MUltraMaximizer

Overview

MUltraMaximizer is a state-of-the-art mastering multiband brickwall limiter that makes your recordings sound louder with minimum distortion or artifacts and you do not even need to be a scientist to use it!

Introduction

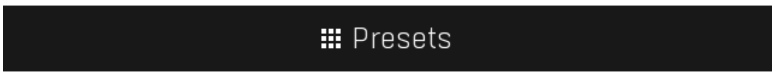
On opening the plugin window, the main function is the **Input gain** control. As the input gain is raised, the audio signal becomes more limited, which increases the loudness, but also progressively removes the dynamics, which can even lead to distortion if overused. Be careful not to set the input gain too high.

Peak meters on the right show the output peaks of the master, which should reach 0dB, but only temporarily. If you do not see them, click the **Meters** button

After the **Input gain** has been set, attention turns to the **Output gain**, which controls the output level and the amount of saturation. This parameter is used to move the output peak meter up to reach the 0dB level. The more it touches the top of the output peak meter, the more saturation and loudness you get, but also the more the sound is distorted.

Finally the **Slope**, controls the frequency balance. This is able to add both bass boom and highs, if needed.

Saturation panel contains the parameters of the saturator used on the output signal, and ensures there will be no peaks above 0dB. This can also improve sound clarity and increase the loudness, but as the concept of saturation is about distorting the sound, it should be used sparingly. The saturation can be disabled completely, which results in the output no longer being limited to 0dB and allows the use of an additional single-band limiter if desired.



Presets

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

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

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

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

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

Left arrow
Left arrow button loads the previous preset.

Right arrow
Right arrow button loads the next preset.

Randomize
Randomize button loads a random preset.

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

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

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

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

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

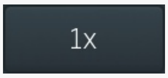
Dry/wet affects determines, for Multiband plug-ins, which multiband parameters are affected by the Global dry/wet control.

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

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

Plugin toolbar

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



Upsampling

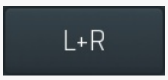
Upsampling can potentially improve sound quality by processing at a higher sample rate. Processors such as compressors, saturators, distortions etc., which employ nonlinear processing generate higher harmonics of the existing frequencies. If these frequencies exceed the Nyquist rate, which equals half of the sampling rate, they get mirrored back under the Nyquist rate. This is known as aliasing and is almost always considered an artifact. This is because the mirrored frequencies are no longer harmonic and sound as digital noise as this effect does not physically occur in nature. Upsampling (or oversampling) reduces the problem by temporarily increasing the sampling rate. This moves the Nyquist frequency which in turn, diminishes the level of the aliased harmonics. Note that the point of upsampling is not to remove harmonics, we usually add them intentionally to make the signal richer, but to reduce or attenuate the harmonics with frequencies so high, that they just cannot be represented within the sampling rate.

To understand aliasing, try this experiment: Set the sampling rate in your host to 44100 Hz. Open MOscillator and select a "rectangle" or "full saw" waveform. These simple waveforms have lots of harmonics and without upsampling even they become highly aliased. Now select 16x upsampling and listen to the difference. If you again select 1x upsampling, you can hear that the audio signal gets extensively "dirty". If you use an analyzer (MAnalyzer or MEqualizer for example), you will clearly see how, without upsampling, the plugin generates lots of inharmonic frequencies, some of them which are even below the fundamental frequency. Here is another, very extreme example to demonstrate the result of aliasing. Choose a "sine" shape and activate 16x upsampling. Now use a distortion or some saturation to process the signal. It is very probable that you will be able to hear (or at least see in the analyzer) the aliased frequencies.

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

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

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



Channel mode

Channel mode button shows the current processing channel mode, e.g. **Left+Right (L+R)** indicates the processing of left and right channels. This is the default mode for mono and stereo audio material and effectively processes the incoming signal as expected.

However the plugin also provides additional modes, of which you may take advantage as described below. Mastering this feature will give you unbelievable options for controlling the stereo field.

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

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

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

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

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

As well as a simple gain control there are various creative uses for this channel mode.
Using a **compressor** on the mid channel can widen the stereo image, because in louder parts the mid part gets attenuated and the stereo becomes more prominent. This is a good trick to make the listener focus on an instrument whenever it is louder, because a wider stereo image makes the listener feel that the origin of the sound is closer to, or even around them.
A **reverb** on the mid part makes the room appear thin and distant. It is a good way to make the track wide due to the existing stereo content, yet spacey and centered at the same time. Note that since this effect does not occur naturally, the result may sound artificial on its own, however it may help you fit a dominant track into a mix.
An **equalizer** gives many possibilities - for example, the removal of frequencies that are colliding with those on another track. By processing only the mid channel you can keep the problematic frequencies in the stereo channel. This way it is possible to actually fit both tracks into the same part of the spectrum - one occupying the mid (centre) part of the signal, physically appearing further away from the listener, the other occupying the side part of the signal, appearing closer to the listener.
Using various **modulation effects** can vary the mid signal, to make the stereo signal less correlated. This creates a wider stereo image and makes the audio appear closer to the listener.

Side (S) mode is complementary to M mode, and allows processing of only the side (stereo) part of the signal leaving the mid intact. The same techniques as described for M mode can also be applied here, giving the opposite results.
Using a **gain** control with positive gain will increase the width of the stereo image.
A **compressor** can attenuate the side part in louder sections making it more monophonic and centered, placing the origin a little further away and in front of the listener.
A **reverb** may extend the stereo width and provide some natural space without affecting the mid content. This creates an interesting side-effect - the reverb gets completely cancelled out when played on a monophonic device (on a mono radio for example). With stereo processing you have much more space to place different sounds in the mix. However when the audio is played on a monophonic system it becomes too crowded, because what was originally in two channels is now in just one and mono has a very limited capability for 2D placement. Therefore getting rid of the reverb in mono may be advantageous, because it frees some space for other instruments.
An **equalizer** can amplify some frequencies in the stereo content making them more apparent and since they psycho acoustically become closer to the listener, the listener will be focused on them. Conversely, frequencies can be removed to free space for other instruments in stereo.
A **saturator / exciter** may make the stereo richer and more appealing by creating higher harmonics without affecting the mid channel, which could otherwise become crowded.
Modulation effects can achieve the same results as in mid mode, but this will vary a lot depending on the effect and the audio material. It can be used in a wide variety of creative ways.

Mid+Side (M+S) lets the plugin process both mid and side channels together using the same settings. In many cases there is no difference to L+R mode, but there are exceptions.
A **reverb** applied in M+S mode will result in minimal changes to the width of the stereo field (unless it is true-stereo, in which case mid will affect side and vice versa), it can be used therefore, to add depth without altering the width.
A **compressor** in M+S mode can be a little harder to understand. It basically stabilizes the levels of the mid and side channels. When channel linking is disabled in the compressor, you can expect some variations in the sound field, because the compressor will attenuate the louder channel (usually the mid), changing the stereo width depending on the audio level. When channel linking is enabled, a compressor will usually react similarly to the L+R channel mode.
Exciters or saturators are both nonlinear processors, their outputs depend on the level of the input, so the dominant channel (usually mid) will be saturated more. This will usually make the stereo image slightly thinner and can be used as a creative effect.

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

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

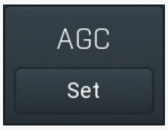
There are also 4 subsidiary modes: **Left & zero Right (L(R0))**, **Right & zero Left (R(L0))**, **Mid & zero Side (M(S0))** and **Side & zero Mid (S(M0))**. Each of these processes one channel and silences the other.

Surround mode is not related to stereo processing but lets the plugin process up to 8 channels, depending on how many the host supplies. For VST2 plugins you have to first activate surround processing using the **Activate surround** item in the bottom. This is a global switch for all MeldaProduction plugins, which configures them to report 8in-8out capabilities to the host, on loading. It is disabled by default, because some hosts have trouble dealing with such plugins. After activation, restart your host to start using the surround capabilities of the plugins. Deactivation is done in the same way. Please note that all input and output busses will be multi-channel, that includes side-chain for example. For VST3/AU/AAX plugins the activation is not necessary.

