

# MXXX



## Easy screen vs. Edit screen

The plugin provides 2 user interfaces - an **easy screen** and an **edit screen**. Use the Edit button to switch between the two.

By default most plugins open on the **easy screen** (edit button released). This screen is a simplified view of the plugin which provides just a few controls. On the left hand side of the plugin you can see the list of available **devices / instruments** (previously called 'active presets'), that is, presets with controls. These controls are actually nothing more than multiparameters (single knobs that can control one or more of the plug-in's parameters and sometimes known as Macro controls in other plug-ins) and are described in more detail later. Each device may provide different controls and usually is intended for a specific purpose. The easy screen is designed for you to be able to perform common tasks, quickly and easily, without the need to use the advanced settings (that is, those available on the Edit screen).

In most cases the devices are highlighted using different text colors. In some cases the colors only mark different types of processing, but in most cases the general rule is that **black/white devices** are the essential ones designed for general use. **Green devices** are designed for a specific task or audio materials, e.g. de-essing or processing vocals in a compressor plugin. **Red devices** usually provide some very special processing or some extreme or creative settings. In a distortion plugin, for example, these may produce an extremely distorted output. **Blue devices** require an additional input, a side-chain or MIDI input usually. Without these additional inputs these **Blue** presets usually do not function as intended. Please check your host's documentation about routing side-chain and MIDI into an effect plugin.

To the right of the controls are the meters or time-graphs for the plugin; the standard plugin Toolbar may be to the right of these or at the bottom of the plugin.

By clicking the **Edit button** you can switch the plugin to **edit mode** (edit button pushed). This mode provides all the of the features that the plugin offers. You lose no settings by toggling between edit mode and the easy screen unless you actually change something. This way you can easily check what is "under the hood" for each device, or start with an device and then tweak the plugin settings further.

Devices are factory specified and cannot be modified directly by users, however you can still make your own and store them as normal presets. To do so, configure the plugin as desired, then define each multiparameter and specify its name in its settings. You can then switch to the easy screen and check the user interface that you have created. Once you are satisfied with it, save it as a normal preset while you are on the easy screen. Although your preset will not be displayed or selected in the list of available devices, the functionality will be exactly the same. For more information about multiparameters and devices please check the [online video tutorials](#).

# Edit mode



## Global 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.

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.

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

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

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

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

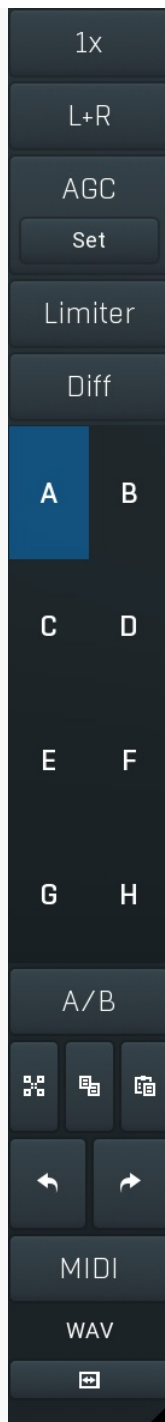
**Smart interpolation** adjusts the interpolation algorithm used when changing parameter values; the higher the setting the higher the audio quality and the lower the chance of zippering noise, but more CPU will be used.



## WWW

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

# Plugin toolbar



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

1x

## Upsampling

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

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

The plugin implements a high-quality upsampling algorithm, which essentially works like this: First the audio material is upsampled to a

higher sampling rate using a very complicated filter. It is then processed by the plugin. Further filtering is performed in order to remove any frequencies above the Nyquist rate to prevent aliasing from occurring, and then the audio gets downsampled to the original sampling rate.

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

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

L+R

## Channel mode

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

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

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

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

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

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

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

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

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

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

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

**Side (S) mode** is complementary to M mode, and allows processing of only the side (stereo) part of the signal leaving the mid intact. The same techniques as described for M mode can also be applied here, giving the opposite results.

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

A **compressor** can attenuate the side part in louder sections making it more monophonic and centered, placing the origin a little further away and in front of the listener.

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

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

A **saturator / exciter** may make the stereo richer and more appealing by creating higher harmonics without affecting the mid channel, which could otherwise become crowded.

**Modulation effects** can achieve the same results as in mid mode, but this will vary a lot depending on the effect and the audio material. It can be used in a wide variety of creative ways.

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

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

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

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

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

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

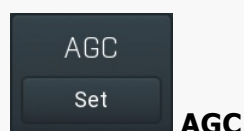
There are also 4 subsidiary modes: **Left & zero Right (L(R0))**, **Right & zero Left (R(L0))**, **Mid & zero Side (M(S0))** and **Side & zero Mid (S(M0))**. Each of these processes one channel and silences the other.

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

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

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

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



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

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

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

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

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

Set

## Set

Set button uses the AGC (automatic gain compensation) processor to calculate the ideal output gain to ensure that the output audio loudness is equal to the input level. To use it, simply enable playback in your host and click the button. The plugin's output gain will be adjusted to match the input and output levels as closely as possible.

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

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

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

Limiter

## 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.

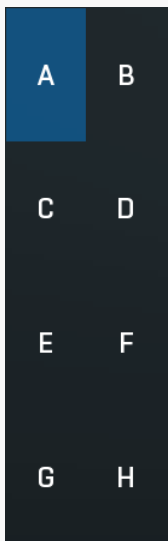
It is generally advised to keep your audio below 0dB at all times in all stages of your processing chain. However, several plugins may cause high level outputs with certain settings, often due to unprevented resonances with specific audio materials. The safety limiter prevents that.

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.

Diff

## Diff

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



## A-H presets selector

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

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

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

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

A/B

## 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

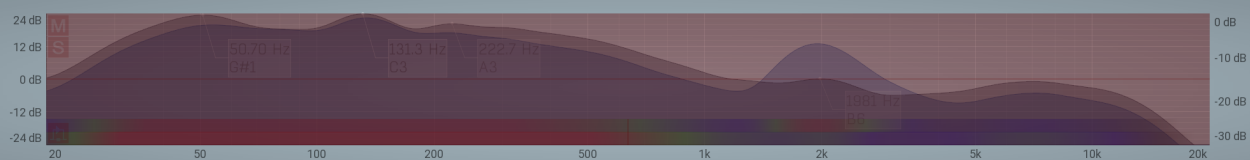
WAV button lets you process a file using the plugin with current settings. You can either click the button and select a file, or drag & drop the file (or multiple files) onto the button. If you let the plugin process WAV files, these will be saved with the original settings. If you use a different file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



Band

## editor

Band editor displays the available frequency bands, the crossover frequencies delimiting them, and the input gains and panoramic positions.

Use the left mouse button to drag the band boundaries (the vertical lines between bands), the band itself (the central dot in each band) and the input gains (the horizontal bars in each band). The short vertical bars in the bottom of each band control the input panoramic positions (when L+R Channel Mode is selected) or the input Widths (when M+S Channel Mode is selected).

Use the right mouse button to open the **Band Configuration** window where you can manage the bands and crossover filters and the appearance of the analyzer waveforms in the band editor.

Buttons to the left-hand side of each band let you mute, solo and bypass the processing in each band. Please note that the **Mute** and **Solo** buttons act on the output for each band, that is after the actual band processing.

The Collapse button to the right of the Band Editor minimises the editor, releasing space for other editors in the plug-in.

# Band menu

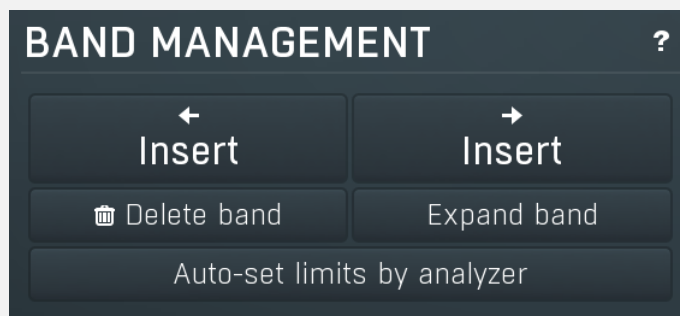


Band menu provides features to control the set of bands and copy & paste band settings (**Band management** section), reset band input gain & panorama (**Band gain & panorama** section), and to select and customize the crossover (**Crossover** section) and analyzer options.

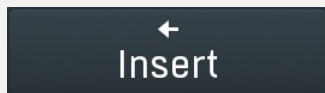
You can display this menu by **right-clicking** on the band editor.

One of the essential things to control in the band menu is the number of bands. The plugin can either operate as a single bundle plugin. In this case there is no crossover employed of any kind and the first and only band receives all MIDI data if the plugin makes use of it somehow. If there are 2 or more bands however, the plugin somehow produces signals for each band using the crossover, based on the spectrum or level for example, and there's a change in MIDI behaviour as well - 1st band receives only MIDI channel 1, 2nd receives only MIDI channel 2 etc.

## Band management panel

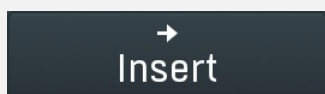


Band management panel contains basic features to create, delete and manipulate bands.



### Insert left

Insert left button inserts a new band to the left of the currently-selected band (the last one that you clicked on).



### Insert right

Insert right button inserts a new band to the right of the currently-selected band (the last one that you clicked on).



### Delete

Delete button deletes the currently-selected band (the last one that you clicked on).



### Expand band

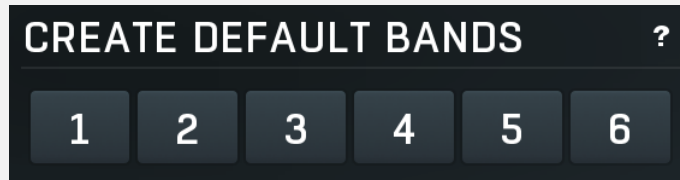
Expand band button solos (or unsolos) the band that you clicked on and disables the crossover temporarily, so that you can audition what the settings of this band would do to the entire signal, without any of the other bands having any affect.

Auto-set limits by analyzer

### Auto-set limits by analyzer

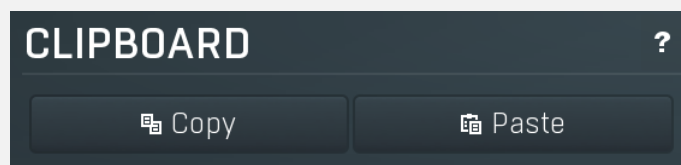
Auto-set limits by analyzer button adjusts the band limits using the current analyzer state, so that there's approximately the same signal level in each band. It is often useful to increase the averaging in the analyzer settings, so that the analysis doesn't 'jump' that quickly.

## Create default bands panel



Create default bands panel lets you easily create a predefined set of bands. This is the easiest way to say create default plugin settings with 4 bands.

## Clipboard panel



Clipboard panel contains features to transfer band settings via the system clipboard. Note that as always you can paste the settings as text into an email or forum post for example.

Copy

### Copy

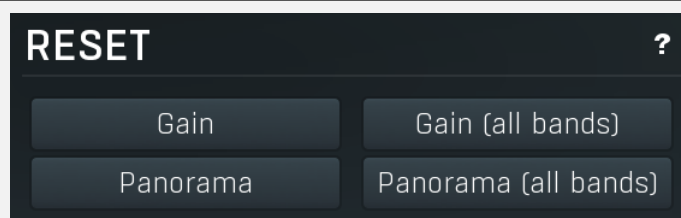
Copy button copies the band settings into the system clipboard. Note that the plugin band parameter settings are not copied; only the band limits, gains and panoramas.

Paste

### Paste

Paste button loads the band settings from the system clipboard. Note that the plugin band parameter settings are not pasted; only band limits, gains and panoramas.

## Reset panel



Reset panel lets you reset some band parameters.

Gain

### Gain

Gain button resets the input gain of the currently-selected band (the last one that you clicked on) to 0dB.

Gain (all bands)

### Gain (all bands)

Gain (all bands) button resets the input gain of all bands to 0dB.

**Panorama**

Panorama button resets the input panorama of the currently-selected band (the last one that you clicked on) to center.

**Panorama (all bands)**

Panorama (all bands) button resets the input panorama of all bands to center.

## Crossover panel

**CROSSOVER** ?

<b>Analog</b>	Analog LP
Linear-phase	Hybrid
Level	Panorama
Mid/Side	Disabled
Spectrum	Tonal/Transient
Slope	<div style="display: flex; justify-content: space-between; font-size: 0.8em;"> <span>6dB/oct A</span> <span>6dB/oct B</span> <span>12dB/oct</span> </div> <div style="display: flex; justify-content: space-between; font-size: 0.8em;"> <span style="background-color: #0070c0; color: white; padding: 2px;">24dB/oct</span> <span>48dB/oct</span> <span>72dB/oct</span> </div> <div style="display: flex; justify-content: space-between; font-size: 0.8em;"> <span>96dB/oct</span> <span>120dB/oct</span> </div>
Level	silence
Level slope	50.0%
Transient release	50 ms
Transient resolution	50 ms
Smoothing	0.00%
Tone	0.00 dB
Spectral resolution	50.0%

Crossover panel contains configuration of the crossover used to separate the signals for each band. **Crossover** is a technical term for an algorithm or device which splits a signal into multiple bands (or signals), which when mixed back together recreate the original signal (meaning that the crossover is transparent). The plugin provides several types of crossover with a flat (or nearly flat) response, which means that whichever crossover you choose and whatever signal you send into the plugin, the output levels of each frequency, after the bands are mixed back together to get the output signal, will be (almost) exactly the same, unless there is some processing applied in the bands themselves. Most of the available crossover types produce bands with different frequency ranges; however there are also a few more creative ones.

**Analog crossovers** have no latency, but they exhibit a phase-shift. That is usually irrelevant unless you are going to mix the output with the input later on. Analog crossovers are based on the classic analog components that you can find in speaker systems for example, however they are perfectly accurate and their slope (band separation) ranges from 6dB/octave to a very steep 120dB/octave. The higher the slope is, the more separated is each band (that is, there is less overlap between bands), but also the bigger is the phase shift. That can reach such an extent that some bassy materials become severely phasey, which may or may not be a good thing. An exception to the rule is the 6dB/oct crossover, which is zero-phase naturally. Its disadvantage is that the separation between bands is rather low, 6dB/oct is often not enough.

**Analog LP crossover** is a linear-phase equivalent to the **Analog** crossover. It introduces latency as does any linear-phase filter, but it does not cause a phase-shift. This may be especially advantageous for higher filter slopes, which, with classic analog crossovers, would cause severe transient smearing. Please note that the crossover type may not be 100% transparent, especially with small bands in bass spectrum and high slopes.

**Linear-phase crossover** is a fully digital crossover with a high slope (frequency-dependent), which introduces latency, but exhibits no phase-shift. This crossover mode is designed specifically for mastering.

**Hybrid crossover** is linear-phase as well, hence it introduces latency, but no phase-shift. However, its slope is more similar to the slopes of the analog crossovers.

**Level crossover** is a very specialized tool, which doesn't filter the input signal at all (hence it is not only linear-phase, but also zero-phase). Instead of filtering, it simply performs a gain on each band in such a way that when all the bands are mixed back together, the output is the original signal again. When you select this crossover, the spectrum analyzer graph disappears and the X axis in the band editor changes from frequencies to dB levels. So the band limits are not frequencies anymore, but rather sound levels.

The current level displayed in the graph area is controlled by the **Level** value below and you are likely to use a modulator, most likely in **Follower** mode, to control this latter value. The crossover then applies gain to each band depending on how much the current level fits into the band. The **Slope** parameter controls how quickly each band fades into the adjacent one. This crossover effectively turns the plugin into a very advanced dynamics processor; using a Follower Modulator the band used to process the input audio depends on the audio level.

*The are many possibilities for this crossover. But the basic principle is to select a spare Modulator, configure it as a Follower and select the Global parameter "Crossover Level value" as its target, with a "Full range" range mode. After configuring the Modulator, you will be able to see the detected value curve in the Modulator's Level graph. Then if the input signal is strongest, the right most band is processed etc. So if you for example use a delay with 2 bands and set the band limit high enough, the 2nd band will be processing only the loud parts of the signal and vice versa.*

**Panorama crossover** is another specialized tool, similar to the level crossover; it splits the signal into bands according to the panorama. If, for example, you create 3 evenly spaced bands (100%L to 33%L, 33%L to 33%R, 33%R to 100%R), then the leftmost band will contain mainly the signals located in the left speaker, the rightmost band will contain mainly signals from the right speaker and the middle band will contain centred signals. Please note that this doesn't mean the crossover attempts to analyze the space the signals are coming from and send them to the respective bands, which is probably what your brain would attempt.

This crossover is useful only when processing stereophonic (or surround, in which case the channels from 3 upwards are kept intact) signals and can be used for all kinds of mixing and creative processing. For example, using a multiband compressor with this crossover can be used to effectively control the stereo image as each band would be processing a different part of the stereo image. To mention another example, a multiband delay or reverb can be used to produce a different ambience for different parts of the stereo image.

**Mid/side crossover** is similar to panorama crossover, but it splits the signal according to their position in mid/side location. In other words, the more to the left a band, the more centred is the signal in it. Similarly the more to the right a band, the more "to the side" is the signal in it. You can think of it as the panorama view folded back on itself, around the center position. If, for example, you create 3 evenly spaced bands (centre to 33% L or R, 33% L or R to 67% L or R, 67% L or R to 100% L or R), then the leftmost band will contain the centred signal, the rightmost band will contain the signals to the extreme left or right and the middle band will contain signals in between. It can be used for similar tasks as the panorama crossover.

**Disabled crossover** is not a crossover actually, it simply disables the crossover and as a result each band processes the full input signal. As a consequence there is likely to be an increase in output level, so take care and turn down the output level first. In practice this "not really crossover" mode lets you process multiple streams of the input audio signal in parallel. For example, if you use a compressor, this in effect produces an extreme parallel compression. As another example, you can use a reverb to produce several rooms in parallel, potentially leading to a fuller space for example.

**Spectrum crossover** is the first of the spectral crossovers. It splits the signal into individual frequencies, analyzes their levels and sends the frequencies with the highest level into the highest band etc. It marks each frequency with its level (as you can see on the dB scale on the X axis in the crossover band editor) and puts it into the appropriate band. The crossover is linear-phase and fully transparent.

It provides a huge (not only) creative potential as it lets you process the dominant and weak parts of the signal individually. For instance, by compressing the dominant frequencies using MDynamicsMB you can bring more attention to the unsubstantial frequencies in the signal and in a way stabilize it without disrupting the silent parts of it. Note that this is NOT the same thing as using a normal compressor, because this way it treats only the loud frequencies even if the weak frequencies are present at the same time. Another example could be using MDelayMB to generate echoes only to the dominant parts of the signal, such as snare and bass drums in a drum loop.

**Transient crossover** is also a spectral crossover. It splits the signal into individual frequencies and sends the transient parts for each of them into the highest band etc. It marks each frequency with its "current transientness" (defined by the percentage scale that you can see on the X axis in the crossover band editor) and puts it into the appropriate band. The crossover is linear-phase and fully transparent.

It provides a huge (not only) creative potential as it lets you process split the signal into tonal and transient parts (and anything in between) and treat each individually. For instance, by compressing the transients using MDynamicsMB you can easily control the attack of drums. Note that this is NOT the same thing as using a normal compressor, because this way you can treat only the attacks in an already mixed signal without affecting the remaining part of the signal. Another example could be using MDelayMB to generate echoes only for the attacks of each drum.

	6dB/oct A	6dB/oct B	12dB/oct
Slope	24dB/oct	48dB/oct	72dB/oct
	96dB/oct	120dB/oct	

**Slope**

Slope defines the slope of each band transition and is used only by analog crossovers (including the linear-phase versions). It essentially controls the separation between the bands - the higher the slope, the lower the overlap between bands. Higher slopes require more CPU power and exhibit higher phase shift, which may be a problem especially when percussive materials. In these cases it may be necessary to switch to a linear-phase version.

Interesting exception to the classic rule are the 6dB/oct crossovers, which are linear-phase by nature (while still being zero latency), because the bands compensate for each other's phase shift. A side-effect of this is that the signal level in each band is

much higher than using other crossovers, so you may expect these crossovers sound considerably different to the other modes.

Level silence **Level value**

Level value is used only with **Level crossover** and controls the level at which the signal is split into each band. You will probably want to attach this parameter to a modulator in Follower mode for instance.

Level slope 50.0% **Level slope**

Level slope is used only with some crossover modes (Level, Spectrum and Tonal/Transient) and controls how quickly each band fades into the next one. It's similar to the **Slope** parameter used with analog crossovers.

Transient release 50 ms **Transient release**

Transient release is only used by the **Tonal/Transient** crossover and controls the release time of each transient. The transients detected by the crossover are naturally very short, so this provides a way to make them longer, hence send more signal to the higher bands of the crossover (receiving transients) and less to the lower bands (receiving the remaining part of the signal).

Transient resolution 50 ms **Transient resolution**

Transient resolution is only used by the **Tonal/Transient** crossover and controls the behaviour of the spectral transient detector. You can use it to adjust the crossover to your audio material and we would recommend a simple trial-and-error approach.

Smoothing 0.00% **Smoothing**

Smoothing is only used by spectral crossovers and controls how frequencies affect their surroundings. Without smoothing the individual bands may sound a bit artificial, because human brain general dislikes separated frequencies. It usually doesn't matter unless you audition the bands separately, but sometimes when more "brutal" processing is used on each band, it may become audible, which is where the smoothing can provide a solution at the cost of additional CPU and lower separation between bands, because it naturally makes the frequencies "more alike".

Tone 0.00 dB **Tone**

Tone is only used by spectral crossovers and controls the spectral slope applied by the detector. It is exactly the same feature as the **Slope** in analyzers and the crossover uses it to determine how to spread the frequencies between the bands. Higher slope gives more energy to higher frequencies and vice versa. Note that whatever the settings are, the crossover still produces signals that perfectly sum to the original input signal, meaning that it is perfectly transparent and unless the bands are actually doing something, you won't be able to hear a difference when changing this parameter.

Spectral resolution 50.0% **Spectral resolution**

Spectral resolution is only used by spectral crossovers and controls the spectral transformation settings. The higher the value is, the higher FFT size and overlap size is used, and therefore more CPU is usually required as well. Whether higher/lower value is good or not depends on the actual signal, the default 50% should work well with most audio materials. Higher values will generally provide better frequency resolution (usually good for less percussive sounds), lower values will provide better time resolution (usually good for more percussive sounds), eventually it is always about a compromise.

## Analyzer panel

## ANALYZER ?

Analyzer

Fill

Sonogram

Pause

Normalize

 Settings

Analyzer panel lets you configure the fully featured integrated analyzer and sonogram.

 Settings

### Settings

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

## Analyzer settings



 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.

