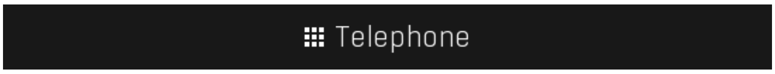
MFreeformEqualizer



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

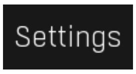
Right arrow button loads the next preset.

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).

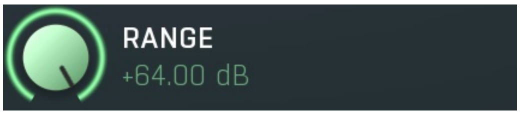
Advanced settings configures several processing options for the plug-in.

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.

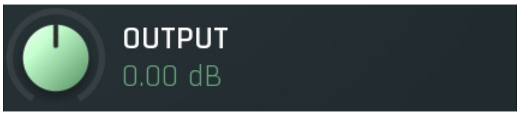
WWW

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.



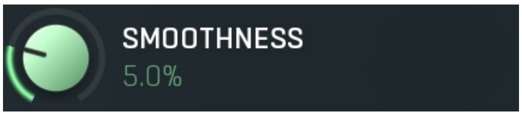
Range

Range defines the range of the equalizer in dB.
Range: +1.00 dB to +64.00 dB, default +24.00 dB



Output gain

Output gain defines the power modification applied to the output signal.
Range: -24.00 dB to +24.00 dB, default 0.00 dB



Smoothness

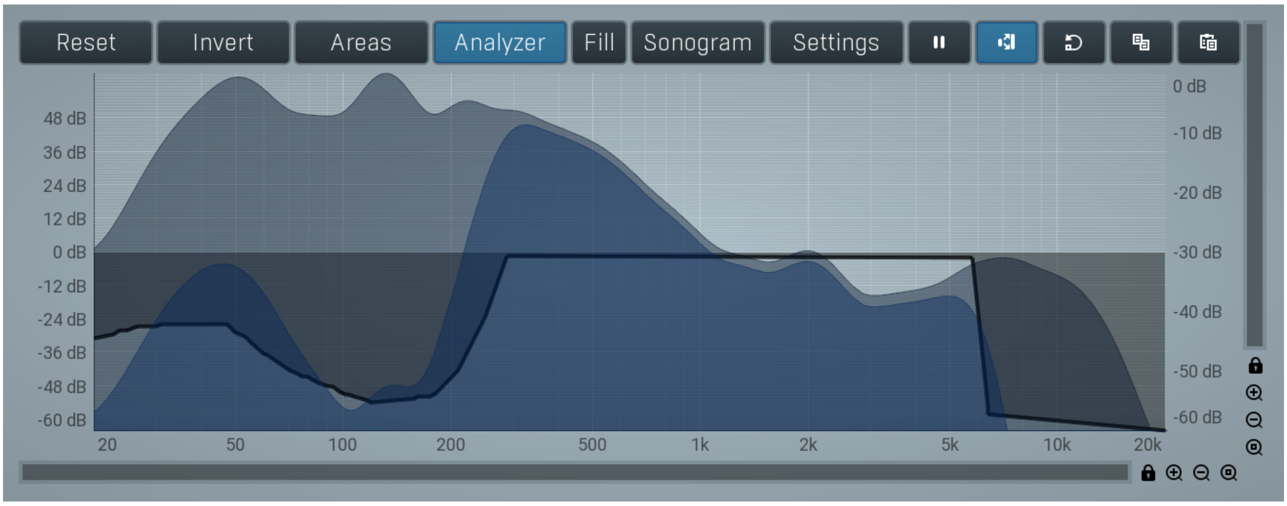
Smoothness makes the analyzer smooth out the curve, so it contains less bumping up and down. It approximates the energy in each frequency and the resulting graph should be easier to understand. Also the smoothness affects the automatic equalization. Usually higher value provides more natural results,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

Quality

Quality defines size of the FFT kernel, which hugely affects CPU consumption and audio processing accuracy.

