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.

**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.

### 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.

Range: 0.00% to 100.0%, default 100.0%

**Gain**

Gain defines output gain applied after the equalization.

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%

**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** mode 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.
**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**

Averaging makes the analyzer 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

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**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 button**

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.

**Analyzer settings**

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

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

**View**

Freeze button stops processing temporarily.

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 button

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

View type

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

Opacity

Opacity controls the opacity of all analyzer graphs.

Resolution

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

Analysis

Source

Source mode defines which audio stages should be analyzed. By default both input & output are selected and analyzed. However you may want to analyze only the input, or the output (or the external sidechain, where available).

Channel mode

Channel mode defines which channels should be analyzed. By default all channels are merged into a mono sum, which is then analyzed. However you may want to analyze separate channels or display both the left and right channels separately. Please note that if both input & output are displayed at the same time then mix mode is used instead of left & right mode.

Decay

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

Super-resolution mode

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

Enable when hidden

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

Global normalization

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

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

### Basic settings

#### BASIC SETTINGS

- **Slope**
  - Value: +3.00 dB
  - **Slope** makes the analyser increase the magnitude of higher frequencies, since they are typically lower in energy. 3dB per octave is a typical value, which makes pink noise horizontal as pink noise contains equal energy in each octave. Therefore if you set slope to 3dB, the response would be the same for the FFT and 1/3 octave graphs.

- **Gain**
  - Value: 0.00 dB
  - **Gain** makes all frequencies change magnitude by the specified amount. This has no meaning when normalization is enabled.

- **Time resolution**
  - Value: 0.00%
  - **Time resolution** improves the time resolution, but lowers the spectral resolution. This is typically useful for more scientific analyses, where the signal is moving quickly and you need to follow its movements quickly. This is often advantageous for sonograms with very high FFT sizes.

- **Deharmonize**
  - Value: 0.00%
  - **Deharmonize** tries to remove harmonics in the content and leave only fundamentals. This may help you find the dominant frequencies in the signal.

### Peak detection

**Peak detection**

- **Peak detection**
  - Value: 0.00%

- **Peak threshold**
  - Value: -40.00 dB

### Scientific settings

**Overlap**

- Value: 4x

**Window type**

- Value: Hann

**FFT size**

- Value: 4096

**Analytical smoothing**

- Enabled

**Logarithmic averaging**

- Enabled
Peak detection

Peak detection tries to remove skirts of separate sinusoids letting you view the frequencies contained in your audio material. This may be handy when performing more scientific analyses.

Peak threshold

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

Scientific settings

Overlap

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

FFT size

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

Window type

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

Analytical smoothing

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

Logarithmic averaging

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

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

### Average

**Average**

- **Copy analysis** button
  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.

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

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

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

- **Thick**
  Thick makes the analysis graph use a thick line.

- **Fill area**
  Fill area makes the analysis graph area below the line filled. Please note that this takes additional CPU power.

- **Color**
  Color changes the color of the graph to the color of the editable rectangle next to it.

### Average (infinite)

### Maximum

### Maximum (infinite)

### Maximum - Average (infinite)

### Comparison

- **Fill area**
  Fill area makes the analysis graph area below the line filled.
Copy analysis button
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.

Maximum

Maximum (infinite)

Maximum - Average (infinite)

Comparison

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

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

### Presets button
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 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.

### Opacity
Opacity controls the opacity of the sonogram.

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

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

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**Presets button**

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 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.

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

Random button
Random button generates random settings using the existing presets.

Presets button
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 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.

Copy button
Copy button copies the settings onto the system clipboard.

Paste button
Paste button loads the settings from the system clipboard.

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

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

Insert point
Insert point button creates a point at mouse position.

Step sequencer button
Step sequencer button generates the envelope from step sequencer.
Randomize button
Randomize button slightly modifies the Y coordinates.

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

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

Clear points button
Clear points button deletes all points.

Curvature

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

Smoothing

Lock sides
Lock sides makes the smoothing factor equal on both sides.

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

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

Pause 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 (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 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 a 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 the 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.

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

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

Maximal count

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

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

Bands panel

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

Reset button
Reset button restores the original equalizer settings.

Invert button
Invert button inverts the gains of all bands.

Copy button
Copy button copies the equalizer settings onto the system clipboard.

Paste button
Paste button pastes the equalizer settings from the system clipboard.

Automatic equalizer panel

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

Settings button
Settings button shows additional settings of the automatic equalizer algorithm.
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
Graph editor lets you edit the envelope graph.