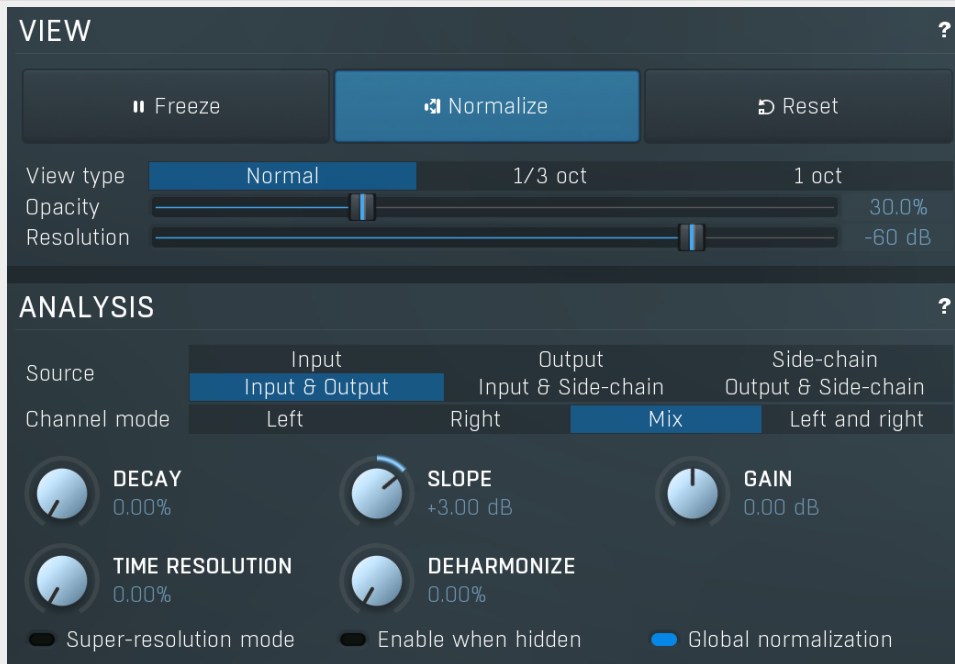
**MAIN SETTINGS** ADVANCED GRAPHS SONOGRAM PREFILTERING

### Tab selector

Tab selector switches between subsections.

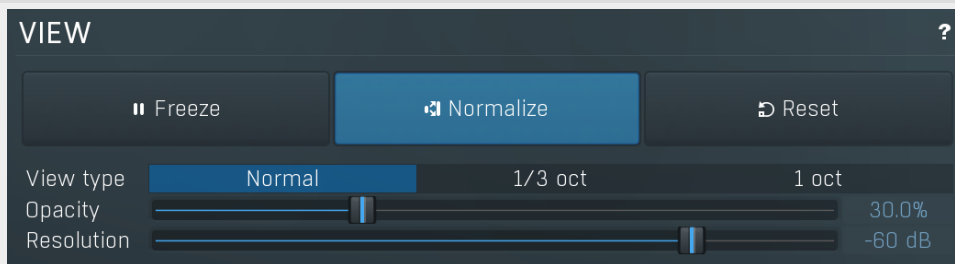


## Main settings panel



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

## View



Freeze

### Freeze

Freeze button stops processing temporarily.

Normalize

### Normalize

Normalize button enables or disables the visual normalization, which makes the loudest frequency be displayed at the top of the analyser area (0dB); it does not normalise the sound. This is very useful for comparing frequency levels, however it does hide the actual level.

When comparing 2 spectrums you are usually interested mainly in the frequency level differences. In most cases both audio materials will have different overall levels, which would mean that one of the graphs would be "lower" than the other, making the comparison quite difficult. Normalize fixes this and makes the most prominent frequencies of the spectrum reach the top of the analyzer area (or have the most highlighted color in case of sonogram).

Reset

### Reset

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

View type **View type**

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

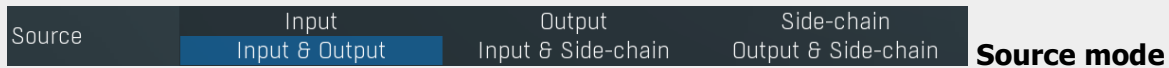
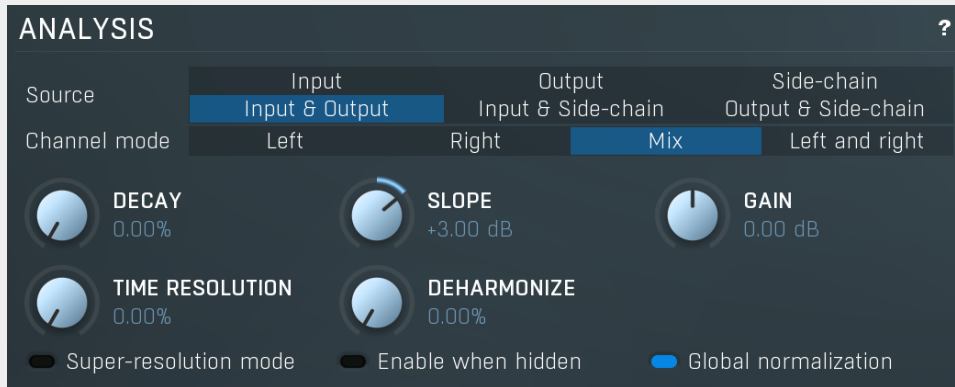
Opacity **Opacity**

Opacity controls the opacity of all analyzer graphs.

Resolution **Resolution**

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

## Analysis



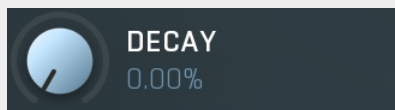
Source mode defines which audio stages are to be analyzed. By default both input & output are selected and analyzed. However you may want to analyze only the input, or the output (or the external side-chain, where available, on its own or with the input or output).



Channel mode defines which channels are to be analyzed. By default all channels are merged into a mono sum (Mix mode), which is then analyzed. However you may want to analyze separate channels or display both the left and right channels separately. Please note that if two channels (for example: input & output, or input & side-chain) are displayed at the same time then mix mode is used instead of left & right mode. Similarly, when the plug-in is in Surround mode then Mix mode is used.

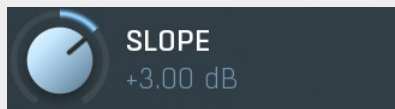
Also please note that when the plug-in is in one of the Mid / Side modes of operation, then you should read 'Left' as 'Mid' and 'Right' as 'Side'.

Different analyser combinations can, of course, be saved as different named presets.



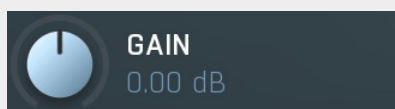
### Decay

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



### Slope

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



### Gain

Gain makes all frequencies change magnitude by the specified amount. This has no meaning when normalization is enabled.



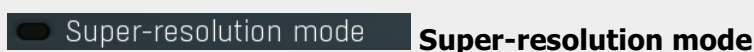
### Time resolution

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



### Deharmonize

Deharmonize tries to remove harmonics in the content and leave only fundamentals. This may help you find the dominant frequencies in the signal.



### Super-resolution mode

Super-resolution mode activates a special processing algorithm, which provides high resolution even in the low frequency spectrum. Using standard FFT algorithms you can increase the FFT size to get better bass resolution, but this also slows

down the response. Super-resolution mode keeps the quick response in high frequencies as they are naturally quicker, but also highly enhances the bass spectrum resolution. It requires additional CPU power.

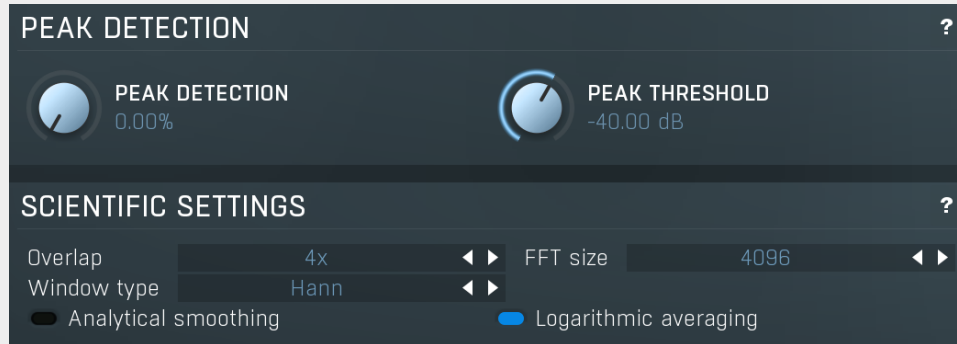
### Enable when hidden **Enable when hidden**

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

### Global normalization **Global normalization**

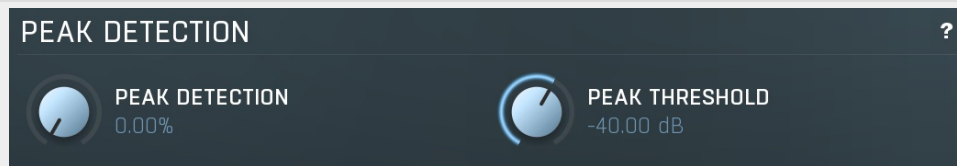
Global normalization makes the **normalization** work based on the maximum of all graphs visible at the time. This means that the levels between the graphs will stay the same, but the maximum level will be 0dB. This is useful for comparing relative levels. If you disable this, all graphs will be normalized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

## Advanced panel



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

## Peak detection



### Peak detection

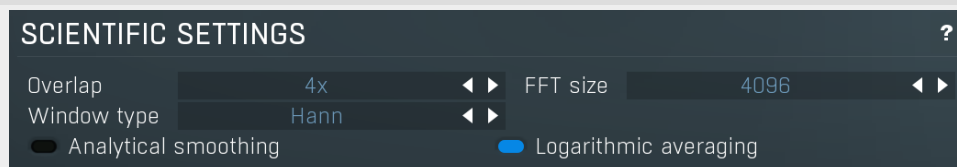
Peak detection tries to remove skirts of separate sinusoids letting you view the frequencies contained in your audio material. This may be handy when performing more scientific analyses.



### Peak threshold

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

## Scientific settings



### Overlap **Overlapping**

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

### FFT size **FFT size**

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

### Window type **Window type**

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

## Analytical smoothing

## Analytical smoothing

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

## Logarithmic averaging

## Logarithmic averaging

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

## Graphs panel

The screenshot shows the Graphs panel with the following settings:

Graph Type	Peaks	Micro-sonogram	Sonogram fill	Line opacity	Line width	Fill opacity	Color 1	Color 2
AVERAGE	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	100.0%	1	100.0%	Dark Blue	Blue
AVERAGE (INFINITE)	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	100.0%	1	100.0%	Dark Green	Green
MAXIMUM	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	100.0%	1	100.0%	Dark Red	Red
MAXIMUM (INFINITE)	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	100.0%	1	100.0%	Dark Orange	Orange
MAXIMUM - AVERAGE (INFINITE)	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	100.0%	1	100.0%	Dark Orange	Orange
COMPARISON	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	100.0%	2	30.0%	Red	Red

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

## Average

The screenshot shows the Average graph settings panel with the following settings:

Graph Type	Peaks	Micro-sonogram	Sonogram fill	Line opacity	Line width	Fill opacity	Color 1	Color 2
AVERAGE	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	100.0%	1	100.0%	Dark Blue	Blue

## Copy analysis

## Copy analysis

Copy analysis button copies the current state of the analysis into the system clipboard so that you can paste it into another analyzer for comparison. Hold **ctrl** to export the analysis into a CSV file.

## Peaks

## Peaks

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

Line opacity 100.0%

## Line opacity

Line opacity controls the opacity of the graph outline.

## Micro-sonogram **Micro-sonogram**

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

### Line width **Line width**

Line width controls the width of the graph online.

### Sonogram fill **Fill**

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

### Fill opacity **Fill opacity**

Fill opacity controls the opacity of the graph interior fill.

## Average (infinite)

**AVERAGE (INFINITE)** Copy analysis Enable ? ⌵

<input checked="" type="checkbox"/> Peaks	Line opacity	<input type="text" value="100.0%"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input checked="" type="checkbox"/> Micro-sonogram	Line width	<input type="text" value="1"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input checked="" type="checkbox"/> Sonogram fill	Fill opacity	<input type="text" value="100.0%"/>	<input type="checkbox"/>	<input type="checkbox"/>

### Copy analysis **Copy analysis**

Copy analysis button copies the current state of the analysis into the system clipboard so that you can paste it into another analyzer for comparison. Hold **ctrl** to export the analysis into a CSV file.

## Maximum

**MAXIMUM** Copy analysis Enable ? ⌵

<input checked="" type="checkbox"/> Peaks	Line opacity	<input type="text" value="100.0%"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input checked="" type="checkbox"/> Micro-sonogram	Line width	<input type="text" value="1"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input checked="" type="checkbox"/> Sonogram fill	Fill opacity	<input type="text" value="100.0%"/>	<input type="checkbox"/>	<input type="checkbox"/>

### Copy analysis **Copy analysis**

Copy analysis button copies the current state of the analysis into the system clipboard so that you can paste it into another analyzer for comparison. Hold **ctrl** to export the analysis into a CSV file.

## Maximum (infinite)

**MAXIMUM (INFINITE)** Copy analysis Enable ? ⌵

<input checked="" type="checkbox"/> Peaks	Line opacity	<input type="text" value="100.0%"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input checked="" type="checkbox"/> Micro-sonogram	Line width	<input type="text" value="1"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input checked="" type="checkbox"/> Sonogram fill	Fill opacity	<input type="text" value="100.0%"/>	<input type="checkbox"/>	<input type="checkbox"/>

### Copy analysis **Copy analysis**

Copy analysis button copies the current state of the analysis into the system clipboard so that you can paste it into another analyzer for comparison. Hold **ctrl** to export the analysis into a CSV file.

## Maximum - Average (infinite)

**MAXIMUM - AVERAGE (INFINITE)** Copy analysis Enable ? ⌵

<input checked="" type="checkbox"/> Peaks	Line opacity	<input type="text" value="100.0%"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input checked="" type="checkbox"/> Micro-sonogram	Line width	<input type="text" value="1"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input checked="" type="checkbox"/> Sonogram fill	Fill opacity	<input type="text" value="100.0%"/>	<input type="checkbox"/>	<input type="checkbox"/>

### Copy analysis **Copy analysis**

Copy analysis button copies the current state of the analysis into the system clipboard so that you can paste it into another analyzer for comparison. Hold **ctrl** to export the analysis into a CSV file.

## Comparison

**COMPARISON** Paste analysis  Enable  ?

<input type="checkbox"/> Peaks	Line opacity	<input type="range" value="100.0%"/>	<input type="checkbox"/>
<input type="checkbox"/> Micro-sonogram	Line width	<input type="text" value="2"/>	<input type="checkbox"/>
<input type="checkbox"/> Sonogram fill	Fill opacity	<input type="range" value="30.0%"/>	<input type="checkbox"/>

**Paste analysis**

**Paste analysis**

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

**Sonogram panel**

**SONOGRAM** Presets    ?

Opacity

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

**Presets**

**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.



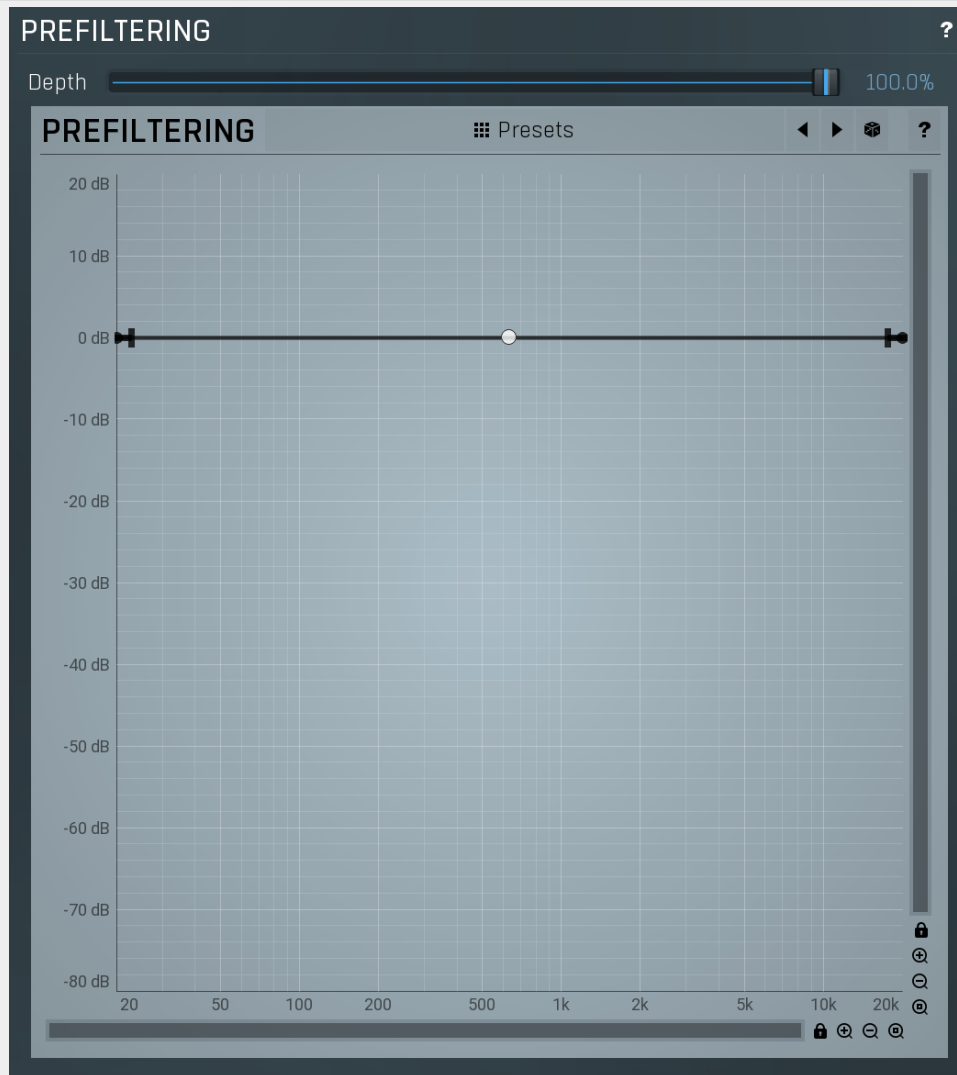
**Randomize**

Randomize button loads a random preset.

Opacity  **Opacity**

Opacity controls the opacity of the sonogram.

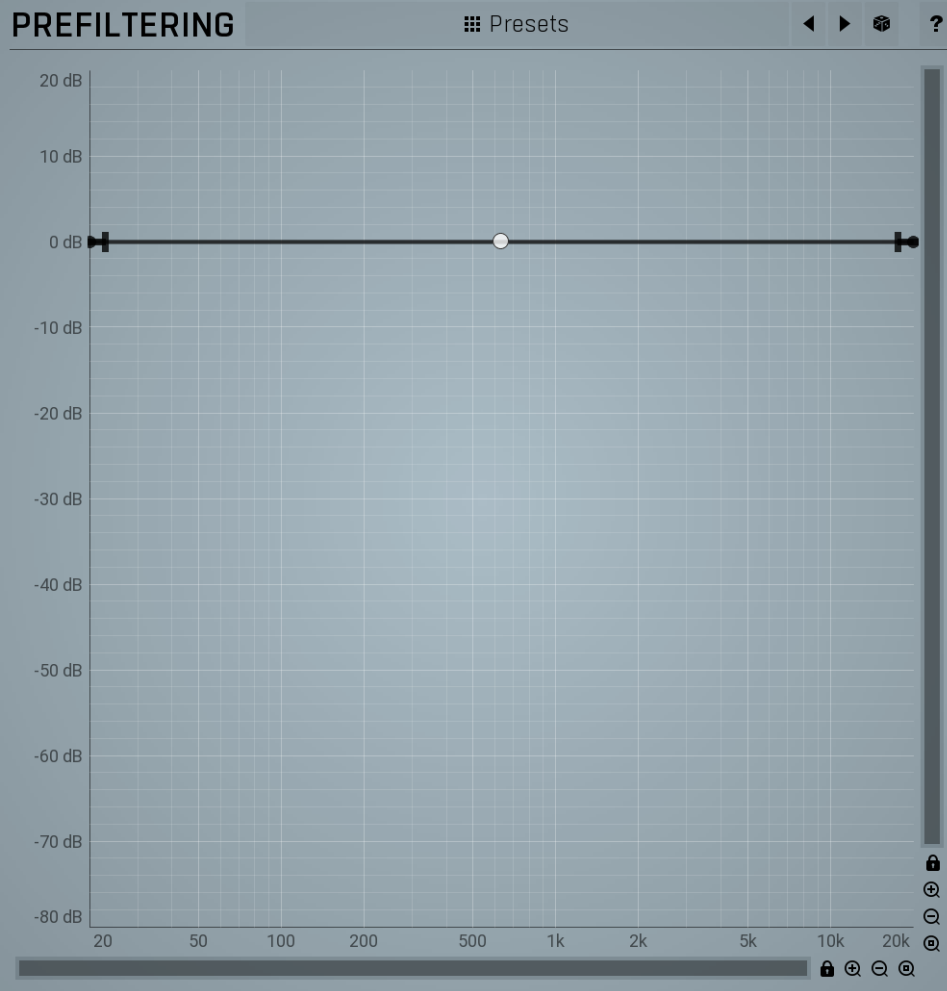
## Prefiltering panel



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

Depth  Depth

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



Prefiltering

## Envelope graph

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

- **Left mouse button** can be used to select points. If there is a *point*, you can move it (or the entire selection) by dragging it. If there is a *curvature circle*, you can set up its tension by dragging it. If there is a *line*, you can drag both edge points of it. If there is a *smoothing controller*, you can drag its size. Hold **Shift** to drag more precisely. Hold **Ctrl** to create a new point and to remove any points above or below.
- **Left mouse button double click** can be used to create a new point. If there is a *point*, it will be removed instead. If there is a *curvature circle*, zero tension will be set. If there is a *smoothing controller*, zero size will be set.
- **Right mouse button** shows a context menu relevant to the object under the cursor or to the entire selection. Hold **Ctrl** to create or remove any points above or below.
- **Middle mouse button** drag creates a new point and removes any points above or below. It is the same as holding Ctrl and dragging using left mouse button.
- **Mouse wheel** over a point modifies its smoothing controller. If no point is selected, then all points are modified.
- **Ctrl+A** selects all points. **Delete** deletes all selected points.

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.

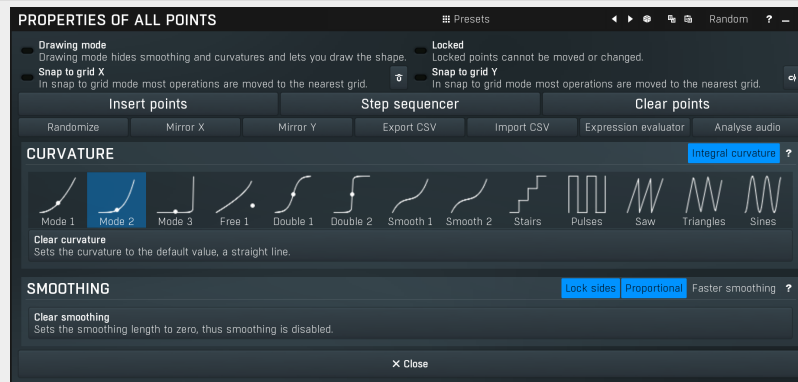


#### Randomize

Randomize button loads a random preset.



## Envelope graph menu



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

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.

Random

### Random

Random button generates random settings using the existing presets.

Snap to grid X

In snap to grid mode most operations are moved to the nearest grid.



### Snap to grid X

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

Snap to grid Y

In snap to grid mode most operations are moved to the nearest grid.



### Snap

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

Insert points

### Insert point

Insert point button creates a point at mouse position.

Step sequencer

### Step sequencer

Step sequencer button generates the envelope from step sequencer.

Clear points

### Clear points

Clear points button deletes all points.

Randomize

### Randomize

Randomize button slightly modifies the Y coordinates.

Mirror X

### Mirror X

Mirror X button inverts the X coordinates of all points.

Mirror Y

### Mirror Y

Mirror Y button inverts the Y coordinates of all points.

