# **MCharmVerb**



## 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, selecting via the buttons or by using 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. The arrow buttons next to the preset button can be used to switch between presets easily.

Holding Ctrl while pressing the button loads a random preset. There must be some presets for this feature to work of course.

Presets can be backed up by 3 different methods:

- A) Using "Backup" and "Restore" buttons in each preset window, which produces a single archive of all presets on the computer.
- B) Using "Export/Import" buttons, which export a single folder of presets for one plugin.
- C) By saving the actual preset files, which are found in the following directories (not recommended):

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

Mac OS X: /Library/Application support/MeldaProduction

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

Please note that prior to version 16 a different format was used and the naming was "{pluginname}presets.xml". *The plugin also supports an online preset exchange. If the computer is connected to the internet, the plugin connects to our server once a week, submits your presets and downloads new ones if available. This feature is manually maintained in order to remove generally unusable presets, so it may take some time before any submitted presets become available. This feature relies on each user so we strongly advise that any submitted presets be named and organised in the same way as the factory presets, otherwise they will be removed.* 

# Left arrow

Left arrow button loads the previous preset.

# Right arrow

Right arrow button loads the next preset.



## Random

#### Randomize

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

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

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

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

# 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

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

**Licence manager** lets you activate/deactivate the plugins and manage subscriptions. While you can simply drag & drop a licence file onto the plugin, in some cases there may be a faster way. For instance, you can enter your user account name and password and the plugin will do all the activating for you.

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.

**Global system settings** contains some settings for all MeldaProduction plugins. Once you change any of them, restart your DAW if needed, and it will affect all MeldaProduction plugins.

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

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

## ♠ www

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

### Sleeping

## Sleep indicator

Sleep indicator informs whether the plugin is currently active or in sleep mode. The plugin can automatically switch itself off to save CPU, when there is no input signal and the plugin knows it cannot produce any signal on its own and it generally makes sense. You can disable this in Settings / **Intelligent sleep on silence** both for individual instances and globally for all plugins on the system.

# **Globals panel**



Globals panel contains basic audio processing features.



### **Dry/Wet**

Dry/Wet defines the ratio between the dry signal and the produced reverberation signal. If you use the reverb as a send effect, keep this at 100% and put the reverb on a dedicated bus in your host to which you send all the tracks to be processed. This way you control the amount of reverberation by adjusting the levels of the tracks that you send to the bus. If you use the reverb as an insert, use this parameter to control the amount of ambience versus the direct signal. Usually the higher the dry/wet value is, the more distant the audio seems.

#### Range: 0.00% to 100.0%, default 25.0%



#### Length

Length controls the decay length, an approximate time until the reverb decays to -60dB. Note that the actual decay time may vary with several other settings.

Range: 100 ms to 60000 ms, default 2000 ms



#### Size

Size controls the virtual room size. Principially this controls lengths of the various delays used by the algorithm. Range: 0.00% to 100.0%, default 50.0%



#### Low-cut

Low-cut defines the low-cut filter frequency used to remove bass from the reverberation signal. Range: Off to 2000 Hz, default Off



#### High-cut

High-cut defines the high-cut filter frequency used to remove treble from the reverberation signal. Range: 1000 Hz to Off, default Off

## Predelay 0 ms Predelay

Predelay defines the initial delay before the actual response, which simulates the space between the sound source and the listener. The longer the predelay is, the further away the source seems. At some point (around say 100ms) the brain stops understanding that that the reverberation belongs to the dry signal and starts interpreting them separately, and then the dry signal becomes close to the listener again. Predelay can therefore be used to control the distance from the source to the destination, but detaching the 2 signals can also be useful for instance to fill up a mix that isn't full enough without smearing the signal with the reverb itself. Range: 0 ms to 1000 ms, default 0 ms

## Gain 0.00 dB Gain

Gain defines the gain performed on the reverberation signal. You can use it to match the input level, so that **Dry/Wet** becomes easy to use and won't fool your ears by changing the output loudness. Range: -24.00 dB to +24.00 dB, default 0.00 dB

#### Widenina 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.

Please note that this algorithm is applied to the reverberation signal only. This is one of the ways to push the audio further away (lower values) or closer (higher values). The advantage is that the original signal's width is kept intact, yet the brain is often able to understand the distance clues.

