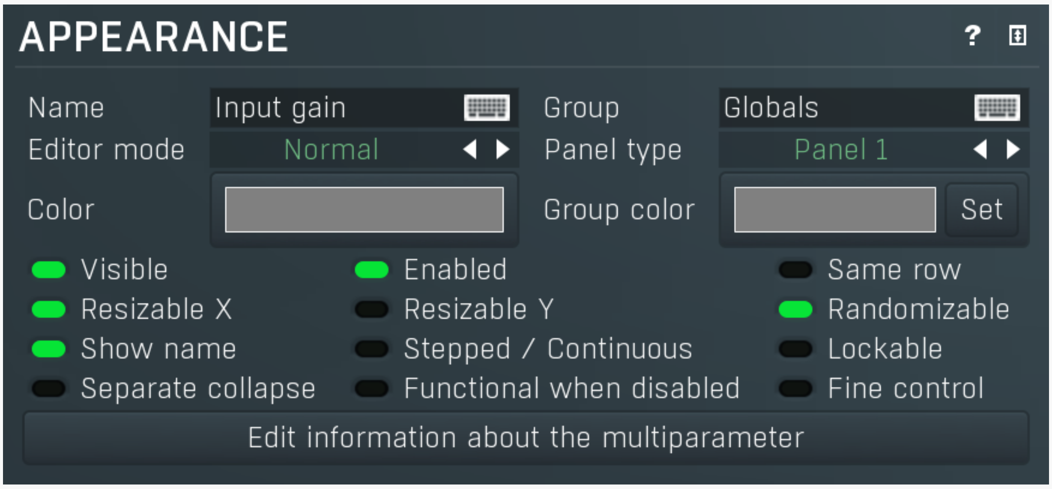
Set current value stores the current value as the origin for the multiparameter.

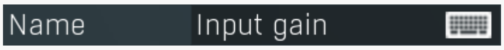


Set center

Set center sets the center (50%) as the origin for the multiparameter. This is often the case for parameters such as gain and panorama and is the only one supported by knobs, so it deserves a dedicated button.

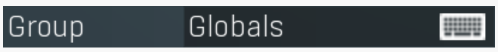
Appearance





Name

Name specifies the name of the multiparameter, which is shown on the multiparameter button. The name is also used for devices - the multiparameter serves as a parameter for the device (on the Easy screen). If no name is specified or if the first character is an *, then the parameter is hidden. This is useful if you need some internal multiparameters which you don't want to show on the Easy screen for some reason.



Group

Group can be used to put some multiparameters into the same group, which results in them being placed in the same panel on the Easy screen (the device editor). Additionally you can actually place the groups into tabs by setting group to "tabname#groupname". The name of the tab needs to be there only for the first parameter of the new group. This makes it possible to build a complex devices with dozens of parameters.

Editor mode

Editor mode controls the way the multiparameter are to be displayed on the Easy screen.
Normal is the default mode and is represented by a small knob or button.
Big mode is similar, but uses a big knob or big button.
Button mode displays a value button, which is usually more compact than knobs.
Check-boxes makes the multiparameter displayed as a set of checkboxes (also called radio buttons). It is relevant only in **Banks** mode.
Check-boxes horiz & below is similar but displays the checkboxes in a single row, hence horizontally. Below mark makes the label underneath the actual checkbox.
Switcher and Selectors are useful for selecting a number of discrete values and similarly to check-boxes these are working only in

Banks mode.
Title button places the control into the title bar of the panel to which it belongs.
Title enable button places the control into the title bar of the panel to which it belongs and makes it a standard enable button (which also makes all controls within the panel unavailable if it is itself disabled).
XY pad creates a 2 dimensional XY pad control, that edits this multiparameter in the X axis and the next multiparameter in the Y axis. There are multiple versions of this control, all of them differ only by size.
Spacer is a helper mode for device design, which doesn't display anything and only keeps empty space.
Meter creates a simple meter instead. You will probably want to set the multiparameter to Meter mode as well or attach it to a modulator. Meters don't really control anything and their purpose is purely to get a visual feedback. The meters can be horizontal or vertical and they can be up or down. Up is the usual choice useful for peak meters for example. Down is useful for gain reduction meters.
Bars start/end mode creates an editor, similar to step sequencer editor, where each parameter has its own bar. The **Bars start** starts the editor and all multiparameters are then added to it until a multiparameter with **Bars end** mode is found or until there are no remaining multiparameters. Note that this kind of editor doesn't show units and may have several other limitations.
Order is a very specific editor for Order modules available in modular systems such as MXXX. It lets you provide an processing order editor on the easy screen. To use it, attach the MP to Order parameter of the Order module and edit the MP information field, so that it contains all the items to be ordered, separated by ';'. The number of items must match the number of items in the Order module, otherwise the order won't work properly. You can also include colors for each item separated by #. These can be specified using hexadecimal numbers, or you can even use standard Melda categories of following set: Dynamics, Distortion, Modulation, Stereo, Spectral, Synthesis, Instrument, MDrummer, Reverb, Delay, EQ, Filter, Saturation, Limit, Time, Pitch, FX.
Example of the info: Compressor#Dynamics;EQ#EQ;Limiter#007F7F;Something

Panel type

Panel type defines the type of panel in which multiple controls of the same group are placed. These differ only in their graphics display.

Color

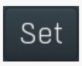
Color defines colorization for the element on the Easy screen (if the plugin has one). The feature is disabled if the Alpha value of the color is 0. Using this feature often increases memory consumption of the plugin, so make sure you use it only if necessary and try to use as low a number of different colors as possible. It is recommended to use only the snapshot colors to make sure the same colors are used in most cases, reducing the memory consumption. It is also highly recommended to use colors with a value (lightness) of 128 (the middle value), which makes sure that the lightness of the elements won't be changed. This works best for most styles. Please note that the style may be configured to simply ignore this color, so there may be no change at all. If you use this feature, make sure that you test it with all styles.

For the sake of workflow the colors have predefined meanings. It's highly recommended to follow this standard:
Orange - dynamics
Green - equalization, filtering
Brown/yellow - reverb, delay
Blue - modulation
Red - limiting, saturation, distortion
Cyan/yellow - stereo
Purple/pink - time, pitch, unison...
Grey - utilities, tools

Group color

Group color defines colorization for the group panel on the Easy screen (if the plugin has one) and is ignored for all multiparameters except for the first one in a group. The feature is disabled if the Alpha value of the color is 0. Using this feature often increases memory consumption of the plugin, so make sure you use it only if necessary and try to use as low number of different colors as possible. It is recommended to use only the snapshot colors to make sure the same colors are used in most cases, reducing the memory consumption. It is also highly recommended to use colors with a value (lightness) of 128 (the middle value), which makes sure that the lightness of the elements won't be changed. This works best for most styles. Please note that the style may be configured to simply ignore this color, so there may be no change at all. If you use this feature, make sure you test it with all styles.

For the sake of workflow the colors have predefined meanings. It's highly recommended to follow this standard:
Orange - dynamics
Green - equalization, filtering
Brown/yellow - reverb, delay
Blue - modulation
Red - limiting, saturation, distortion
Cyan/yellow - stereo
Purple/pink - time, pitch, unison...
Grey - utilities, tools



Set

Set button sets the color and group color for all multiparameters in the same group. It is pretty sensible to do that as all controls should look similar within each group. This can also be done by editing each parameter, but this way is easier.

Visible

Visible checkbox controls if the parameter is visible on the Easy screen (if the plugin has one). Its effect is similar to the '*' prefix in the parameter name, but the multiparameter's name is also available to the plug-in host. This is useful when you wish to automate that multiparameter from the host but not show it on the Easy screen. This parameter can also be attached to another multiparameter for example in order to change the GUI somehow.

Enabled

Enabled switch enables/disables the multiparameter. If disabled, it is grayed on the easy screen.

Same row

Same row checkbox defines if the parameter should be displayed next to the previous one on the Easy screen. Otherwise it will be placed on the next row. This setting serves as a hint and the plugin may ignore it, if it is impossible to do.

Resizable X

Resizable X switch lets you specify if the panel could be resized. It is on by default to make sure everything gets resized, however when using multiple panels next to each other, it may be advantageous to disable resizing of some of them to save space. Otherwise each panel's size is proportional to number of controls it contains, which could make some of the panels larger than actually necessary.

Resizable Y

Resizable Y switch lets you specify if the panel could be resized vertically. It is off by default to make sure everything has the minimum size it requires, but for aesthetic reasons you may want to make all groups on the same row the same size even if the controls inside them are not.

Randomizable

Randomizable option defines if the multiparameter can be randomized on the easy screen. You may want to disable this for input/output gain for example.

Show name

Show name option lets you show or hide the name of the multiparameter for some editor modes. The option has no effect for several editor modes.

Stepped / Continuous

Stepped / Continuous option tells the engine that the multiparameter can be in 2 modes, stepped or continuous. If so, it is assumed that you either used **Banks mode** or **Steps** to produce some sort of predefined set of values for the stepped mode. By enabling this option you allow the engine to convert the multiparameter to continous mode by either ignoring the steps or interpolating the bank values. It can be used when designing devices.

Lockable

Lockable option creates a lock button next to the parameter on the Easy screen, allowing the user to browse through presets without this parameter changing. Please note that this feature is available only for some editor modes.
When the parameter is first locked on the Easy screen it is added to the set of lockable parameters (which are listed in the Global Lock window).

Separate collapse

Separate collapse checkbox makes the panel collapsable separately on the Easy screen. By default it is disabled and that makes the engine find all panels on the same row and collapse all of then or none of them.