Export CSV

### Export CSV

Export CSV feature lets you export the graph to a CSV file. CSV file is a simple text format, which has multiple lines with X and Y coordinates delimited by ';'. For example:

```
0.275;0.2
0.438;0.5
0.775;0.67
```

Import CSV

### Import CSV

Import CSV feature lets you select a CSV file and imports the graph points from it. CSV file is a simple text format, which has multiple lines with X and Y coordinates delimited by ';'. For example:

```
0.275;0.2
0.438;0.5
0.775;0.67
```

Expression evaluator

### Expression evaluator

Expression evaluator lets you generate points based on a mathematic formula. The only input variable is 'x', so as an example you may write 'ln(x^3 + 1) - sin(x\*x)'.  
Expression evaluator uses traditional C/C++ style formatting, which is natural for most people. It provides arithmetics, logical and conditional operators. Following terms are supported:

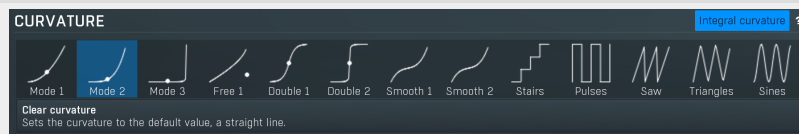
Unary operators: - (negative sign), ! (logical negation)  
Binary arithmetic operators: +, -, \*, /, ^, %, max, min  
Binary logical operators: ==, !=, <, <=, >, >=, &&, ||, ^^  
Unary functions: log, sqrt, sqr, abs, exp, sin, cos, tan, sinh, cosh, tanh, inv, asin, acos, atan, ln, log10, sgn, floor, ceil, round, rand, f01 (frequency from 20...20000 into log scale 0..1)  
Ternary logical operator: a ? b : c (if a is true, then the result is b, otherwise it is c)  
Constants: pi, e, sqrt2, ln2

Analyse audio

### Analyse audio

Analyse audio lets you analyse a portion of an audio file at specified intervals, extract its level envelope and use those levels to construct the graph's curve.

## Curvature

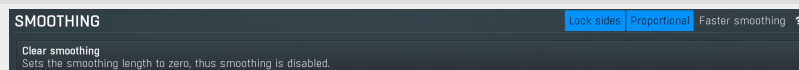


Integral curvature

### Integral curvature

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

## Smoothing



Lock sides

### Lock sides

Lock sides makes the smoothing factor equal on both sides.

Proportional

### Proportional

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

Faster smoothing

### Faster smoothing

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



## Collapse

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

# Band panel



Band panel contains parameters of a particular band. You can select a band using the band editor above, just click on the band in the graph.

## 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.



## Copy

Copy button copies the settings onto the system clipboard.



## Paste

Paste button loads the settings from the system clipboard.

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.

Reset

## Reset

Reset button loads the default settings.

Link

## Link

Link button enables parameter linking between bands. Every parameter change performed with this enabled changes that parameter in all bands. But **Pasting** settings from the system clipboard does not change the other bands.



## Left

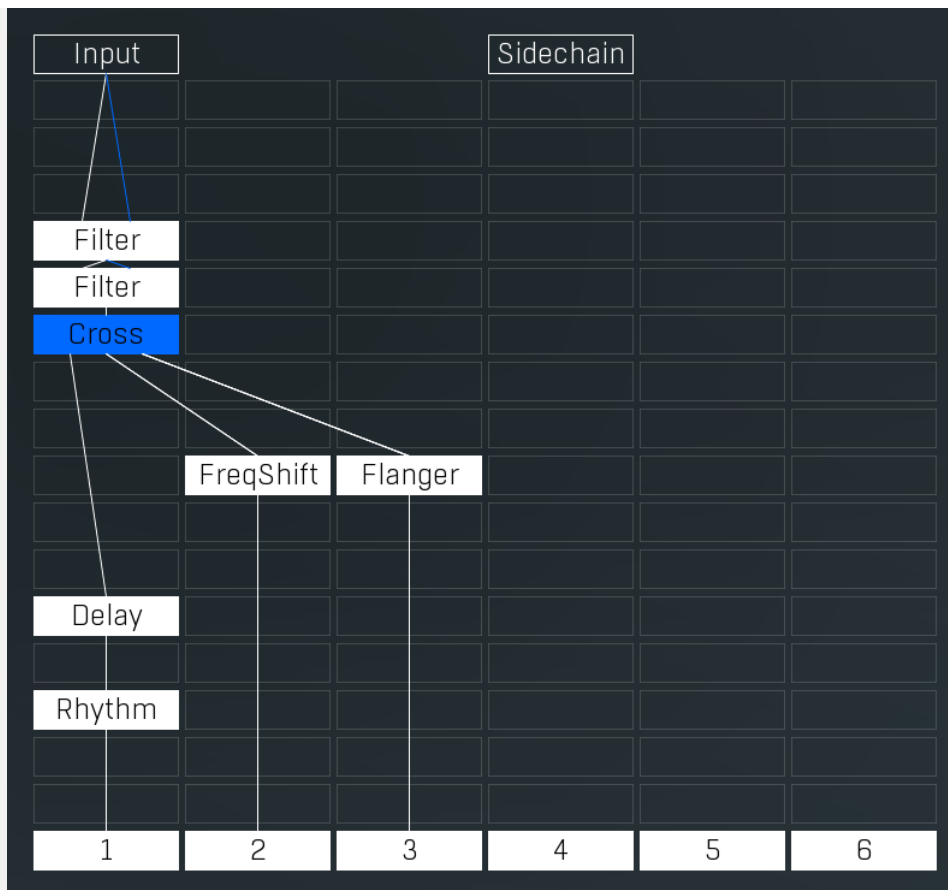
Left button selects the previous band. If this is the first band, it selects the last one instead. This way you can easily cycle between the bands if selecting them in the band editor is hard because they are modulated for example.



## Right

Right button selects the next band. If this is the last band, it selects the first one instead. This way you can easily cycle between the bands if selecting them in the band editor is hard because they are modulated for example.

# Modular editor



Modular editor lets you edit the processing flow. There is at least one input at the top, several outputs at the bottom and several processing boxes in-between. Processing is always performed from top to bottom and is indicated by a solid line. All outputs except the first one are disabled by default, unless you put a plugin into its chain. Every output can be disabled by clicking on it.

Each processing box can contain one plugin. By default all plugins are taking their inputs directly from above (thus from the same chain that they are in), but if you for example put a plugin in the first row of the second chain, there is no input above, so it is redirected to the first input. Some plugins may have multiple input channels or even a variable number of input channels. Most plugins have just one output channel, in which case they place the output into the same chain that they are in. Some plugins have variable output channels in which case they send them to the same channel, the next one to the right and so on.

If you **click** on an empty box a menu with all available processing plugins will be shown. If you click on a box which already contains a plugin, it will be selected and its editor shown to the right. The currently-selected plugin is highlighted.

If you hold **Ctrl** and click on the plugin or click using **right mouse button**, the advanced settings window will be displayed. You can use it to configure the plugin input and output channels and much more.

**Double-click** on the plugin to enable or disable it.

Hold **Alt** and click on the plugin to delete it.

Hold **Shift** and click on the plugin to show its editor in a pop-up window.



**Collapse**

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

## Crossover 1



## 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.

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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.

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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.

All

## Channel mode

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

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

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

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

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

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

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

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

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

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

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

**Side (S) mode** is complementary to M mode, and allows processing of only the side (stereo) part of the signal leaving the mid intact. The same techniques as described for M mode can also be applied here, giving the opposite results.

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

A **compressor** can attenuate the side part in louder sections making it more monophonic and centered, placing the origin a little further away and in front of the listener.

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

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

A **saturator / exciter** may make the stereo richer and more appealing by creating higher harmonics without affecting the mid channel, which could otherwise become crowded.

**Modulation effects** can achieve the same results as in mid mode, but this will vary a lot depending on the effect and the audio material. It can be used in a wide variety of creative ways.

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

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

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

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

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

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

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

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

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

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

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

1x

## Upsampling

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

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

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



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

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



### Show window

Show window button displays the plugin in a dedicated pop-up window. You can do the same thing by Shift + click on the processor item in the modular grid.

	Analog	Analog LP	Linear-phase
Crossover	Hybrid	Level	Panorama
	Mid/Side	Spectrum	Tonal/Transient

### Crossover panel

Crossover panel contains configuration of the crossover used to separate the signals for each band. **Crossover** is a technical term for an algorithm or device which splits a signal into multiple bands (or signals), which when mixed back together recreate the original signal (meaning that the crossover is transparent). The plugin provides several types of crossover with a flat (or nearly flat) response, which means that whichever crossover you choose and whatever signal you send into the plugin, the output levels of each frequency, after the bands are mixed back together to get the output signal, will be (almost) exactly the same, unless there is some processing applied in the bands themselves. Most of the available crossover types produce bands with different frequency ranges; however there are also a few more creative ones.

**Analog crossovers** have no latency, but they exhibit a phase-shift. That is usually irrelevant unless you are going to mix the output with the input later on. Analog crossovers are based on the classic analog components that you can find in speaker systems for example, however they are perfectly accurate and their slope (band separation) ranges from 6dB/octave to a very steep 120dB/octave. The higher the slope is, the more separated is each band (that is, there is less overlap between bands), but also the bigger is the phase shift. That can reach such an extent that some bassy materials become severely phasey, which may or may not be a good thing. An exception to the rule is the 6dB/oct crossover, which is zero-phase naturally. Its disadvantage is that the separation between bands is rather low, 6dB/oct is often not enough.

**Analog LP crossover** is a linear-phase equivalent to the **Analog** crossover. It introduces latency as does any linear-phase filter, but it does not cause a phase-shift. This may be especially advantageous for higher filter slopes, which, with classic analog crossovers, would cause severe transient smearing. Please note that the crossover type may not be 100% transparent, especially with small bands in bass spectrum and high slopes.

**Linear-phase crossover** is a fully digital crossover with a high slope (frequency-dependent), which introduces latency, but exhibits no phase-shift. This crossover mode is designed specifically for mastering.

**Hybrid crossover** is linear-phase as well, hence it introduces latency, but no phase-shift. However, its slope is more similar to the slopes of the analog crossovers.

**Level crossover** is a very specialized tool, which doesn't filter the input signal at all (hence it is not only linear-phase, but also zero-phase). Instead of filtering, it simply performs a gain on each band in such a way that when all the bands are mixed back together, the output is the original signal again. When you select this crossover, the spectrum analyzer graph disappears and the X axis in the band editor changes from frequencies to dB levels. So the band limits are not frequencies anymore, but rather sound levels.

The current level displayed in the graph area is controlled by the **Level** value below and you are likely to use a modulator, most likely in **Follower** mode, to control this latter value. The crossover then applies gain to each band depending on how much the current level fits into the band. The **Slope** parameter controls how quickly each band fades into the adjacent one. This crossover effectively turns the plugin into a very advanced dynamics processor; using a Follower Modulator the band used to process the input audio depends on the audio level.

*The are many possibilities for this crossover. But the basic principle is to select a spare Modulator, configure it as a Follower and select the Global parameter "Crossover Level value" as its target, with a "Full range" range mode. After configuring the Modulator, you will be able to see the detected value curve in the Modulator's Level graph. Then if the input signal is strongest, the right most band is processed etc. So if you for example use a delay with 2 bands and set the band limit high enough, the 2nd band will be processing only the loud parts of the signal and vice versa.*

**Panorama crossover** is another specialized tool, similar to the level crossover; it splits the signal into bands according to the panorama. If, for example, you create 3 evenly spaced bands (100%L to 33%L, 33%L to 33%R, 33%R to 100%R), then the leftmost band will contain mainly the signals located in the left speaker, the rightmost band will contain mainly signals from the right speaker and the middle band will contain centred signals. Please note that this doesn't mean the crossover attempts to analyze the space the signals are coming from and send them to the respective bands, which is probably what your brain would attempt.

This crossover is useful only when processing stereophonic (or surround, in which case the channels from 3 upwards are kept intact) signals and can be used for all kinds of mixing and creative processing. For example, using a multiband compressor with this crossover can be used to effectively control the stereo image as each band would be processing a different part of the stereo image. To mention another example, a multiband delay or reverb can be used to produce a different ambience for different parts of the stereo image.

**Mid/side crossover** is similar to panorama crossover, but it splits the signal according to their position in mid/side location. In other words, the more to the left a band, the more centred is the signal in it. Similarly the more to the right a band, the more "to the side" is the signal in it. You can think of it as the panorama view folded back on itself, around the center position. If, for example, you create 3 evenly spaced bands (centre to 33% L or R, 33% L or R to 67% L or R, 67% L or R to 100% L or R), then the leftmost band will contain the centred signal, the rightmost band will contain the signals to the extreme left or right and the middle band will contain signals in between. It can be used for similar tasks as the panorama crossover.

**Disabled crossover** is not a crossover actually, it simply disables the crossover and as a result each band processes the full input signal. As a consequence there is likely to be an increase in output level, so take care and turn down the output level first. In practice this "not really crossover" mode lets you process multiple streams of the input audio signal in parallel. For example, if you use a compressor, this in effect produces an extreme parallel compression. As another example, you can use a reverb to produce several rooms in parallel, potentially leading to a fuller space for example.

**Spectrum crossover** is the first of the spectral crossovers. It splits the signal into individual frequencies, analyzes their levels and sends the frequencies with the highest level into the highest band etc. It marks each frequency with its level (as you can see on the dB scale on the X axis in the crossover band editor) and puts it into the appropriate band. The crossover is linear-phase and fully transparent.

It provides a huge (not only) creative potential as it lets you process the dominant and weak parts of the signal individually. For instance, by compressing the dominant frequencies using MDynamicsMB you can bring more attention to the unsubstantial frequencies in the signal and in a way stabilize it without disrupting the silent parts of it. Note that this is NOT the same thing as using a normal compressor, because this way it treats only the loud frequencies even if the weak frequencies are present at the same time. Another example could be using MDelayMB to generate echoes only to the dominant parts of the signal, such as snare and bass drums in a drum loop.

**Transient crossover** is also a spectral crossover. It splits the signal into individual frequencies and sends the transient parts for each of them into the highest band etc. It marks each frequency with its "current transientness" (defined by the percentage scale that you can see on the X axis in the crossover band editor) and puts it into the appropriate band. The crossover is linear-phase and fully transparent.

It provides a huge (not only) creative potential as it lets you process split the signal into tonal and transient parts (and anything in between) and treat each individually. For instance, by compressing the transients using MDynamicsMB you can easily control the attack of drums. Note that this is NOT the same thing as using a normal compressor, because this way you can treat only the attacks in an already mixed signal without affecting the remaining part of the signal. Another example could be using MDelayMB to generate echoes only for the attacks of each drum.

Slope	6dB/oct A	6dB/oct B	12dB/oct
	24dB/oct	48dB/oct	72dB/oct
	96dB/oct	120dB/oct	

## Slope

Slope defines the slope of each band transition and is used only by analog crossovers (including the linear-phase versions). It essentially controls the separation between the bands - the higher the slope, the lower the overlap between bands. Higher slopes require more CPU power and exhibit higher phase shift, which may be a problem especially when percussive materials. In these cases it may be necessary to switch to a linear-phase version.

Interesting exception to the classic rule are the 6dB/oct crossovers, which are linear-phase by nature (while still being zero latency), because the bands compensate for each other's phase shift. A side-effect of this is that the signal level in each band is much higher than using other crossovers, so you may expect these crossovers sound considerably different to the other modes.

Level value	-12.0 dB	Level value
-------------	----------	-------------

Level value is used only with **Level crossover** and controls the level at which the signal is split into each band. You will probably want to attach this parameter to a modulator in Follower mode for instance.

Range: silence to 0.00 dB, default -12.0 dB

Level slope	50.0%	Level slope
-------------	-------	-------------

Level slope is used only with some crossover modes (Level, Spectrum and Tonal/Transient) and controls how quickly each band fades into the next one. It's similar to the **Slope** parameter used with analog crossovers.

Range: 0.00% to 100.0%, default 50.0%

Transient release	50 ms	Transient release
-------------------	-------	-------------------

Transient release is only used by the **Tonal/Transient** crossover and controls the release time of each transient. The transients detected by the crossover are naturally very short, so this provides a way to make them longer, hence send more signal to the higher bands of the crossover (receiving transients) and less to the lower bands (receiving the remaining part of the signal).

Range: 0 ms to 1000 ms, default 50 ms

Transient resolution	50 ms	Transient resolution
----------------------	-------	----------------------

Transient resolution is only used by the **Tonal/Transient** crossover and controls the behaviour of the spectral transient detector. You can use it to adjust the crossover to your audio material and we would recommend a simple trial-and-error approach.

Range: 0 ms to 200 ms, default 50 ms

Smoothing	0.00%	Smoothing
-----------	-------	-----------

Smoothing is only used by spectral crossovers and controls how frequencies affect their surroundings. Without smoothing the individual bands may sound a bit artificial, because human brain general dislikes separated frequencies. It usually doesn't matter unless you audition the bands separately, but sometimes when more "brutal" processing is used on each band, it may become

audible, which is where the smoothing can provide a solution at the cost of additional CPU and lower separation between bands, because it naturally makes the frequencies "more alike".

Range: 0.00% to 100.0%, default 0.00%

## Spectral resolution 50.0% **Spectral resolution**

Spectral resolution is only used by spectral crossovers and controls the spectral transformation settings. The higher the value is, the higher FFT size and overlap size is used, and therefore more CPU is usually required as well. Whether higher/lower value is good or not depends on the actual signal, the default 50% should work well with most audio materials. Higher values will generally provide better frequency resolution (usually good for less percussive sounds), lower values will provide better time resolution (usually good for more percussive sounds), eventually it is always about a compromise.

Range: 0.00% to 100.0%, default 50.0%

## Tone 0.00 dB **Tone**

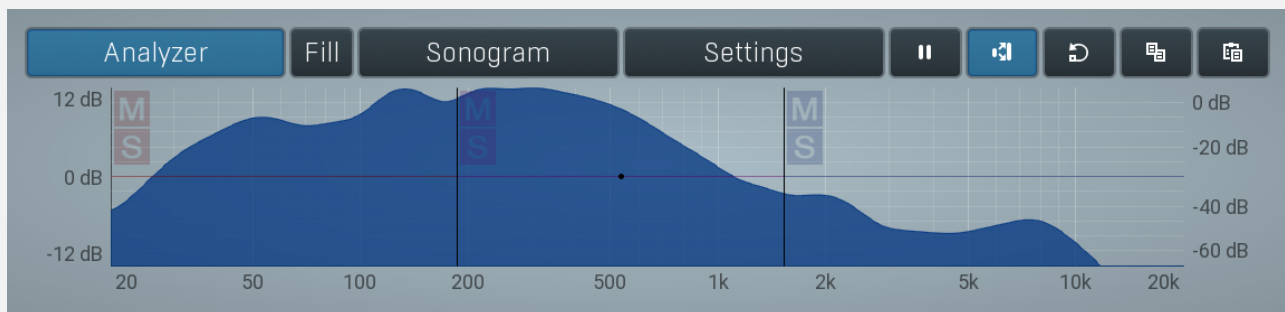
Tone is only used by spectral crossovers and controls the spectral slope applied by the detector. It is exactly the same feature as the **Slope** in analyzers and the crossover uses it to determine how to spread the frequencies between the bands. Higher slope gives more energy to higher frequencies and vice versa. Note that whatever the settings are, the crossover still produces signals that perfectly sum to the original input signal, meaning that it is perfectly transparent and unless the bands are actually doing something, you won't be able to hear a difference when changing this parameter.

Range: -12.00 dB to +12.00 dB, default 0.00 dB

Set default band limits

## Set default band limits

Set default band limits button sets the band limits equally spaced along the X axis.



## Band editor

Band editor shows the available bands, the cross-over frequencies delimiting them, the input gains and panoramic positions. Use the left mouse button to change cross-overs, input gains or pan positions. Use the right mouse button to add more bands, solo them etc.

## Analyzer **Analyzer**

Analyzer button enables or disables the spectrum analyzer, which shows the levels of individual frequencies. In most practical cases it is more convenient to use the sonogram, which shows the frequencies in time, but provides a lower level resolution as the levels are differentiated by color. The spectrum analyzer also provides a micro-sonogram (shown in the bottom of the panel) which uses the same color-based view as the sonogram.

## Fill **Fill**

Fill button enables or disables the full-sized analyzer micro-sonogram. This means that the micro-sonogram at the bottom of the equalizer graph will fill the whole analyzer view. Color differentiation is often easier to understand than the classical spectrum analyzer, so this might help you better understand the spectrum of your audio material.

An alternative is to use the spectrum sonogram.

## Sonogram **Sonogram**

Sonogram button enables or disables the spectrum sonogram, which shows levels of individual frequencies in time. Levels are differentiated by color, so the accuracy is not as good as when using the spectrum analyzer. However, the time axis improves the visual orientation in the spectrum for typical audio signals. In contrast, the spectrum analyzer is more of a scientific tool.

## Settings **Settings**

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

## Pause **Pause**

Pause button stops the analyzer temporarily.

## Normalize **Normalize**

Normalize button enables or disables the visual normalization, which makes the loudest frequency be displayed at the top of the analyzer area (0dB); it does not normalise the sound. This is very useful for comparing frequency levels, however it does hide the actual level.

When comparing 2 spectrums you are usually interested mainly in the frequency level differences. In most cases both audio materials will have different overall levels, which would mean that one of the graphs would be "lower" than the other, making the comparison quite difficult. Normalize fixes this and makes the most prominent frequencies of the spectrum reach the top of the analyzer area (or have the most highlighted color in case of sonogram).



### Reset

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



### Copy

Copy button copies the current analysis to the system clipboard. Then you can use the paste button to show the analysis as a comparison in any of analyzer instanced.



### Paste

Paste button pastes the analysis from the system clipboard and displays it as the comparison in the graph.

## Global parameters panel

Dry/Wet	75.0%	0%	100%
Input	0.00 dB		
Output	+6.00 dB		
Temporary gain	0.00 dB		

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

Dry/Wet  **Dry/wet**

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

0% **0%**

0% button makes the **Dry/Wet** virtually 0%. You can use it for comparison.

100% **100%**

100% button makes the **Dry/Wet** virtually 100%. You can use it for comparison.

Input  **Input gain**

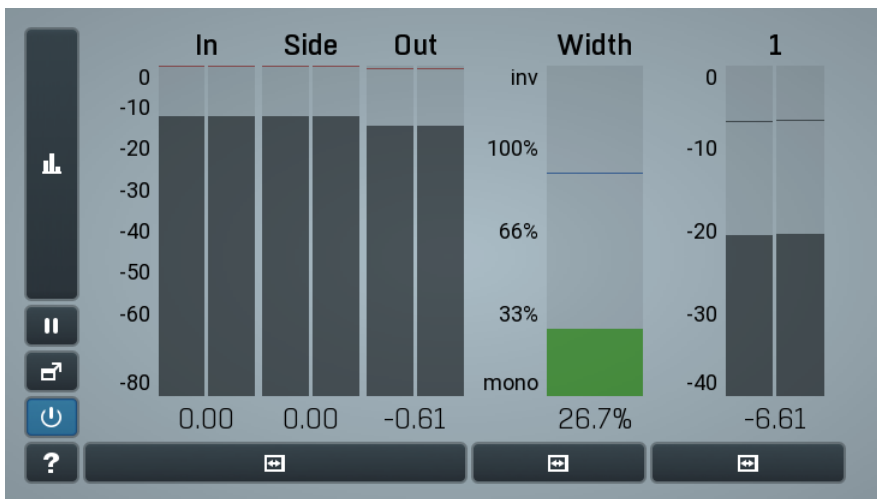
Input gain defines the power modification applied to the incoming signal, before it is split into bands.