Range: Mono to 200.0%, default 0.00%

# **Damp Low panel**



Damp Low panel contains parameters of the low frequency dampening filter. Its purpose is to simulate absorption by air and walls.



# FREQUENCY

### Frequency

Frequency controls the filter frequency. The higher it is, the more of the bass content is removed and less muffled the sound will be. Range: 20.00 Hz to 20.0 kHz, default 200.0 Hz

#### Gain Gain

0.40

Gain controls the filter gain, hence amount of the dampening. The lower it is, the more of the bass content is removed and less muffled the sound will be.

Range: -20.00 dB to 0.00 dB, default -3.00 dB

0 Q controls the filter Q. The lower it is the smoother the effect will be, but it will also be less focused the low frequencies desired to attenuate.

Range: 0.05 to 0.71, default 0.40

# **Damp High panel**



Damp High panel contains parameters of the high frequency dampening filter. Its purpose is to simulate absorption by air and walls.



### Frequency

Frequency controls the filter frequency. The lower it is, the darker the sound will be. Range: 20.00 Hz to 20.0 kHz, default 8000 Hz

### Gain -3.00 dl

## Gain

Q

Gain controls the filter gain, hence amount of the dampening. The lower it is, the darker the sound will be. Range: -20.00 dB to 0.00 dB, default -3.00 dB

#### 0.40

Q controls the filter Q. The lower it is the smoother the effect will be, but it will also be less focused the high frequencies desired to attenuate.

Range: 0.05 to 0.71, default 0.40

# **Designer panel**



Designer panel controls the advanced parameters of the reverberation algorithm. Presets and randomization are the safest bet if you want to play it safe.

### 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.

# Random Random

Random button generates random settings using the existing presets.



### Complexity

Complexity controls the structure of reverb. The higher the value is, the more complex and likely more natural the output will be, however the CPU requirements will increase as well. Range: 1 to 64, default 8



### Depth

Depth controls the depth of the input modulation useful to further improve the realisticity of the produced reverberation. If you enable it, the input signal is modulated first, which often provides a pleasing evolving character at the expense of additional CPU requirements. It can sound very artificial when overused.



#### Seed

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Seed controls the pseudorandom value generator for the internal reveration algorithm. When the seed is the same, the reverb always produces the same results. By clicking the button you can create a completely new sequence, hence it is an easy way to randomly change the results without messing with complicated parameters the reverb is based on. There are billions of combinations the generator can produce.

Range: 0 to 2147483647, default 12345678

#### Delay min

Delay min controls the minimum for the delay length used in the reverb. This way you can make sparser or denser reflections and control the algorithm in general.

Range: 0.00% to 100.0%, default 5.0%

#### Delay max

Delay min

#### Delay max

Delay max controls the maximum for the delay length used in the reverb. This way you can make sparser or denser reflections and control the algorithm in general.

Range: 0.00% to 100.0%, default 50.0%

#### Delay focus -100.0% Delay focus

5.0%

Delay focus controls the preference for the delay lengths used in the reverb. Lower values tend to produce shorter delays, which in effect causes quicker increase in the reflection density over time. Conversely, higher values tend to produce longer delays, which then slow down the density. However the output can still become very dense, just in longer time. Range: -400.0% to 400.0%, default -100.0%

#### Delay width 20.0%

#### Delay width

Delay width controls the tendency to produce different delay lengths for different channels, hence to provide wider stereo spectrum as a result of different processing in each channel. It is extended to surround as well. Range: 0.00% to 100.0%, default 20.0%

Range. 0.00% to 100.0%, default 20.0%

## Delay order Random < > Delay order

0.2000 Hz

Delay order controls the order of the delay lengths. On its own it doesn't make much sense, but along with other order settings it lets you set lowest coefficients for smallest delays for example. Think about it this way: an algorithmic reverb is always based on some form of complex delay system. For starters you can think of several delays in series, one after the other. Now if the first delay is short, while the next delays are longer, the first echo will occur very quickly and then the density will get lower. Conversely if the first delay is long, the first echo will arrive sooner and the density will increase afterwards. If they are not sorted, anything is possible. In practice the situation is always far more complex, but the classic advice is try every option and see what suits you best. In most cases sorting provides tigher results, while no sorting results in more ambient results.