**Envelope graph**

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

- **Left mouse button** can be used to select points. If there is a point, you can move it (or the entire selection) by dragging it. If there is a curvature circle, you can set up its tension by dragging it. If there is a line, you can drag both edge points of it. If there is a smoothing controller, you can drag its size. Hold Shift to drag more precisely. Hold Ctrl to create a new point and to remove any points above or below.

- **Left mouse button double click** can be used to create a new point. If there is a point, it will be removed instead. If there is a curvature circle, zero tension will be set. If there is a smoothing controller, zero size will be set.

- **Right mouse button** shows a context menu relevant to the object under the cursor or to the entire selection. Hold Ctrl to create or remove any points above or below.

- **Middle mouse button** drag creates a new point and removes any points above or below. It is the same as holding Ctrl and dragging using left mouse button.

- **Mouse wheel** over a point modifies its smoothing controller. If no point is selected, then all points are modified.

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

**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 loads 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.
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.

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**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.
**Multiparameters**

![Multiparameters](image)

**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.

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

**Upsampling**

Upsampling can potentially improve sound quality by processing at a higher sample rate. Processors such as compressors, saturators, distortions etc., which employ nonlinear processing generate higher harmonics of the existing frequencies. If these frequencies exceed the Nyquist rate, which equals half of the sampling rate, they get mirrored back under the Nyquist rate. This is known as aliasing and is almost always considered an artifact. This is because the mirrored frequencies are no longer harmonic and sound as digital noise as this effect does not physically occur in nature. Upsampling (or oversampling) reduces the problem by temporarily increasing the sampling rate. This moves the Nyquist frequency which in turn, diminishes the level of the aliased harmonics. Note that the point of upsampling is not to remove harmonics, we usually add them intentionally to make the signal richer, but to reduce or attenuate the harmonics with
To understand aliasing, try this experiment: Set the sampling rate in your host to 44100 Hz. Open MOscillator and select a “rectangle” or “full saw” waveform. These simple waveforms have lots of harmonics and without upsampling even they become highly aliased. Now select 16x upsampling and listen to the difference. If you again select 1x upsampling, you can hear that the audio signal gets extensively “dirty”. If you use an analyzer (MANalyzer or MEqualizer for example), you will clearly see how, without upsampling, the plugin generates lots of inharmonic frequencies, some of them which are even below the fundamental frequency. Here is another, very extreme example to demonstrate the result of aliasing. Choose a “sine” shape and activate 16x upsampling. Now use a distortion or some saturation to process the signal. It is very probable that you will be able to hear (or at least see in the analyzer) the aliased frequencies.

The plugin implements a high-quality upsampling algorithm, which essentially works like this: First the audio material is upsamled to a higher sampling rate using a very complicated filter. It is then processed by the plugin. Further filtering is performed in order to remove any frequencies above the Nyquist rate to prevent aliasing from occurring, and then the audio gets downsamled to the original sampling rate.

Upsampling also has several disadvantages of which you should be aware before you start using it. Firstly, upsampled processing induces latency (at least in high-quality mode, although you can select low-quality mode in the plugin settings), which is not very usable in real time applications. Secondly, upsampling also takes much more CPU power, due to both the processing being performed at a higher sampling rate (for 16x upsampling at 44100 Hz, this equates to 706 kHtz!), and the complex filtering. Finally, and most importantly, upsampling creates some artifacts of its own and for some algorithms processing at higher sampling rates can actually lower the audio quality, or at least change the sound character. Your ears should always be the final judge.

As always, use this feature ONLY if you can actually hear the difference. It is a common misconception that upsampling is a miraculous cure all that makes your audio sound better. That is absolutely not the case. Ideally, you should work in a higher sampling rate (96kHz is almost always enough), while limiting the use of upsampling to some heavily distorting processors.

Channel mode button

Channel mode button shows the current processing channel mode, e.g. Left+Right (L+R) indicates the processing of left and right channels. This is the default mode for mono and stereo audio material and effectively processes the incoming signal as expected. However the plugin also provides additional modes, of which you may take advantage as described below. Mastering this feature will give you unbelievable options for controlling the stereo field.

Note that this is not relevant for mono audio tracks, because the host supplies only one input and output channel.

Left (L) mode and Right (R) mode allow the plugin to process just one channel, only the left or only the right. This feature has a number of simple uses. Equalizing only one channel allows you to fix spectral inconsistencies, when mids are lower in one channel for example. A kind of stereo expander can be produced by equalizing each side differently. Stereo expansion could also be produced by using a modulation effect, such as a vibrato or flanger, on one of these channels. Note however that the results would not be fully mono compatible.