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

Ambisonics mode provides support for the modern 3D systems (mostly cinema and VR) with up to 64 channels (ambisonics 7th order). Support for this is still quite rare among the DAWs, so this needs to be activated in all DAWs using the **Activate ambisonics** item in the bottom. This is a global switch for all MeldaProduction plugins, which configures them to report 64in-64out capabilities to the host, on loading. After activation, restart your host to start using the ambisonics capabilities of the plugins. Deactivation is done in the same way. Please note that all input and output busses will be multi-channel, that includes side-chain for example.

First place the plugin on an ambisonics track, supported are all orders from 1st (4 channels) to 7th (64 channels). Then select **Ambisonics** from the plug-in's Channel Mode menu. Finally select the **Ambisonics settings** in the menu and configure the Ambisonics order and other settings if needed. The plugins will regard this mode as a natural extension of 2 channel processing. For example, a compressor will process each channel separately or measure the level by combining the levels of all of the inputs provided.



AGC

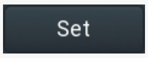
AGC button enables or disables the automatic gain control - the automatic adjustment of the output volume such that it matches the input volume. Human hearing is very adaptable. In fact differences in loudness, for example when loading a preset, may go unnoticed and instead be perceived by the listener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

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

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

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

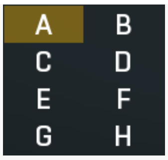
As a general rule of thumb as for all audio processing tasks, use it only if you know you need it. AGC is a powerful tool that can make your workflow easier, but it can also be damaging.



Set

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

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



A-H presets selector

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

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

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

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

A/B

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

Morph

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

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

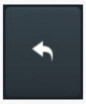
Please note that the ABCD morphing itself cannot be automated and that, while morphing, the changes to the underlying parameters are not notified to the host (there may be hundreds of change events).

Copy

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

Paste

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



Undo

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

Redo

Redo button reverts the last undo operation.

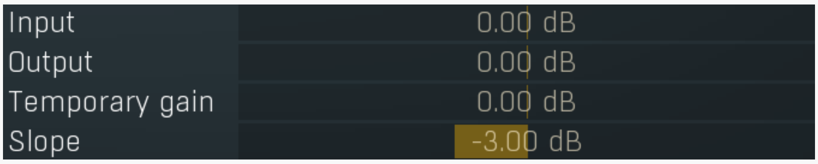
WAV

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

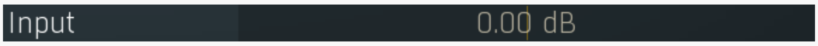
Please note that the files will be overwritten, so make a copy first if you want to keep the original.

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

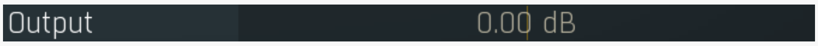
Global parameters panel



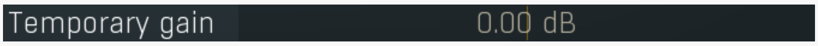
Global parameters panel contains global controls, which are usually relevant to global processing performed either before the signal reaches the crossover and gets split into bands, or after the signals are processed and summed back to the master signal.



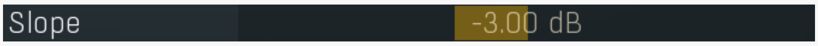
Input gain
Input gain defines the power modification applied to the incoming signal, before it is split into bands.



Output gain
Output gain defines the gain applied to the output signal, right after it is summed from the bands, but before the signal is passed into the saturator. By increasing this value you can get more saturation, hence the sound will be louder but still won't get above the 0dB limit.

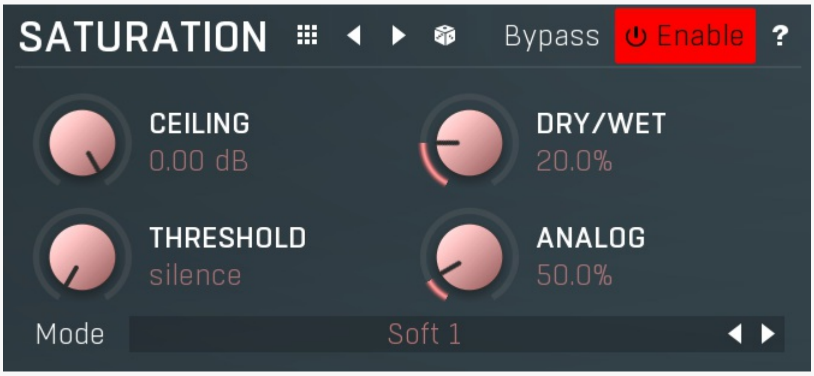


Temp gain
Temp gain defines a temporary power modification applied to the input signal and then reversed on the output. You can achieve the same effect by setting **Input gain** to a value **G** and **Output gain** to value **-G**.



Slope
Slope defines the balance between the band thresholds therefore it basically defines the resulting sound character. A negative value provides more energy for bass frequencies, which is often desirable.

Saturation panel



Saturation panel contains parameters of the master effect that is processed after the individual bands are processed and the signal is mixed back together.

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Left arrow

Left arrow button loads the previous preset.

Right arrow

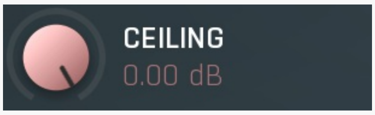
Right arrow button loads the next preset.

Randomize

Randomize button loads a random preset.

Enable

Enable button enables or disables the effect.



Ceiling

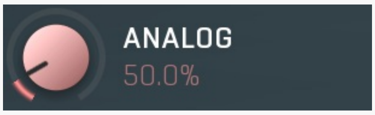
Ceiling defines the gain applied to the output after the saturation, hence this is the maximal level that the output can reach.
Range: -24.00 dB to 0.00 dB, default 0.00 dB

Dry/wet

Dry/wet defines ratio between dry and wet signals. 100% means fully processed, 0% means no processing at all.
Range: 0.00% to 100.0%, default 20.0%

Threshold

Threshold determines the minimal signal level above which the effect starts to apply. By lowering the threshold you increase loudness and also distortion.
Range: silence to 0.00 dB, default silence

