

# MAutoDynamicEq



## 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.

## Easy screen vs. Edit screen

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

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

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

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

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

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

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

# Edit mode



## Presets

### Presets

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

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

Presets can be backed up by 3 different methods:

- A) Using "Backup" and "Restore" buttons in each preset window, which produces a single archive of all presets on the computer.
- B) Using "Export/Import" buttons, which export a single folder of presets for one plugin.
- C) By saving the actual preset files, which are found in the following directories (not recommended):

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

Mac OS X: /Library/Application support/MeldaProduction

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

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



### Left arrow

Left arrow button loads the previous preset.



### Right arrow

Right arrow button loads the next preset.



### Randomize

Randomize button loads a random preset.



### Panic

Panic button resets the plugin state. You can use it to force the plugin to report latency to the host again and to avoid any audio problems. For example, some plugins, having a look-ahead feature, report the size of the look-ahead delay as latency, but it is inconvenient to do that every time the look-ahead changes as it usually causes the playback to stop. After you tweak the latency to the correct value, just click this button to sync the track in time with the others, minimizing phasing artifacts caused by the look-ahead delay mixing with undelayed audio

signals in your host. It may also be necessary to restart playback in your host.

Another example is if some malfunctioning plugin generates extremely high values for the input of this plugin. A potential filter may start generating very high values as well and as a result the playback will stop. You can just click this button to reset the plugin and the playback will start again.

Settings

## Settings

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

**Licence manager** lets you activate/deactivate the plugins and manage subscriptions. While you can simply drag & drop a licence file onto the plugin, in some cases there may be a faster way. For instance, you can enter your user account name and password and the plugin will do all the activating for you.

There are 4 groups of settings, each section has its own detailed help information: **GUI & Style** enables you to pick the GUI style for the plug-in and the main colours used for the background, the title bars of the windows and panels, the text and graphs area and the highlighting (used for enabled buttons, sliders, knobs etc).

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

**Global system settings** contains some settings for all MeldaProduction plugins. Once you change any of them, restart your DAW if needed, and it will affect all MeldaProduction plugins.

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

**Smart interpolation** adjusts the interpolation algorithm used when changing parameter values; the higher the setting the higher the audio quality and the lower the chance of zippering noise, but more CPU will be used.



WWW

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

Sleeping

## Sleep indicator

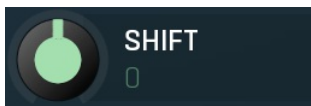
Sleep indicator informs whether the plugin is currently active or in sleep mode. The plugin can automatically switch itself off to save CPU, when there is no input signal and the plugin knows it cannot produce any signal on its own and it generally makes sense. You can disable this in Settings / **Intelligent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



## 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.

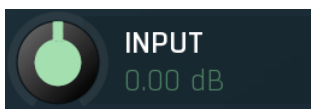
Range: 0.00% to 100.0%, default 100.0%



## Shift

Shift lets you pitch shift all bands by specified number of semitones. It doesn't change the actual band points, but changes the resulting EQ shape appropriately.

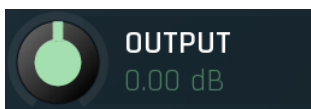
Range: -24.00 to +24.00, default 0



## Input gain

Input gain defines input gain applied before the equalization. Therefore this affects all dynamics-based processes.

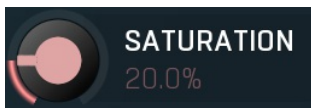
Range: -24.00 dB to +24.00 dB, default 0.00 dB



## 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.

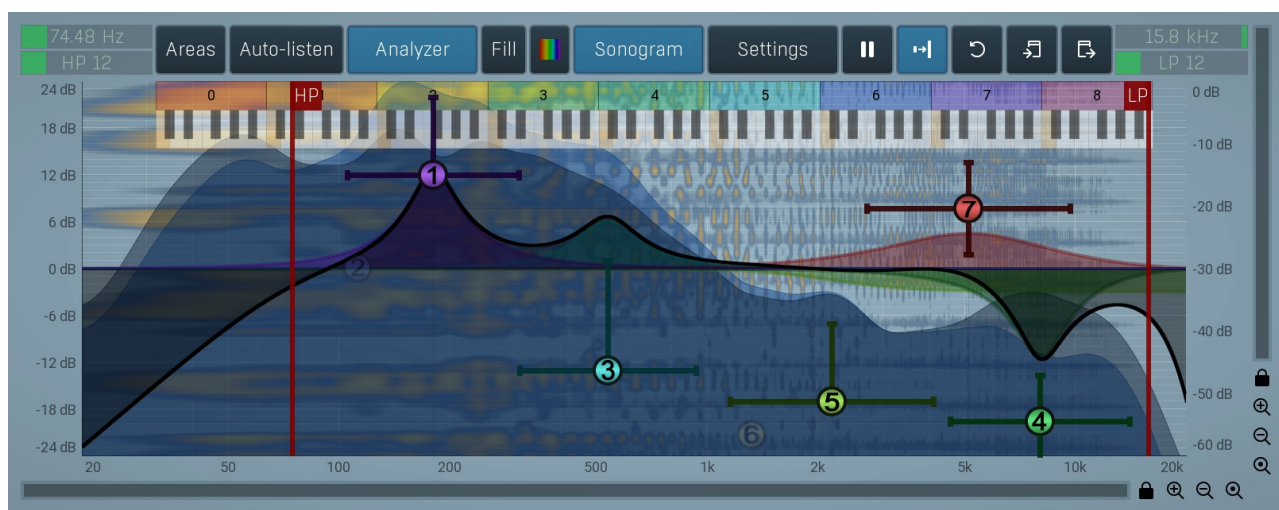
Range: 0.00% to 20.0%, default 5.0%



## Nonlinear gain

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

Range: 0.00% to 100.0%, default 0.00%



## Equalizer shape graph

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

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

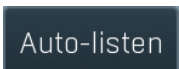
The equalizer graph also contains 2 **red vertical lines** on the right and on the left. These are the high-pass and low-pass filters, conveniently placed so that you can perform this often-used task quickly, efficiently and most of all using the highest quality filters available on the market. The high-pass 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 bandwidth. You can also hold **shift** to get this feature if auto-listen is not enabled.