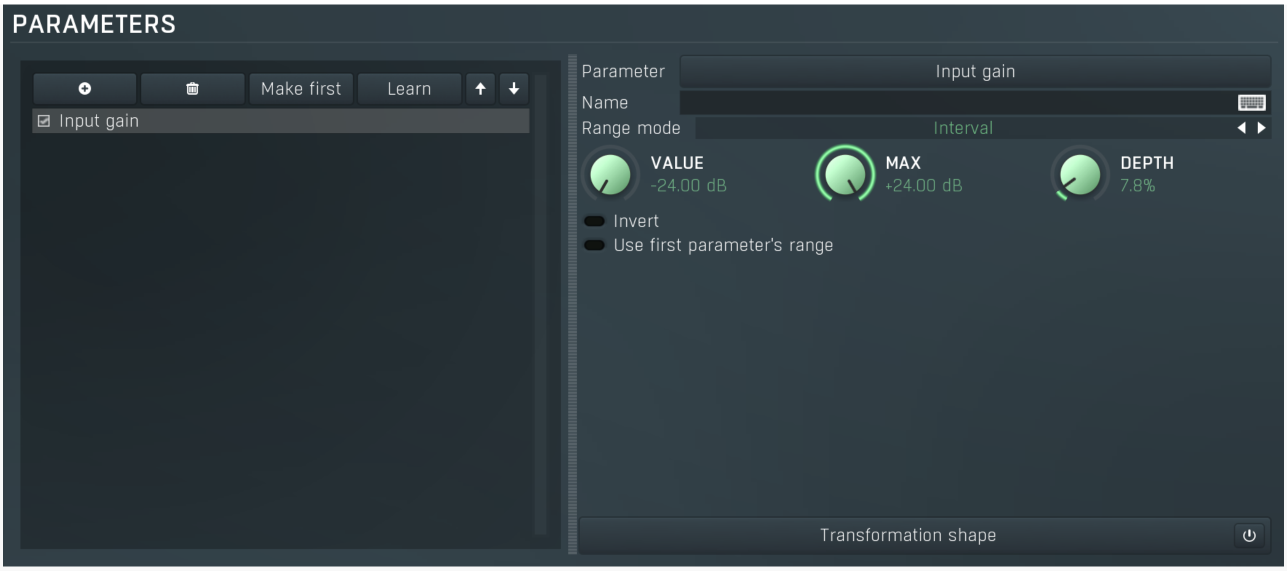
Functional when disabled

Functional when disabled switch makes the multiparameter work even when disabled. This may be useful in some complex scenarios, where you need to make the MP control the target parameters and only use the **Enabled** flag to grey out the controls on the easy screen.

Fine control

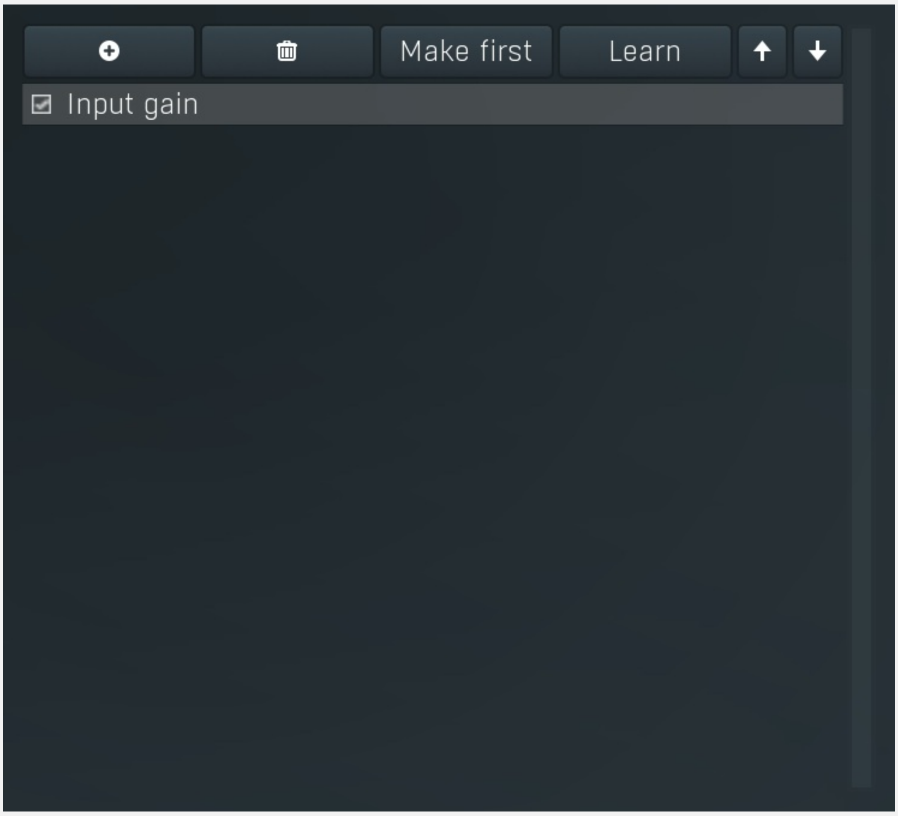
Fine control switch makes the multiparameter editor steps extra small, which is useful, when you need very high precision. This is often handy when using banks mode with many banks interpolating.

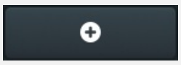
Parameters panel



Parameters panel configures how the multiparameter assigns values to the target parameters.

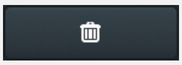
ParameterList





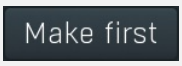
Add

Add button adds a parameter to the list of controlled parameters. Alternatively you can use the learn feature available by right-clicking the multiparameter button.



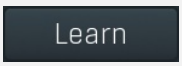
Delete

Delete button deletes the selected parameter from the list of controlled parameters.



Make first

Make first button moves the selected parameter to the first item in the list. This is useful for sake of the **By first parameter** value mode, which makes the multiparameter show the units of the first parameter in the list. Please note that if you have some other multiparameter, modulator or another subsystem access the ranges of individual parameters, this function will reorder them, so these connections will no longer be correct.



Learn

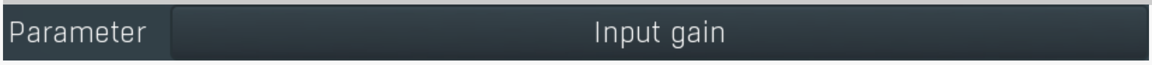
Learn button starts or stops the learning. Click it, then move some parameters in the plugin, then click it again. Learning can also be accessed from the global multiparameter menu.

Up

Up button moves the selected parameter up one item, if possible. This may be useful when keeping things organized, but please note that if you have some other multiparameter, modulator or another subsystem access the ranges of individual parameters, this function will reorder them, so these connections will no longer be correct.

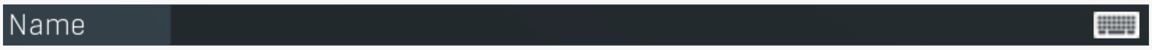
Down

Down button moves the selected parameter down one item, if possible. This may be useful when keeping things organized, but please note that if you have some other multiparameter, modulator or another subsystem access the ranges of individual parameters, this function will reorder them, so these connections will no longer be correct.



Parameter

Parameter defines the target parameter which is being modulated. The set contains all automatable parameters.



Name

Name lets you name the parameter somehow and may be helpful in situations, where there are many parameters being edited without obvious meanings.

Transformation shape

Transformation shape button displays the graph editor, which lets you tweak the shape of the curve used to control the selected parameter. The X axis shows the original values, the Y axis defines the results. Please note that this takes some CPU, therefore you have to enable it using the enable button in the title bar.

Range mode

Range mode defines how the parameter range is selected. While sometimes it is better to specify minimum and maximum, other times it is better to use a nominal center and depth (% of full scale). This control allows you to define which one it will be.

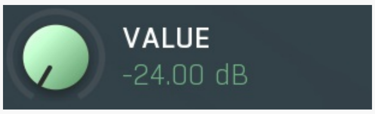
Up and down mode makes the values go above and below the selected **Value**, which is considered the center. The interval is made smaller if necessary.

Full range mode is similar, except the range is symmetrically constrained, so the selected **Value** may not be the center anymore.

Up/down only modes goes from the selected value up/down only.

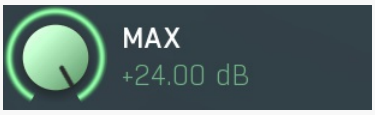
Let's compare these 4 modes. Taking a value of -12dB value, with a depth of 75% and a scale of +/- 24dB. The nominal range is therefore = +/-24 dB * 75% = 36dB. With values of 0%, 50% and 100% the outputs are:
Up and down: -24, -12, 0 (range constrained to 12 dB either side)
Full range: -24, -6, 12 (range limited to minimum, but not constrained)
Up only: -12, 6, 24 (range not constrained = +/-24 dB * 75% = 36dB)
Down only: -12, -18, -24 (range limited to minimum)

Interval mode is the most simple one and goes from **Value** to **Maximal value**.



Value

Value defines the center of the target parameter's range or the minimum if the **Range mode** is set to **Interval**.



Maximal value

Maximal value defines the upper limit of the target parameter's range. It is available only if the **Range mode** is set to **Interval**. This value can be lower than **Value**. 0% is always mapped to reference>Value and 100% to reference>Maximal value.

Depth

Depth defines size of the target parameter's range. It is used only if the **Range mode** is not set to **Interval**.

Invert

Invert checkbox inverts the target parameter's range, so that minimum becomes maximum and vice versa.

