

# MAutoStereoFix



## Overview

MAutoStereoFix is an essential tool for processing stereo tracks, whether these are drum overheads, rooms or any kind of stereo signals including master tracks. It lets you first analyze the audio and then apply several "fixes" to improve the stereo field and make it sound centered as it should be. We would recommend using headphones when analyzing the stereo field, since doing that with monitors is nearly impossible even in the best mixing rooms.

When using 2 microphones to record a certain instrument (or instruments), there is nearly no chance that the stereo field will be balanced. That would be possible only for an infinitely small instrument recorded with a pair of identical microphones placed at the exact same distance from the instrument, and recorded using a perfectly flat interface with no phase alteration. None of these requirements are even remotely realistic.

Most engineers adjust manually the level between the 2 microphones and that's pretty much it. Unfortunately that's not enough to keep the stereo field balanced. For example, if one microphone is further away from the instrument (or its prominent part - snare drum in case of drum overheads for example), the sound it will record is going to be delayed and the track will psychoacoustically appear "sounding from the other side". Trying to fix that by increasing the volume of that microphone would then only make things worse and sadly that's the usual way to deal with this problem. Delaying one of the microphones would solve the problem, but manually searching for a perfect delay would take a tremendous amount of time. Even searching for a good volume balance takes a lot of time, adding delay to the equation is nearly impossible even for a skilled engineer.

It gets worse when it comes to spectrum. For example, one of the microphones may record a little less high frequency content due to a different high frequency response, location close to a dampening surface, or even the room acoustics. The instrument will then appear closer to the other side again, because high frequencies are absorbed by the air and our brains know that. Fixing that using a linear-phase equalizer would be even worse than setting up delays and volumes.

And the final stage is phase. Any of the devices in the recording chain can alter the phase, whether it is the microphone, AD converters and especially if you are using an analog deck. Fixing phase manually would probably mean using an equalizer featuring allpass filters and that would take forever.

The worst part is that a phase or spectrum difference would not only make the stereo field appear out of the center, it would also sound confusing, because the clues picked by the brain would just don't make sense. After all stereo recording is generally everything but natural.

**MAutoStereoFix solves all of these problems automatically.** It analyzes the audio and calculates the optimal levels, delays, polarity inversion, equalization and spectral phase correct, so that the stereo field becomes as centered as possible. Please note however that the plugin cannot magically fix poor recording conditions such as room acoustics or crazy microphone placement.

Presets

## Presets

Presets button shows a window with all available presets. A preset can be loaded from the preset window by double-clicking on it, 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.

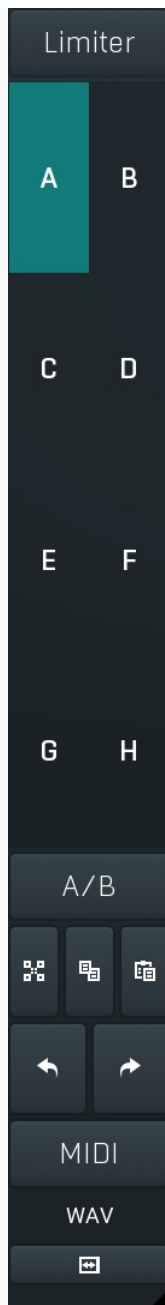
**Smart interpolation** adjusts the interpolation algorithm used when changing parameter values; the higher the setting the higher the audio quality and the lower the chance of zipping noise, but more CPU will be used.



### 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.

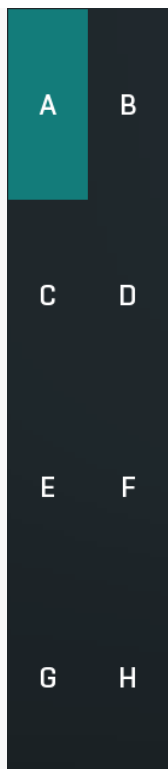
Limiter

### Limiter

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

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

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



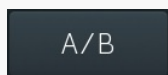
### A-H presets selector

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

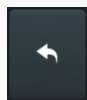
Copy button copies the current settings to the system clipboard. Other presets, upsampling, channel mode and other global settings are not copied.

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

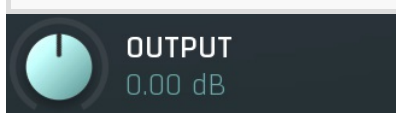
WAV button lets you process a file using the plugin with current settings. You can either click the button and select a file, or drag & drop the file (or multiple files) onto the button. If you let the plugin process WAV files, these will be saved with the original settings. If you use a different file 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

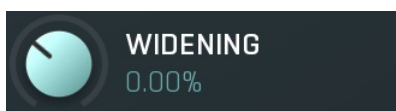
Collapse button minimizes or enlarges the panel to release space for other editors.



### Output gain

Output gain defines the power modification applied to the output signal.

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

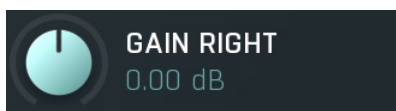


### Widening

Widening defines the broad-band stereo field widening depth. The algorithm is fully mono-compatible as it only extends the existing stereo field and no new signal is added. This parameter should only be used to control the existing stereo field.

Widening converts the audio into its mid (mono) and side channels, leaving the mid intact and applying a gain to the side channel, then converts the signal back to left and right channels. As a result the stereo image becomes wider (for widening above 0%) or narrower (for widening below 0%). This method of widening the stereo image may initially sound pleasing, however it can quickly become fatiguing on the ear and often sounds unnatural, especially for larger amounts of widening. Use this parameter to control the existing stereo field and as a special effect. Use it to increase width only with caution.

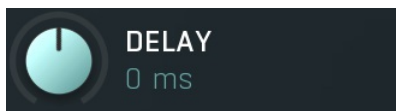
Range: Mono to 200.0%, default 0.00%



### Gain right

Gain right defines the power modification applied to the right channel of the output signal. You can use it to manually balance the stereo field if needed.

Range: -10.00 dB to +10.00 dB, default 0.00 dB

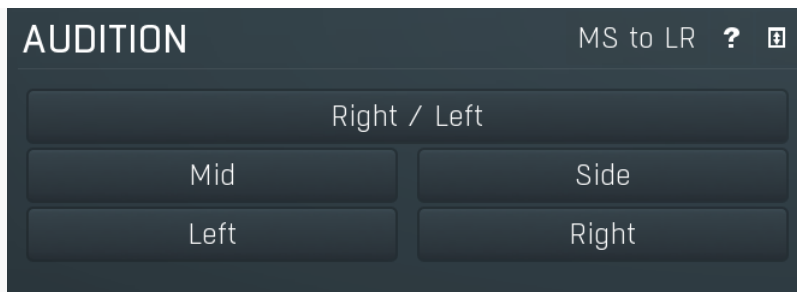


### Delay

Delay controls the delay applied on the left channel of the output signal. If it is negative, it is applied on the right channel instead. This way it in a way works as a panorama control. You can use it to manually balance the stereo field if needed. It is recommended to use this only if you have a creative reason to do that, or to adjust the delay between the channels if you feel that the automatic detection didn't produce ideal results.

Range: -25 ms to 25 ms, default 0 ms

## Audition panel



Audition panel contains multiple buttons, which let you audition specific part of the stereo signal. This can be used to understand the stereo field properly and check if the output signal is well balanced.