## Analyzer

### 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

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

An alternative is to use the spectrum sonogram.



### Analyzer Rainbow Colors

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

## 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

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



### Pause

Pause button stops the analyzer temporarily.



### Normalize

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



### Reset

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



### Copy analysis

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




### 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

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



### Left arrow

Left arrow button loads the previous preset.



### Right arrow

Right arrow button loads the next preset.



### Randomize

Randomize button loads a random preset.



### Copy

Copy button copies the settings onto the system clipboard.



### Paste

Paste button loads the settings from the system clipboard.

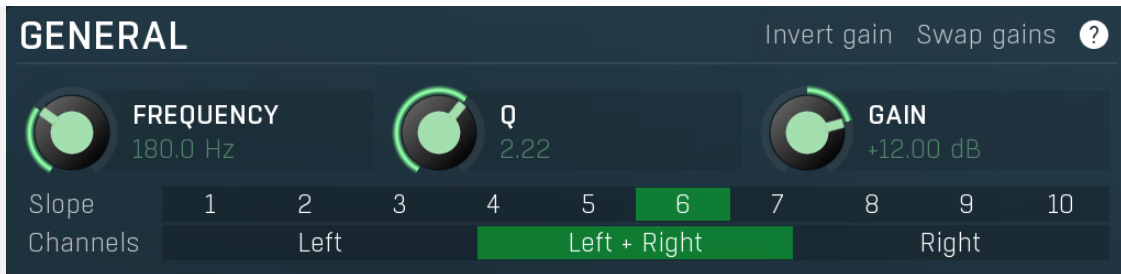


### Random

Random button generates random settings using the existing presets.

## General panel





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

Invert gain

### Invert gain

Invert gain inverts the gain of the band, e.g. makes -6dB from +6dB.

Swap gains

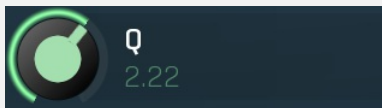
### Swap gains

Swap gains button swaps values between gain and dynamics gain.



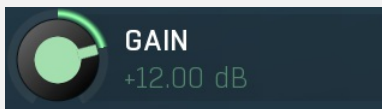
### Frequency

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



### Q

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



### Gain

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



### Slope

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



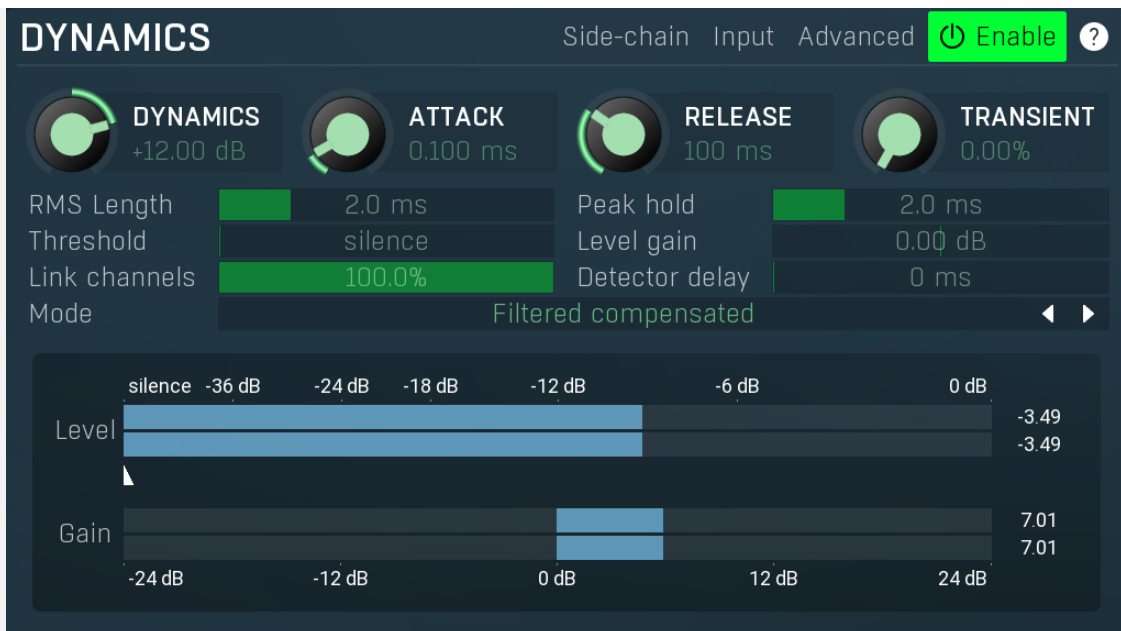
### Channels

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

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

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

## 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.

### Side-chain **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**

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**

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

### Enable **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.

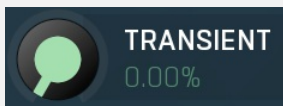
To understand the working of a level detector, it is best to cover the typical cases:

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.



