Overview

An equalizer is doubtless the most important audio processing tool. Unfortunately it is also one of the hardest to master too, with years of experience necessary in order to use it properly. MAutoEqualizer not only sounds great, it also greatly simplifies this task.

MAutoEqualizer is a revolutionary mastering plugin that can also be used for mixing and creative effects. It combines a powerful equalizer (including a state-of-the-art linear-phase version) and an analyser to achieve the first truly automatic equalization functionality. It is the first plugin to feature our MeldaProduction Filter Adaptation (MFA) technology, which can actually perform the equalization for you based on an analysis of your recording, another recording or indeed any spectral content that you can literally "draw" using our MeldaProduction Envelope System (MES).

Do not be mistaken, this has nothing to do with FFT. MAutoEqualizer is a parametric equalizer. The filter adaptation really configures the bands only. It does not suffer from distortion, transient smearing or other artifacts caused by simple FFT algorithms.

MAutoEqualizer provides an automatic equalization feature that lets you focus on what spectrum you want to get, not how to get it. With a standard equalizer you are listening to the whole spectrum while amplifying or attenuating frequencies. This is very difficult even for a very experienced user with advanced listening skills. With MAutoEqualizer you can make your recording sound like a commercial song. You no longer need to be worried if your ears aren’t objective enough.

You can give your recordings a professional sound. And you can ensure that all of the recordings on your album sound uniform. Or you can draw your desired frequency response. MAutoEqualizer will work out how to do what you want and configure the parametric equalizer bands for you.

Introduction

The MAutoEqualizer package contains two plugins - the linear phase version (MAutoEqualizerLinearPhase) and the minimum phase version (MAutoEqualizer). The linear phase version implements three equalization algorithms - minimum phase, linear phase and FFT based linear phase. The disadvantage of the linear phase version is that it induces latency. Please note that the response of the linear phase version in minimum phase mode is different from the response of the native minimum phase version.

First look at the spectral analyser view. By default, the view is set to display the input and output levels: the moving green line displays the power of the output signal after processing and the moving dark green line shows the power of the incoming signal before processing. The static black or white line defines the equalizer frequency response. You can use the band points or the controls in the Bands panel below to control the equalizer manually.

Automatic equalization

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

1. Analyse your recording - start playback and press the Analyse target button. Most of the graphs in the spectral view will disappear and a blue 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 the Analyse source button (by the same method used to analyse the target), draw the requested frequency response using the Draw button or even analyze an audio file using the File button.

3. Click the Equalize or Separate button and the plugin will adjust the bands.

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.

If you are having problems mixing in a particular track, you can also analyse your whole mix (without the problematic track) and let the plugin help fit the track into the mix, and
make the mix clearer, less muddy. See Separate button for more information.

### Presets button

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 button

Left arrow button loads the previous preset.

#### Right arrow button

Right arrow button loads the next preset.

#### Randomize button

Randomize button loads a random preset.

#### Save button

Save button replaces the current preset.

#### Panic button

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 button

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 button**

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.

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### General parameters panel

General parameters panel contains general processing parameters, such as the gain and dry/wet controls.

**Dry/wet**

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

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

Gain defines output gain applied after the equalization.

Soft saturation defines amount of saturation simulating analog equalizers.

Gain

Gain

0.00 dB

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

Soft saturation

0.00%

Range: 0.00% to 100.0%, default 0.00%

Mode

Mode controls the equalizer algorithm. The processor provides 3 different algorithms:

Normal is the standard minimum-phase algorithm as used in other MeldaProduction equalizers. It is provided so you can compare the linear-phase and minimum-phase algorithms on your particular material.

Linear-phase is the high-quality linear-phase algorithm, implemented using an improved bidirectional method, which usually provides the best audio quality.

Linear-phase FFT based modes uses the trivial FFT algorithm used in common linear-phase equalizers. Its main disadvantage is reduced accuracy for the low end of the audio spectrum.

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Analyser panel

Analyser panel contains settings of the analyser engine. None of the parameters affect the resulting sound, however Smoothness does affect automatic equalization.

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%

Average length 300 ms

Averaging

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

Range: 0 ms to 5000 ms, default 300 ms

Spectrum viewer

Spectrum viewer shows spectrum of the input and output signals and various analyses. The white or black line defines the equalizer frequency response. You can use it to edit each eq band or you can use band editors below. Double-click on a band point to enable or disable it. Click using the right mouse button on a band point to change its properties. Equalizer follows the units on the left ranging from -24dB to 24dB.

Green area contains spectrum of the output signal after equalization. Dark green line describes spectrum of incoming signal before equalization. Blue line displays target analysis, thus kind of long-term frequency spectrum. Orange line shows source analysis - usually a requested spectrum. Automatic equalization tries to make the analysis (blue line) similar to source analysis (orange line).

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 button**

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

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

**Analyzer button**

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 button**

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 button**

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 button**

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

**Pause button**

Pause button stops the analyzer temporarily.

**Normalize button**

Normalize button enables or disables the visual normalization, which makes the loudest frequency be displayed at the top of the analyzer area (0 dB); 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 button**

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

**Copy button**

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 button**

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.

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

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

**Frequency**
Frequency defines the band's central frequency, which has different meaning depending on 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 use 4-pole 24 dB/octave filters instead etc. To see the actual slope of each filter look into the filter type list on the left.

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

When one of more bands are set to process a single channel, then 2 EQ curves are displayed, in blue 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.

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** Harmonics panel **

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

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

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

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

Bands panel contains all equalizer bands. You can manipulate them here or use the graph.

Reset button restores the original equalizer settings.

Invert button inverts the gains of all bands.

Copy button copies the equalizer settings onto the system clipboard.

Paste button pastes the equalizer settings from the system clipboard.

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

Settings button shows additional settings of the automatic equalizer algorithm.

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

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

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

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

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

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

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

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

- **Default** provides the most accurate results, but it takes lots of CPU. Note, as mentioned above, that this CPU usage is required only for the pre-computing, the actual audio processing requirements are the same for all algorithms.
- **Optimized** algorithm is much faster, however it can produce slightly less accurate results. It may be more suitable because it smoothes out the peaks in the frequency response.
**Super-fast** mode is the fastest of all of them, but it trades speed for accuracy in its results. You may find it useful in specific cases.

**Never amplify**

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

**Disable pass/notch**

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

### 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.

### 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 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 curve, 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.
Swap source and target button

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'.

Analyze source button

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.

File button

File button lets you choose a file in your file system and analyse it completely resulting in the source analysis.

Draw button

Draw button enables drawing mode, which you can use to draw the desired frequency response instead of analyzing spectrum of another recording.

Load button

Load button leads a source analysis graph which you can then use for automatic equalization.

Save button

Save button stores the current source analysis graph (orange line). It is generally useful to save source analyses created using Draw button.

Analyze target button

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 blue 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 button

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 blue (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 blue 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 button

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 blue (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 blue 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.
Global meter view provides a powerful metering system. If you do not see it in the plug-in, click the Meters or Meters & Subsystems 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 button**

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 button**

Pause button pauses the processing.

**Popup button**

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 button**

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

**Collapse button**

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

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**Multiparameters**

**Map button**

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

**Multiparameter button**

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 button**

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.

**Lock button**

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 button**

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