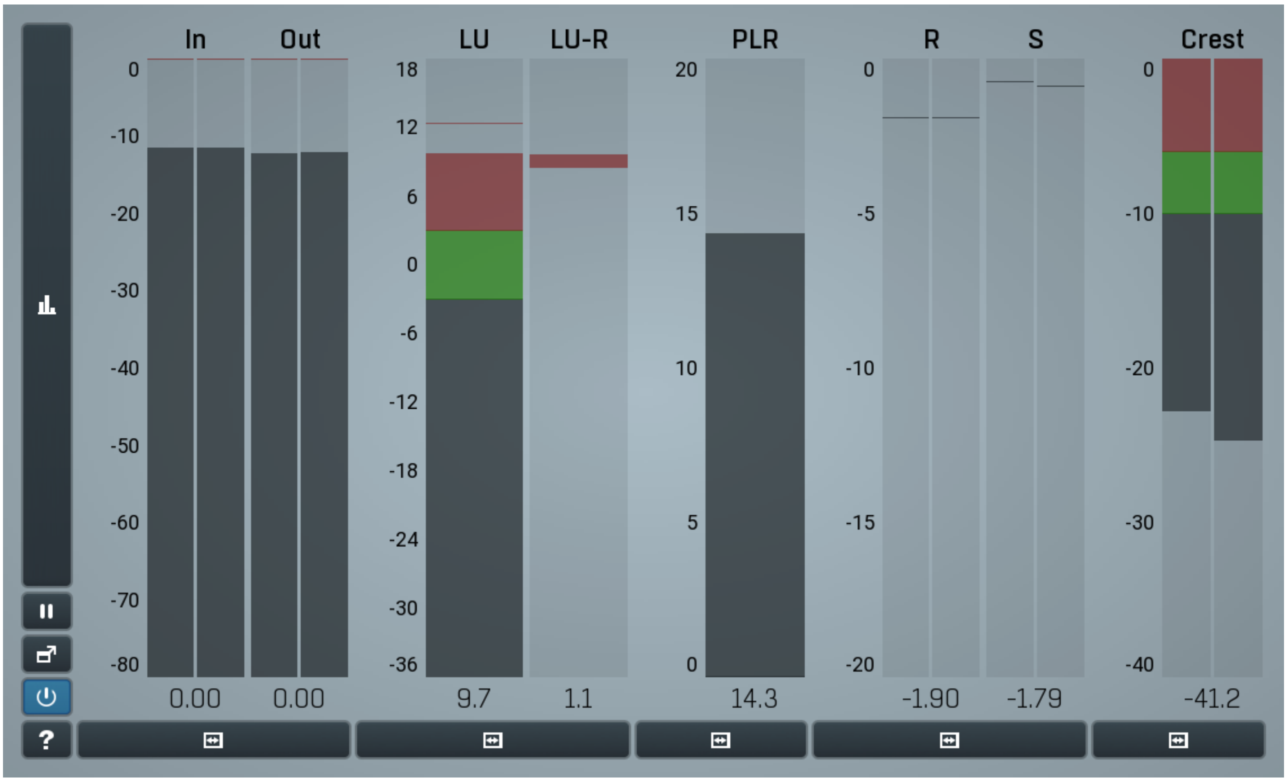


Analog

Analog determines the amount of even harmonics added to the signal in addition to the main saturation, which is typical for analogue saturation. The amount of the harmonics is dependent on the saturation signal, unlike the full harmonic control in the **Harmonics** panel, which is completely independent of the actual saturation processing.
Range: 0.00% to 500.0%, default 50.0%

Mode

Mode defines the limiting shape.



Global meter view

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

Numbered band meters display the input levels for each band, but also indicate the gain reduction by the colored top part. If the top part is red, it is indicating a gain reduction. If it is green, the band is actually increasing the gain, which usually means expansion.

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.

R meter shows gain reduction for each channel. Negative values, running down from the top, mean that compression or limiting is occurring. The lower the value, the stronger the effect. For maximum transparency you should try to achieve the least amount of gain reduction. Expansion is not indicated in this meter.

S meter shows the saturation level for each channel. This basically shows how much signal over 0dB has passed through the limiting section. A limiter can remove all of the peaks above 0dB only if it is in single-band mode and attack is set to 0ms or using the look-ahead + true hold trick. In all other cases, there can and will be peaks above 0dB passing through the main dynamics unit. These peaks are then saturated or clipped and are indicated using this saturation meter. Saturation provides a crisper signal, in higher levels it can cause audible distortion, which may be desired as it brightens the signal. Less saturation provides a cleaner signal. In higher levels you may hear severe pumping however.

LU meter shows the output loudness in EBU-18 scale. The loudness metering follows the ITU-R BS.1770-3 and EBU 3341 specifications. The metering units used are LU (Loudness Units) with 0 LU defined as -23 LUFS (LU Full Scale) and you should consider the LU values to be relative - using them to compare the loudness values between different signals. If the difference in loudness between 2 signals is 10 LU, it is approximately 10 dB as well.

Please note that you should still use your ears to judge loudness properly as there is still no accurate model of human loudness perception and every measurement is only an approximation. Loudness perception is also individual.

If you right click on the meter, additional settings will be displayed. Maximum value displays the maximum since the analysis started, rather than the recent maximum. Loudness pre-filtering uses EBU standard filters to simulate human perception. However, you may want to disable this to get more technical measurements.

There are 3 types of loudness measurements, all following the EBU specifications.
Momentary loudness uses an RMS sliding analysis window of 400 milliseconds; therefore it shows quick fluctuations in loudness.

Short-term loudness works in the same way, but uses a window of 3 seconds, therefore it provides more stable loudness measurements. **Integrated loudness** shows the overall loudness, hence it is affected by the whole track from the beginning of the playback until you reset it by clicking on the value field. The host may reset it too; it depends on your host.

Please note that the **Integrated loudness** is NOT the same as an averaged loudness, as it ignores quiet passages. Imagine a track which is generally quiet but has a few loud sections. The averaged loudness will be less than the Integrated loudness. Its calculation uses gating to ignore those quiet passages (levels less than 10 LU less than the current ungated level) of the track. Essentially, **Integrated loudness** is a measure of the loudest sections of the track.

LU-R meter shows the output loudness range in EBU-18 scale. The loudness meter, **LU**, follows the ITU-R BS.1770-3 and EBU 3342 specifications, so 0 LU (Loudness Units) represents -23 LUFS. The loudness range meter, **LU-R**, essentially describes how dynamic the signal is, measured over a sliding analysis window of 3 seconds. Too low a dynamic range usually means the signal is over-compressed. You should compare the dynamic range of your material with other mastered recordings in the same style.

Green bars indicate that the loudness or range is in the preferred regions, red bars suggest that the levels or range need attention.

PLR meter displays the true-peak-to-loudness ratio. It's similar to the crest meters, but based on the modern EBU loudness standard. It ranges from 0 to 20 and the higher the value, the less compressed (more dynamic) is the audio material. It tells you the difference between the maximum peak level and the **Integrated** loudness.

This is very useful when submitting audio tracks to services such as Spotify or iTunes, which normalize the loudness of the audio. Let's show this with an example: You are going to submit a song which has the true peak level of -1 dBTP and integrated loudness of -16 LUFS (which is the usual recommended loudness), so the PLR is -1 - (-16) = 15. This means that the song requires 15 dB of headroom above the normalization level to play without clipping. The song will play without clipping in iTunes (normalization level -16 LUFS, maximum true-peak level -1 dBTP, hence 15dB headroom), but the song might be turned down (by e.g. Tidal and YouTube), limited (by e.g. Spotify) or clipped when played on a platform with less headroom.

As a different example, you may be working on a cinematic score, with target loudness of -26 LU, and you might have used all the headroom for the requested maximum true peak level of -1 dBTP. The PLR is therefore 25. If you try to submit such a file to iTunes, it will normalize it to -16 LUFS, which means increasing the loudness, but your song requires 25 dB headroom and iTunes provides only 15, therefore it will either be rejected or processed in some way (clipped, limited...). Neither of these options are "good", therefore you will need to provide a different master for this platform.

Crest meter shows the output crest factor (calculated as RMS divided by peak level), which essentially indicates how extreme the peaks are in the output waveform. The lower the value is, the more peaks there are in the output, the more dynamic it is. If the value reaches 0dB, then the output is over-compressed and flattened and you should consider going easier on the compressor and limiter.

The range from 0dB to -6dB is red and you should prevent your master output from remaining in this range as that would mean it is extremely over-processed. The green range from -6dB to -10dB is the range most that recordings are in, usually jumping below the -10dB.

Please note that others calculate the crest factor as the peak level of the waveform divided by the RMS value of the waveform, and the higher the value, the more peaks there are in the output, the more dynamic it is.

Time graph

Time graph button switches between the metering view and the time-graphs. The metering view provides an immediate view of the current values including a text representation. The time-graphs provide the same information over a period of time. Since different time-graphs often need different units, only the most important units are provided.

Pause

Pause button pauses the processing.

Popup

Popup button shows a pop-up window and moves the whole metering / time-graph system into it. This is especially useful in cases where you cannot enlarge the meters within the main window or such a task is too complicated. The pop-up window can be arbitrarily resized. In metering mode it is useful for easier reading from a distance for example. In time-graph mode it is useful for getting higher accuracy and a longer time perspective.