Use first parameter's range

Use first parameter's range makes the parameter display use the same range as the first parameter in the list. This is often useful if want to control the range in some way and apply the range to multiple parameters.

Cyclic mode

Cyclic mode switches the multiparameter into so-called cyclic mode. If you have say 4 banks, called A, B, C and D, and gradually increase the multiparameter value, it starts with A, then interpolates to B, then to C and finally to D. But after that you cannot interpolate back to A, because D is the last one, the maximum value. In cyclic mode the multiparameter behaves as if there were a clone of A at the end, hence after D is reached, the multiparameter interpolates back to A and creates a full circle A->B->C->D->A. This is handy for example if you use a saw wave modulator to drive the multiparameter and want to repeat the sequence of the banks.

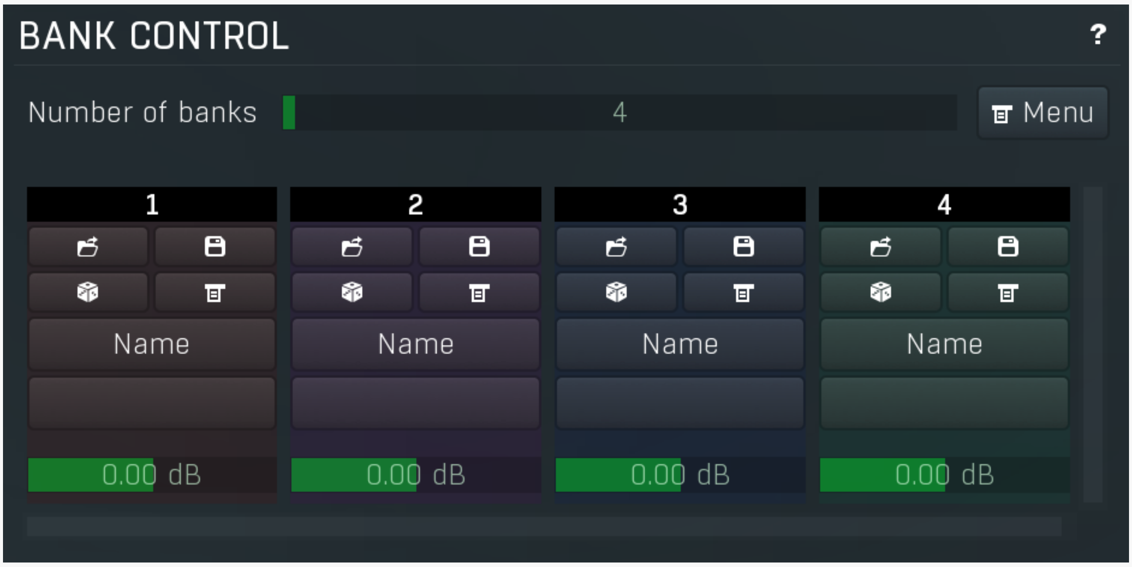
Interpolate values

Interpolate values controls if the parameter value is to be interpolated between the bank values or if it will take the value from the nearest bank. For example, when bank A contains the value 0% for the parameter and bank B contains 100% and you set the multiparameter to 30%, then when interpolation is enabled, 30% is selected for that parameter, when the interpolation is disabled, the nearest value, 0%, is selected. If you want the parameter to step from one bank value to another then disable interpolate values.

Set interpolate to all parameters buttons

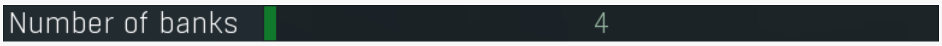
Set interpolate to all parameters buttons sets the interpolate values setting for all parameters controlled by that multiparameter.

Bank control panel



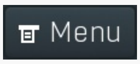
Bank control panel is available only in **Banks mode** and contains tools to define the banks between which the multiparameter is interpolating. The multiparameter stores parameter values for each bank. Here you can load and save these values.
Each bank has 5 buttons and a value for each controlled parameter. Click the **load button** to load the bank values into the plug-in. If you want to change say bank 3, you first click its **load button**, change whatever you need and resave the settings.
By clicking the **save button** you overwrite the bank's settings from those currently set in the plug-in. A typical approach to define the multiparameter's behaviour is to set the number of banks, then go to the plugin editor, set all associated parameters to the values you would like to have in bank 1 and click the save button for bank 1, then modify the parameters to whatever you want in bank 2 and click the save button for bank 2, etc.
You can also use the **Random button** to generate random values using the smart-randomization engine for each of the banks.
And the **menu button** enables you to re-order the banks

For each bank, the values for each parameter are shown and can be changed as desired.



Number of banks

Number of banks controls the number of settings that the multiparameter stores for all parameters. By changing the multiparameter value all associated parameters are then modified according to these settings. Please note that when you change the number of banks, the multiparameter will behave differently, because the multiparameter's range from 0% to 100% will now be distributed between a different number of presets. If you had automated the multiparameter value in your host for example you will almost certainly need to edit / rewrite the automation envelope.



Menu

Menu button provides some additional features for processing the entire set of banks.
Sort banks (up) reorders the banks so that the values of the selected parameter are in increasing order.
Sort banks (down) reorders the banks so that the values of the selected parameter are in decreasing order.
Reverse reverses the order of banks, so that the first bank contains values of the previously last one and so on.
Interpolate lets you change the number of banks, but keeps the values as they are now by calculating values of parameter for all banks. It is usually useful when you want to provide 'banks in between current banks', without manually calculating the new values.
Auto-gain (if available) temporarily enables AGC and automatically sets up the main plugin gain to each bank so that all banks provide similar output loudness. To use it, ensure that the main gain parameter is attached to the multiparameter, start playback of your sound

material and press this button. It will take several seconds to complete depending on the number of the banks.
Set names by values sets the names for each bank to the values of the selected parameter. It may be handy when replicating existing parameters for example.

Load

Load button loads the bank settings by setting all associated parameters to the values in the particular bank.

Save

Save button saves the current values of all associated parameters into the particular bank. So you can edit all those parameters in the plugin then click the save button to store them in the bank.

Randomize

Randomize button loads random settings to the bank using the smart randomization engine. Only parameters associated with the multiparameter are randomized.

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.

Hold **Shift** while clicking the button to undo the previous randomization.

Menu

Menu button provides some additional options related to the bank.

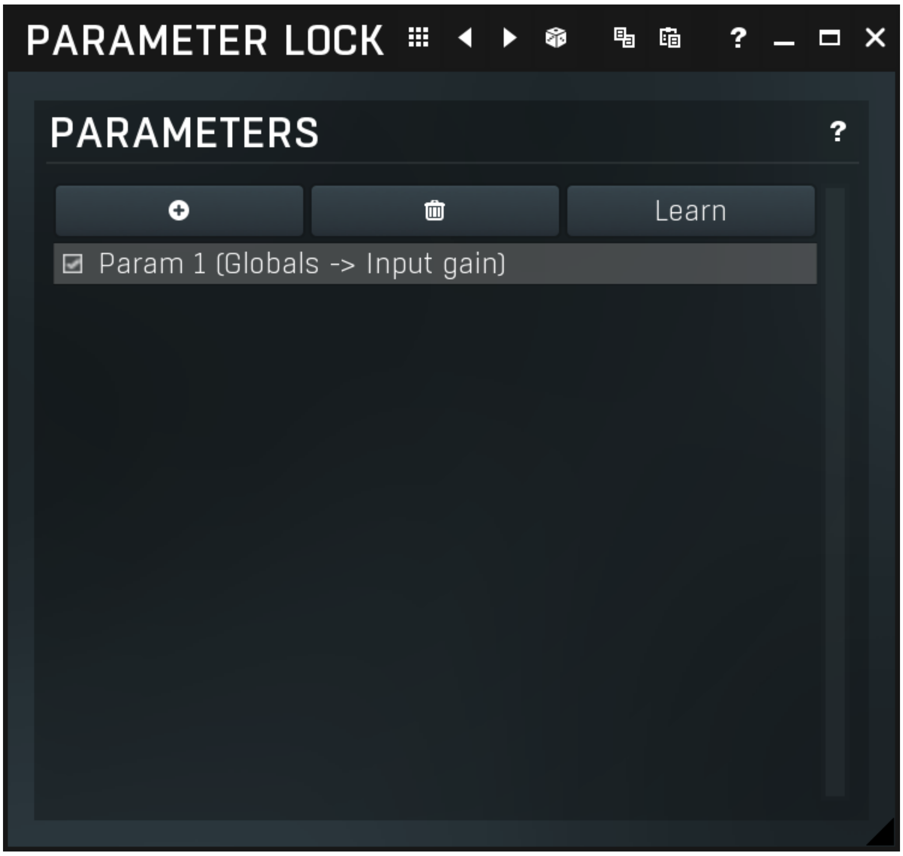
Name

Name button lets you rename the bank.

Name check

Name check button lets you rename the bank. This is a secondary name used for controls such as checkboxes and selectors if defined.

Parameter lock editor



Lock provides a simple way to keep some parameters unchanged when using randomization or browsing presets. You can still change these locked parameters by adjusting the control directly. You simply use the learn feature (right click) in the same way you would with modulators or multiparameters, and touch every parameter you want to keep locked. You can also select them directly in the Parameter Lock window where you can also save them as presets, copy & paste etc. Learning mode is ended by clicking the button again. Please note that this list is not saved with global plugin presets for obvious reasons. The parameters can be locked or unlocked directly in the list or by clicking the lock button associated with the parameter on the Easy screen.

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.