Output  **Output gain**

Output gain defines the power modification applied to the output signal, right after it is summed from the bands.

Temporary gain  **Temp gain**

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



## Global meter view

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

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

**Numbered band meters** display the input levels for each band.

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

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

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

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

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

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

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

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

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

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



## Time graph

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.

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## Utilities



### **Map**

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

### **Modulator**

Modulator button displays settings of the modulator. It also contains a checkbox, to the left, which you can use to enable or disable the modulator. Click on it using your right mouse button or use the **menu button** to display an additional menu with learning capabilities - as described below.

### **Menu**

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

**Learn** activates the learning mode and displays "REC" on the button as a reminder, **Clear & Learn** deletes all parameters currently associated with the modulator, then activates the learning mode as above. After that every parameter you touch will be associated to the modulator along with the range that the parameter was changed. Learning mode is ended by clicking the button again.

In smart learn mode the modulator 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 modulator and also records the range of values that you set.

*For example, to associate a frequency slider and make a modulator 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 modulator window too). Then disable the learning mode by clicking on the button.*



## Menu

Menu button displays additional menu containing features for modulator presets and randomization.



## Randomize

Randomize button generates random arpeggiator settings.



Lock



## Lock

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

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

Dry/Wet

75.0%



## 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.



## 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.*



## **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.

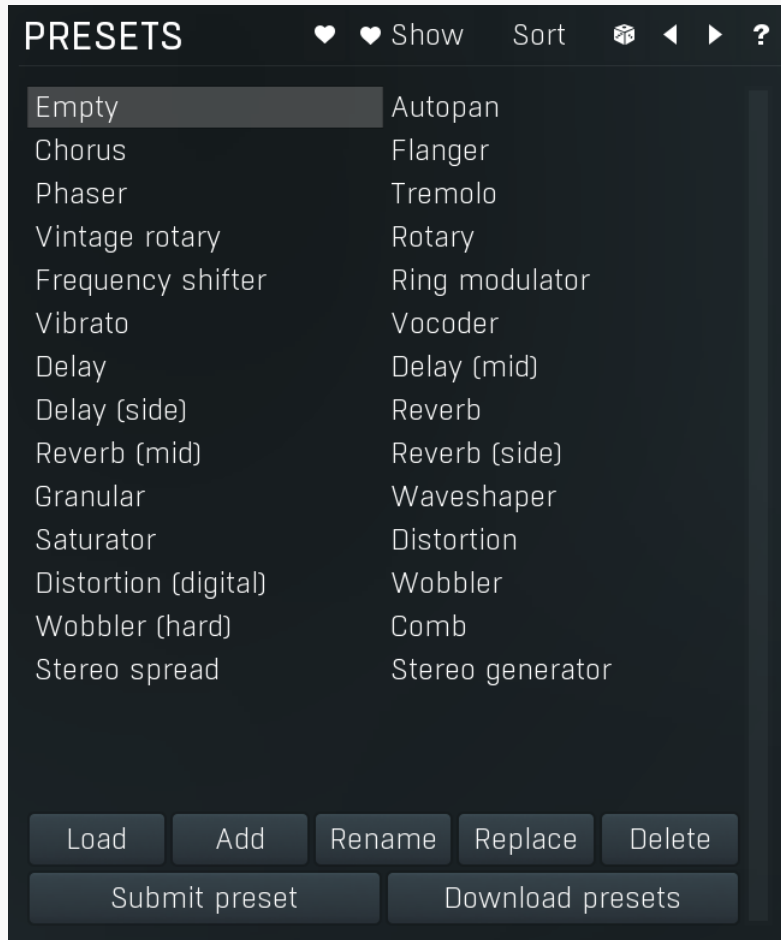
### Export **Export**

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 button shows only the favourite presets and hides the others.

## **Sort**

Sort button shows the presets sorted alphabetically.

## **Random**

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

## **Previous**

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

## **Next**

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

Submit preset

## **Submit preset**

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

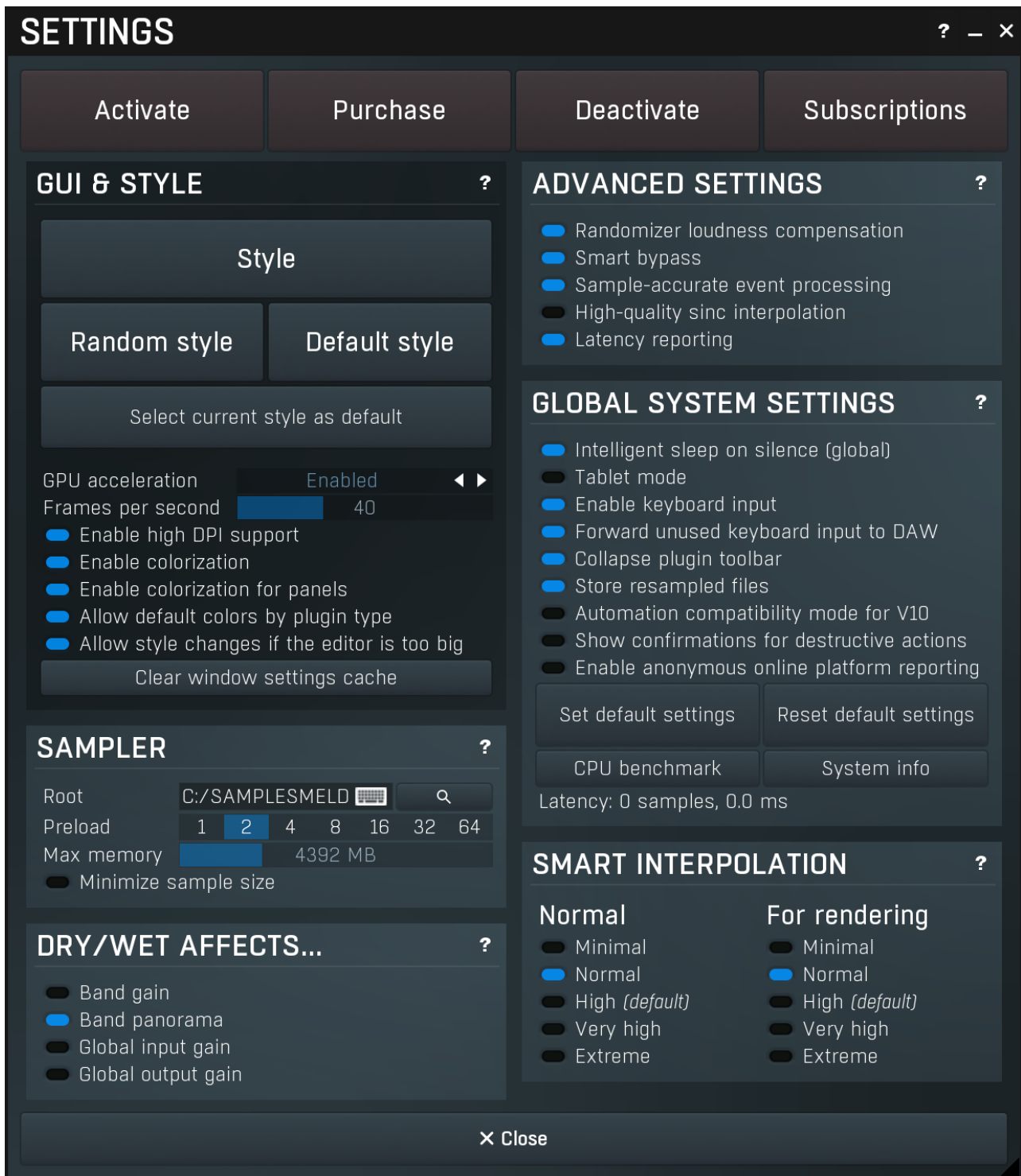
Edit ? 

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

Activate

Purchase

Deactivate

Subscriptions

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

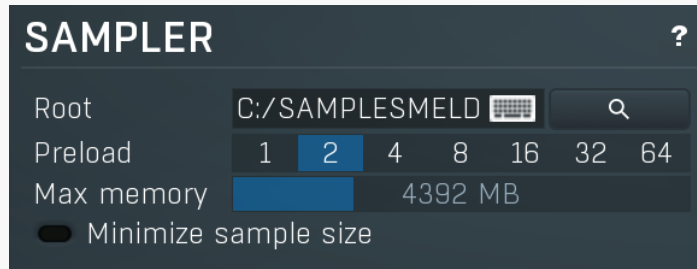
Allow style changes if the editor is too big is on by default and makes the plugin change its style, editor mode and other settings if it finds out it is too big to fit the current screen resolution.

## Clear window settings cache

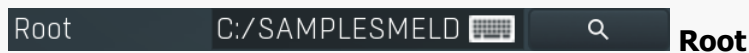
### Clear window settings cache

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.

# Sampler panel



Sampler panel contains settings of the sampler module.



**Root**  
Root defines the main path for samples and samples banks. By placing all factory content in a single folder the plugin always knows where the data is, since these may be referenced from various places. It is recommended to use a high-speed SSD drive for this data for optimal performance.



**Preload**  
Preload controls how much of the samples are loaded initially when loading an instrument/project. Increasing this value makes the loading slower and increases memory consumption, but avoids possibility of required samples not being loaded in time producing audible clicks and lags. Lower this value if your computer (and specifically your hard drives) is fast enough. The compression algorithm organizes the samples typically in 64kB blocks and this value controls the number of blocks being loaded.

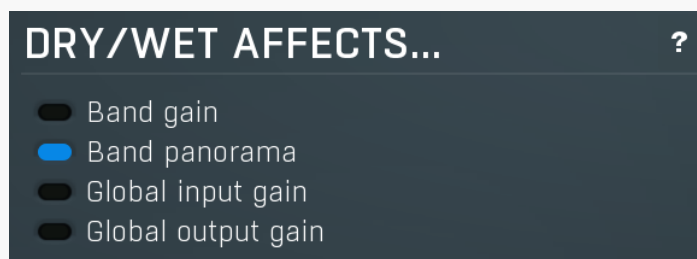


**Max memory**  
Max memory controls the maximum memory occupied by samples above which the engine starts releasing the unused ones. Please note that this value is only a hint, the consumed memory may exceed this value when the samples are needed. Lowering this value will reduce the memory usage, however it will also require additional CPU/disk processing to reload the samples when needed. Therefore it is best to keep this value as high as possible. If the OS runs out of system memory, it should still be able to operate properly using the so-called virtual memory, which means that unused memory blocks are swapped to HDD.



**Minimize sample size**  
Minimize sample size makes the sampling engine lower the memory overhead by keeping the samples in a semicompressed form. Note that this will lower the memory consumption at the expense of potentially slightly higher CPU consumption.

# Dry/wet affects... panel



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



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



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



## Global input gain

### Global input gain

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

## Global output gain

### Global output gain

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

## Advanced settings panel

### ADVANCED SETTINGS ?

- Randomizer loudness compensation
- Smart bypass
- Sample-accurate event processing
- High-quality sinc interpolation
- 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.

## Randomizer loudness compensation

### Randomizer loudness compensation

Randomizer loudness compensation enables the automatic detection of loudness after new settings have been generated using the main **Random** button and using the output gain of the plugin to get some predefined level. This is useful in most cases since normally randomized settings can produce various output levels, so this can mitigate the problem.

## 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.

## High-quality sinc interpolation

### High-quality sinc interpolation

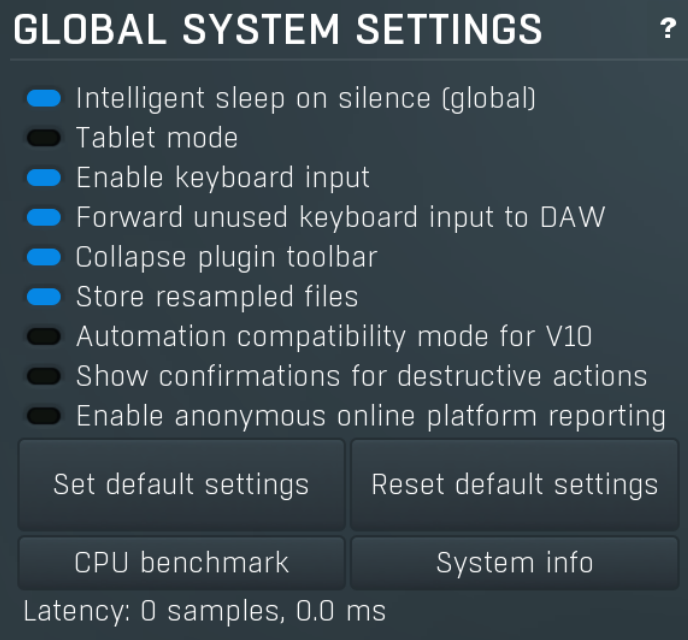
High-quality sinc interpolation activates the maximum audio quality interpolation algorithm, which is used by several modulation processors, such as a flanger or chorus. Unlike standard interpolation sinc interpolation avoids distortion and improves response for high frequencies. The disadvantage is that it introduces some latency and requires much 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 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

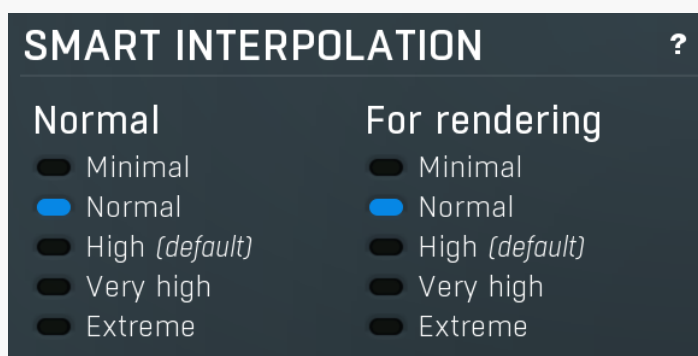
CPU benchmark button calculates the performance of the plugin with the current settings.

System info

### System info

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

## Smart interpolation panel



Smart interpolation 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.*

# Modulator editor



Modulator is an extremely advanced feature, which lets you change parameters automatically depending on various inputs. You can use this to add movement to your sound, respond to some plugins differently for louder sections, or even follow the pitch of the input.

The modulator edit window has two parts: on the left side you can configure the mode of the modulator (the way the modulator works) and on the right side there is a list of parameters to modulate. A modulator can control all automatable parameters (and often more than that) including the parameters of other modulators. Each modulator can control as many parameters as is needed and each of the parameters has its own range and transformation shape. The values and ranges of the first 4 parameters associated with the other modulators can also be modulated/automated. The following modulator modes are available:

**Normal** mode makes the modulator behave like an ordinary low-frequency oscillator (LFO). There are various ways to control its shape as with all oscillators in our plugins. Each modulator can synchronize to the host in the **Synchronization** panel. Modulators can also synchronize with each other using the **Sync groups**. Using **MIDI reset** you can reset the oscillator to any phase using MIDI notes, but obviously to-host synchronization must be disabled in order for this to work.

Note that the settings in this mode are used even if the modulator is actually in a different mode by using "LFO modulation". This basically blends between the actual mode, which may for example detect the input signal level, and give it some additional movement using the LFO depending on the LFO modulation parameter available for each of the remaining modes.

**Follower** mode makes the modulator detect the input signal level. It contains an extremely advanced and accurate level detector taken from our MDynamics plugin. The level follower is an immensely useful feature, yet it may be a little difficult for beginners to comprehend, so we will cover it here in more detail.

It is often necessary to adjust the follower slightly for new material. First, it has the standard parameters - attack, release, hold and RMS length. These are fairly standard features and help is available for each of them. **Level min and max** controls the range of input levels. When the input level is equal to or below the min level, the modulated parameters' values will be minimal. Similarly, when it reaches the max level, the modulated parameters' values will be at their maximum. This allows for adjustments to the range of input levels, which are certainly different for any audio material and settings. It can be used creatively too - for example, by using very low values for both limits we can differentiate between silent and non-silent parts, similar to the way a gate effect works.

**Advanced detector settings** provide some extraordinary features, such as psycho-acoustic pre-filtering, which forces the modulator to detect loudness instead of raw input levels, custom input signal pre-filtering using a fully featured 6-band equalizer, and custom attack and release shapes. **Band-pass panel** pre-filters the level detection signal using a band-pass filter, so this is like a very simplified version of the equalizer from the advanced detector settings. **Side-chain** makes the modulator measure side-chain input if the plugin has one. For modular plugins the modulator can also be driven by a feedback signal. The **advanced panel** provides some further level processing features that you can take advantage of creatively or to further adjust to your actual audio material.

**Project onto LFO shape** is a more advanced concept, which is available for other modulator modes too. You can easily imagine, that the modulator in any mode generates values for each parameter, we can say it is between 0 and 1, where 0 sets minimum parameter value, and 1 sets the maximum. Project onto LFO shape forces the modulator to use this range in the oscillator shape, which can then be configured in normal mode. The value is basically transformed by the oscillator shape, where the values generated by the modulator are on

the horizontal axis (phase) and the output is the actual oscillator value. This feature has no physical meaning and can only be used creatively - to transform the more or less linear results of the level follower into a much more complicated curve.

*Let us demonstrate the follower mode with an example - the idea is to apply a delay to a snare drum within a previously mixed drumset. This is commonly used on reggae/dub rhythms for example, however in these cases the snare track is usually available separately. Using the modulators you can get somewhat interesting results even with an already mixed drumset. The idea is to increase the input gain whenever the snare is playing, so that only the snare drum (and potentially other instruments playing at the same moment) are passed into the delay. So first teach the modulator to control input gain parameter of the delay and set it to follower mode, potentially configure some of the parameters to get the desired response. Now the louder the input is, the more delay you get. To make it respond only to snare drum, enable the band-pass and set the filter limits accordingly, e.g. 500Hz to 1k. This makes the input gain increased depending on the input level in this part of the spectrum, which contains the snare drum.*

**Envelope** mode causes the modulator to generate an arbitrary envelope, similar to those from synthesizers. It can either follow MIDI - the envelope starts when a key is pressed, goes through the attack and decay stages, then holds in sustain stage until the key is released when the release stage begins, or it can follow audio - when the audio level exceeds **Threshold on** it behaves the same way as when a note is pressed in MIDI mode, and then when the input level drops below **Threshold off** it behaves like a key release. As with most modes there is LFO modulation and LFO projection and the input level can be driven by the side-chain or feedback if available. The envelope shape can be adjusted using several controls (lengths of each stage etc.) and you can even draw your own shape.

**Random** mode is a smooth random generator. It is very handy if you want some parameters to change over time, but do not actually want them to be periodic like LFOs. A modulator in random mode does not actually generate random values, the results will always be the same at each position in your arrangement in the host. This allows a pseudo synchronization with the host and ensures a "what you hear is what you get" performance. **Speed** parameter controls the speed of change and any slight change to this parameter will change the whole stream.

**Pitch** detects the pitch of the input signal assuming it is not polyphonic (here it can work too and will probably detect the lowest note, however it is definitely not suitable for percussive signals, which do not have a pitch). It is very useful, enabling you to tune an oscillator to follow your singing, or allow an equalizer to control separate harmonics of a vocal, use a distortion to get more drive for higher notes in a guitar solo and much more. The pitch detection may be a little tricky to understand, so we will discuss it in more detail.

A pitch detector takes the input signal and tries to approximate the pitch of the fundamental frequency in it. It is physically impossible to detect pitch instantly, as an extreme example, 20Hz takes 50ms for the signal to evolve enough to detect that there is actually a 20Hz frequency in the signal. For this and many other reasons any pitch detector employs several limitations. These are available in the **Detector panel**. The defaults will work well for most audio material, however, it is useful to understand the parameters, so that you can let the detector adapt better to your particular audio materials if necessary, and also in order to be more creative.

**Min and max frequency** parameters in the Detector panel control the limits of the frequencies you expect in the input. For example, a female voice is unlikely to sing below 100Hz, so it is customary to set the minimum frequency to 100Hz or even higher. Voice signals contain several artifacts, blows and pops, all of which can temporarily create frequencies below the actual pitch of the voice, so setting these limits is preferable to avoid "jumps" to incorrect pitches. **Stabilization** and **Speed** also prevent these jumps by restricting how quickly the pitch can change. These can also be used creatively. **Threshold** controls the minimum level of the input signal to be considered "not-silent and probably having pitch". This acts as a form of gate, which prevents the detector from analyzing irrelevant rumble in between actual performances. **Shift panel** allows the detected pitch to be shifted up or down and **Auto-tune panel** moves it to the closest note - similar to the automatic pitch changing function from MAutoPitch, except no pitch shifting is actually done and the results are used purely to control some parameters.

**Min and max frequency** parameters in the top of the editor have a very different meaning than the parameters of the same name in the detector panel. From now on we will assume that the pitch has been detected successfully and are now considering what to do with the results. Again, we may assume the modulator generates values from 0 to 1, where at 0 the modulated parameters' values become minimal and reach maximum at 1. When the input pitch is equal or below the min frequency parameter, the modulator's value is 0, hence modulated parameters will have a minimal value as well. Similarly when the pitch reaches max frequency, the modulated parameters will get to the maximum.

Now you may say this makes no sense, because the detected pitch cannot exceed the limits specified in the **Detector panel** anyway. The reason for this is that most "frequency" parameters of all plugins are limited from 20Hz to 20kHz, whether it is the frequency of a band in an equalizer, or a high-pass frequency in a phaser for example. It is a reasonable solution since physiologically speaking these figures are on or around the range of our hearing limits.

*Let us explain the concept with an example. We want to modulate a band of an equalizer, so that it always follows the fundamental frequency, the pitch, of our audio material. All we need to do is to switch the modulator to pitch mode, allow it to control the band frequency parameter and set the range for this parameter to the full range, from 20Hz to 20kHz. The pitch detector may then detect frequencies from 50Hz to 2kHz, but the modulator takes it that the actual limits (converted to 0..1) are 20Hz to 20kHz and that exactly the same range is configured for the band frequency parameter, so you could say that "they understand each other". We did not need to touch the min and max frequency parameters at all.*

*Here is one more example, where we would actually want to adjust the min and max frequency parameters. We want to control a drive parameter of a distortion for a guitar so that the higher the guitarist plays the more distortion he gets. Again, we teach a modulator to control the drive parameter, for any range we want, and switch the modulator to pitch mode. Now the modulator will move the drive parameter, but only slightly, because it assumes the pitch can vary from 20Hz to 20kHz, but the guitar may actually only play from about 100Hz to 1kHz. So we can use the min and max frequency parameters to say "what is high and what is low", to limit the frequency range. There are no general rules here, you have to experiment, because every instrument and parameter is different.*

To sum things up, the difference between controlling a frequency parameter and a drive parameter is simply the fact that a frequency parameter is compatible with the pitch. After all, pitch is nothing more than a frequency (strictly speaking it is a logarithmic representation of frequency).

## 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. Note that unlike copy & paste, presets & randomization do NOT affect the set of parameters being modified, hence it serves to optimize adjustment of the modulator behaviour assuming that you already specified the set of parameters to control.

If you hold **Shift**, the plugin will undo previous randomization.



### R

R button enables automation read. This way you can actually automate the modulation value. First you use **W button** to record the modulator values over time. After that you can modify it in some way and enable automation read to override the normal modulator behaviour. Note that the results may be different when automation is used with potentially lower audio quality and slower response.



### W

W button enables automation write. This way you can actually automate the modulation value. Use the button to record the modulator values over time. After that you can modify it in some way and enable automation read to override the normal modulator behaviour. Note that the results may be different when automation is used with potentially lower audio quality and slower response.



### Map

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

## Parameters panel



Parameters panel contains the list of the parameters that the modulator is controlling, their ranges etc.