 **Enable**

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

 **Collapse**

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

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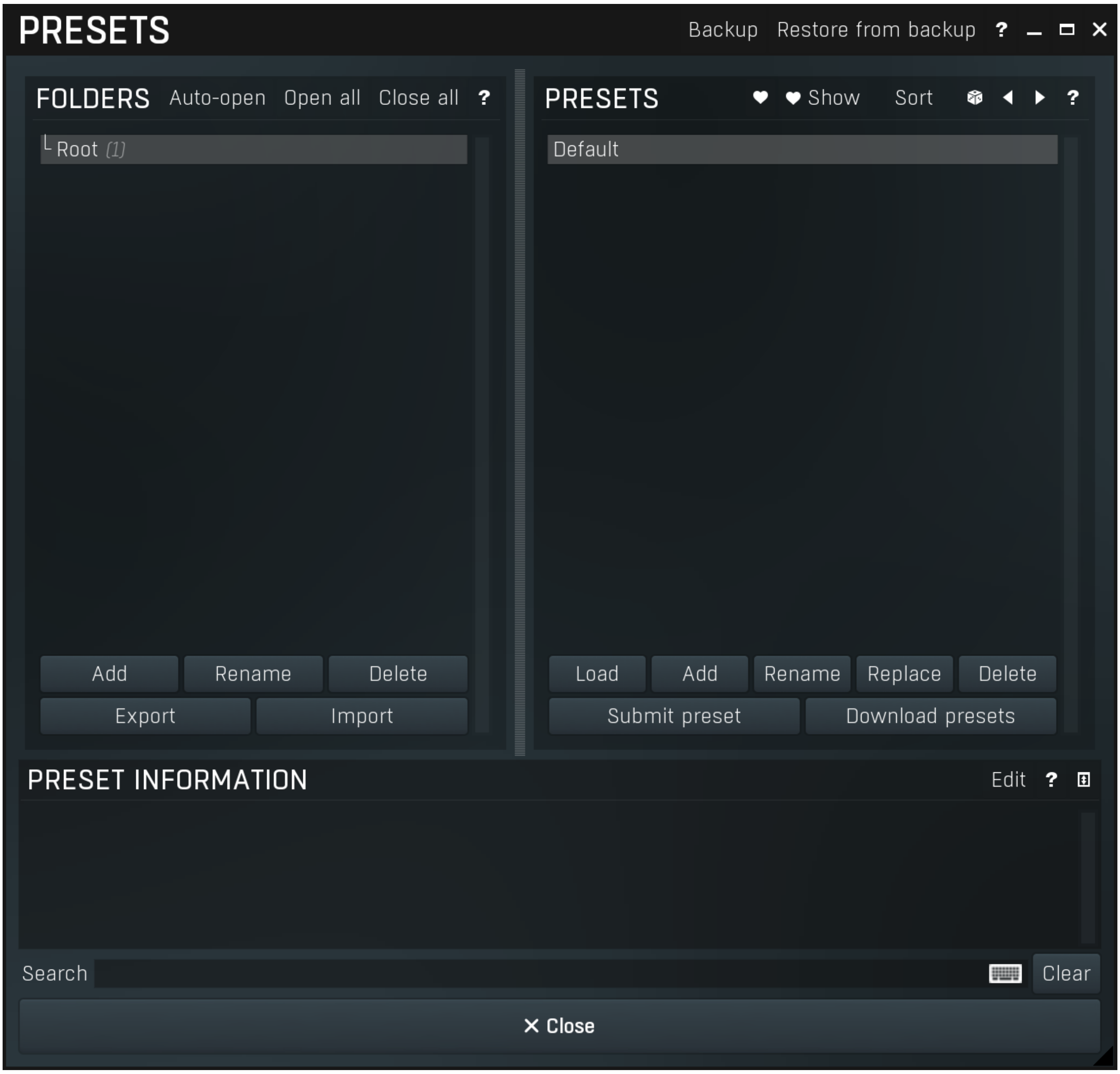
Preset selector



Preset management window provides management for your presets.

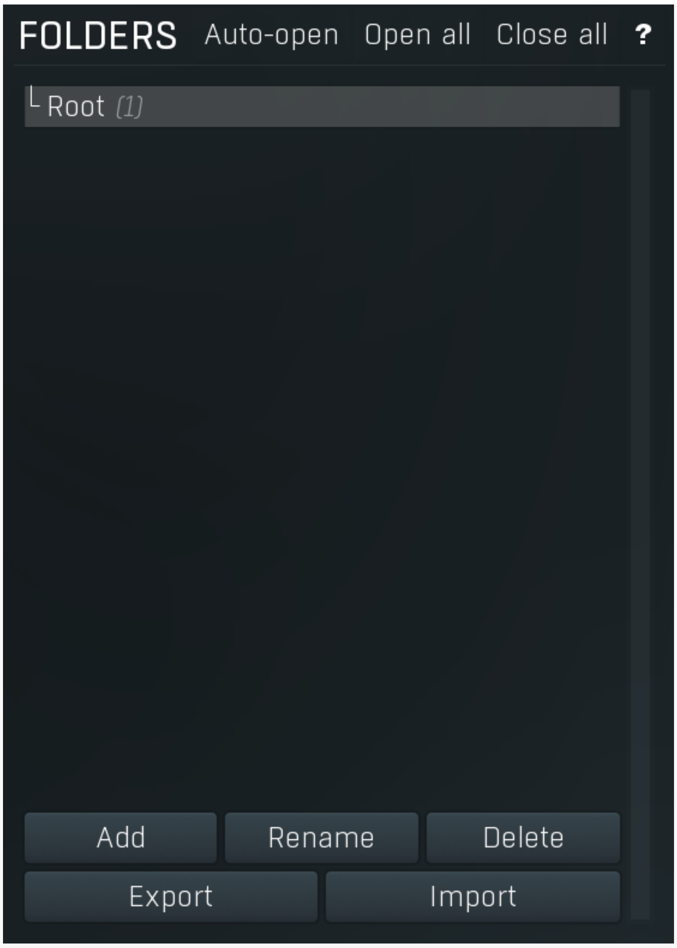
Backup

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

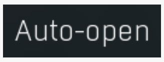
Restore from backup

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

Folders tree

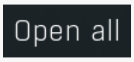


Folders tree lets you organize your presets into any number of folders. Use the buttons at the bottom of the window to create, rename or delete sub-folders. Note that these are not actual files & folders on disk, but are records in the preset database.



Auto-open

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

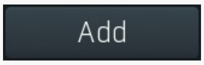


Open all

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

Close all

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

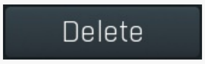


Add

Add button creates a new folder in the tree

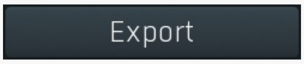
Rename

Rename button lets you rename the selected folder.



Delete

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



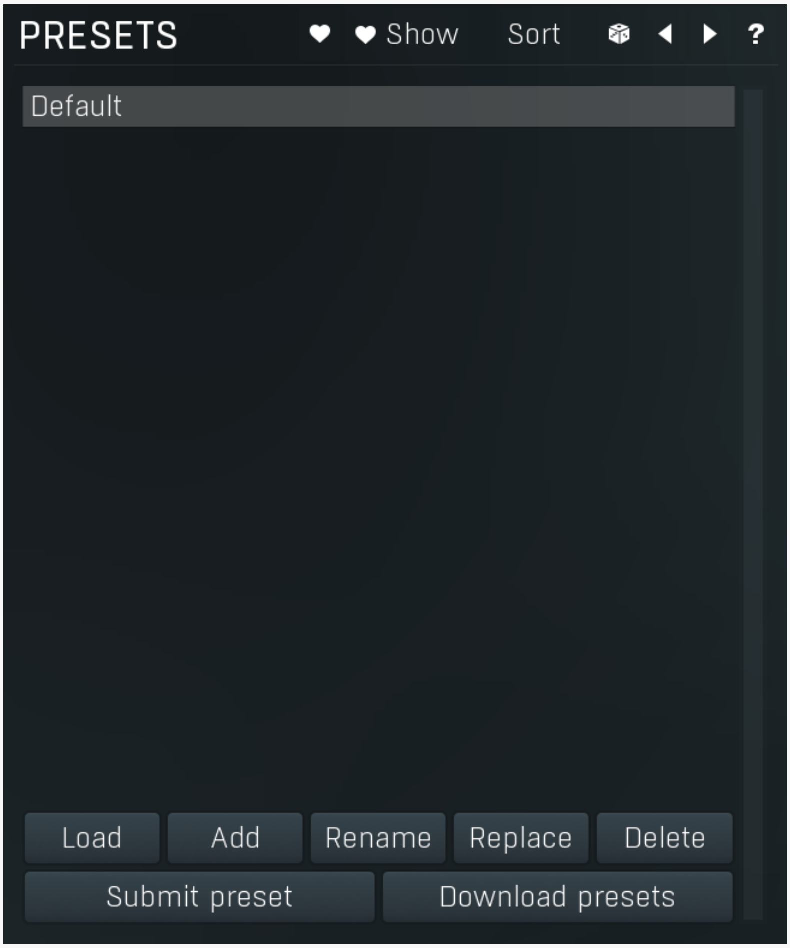
Export

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

Import

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

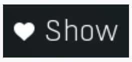
Presets list



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

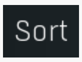
Favourite

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



Show

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



Sort

Sort button shows the presets sorted alphabetically.