Minimum phase

Minimum phase switch activates the minimum phase mode. By default the plugin is in linear-phase mode, which means that the output is slightly delayed (exhibits latency), but the phase of each frequency is not changed. Minimum phase mode is known from classic analog equalizers and has no latency. However the phase of each affected frequency is changed, hence it will sound different. The frequency response might also be slightly different, especially in lower quality modes.
The most typical difference between linear-phase and minimum-phase equalizers is in ringing, which is an inevitable part of their sound. You might imagine that each frequency being manipulated in some way has some sort of tail, which is especially audible when processing transient sounds. After each drum hit, for example, you can hear some sort of decay, a very short one, depending on what the equalizer does. While minimum phase equalizers produce this tail after the transient, linear-phase equalizers generate a tail BEFORE as well, called pre-ringing. That's a very unnatural thing as it essentially sounds like something that 'has not happened yet', and human ears are especially sensitive to that. This is what makes linear-phase equalizers sound inferior in many cases.
It is important to understand that the concept of linear-phase filtering is purely scientific and has no meaning in nature. However there is one advantage of it as well - changing phase might introduce comb filtering when the processed audio is mixed with itself (or with another recording of the same instrument). Since linear-phase equalizers do not cause this problem, they are often used for orchestral recordings for example, where there are many microphones recording the same audio material.



Equalizer shape graph

Equalizer shape graph defines the desired frequency response. By default the editor is in drawing mode which means you can just use your left mouse button to draw any frequency response you want. Use right mouse button to disable the drawing mode and to tweak the graph more accurately.

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.

Reset

Reset button restores the original settings (0dB everywhere).

Invert

Invert button inverts the current graph vertically, such that what was being amplified will be attenuated and vice versa.

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.

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.

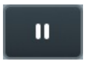
Fill

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.



Pause
Pause button stops the analyzer temporarily.

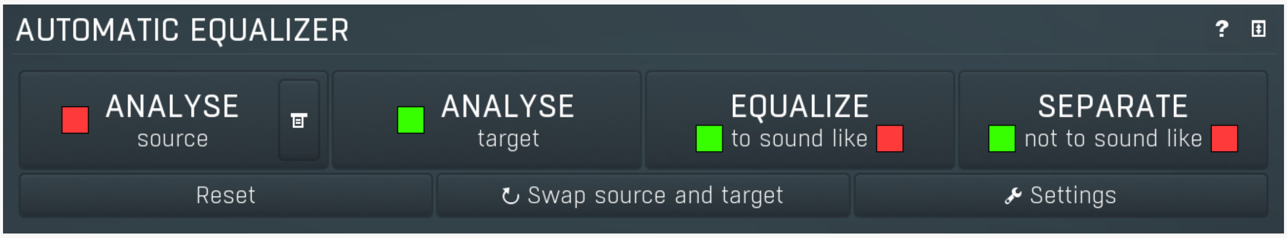
Normalize
Normalize button enables or disables the visual normalization, which makes the loudest frequency be displayed at the top of the analyser area (0dB); it does not normalise the sound. This is very useful for comparing frequency levels, however it does hide the actual level. When comparing 2 spectrums 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.

Automatic equalizer panel



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

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

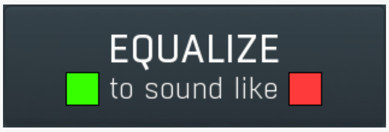
Alternatively, you can analyze an audio file (WAV, MP3, FLAC etc.) offline by clicking the **File** button and browsing to the file or by dragging & dropping the file from your host or Explorer / Finder onto the **Analyze source** button (this latter method may not work in all hosts, especially on Mac).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

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

Alternatively, you can analyze an audio file (WAV, MP3, FLAC etc.) offline by dragging & dropping the file from your host or Explorer /

Finder onto the **Analyze target** button (this may not work in all hosts, especially on Mac).