### Transient

Transient lets you mix the level follower output with a transient detector output. This lets you follow signal level, transients or both. Note that since transient level is usually lower than level detector's output, **Level gain** is only applied on the level detector's signal, so you can use this to compensate for the difference in level.



### 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 2.0 ms **Peak hold**

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

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

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

Threshold silence **Threshold**

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

Level gain 0.00 dB **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 100.0% **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 0 ms **Detector delay**

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

Mode Filtered compensated **Mode**

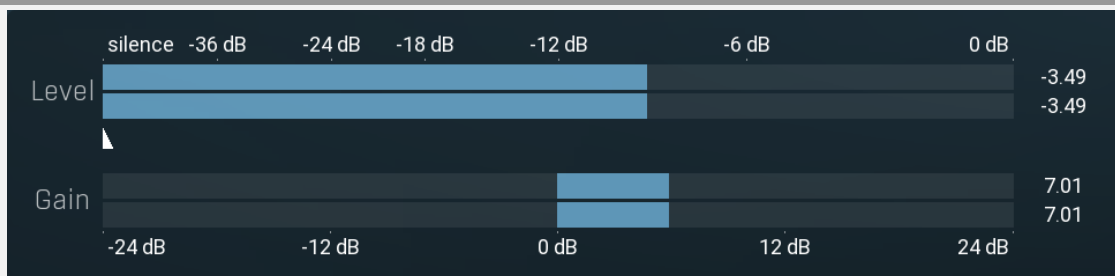
Mode controls the way the band reacts to the input signal. It has no meaning if the dynamic gain is 0dB.

**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

Linear Dynamics by fundamental ?



HARMONICS  
0.00%



SEMITONES  
12.00



MAXIMAL COUNT  
16

Harmonics



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

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

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  
0.00%

### 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  
12.00

### Semitones

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

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



MAXIMAL COUNT  
16

### Maximal count

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

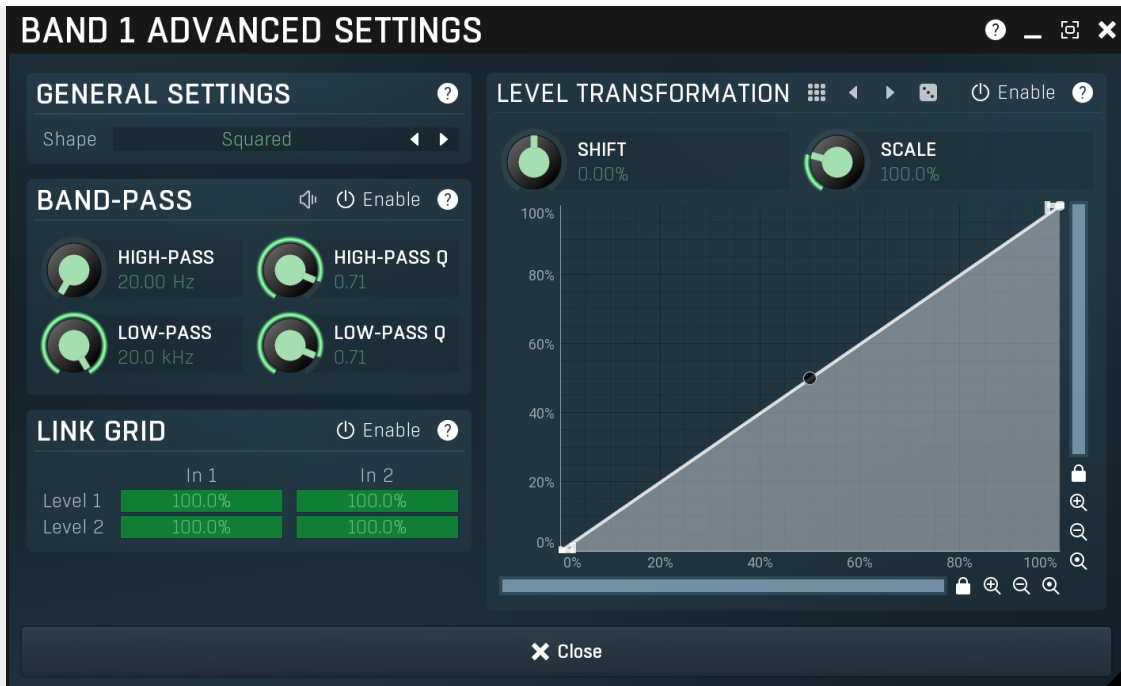
## Harmonics grid

Harmonics



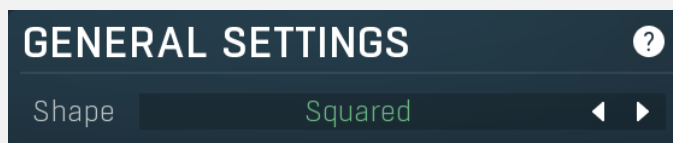
Harmonics grid is useful to turn on/off particular harmonics manually. Click any one to enable / disable it.

## Band advanced settings

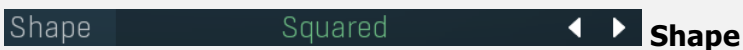


Band advanced settings contains additional settings for the band. These contain some more esoteric features, such as a dynamic transformation shape. It can be displayed by clicking the right mouse button on a band while holding **Ctrl**, from the basic band settings window, or from the band list if provided.