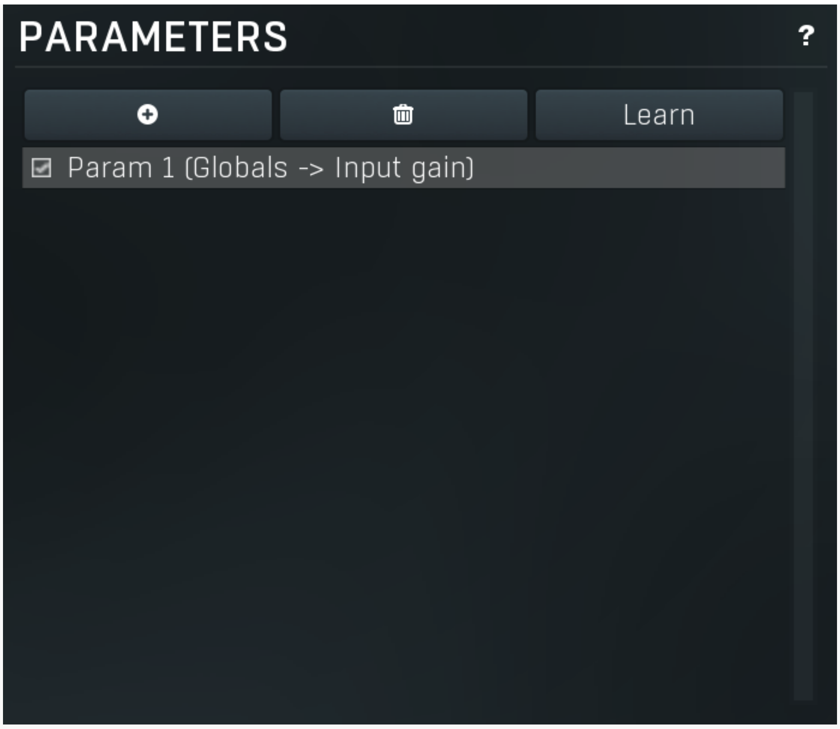
Right arrow
Right arrow button loads the next preset.

Randomize
Randomize button loads a random preset.

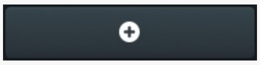
Copy
Copy button copies the settings onto the system clipboard.

Paste
Paste button loads the settings from the system clipboard.

Parameters panel

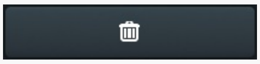


Parameters panel configures the list of the parameters which are locked.



Add

Add button adds a parameter to the list of locked parameters. Alternatively you can use the learn feature available by right-clicking the paramlock button for example.



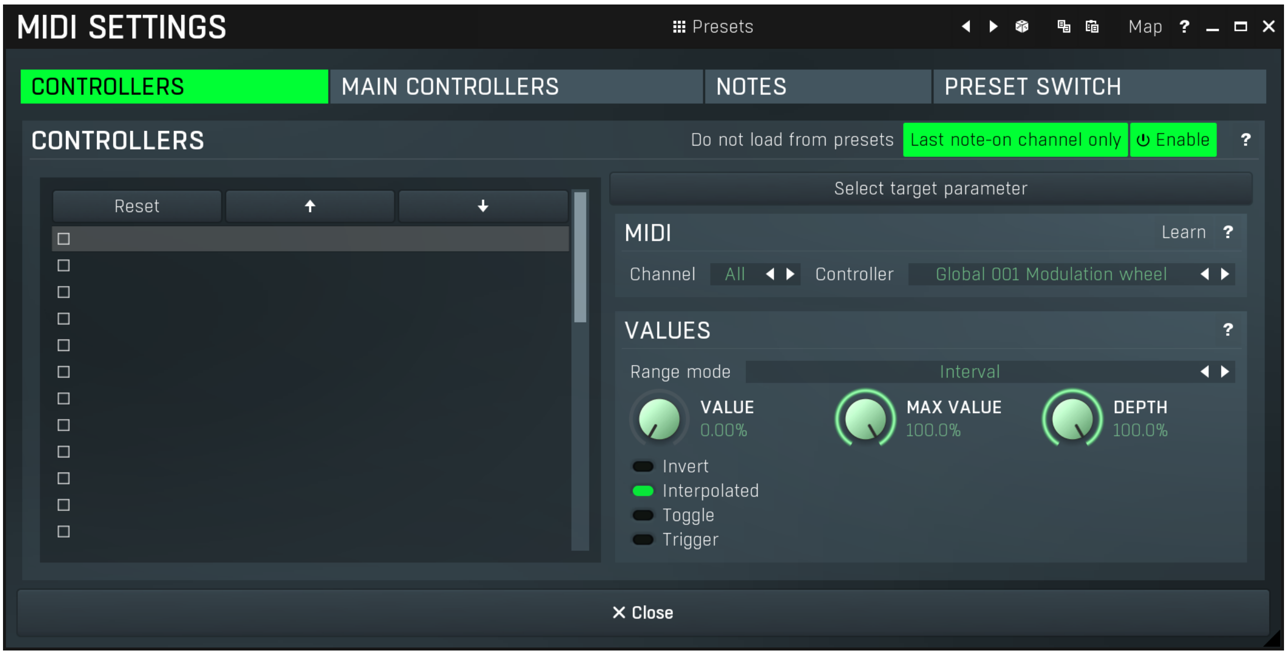
Delete

Delete button deletes the selected parameter from the list of controlled parameters.

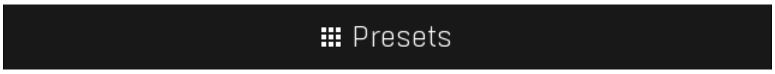
Learn

Learn button starts or stops the learning. Click it, then move some parameters in the plugin, then click it again. Learning can also be accessed from the global parameter lock menu.

MIDI editor



MIDI settings window lets you configure, how the plugin reacts to various MIDI messages. You can use MIDI controllers or MIDI notes and you can also configure a controller to switch between presets, which is especially useful for realtime performances.



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

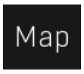
Randomize button loads a random preset.

Copy

Copy button copies the settings onto the system clipboard.

Paste

Paste button loads the settings from the system clipboard.



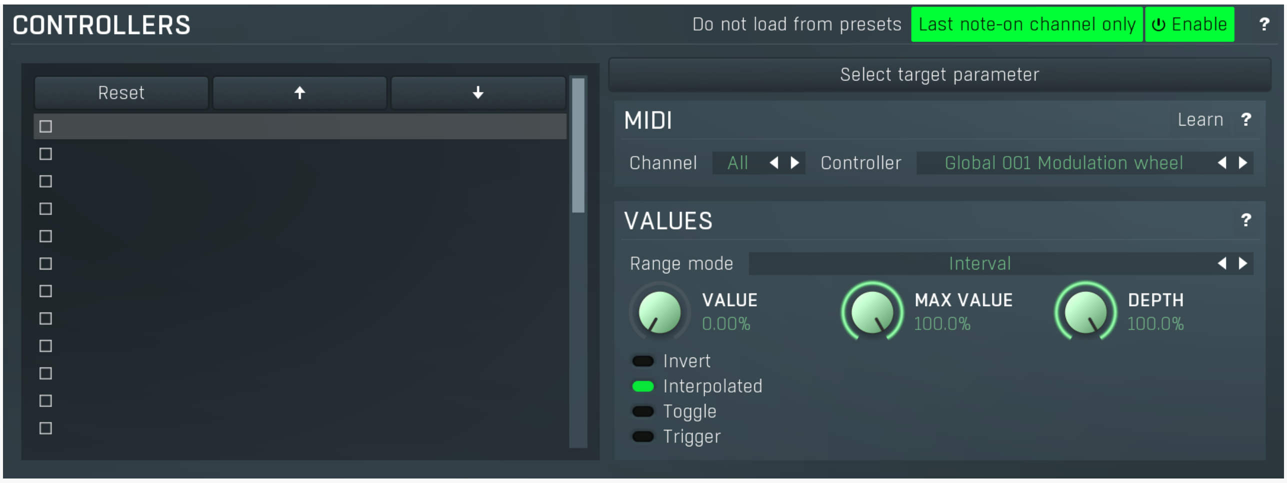
Map

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

Tab selector

Tab selector switches between subsections.

Controllers panel



Controllers panel contains settings of MIDI controllers.

Do not load from presets

Do not load from presets button disables loading the controllers from presets. This may be handy if you have configured specific MIDI controllers with target parameters and you want to browse the presets without the need to configure them every time. Please note that some presets may rely on specific controllers though. For example, if a preset requires a velocity controller to provide velocity-dependent response, this option will avoid loading it, so the preset won't be complete, until you reconfigure it.

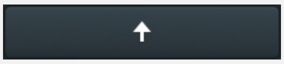
Last note-on channel only

Last note-on channel only button makes the engine more suitable for voice-per-channel devices. These devices are able to send different controllers for each note you press, which however means that these could collide. This option makes the engine pass only the controllers that are related to the last note you pressed. For classic keyboards it is not relevant as you will usually use a single MIDI channel to transmit both the controllers and notes. Some more modern keyboard controllers will allow you to select one MIDI channel for the notes and a different one (or the same one) for the controllers.

Controllers

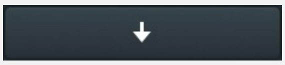
Reset

Reset button resets the selected controller to undefined state.



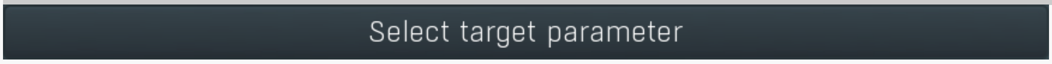
Up

Up button moves the selected controller up one item, if possible. This may be useful when keeping things organized, but please note that if you have some multiparameter, modulator or another subsystem access the ranges of individual controllers, this function will reorder them, so these connections will no longer be correct.