### 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.

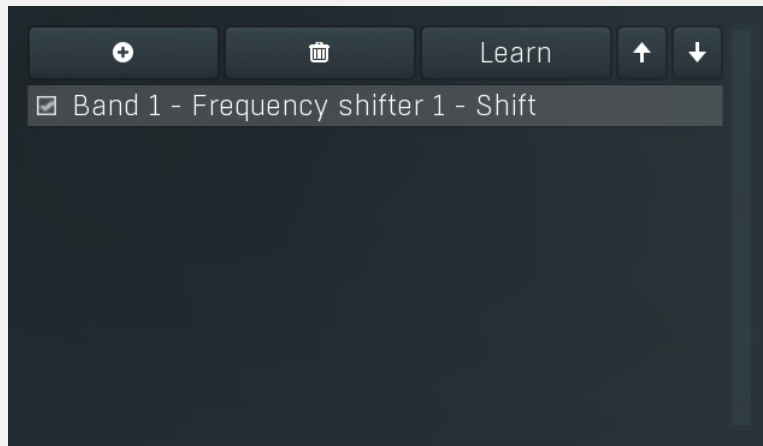
### Randomize

Randomize button loads a random preset.

### Restore original values when disabled **Restore original values when disabled**

Restore original values when disabled makes the modulator restore the original parameter values when it is disabled by automation or modulation. Normally when you manually disable the modulator, the original values are restored as that is usually desired. However when you control the modulator enable state by automation or modulation, you may or may not want this to happen.

# ParameterList



## Add

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



## Delete

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

Learn

## Learn

Learn button starts or stops the learning. Click it, then move some parameters in the plugin, then click it again. Learning can also be accessed from the global modulator menu.



## Up

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



## Down

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

Parameter

Band 1 - Frequency shifter 1 - Shift

Parameter

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

Name



Name

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

Range mode

Up and down



Range mode

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

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

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

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

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

Up and down: -24, -12, 0 (range constrained to 12 dB either side)

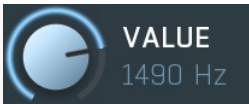
Full range: -24, -6, 12 (range limited to minimum, but not constrained)

Up only: -12, 6, 24 (range not constrained = +/-24 dB \* 75% = 36dB)

Down only: -12, -18, -24 (range limited to minimum)

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





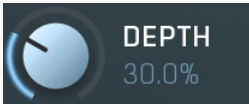
### Value

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



### Maximal value

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



### Depth

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



### Invert

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



### Use first parameter's range

Use first parameter's range makes the parameter display use the same range as the first parameter in the list. This is often useful if you want to control the range in some way and apply the range to multiple parameters.



### Transformation shape

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



Mode defines the way in which the modulator works. The modulator is like a black box that generates one number in range 0% to 100% at each moment and then assigns the appropriate value to each of the target parameters. The mode defines what this number will be. Select the particular tab to control the modulator's behaviour.

**Normal** mode uses a standard low-frequency oscillator (LFO) to drive the parameters.

**Follower** mode uses the level of the input signal.

**Envelope** generates an envelope using MIDI notes or by following input signal level.

**Random** generates randomized output which is however the same every time you render the song.

**Pitch** detects and follows the pitch of the input signal.

# Normal mode

**MODULATOR SHAPE** Default [Navigation icons] Random ?

Shape Sine Custom 25.0% Step 25.0% Smooth 50.0% Advanced

Normal [Waveform icons] Edit Edit

Harmonics [Waveform]

**RATE** Sync ?

Frequency 1.000 Hz

Sync group Disabled 1 2 3 4

Reset on ARP

Length 1 / 2 Type Straight Set frequency

Phase 90° (25.0%) Count 1

**MIDI RESET - (RE)TRIGGER** Enable ?

Note-on  Note-on only first  Note-off  Note-off only last  Single shot  Single shot reset

Min velocity 0.00% Max velocity 100.0%

Min note 0 [C-1] Max note 100 [E7]

Phase 0° (0%) Channel All

Normal mode makes the modulator work as a traditional low-frequency oscillator (LFO). Note that even if the modulator itself is running in a different mode, you can still blend this LFO using the **LFO modulation** parameter available on each tabbed page. The LFO parameters themselves are available on the first tabbed page only though.

## Signal generator

**MODULATOR SHAPE** Default [Navigation icons] Random ?

Shape Sine Custom 25.0% Step 25.0% Smooth 50.0% Advanced

Normal [Waveform icons] Edit Edit

Harmonics [Waveform]

Signal generator defines the modulation LFO shape. It is used by the LFO generator, but also for the **Project** feature. Signal-generator is an incredibly versatile generator of low & high frequency signals. It offers 2 distinct modes - Normal and Harmonics. **Normal mode** is appropriate for low-frequency oscillators, where the graphical shape is relevant and is used to drive some form of modulation. For example, a tremolo uses this modulation to change the actual signal level in time. Frequencies for such oscillators usually do not exceed 20Hz as this is a sort of limit above which the frequencies become audible.

**Harmonics** mode is designed for high-frequency oscillators, where the actual shape is not as important as the harmonic content of the resulting signal, hence it is especially useful for actual audio signals. Please note that since a shape can contain more harmonics than those available from the harmonic generator, the results may not be exactly the same. As an example, a rectangular wave in normal mode may sound fuller than when converted to the harmonic mode.

Use the arrow-down button to switch from normal mode to harmonics mode or click the **Normal** and **Harmonics** buttons

## Normal mode

The generator first uses a set of predefined signal shapes (sine, triangle, rectangle...), which you can select directly by right-clicking on the editor and choosing the requested shape from the menu. This menu also provides a link to the modulator shapes preset manager, normalization and randomization. You can also use the **Main shape** parameter, which generates a combination of adjacent signals to provide a nearly inexhaustible number of basic shapes.

The engine then combines the predefined shape with a **Custom shape**, which may be anything you can draw using the advanced envelope engine, depending on the level set by the **Custom shape** control. Use the **Edit** button to edit the custom shape.

You can also combine those results with a fully featured step sequencer, with variable number of steps and several shapes for each of them, depending on the level set by the **Step sequencer** control. Use the lower **Edit** button to edit the step sequence.

Those results may be mixed with a custom sample, which is available from the advanced settings, accessed by clicking the **Advanced** button.

**Smoothness** softens any abrupt edges, generated by the step sequencer for example.

Finally there are **Advanced** features providing more complex transformations, adding harmonics etc. or you can click the **Randomize** button in the top-left corner to generate a random, but reasonable, modulator shape.

## Harmonics mode

Harmonics mode represents the signal as a series of harmonics (that is, multiples of the base frequency). For example, when your oscillator has a frequency of 2Hz (set in the **Rate** panel), then the harmonics are 2Hz, 4Hz, 6Hz, 8Hz etc. In theory, any signal can be created by mixing a potentially infinite number of these harmonics.

The harmonics mode lets you control the levels and phases of each harmonic. The top graph controls the levels of individual harmonics, while the bottom one controls their phases. Use the left-mouse button to change the values in each graph, the right-mouse button sets the default for the harmonics - 0% level and 0% phase. In both graphs the harmonics of power 2 (that is octaves) are highlighted. Other harmonics may actually sound disharmonic, despite their names.

*For example, if you reset all harmonics to the defaults and increase only the first one, you will get a simple sine wave. By adding further harmonics you make the output signal more complex.*

**Harmonics** controls the number of generated harmonics. The higher the number is, the richer the output signal is (unless the levels are 0% of course). This is useful to make the sound cleaner. For example, if you transform a saw-tooth wave to harmonics, it would not sound like a typical saw-tooth wave anymore, but more like a low-passed version of one. The more harmonics you use, the closer you get to the original saw-tooth wave.

**Generator** is a powerful tool for generating the harmonics, which are otherwise rather clumsy to edit. The generator provides several parameters based upon which it creates the entire series of harmonic levels and phases. These parameters are usually easier to understand than the harmonics themselves. Part of the generator is the randomizer available via the **Random seed** button, which smartly generates random settings for the generator. This makes the process of getting new sounds as simple as possible.

## Signal generation fundamentals

The signal generator produces a periodic signal with specified wave shape. This means that the signal is repeating over and over again. As a result it can only contain multiples of the fundamental frequency. For example, if the generator is producing 100Hz signal, then it can contain 100Hz (fundamental or 1st harmonic), 200Hz (2nd harmonic), 300Hz (3rd harmonic), 400Hz (4th harmonic) etc. However, it can never produce 110Hz. You can then control the level of each harmonic and their relative phases. It does not matter whether you use the normal mode using oscillator shapes, or harmonics mode where you can control the harmonics directly. If both modes result in the same wave shape (such as sine wave vs. 1st harmonic only), then the result is exactly the same.

Sine wave is the simplest of all as it contains the fundamental frequency only. The "sharper" the signal shape is, the more harmonics it contains. The biggest source of higher harmonics is a "discontinuity", which you can see in both rectangle and saw waves. In theory, these signals have an infinite number of harmonics. However since our hearing is highly limited to less than 20kHz, the number of harmonics which are relevant is actually pretty small. If you generate a 50Hz signal, which is very low, and assuming that you have extremely good ears and you actually hear 20kHz, then the number of harmonics audible for you is  $20000 / 50 = 400$ .

## What happens above 20kHz?

Consider the example above again, what happens with harmonics above 400? These either stay there and simply are not audible, disappear if anti-aliasing is used, or get aliased back under 20kHz in which case you get the typical digital dirt.

When you convert a rectangle wave to harmonics mode, only the first 256 harmonics are used, so it basically works like an infinitely steep low-pass filter. What is the limit then?  $50 \text{ Hz} * 256 = 12.8\text{kHz}$ . The harmonic mode will not produce anything above this limit if you are generating a 50Hz signal. Most people do not hear anything above 15kHz, so this is usually enough, but if not, you may need to use the normal mode where you get the "infinite" number of harmonics.

## What you see is not always what you get!

Say you want a rectangle wave and play a 440Hz tone(A4). You would expect the output signal to be a really quick rectangle wave, right? Wrong! If you would do that, and actually most synthesizers on the market do that, you would get the infinite number of harmonics. And, since you are working in say 48kHz sampling rate, the maximum frequency that can actually exist in your signal is 24kHz. So everything above it would get aliased below 24kHz, and there would be a lot of aliased dirt.

The "good" synthesizers perform a so-called anti-aliasing. There are several methods, most of them require quite a lot of CPU or have other limitations. The goal is to remove all frequencies above the 24kHz in our case or in reality, it is more about removing all aliased frequencies above 20kHz - this means, that we do not care about frequencies above 20kHz, because we do not hear them anyway. But we will keep it simple. Let's say we remove everything above 20kHz. You already know that the rectangle wave can be created using an infinite number of harmonics or sine waves. We removed everything above the 45th harmonic ( $20000 / 440$ ) so our rectangle wave is trying to be formed using just 45 harmonics, so it will not really look like a rectangle wave.

After some additional filtering (like DC removal), the rectangle wave may look completely different than a true rectangle wave, yet it would sound the same! Does it matter? Not really. You simply edit the shape as a rectangle wave and let the synthesizer do the ugly stuff for you. But do not check the output, because it may be very different than what you would expect ;).

## How can I generate non-harmonic frequencies?

Ok, so now you are playing a 440Hz (A4) saw wave, it contains 440Hz, 880Hz, 1320Hz etc. Anything generated using the signal generator can contain only these frequencies, the only difference is the levels and phases of each of them. What if you want to make the signal dirty by adding say 500Hz? Well, that is not that simple! Here we are getting into audio synthesizer stuff, so let us just give you a few hints.

The traditional way is to use modulation. One particular method is called frequency modulation (FM). Instead of generating a 440Hz saw wave with your generator, you change the pitch, up and down. You are modulating the frequency, that's why FM. It is basically a vibrato, but as you increase the speed of the vibrato, it gets so quick that you stop noticing the pitch changes (that's very simplified but it serves the purpose) and instead it starts producing a very complex spectrum. Will the 500Hz be there? Well, if setup correctly, yes, but there will also be lots of other non-harmonic frequencies.

Another way is possible without any other tools. Let's say you do not want 440Hz, but 660Hz. Then you may generate 220Hz instead of 440Hz (which is one octave below it) and voila, 660Hz is the 3rd harmonic ( $3 \times 220$  is 660)! But you need to shift the saw wave one octave above. Fortunately it is not that hard here - go to the normal mode, select saw tooth, click advanced, and use the harmonics panel to remove the fundamental and leave just the 2nd harmonic, then convert it to harmonic mode. Well, it's not that hard, but it's not exactly simple either...

The only way is, of course, additive synthesis. In that case you do not use one oscillator, but many of them. It lets you generate just about anything. But there is a catch, actually many of them. First, you need to say "ok I want this frequency and that frequency...", the setup is actually infinitely hard as there may be an infinite number of frequencies :). And the second is, of course, CPU requirements.

So is there some ultimate solution? Nope, sorry. The good thing is, you will not probably need it, because while what you see is not always what you get, also what you want is often not what you really want to hear :).

 Default

### 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.

Random

## Random

Random button generates random settings using the existing presets.

Normal

## Normal

Normal button switches the generator into the normal mode, which lets you edit the shape of the oscillator. This is especially advantageous for low-frequency oscillators, where the shape matters even though it doesn't have any physical meaning.



## Convert

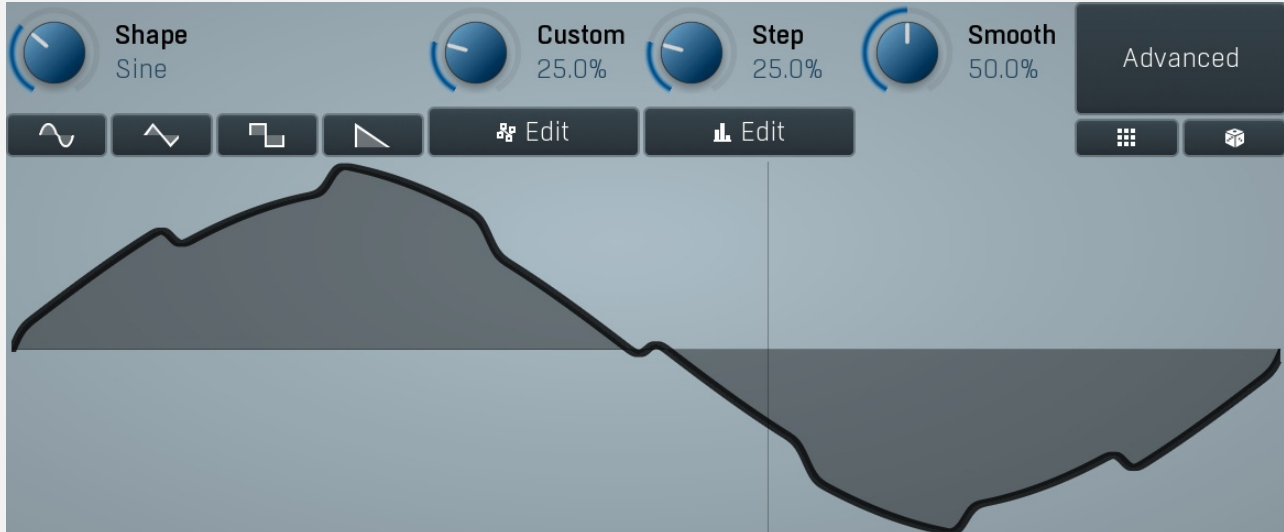
Convert button converts the current shape into harmonic-based representation. Please note that since the number of harmonics is limited, the result will not perfectly resemble the original shape.

Harmonics

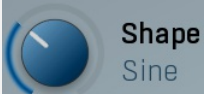
## Harmonics

Harmonics button switches the generator into the harmonics mode, which lets you edit the levels and phases of individual harmonics. This is especially advantageous for high-frequency oscillators, hence sound generators.

## Signal generator in Normal mode



Signal generator in Normal mode works by generating the oscillator shape using a combination of several curves - a predefined set of standard curves, custom shape, step sequencer and custom sample. It also post-processes the shape using several filters including smoothing to custom transformations. This is especially useful when using the oscillator as an LFO (low-frequency-oscillator), where the harmonic contents does not really matter, but the shape does.



## Shape

Shape controls the main shape used by the signal generator. There are several predefined shapes: exponential, triangle, sine power 8, sine power 4, sine square, sine, harmonics, more harmonics, disharmonics, sine square root, sine 4 root, rectangle, rect-saw, saw, noise and mess. You can choose any of them or interpolate between any 2 adjacent shapes using this control.



## Custom

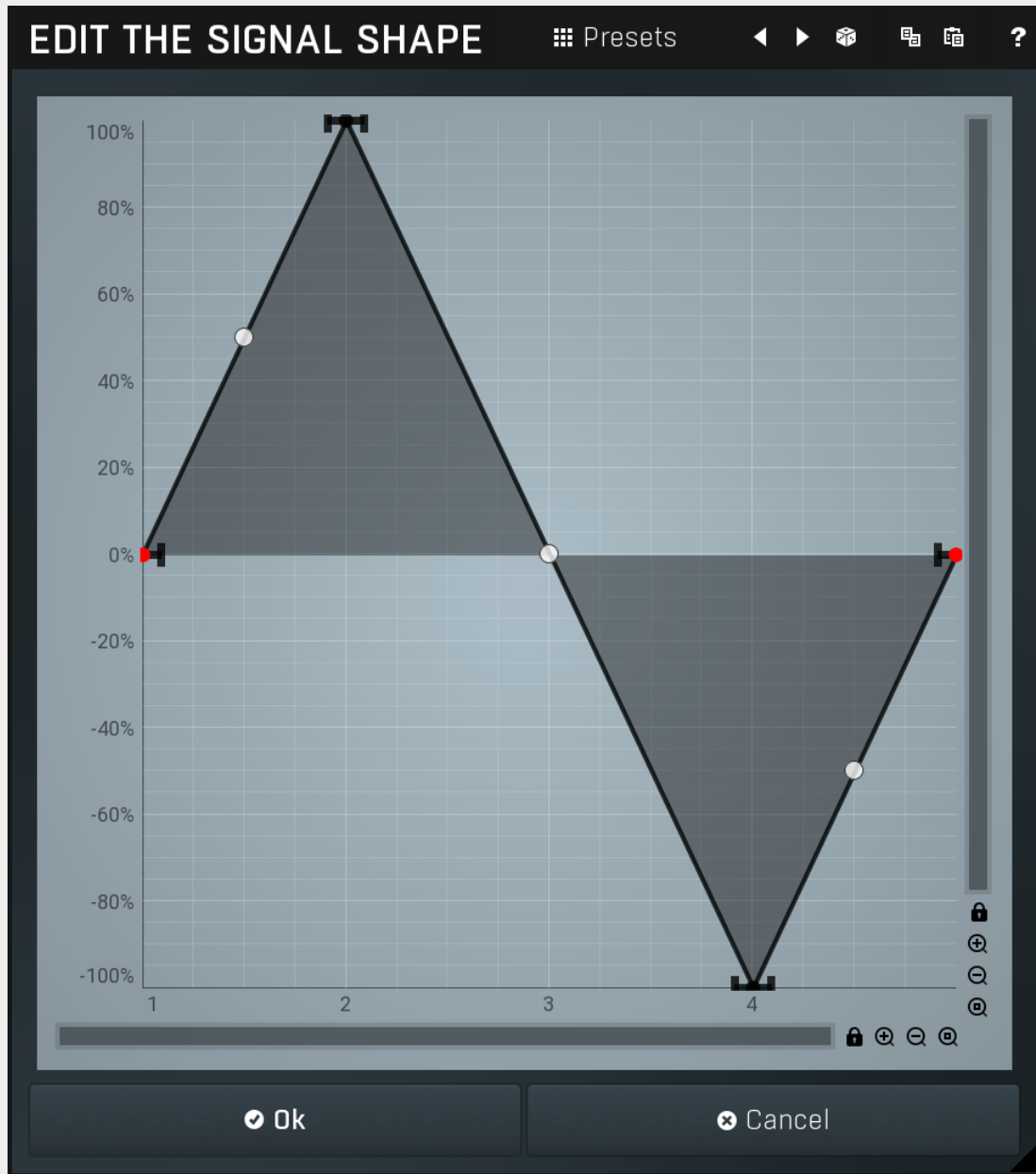
Custom controls the amount of the custom shape that is blended into the main shape.

Edit

## Edit

Edit button shows the custom shape editor.

### Signal generator custom shape editor



Signal generator custom shape editor controls the custom shape. You can edit virtually any shape that you can imagine and then blend it with the standard shapes, the step sequencer etc.

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.



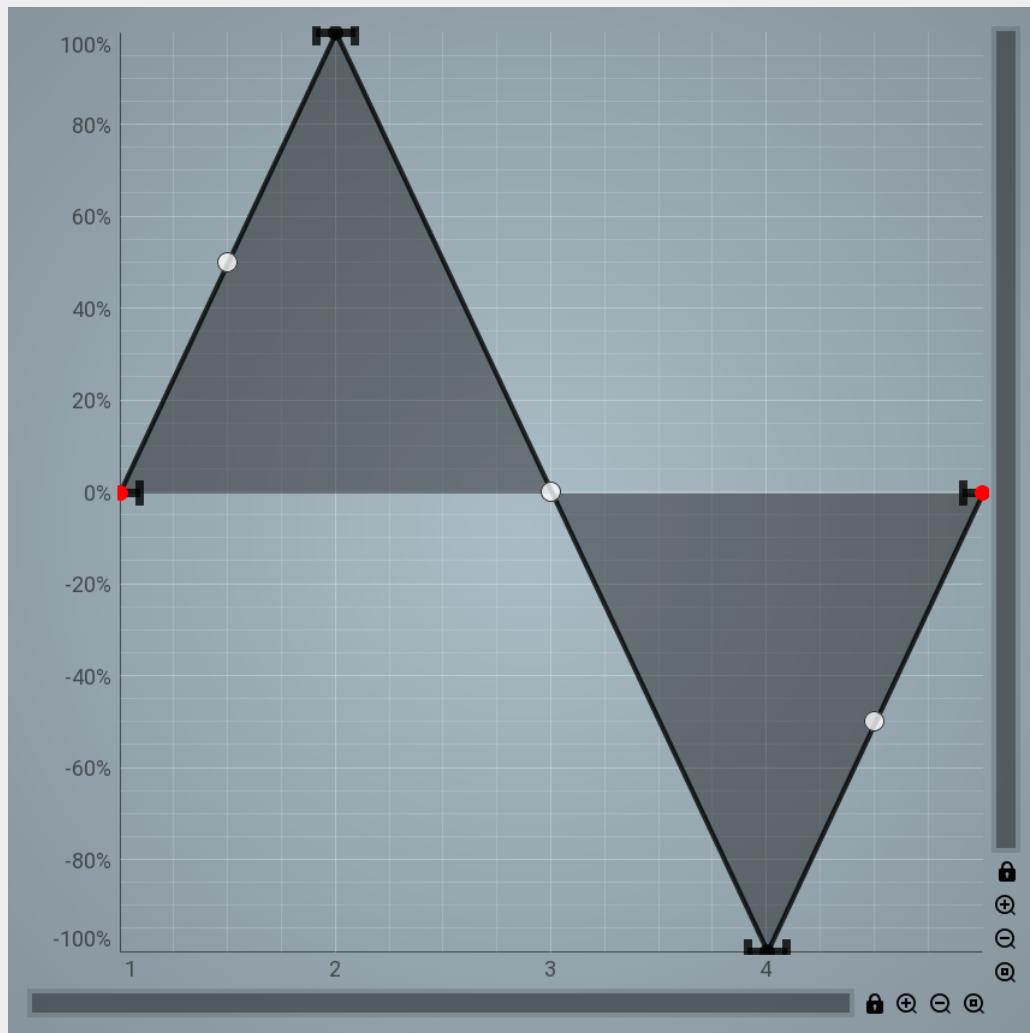
### Copy

Copy button copies the settings onto the system clipboard.