### MS to LR **MS to LR**

MS to LR lets you perform Mid/Side to Left/Right encoding. In certain recording techniques this is a necessity as one of the microphones is design as mid and the other is side, however it turns out that even with classic recording techniques it can be advantageous to use this feature. If you turn this on, you will usually notice wider stereo image, which is however perfectly mono compatible, because the right channel (now serving as side signal) will disappear in mono. Another positive aspect of this is that there won't be any phase cancellation when playing the signal as mono. Be careful though that this increased width may often sound unnatural and fatiguing, therefore it is advised to check repeatedly whether this feature generates an actual improvement.

### Right / Left

### **Right / Left**

Right / Left swaps samples between the left and right channels of the output signal, so that you listen right / left instead of left / right. You can use it to check whether the stereo is ballanced. If it is not, you will hear a serious unnatural change of the stereo field when swapping the channels. In an ideal case you should hear the difference, but it shouldn't sound "strange" and unpleasant.

### Mid

### **Mid**

Mid button lets you audition the mid signal only, that is the mono part of the output. You can use it to check for potential phase cancellations and to get your ears adjusted to the actual center.

### Side

### **Side**

Side button lets you audition the side signal only, that is the stereo part of the output. You can use it to understand what exactly is the stereo signal created from and if there are some specific problems you need to address only in the side signal.

### Left

### **Left**

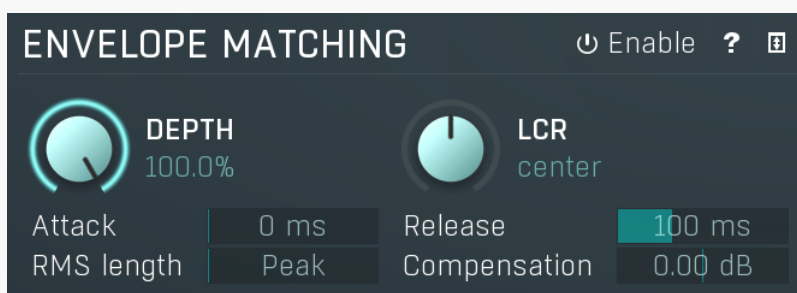
Left button lets you audition the left channel only. It is often useful to switch between Left and Right channels, to understand what is in each of them, since in most cases these have been recorded with different microphones. It is pretty common to hear a specific difference in one of them, such as a certain room resonance, longer instrument sustain, different spectrum response etc. Despite the automatic fix should correct many of these problems, it cannot perform miracles by for example changing the sustain in one channel. And these are the things that may make the signal impossible to perfectly align to the center.

### Right

### **Right**

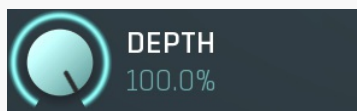
Right button lets you audition the right channel only. It is often useful to switch between Left and Right channels, to understand what is in each of them, since in most cases these have been recorded with different microphones. It is pretty common to hear a specific difference in one of them, such as a certain room resonance, longer instrument sustain, different spectrum response etc. Despite the automatic fix should correct many of these problems, it cannot perform miracles by for example changing the sustain in one channel. And these are the things that may make the signal impossible to perfectly align to the center.

## Envelope matching panel



Envelope matching panel lets you automatically match the level envelope in one channel to the other. In most cases one microphone is

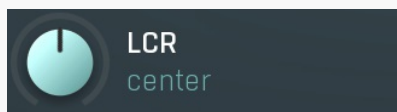
picking up more ambience than the other for example. This can cause disturbing differences in audio level between the left and right channels, especially with percussive signals. Envelope matching provides a simple and efficient way to fix that by letting the plugin follow level of one channel with the other.



### Depth

Depth controls the amount of envelope matching. 0% means no processing at all, 100% means full level matching.

Range: 0.00% to 100.0%, default 100.0%



### LCR

LCR controls which channels will be affected.

Range: 100% left to 100% right, default center



### Attack

Attack defines the attack time, that is how quickly the level detector increases the measured input level. When the input peak level is higher than the current level measured by the detector, the detector moves into the attack mode, in which the measured level is increased depending on the input signal. The higher the input signal, or the shorter the attack time, the faster the measured level rises. Once the measured level exceeds the **Threshold** then the dynamics processing (compression, limiting, gating) will start.

There must be a reasonable balance between attack and **release** times. If the attack is too long compared to the release, the detector will tend to keep the measured level low, because the release would cause that level to fall too quickly. In most cases you may expect the attack time to be shorter than the release time.

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

*In a **compressor** the attack time controls how quickly the measured level moves above the threshold and the processor begins compressing. As a result, a very short attack time will compress even the beginning transient of a snare drum for example, hence it would remove the punch. With a very long attack time the measured level may not even reach the threshold, so the compressor may not do anything.*

*In a **limiter** the attack becomes a very sensitive control, defining how much of the signal is limited and how much of it becomes saturated/clipped. If the attack time is very short, limiting starts very quickly and the limiter catches most peaks itself and reduces them, providing lower distortion, but can cause pumping. On the other hand, a higher attack setting (typically above 1ms) will let most peaks through the limiter to the subsequent in-built clipper or saturator, which causes more distortion of the initial transient, but less pumping.*

*In a **gate** the situation is similar to a compressor - the attack time controls how quickly the measured level can rise above the threshold at which point the gate opens. In this case you will usually need very low attack times, so that the gate reacts quickly enough. The inevitable distortion can then be avoided using look-ahead and hold parameters.*

*In a modulator, the detector is driving other parameters, a filter cut-off frequency for example, and the situation really depends on the target. If you want the detector to react quickly on the input level rising, use a shorter attack time; if you want it to follow the flow of the input signal slowly, use longer attack and release times.*

Range: 0 ms to 100 ms, default 0 ms



### Release

Release defines the release time, that is how quickly the level detector decreases the measured input level. The shorter the release time, the faster the response is. Once the attack stage has been completed, when the input peak level is lower than the current level measured by the detector, the detector moves into the release mode, in which the measured level is decreased depending on the input signal. The lower the input signal, or the shorter the release time, the faster the measured level drops. Once the measured level falls under the **Threshold** then the dynamics processing (compression, limiting, gating) will stop.

There must be a reasonable balance between **attack** and release times. If the attack is too long compared to release, the detector would tend to keep the level low, because release would cause the level to fall too quickly. Hence in most cases you may expect the attack time to be shorter than the release time.

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

*In a **compressor** the release time controls how quickly the measured level falls below the threshold and the compression stops. As a result a very short release time makes the compressor stop quickly, for example, leaving the sustain of a snare drum intact. On the other hand, a very long release keeps the compression working longer, hence it is useful to stabilize the levels.*

*In a **limiter** the release time keeps the measured level above the limiter threshold causing the gain reduction. Having a very long release time in this case doesn't make sense as the limiter would be working continuously and the effect would be more or less the same as simply decreasing the input gain manually. However too short a release time lets the limiter stop too quickly, which usually causes distortion as the peaks through the limiter to the subsequent in-built clipper or saturator. Hence release time is used to avoid distortion at the expense of decreasing the output level.*

*In a **gate** the situation is similar to a compressor - the release time controls how quickly the measured level can fall below the threshold at which point the gate closes. Having a longer release time in a gate is a perfectly acceptable option. The release time will basically control how much of the sound's sustain will pass.*

In a modulator, the detector is driving other parameters, a filter cut-off frequency for example, and the situation really depends on the target. If you want the detector to react quickly on the input level falling, use a shorter release time; if you want it to follow the flow of the input signal slowly, use longer attack and release times.