Down

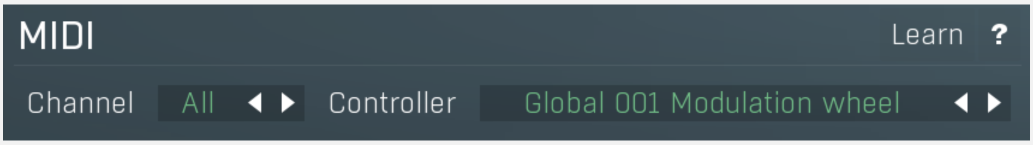
Down button moves the selected controller down one item, if possible. This may be useful when keeping things organized, but please note that if you have some multiparameter, modulator or another subsystem access the ranges of individual controllers, this function will reorder them, so these connections will no longer be correct.

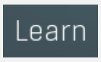


ParameterIndex

ParameterIndex button lets you choose the parameter being controlled. The set contains all automatable parameters.

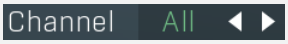
MIDI





Learn

Learn enables or disables MIDI learn. When enabled, the plugin listens to both the MIDI CC messages from the controllers that you touch and the target parameters that you touch and associates the last-touched of each with the selected slot. You can perform several mappings by selecting another slot, adjusting a hardware controller. Then adjusting a target parameter, and repeating those steps for each mapping desired.



Channel

Channel defines the controller MIDI channel.

Controller

Controller defines the source controller.

Values

Range mode

Range mode defines how the parameter range is selected. While sometimes it is better to specify minimum and maximum, other times it is better to use a nominal center and depth (% of full scale). This control allows you to define which one it will be.

Up and down mode makes the values go above and below the selected **Value**, which is considered the center. The interval is made smaller if necessary.

Full range mode is similar, except the range is symmetrically constrained, so the selected **Value** may not be the center anymore.

Up/down only modes goes from the selected value up/down only.

Let's compare these 4 modes. Taking a value of -12dB value, with a depth of 75% and a scale of +/- 24dB. The nominal range is therefore = +/-24 dB * 75% = 36dB. With values of 0%, 50% and 100% the outputs are:
Up and down: -24, -12, 0 (range constrained to 12 dB either side)
Full range: -24, -6, 12 (range limited to minimum, but not constrained)
Up only: -12, 6, 24 (range not constrained = +/-24 dB * 75% = 36dB)
Down only: -12, -18, -24 (range limited to minimum)

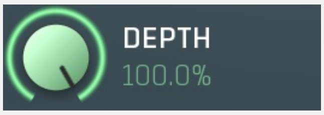
Interval mode is the most simple one and goes from **Value** to **Maximal value**.

Value

Value defines the center of the target parameter's range or the minimum if the **Range mode** is set to **Interval**.

Maximal value

Maximal value defines the upper limit of the target parameter's range. It is available only if the **Range mode** is set to **Interval**. This value can be lower than **Value**. 0% is always mapped to reference>Value and 100% to reference>Maximal value.



Depth

Depth defines size of the target parameter's range. It is used only if the **Range mode** is not set to **Interval**.

Invert

Invert checkbox inverts the controller shape, so the minimum becomes the maximum etc.

Interpolated

Interpolated makes the controller value interpolated over the time using the smart interpolation. This approach ensures there won't be abrupt changes, which could lead to clicks and pops. However sometimes you may want to apply these changes immediately - for example when changing ADSR based on the note velocity, in which case this parameter should be disabled.

Toggle

Toggle mode makes the controller switch between the maximum and minimum of the target parameter whenever triggered. By default triggering it means going from values below 50% to above 50%. By enabling **Trigger** you can make it perform the trigger everytime the value is changed.

Trigger

Trigger mode makes the controller automatically produce maximum and the minimum right after it. It can be handy with some buggy MIDI controllers providing buttons, which however do not send value 0, and only repeat value 127. Trigger makes it behave like the minimum was actually sent by the MIDI controller a little bit after the original message.

Main controllers panel

Main controllers panel lets you define the set of main MIDI controllers on your MIDI device. These are not stored with the presets, so using them lets you easily switch between MIDI controllers, create presets that will work for users of other MIDI controllers etc. Using the Main controllers is no different than using the standard MIDI controllers, but the extra 'layer' can make things simple when using multiple controllers and also in general situations where your MIDI device has several controllers with quite 'random' numbers.

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.

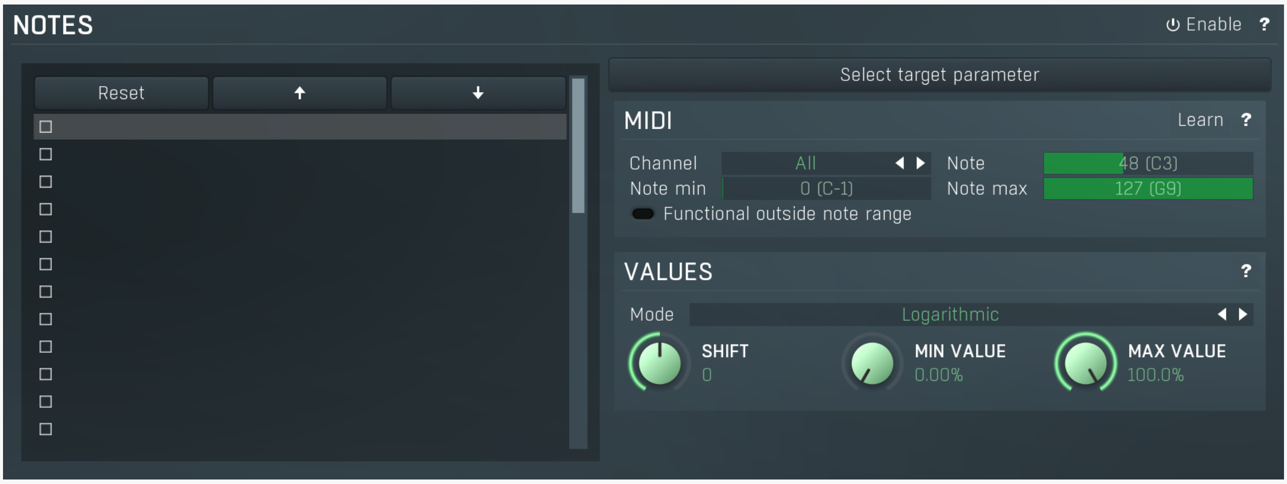
Controller

Controller defines the MIDI controller associated to this Main controller.

Learn

Learn enables or disables MIDI learn. When enabled, the plugin listens to the controllers you touch and associates them to the main controller.

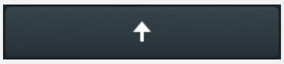
Notes panel



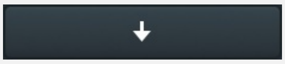
Notes panel contains settings of MIDI note controllers, if you want to control parameters using MIDI keys.

NoteControllers

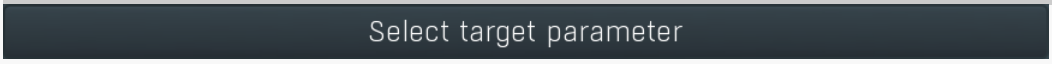
Reset
Reset button resets the selected controller to undefined state.



Up
Up button moves the selected controller up one item, if possible. This may be useful when keeping things organized, but please note that if you have some multiparameter, modulator or another subsystem access the ranges of individual controllers, this function will reorder them, so these connections will no longer be correct.

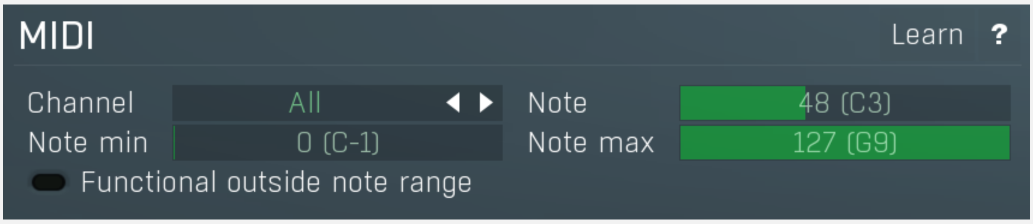


Down
Down button moves the selected controller down one item, if possible. This may be useful when keeping things organized, but please note that if you have some multiparameter, modulator or another subsystem access the ranges of individual controllers, this function will reorder them, so these connections will no longer be correct.



Learn
Learn enables or disables MIDI learn. When enabled, the plugin listens to both the notes you touch and the parameters you touch and associates them with the selected slot.

MIDI



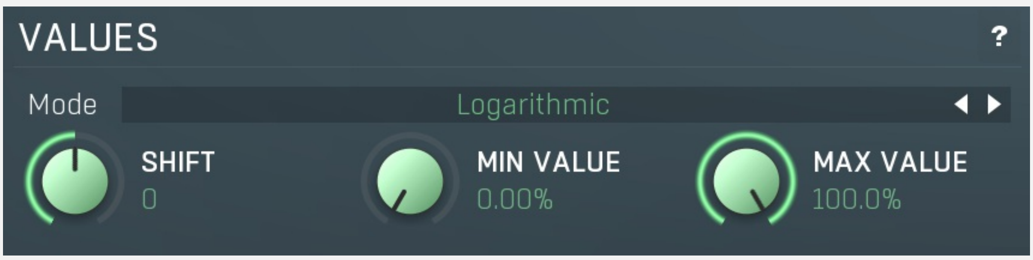
Channel
Channel defines the controller MIDI channel.