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



Equalize

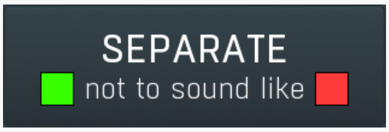
Equalize button performs automatic equalization - it adjusts the bands to match the source and target analyses as closely as possible. To do that, you need to have the analyses of both the source and target audio first. When you have both analyses ready, you will see the **red** (source) and **green**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

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

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

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

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



Separate

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

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

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

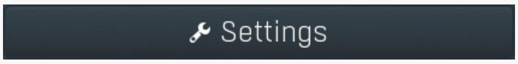
When you have both analyses ready, click this button to perform the separation. In most cases the result will be too strong so it is worth considering lowering the **Dry/wet** parameter to say 30-40% to get more natural output. You can also make use of the **Smoothness** parameter, above the graph area.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

Reset

Reset button clears both the source and target analyses.

Swap source and target

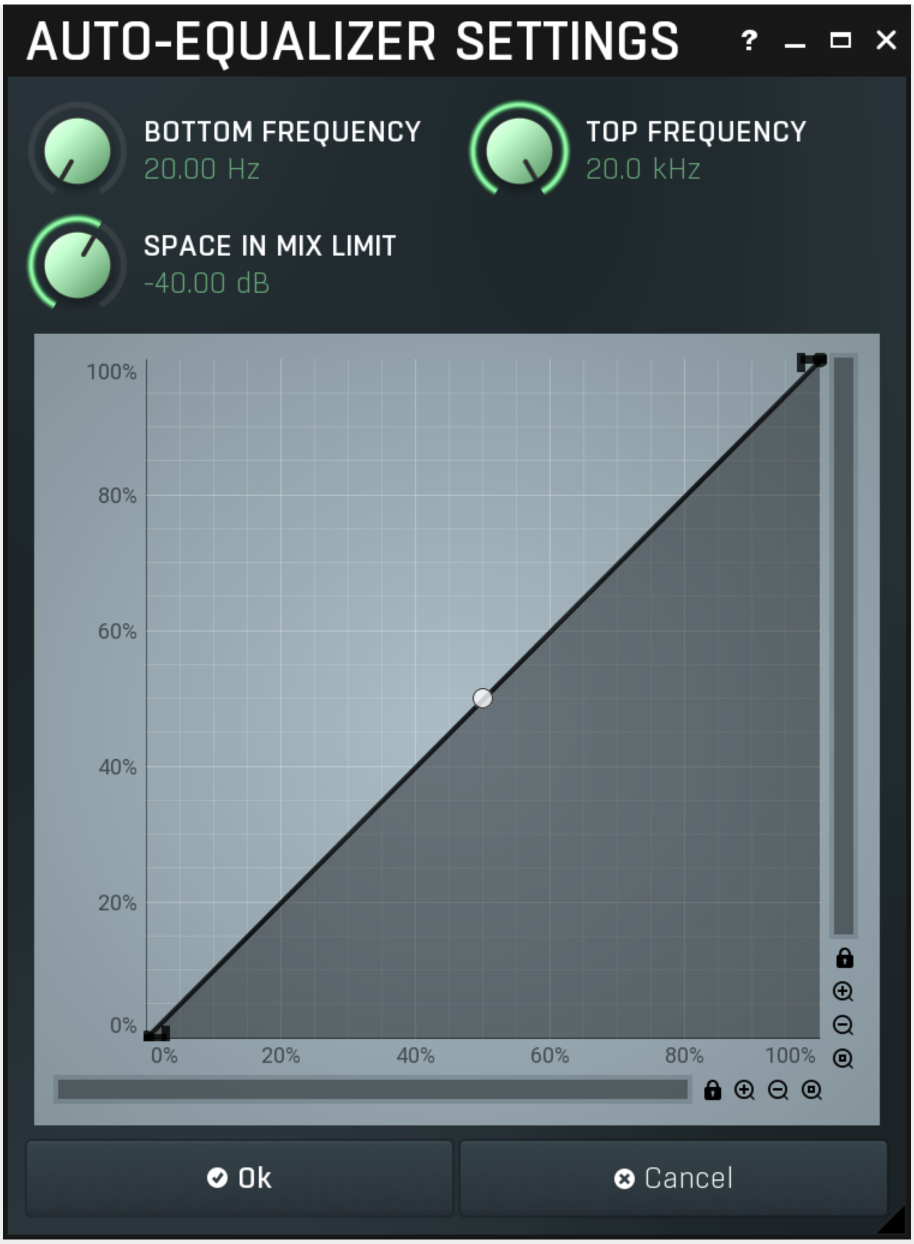
Swap source and target button swaps the analyses of source and target and can be helpful when you want to try equalizing 'the other way around'.