## Paste

Paste button loads the settings from the system clipboard.



Graph editor

Graph editor lets you edit the envelope graph.

## Envelope graph

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

- **Left mouse button** can be used to select points. If there is a *point*, you can move it (or the entire selection) by dragging it. If there is a *curvature circle*, you can set up its tension by dragging it. If there is a *line*, you can drag both edge points of it. If there is a *smoothing controller*, you can drag its size. Hold **Shift** to drag more precisely. Hold **Ctrl** to create a new point and to remove any points above or below.
- **Left mouse button double click** can be used to create a new point. If there is a *point*, it will be removed instead. If there is a *curvature circle*, zero tension will be set. If there is a *smoothing controller*, zero size will be set.
- **Right mouse button** shows a context menu relevant to the object under the cursor or to the entire selection. Hold **Ctrl** to create or remove any points above or below.
- **Middle mouse button** drag creates a new point and removes any points above or below. It is the same as holding Ctrl and dragging using left mouse button.
- **Mouse wheel** over a point modifies its smoothing controller. If no point is selected, then all points are modified.
- **Ctrl+A** selects all points. **Delete** deletes all selected points.



Step  
25.0%

## Step

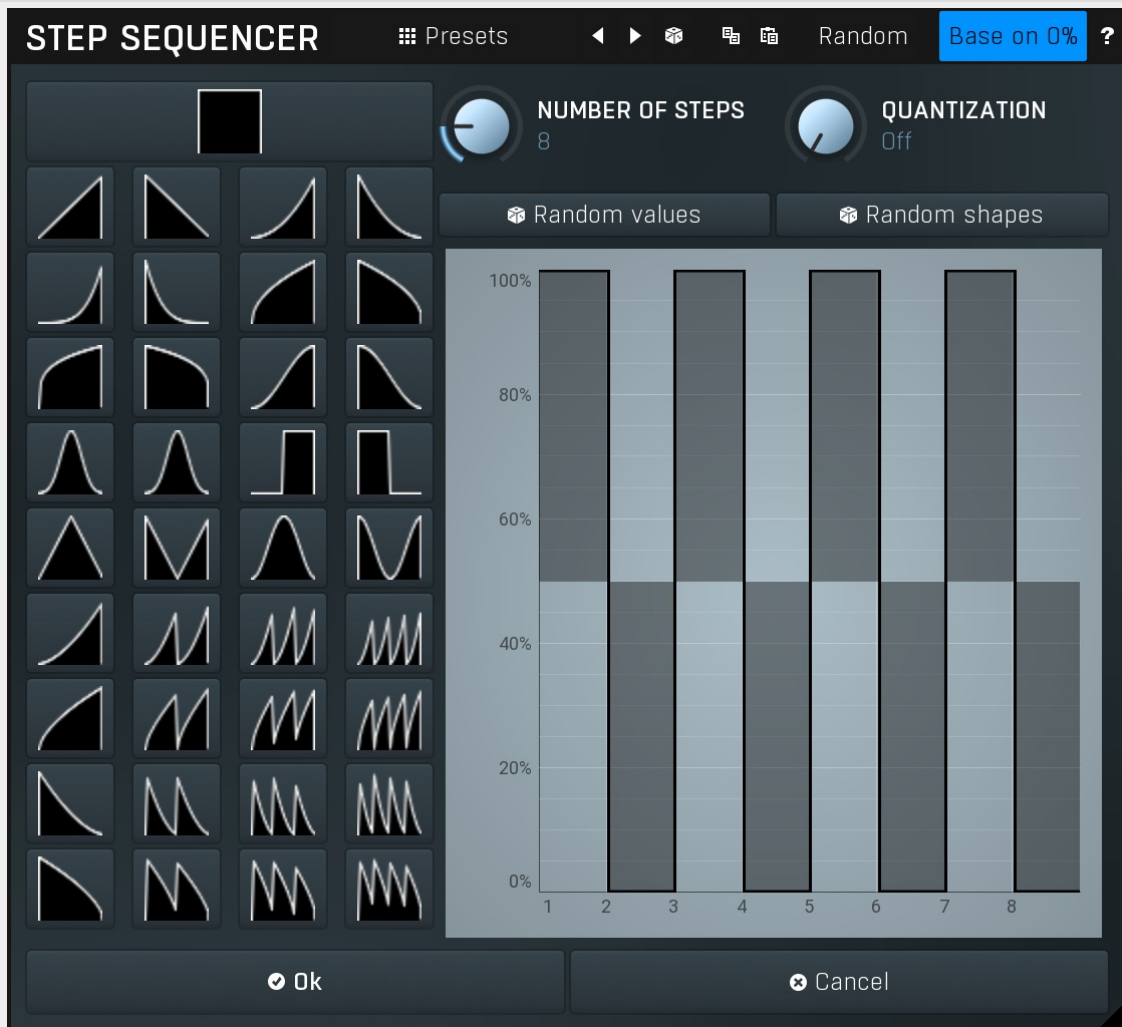
Step controls the amount of the step sequencer shape that is blended into the main shape (which has already been blended with the custom shape).

⬆ Edit

## Edit

Edit button shows the step sequencer editor.

### Signal generator step sequencer editor



Signal generator step sequencer editor controls the step sequencer shape. You can have various numbers of steps each with a different value and shape. Note that for classic rectangular shapes the output can be very rough, hence it may be worth considering using **Smoothness** parameter to smooth out the resulting shape. This will use additional CPU power of course, but that should be negligible unless you modulate any of the signal generator parameters.

⌘ 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.



## Random

### Random

Random button generates random settings using the existing presets.

#### Random values

### Random values

Random values button generates random sequence of values, but keeps the shape of each step.

#### Random shapes

### Random shapes

Random shapes button generates random sequence of shapes, but keeps the values of each step.



## Smooth

50.0%

### Smooth

Smooth controls the amount of smoothing. Many shapes, especially those produced by the step sequencer, have rough jagged edges, which may be advantageous, but when used to modulate certain parameters, the output may be clicking or causing other artifacts. Smoothness helps it by smoothing the whole signal shape out and removing these rough edges.

## Advanced

### Advanced

Advanced button displays an additional window with more advanced settings for post-processing the signal shape, such as harmonics or custom transformations.

## Advanced settings

Order	Amplitude	Phase
1	100.0%	0° (0%)
2	0.00%	0° (0%)
3	0.00%	0° (0%)
4	0.00%	0° (0%)
5	0.00%	0° (0%)
6	0.00%	0° (0%)
7	0.00%	0° (0%)
8	0.00%	0° (0%)

## 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.



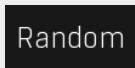
### Copy

Copy button copies the settings onto the system clipboard.



### Paste

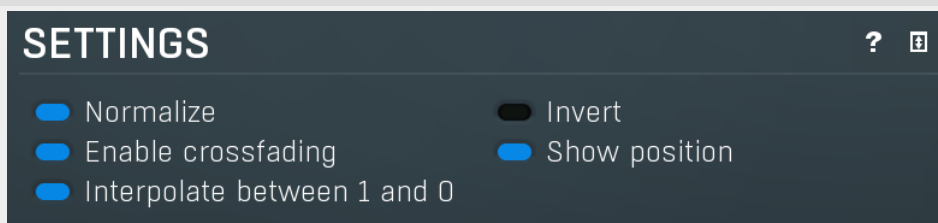
Paste button loads the settings from the system clipboard.



### Random

Random button generates random settings using the existing presets.

## Settings panel



Settings panel contains some global settings of the oscillator.



Normalize

### Normalize

Normalize switch enables normalization to  $-1..+1$ . It is generally desirable since even if you draw a custom shape, you usually want it to have the full range. You may want to disable it if you want to create some custom shapes, where the level actually matters.



Invert

### Invert

Invert switch simply inverts the output shape vertically.



Enable crossfading

### Enable crossfading

Enable crossfading enables interpolation between shapes when the shape is changing. This requires more CPU, but can avoid zipper noise when the shape is being modulated for example.



Show position

### Show position

Show position makes the editor display a position indicator.

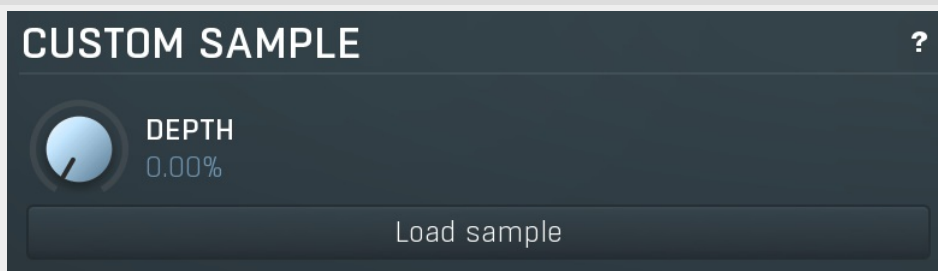


Interpolate between 1 and 0

### Interpolate between 1 and 0

Interpolate between 1 and 0 smoothens the discontinuity between 1 and 0 values, which is inevitable for shapes such as saw or rect for example. However when this is a high frequency oscillator (HFO), this discontinuity is what creates the highest frequencies, so it is actually desirable. When using it as an LFO, you may also want the discontinuity in some extreme cases.

## Custom sample panel



Custom sample panel contains parameters of the custom sample that you can load and mix with the other sources. Do NOT confuse this with a sampler, the custom sample is taken as one period of the waveform. It can be used for creative effects

and it can be used to import a custom waveform. The custom sample is then stored with limited precision within the settings, so the sample does not need to be kept on the system, but note that these settings may be quite large. To limit the space required by the settings, the sample is stored only if the depth is not 0%, meaning only if the sample is actually used.

A circular knob with a blue needle pointing to the 0% mark. The text "DEPTH" and "0.00%" is displayed to the right of the knob.

### Depth

Depth controls the amount of custom sample mix. 0% means the sample is not used even if there actually is one loaded. 100% means the sample completely overrides the basic shape, custom shape, step sequencer... However, transformations are still performed on the sample.

A dark rectangular button with the text "Load sample" in white.

### Load sample

Load sample button displays a file selection window, which lets you select the custom sample file.

## Shape panel

A dark panel titled "SHAPE" with a question mark icon in the top right. It contains four knobs: "PHASE" at 0° (0%), "SKEW" at 0.00%, "FULL SKEW" at 0.00%, and "HALVE" at 0.00%.

Shape panel contains parameters performing various transformations on the signal shape. Please note that most transformation require a significant amount of CPU resources, so you should not automate or modulate the signal shape if you are using them.

## Harmonics panel

A dark panel titled "HARMONICS" with a question mark icon in the top right. It contains a table with 8 rows and 3 columns. The first column is the harmonic number (1-8), the second is the amplitude percentage, and the third is the phase angle.

Harmonic	Amplitude	Phase
1	100.0%	
2	0.00%	0° (0%)
3	0.00%	0° (0%)
4	0.00%	0° (0%)
5	0.00%	0° (0%)
6	0.00%	0° (0%)
7	0.00%	0° (0%)
8	0.00%	0° (0%)

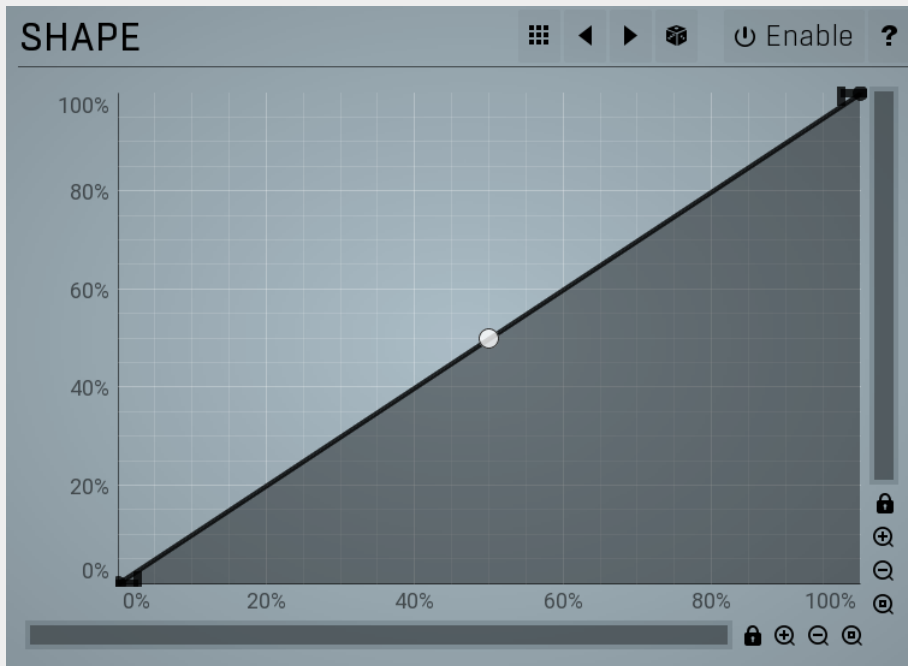
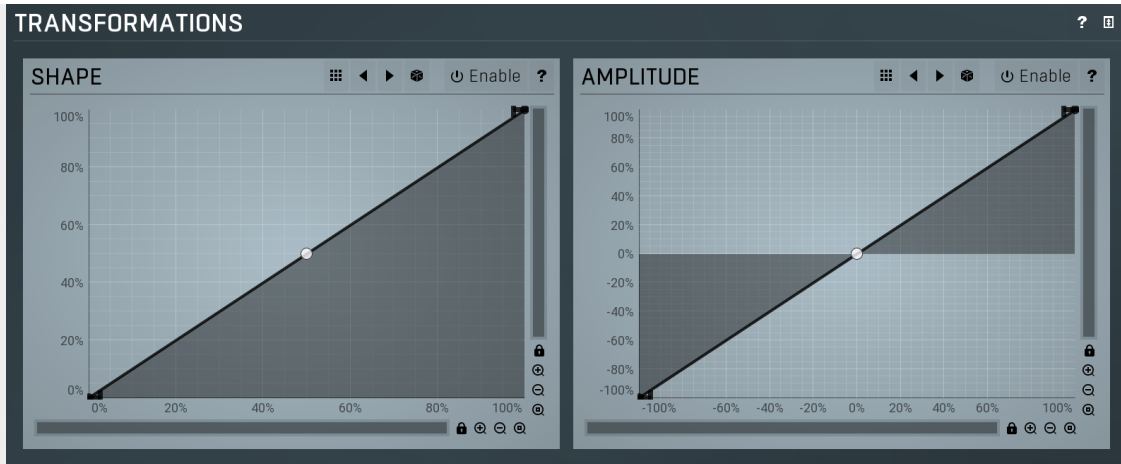
Harmonics panel lets you add separate harmonics of the original signal.

## Post-processing panel

A dark panel titled "POST-PROCESSING" with "Wrap" and "Enable" (with a power icon) and a question mark icon in the top right. It contains two knobs: "OFFSET" at 0.00% and "SCALE" at 100.0%.

Post-processing panel lets you post-process the shape after all the previous generator items.

## Transformations



**Shape transformation**

## graph

Shape transformation graph lets you perform arbitrary modification of the graph shape. Basically this graph lets you modify the shape "in time". The Y axis represents the position in the source signal related to the position in the target signal. The best way to check what it does is simply to try it.



### 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.



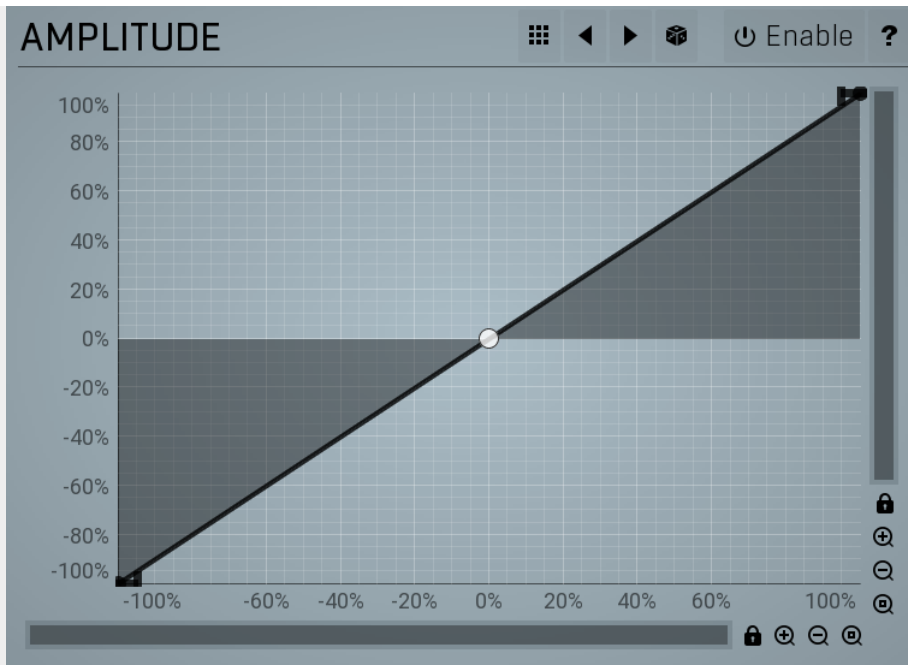
### Right arrow

Right arrow button loads the next preset.



### Randomize

Randomize button loads a random preset.



**Amplitude**

### transformation graph

Amplitude transformation graph lets you perform arbitrary modification of the graph amplitude. Basically this graph lets you modify the shape's level, vertical axis. The X axis represents the original values, the Y axis defines the resulting values. The best way to check what it does is simply to try it.

#### 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.

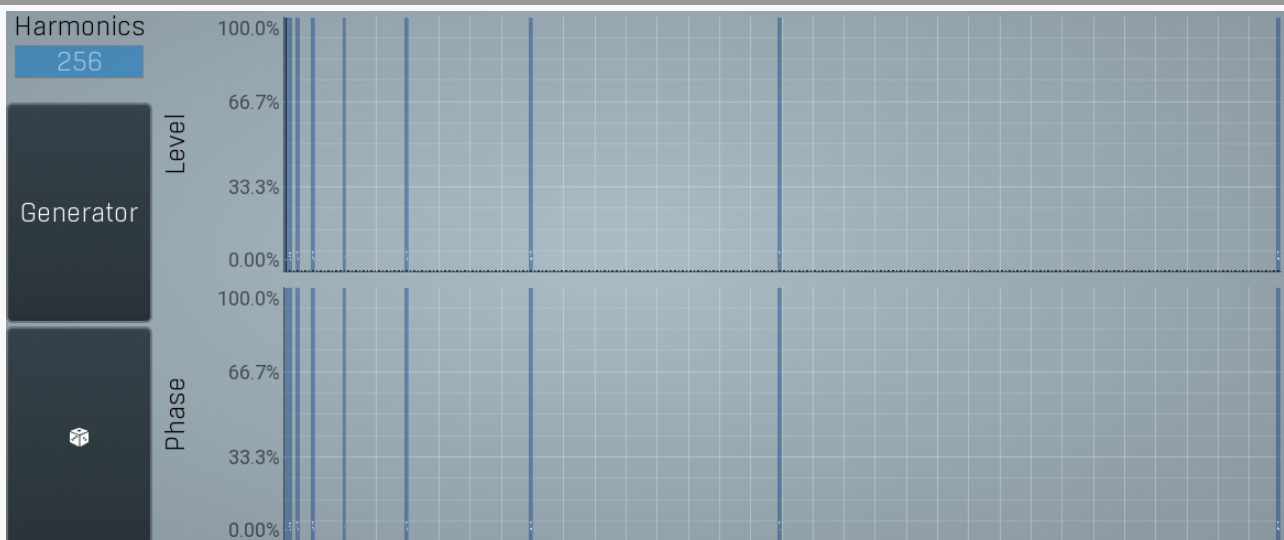
#### Right arrow

Right arrow button loads the next preset.

#### Randomize

Randomize button loads a random preset.

## Signal generator in Harmonics mode



Signal generator in Harmonics mode works by generating the oscillator shape using individual harmonics. Essentially a harmonic is a sine wave. The first harmonic, known as the fundamental, fits once in the oscillator time period, hence it is the same as selecting sine wave in the **Normal mode**. The second harmonic fits twice, the third three times etc. In theory, any shape you create in normal mode can be converted into harmonics. However, this approach to signal generation needs an enormous number of harmonics, which is both inefficient to calculate and mostly hard to edit. Therefore, the harmonic mode can process up to 256 harmonics, which is enough for

very complex spectrums, however it is still not enough to generate an accurate square wave for example. If your goal is to create basic shapes, it is better to use the normal mode.

It is nearly impossible to say how a particular curve will sound when used as a high-frequency oscillator in a synthesizer, just by looking at its shape. Harmonics mode, on the other hand, is directly related to human hearing and makes this process very simple. In general, the more harmonics you add, the richer the sound will be. The higher the harmonic, the higher the tone. Usually, one leaves the first harmonic enabled too, as this is the fundamental tone, however you may experiment with more dissonant sounds without it.

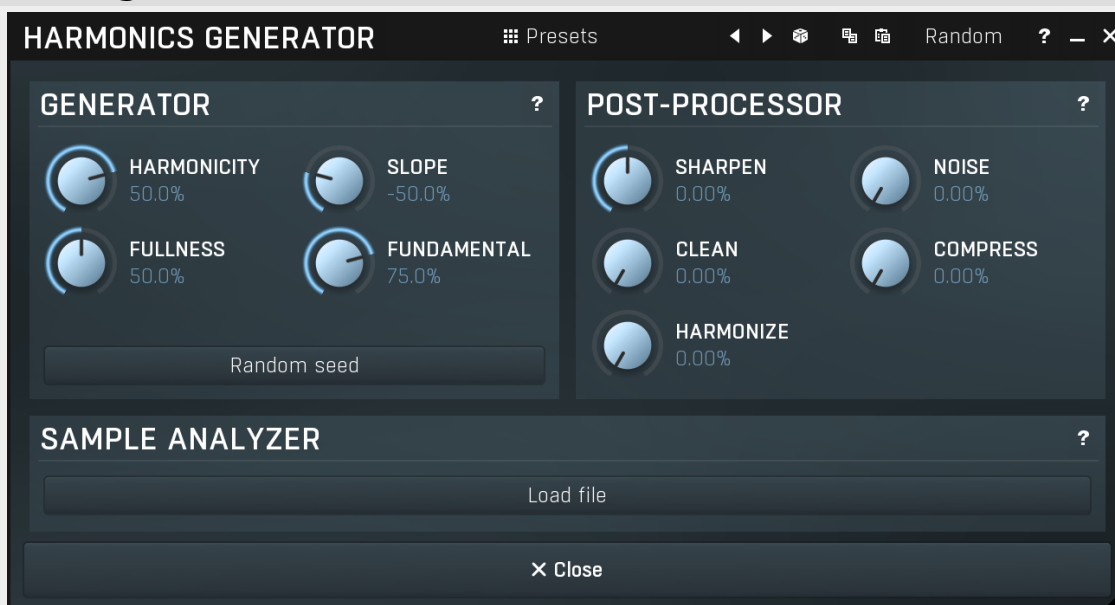
Editing harmonics can be time consuming unless you hear what you want, so a signal generator is also available. This great tool lets you generate a random spectrum by a single click. You can also open the **Generator** settings and edit its parameters, which basically control the audio properties in a more natural way - using parameters such as complexity, harmonicity etc.