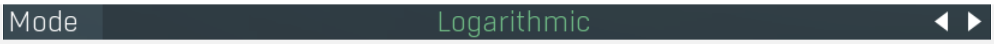
Note
Note defines the controller's target MIDI note. It is used only in On/off and Switch modes, which you can set using **Mode** parameter (in the **Values** panel).

Note min
Note min controls the lowest note to be used by a controller in Linear or Logarithmic mode. The minimum value of the target parameter will then be associated to this note.
If both Note min and Note max parameters are default, the plugin takes the actual frequency of each note, and transforms it into the range 20Hz to 20kHz, which is the range used by all equalizers and filters, so that you can literally play a parameter on a MIDI keyboard. If you change either of these 2 parameters however, the plugin takes the range of notes as the requested interval. This is useful for example if you have a small MIDI keyboard used for soloing and you want increase some parameter the higher you play. In the default mode it would be difficult, since the range of frequencies is much bigger than the range of your MIDI keyboard.*Set the **Note min** and **Note max** to C0 and B0 respectively, the **Mode** to Logarithmic and select a suitable target parameter (Dry/Wet is fine). Send MIDI notes in the specified range to the plugin and you will see the target parameter increase (by 9.09% (= 100 / (12-1)) for a 100% range).*

Functional outside note range
Functional outside note range makes the note controller work even if the note isn't in the specified range, clamping the value to the minimum or maximum.

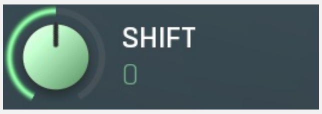
Values





Mode

Mode controls how the controller works.
Key takes the note index and transforms it into 0..1, which is the output of any controller. This mode is useful for scale switches for example - if you want to use MIDI keys to change values linearily.
Linear converts the notes into frequencies and then transform them into the linear scale from 20Hz to 20kHz.
Logarithmic converts the notes into the frequencies and then into the logarithmic scale from log(20) to log(20000). A typical use case is when you want to control an equalizer band using a MIDI keyboard. Since EQ frequencies work in logarithmic scale, this mode makes both things compatible and the EQ frequency will be set to the note frequency.
On/off modes react only to single notes and can be used for triggers. When the Note On is received the parameter is changed to its **Max value** and when the Note Off is received the parameter is changed to its **Min value**. So this mode can also be used to change between any 2 parameter values.
Switch modes are similar, but only recognize when a note is pressed. The Note Offs are ignored. Note Ons select the **Max value** and **Min value** alternately. In all octaves mode it doesn't matter which octave is used. For example, this is useful when you want to use any note C to switch something on and off.
Velocity modes do not actually follow the note number being pressed, but it's velocity instead. While you can do the same thing with normal MIDI controllers using the special Velocity controllers, this one allows you to select only some notes to follow.



Shift

Shift lets you shift the original note up or down by the specified number of semitones.

Min value

Min value defines the minimum value for the target parameter.

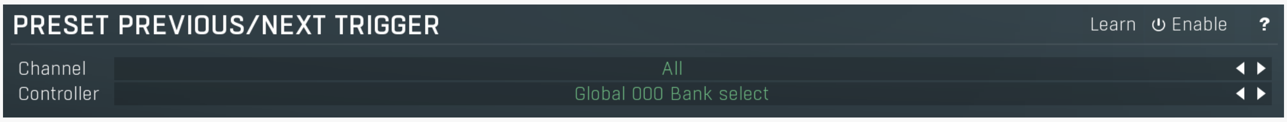
Max value

Max value defines the maximum value for the target parameter.

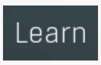
Enable MIDI program change

Enable MIDI program change enables processing program change MIDI message.

Preset previous/next trigger panel



Preset previous/next trigger panel lets you select a MIDI controller, which will switch presets. It provides the same action as clicking the arrows next to the main preset button. When the controller value gets below 33%, the previous preset is loaded. When the controller value gets above 66%, the next preset is loaded.



Learn

Learn enables or disables MIDI learn.

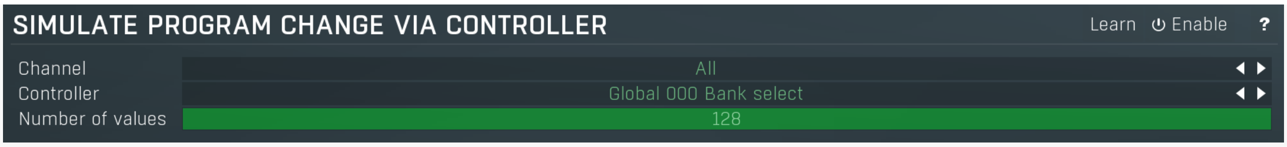
Channel

Channel defines the controller MIDI channel.

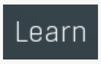
Controller

Controller defines the source controller.

Simulate program change via controller panel



Simulate program change via controller panel lets you select a MIDI controller, that will work as program change, for convenience. You can use it then to switch between A-H presets or presets via panel below.

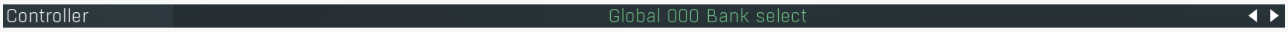


Learn

Learn enables or disables MIDI learn.

Channel

Channel defines the controller MIDI channel.



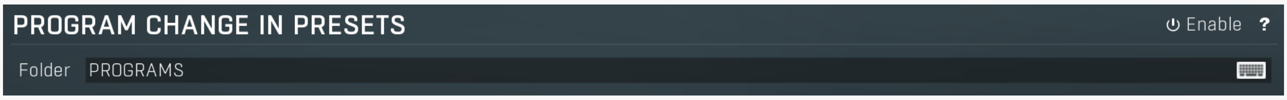
Controller

Controller defines the source controller.

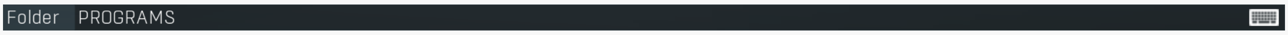
Number of values

Number of values defines the number of programs to switch between. By default Program change MIDI standard offers 128 programs. However it may by too many and could be hard to actually control with the specific controller. Hence you can lower the number of actual programs.

Program change in presets panel



Program change in presets panel enables the MIDI program change processing. If disabled, the plugin follows Program Change messages by changing the A-H presets. The obvious disadvantage is that this way there are just 8 presets. By enabling this feature the plugin stops selecting A-H presets and rather loads different presets from the specified preset folder, including all sub-folders. The default folder is called "Programs". To use it, you simply need to create a preset folder called Programs and put the presets into it. Note that the order matters of course. And you can change the folder name at any time, so you can have several sets of selectable presets.



Folder

Folder defines the preset folder from which the presets for program-change MIDI messages are taken.

Used controls

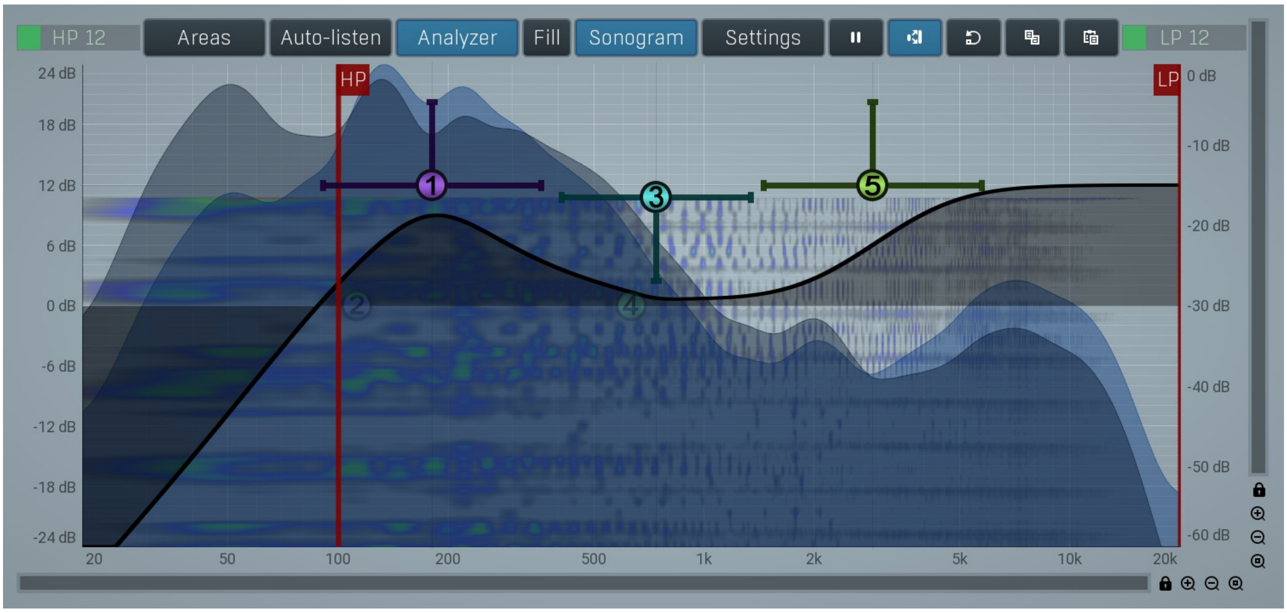
Here we discuss the general properties of all application controls. As a most important rule you should note, that you can always use any question mark button or F1 (or Ctrl+F1 or Ctrl+H) key with the mouse cursor over a specified control to get detailed information about what it does and how to use it.