Left and right channels can be processed separately with different settings, by creating two instances of the plugin in series, one set to ‘L’ mode and the other to ‘R’ mode. The instance in ‘L’ mode will not touch the right channel and vice versa. This approach is perfectly safe and is even advantageous, as both sides can be configured completely independently with both settings visible next to each other.

Mid (M) mode allows the plugin to process the so-called mid (or mono) signal. Any stereo signal can be transformed from left and right, to mid and side, and back again, with minimal CPU usage and no loss of audio quality. The mid channel contains the mono sum (or centre), which is the signal present in both left and right channels (in phase). The side channel contains the difference between the left and right channels, which is the ‘stereo’ part. In ‘M mode’ the plugin performs the conversion into mid and side channels, processes mid, leaves side intact and converts the results back into the left and right channels expected by the host.

To understand what a mid signal is, consider using a simple gain feature, available in many plugins. Setting the plugin to M mode and decreasing gain, will actually lower or attenuate the mono content and the signal will appear "wider". There must be some stereo content present, this will not work for monophonic audio material placed in stereo tracks of course. Similarly amplifying the mono content by increasing the gain, will make the mono content dominant and the stereo image will become "narrower".

As well as a simple gain control there are various creative uses for this channel mode.

Using a compressor on the mid channel can widen the stereo image, because in louder parts the mid part gets attenuated and the stereo becomes more prominent. This is a good trick to make the listener focus on an instrument whenever it is louder, because a wider stereo image makes the listener feel that the origin of the sound is closer to, or even around them.

A reverb on the mid part makes the room appear thin and distant. It is a good way to make the track wide due to the existing stereo content, yet spacey and centered at the same time. Note that since this effect does not occur naturally, the result may sound artificial on its own, however it may help you fit a dominant track into a mix.

An equalizer gives many possibilities - for example, the removal of frequencies that are colliding with those on another track. By processing only the mid channel you can keep the problematic frequencies in the stereo channel. This way it is possible to actually fit both tracks into the same part of the spectrum - one occupying the mid (centre) part of the signal, physically appearing further away from the listener, the other occupying the side part of the signal, appearing closer to the listener.

Using various modulation effects can vary the mid signal, to make the stereo signal less correlated. This creates a wider stereo image and makes the audio appear closer to the listener.

Side (S) mode is complementary to M mode, and allows processing of only the side (stereo) part of the signal leaving the mid intact.

The same techniques as described for M mode can also be applied here, giving the opposite results.

Using a gain control with positive gain will increase the width of the stereo image.

A compressor can attenuate the side part in louder sections making it more monophonic and centered, placing the origin a little further away and in front of the listener.
A reverb may extend the stereo width and provide some natural space without affecting the mid content. This creates an interesting side-effect - the reverb gets completely cancelled out when played on a monophonic device (on a mono radio for example). With stereo processing you have much more space to place different sounds in the mix. However when the audio is played on a monophonic system it becomes too crowded, because what was originally in two channels is now in just one and mono has a very limited capability for 2D placement. Therefore getting rid of the reverb in mono may be advantageous, because it frees some space for other instruments.

An equalizer can amplify some frequencies in the stereo content making them more apparent and since they psycho acoustically become closer to the listener, the listener will be focused on them. Conversely, frequencies can be removed to free space for other instruments in stereo.

A saturator / exciter may make the stereo richer and more appealing by creating higher harmonics without affecting the mid channel, which could otherwise become crowded. Modulation effects can achieve the same results as in mid mode, but this will vary a lot depending on the effect and the audio material. It can be used in a wide variety of creative ways.

Mid+Side (M+S) lets the plugin process both mid and side channels together using the same settings. In many cases there is no difference to L+R mode, but there are exceptions.

A reverb applied in M+S mode will result in minimal changes to the width of the stereo field (unless it is true-stereo, in which case mid will affect side and vice versa), it can be used therefore, to add depth without altering the width.

A compressor in M+S mode can be a little harder to understand. It basically stabilizes the levels of the mid and side channels. When channel linking is disabled in the compressor, you can expect some variations in the sound field, because the compressor will attenuate the louder channel (usually the mid), changing the stereo width depending on the audio level. When channel linking is enabled, a compressor will usually react similarly to the L+R channel mode.

Exciters or saturators are both nonlinear processors, their outputs depend on the level of the input, so the dominant channel (usually mid) will be saturated more. This will usually make the stereo image slightly thinner and can be used as a creative effect.