Random

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

Previous

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

Next

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

Submit preset

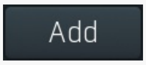
Submit preset button submits the selected preset to the online exchange servers and retrieves all the presets currently in the database. This feature serves as an online database of presets available for all the user community. Please do not submit garbage presets.

Download presets

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

Load

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

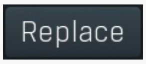


Add

Add button creates a new preset using the current settings.

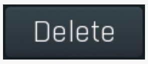
Rename

Rename button lets you rename the selected preset.



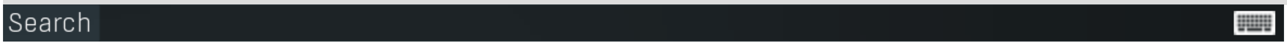
Replace

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



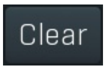
Delete

Delete button deletes the selected preset.



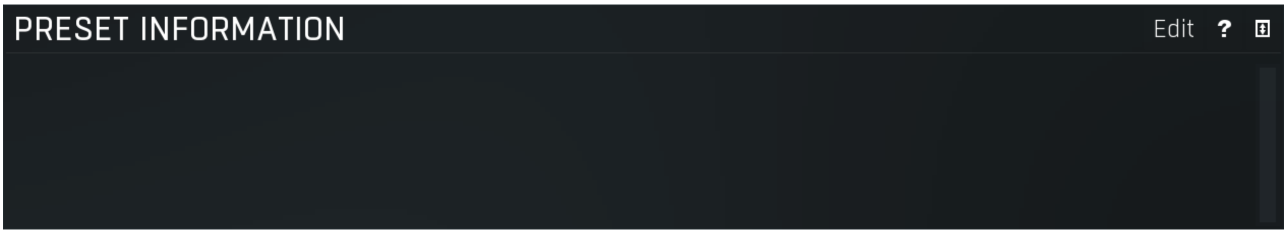
Search

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



Clear

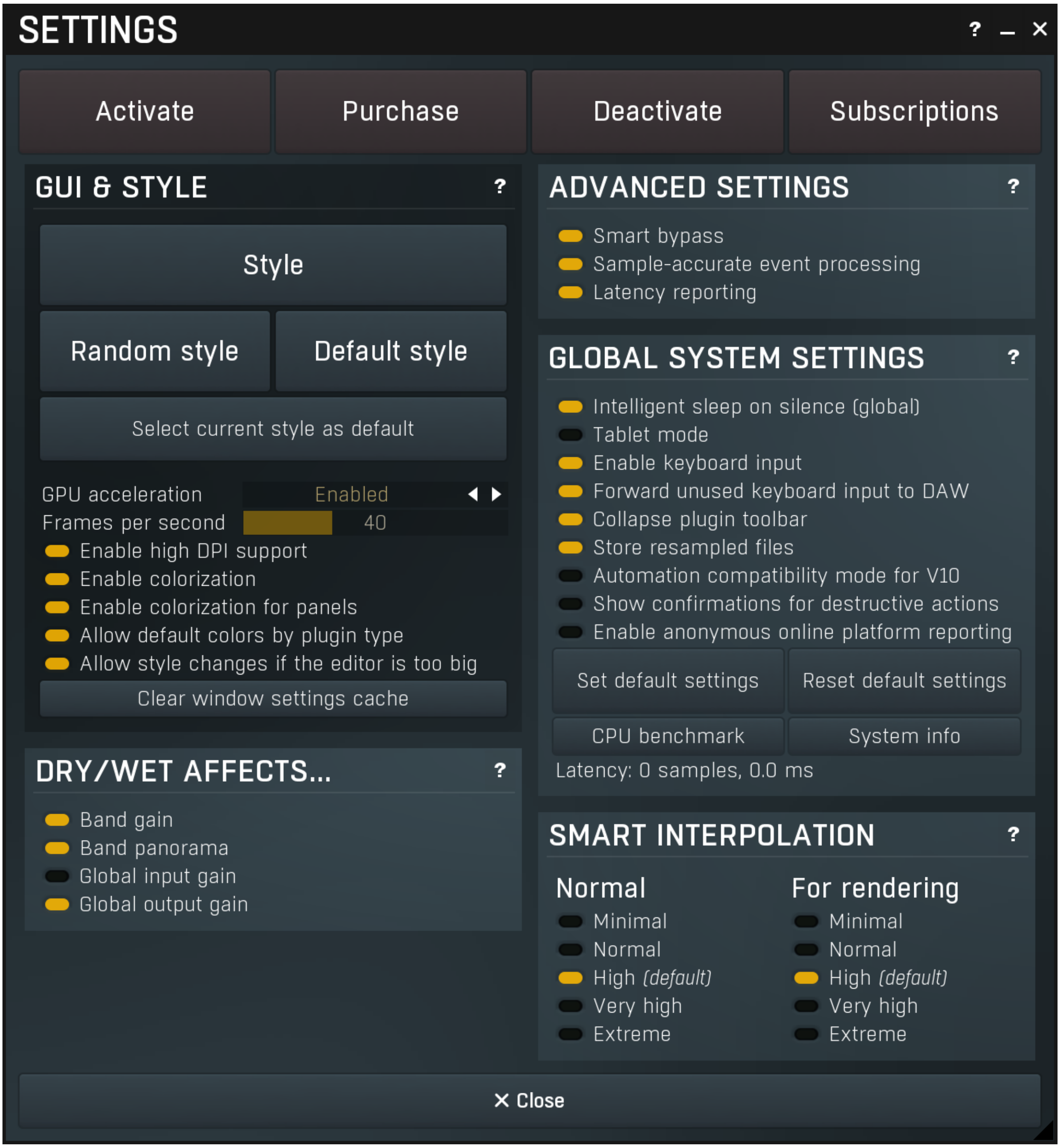
Clear button deletes all text in the search field.



Preset information

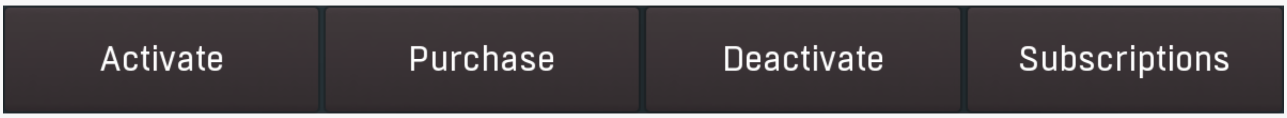
Preset information field contains optional information about the preset, which you can edit when creating or renaming the preset.

Plugin settings

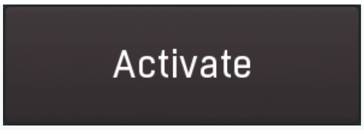


Plugin settings window offers more advanced settings and is available via the Settings button.

Licence panel

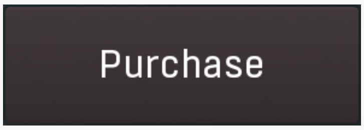


Licence panel lets you manage licences on this computer.