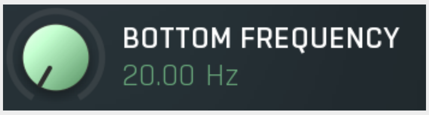
Settings

Settings button shows additional settings of the automatic equalizer algorithm.

Auto-equalizer settings

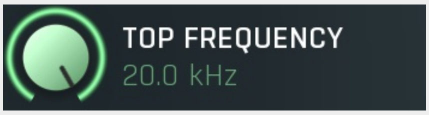


Auto-equalizer settings provides additional settings for the automatic equalization algorithm.



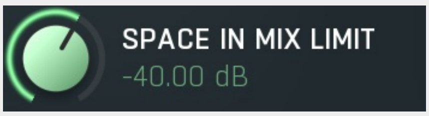
Bottom frequency

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



Top frequency

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

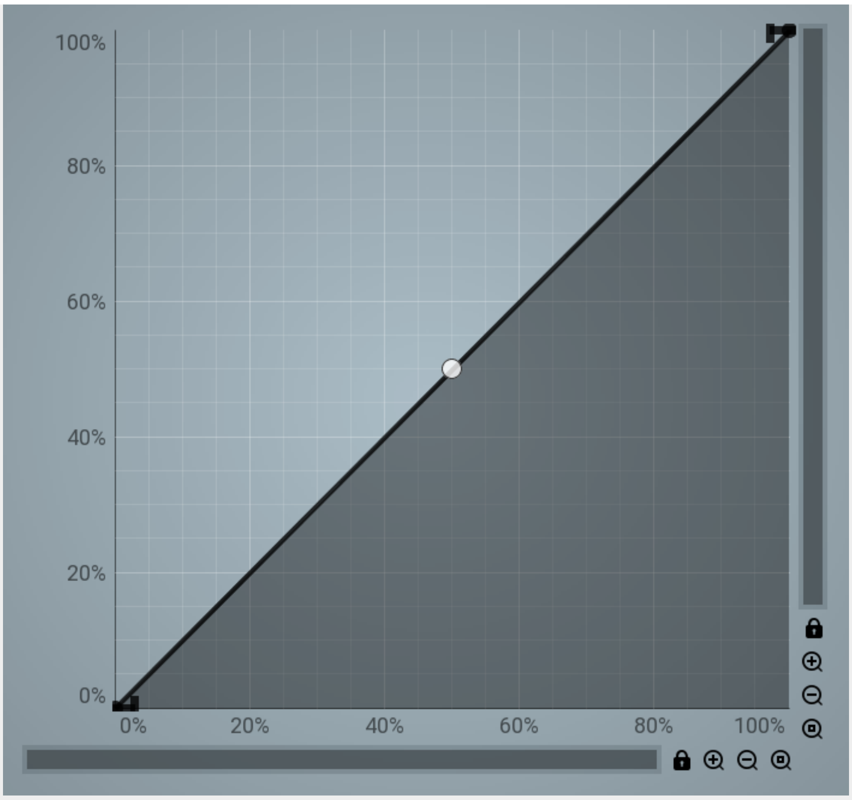


Space in mix limit

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.