How to modify mid and side with different settings? The answer is the same as for the L and R channels. Use two instances of the plugin one after another, one in M mode, the other in S mode. The instance in M mode will not change the side channel and vice versa.

Left+Right(neg) (L+R-) mode is the same as L+R mode, but the the right channel's phase will be inverted. This may come in handy if the L and R channels seem out of phase. When used on a normal track, it will force the channels out of phase. This may sound like an extreme stereo expansion, but it is usually extremely fatiguing on the ears. It is also not mono compatible - on a mono device the track will probably become almost silent. Therefore be advised to use this only if the channels are actually out of phase or if you have some creative intent.

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

Surround mode is not related to stereo processing but lets the plugin process as many channels as the host supplies (up to 8). To use it, you have to first activate surround processing, by selecting the menu item. This is a global switch for all MeldaProduction plugins, which configures them to report 8in-8out capabilities to the host, on loading. It is disabled by default, because some hosts have trouble dealing with such plugins. After activation, restart your host to start using the surround capabilities of the plugins. Deactivation is done in the same way. Please note that the sidechain inputs will be multi-channel too.

First place them on a surround track - a track that has more than 2 channels. Then select Surround from the plug-in's Channel Mode menu. The plugins will regard this mode as a natural extension of 2 channel processing. For example, a compressor will process each channel separately or measure the level by combining the levels of all of the inputs provided. Further surround processing properties, to enable/disable each channel or adjust its level, can be accessed via the Surround settings in the menu.

AGC button

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

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

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

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

As a general rule of thumb as for all audio processing tasks, use it only if you know you need it. AGC is a powerful tool that can make your workflow easier, but it can also be damaging.
Set button uses the AGC (automatic gain compensation) processor to calculate the ideal output gain to ensure that the output audio loudness is equal to the input level. To use it, simply enable playback in your host and click the button. The plugin’s output gain will be adjusted to match the input and output levels as closely as possible.

If the AGC is already enabled, the change will be instant and you can disable the AGC afterwards. Typically you will browse presets, generate random settings etc. During the entire time you will have AGC enabled to prevent you from experiencing different output loudness levels. When you find a sonically ideal setup, you simply click the Set button to set the output gain automatically and disable the AGC as you won’t need it anymore.

If the AGC is not already enabled, clicking the Set button displays a window with progress bar for a few seconds, while the plugin temporarily enables AGC and analyses input and output of the plugin. After that the AGC is disabled again.

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

**Limiter button**

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

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

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

**Diff button**

Diff button lets you audition the difference between input and output. This is especially useful for dynamic processors, such as compressors, where you can simply listen to the parts being modified. The output may give you insight about which parts of the signal are being processed and how.

**A-H presets selector**

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

For example, this feature can be used to keep multiple settings, when you are not sure about the ideal configuration. When you change any parameter, only the currently selected preset is modified.
The four buttons below enable you to switch between the last 2 selected sets using the A/B button, morph between the first 4 sets using the morphing button and copy & paste settings from one preset to another (via the clipboard).

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

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

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

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

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

### Copy button
Copy button copies the current settings to the system clipboard. Other presets, upsampling, channel mode and other global settings are not copied.

Hold Ctrl to save the settings as a file instead. That may be necessary for complex settings, which may be too long for system clipboard to handle. It may also be advantageous when you want to send the settings via email. You can load the settings by drag & dropping them to a plugin or holding Ctrl and clicking Paste.

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

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

### Redo button
Redo button reverses the last undo operation.

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

Please note that the files will be overwritten, so make a copy first if you want to keep the original.
Preset management window provides management for your presets.

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

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

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

**Auto-open**

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

**Open all button**

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

**Add button**

Add button creates a new folder in the tree.

**Rename button**

Rename button lets you rename the selected folder.

**Delete button**

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

**Export button**

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

**Import button**

Import button lets you import a file containing presets and sub-folders and add it to the selected folder. The importer will ask you whether to destroy the original contents, so that the new presets replace previous ones, or to keep both.
Presets list contains all presets available in the selected folder. **Double-click** on a preset or use **Load** button to load a preset. Use the buttons at the bottom of the list to perform additional changes. Please note that these are not actual files & folders on disk, but are records in the preset database.

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

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

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

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

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

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

**Add button**
Add button creates a new preset using the current settings.

**Replace button**
Replace button replaces the selected preset by one with current settings.
Delete button deletes the selected preset.
Plugin settings window offers more advanced settings and is available via the Settings button.