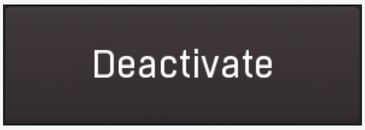
Activate

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



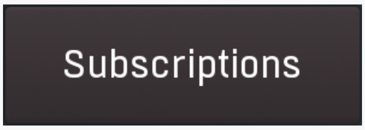
Purchase

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



Deactivate

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

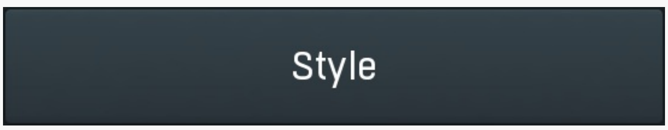


Subscriptions

Subscriptions button lets you manage the subscription based licencing.

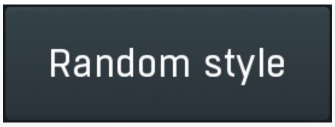
GUI & Style panel

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



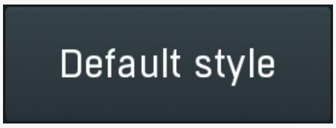
Style

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



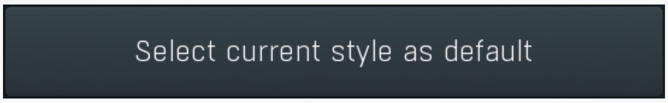
Random style

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



Default style

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



Select current style as default

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

GPU acceleration

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

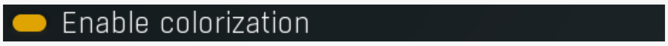
Frames per second

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

Enable high DPI / retina support

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

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



Enable colorization

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

Enable colorization for panels

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

Allow default colors by plugin type

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

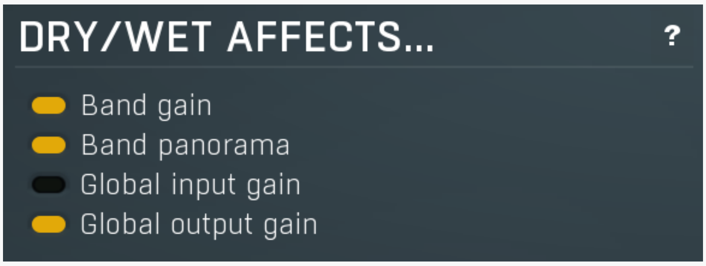
Allow style changes if the editor is too big

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

Clear window settings cache

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

Dry/wet affects... panel



Dry/wet affects... panel controls which multiband parameters are affected by the global dry/wet controls. Dry/wet generally controls the mix between the processed and input signal, however it is often advantageous to support its function with other parameter. In most cases it is good to enable everything except for global input gain. However note that this 'default' is not specified for most plugins for sake of maintaining backwards compatibility.

Band gain

Band gain makes the global **dry/wet** parameter affect the input gain of each band. In a way you may say that the input gain settings of all bands are applied to wet only when this is enabled. This is suitable in most cases, because that way lowering dry/wet to 0% provides a true "dry" response. If this is not enabled and dry/wet is 0% the output will still be affected by the input gain of each band.

Band panorama

Band panorama makes the global **dry/wet** parameter affect the input panorama of each band. In a way you may say that the input panorama settings of all bands are applied to wet only when this is enabled. This is suitable in most cases, because that way lowering dry/wet to 0% provides a true "dry" response. If this is not enabled and dry/wet is 0% the output will still be affected by the input panorama of each band.

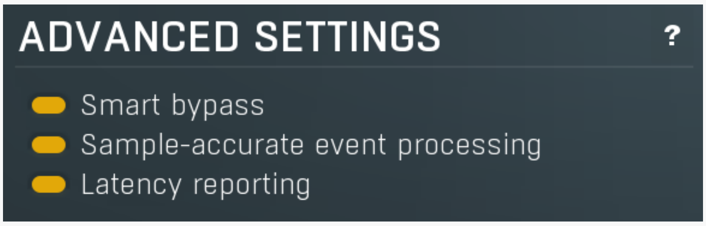
Global input gain

Global input gain makes the global **dry/wet** parameter affect the global input gain. This is usually not desired, because the input gain affects the functionality of most plugins, so this would affect the processed signal, which is not necessary, because the dry/wet itself is creating the mix between the dry and wet signals. However this control may be handy in specific creative scenarios.

Global output gain

Global output gain makes the global **dry/wet** parameter affect the global output gain. In a way you may say that the output gain is applied to wet only when this is enabled. This is suitable in most cases, because that way lowering dry/wet to 0% provides a true "dry" response. If this is not enabled and dry/wet is 0% the output will still be affected by the global output gain.

Advanced settings panel



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

Smart bypass

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

Sample-accurate event processing

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

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

Latency reporting

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

Global system settings panel

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

Intelligent sleep on silence (global)

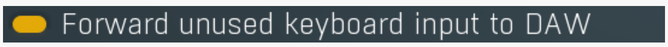
Intelligent sleep on silence (global) is a global switch, which disables the **Auto disable on silence** feature in all plugins on the system. It is provided "just in case" something goes wrong.

Tablet mode

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

Enable keyboard input

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



Forward unused keyboard input to DAW

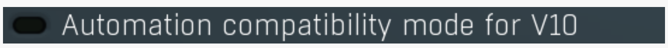
Forward unused keyboard input to DAW makes the plugin forward unused keyboard events to the DAW from its popups. If this is disabled, pressing say spacebar commonly used to start/stop playback won't work if a popup window is active. Enabling this makes this work and it is optional just in case your DAW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

Collapse plugin toolbar

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

Store resampled files

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



Automation compatibility mode for V10

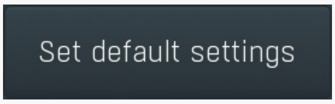
Automation compatibility mode for V10 reverts the set of automation parameters back to version 10 and earlier. Use this if you need the plugins to work with projects, which contain autmation, made using version 10 or older. In version 11 the list of automatable parameters have been highly simplified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

Show confirmations for destructive actions

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

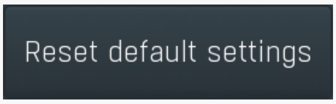
Enable anonymous online platform reporting

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



Set default settings

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



Reset default settings

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

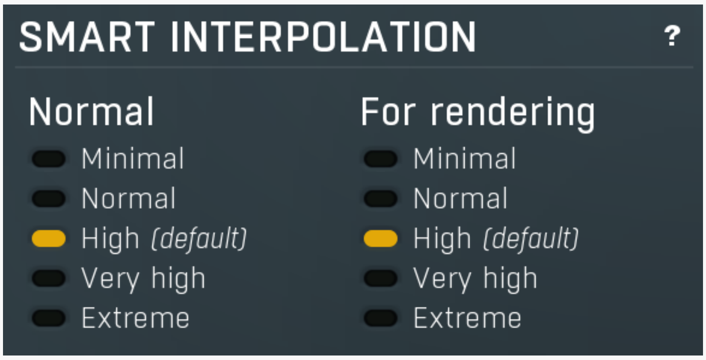
CPU benchmark

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

System info

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

Smart interpolation panel



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

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

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

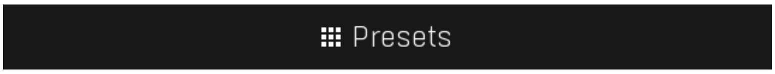
The smart interpolation mode controls the maximum number of samples being processed before the parameters are updated. **Minimal mode** uses 2048 samples and rarely will do anything unless processing offline. **Normal mode** uses 256 samples and usually is enough to achieve good quality results. **High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

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

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

MIDI editor

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



Presets

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

Left arrow

Left arrow button loads the previous preset.

Right arrow

Right arrow button loads the next preset.

Randomize

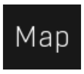
Randomize button loads a random preset.

Copy

Copy button copies the settings onto the system clipboard.

Paste

Paste button loads the settings from the system clipboard.