Range: 0 ms to 10000 ms, default 100 ms

## RMS length Peak RMS length

RMS length smoothes out the values of the input levels (not the input itself), such that the level detector receives the pre-processed signal without so many fluctuations. When set to its minimum value the detector becomes a so-called "peak detector", otherwise it is an "RMS detector".

When you look at a typical waveform in any editor, you can see that the signal is constantly changing and contains various transient bursts and separate peaks. This is especially noticeable with rhythmical signals, such as drums. Trying to imagine how a typical attack/release detector works with such a wild signal may be complex, at least. RMS essentially takes the surrounding samples and averages them. The result is a much smoother signal with fewer individual peaks and short noise bursts.

RMS length controls how many samples are taken to calculate the average. It stabilizes the levels, but it also causes a slower response time. As such it is great for mastering, when you want to lower the dynamic range in a very subtle way without any instabilities. However, it is not really desirable for processing drums, for example, where the transient bursts may actually be individual drum hits, hence it is usually recommended to use peak detectors for percussive instruments.

Note that the RMS detector has 2 modes - a simplified approximation is used by default, and a true RMS processor can be enabled from the advanced settings (if provided). Both respond differently, neither of them is better than the other, they are simply different.

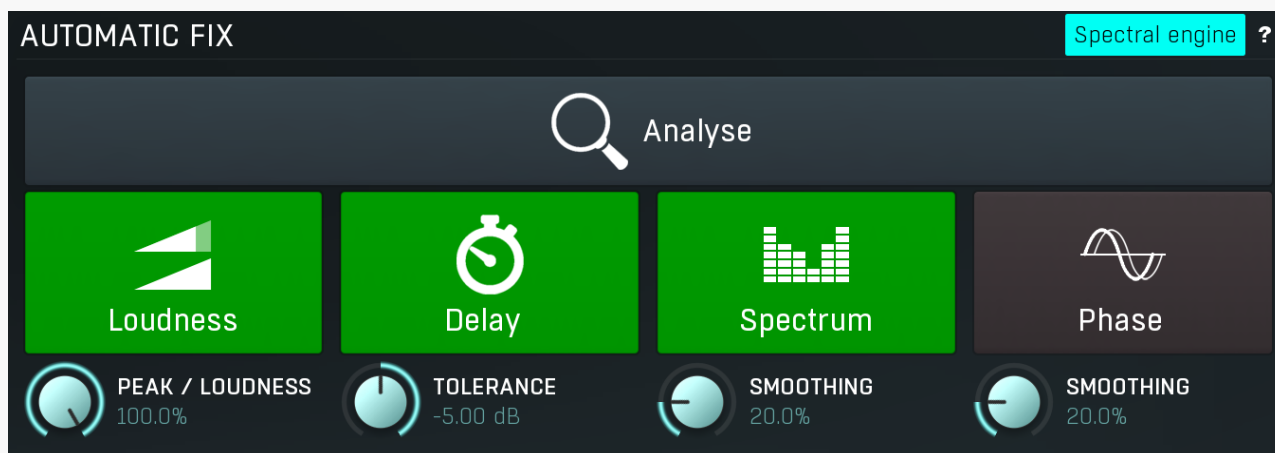
Range: Peak to 1000 ms, default Peak

## Compensation Compensation

Compensation defines the power modification applied to the right channel of the output signal when **Depth** is 100%. You can use it to manually balance the stereo field if needed, since the envelope matching can change it.

Range: -10.00 dB to +10.00 dB, default 0.00 dB

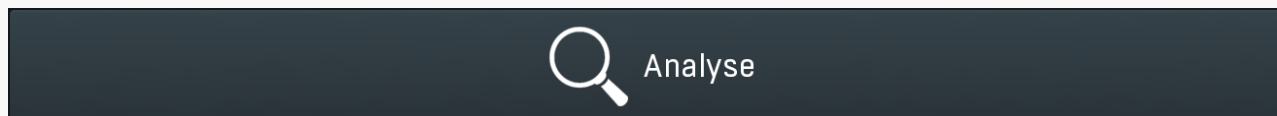
# Automatic fix panel



Automatic fix panel contains the core of the plugin - the automatic detection and fixing of various problems. First enable the playback and use the **Analyse** button to analyse your audio. Use the remaining buttons to enable or disable all each of the fixes that the plugin offers to improve the stereo field of your audio material.

## Spectral engine Spectral engine

Spectral engine switch enables the spectral engine; this is required for the **Spectrum fix** and **Phase fix**. Please note that this makes the plugin introduce latency.



### Analyse

Analyse button starts or stops the analysis. First start playback in your host so that the plugin receives your audio material. Then press this button and a percentage display will appear and will count from 0% to 100%, where 100% is the reasonable minimum required for the actual analysis algorithm. After reaching 100% the counter stops but it is advised to keep the analysis going. In theory the longer the analysis, the more accurate the results will be, but in most cases it is more than enough to analyse say 10 to 20 seconds of your material. The algorithm will actually let you proceed at any point after 20%, but the results may be much less accurate. Press the button again afterwards to finish the analysis. It is essential to choose a representative part of your audio material for analysis. Different sections of the recording may produce very different analysis results. We would advise choosing a medium part, where all instruments



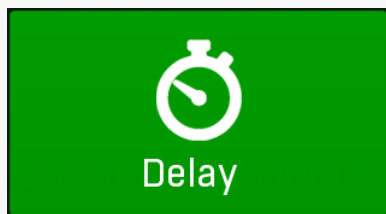
being recorded are audible, but there is no extreme signal.

For instance it is quite common when recording a drumset that one of the microphones is placed close to a ride cymbal. When the drummer starts playing this cymbal, the sound from it is too prominent and is effectively killing everything else. Analysing such a part of the song may misguide the plugin to think that this microphone produces very specific loud "cymbal" sound. The results could actually be working very well for that part of the song where the cymbal is indeed playing, but the rest of the song would then sound bad, as the same fix would be applied through the song.



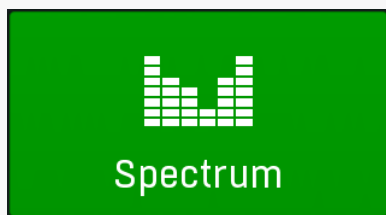
### Loudness

Loudness button activates the loudness fix. It ensures that the loudness of the left channel is the same as loudness of the right channel, which is an essential requirement for a good stereo image. Please note that the output peak meters may not show the same values for both channels, which is entirely correct, since the peak meters don't say too much about the actual perceived loudness. This feature only performs a gain on the right channel, so it is technically fully transparent.