Licence panel

Licence panel lets you manage licences on this computer.

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

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

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

**SUBSCRIPTIONS**  
**Subscriptions button**  
Subscriptions button lets you manage the subscription based licencing.
GUI & Style panel lets you configure the plugin's style (and potentially styles of other plugins) and other GUI properties.

**STYLE**

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

**RANDOM STYLE**

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

**DEFAULT STYLE**

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

**SELECT CURRENT STYLE AS DEFAULT**

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

**Editor mode**

Editor mode selects the default control used by the plugin editors. Each control is manipulated in a slightly different way, takes a different amount of space and looks different. To make the editor as small as possible, it is usually best to use Buttons.

**GPU acceleration**

GPU acceleration controls how much the GPU is used for visual rendering to save CPU power.

- **Enabled mode** provides maximum speed and lets the GPU perform as many drawing operations as possible.
- **Compatibility mode** uses the GPU for drawing, but doesn't use modern technologies for maximum performance. Use it if you experience occasional problems with drawing, the usual case for older ATI graphics cards. With Pro Tools on OSX this mode is always used instead of Enabled mode due to compatibility problems with this host.
- **Disabled mode** disables GPU acceleration completely, drawing is then performed by the CPU. Use only if you experience technical difficulties.

A known problem may occur when using multiple displays with multiple graphical interfaces. When moving the plugin window from one display to another, it may stop displaying correctly until you move it back to the original display.
Frames per second controls the refresh rate of the visual engine. The higher the number is the smoother everything is, but the more CPU it requires. You might want to lower this value if your computer is running out of CPU power.

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

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

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

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

Enable gradients
Enable gradients enables allows the plugins to use gradients for various graphs. Disabling this will save some CPU.

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

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

Set default editor size button
Set default editor size button stores the current editor size as its default. You can drag the bottom-right corner of most plugins to change their size. This can be advantageous as it allows several controls to be bigger and easier to work with. After clicking this item, the current size will be stored and any new instance will open with this size by default. Default sizes are usually the smallest available, so that people with lower resolution displays can still use the plugins. This item is especially useful for users who want to enjoy the advantages of large hi-res displays.
Advanced settings panel contains settings that control the behaviour of this instance. These are properties that rarely need to be changed, so they have been moved here.

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

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

**High-quality upsampling**
High-quality upsampling enables the high-quality linear-phase upsampling algorithm. This is relevant only if you use upsampling. Linear-phase upsampling provides the maximum possible quality, however it also requires more CPU and introduces latency. If you need to use upsampling in real-time or want to save resources, you can deactivate this high-quality upsampling option. That will switch to the minimum-phase upsampling algorithm, which offers a superb audio quality as well and does not introduce latency, but it does alter the phase, which may not be acceptable in some cases.

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

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

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

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

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

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

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

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

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

**System info**
System info button displays some technical information about the build and the machine.
Smart interpolation panel controls the depth of the smart interpolation algorithm, which controls the parameters in order to provide maximum audio quality and lower the chance of zipper noise. Smart interpolation is engaged whenever you change any parameter via the GUI, modulators, multiparameters, MIDI or automation.

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

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

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

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

Please note that modulating certain parameters without artifacts is impossible. For example, when modulating a delay very quickly, the physics of such a process just cannot occur in the natural world and the results are appropriately unnatural. These physically impossible processes usually manifest themselves as distortion or zipper noise.
MultiParameter is a powerful structure, which can speed up your workflow significantly and even perform automatic tasks, often useful when performing in real-time for example. Essentially a multiparameter is a controller which controls other parameters, in fact, an unlimited number of them. Each parameter has limits and potentially a transformation curve for more advanced processing. By manually moving the multiparameter (or automating/modulating it) you can control all of the associated parameters at once.

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

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

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

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

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

Now that we have shown you what is possible with multiparameters, it is worth mentioning that they are used extensively for building active presets on the easy screens of most Melda plugins. Every multiparameter given a name in the Information panel will be shown on the Easy screen (if the plugin has one). Check our online video tutorials to get more information about multiparameters and building active presets.

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

Presets button
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 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.

Copy button
Copy button copies the settings onto the system clipboard.

Paste button
Paste button loads the settings from the system clipboard.

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

Behaviour

Normal mode makes the multiparameter work like any other control.