Generator

## Generator

Generator button shows a powerful harmonics generator, which can create unlimited number of various timbres and even analyze a sample and extract harmonics from it.

### Harmonics generator



Harmonics generator is a powerful tool, that can generate various harmonics-based timbres and even analyze a sample file and extract harmonics from it.

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.

Random

## Random

Random button generates random settings using the existing presets.

## Generator panel



Generator panel contains parameters of the harmonics generator. By changing any of the parameters, the harmonics are changed, however only **Random seed** button changes the structure completely. The other parameters can be used to tweak the results.



### Harmonicity

Harmonicity controls the ratio between natural harmonics and those which sound disharmonic (despite the title "harmonics"). Assuming that the 1st harmonic is the fundamental, 2nd harmonic is 1 octave above, 4th is 2 octaves above, both can be considered very natural. 3rd harmonic is 1 octave and a 5th above the fundamental, and is still pretty harmonic, but less than the octaves. 5th harmonic is 2 octaves and a major 3rd above the fundamental. Such a tone may sound very disharmonic, in minor scales for example. Higher harmonics are often very disharmonic and produce typical ringing timbres. When harmonicity parameter is set to 100%, only octaves are allowed. By lowering the value more and more disharmonics are created and with 0% all frequencies are allowed. For values below 0% disharmonics are preferred, hence you can expect more ringing timbres.



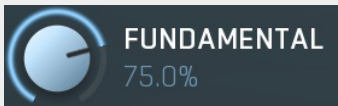
### Slope

Slope defines the amount of higher harmonics compared to lower ones. When 0%, the higher harmonics have the same levels as lower ones. Typically you use values below 0%, which attenuates the higher harmonics making the resulting sound darker. Similarly values above 0% make the sound brighter.



### Fullness

Fullness controls the number of generated harmonics. With values around 0% the resulting timbres will contain only a few harmonics making the sound clear. Higher values increase number of harmonics making the timbre rich.



### Fundamental

Fundamental controls the minimum level of the fundamental (the 1st harmonic). Most sounds have a very strong fundamental as it carries the pitch.

Random seed

### Random seed

Random seed button generates a new series of harmonics. Pressing this button will create a whole new timbre.

## Post-processor panel

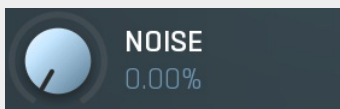


Post-processor panel contains parameters of the harmonics post-processor. The generator and sample analyzer first create a series of harmonics, the timbre. These harmonics are mixed depending on the **Sample ratio** parameter. After that the post-processor is engaged, which can further transform the harmonics in several ways.



### Sharpen

Sharpen is a sort of soft compression/expanding unit. Values below 0% decrease the level of quiet harmonics, while values above 0% increase their level.



### Noise

Noise defines amount of noise added to the timbre. Noise can make the results dirty providing much richer timbres.



### Clean

Clean controls the threshold of a gate. It basically attenuates or removes harmonics below this level making the output cleaner.



### Compress

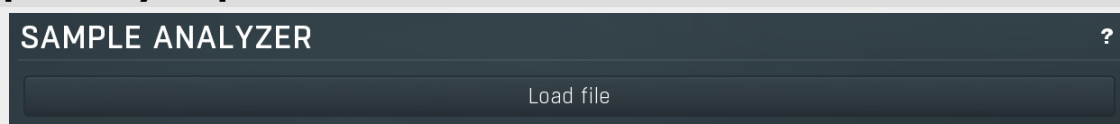
Compress reduces the dynamic range of the harmonics by increasing levels of the quiet ones, but keeping the levels of the loud ones.



### Harmonize

Harmonize creates additional higher harmonics from existing ones. This is especially useful to transform rich dirty disharmonic timbres into similarly rich but more harmonic timbres.

## Sample analyzer panel



Sample analyzer panel contains parameters of the sample analyzer. If there is no sample loaded, the sample analyzer is turned off. The analyzer takes the selected sample and a position within it, analyses one period of the signal waveform and produces the output set of harmonics. You can then combine these harmonics with the output of the generator using **Sample ratio** parameter.

The sample itself is not store with the plugin settings. Instead the path to the target sample file is stored along with the analyzed harmonics. If the sample file is not available, you cannot modify the analysis parameters and the last analyzed harmonics are used. This means that you actually don't need to have the sample file available on the computer on which you are using the settings.



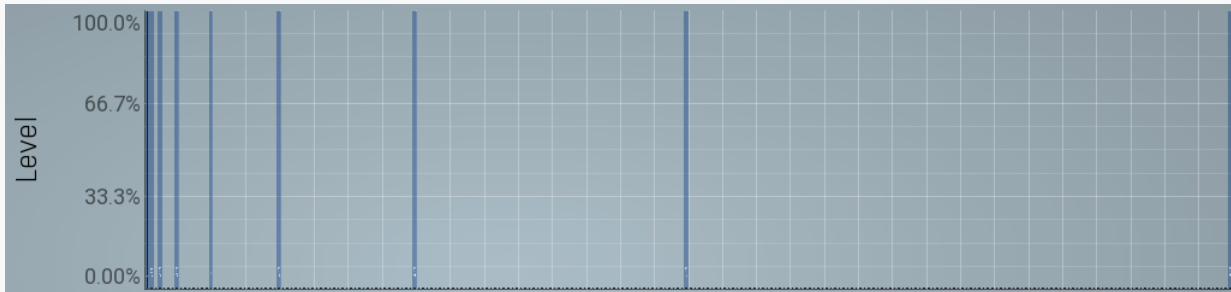
Load file button lets you select a sample file to analyse.





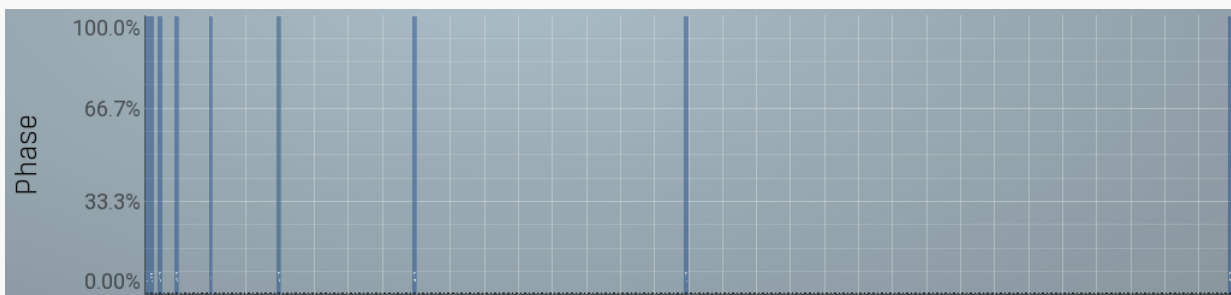
## Randomize

Randomize button selects random parameters for the harmonics generator, so you can use it to get a random sound character instantly. Hold **Ctrl** to slightly modify existing generator settings instead of completely changing them.



### Magnitudes graph

Magnitudes graph contains the levels of the individual harmonics. The highlighted bars are octaves, thus the 1st, 2nd, 4th, 8th harmonic etc.



### Phases

### graph

Phases graph contains the phases of the individual harmonics. The highlighted bars are octaves, thus the 1st, 2nd, 4th, 8th harmonic etc.

## Rate panel

**RATE** Sync ?

Frequency

Sync group  1 2 3 4

Reset on ARP

Length   Type

Phase  Count

Rate panel contains parameters controlling the speed of the LFO, whether the modulator is set to **Normal** mode or any other mode while the **LFO modulation** is used.

Frequency

### Frequency

Frequency defines the modulation speed.

Sync group  1 2 3 4

### Sync group

Sync group lets you synchronize the modulators with each other and potentially with other parts of the plugin. It can be controlled only when **to-host synchronization** is disabled, otherwise it is overridden by synchronization from the host. By using the same synchronization group for all modulators you ensure they will always be in-sync even though no other synchronization is used. This can be useful, for example, when you want to modulate different parameters with different shapes or when using some more advanced method, such as using a follower. When the synchronization is enabled, it works on the 'first is the leader' basis, hence the first modulator controls the rest of the modulators in the same group.

Reset on ARP

### Reset on ARP

Reset on ARP makes the LFO phase reset every time the ARP position jumps to the beginning. This is useful if you want to make

the ARP and LFO in sync in a way.

## Synchronization panel

Length	1 / 2	◀ ▶	Type	Straight	◀ ▶	Set frequency
Phase	90° (25.0%)		Count	1		

Synchronization panel contains parameters for the to-host synchronization.

Length	1 / 2	◀ ▶	<b>Length</b>
--------	-------	-----	---------------

Length defines the note length to be used.

Type	Straight	◀ ▶	<b>Type</b>
------	----------	-----	-------------

Type defines the note type, such as straight notes or triplets, to be used. Together the **Length** and **Type** determine the actual time/delay.

*Example: '1/4 Straight' at 120 bpm = a delay of 500 ms, '1/4 Triplet' at 160 bpm = a delay of 281.25 ms.*

Phase	90° (25.0%)	<b>Phase</b>
-------	-------------	--------------

Phase defines the phase offset of the to-host synchronization.

Range: 0° (0%) to 360° (100.0%), default 90° (25.0%) <

Count	1	<b>Count</b>
-------	---	--------------

Count defines the number of the units, hence multiplies of the sync length.

Range: 1 to 64, default 1

Set frequency	<b>Set frequency</b>
---------------	----------------------

Set frequency button sets the **Frequency** parameter available for the frequency mode so that it matches the current synchronization. That way you can set the modulator's frequency to the current synchronization and then change it a little for example.

## MIDI reset panel

MIDI RESET - (RE)TRIGGER					🔌 Enable	?	🗄
<input checked="" type="checkbox"/> Note-on	<input type="checkbox"/> Note-on only first	<input type="checkbox"/> Note-off	<input type="checkbox"/> Note-off only last	<input type="checkbox"/> Single shot	<input checked="" type="checkbox"/> Single shot reset		
Min velocity	0.00%		Max velocity	100.0%			
Min note	0 (C-1)		Max note	100 (G7)			
Phase	0° (0%)		Channel	All			◀ ▶

MIDI reset panel configures the MIDI reset feature, which will reset the oscillator when a MIDI note is received or its **MIDI reset parameter** is a target of another modulator or multiparameter. This way you can make the oscillator perform "in-sync" with your playing. Please note that once you enable it, the oscillator will not be in phase-sync with the host.

🔌 Enable	<b>Enable</b>
----------	---------------

Enable button enables or disables the feature.

<input checked="" type="checkbox"/> Note-on	<b>Note-on</b>
---	----------------

Note-on controls if the MIDI reset should occur when a note is pressed.

<input type="checkbox"/> Note-on only first	<b>Note-on only first</b>
---	---------------------------

Note-on only first controls if the MIDI reset should occur when a note is pressed only if it is the first note (thus no other note is being held).

<input type="checkbox"/> Note-off	<b>Note-off</b>
-----------------------------------	-----------------

Note-off controls if the MIDI reset should occur when a note is released.

<input type="checkbox"/> Note-off only last	<b>Note-off only last</b>
---	---------------------------

Note-off only last controls if the MIDI reset should occur when a note is released only if it is the last note (that is, no other note is being held afterwards).

Min velocity	0.00%	<b>Min velocity</b>
--------------	-------	---------------------

Min velocity defines the minimum velocity that will reset the oscillator.

Max velocity	100.0%	<b>Max velocity</b>
--------------	--------	---------------------

Max velocity defines the maximum velocity that will reset the oscillator.

Min note

0 (C-1)

**Min note**

Min note defines the minimum note that will reset the oscillator.

Max note

111 (F7)

**Max note**

Max note defines the maximum note that will reset the oscillator.

Phase

0% (0%)

**Phase**

Phase defines the initial oscillator phase after a reset.

Channel

All

**Channel**

Channel defines note MIDI channel to reset the oscillator.

Single shot

**Single shot**

Single shot button activates the single shot mode in which the LFO doesn't cycle around but instead only goes once from left to right, then stops until the MIDI reset occurs.

Single shot reset

**Single shot reset**

Single shot reset button defines if the phase should reset to 0 after a single shot period ends. For most waves such as sine it doesn't really matter since the value at 0 (the start of the cycle) is the same as value at 1 (the end of the cycle). But it might matter for saw wave for example.

## Follower mode

The screenshot displays a software interface with several control sections:

- LFO MODULATION:** A knob set to 0.00%.
- LEVEL MIN:** A knob set to -50.00 dB.
- LEVEL MAX:** A knob set to 0.00 dB.
- Input from feedback:** A dropdown menu set to "Off".
- DETECTOR:**
  - Side-chain ? ⓘ
  - ATTACK: 10 ms
  - RELEASE: 100 ms
  - HOLD: 0 ms
  - RMS length: 10 ms
  - Pitch modulation: 0.00%
  - Transient mode: Enhanced ◀ ▶
  - Delay: 0 ms
  - Transient modulation: 0.00%
  - Release mode: Manual ◀ ▶
  - Advanced detector settings
- BANDPASS ? ⓘ:**
  - Min: Off
  - Max: Off
  - Q: 0.7071
- PROJECTION ⏻ ? ⓘ:**
  - Phase: 270° [75.0%]
  - Interval size: 180° [50.0%]
- ADVANCED ? ⓘ:**
  - Mode: Linear ◀ ▶
  - Maximize: -32.0 dB
- LEVEL ? ⓘ:**
  - Graph showing signal level in dB (left axis, -80 to 0) and percentage (right axis, 0 to 100).
  - Level Max is at 0 dB.
  - Level Min is at -50 dB.
  - Graph controls: +, -, ||, ▶▶, 🔍, 🔒, 🔍, 🔍.

Follower mode makes the modulator follow the input signal level.

### LFO modulation

LFO modulation defines the amount of LFO modulation to be applied in addition to the follower. With 0% the modulator uses only the follower; with 100% the modulator does the same job as if the modulator were in **Normal** mode. To set the LFO parameters switch to normal mode temporarily.

### Level min

Level min defines the minimum input level that is transformed into a modulator value of 0%. For example if you set the minimum / maximum levels to -50dB ... -20dB, then an input level of -50dB or lower results in a value of 0% and an input level at -20dB or above results in 100%.



LEVEL MAX  
0.00 dB

### Level max

Level max defines the maximum input level that is transformed into a modulator value of 100%. For example if you set the minimum / maximum levels to -50dB ... -20dB, then an input level of -50dB or lower results in a value of 0% and an input level at -20dB or above results in 100%.

## Detector panel



Detector panel contains the dynamic detector parameters, which control how the signal level is measured.

### Side-chain

#### Side-chain input

Side-chain input makes the modulator analyze the side-chain input instead of the regular input.



ATTACK  
10 ms

### 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.*



RELEASE  
100 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.*



### Peak hold

Peak hold defines the time that signal level detector holds its maximum before the release stage is allowed to start. As an example, you can imagine that when an attack stage ends there can be an additional peak hold stage and the level is not yet falling, before the release stage starts. This is true only when **true peak** mode is enabled (check the advanced detector settings if available).

It is often used in **gates** to avoid the gated level falling below the threshold too quickly, while having short release times. If you want the gate to close quickly, you need a short release time. But in that case the ending may be too abrupt and even cause some distortion. So you use the peak hold to delay the release stage.

It is also used along with **look-ahead** to avoid distortion in **limiters and compressors**. If you need a very short attack, the attack stage may be too quick and cause distortions. In limiters this attack time is often 0ms, in which case it becomes a clipper. Setting look-ahead and peak hold to the same value will make the detector move ahead in time, so that it can react to attack stages before they actually occur and yet hold the levels for the actual signal to come.



### 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 is 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.



### Delay

Delay defines how much the follower output should be delayed. It is a powerful way to keep attacks intact for example.



### Pitch modulation

Pitch modulation lets you employ the pitch detector (configurable from the **Pitch tab**) in the detector. This may sound odd at first, but thinking of the input signal, you may measure its level, but you can also measure other properties, such as its pitch, and use them in exactly the same way. While an input level is usually understood as a value in decibels, pitch is a frequency in Hz, so the plugin smartly transforms the frequency to mimic the level axis. When you look at the detector graph afterwards, you can hardly tell the exact pitch in Hz, but that's not really relevant or necessary.

What is this for? Let's show it with an example. Let's say you have an instrument, say a bass, which is playing legato, and you want some kind of effect at the beginning of the note (in case of **Follower** mode) or you just want to restart some kind of filter at the beginning of each note (in case of **Envelope** mode). But since the performance is legato, meaning there are no gaps in between the notes, the level graph is just a steady horizontal line, which is pretty much useless for us. The pitch modulation lets you replace this horizontal line with something much more useful - the pitch. While the level isn't changing much at all, the pitch is changing. The plugin then takes the actual pitch as the input signal, so you can let the plugin follow it in some way, start envelopes when a

certain pitch is exceeded, or using **Transformation** you can even let the plugin restart an envelope every time the pitch changes. By setting the pitch modulation half-way you can let the plugin react to both properties, the level and the pitch.

Transient modulation 0.00% **Transient modulation**

Transient modulation lets you detect transients and blend that detection with the control signal. This way you may let the modulator be controlled not by level (alone or in combination with pitch for example), but also by transients detected in either of these properties - level or pitch.

Transient mode Enhanced ◀ ▶ **Transient mode**

Transient mode controls the way in which the transients are detected. These simply provide different results, so you should just try the alternative modes if the default one doesn't suit your audio material. **Attack only modes** ignore sustain transients - those moments when the level decreases.

Release mode Manual ◀ ▶ **Release mode**

Release mode defines how the plug-in performs when decreasing level. In **manual mode** this is based only on the **release time**, which is suitable for most cases when the signal has constant characteristics. Automatic release modes can adapt to signals with unstable characteristics.

**Automatic** and **Automatic fast** modes: the longer the level stays above the threshold, the longer the release time will be and thus, the longer it will take to move below the threshold and end the release stage. The idea is that if the input is loud for some time, it will most likely stay that way for some more time, hence it should be stabilized to avoid unnecessary temporary fluctuations, which could result in pumping.

Both automatic modes increase the release time when the input signal is above the threshold and vice versa. The speed of the increase depends on the **Auto speed** parameter. Automatic fast mode uses full speed immediately after crossing the threshold, automatic mode varies the speed according to the current signal level.

*For example, when a guitarist plays softly, the level is low and fluctuates around the threshold and the release time gets slower. So the processor quickly responds to sudden changes. However, when the guitarist starts playing a solo, the level rises and, the longer the solo is, the longer the release time becomes, hence the response becomes slower avoiding unnecessary fluctuations (pumping) when the solo contains small silent sections.*

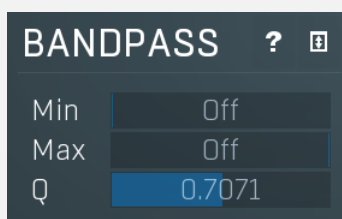
**Linear 1** and **Linear 2** modes: the higher the level is, the longer the release. The idea is that if the input is very loud, it will probably stay that way for some time, so it is wise to keep the levels up too. This is similar to the automatic modes, however the main factor is not how long the level is high, but how high it is.

Below the threshold the release time is the same as the attack time, above the threshold the release time rises from the attack time up to the specified release time parameter. Linear 1 mode usually provides higher release times than does Linear 2.

**Opto** mode: the higher the level is, the shorter the release. So this is kind of the opposite of linear modes. The idea is, that you are expecting short transients, which you wish to deal with. Normally the higher the level would get in such a transient, the longer it would take to get the level below the threshold, so, when used in a compressor for example, these transients would cause unnecessary compression in the sustain stage. The opto detector lowers the level quickly, minimizing the amount of compression in the sustain stage.

*For example, let's say you are compressing a full drumset, but there is a very dominant sharp and short hi-hat sound, so it is appropriate to have short release times. You would use **Opto** mode. But the rest of the drumset deserves a softer treatment, so you want to keep longer release times. Use one of the other modes.*

## Band-pass panel



Band-pass panel contains parameters of the follower band-pass filter. Using this feature you can make the follower detect the level of just part of the spectrum instead of all frequencies. For example, when using band-pass from 20Hz to 100Hz the modulator will react mainly to a bass or bass-drum signal.

Min Off **Minimum**

Minimum defines the high-pass filter cut-off frequency. The band-pass is disabled if both the minimum and maximum frequencies are set to their limits, thus from 20Hz to 20kHz.

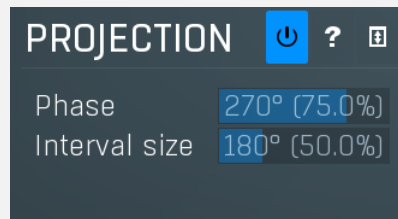
Max Off **Maximum**

Maximum defines the low-pass filter cut-off frequency. The band-pass is disabled if both the minimum and maximum frequencies are set to their limits, thus from 20Hz to 20kHz.

Q 0.7071 Q

Q defines the bandwidth for the high-pass and low-pass filters.

## Projection panel



Projection panel contains parameters of projection onto the LFO oscillator shape, which takes the value generated by the modulator and puts it onto the LFO oscillator shape. This features is useful for several creative effects.

### Enable

Enable button enables or disables the projection onto the LFO oscillator shape.

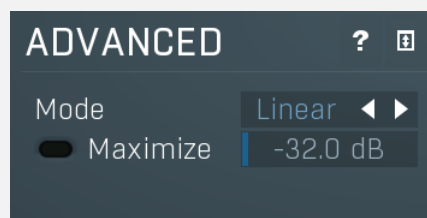
### Phase 270° (75.0%) Phase

Phase defines the offset from zero of the signal curve. By default it is 75%, because when you look at common oscillator shapes, such as a sine or triangle, at position 75% its value is minimal. Then when you look at the right side, the value is growing up to the 25%, where it becomes the maximum.

### Interval size 180° (50.0%) Interval

Interval defines the size of the interval from the oscillator shape in addition to **Phase**. As a result, phase defines where you start on the shape and interval specifies size of the window on the shape. Default value is 50% as for example sine grows from minimum to maximum in 50% of the period.

## Advanced panel



Advanced panel contains some more advanced features of the level follower.

### Mode Linear Mode

Mode controls the way in which the audio level is treated. **Linear** mode takes the audio level and uses it directly. This often tends to result in very low modulator values. **Squared** mode treats the squared levels. This is a compromise between linear and logarithmic modes. **Logarithmic** mode is the most aggressive one and usually also the most natural as it emulates the logarithmic behaviour of our ears.