Value button

Value button is an alternative to the knobs and its main advantage is that it is very small. In some cases the button simply serves as a clickable item and a menu is shown when clicked. However the mouse wheel and other controls still do work.

- **Click and drag using the left mouse button** to change the value.
- **Right mouse button** selects the default value.
- **Mouse wheel**, **arrow keys** and vertical drag using **middle mouse button** or using **left mouse button while holding Ctrl** modifies the value more precisely.
- **Home key** configures the minimal possible value, conversely **end key** setups the maximal one.
- **Esc or Backspace keys** restore the original value when either one is pressed during dragging.
- **Shift + left mouse button** or **double-click using left mouse button** lets you edit the value as text. You can use the virtual keyboard or type on your computer keyboard. In some cases this shows a menu with all possible values instead.
- **Alt + press then release** measures the time between the press and the release and applies it as time/frequency tap. Usable only for certain values of course.

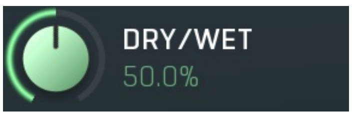
Graph editor



Graph editor will show and edit one or more graphs.

- **Zoomers** below and on the right control the zoom amount and position of the view.
- **Mouse wheel** zooms in or out. Hold **Ctrl** to zoom horizontally, hold **Shift** to zoom vertically. Alternatively you can zoom in using **Alt + right button double click** and out using **Alt + left button double click**. You can also use keyboard **numbers 0 to 9** to quickly set the zoom level.
- **Drag a rectangle using the left mouse button while holding Alt** zooms into the selected rectangle if possible.
- **Drag using the left mouse button while holding Alt and Ctrl** to scroll the view. This is not possible when zoomed all the way out as there is nothing to scroll.

Knob



Knob simulates physical knobs used to edit various values.

- **Click and drag using the left mouse button** to change the value.
- **Right mouse button** selects the default value.

- **Mouse wheel**, **arrow keys** and vertical drag using **middle mouse button** or using **left mouse button while holding Ctrl** modifies the value more precisely.
- **Home key** configures the minimal possible value, conversely **end key** setups the maximal one.
- **Esc or Backspace keys** restore the original value when either one is pressed during dragging.
- **Shift + left mouse button** or **double-click using left mouse button** lets you edit the value as text. You can use the virtual keyboard or type on your computer keyboard. In some cases this shows a menu with all possible values instead.
- **Alt + press then release** measures the time between the press and the release and applies it as time/frequency tap. Usable only for certain values of course.

Zoomer

Zoomer provides a simple way to zoom and move in an enlargeable view.

- **Plus button** zooms-in.
- **Minus button** zooms-out.
- **Zoom default button** zooms to the default ratio, which typically means full zoom-out.
- **Lock button** locks the zoom ratio.

Installation, activation, introduction to audio plugins

Installation

All MeldaProduction plugins are currently available for Windows and Mac OS X operating systems, both 32-bit and 64-bit versions. You can download all software directly from our website. Since the installation procedures for the two operating systems are quite different, we will cover each one separately.

The download files for the effects include all the effects plug-ins and MPowerSynth. During the installation process you can select which plug-ins or bundles to install. If you have not licensed all of the plugins in a bundle then you just need to activate each plugin separately.

If you have multiple user accounts on your computer, always install the software under your own account! If you install it under one account and run it under a different one, it may not have access to all the resources (presets for example) or may not even be able to start.

Installation on Windows

All plugins are available for VST, VST3 and AAX interfaces. The installer automatically installs both the 32-bit and 64-bit versions of the plugins.

Note: Always use 32-bit plugins in 32-bit hosts, or 64-bit plugins in 64-bit hosts. 64-bit plugins cannot work in 32-bit hosts even if the operating system is 64-bit. Conversely, never use 32-bit plugins in 64-bit hosts. Otherwise they would have to be 'bridged' and, in some hosts, can become highly unstable.

You can select the destination VST plugins paths on your system. The installer will try to detect your path, however you should check that the correct path has been selected and change it if necessary. In all cases it is highly recommended to use the current standard paths to avoid any installation issues:
32-bit Windows:
C:\Program files\VstPlugins

64-bit Windows:
C:\Program files (x86)\VstPlugins *(for 32-bit plugins)*
C:\Program files\VstPlugins *(for 64-bit plugins)*

If your host provides both VST and VST3 interfaces, VST3 is usually preferable. If a plugin cannot be opened in your host, ensure the plugin file exists in your VST plugin path and that if your host is 32-bit, the plugin is also 32-bit, and vice versa. If you experience any issues, contact our support via info@meldaproduction.com

Installation on Mac OS X

All plugins are available for VST, VST3, AU and AAX interfaces. Installers create both 32-bit and 64-bit versions of the plugins.

If your host provides multiple plugin interface options, VST3 is usually preferable. If you experience any issues, contact our support via info@meldaproduction.com

Most major hosts such as Cubase or Logic should work without problems. In some other hosts the keyboard input may be partly non-functional. In that case you need to use the virtual keyboard available for every text input field. You may also experience various minor graphical glitches, especially during resizing plugin windows. This unfortunately cannot be avoided since it is caused by disorder in Mac OS X.

Uninstallation on Windows

The Uninstaller is available from the Start menu and Control panel, in the same way as for other applications. If you don't have any of these for any reason, go to Program files / MeldaProduction / MAudioPlugins and run setup.exe.

Uninstallation on OSX

The Uninstaller is available from Applications / MeldaProduction / MAudioPlugins / setup.app.

Deleting all data, presets etc.

Even if you uninstall the plugins, some data will be left behind - because of potential crossdependencies or because these are your presets, settings, configurations etc. If you want to wipe out everything, please manually delete following folders:

Windows:
C:\ProgramData\MeldaProduction

C:\Users\{username}\AppData\Roaming\MeldaProduction

OSX:
Macintosh HD/Library/Application support/MeldaProduction/
HOME/Library/Application support/MeldaProduction

Performance precautions

In order to maximize performance of your computer and minimize CPU usage it is necessary to follow a few precautions. The most important thing is to keep your buffer sizes (latency) as high as possible. There is generally no reason to use latency under 256 samples for 44kHz sampling rates (hence 512 for 96kHz etc.). Increasing buffer sizes (hence also latency) highly decreases required CPU power. In rare cases increasing buffer sizes may actually increase CPU power, in which case you can assume your audio interface driver is malfunctioning.

You should also consider using only necessary features. Usually the most CPU demanding features are upsampling and modulation of certain parameters. You can reduce modulation CPU usage at the cost of lower audio quality in Settings/Settings/Modulator protection.

Troubleshooting

The plugins are generally very stable, there are known problems however.

GPU compatibility

The software uses hardware acceleration to move some of the processing (mainly GUI related) from your CPU (processor) to your GPU (graphics processing unit). It is highly recommended to use a new GPU, as it will provide higher performance improvements, and update your GPU drivers. Older GPUs are slower and may not even provide required features, so the software will have to perform all calculations in the main CPU. We also have had extremely bad experiences with GPUs from ATI and despite the fact that software is now probably bulletproof, it is recommended to use NVidia GPUs as there has not been a single case of a problem with them.

If you experience problems with your GPU (crashing, blank/dysfunctional GUI), and that you cannot disable the GPU acceleration from the plugin's Settings window itself, download this file:

http://www.meldaproduction.com/download/GPU.zip

And place the GPU.xml included in the zip into

Windows: C:\Users\{username}\AppData\Roaming\MeldaProduction
Mac OS X: ~/Library/Application support/MeldaProduction

Memory limits of 32-bit platform

Most hosts are now 64-bit ready, however some of them are not or users willingly choose 32-bit edition, because the required plugins are not 64-bit ready yet. All our software is 64-bit ready. Please note that you must NOT use the 64-bit plugins in 32-bit hosts, even if you have a bridge. If you are stuck with a 32-bit host for any reason, note that there is a memory limit (about 1.5 GB), which you may not exceed. This can happen if you load too many samples or different plugins for example. In that case the host may crash. There is no other solution than to use a 64-bit host.

Updating

You can use "Home/Check for updates" feature in any of the plugins. This will check online if there is a newer version available and open the download page if necessary.

To install a newer (or even older) version you simply need to download the newest installer and use it. There is no need to uninstall the previous version, the installer will do that if necessary. You also do not need to worry about your presets when using the installer. Of course, frequent backup of your work is recommended as usual.

Using touch-screen displays

Touch screen displays are supported on Windows 8 and newer and the GUI has been tweaked to provide a good workflow. Up to 16 connections/fingers/inputs are supported. Any input device such as touch-screens, mouse, tablets are supported. These are the main gestures used by the plugins:
- Tap = left click
- Double tap = double click
- Tap & hold and quickly tap next to it with another finger = right click. Tap & hold is a classic right-click gesture, however that doesn't provide a good workflow, so came up with this method, which is much faster and does not collide with functionality of some elements.

Purchasing and activation

You can purchase the plugin from our website or any reseller, however purchasing directly from our website is always the quickest and simplest option. The software is available online only, purchasing is automatic, easy and instant. After the purchase you will immediately receive a keyfile via email. If you do not receive an e-mail within a few minutes after your purchase, firstly check your spam folder and if the email is not present there, contact our support team using **info@meldaproduction.com** so we can send you the licence again.

To activate the software simply **drag & drop the licence file onto the plugin**. Unfortunately some hosts (especially on Mac OS X) either do not allow drag & drop, or make it just too clumsy, so you can use Home/Activate in any of the plugins and follow the instructions. For more information about activation please check the **online video tutorial**.