Switch mode hides the slider and shows a button instead. The button has 2 states. By pushing the button, the multiparameter value starts rising from 0% to 100% over a specified time interval. By pushing it again the value starts falling back to 0%. You could do the same thing having the multiparameter in normal mode and moving the slider from left to right and then back, but mode this performs that automatically and maintains a constant time period.
**Trigger** mode is similar to switch mode, but the button has only a single state and when you push it, the value automatically goes from 0% to 100% and then back without any need to push the button again.

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

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

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

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

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

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

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

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

**Set the current value as default** stores the current value as the default one for the multiparameter. The multiparameter’s control responds to right-click by setting the default value in the same way that other parameters do. This way you can select the default value. It is also essential when building your own active-presets.
Appearance

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

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

**Editor mode**
Editor mode controls the way the multiparameter are to be displayed on the Easy screen.
- **Normal** is the default mode and is represented by a small knob or button.
- **Big** mode is similar, but uses a big knob or big button.
- **Button** mode displays a value button, which is usually more compact than knobs.
- **Check-boxes** makes the multiparameter displayed as a set of checkboxes (also called radio buttons). It is relevant only in **Banks** mode.
- **Check-boxes horiz & below** is similar but displays the checkboxes in a single row, hence horizontally. Below mark makes the label underneath the actual checkbox.
- **Switcher and Selectors** are useful for selecting a number of discrete values and similarly to check-boxes these are working only in **Banks** mode.
- **Title button** places the control into the title bar of the panel to which it belongs.
- **Title enable button** places the control into the title bar of the panel to which it belongs and makes it a standard enable button (which also makes all controls within the panel unavailable if it is itself disabled).
- **XY pad** creates a 2 dimensional XY pad control, that edits this multiparameter in the X axis and the next multiparameter in the Y axis. There are multiple versions of this control, all of them differ only by appearance and size.
- **Spacer** is a helper mode for active preset design, which doesn't display anything and only keeps empty space.
- **Meter** creates a simple meter instead. You will probably want to set the multiparameter to Meter mode as well or attach it to a modulator. Meters don't really control anything and their purpose is purely to get a visual feedback. The meters can be horizontal or vertical and they can be up or down. Up is the usual choice useful for peak meters for example. Down is useful for gain reduction meters.
- **Bars start/end** mode creates an editor, similar to step sequencer editor, where each parameter has its own bar. The **Bars start** starts the editor and all multiparameters are then added to it until a multiparameter with **Bars end** mode is found or until there are no remaining multiparameters. Note that this kind of editor doesn't show units and may have several other limitations.

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

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

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

Group color

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

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Grey - utilities, tools

Set button

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

Visible

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

Same row

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

Resizable X

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

Resizable Y

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

Enabled

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

Show name

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

Enable Stepped / Continuous

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

Lockable

Lockable option creates a lock button next to the parameter on the Easy screen, allowing the user to browse through presets without this parameter changing. Please note that this feature is available only for some editor modes.

When the parameter is first locked on the Easy screen it is added to the set of lockable parameters (which are listed in the Global Lock window).
Parameters panel configures how the multiparameter assigns values to the target parameters.

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

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

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

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

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

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

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

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

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

Let’s compare these 4 modes. Taking a value of -12dB value, with a depth of 75% and a scale of +/-24dB. The nominal range is therefore = +/-24 dB * 75% = 36dB. With values of 0%, 50% and 100% the outputs are:

- Up and down: -24, -12, 0 (range constrained to 12 dB either side)
- Full range: -24, -6, 12 (range limited to minimum, but not constrained)
- Up only: -12, 0, 24 (range not constrained = +/-24 dB * 75% = 36dB)
- Down only: -12, -18, -24 (range limited to minimum)

**Interval mode** is the most simple one and goes from Value to Maximal value.
**Value**
Value defines the center of the target parameter's range or the minimum if the **Range mode** is set to **Interval**.

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

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

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

**Use first parameter's range**
Use first parameter's range makes the parameter display use the same range as the first parameter in the list. This is often useful if want to control the range in some way and apply the range to multiple parameters.
Cyclic mode switches the multiparameter into so-called cyclic mode. If you have say 4 banks, called A, B, C and D, and gradually increase the multiparameter value, it starts with A, then interpolates to B, then to C and finally to D. But after that you cannot interpolate back to A, because D is the last one, the maximum value. In cyclic mode the multiparameter behaves as if there were a clone of A at the end, hence after D is reached, the multiparameter interpolates back to A and creates a full circle A->B->C->D->A. This is handy for example if you use a saw wave modulator to drive the multiparameter and want to repeat the sequence of the banks.

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

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

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

You can also use the Random button to generate random values using the smart-randomization engine for each of the banks. And the menu button enables you to re-order the banks.