## General settings panel

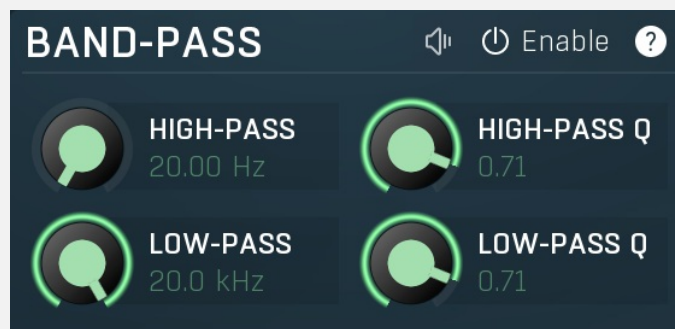


General settings panel contains additional parameters, which are too scientific to be available from the main band settings.



Shape 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.

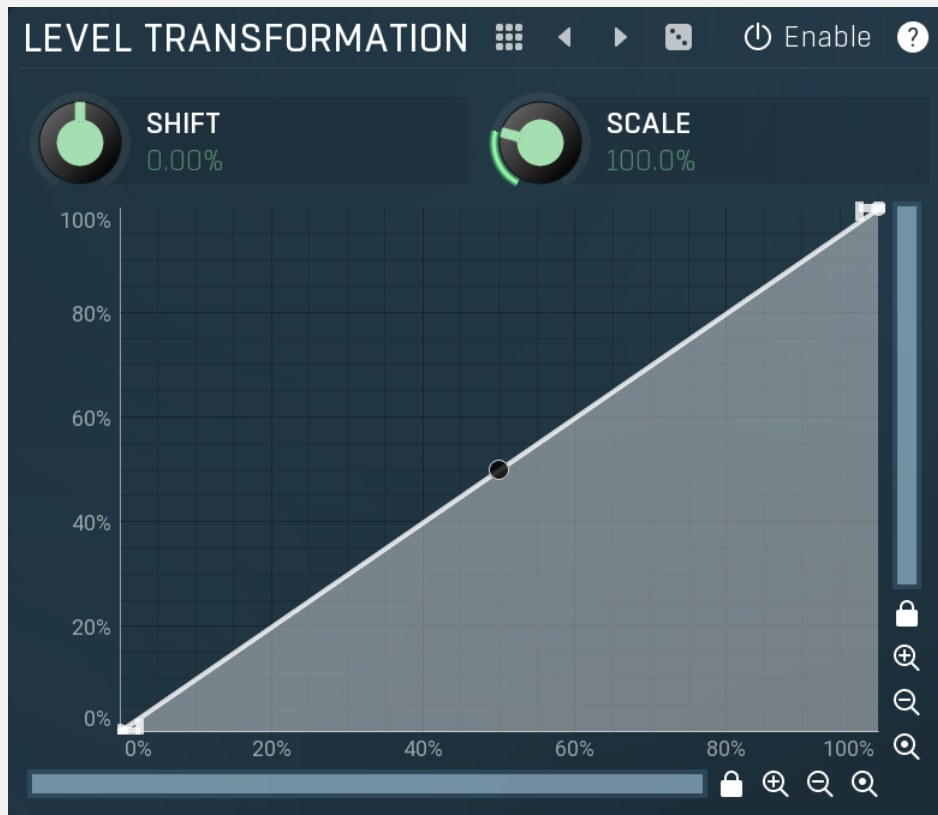


Play button enables the band-pass monitoring and hence could be useful to tweak the band pass.

 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

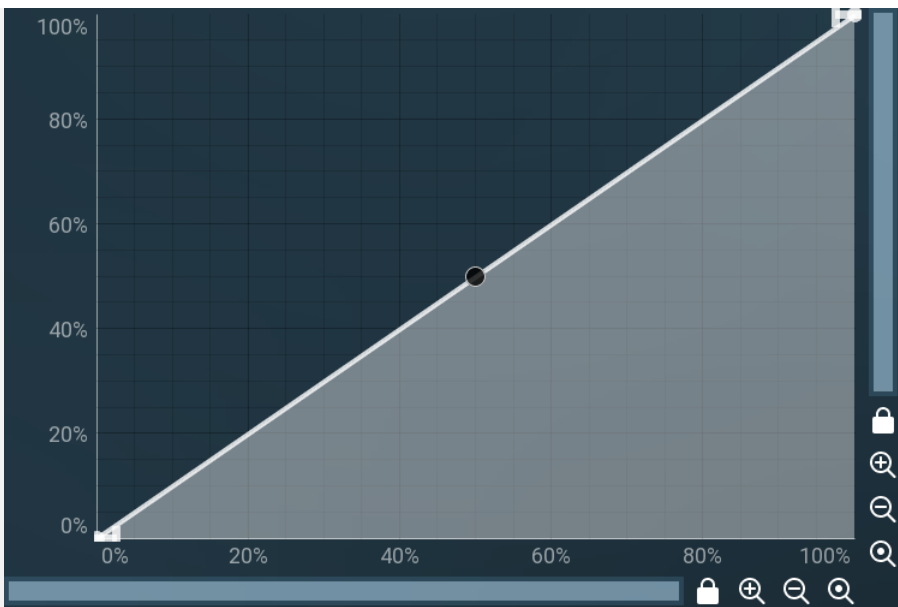
Right arrow button loads the next preset.