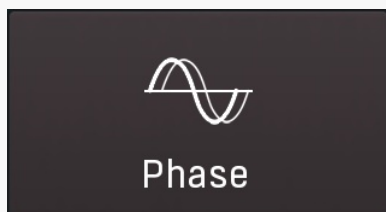
### Delay

Delay button activates the delay and polarity fix. It can delay the left or right channels and invert polarity of one of them if needed, to ensure that the correlation of both channels is as high as possible. In less scientific words, it fixes any potential time error between the channels. This feature only performs a delay and potential polarity inversion on left/right channels, so it is technically fully transparent. Time error occurs in most cases since the microphones are rarely exactly the same distance from the instruments being recorded. That is rarely even possible anyway due to the physical nature of the instruments and their surroundings. As a result the prominent signal (the loudest instrument usually) arrives at each microphone at a slightly different time and that shifts the stereo field (As perceived by your ears and brain) to the left or to the right. The plugin fixes that by shifting the channels in time, so that the prominent instrument sounds at the same moment in both channels and that makes the stereo field more centered.



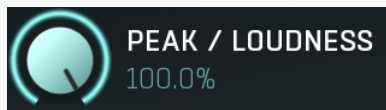
### Spectrum

Spectrum button activates the spectrum fix. It takes the spectrum analysis of the left and right channels and applies any necessary linear-phase equalization on the right signal to match the left channel. This is necessary, since in most cases the spectrum recorded by one microphone is different from the other. It is caused by differences between the microphones, recording equipment, position and surroundings etc. Please note that since this feature is actually changing the spectrum, it is not fully transparent. It also introduces latency.



### Phase

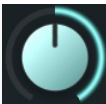
Phase button activates the phase fix. It takes the spectrum analysis of the left and right channels and applies the ideal spectral phase compensation to minimize phase cancellations between them. This is advantageous not only for mono compatibility, but the human brain also picks up phase differences of individual frequencies in the signal and uses them as clues for 3D positioning. Unfortunately since various elements from recording equipment to microphone position affect the phase, the resulting clues are usually very confusing for the brain and this feature can help minimizing them and make the stereo field tighter and centered. Please note that since this feature is actually changing the spectrum, it is not fully transparent. It also introduces latency. Please also note that this feature may sometimes have severe side-effects, especially when it comes to transients. Use the **Phase threshold** and **Phase smoothing** parameters to minimize that if needed.



### Peak / Loudness

Peak / Loudness controls how much the analyzer uses peak level (maximum when the control is at 0%) and how much it uses loudness estimate (maximum when at 100%). In most cases the loudness measure is mainly relevant, but for some percussive instruments the peak levels sometimes differ too much between the channels, which causes a sort of a psychoacoustic confusion. In these cases it may be worth trying to lower this value before analysis. Please note that this parameter is relevant only during analysis.

Range: 0.00% to 100.0%, default 100.0%



## TOLERANCE

-5.00 dB

### Delay tolerance

Delay tolerance sets the limit for searching for minimum delay. There is rarely a one perfect delay time between the channels. Instead there are several possible delays, which all reasonably avoid phase cancellation, but while this scientifically seems good, it may not be so psychoacoustically. The analysis may tend to select delays that are too long, because they are technically better, but they are in fact worse in practice, since they may make the output not rhythmical.

When analysing the audio material each possible delay is investigated and marked by a quality factor. The delay with the highest quality is then selected. The delay tolerance then specifies how much lower the quality may be compared with the scientifically best result, so the plugin may try to search for a shorter delay, which could then improve the psychoacoustic results while still providing good scientific quality. Please note that this parameter is relevant only during analysis.

Range: -10.00 dB to 0.00 dB, default -5.00 dB



## SMOOTHING

20.0%

### Spectrum smoothing

Spectrum smoothing lets you smooth the spectrum compensation that is applied on the right channel signal. It could be helpful if the automatic spectrum fix is too brutal, causing various artefacts such as transient smearing for percussive signals. However, please note that by using this parameter you also get further away from the theoretically ideal spectrum processing, so it technically lowers the effect of the compensation itself.

Range: 0.00% to 100.0%, default 20.0%



## SMOOTHING

20.0%

### Phase smoothing

Phase smoothing lets you smooth the phase envelope used by the phase fix. It could be helpful if the processing is too brutal, causing various artefacts such as transient smearing for percussive signals. However, please note that by using this parameter you also get further away from the theoretically ideal phase processing, so it technically lowers the effect of the compensation itself.

Range: 0.00% to 100.0%, default 20.0%

## Advanced panel

### ADVANCED

Maximum delay

Spectrum limit

Phase threshold

Allow polarity invert

Advanced panel contains advanced parameters, which you may need to use only in extreme cases.

Maximum delay

### Maximum delay

Maximum delay defines maximum time offset to be detected for the delay fix. You may decrease this value if you feel the plugin is actually introducing delay. And conversely you can increase it if there's a bigger delay than the plugin can detect. Please note that this parameter is relevant only during analysis.

Range: 0 ms to 1000 ms, default 100 ms

Spectrum limit

### Spectrum limit

Spectrum limit lets you limit the gain performed for each frequency by the spectrum fix. It is an alternative to **Spectrum smoothing**, which also reduces the effect and can therefore be used if the spectrum fix is too brutal.

Range: 0.00 dB to +40.00 dB, default +20.00 dB

Phase threshold

### Phase threshold

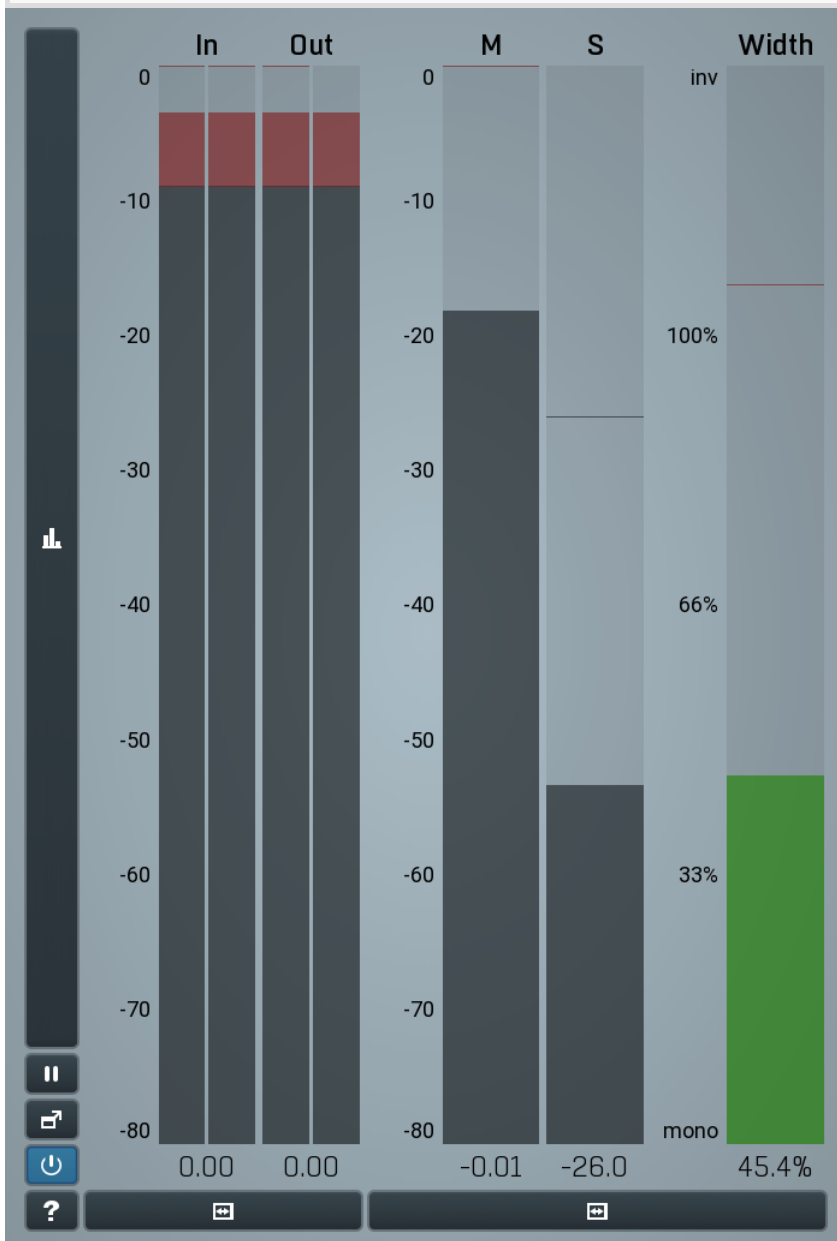
Phase threshold defines the minimum level to be considered relevant for the phase fix. In most signals all frequencies are actually present, but most of them are only noise, distortion and other artefacts. Shifting such signals wouldn't be a problem on its own, but the more the phase is changed, the bigger the chance that something will go wrong. In percussive signals this could lead to smearing of the initial transient for example. This parameter could be used to let the plugin consider only the prominent frequencies relevant for the analysis. All frequencies will then be processed anyway, but this can make the resulting phase envelope much less complex, which in effect improves the phase coherency and minimizes the chance that something goes wrong.