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

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

Sort banks (down) button
Sort banks (down) button reorders the banks so that the values of the selected parameter are in decreasing order.

Reverse button
Reverse button reverses the order of banks, so that the first bank contains values of the previously last one and so on.

Interpolate button
Interpolate button lets you change the number of banks, but keeps the values as they are now by calculating values of parameter for all banks. It is usually useful when you want to provide ‘banks in between current banks’, without manually calculating the new values.

Auto-gain button
Auto-gain button temporarily enables AGC and automatically sets up the main plugin gain to each bank so that all banks provide similar output loudness. To use it, ensure that the main gain parameter is attached to the multiparameter, start playback of your sound material then click the button and press this button. It will take several seconds to complete depending on the number of the banks.

Set names by values button
Set names by values button sets the names for each bank to the values of the selected parameter. It may be handy when replicating existing parameters for example.

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

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

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

Generally, randomization in plug-ins works by selecting random values for all parameters, but rarely achieves satisfactory results, as the more parameters that change the more likely one will cause an unwanted effect. Our plugins employ a smart randomization engine that learns which settings are suitable for randomization (using the existing presets) and so is much more likely to create successful changes.

In addition, there are some mouse modifiers that assist this process. The smart randomization engine is used by default if no modifier keys are held.

Holding Ctrl while clicking the button constrains the randomization engine so that parameters are only modified slightly rather than completely randomized. This is suitable to create small variations of existing interesting settings.

Holding Alt while clicking the button will force the engine to use full randomization, which sets random values for all reasonable automatable parameters. This can often result in “extreme” settings. Please note that some parameters cannot be randomized this way.

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

Menu button
Menu button provides some additional options related to the bank.

Name button
Name button lets you rename the bank.

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

Presets button
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 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.

Copy button
Copy button copies the settings onto the system clipboard.

Paste button
Paste button loads the settings from the system clipboard.
Parameters panel
Parameters panel configures the list of the parameters which are locked.

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

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

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

**Presets button**
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 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.

**Copy button**
Copy button copies the settings onto the system clipboard.

**Paste button**
Paste button loads the settings from the system clipboard.

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

**Tab selector**
Tab selector switches between subsections.
Controllers panel contains settings of MIDI controllers.

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

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

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

### MIDI

<table>
<thead>
<tr>
<th>Channel</th>
<th>All</th>
<th>Controller</th>
<th>Global 001 Modulation wheel</th>
</tr>
</thead>
</table>

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

- **Channel**
  Channel defines the controller MIDI channel.

- **Controller**
  Controller defines the source controller.
Values

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

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

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

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

Let’s compare these 4 modes. Taking a value of -12dB value, with a depth of 75% and a scale of +/- 24dB. The nominal range is therefore = +/-24 dB * 75% = 36dB. With values of 0%, 50% and 100% the outputs are:

- **Up and down**: -24, -12, 0 (range constrained to 12 dB either side)
- **Full range**: -24, -6, 12 (range limited to minimum, but not constrained)
- **Up only**: -12, 6, 24 (range not constrained = +/-24 dB * 75% = 36dB)
- **Down only**: -12, -18, -24 (range limited to minimum)

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

**Value**

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

**Maximal value**

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

**Depth**

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

**Invert**

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

**Interpolated**

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

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

### MIDI

**Channel**
Channel defines the controller MIDI channel.

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

**Note min**
Note min controls the lowest note to be used by a controller in Linear or Logarithmic mode. The minimum value of the target parameter will then be associated to this note.

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

**Note max**
Note min controls the highest note to be used by a controller in Linear or Logarithmic mode. The maximum value of the target parameter will then be associated to this note.

If both Note min and Note max parameters are default, the plugin takes the actual frequency of each note, and transforms it into the range 20Hz to 20kHz, which is the range used by all equalizers and filters, so that you can literally play a parameter on a MIDI keyboard. If you change either of these 2 parameters however, the plugin takes the range of notes as the requested interval. This is useful for example if you have a small MIDI keyboard used for soloing and you want increase some parameter the higher you play. In the default mode it would be difficult, since the range of frequencies is much bigger than the range of your MIDI keyboard. Set the **Note min** and **Note max** to C0 and B0 respectively, the **Mode** to Logarithmic and select a suitable target parameter (Dry/Wet is fine). Send MIDI notes in the specified range to the plugin and you will see the target parameter increase (by 9.09% (= 100 / (12-1)) for a 100% range).
Mode controls how the controller works. Logarithmic scale is useful for oscillator frequencies, however it may not be useful for general parameters where Linear scale will be more useful.