### 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.



EnvelopeEditorGraph

## Envelope graph

Envelope graph provides an extremely advanced way to edit any kind of shape that you can imagine. An envelope has a potentially unlimited number of points, connected by several types of curves with adjustable curvature (drag the dot in the middle of each arc) and the surroundings of each point can also be automatically smoothed using the smoothness (horizontal pull rod) control. You can also literally draw the shape in drawing mode (available via the main context menu).

- **Left mouse button** can be used to select points. If there is a *point*, you can move it (or the entire selection) by dragging it. If there is a *curvature circle*, you can set up its tension by dragging it. If there is a *line*, you can drag both edge points of it. If there is a *smoothing controller*, you can drag its size. Hold **Shift** to drag more precisely. Hold **Ctrl** to create a new point and to remove any points above or below.
- **Left mouse button double click** can be used to create a new point. If there is a *point*, it will be removed instead. If there is a *curvature circle*, zero tension will be set. If there is a *smoothing controller*, zero size will be set.
- **Right mouse button** shows a context menu relevant to the object under the cursor or to the entire selection. Hold **Ctrl** to create or remove any points above or below.
- **Middle mouse button** drag creates a new point and removes any points above or below. It is the same as holding Ctrl and dragging using left mouse button.
- **Mouse wheel** over a point modifies its smoothing controller. If no point is selected, then all points are modified.
- **Ctrl+A** selects all points. **Delete** deletes all selected points.

**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			Enable	?
	In 1	In 2		
Level 1	100.0%	100.0%		
Level 2	100.0%	100.0%		

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

## Bands panel



	1	2	3	4	5	6	7
Filter							
Frequency	180.0 Hz	112.5 Hz	535.7 Hz	8030 Hz	2185 Hz	1319 Hz	5113 Hz
Q	2.22	0.71	2.00	1.82	0.70	20.00	0.71
Gain	+12.00 dB	0.00 dB	-13.08 dB	-19.62 dB	-17.05 dB	-21.35 dB	+7.72 dB
Dynamics	+12.00 dB	0.00 dB	+24.00 dB	0.00 dB	+12.00 dB	0.00 dB	0.00 dB
Attack	0.100 ms	Auto	0.100 ms	Auto	0.100 ms	Auto	Auto
Release	100 ms	Auto	100 ms	Auto	100 ms	Auto	Auto
Threshold	silence	silence	silence	silence	silence	silence	silence

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

 **Reset**

Reset button restores the original equalizer settings.

 **Invert**

Invert button inverts the gains of all bands.

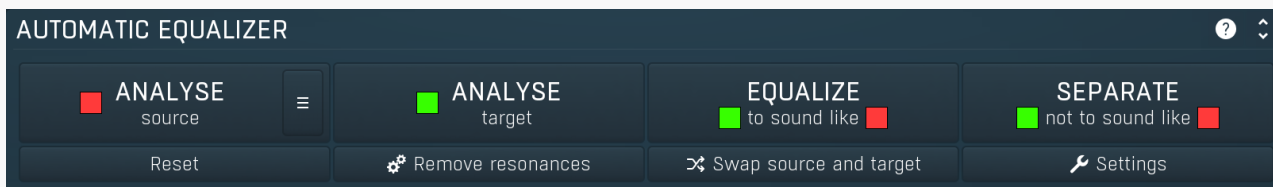
 **Swap gains**

Swap gains button swaps values between gain and dynamics gain.

 **Band graphs**

Band graphs button enables/disables display of band graphs.

## Automatic equalizer panel



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



**Analyze source**

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

Alternatively, you can analyze an audio file (WAV, MP3, FLAC etc.) offline by clicking the **File** button and browsing to the file or by dragging & dropping the file from your host or Explorer / Finder onto the **Analyze source** button (this latter method may not work in all

hosts, especially on Mac). Three other 3 buttons let you save and load an analysis or even draw the desired response manually.

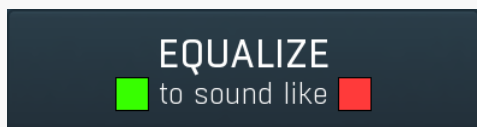


### Analyze target

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

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

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



### Equalize

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

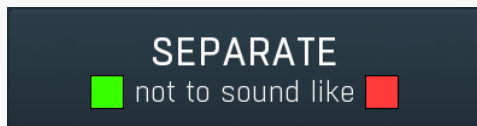
First get the source analysis using the **Analyze source** button. The Source is the reference audio material that you want your track to sound like.

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

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

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

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



### Separate

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

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

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

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



### Reset

Reset button clears both the source and target analyses.



### Remove resonances

Remove resonances button takes the target analysis, tracks peaks in it and creates notch filters to remove them. It is especially useful with various audio materials, such as drums, where resonances of the instruments are too prominent. You simply analyze the part of

the audio material, where the resonances are mostly audible, and click this button. Then you can just disable the bands that remove the frequencies that you actually want to keep. It may also be worth trying to play with the Q values of each band to make sure the resonances are completely gone.