Range: silence to 0.00 dB, default -10.0 dB

Allow polarity invert

### Allow polarity invert

Allow polarity invert switch controls whether the polarity is allowed to be inverted by the delay fix if needed. You can try turning it off if you feel the plugin has performed polarity inversion and made things actually worse.



### Global meter view

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

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

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

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

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

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

**Width meter** shows the stereo width at the output stage. This meter requires at least 2 channels and therefore does not work in mono mode. Stereo width meter basically shows the difference between the mid and side channels.

When the value is **0%**, the output is monophonic. From 0% to 66% there is a green range, where most audio materials should remain.

**From 66% to 100%** the audio is very stereophonic and the phase coherence may start causing problems. This range is colored blue. You may still want to use this range for wide materials, such as background pads. It is pretty common for mastered tracks to lie on the edge of green and blue zones.

**Above 100%** the side signal exceeds the mid signal, therefore it is too monophonic or the signal is out of phase. This is marked using red color. In this case you should consider rotating the phase of the left or right channels or lowering the side signal, otherwise the audio will be

highly mono-incompatible and can cause fatigue even when played back in stereo.

For most audio sources the width is fluctuating quickly, so the meter shows a 400ms average. It also shows the temporary maximum above it as a single coloured bar.

If you right click on the meter, you can enable/disable loudness pre-filtering, which uses EBU standard filters to simulate human perception. This may be useful to get a more realistic idea about stereo width. However, since humans perceive the bass spectrum as lower than the treble, this may hide phase problems in that bass spectrum.

**M & S meters** display mid and side levels. These meters require at least 2 channels, so they do not work in mono mode.



### Time graph

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



### Pause

Pause button pauses the processing.



### Popup

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



### Enable

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



### Collapse

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



### Collapse

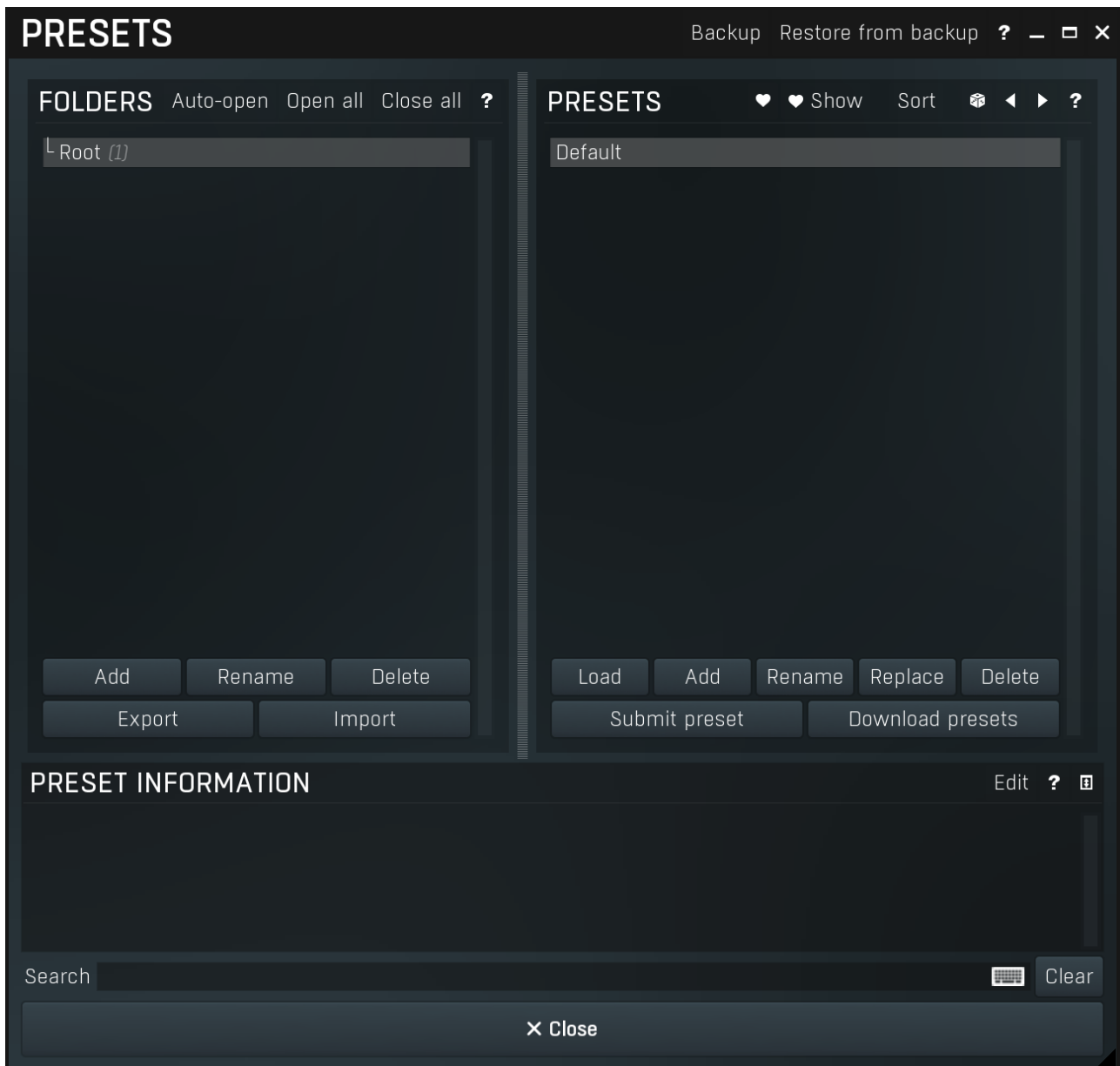
Collapse button minimizes or enlarges the panel to release space for other editors.



### Collapse

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

# Preset selector



Preset management window provides management for your presets.