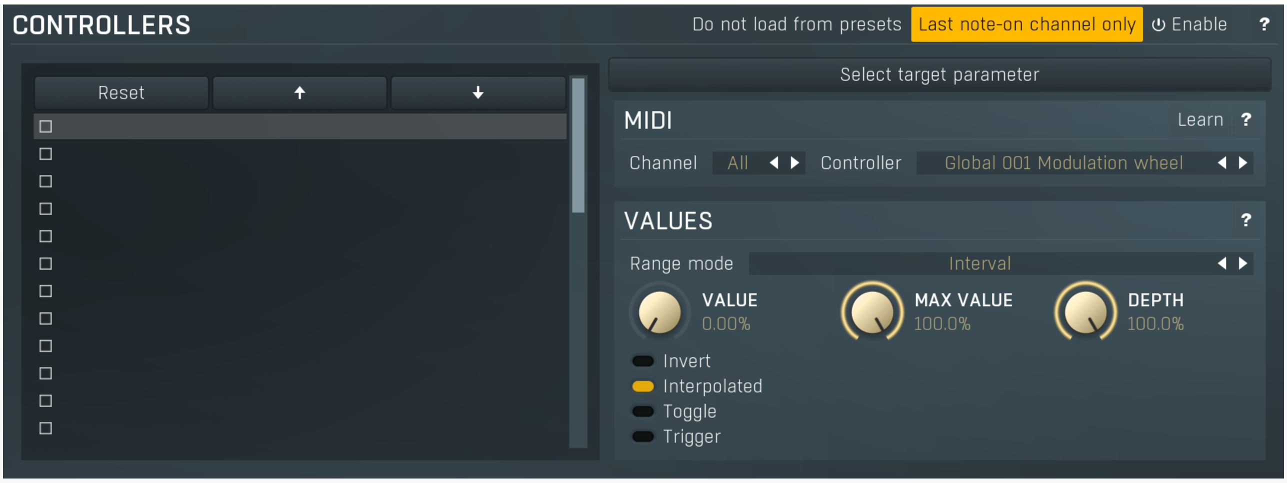
Map

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

Tab selector

Tab selector switches between subsections.

Controllers panel



Controllers panel contains settings of MIDI controllers.

Do not load from presets

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

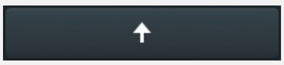
Last note-on channel only

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

Controllers

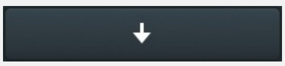
Reset

Reset button resets the selected controller to undefined state.



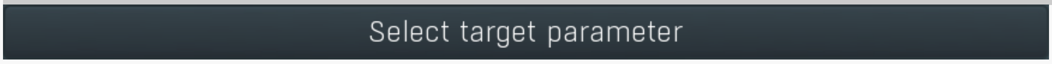
Up

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



Down

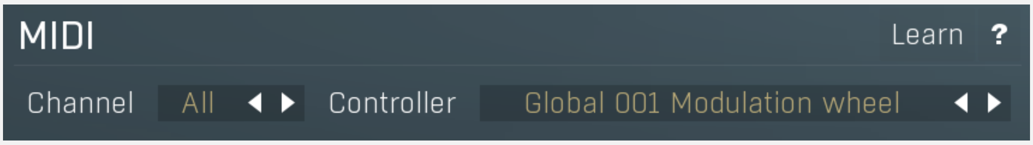
Down button moves the selected controller down one item, if possible. This may be useful when keeping things organized, but please note that if you have some multiparameter, modulator or another subsystem access the ranges of individual controllers, this function will reorder them, so these connections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

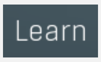


ParameterIndex

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

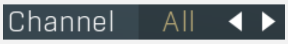
MIDI





Learn

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



Channel

Channel defines the controller MIDI channel.

Controller

Controller defines the source controller.

Values

Range mode

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

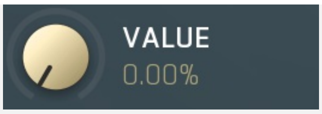
Up and down mode makes the values go above and below the selected **Value**, which is considered the center. The interval is made smaller if necessary.

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

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

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

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

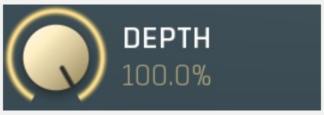


Value

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

Maximal value

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



Depth

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

Invert

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

Interpolated

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

Toggle

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

Trigger

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

Main controllers panel

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

Presets

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

Left arrow
Left arrow button loads the previous preset.

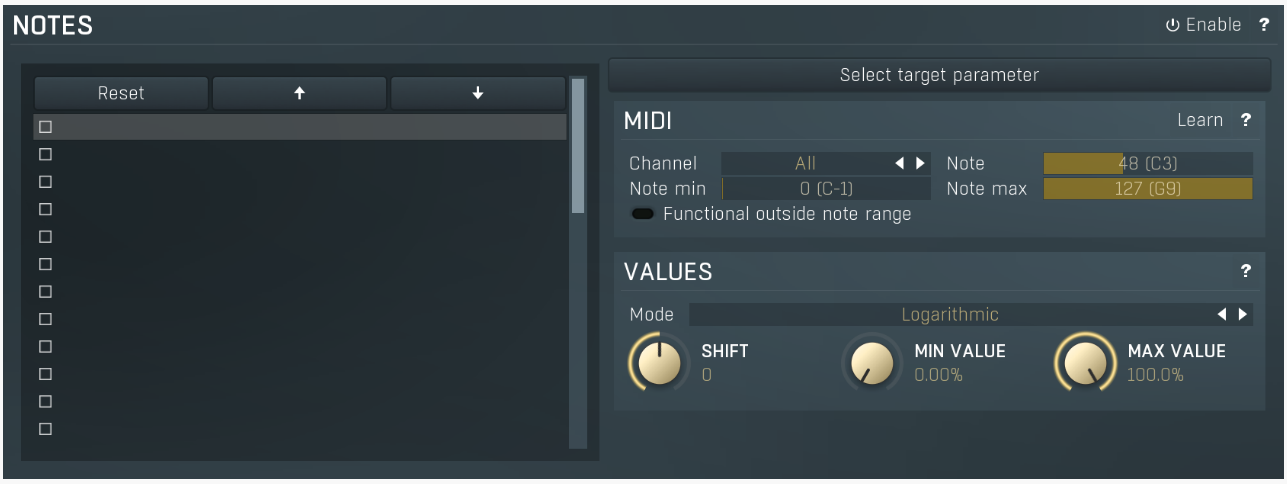
Right arrow
Right arrow button loads the next preset.

Randomize
Randomize button loads a random preset.

Controller
Controller defines the MIDI controller associated to this Main controller.

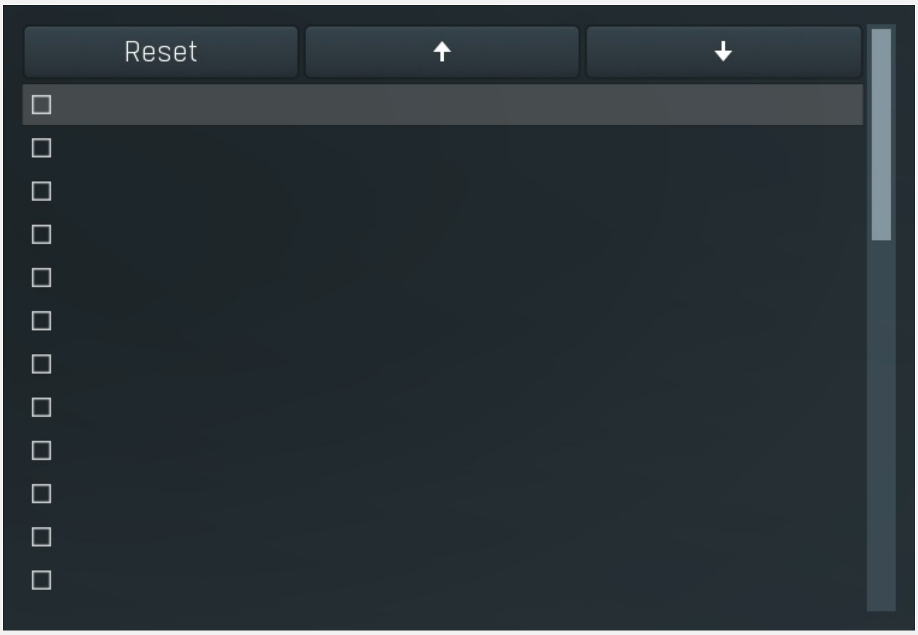
Learn
Learn enables or disables MIDI learn. When enabled, the plugin listens to the controllers you touch and associates them to the main controller.