 Swap source and target

### Swap source and target

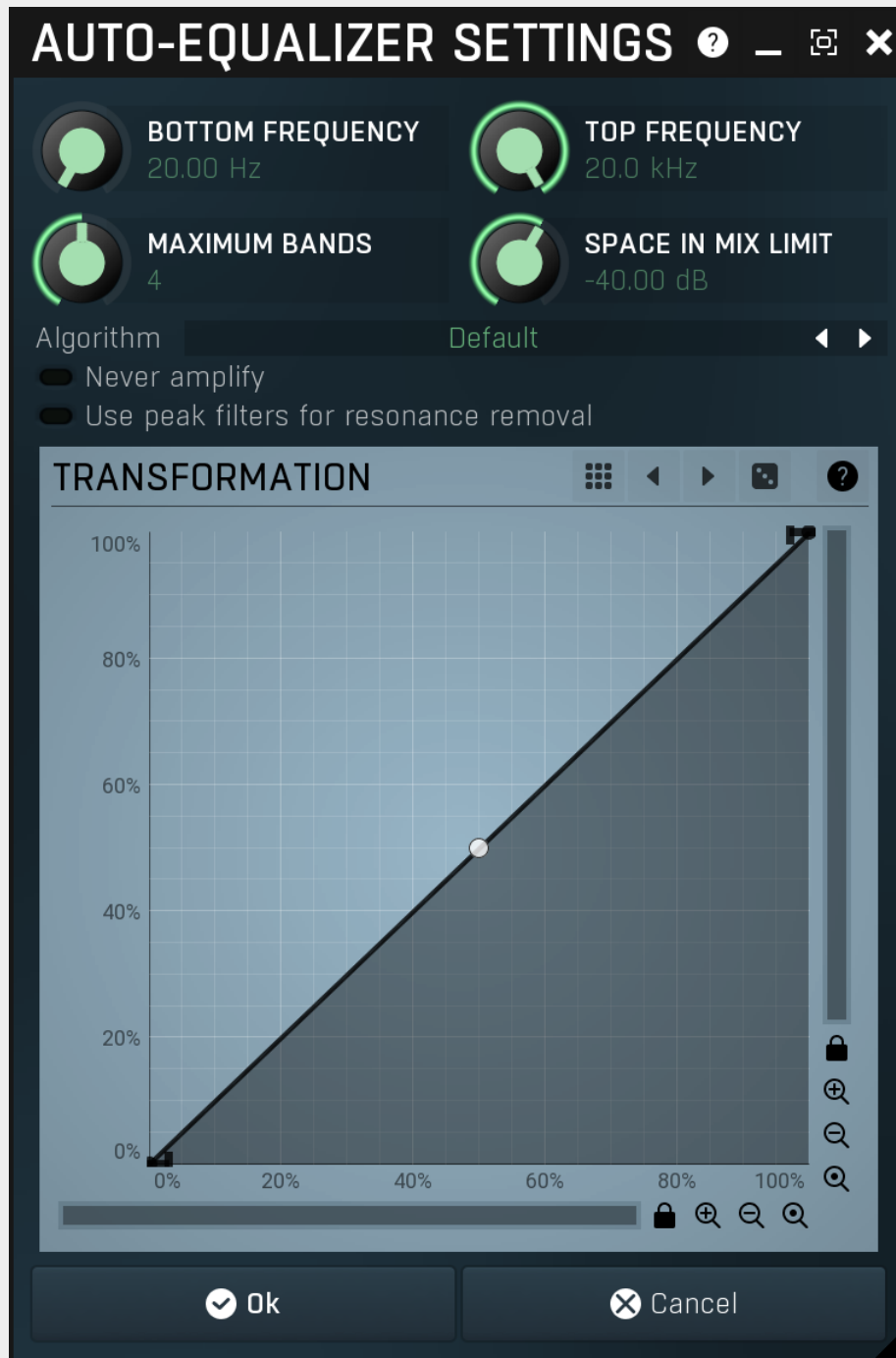
Swap source and target button swaps the analyses of source and target and can be helpful when you want to try equalizing 'the other way around'.

 Settings

### Settings

Settings button shows additional settings of the automatic equalizer algorithm.

## Auto-equalizer settings



**AUTO-EQUALIZER SETTINGS** ? \_ [Maximize] [Close]

**BOTTOM FREQUENCY**  
20.00 Hz

**TOP FREQUENCY**  
20.0 kHz

**MAXIMUM BANDS**  
4

**SPACE IN MIX LIMIT**  
-40.00 dB

Algorithm: Default

Never amplify

Use peak filters for resonance removal

**TRANSFORMATION** [Grid] [Left] [Right] [Reset] [Help]

100%  
80%  
60%  
40%  
20%  
0%

0% 20% 40% 60% 80% 100%

[Lock] [Zoom In] [Zoom Out] [Reset]

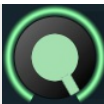
Ok Cancel

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

 **BOTTOM FREQUENCY**  
20.00 Hz

### Bottom frequency

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

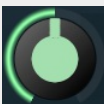


## TOP FREQUENCY

20.0 kHz

### Top frequency

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



## MAXIMUM BANDS

4

### Maximum bands

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



## SPACE IN MIX LIMIT

-40.00 dB

### Space in mix limit

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

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

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

Algorithm

Default



Algorithm

Algorithm selector lets you choose which method is used to convert the frequency response into a configuration of the equalizer bands. (This algorithm itself is actually very complex and requires lots of CPU when adjusting the bands during the equalization/separation processes.) Multiple algorithms are available, each of them with a slightly different output.

**Default** provides the most accurate results, but it takes lots of CPU. Note, as mentioned above, that this CPU usage is required only for the pre-computing, the actual audio processing requirements are the same for all algorithms.

**Optimized** algorithm is much faster, however it can produce slightly less accurate results. It may be more suitable because it smoothes out the peaks in the frequency response.