**Backup**

## **Backup**

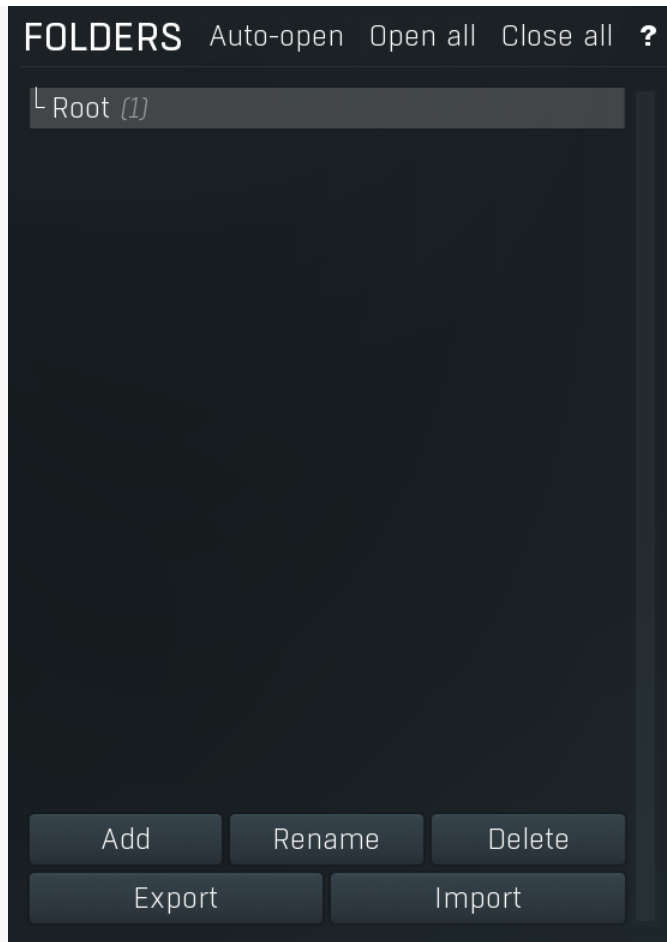
Backup button lets you backup presets for all MeldaProduction software into a single file, so you can transfer it to a different machine and restore the presets there for example.

**Restore from backup**

## **Restore from backup**

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**

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**

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**

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

#### Add **Add**

Add button creates a new folder in the tree

#### Rename **Rename**

Rename button lets you rename the selected folder.

#### Delete **Delete**

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

#### Export **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**

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**

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

## Sort **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**

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**

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

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

Add button creates a new preset using the current settings.

Rename

### Rename

Rename button lets you rename the selected preset.

Replace

### Replace

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

Delete

### Delete

Delete button deletes the selected preset.

Search



Search


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

Clear

### Clear

Clear button deletes all text in the search field.

PRESET INFORMATION

Edit ? 

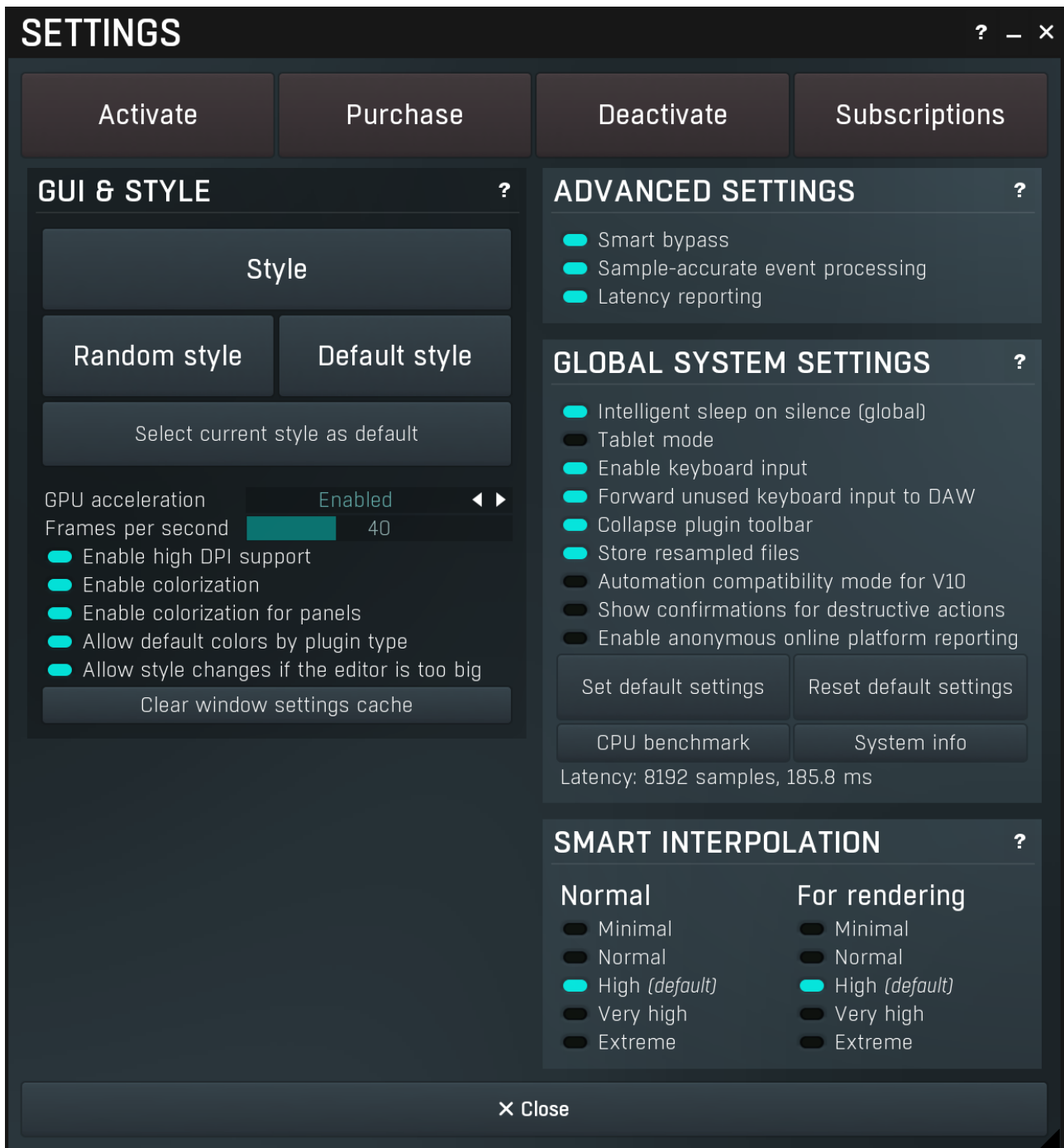
Preset

### 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

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

Purchase

### Purchase

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

Deactivate

### 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

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

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

## Random style

### Random style

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

## Default style

### 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

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

Enabled



## 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

40

## 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 support

### 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

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

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

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

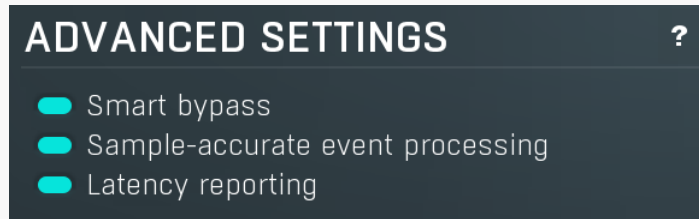
Allow style changes if the editor is too big is on by default and makes the plugin change its style, editor mode and other settings if it finds out it is too big to fit the current screen resolution.