#### Modulation rate

#### **Modulation rate**

Modulation rate controls the speed of the input modulation. The faster it is, the more pronounced the effect is. It can sound very artificial when overused.

#### Range: 0.0100 Hz to 10.00 Hz, default 0.2000 Hz



#### 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 oversampling and other pre-processing). It is always recommended to keep the input level under 0dB. You may need to adjust the previous processing plugins, track levels or gain stages to ensure that it is achieved.

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

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

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

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

When the value is **0%**, the output is monophonic. From 0% to 66% there is a green range, where most audio materials should remain. **From 66% to 100%** the audio is very stereophonic and the phase coherence may start causing problems. This range is colored blue. You may still want to use this range for wide materials, such as background pads. It is pretty common for mastered tracks to lie on the edge of green and blue zones.

**Above 100%** the side signal exceeds the mid signal, therefore it is too monophonic or the signal is out of phase. This is marked using red color. In this case you should consider rotating the phase of the left or right channels or lowering the side signal, otherwise the audio will be highly mono-incompatible and can cause fatigue even when played back in stereo.

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

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



### Time graph

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

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Pause button pauses the processing.

# Popup

Pause

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

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

# Utilities



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



## Multiparameter

Multiparameter button displays settings of the multiparameter. The multiparameter value can be adjusted by dragging it or by pressing Shift and clicking it to enter a new value from the virtual keyboard or from your computer keyboard.

Click on the button using your left mouse button to open the **Multiparameter** window where all the details of the multiparameter can be set. Click on it using your right mouse button or click on the **menu button** to the right to display an additional menu with learning capabilities - as described below.

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#### Menu

Menu button shows the **smart learn** menu. You can also use the right mouse button anywhere on the multiparameter button.

**Learn** attaches any parameters, including ranges. Click this, then move any parameters through the ranges that you want and click the multiparameter button again to finish. While learning is active, "REC" is displayed on the multiparameter button and learning mode is ended by clicking the button again.

**Clear & Learn** clears any parameters currently in the list then attaches any parameters, including ranges. Click this, then move any parameters through the ranges that you want and click the multiparameter button again to finish. While learning is active, "REC" is displayed on the multiparameter button and learning mode is ended by clicking the button again.

Reset resets all multiparameter settings to defaults.

**Quick Learn** clears any parameters currently in the list, attaches one parameter, including its range and assigns its name to the multiparameter. Click this, then move one parameter through the range that you want.

Attach MIDI Controller opens the MIDI Settings window, selects a unused parameter and activates MIDI learn. Click this then move the MIDI controller that you want to assign.

**Reorder to** ... lets you change the order of the multiparameters. This can be useful when creating active-presets. Please note that this feature can cause problems when one multiparameter controls other multiparameters, as these associations will not be preserved and they will need to be rebuilt.

In learning mode the multiparameter does not operate but rather records your actions. You can still adjust every automatable parameter and use it normally. When you change a parameter, the plugin associates that parameter with the multiparameter and also records the range of values that you set.

For example, to associate a frequency slider and make a multiparameter control it from 100Hz to 1KHz, just enable the smart learn mode, click the slider then move it from 100Hz to 1KHz (you can also edit the range later in the Multiparameter window too). Then disable the learning mode by clicking on the button.



Lock button displays the settings of the global parameter lock. Click on it using your left mouse button to open the Global Parameter Lock window, listing all those parameters that are currently able to be locked.

Click on it using your right mouse button or use the **menu button** to display the menu with learning capabilities - **Learn** activates the learning mode, **Clear & Learn** deletes all currently-lockable parameters and then activates the learning mode. After that, every parameter you touch will be added to the lock. Learning mode is ended by clicking the button again. The On/Off button built into the Lock button enables or disables the active locks.

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