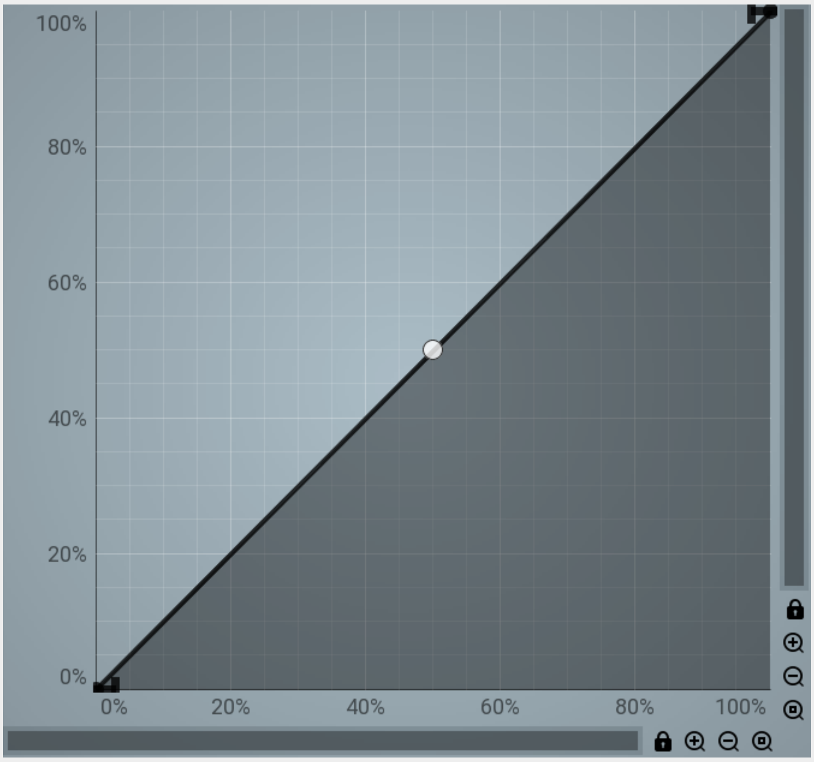
For example, if this value is -20dB and the level of the frequency 1000Hz in the source analysis is -20dB, then the separation engine will consider 1000Hz to be "available" in the target audio and will not try to remove this frequency from the target. However if the level in the source were -10dB, then 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.



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.



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).

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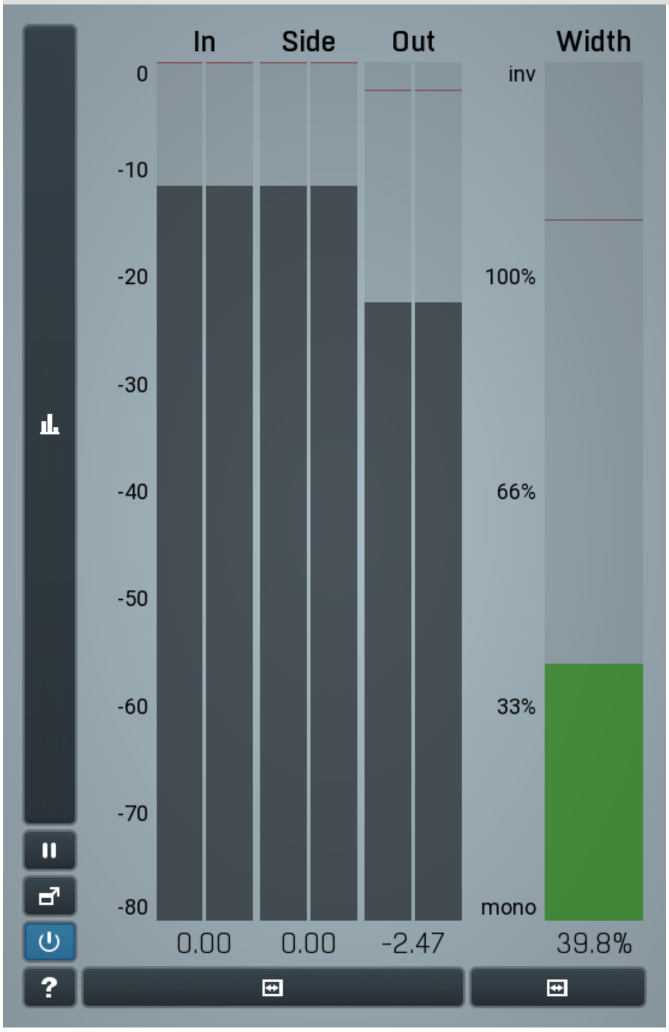

Global meter view

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.

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