## Clear window settings cache

### Clear window settings cache

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.

# Advanced settings panel



ADVANCED SETTINGS ?

- Smart bypass
- Sample-accurate event processing
- Latency reporting

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

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

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.

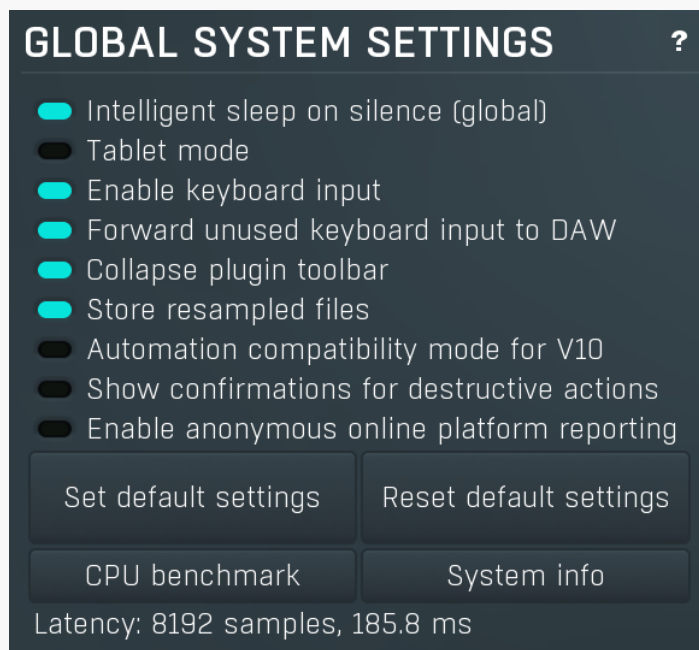
For example, if the block size in your host's audio settings is 1024 samples, this means the plugin is probably processing blocks of 1024 samples, in 44100 Hz sampling rate it is about 23ms. If this setting is disabled, any change in automation, MIDI, modulation etc. may then be granularized to 23ms (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

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 ?

- Intelligent sleep on silence (global)
- Tablet mode
- Enable keyboard input
- Forward unused keyboard input to DAW
- Collapse plugin toolbar
- Store resampled files
- Automation compatibility mode for V10
- Show confirmations for destructive actions
- Enable anonymous online platform reporting

Set default settings    Reset default settings

CPU benchmark    System info

Latency: 8192 samples, 185.8 ms

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)

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

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

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

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

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

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

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

Show confirmations for destructive actions

### Show confirmations for destructive actions

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

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

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

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

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

System info

### System info

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

## Smart interpolation panel

## SMART INTERPOLATION



### Normal

- Minimal
- Normal
- High (default)
- Very high
- Extreme

### For rendering

- Minimal
- Normal
- High (default)
- Very high
- Extreme

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.

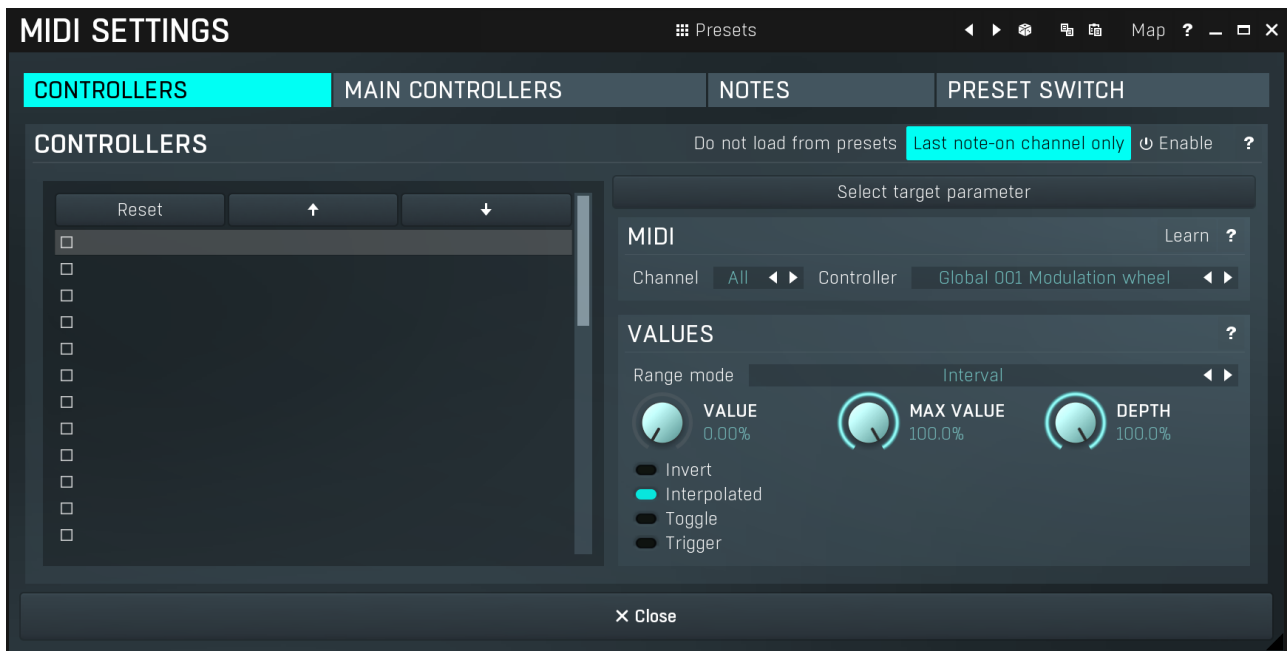
With modulators the situation is more complicated. Besides the updating issue, the modulator itself can perform some pretty advanced processing, hence it is better to perform the processing in blocks. However, the bigger the block, the less often the modulator updates those parameters associated with 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

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).