**On/off** modes react only to single notes and can be used for triggers. When the Note On is received the parameter is changed to its Max value and when the Note Off is received the parameter is changed to its Min value. So this mode can also be used to change between any 2 parameter values.

**Switch** modes are similar, but only recognize when a note is pressed. The Note Offs are ignored. Note Ons select the Max value and Min value alternately. In all octaves mode it doesn't matter which octave is used. For example, this is useful when you want to use any note C to switch something on and off.

**Velocity** modes do not actually follow the note number being pressed, but it's velocity instead. While you can do the same thing with normal MIDI controllers using the special Velocity controllers, this one allows you to select only some notes to follow.

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

**Min value** defines the minimum value for the target parameter.

**Max value** defines the maximum value for the target parameter.

Enable MIDI program change enables processing program change MIDI message.

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

Learn enables or disables MIDI learn.
Simulate program change via controller panel

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

**Learn**
Learn enables or disables MIDI learn.

**Channel**
Channel defines the controller MIDI channel.

**Controller**
Controller defines the source controller.

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

Program change in presets panel

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

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

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

Zoomer

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

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

Value button

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

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

Graph editor

Graph editor will show and edit one or more graphs.

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

Knob
Knob is an alternative to a tracker, which simulates physical knobs.

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

### Switcher

<table>
<thead>
<tr>
<th>Mode</th>
<th>Linear-phase</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Mode</strong></td>
<td><strong>Linear-phase</strong></td>
</tr>
</tbody>
</table>

Switcher is an alternative to a tracker or knob control, but it has a limited set of values.

- **Left mouse button** shows a menu with list of all possible values. This function might be unavailable in certain cases when the number of possible values is too high.
- **Right mouse button** selects the default value.
- **Up** and **Down** arrow keys, **buttons** in the control and **mouse-wheel** increase or decrease the value.
Installation

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

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

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

Installation on Windows

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

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

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

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

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

Installation on Mac OS X

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

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

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

Uninstallation on Windows

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

Uninstallation on OSX

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

Performance precautions

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

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

**Troubleshooting**

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

**GPU compatibility**

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

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

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

And place the GPU.xml included in the zip into

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

**Memory limits of 32-bit platform**

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

**Updating**

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

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

**Using touch-screen displays**

Touch screen displays are supported on Windows 8 and newer and the GUI has been tweaked to provide a good workflow. Up to 16 connections/fingers/inputs are supported. Any input device such as touch-screens, mouse, tablets are supported. These are the main gestures used by the plugins:

- Tap = left click
- Double tap = double click
- Tap & hold and quickly tap next to it with another finger = right click. Tap & hold is a classic right-click gesture, however that doesn't provide a good workflow, so came up with this method, which is much faster and does not collide with functionality of some elements.

**Purchasing and activation**

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

To activate the software simply drag & drop the licence file onto the plugin. Unfortunately some hosts (especially on Mac OS X) either do not allow drag & drop, or make it just too clumsy, so you can use Home/Activate in any of the plugins and follow the instructions. For more information about activation please check the online video tutorial.
You are allowed to use the software on all your machines, but only you are allowed to operate the software. The licences are "to-person" as defined in the licence terms, therefore you can use the software on all your computers, but you are the only person allowed to operate them. MeldaProduction can provide a specialized licence for facilities such as schools with different licence terms.

## Quick start with your host

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

### Cubase

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

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

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

### Logic

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

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

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

### Studio One

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

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

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

### Digital performer

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

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

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

### Reaper

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

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

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

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

### Live

In Session view, select the track you would like to insert the plugin to. At the left top of Ableton Live's interface, click on the Plug-in Device Browser icon (third icon from the top). From the plug-ins list choose the plugin (from MeldaProduction folder), double click on it or drag &
The X/Y grid usually doesn’t provide any parameters of the plugin. This is because the plugins have too many of them, so you have to select them manually. Check Live’s documentation for more information.

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

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

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

**ProTools**

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

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

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

**FL Studio**

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

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

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

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

**GUI styles, editor modes and colors**

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

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

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

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

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

Innovative Thinking

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

Sound Matters

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

Inspiring User Interface

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

Easy to Use, Yet Versatile

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

Never-Ending Improvements
Most companies create a plugin, sell it and abandon it. We improve our plugins, add features, optimize... until there is nothing left to improve and there are no more ideas. Unfortunately that hasn't happened yet :). And the best thing is that the updates are free-for-life!

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

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