**Super-fast** mode is the fastest of all of them, but it trades speed for accuracy in its results. You may find it useful in specific cases.



Never amplify

### Never amplify

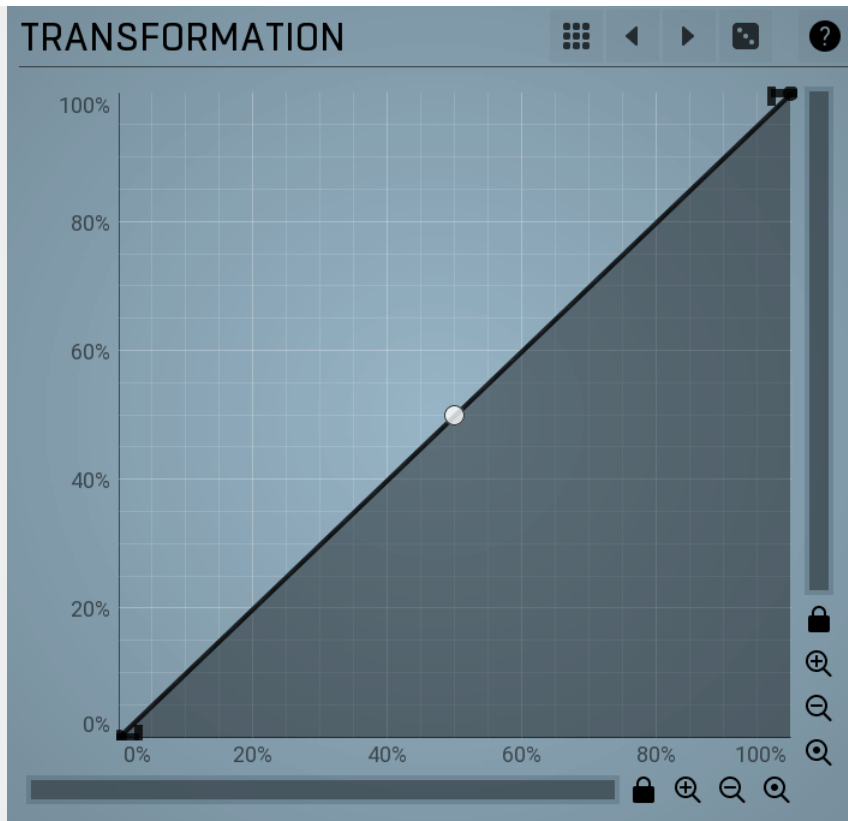
Never amplify ensures that the automatic equalization only attenuates frequencies so that no amplification occurs. This may be desired, since amplification often increases noise level. On the other hand, due to its nature, it will probably lower the output level. It may be worth considering using AGC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



Use peak filters for resonance removal

### Use peak filters for resonance removal

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



## Transformation

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



### Presets

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



### Left arrow

Left arrow button loads the previous preset.



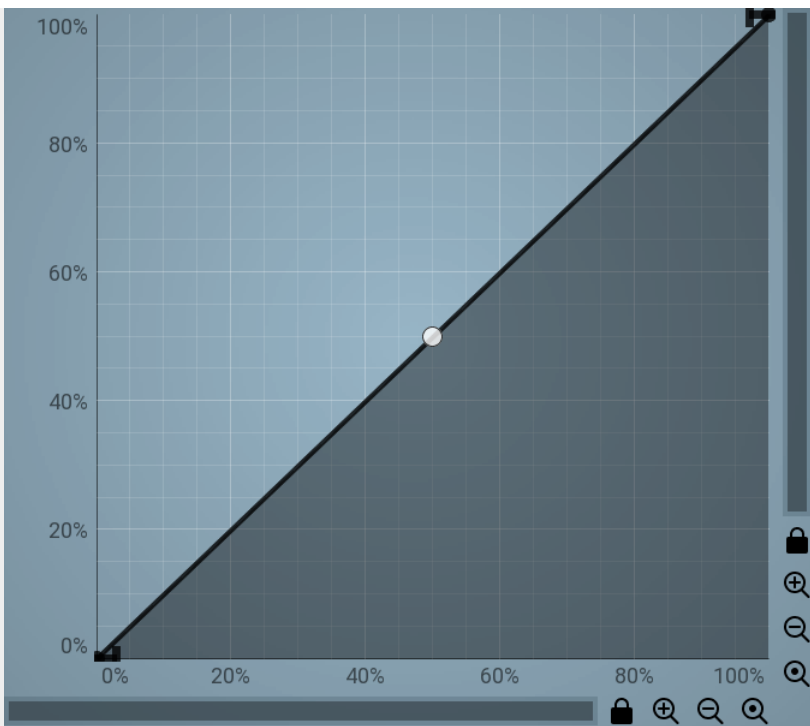
### Right arrow

Right arrow button loads the next preset.



### Randomize

Randomize button loads a random preset.