You are allowed to use the software on all your machines, but only you are allowed to operate the software. The licences are "to-person" as defined in the licence terms, therefore you can use the software on all your computers, but you are the only person allowed to operate them. MeldaProduction can provide a specialized licence for facilities such as schools with different licence terms.

Quick start with your host

In most cases your host will be able to recognize the plugin and be able to open it the same way as it opens other plugins. If it doesn't, ensure you did installation properly as described above and let your host rescan the plugins.

Cubase

Click on an empty slot (in mixer or in track inserts for example) and a menu with available plugins will be displayed. VST2 version is located in MeldaProduction subfolder. However VST3 version is recommended and is located in the correct folder along with Cubase's factory plugins. For example, dynamic processors are available from the "Dynamics" subfolder.

To route an audio to the plugin's **side-chain** (if it has one), you need to use the VST3 version. Enable the side-chain using the arrow button in the Cubase's plugin window title. Then you can route any set of tracks into the plugin's side-chain either by selecting the plugin as the track output or using sends.

To route **MIDI** to the plugin, simply create a new MIDI track and select the plugin as its output.

Logic

Choose an empty insert slot on one of your audio tracks (or instrument tracks for example) and select the plugin from the popup menu. You will find it in the Audio Units / MeldaProduction folder.

To route an audio to the plugin's **side-chain** (if it has one), a side-chain source should be available in the top of the plugin's window, so simply select the source track you want to send to the plugin's side-chain.

To route **MIDI** to the plugin, you need to create a new Instrument track, click on the instrument slot and select the plugin from AU MIDI-controlled Effects / MeldaProduction. The plugin will receive MIDI from that track. Then route the audio you want to process with the plugin to this track.

Studio One

Find the plugin in the Effects list and drag & drop it onto the track you would like to insert the plugin to.

To route an audio track to the plugin's **side-chain** (if it has one), first enable the side-chain using the "Side-chain" button in the Studio One's plugin window title. Then you can route any set of tracks into the plugin's side-chain from the mixer.

To route **MIDI** to the plugin, simply create a new MIDI track and select the plugin as its output.

Digital performer

In the Mixing Board, find an empty slot in the track you would like to insert the plugin to. Click on the field and select the plugin from the effects list.

To route an audio track to the plugin's **side-chain** (if it has one), choose the track you want to send using Side-chain menu, which appears at the top of the DP's plugin window.

To route **MIDI** to the plugin, simply create a new MIDI track in the Track view and select the plugin as its output.

Reaper

Click on an empty slot in the mixer and a window with available plugins will be displayed. Select the plugin you want to open by double clicking on it or using Ok button.

It is highly recommended to select all MeldaProduction plugins in the plugin window the first time you open it, click using your right mouse button and enable "Save minimal undo states". This will disable the problematic Undo feature, which could cause glitches whenever you change certain parameters.

To route an audio track to the plugin's **side-chain** (if it has one), click on I/O button of the side-chain source track in the mixer. Routing window will appear, there you click "Add new send" and select the track the plugin is on. In the created send slot select the channels (after the "=>" mark) for the send, in stereo configuration 3/4 for example. Note that this way the whole track receives the side-chain signal and all plugins with it. It is possible to send it to a single plugin only, but it is more complicated, please check the Reaper's documentation about that.

To route **MIDI** to the plugin, create a new MIDI track and do the same thing as with side-chain, except you don't need to change output channels.

Live

In Session view, select the track you would like to insert the plugin to. At the left top of Ableton Live's interface, click on the Plug-in Device Browser icon (third icon from the top). From the plug-ins list choose the plugin (from MeldaProduction folder), double click on it or drag & drop it into the track.

The X/Y grid usually doesn't provide any parameters of the plugin. This is because the plugins have too many of them, so you have to select them manually. Check Live's documentation for more information.

To route an audio to the plugin's **side-chain** (if it has one), select the track you want to send to the side-chain and in the 'Audio To' menu, choose the audio track that has the plugin on it. Then in the box just below that select the plugin from the menu.

NOTE: Live does NOT support any interface correctly, it doesn't use the reported buses properly, hence it doesn't work with surround capable plugins. Therefore you need to use VST version, which reports only stereo capabilities by default.

To route **MIDI** to the plugin, create a new MIDI track and in the 'MIDI to' menu, choose the audio track that has the plugin on it. Note that in Live only the first plug-in on any track can receive MIDI.

ProTools

In the mixer click an empty slot to insert the plugin to and select the plugin from the tree. The plugin may be present multiple times, once for each channel configuration (mono->stereo etc.). As of now ProTools do not arrange them in the subfolders, which is a workflow dealbreaker, but we cannot do anything about it. The huge empty space on top of each plugin window, which occupies so much of the precious display area, is part of ProTools and every plugin window and again we cannot do anything about it. In some cases you may experience CPU overload messages, in which case please contact Avid for support. Note that ProTools 10 and newer is supported. RTAS compatibility for PT9 and older will never be added.

To route an audio to the plugin's **side-chain** (if it has one), open the plugin, click on the *No key input* button in the plugin title and select the bus you want the audio taken from. You might need to remember the bus number, unless your ProTools version supports bus renaming. ProTools doesn't support stereo (or surround) side-chains at all.

To route **MIDI** to the plugin, create a new MIDI track and in the mixer click the output field for that track and select the plugin, which should already be in the menu.

FL Studio

First make sure plugins are scanned, either a full scan through the Plugin Manager or an automatic fast scan when you open the Plugin Database section of the browser in FL. The scanned plugins will show up in the Plugin Database > Installed section of the FL browser. The Effects and Generators sections in the Plugin Database will show all "favorite" plugins. These can be checked and unchecked in the Plugin Manager or added in some other ways. These favorites also show up in the Add menu, the menu for the "+" button in the channel rack, when you right click an existing channel button to replace or insert, in the plugin slot menu in the mixer and in the plugin picker (F8). The menus with favorite plugins also have a "More" choice that will show all scanned plugins. The full explanation is in our help file, on the page **Installing Plugins**.

To route an audio to the plugin's **side-chain**, first set up the mixer: make sure the track you want to receive audio from is sent to the track the plugin as a sidechain (**help**). Then set up the plugin wrapper: choose the desired input on the **Processing tab** of the wrapper options.

To route **MIDI notes** to the plugin, first configure the sender: choose a MIDI port for the input device in the MIDI settings (for a hardware device), or an output port in the **wrapper options** (for a VST plugin that produces MIDI). For the receiving plugin, set the input port in the wrapper options to the same value you chose in step 1.

To route **MIDI controllers**, the procedure is different. The usual method in FL is to link CC messages to plugin parameters (**help file**). VST plugins will also have 128 CC parameters published (through the wrapper) that can be linkes this way. Those will send the specified CC MIDI message to the plugin, instead of changing a published parameter.

GUI styles, editor modes and colors

MeldaProduction plugins provide a state of the art styling engine, which lets you change the appearance to your liking. The first time you run the plugins a style wizard will appear and let you choose the style and other settings. It may not be available in ProTools and other problematic hosts.

By default each plugin has a certain color scheme, which differs based on what kind of plugin is that. Also, sections of some plugins are colorized differently, again, based on what kind of section is that (this can be disabled in global settings). Despite you can change the colors anyhow you want, it is advantageous to keep the defaults as these are standardized and have predefined meaning, so just by looking at a plugin's color you can immediately say what kind of plugin and section is that. Same rules apply when designing devices for easy screens. This is the current set of colors:

Dynamics = orange

Equalization, filtering = green
Reverb, delay = brown/yellow
Modulation = blue
Distortion, limiting = red
Stereo = cyan/yellow
Time, pitch, unison... = purple/pink
Tools = grey

Special colors:
Synchronization = grey
Detection = blue/green
Side-chain = green
Effects = red
Advanced stuff = grey

MeldaProduction
The only limit is your imagination

About MeldaProduction

The best sound on the market, incredible workflow and versatility beyond your imagination. We create the deepest and the most powerful audio plugins with unbelievable sound and tons of unique features you cannot find anywhere else.

Innovative Thinking

At MeldaProduction, we make the most advanced tools for music production and audio processing. We get inspired by the whole range of tools from the ancient analog gear to the newest digital creations, but we always push forward.
We've always felt the audio industry is extremely conservative, still relying on the prehistoric equipment making the job unnecessarily slow and complicated. That's why we invent new technologies, which make audio processing easier, faster, better sounding and more creative.

Sound Matters

In the world full of audiophiles you just need superb audio quality. And that's why we spend so much time perfecting audio algorithms until they sound unbeatable. Everything from dynamic filters to spectral dynamic processing. Our technologies just sound perfect.

Inspiring User Interface

Modern user interfaces must not only be easy and quick to use, but also versatile and the whole visual appearance should inspire you. MeldaProduction plugins provide the most advanced GUI engine on the market. It is still the first and only GUI engine, which is freely resizable and stylable. Our plugins can look as an ancient vintage gear, if you are working on old-school rock music. Or as super-modern futuristic devices if you are working on modern electronic music.

Easy to Use, Yet Versatile

The only limit is your imagination. Our plugins are with absolutely no doubt the most powerful and versatile tools on the market. Yet we managed to make the plugins easy to use via the devices and smart randomization system. But when you are ready, you are one click away from the endless potential the plugins provide.

Never-Ending Improvements

Most companies create a plugin, sell it and abandon it. We improve our plugins, add features, optimize... until there is nothing left to improve and there are no more ideas. Unfortunately that hasn't happened yet :). And the best thing is that the updates are free-for-life!

MeldaProduction was founded in 2009 by Vojtech Meluzin and is based in Prague, Czech Republic.

www.meldaproduction.com
info@meldaproduction.com