CONTROLLERS

MAIN CONTROLLERS

NOTES

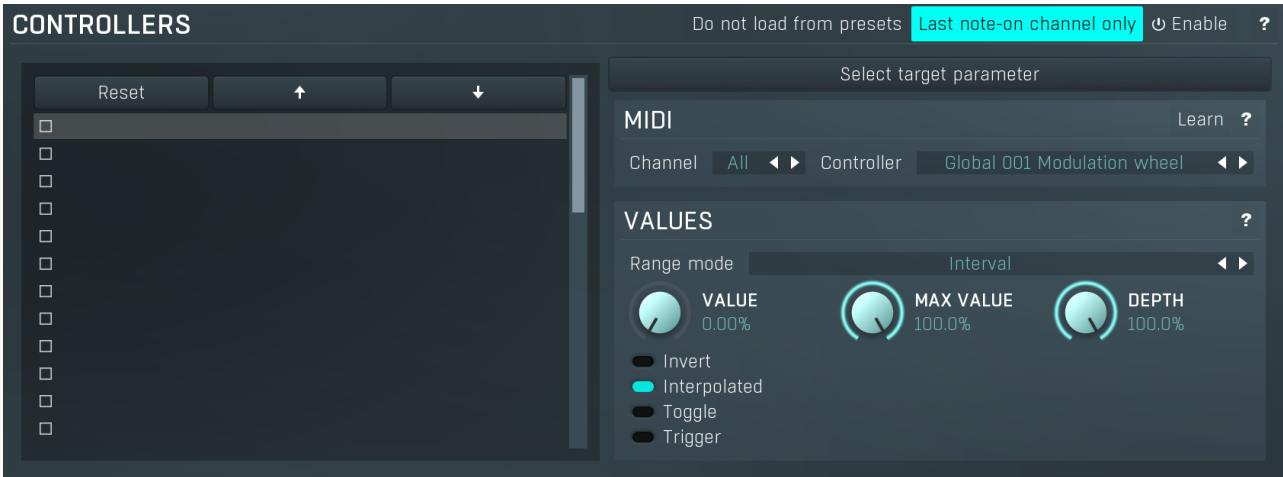
PRESET SWITCH

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

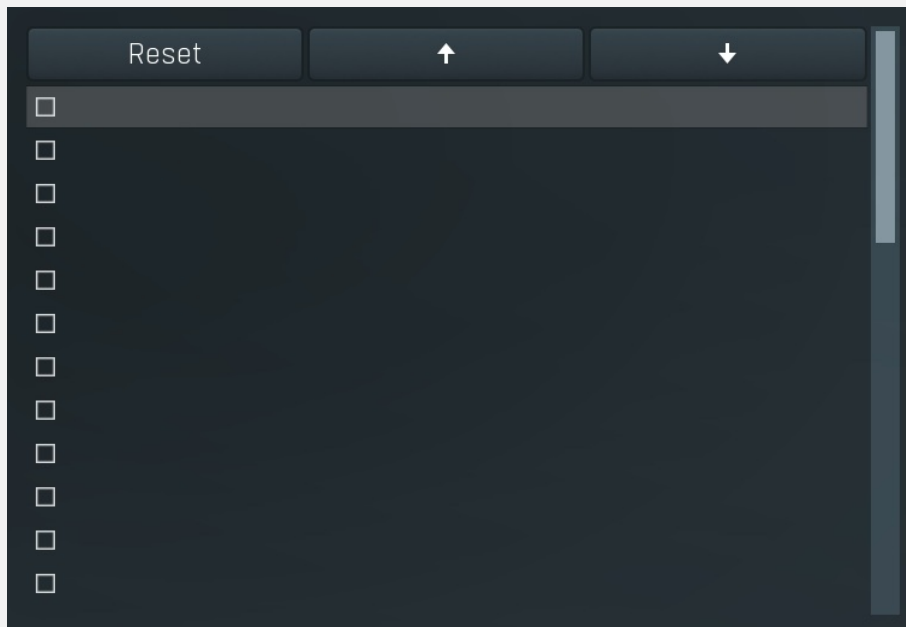
Do not load from presets button disables loading the controllers from presets. This may be handy if you have configured specific MIDI controllers with target parameters and you want to browse the presets without the need to configure them every time. Please note that some presets may rely on specific 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

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

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



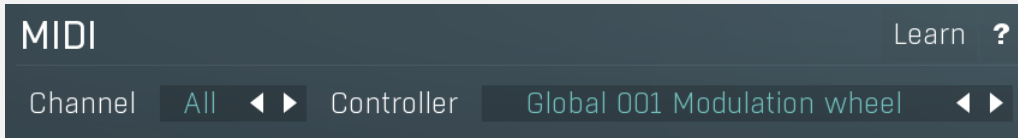
please note that if you have some multiparameter, modulator or another subsystem access the ranges of individual controllers, this function will reorder them, so these connections will no longer be correct.

Select target parameter

ParameterIndex

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

## MIDI



Learn

### 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 All Channel

Channel defines the controller MIDI channel.

Controller Global 001 Modulation wheel Controller

Controller defines the source controller.

## Values



Range mode Interval 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  
0.00%

### Value

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

MAX VALUE  
100.0%

### 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  
100.0%

### 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

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

 Interpolated

### 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

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

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

Slot	Name	Learn	Value	Learn
1	Global 001 Modulation wheel	Learn	9	Learn
2	Global 002 Breath controller	Learn	10	Learn
3	Global 011 Expression	Learn	11	Learn
4	Off	Learn	12	Learn
5	Off	Learn	13	Learn
6	Off	Learn	14	Learn
7	Off	Learn	15	Learn
8	Off	Learn	16	Learn

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

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.

Global 001 Modulation wheel

## ◀▶ Controller

Controller defines the MIDI controller associated to this Main controller.

## Learn 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

The screenshot shows the 'NOTES' panel with an 'Enable' button and a help icon. It features a list of checkboxes on the left and a control area on the right. The control area includes a 'Select target parameter' dropdown, a 'MIDI' section with 'Channel' (All), 'Note' (48 [C3]), 'Note min' (0 [C-1]), and 'Note max' (127 [G9]), and a 'VALUES' section with 'Mode' (Logarithmic), 'SHIFT' (0), 'MIN VALUE' (0.00%), and 'MAX VALUE' (100.0%).

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

# NoteControllers

The screenshot shows the 'NoteControllers' panel with a 'Reset' button and up/down arrow buttons. It features a list of checkboxes on the left.

Reset

### 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.

Select target parameter

### 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

MIDI Learn ?

Channel All ◀ ▶ Note 48 [C3]

Note min 0 [C-1] Note max 127 [G9]

Functional outside note range

Channel

All



### Channel

Channel defines the controller MIDI channel.

Note

48 [C3]

### 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

0 [C-1]

### Note min

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

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

Note max

127 [G9]

### 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

VALUES ?

Mode Logarithmic ◀ ▶

SHIFT 0 MIN VALUE 0.00% MAX VALUE 100.0%

Mode

Logarithmic



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.

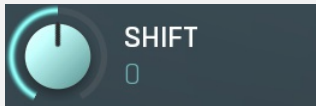
**Linear** converts the notes into frequencies and then transform them into the linear scale from 20Hz to 20kHz.

**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

Enable MIDI program change enables processing program change MIDI message.

## Preset previous/next trigger panel

PRESET PREVIOUS/NEXT TRIGGER

Learn  Enable ?Channel  
ControllerAll  
Global 000 Bank select

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**

Learn enables or disables MIDI learn.

Channel

All

**Channel**

Channel defines the controller MIDI channel.

Controller

Global 000 Bank select

**Controller**

Controller defines the source controller.

## Simulate program change via controller panel

## SIMULATE PROGRAM CHANGE VIA CONTROLLER

Learn  Enable ?

Channel	All	◀ ▶
Controller	Global 000 Bank select	◀ ▶
Number of values	128	

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

Learn enables or disables MIDI learn.

Channel	All	◀ ▶
---------	-----	-----

### Channel

Channel defines the controller MIDI channel.

Controller	Global 000 Bank select	◀ ▶
------------	------------------------	-----

### Controller

Controller defines the source controller.

Number of values	128
------------------	-----

### 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 be 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

Enable ?

Folder	PROGRAMS	<input type="text"/>
--------	----------	----------------------

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	PROGRAMS	<input type="text"/>
--------	----------	----------------------

### 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.

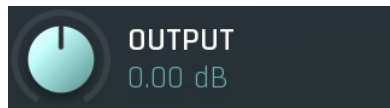
## Value button



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.

# 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](mailto: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](mailto: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](mailto: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 linked 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



**MELDA** production  
*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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