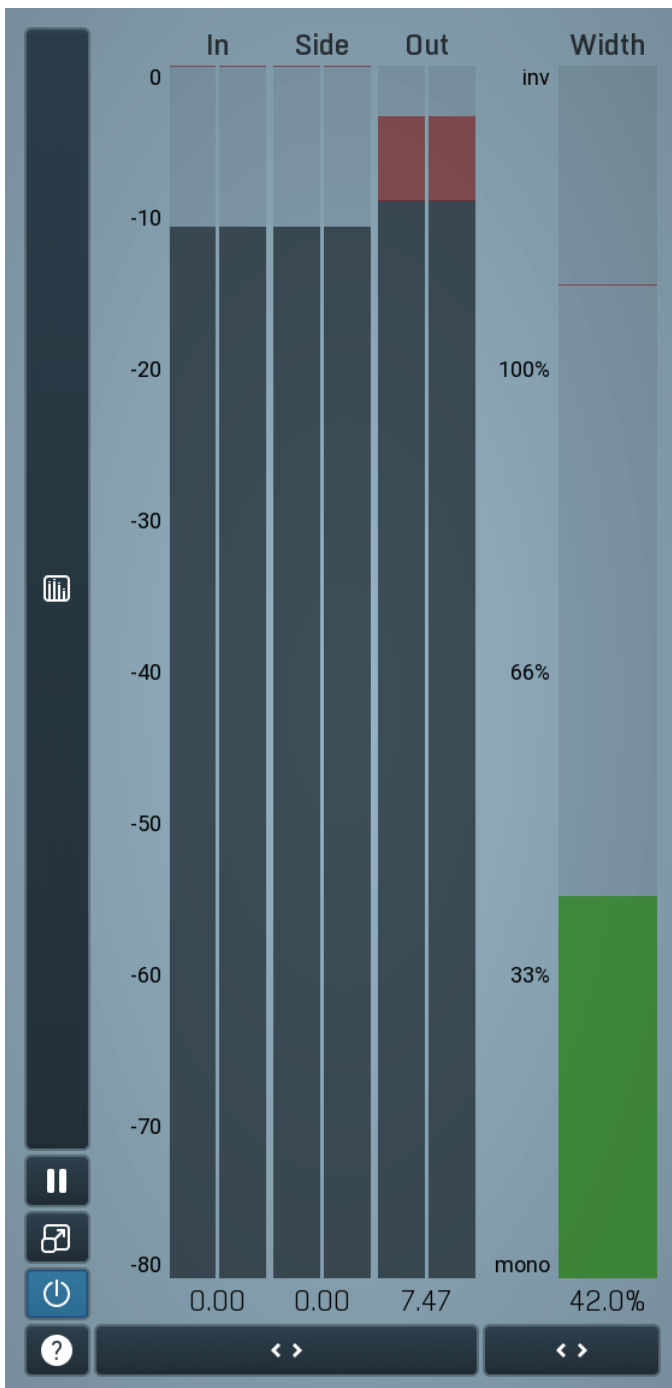


EnvelopeEditorGraph

## Envelope graph

Envelope graph provides an extremely advanced way to edit any kind of shape that you can imagine. An envelope has a potentially unlimited number of points, connected by several types of curves with adjustable curvature (drag the dot in the middle of each arc) and the surroundings of each point can also be automatically smoothed using the smoothness (horizontal pull rod) control. You can also literally draw the shape in drawing mode (available via the main context menu).

- **Left mouse button** can be used to select points. If there is a *point*, you can move it (or the entire selection) by dragging it. If there is a *curvature circle*, you can set up its tension by dragging it. If there is a *line*, you can drag both edge points of it. If there is a *smoothing controller*, you can drag its size. Hold **Shift** to drag more precisely. Hold **Ctrl** to create a new point and to remove any points above or below.
- **Left mouse button double click** can be used to create a new point. If there is a *point*, it will be removed instead. If there is a *curvature circle*, zero tension will be set. If there is a *smoothing controller*, zero size will be set.
- **Right mouse button** shows a context menu relevant to the object under the cursor or to the entire selection. Hold **Ctrl** to create or remove any points above or below.
- **Middle mouse button** drag creates a new point and removes any points above or below. It is the same as holding Ctrl and dragging using left mouse button.
- **Mouse wheel** over a point modifies its smoothing controller. If no point is selected, then all points are modified.
- **Ctrl+A** selects all points. **Delete** deletes all selected points.



### Global meter view

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

In the classical level indicators mode each of the meters also shows the recent maximum value. Click on any one of these values boxes to reset them all.

**In meter** indicates the total input level. The input meter shows the audio level before any specific processing (except potential oversampling and other pre-processing). It is always recommended to keep the input level under 0dB. You may need to adjust the previous processing plugins, track levels or gain stages to ensure that it is achieved.

As the levels approach 0dB, that part of the meters is displayed with **red** bars. And recent peak levels are indicated by single bars.

**Out meter** indicates the total output level. The output meter is the last item in the processing chain (except potential downsampling and other post-processing). It is always recommended to keep the output under 0dB.

As the levels approach 0dB, that part of the meters is displayed with **red** bars. And recent peak levels are indicated by single bars.

**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.

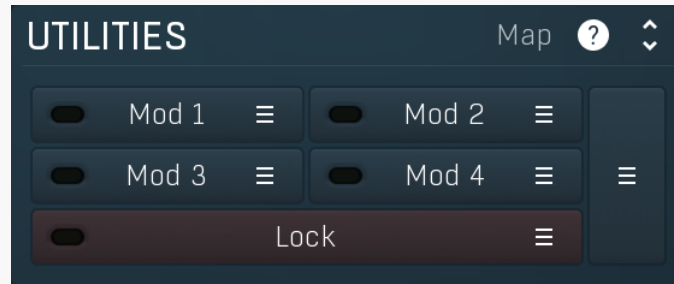


### Collapse

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



# Utilities



## Map **Map**

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

## Modulator **Modulator**

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

## Menu **Menu**

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

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

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

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

## Menu **Menu**

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

## Lock **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**

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