Notes panel



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

NoteControllers



Reset

Reset button resets the selected controller to undefined state.

Up

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

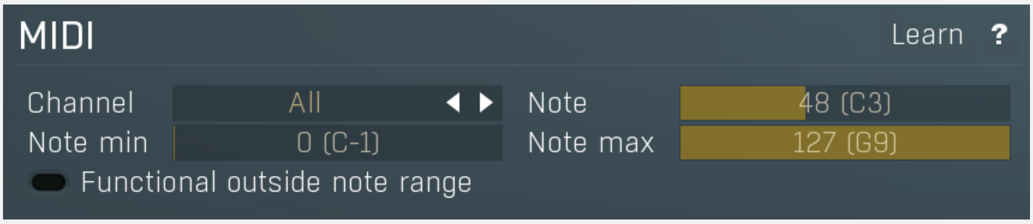
Down

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

Learn

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

MIDI



Channel

Channel defines the controller MIDI channel.

Note

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

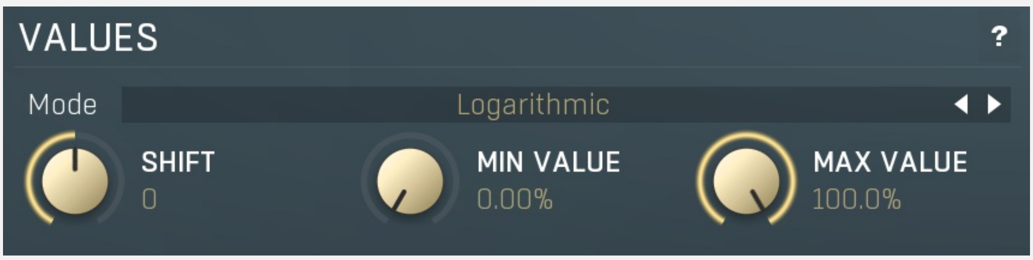
Note min

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

Functional outside note range

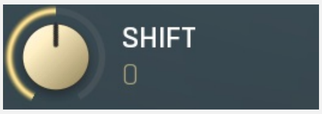
Functional outside note range makes the note controller work even if the note isn't in the specified range, clamping the value to the minimum or maximum.

Values



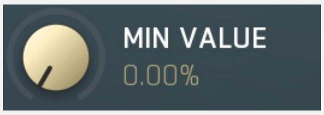
Mode

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



Shift

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



Min value

Min value defines the minimum value for the target parameter.

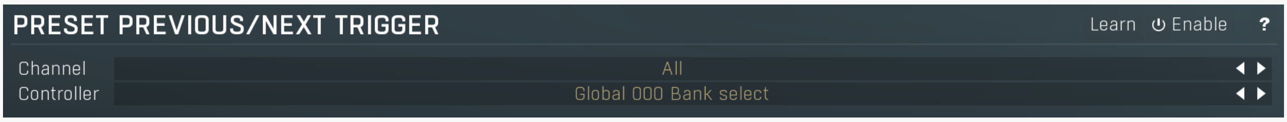
Max value

Max value defines the maximum value for the target parameter.

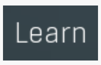
Enable MIDI program change

Enable MIDI program change enables processing program change MIDI message.

Preset previous/next trigger panel

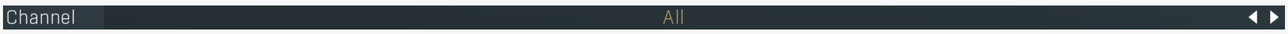


Preset previous/next trigger panel lets you select a MIDI controller, which will switch presets. It provides the same action as clicking the arrows next to the main preset button. When the controller value gets below 33%, the previous preset is loaded. When the controller value gets above 66%, the next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



Learn

Learn enables or disables MIDI learn.



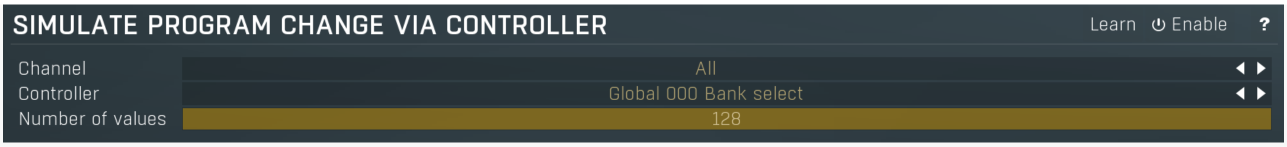
Channel

Channel defines the controller MIDI channel.

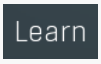
Controller

Controller defines the source controller.

Simulate program change via controller panel

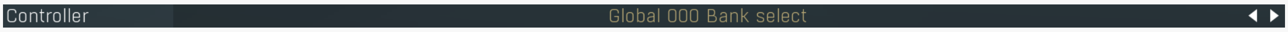


Simulate program change via controller panel lets you select a MIDI controller, that will work as program change, for convenience. You can use it then to switch between A-H presets or presets via panel below.



Learn
Learn enables or disables MIDI learn.

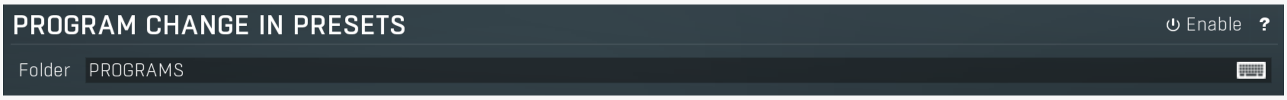
Channel
Channel defines the controller MIDI channel.



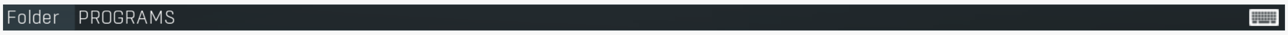
Controller
Controller defines the source controller.

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

Program change in presets panel



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



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

Used controls

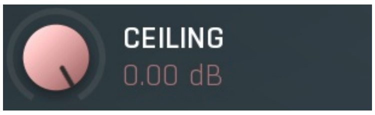
Here we discuss the general properties of all application controls. As a most important rule you should note, that you can always use any question mark button or F1 (or Ctrl+F1 or Ctrl+H) key with the mouse cursor over a specified control to get detailed information about what it does and how to use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

Value button

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

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

Knob



Knob simulates physical knobs used to edit various values.

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

Switcher

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, activation, introduction to audio plugins

Installation

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

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

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

Installation on Windows

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

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

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

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

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

Installation on Mac OS X

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

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

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

Uninstallation on Windows

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

Uninstallation on OSX

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

Deleting all data, presets etc.

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

Windows:
C:\ProgramData\MeldaProduction

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

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

Performance precautions

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

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

Troubleshooting

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

GPU compatibility

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

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

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

And place the GPU.xml included in the zip into

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

Memory limits of 32-bit platform

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

Updating

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

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

Using touch-screen displays

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

Purchasing and activation

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

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

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

Quick start with your host

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

Cubase

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

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

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

Logic

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

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

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

Studio One

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

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

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

Digital performer

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

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

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

Reaper

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

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

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

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

Live

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

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

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

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

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

ProTools

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

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

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

FL Studio

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

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

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

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

GUI styles, editor modes and colors

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

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

Dynamics = orange

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

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

MeldaProduction
The only limit is your imagination

About MeldaProduction

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

Innovative Thinking

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

Sound Matters

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

Inspiring User Interface

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

Easy to Use, Yet Versatile

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

Never-Ending Improvements

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

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

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