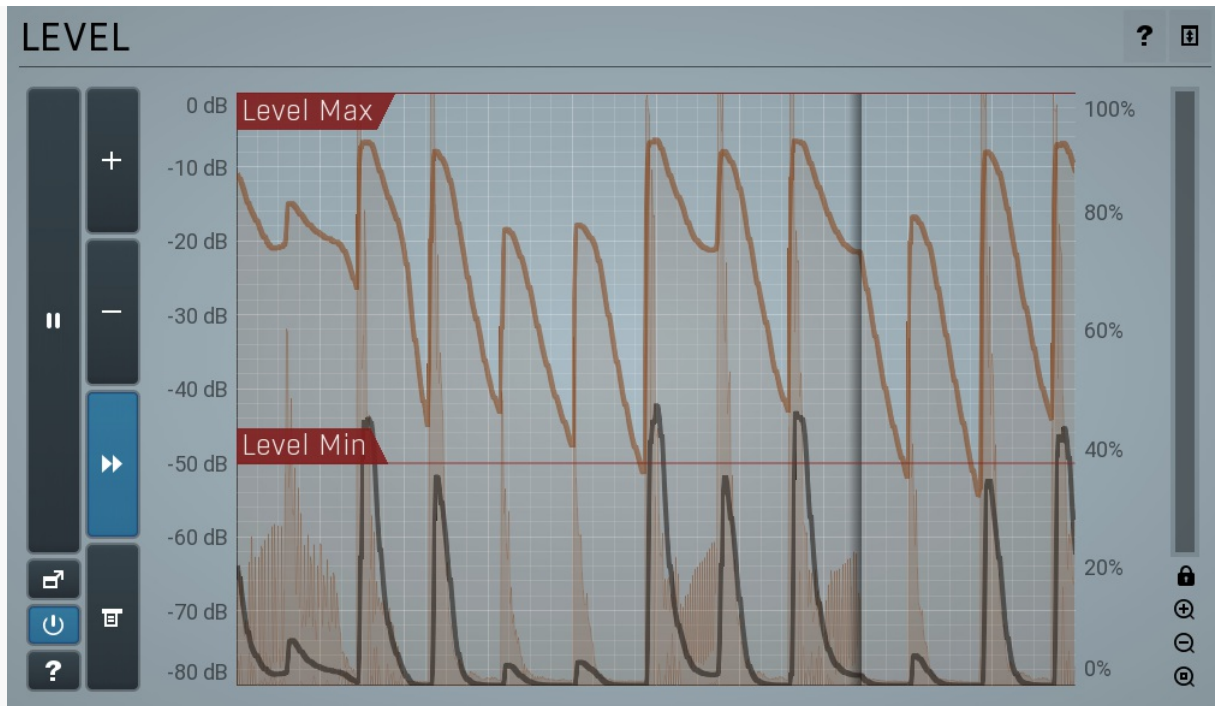
**Direct** mode is quite different as it doesn't really follow the level but instead takes the audio directly without any attack/release processing. It always takes the mid-range value,  $(\text{minimum} + \text{maximum})/2$ , from each audio block. This is mostly useful for control signals.

*For example, let's say your audio level is now -40dB. Then in linear mode it is treated as 1%, because the value of -40dB equals 0.01. In squared mode it becomes 10%. And finally in logarithmic mode it is 33%, because -40dB is 33% of the way from -60dB to 0dB.*

### -32.0 dB Maximize

Maximize enables threshold-based maximization. Normally the input signal is used to drive the level follower. In this mode however each input sample is treated as 0 or 1, depending on whether it is below or above the threshold. As a result you can get very fast and sharp transitions.





**Level**

### panel

Level panel contains the metering system showing the follower level. It is indispensable when setting up the follower.

The **orange graph** displays the measured level, which depends on the detector parameters, such as **Attack** or **Release**. In most cases you will want to set the follower so that it responds well to the full range of the audio material. After the detector parameters the level range is the next most important and is available via **Level min** and **Level max** parameters (these can be also adjusted directly from the analyser). In most cases you will want the minimum level to lie just below the lowest signal peaks and maximum level just above the highest peaks.

The **white graph** displays the output modulator values. It includes all the processing that affects the modulator including **LFO modulation** and **Project** features.



### Pause

Pause button pauses the processing.



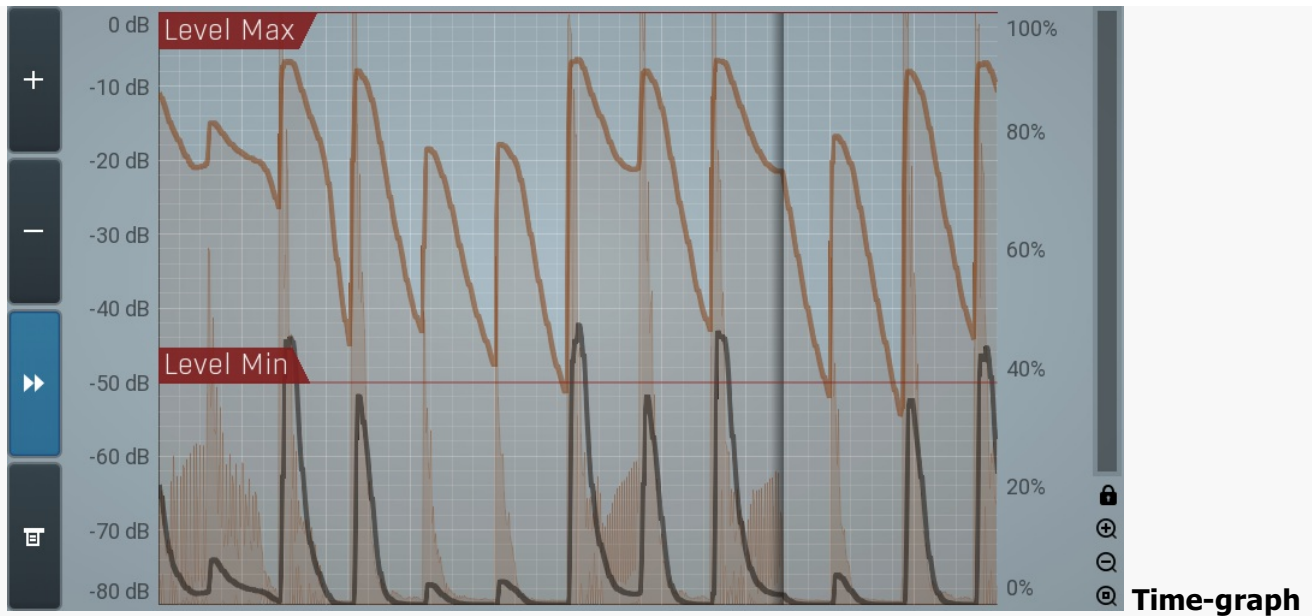
### Popup

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.



**view**

Time-graph view shows the measurements over a period of time.



**Plus**

Plus button increases the time-graph speed (reduces the period that is displayed).



**Minus**

Minus button decreases the time-graph speed (increases the period that is displayed).



**Rewind**

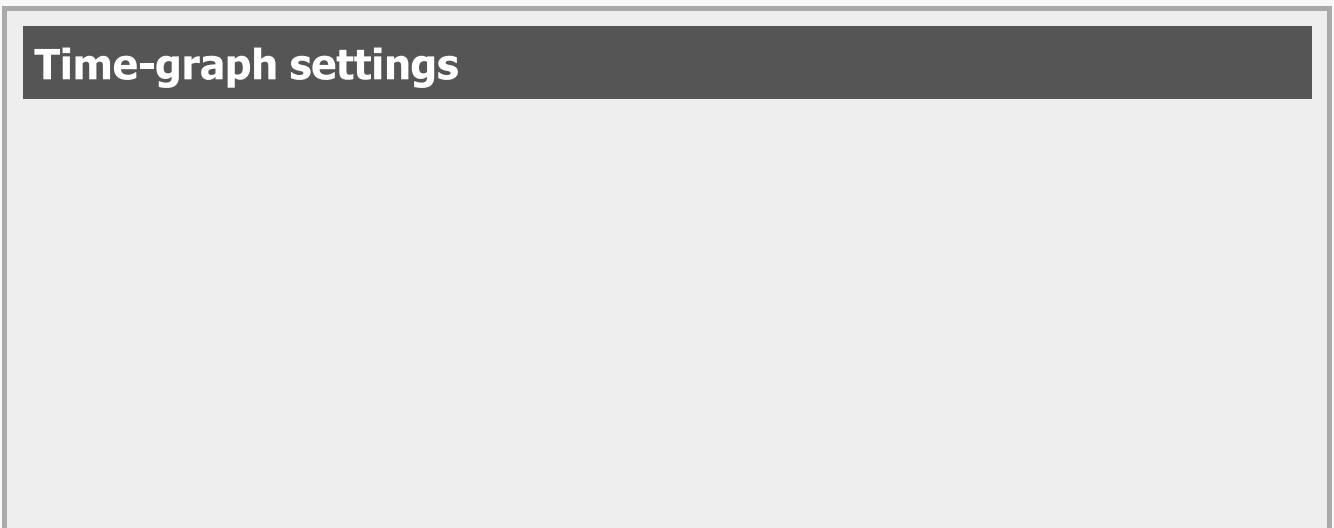
Rewind button enables or disables the time-graph static mode. In static mode the graphs are fixed and the current position cycles from left to right; otherwise the graphs move from right to left and the current position is fixed (at the right-hand side).

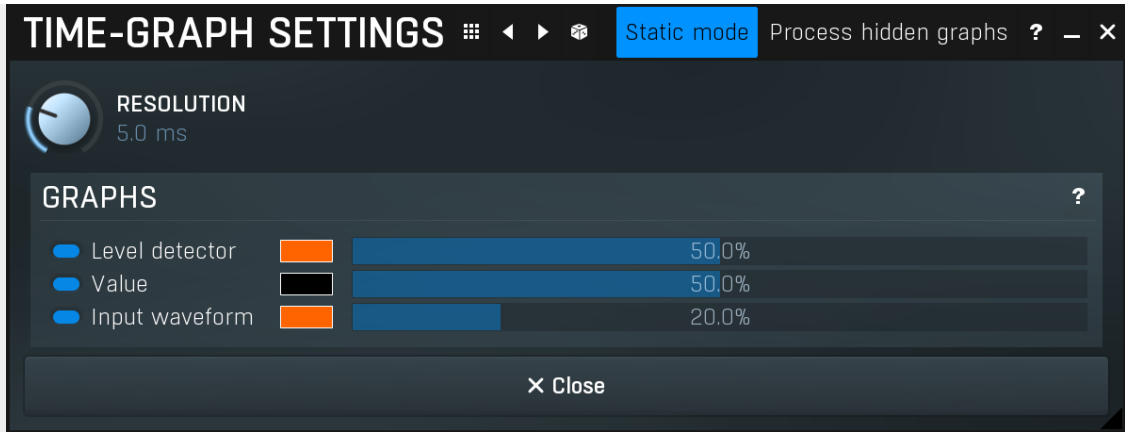


**Menu**

Menu button displays the time-graph settings. In this window you can control which graphs are displayed, the speed and other relevant parameters.

**Time-graph settings**





### 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.

Static mode

### Static mode

Static mode stops the graph from scrolling to the left and makes the graph refresh from left to right instead.

When this is disabled, the entire graph is moving from right to left as the incoming audio is processed. This may make it hard to spot the actual details, which is where the static mode comes to the rescue. Static mode is the default state and in most cases is more practical.

Process hidden graphs

### Process hidden graphs

Process hidden graphs enables measurement of graphs which are actually disabled in the view. This may come handy if you need to repeatedly show and hide several graphs. With this mode disabled, which it is by default, the processor saves CPU resources by computing only those measurements that are actually visible. However, when you show a currently hidden graph, no measurements are available, so you will need to wait for the graph to be generated from the incoming signal. If you enable this option, the graph will be available immediately after you make it visible.

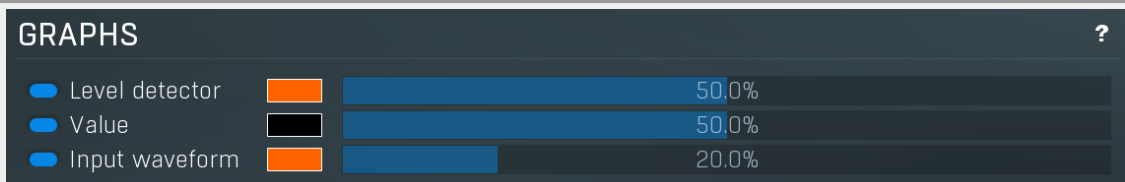


RESOLUTION  
5.0 ms

### Resolution

Resolution controls the time it takes for the graph to move one pixel. Therefore this actually controls the display speed.

## Graphs panel



Graphs panel contains all available graphs and lets you show or hide each of them, and change their visual properties.

# Envelope mode

Mode: MIDI | **Audio** | Input from feedback: Off  
Action ON: Start single | Action OFF: Stop single  
LFO modulation: 0.00% | Trigger

## DETECTOR

Side-chain ?

**THRESHOLD ON** -40.00 dB | **THRESHOLD OFF** -60.00 dB | **DETECTOR HOLD** 20 ms  
**ATTACK** 10 ms | **RELEASE** 10 ms | **RMS LENGTH** 20 ms

Pitch modulation: 0.00% | Transient modulation: 0.00%  
Transient mode: Enhanced | Release mode: Manual

Advanced detector settings

## BANDPASS

?

Min: Off  
Max: Off  
Q: 0.7071

## PROJECTION

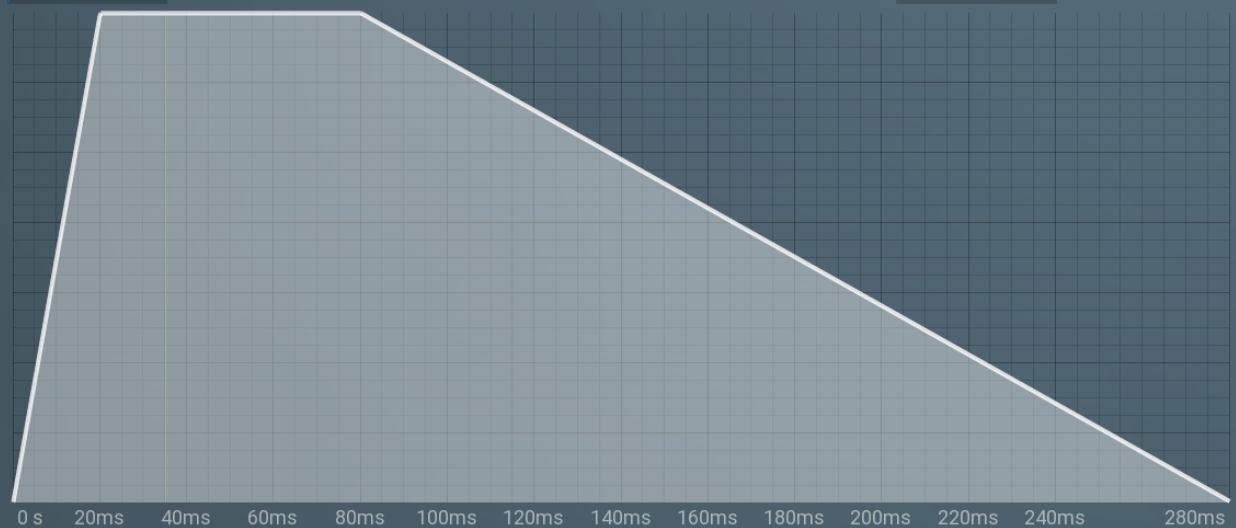
?

Phase: 270° (75.0%)  
Interval size: 180° (50.0%)

## ADSR

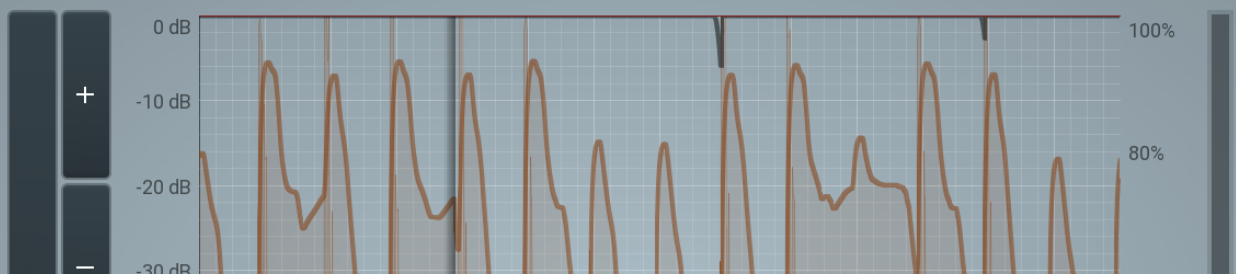
Custom shape Percussive Sync ?

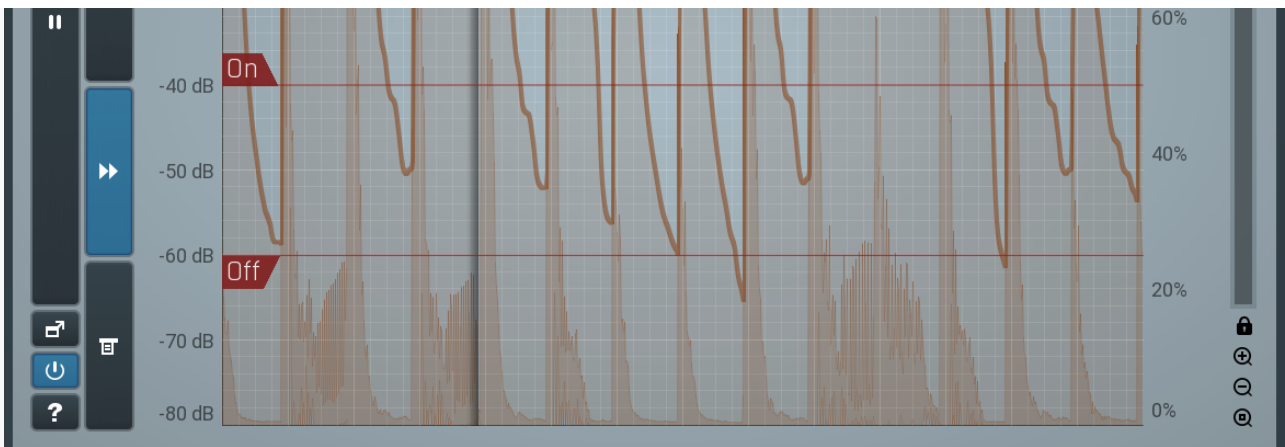
Delay	Attack	Hold	Decay	Sustain	Tremolo	Release
0 ms	20 ms	0 ms	10 ms	0.00 dB	0.00%	200 ms
Smoothness	0.00%	0.00 dB	0.00%		6.000 Hz	0.00%
0.00%					0 ms	



## ANALYZER

?





Envelope mode makes the modulator generate arbitrary envelopes from input MIDI or by analyzing the audio input level. When using MIDI the modulator responds to input note-on and note-off messages. When using an audio input (if available), the modulator detects the input level and when it exceeds the **Threshold On**, it behaves like a note-on MIDI event. Afterwards when the level drops down below **Threshold Off**, it behaves like a note-off MIDI event. Each event can result in just about any action. By default note-on starts the envelope and note-off initiates the release stage, but a different behaviour is also possible depending on **Action ON** and **Action OFF** parameters.

Mode MIDI Audio Mode

Mode controls if the envelope is triggered by audio or by MIDI input.

Action ON Start single Action ON

Action ON controls what happens when a note-on event occurs (either via audio or MIDI).

**Start single** action, which is the default, means that the envelope will start on the note-on event, but only once, it won't start again until you release all of the keys that are relevant for MIDI triggering only. With audio triggering it works the same as **Start** mode.

**Start** makes the envelope start every time you press a key, whether another key is already pressed or not. The envelope will seamlessly jump to the attack stage avoiding any abrupt changes.

**Start forced** is similar, but lets the envelope start from the very beginning every time. So for example if the envelope is currently in a long release stage and the new modulator value is 0.5, then the Start action jumps to a location in the attack stage where there is a value of 0.5 as well, hence avoiding abrupt changes. Start forced action on the other hand starts the whole envelope over from the beginning of the attack stage, where the value is most likely 0.

**Start legato** is in a way an opposite of the **Start single** in that it starts only if there is at least one other note already playing. As such it can only be used with MIDI triggering. With audio triggering it works the same as **Start** mode.

**Ignore** action simply ignores this event.

The remaining actions are rather creative and let you do the opposite - initiate release stage and stop the envelope when you press a key.

Action OFF Stop single Action OFF

Action OFF controls what happens when a note-off event occurs (either via audio or MIDI).

**Stop single** action, which is default, means that the envelope will enter the release stage on the note-off event, but only once, at the moment you release the last key (if you were holding more than one) it is relevant for MIDI triggering only.

**Stop** makes the envelope enter the release stage every time you release a key, whether another key is already pressed or not.

**Ignore** action simply ignores this event.

The remaining actions are rather creative and let you do the opposite - start the envelope when you release a key.

LFO modulation 0.00% LFO modulation

LFO modulation defines the amount of LFO modulation applied in addition to the envelope. With 0% the modulator uses only the envelope; with 100% the modulator does the same job as if the modulator were in **Normal** mode. To set the LFO parameters switch to normal mode temporarily.

Trigger Trigger

Trigger button serves for manual triggering. It can be associated to other modulators as well for example, so it enables you to trigger the envelope pretty much any way.

## Detector panel

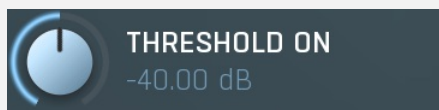


Detector panel contains parameters of the audio event detector.

Side-chain

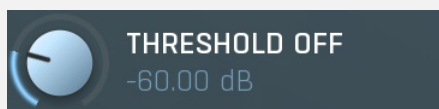
## Side-chain input

Side-chain input makes the modulator analyze the side-chain input instead of the regular input.



### Threshold On

Threshold On defines the note-on level. When the input level rises above it the envelope is started. Then it stays into the sustain stage until the level falls below **Threshold Off** and the release stage is initiated.



### Threshold Off

Threshold Off defines the note off level. When the input level rises above **Threshold On**, the envelope is started. Then it stays into the sustain stage until the level falls below this threshold off and the release stage is initiated.



### Peak hold

Peak hold defines the time that signal level detector holds its maximum before the release stage is allowed to start. As an example, you can imagine that when an attack stage ends there can be an additional peak hold stage and the level is not yet falling, before the release stage starts. This is true only when **true peak** mode is enabled (check the advanced detector settings if available).

It is often used in **gates** to avoid the gated level falling below the threshold too quickly, while having short release times. If you want the gate to close quickly, you need a short release time. But in that case the ending may be too abrupt and even cause some distortion. So you use the peak hold to delay the release stage.

It is also used along with **look-ahead** to avoid distortion in **limiters and compressors**. If you need a very short attack, the attack stage may be too quick and cause distortions. In limiters this attack time is often 0ms, in which case it becomes a clipper. Setting look-ahead and peak hold to the same value will make the detector move ahead in time, so that it can react to attack stages before they actually occur and yet hold the levels for the actual signal to come.



### 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:

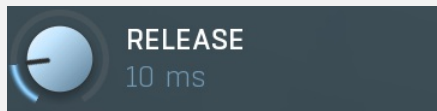
*In a **compressor** the attack time controls how quickly the measured level moves above the threshold and the processor begins compressing. As a result, a very short attack time will compress even the beginning transient of a snare drum for example, hence it would remove the punch. With a very long 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.



### 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.

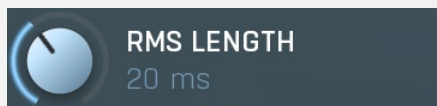
To understand the working of a level detector, it is best to cover the typical cases:

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.



### RMS length

RMS length defines the window length used for smoothing the input. In most cases the input waveform contains lots of separate peaks and short transients. All of them would generate note-on events and the spaces between them would similarly cause note-offs. RMS is used to smooth the input. The longer the window, the longer interval it takes and the longer delay it exhibits. If it is too short, unpredictable behaviour can be expected.



### Pitch modulation

Pitch modulation lets you employ the pitch detector (configurable from the **Pitch tab**) in the detector. This may sound odd at first, but thinking of the input signal, you may measure its level, but you can also measure other properties, such as its pitch, and use them in exactly the same way. While an input level is usually understood as a value in decibels, pitch is a frequency in Hz, so the plugin smartly transforms the frequency to mimic the level axis. When you look at the detector graph afterwards, you can hardly tell the exact pitch in Hz, but that's not really relevant or necessary.

What is this for? Let's show it with an example. Let's say you have an instrument, say a bass, which is playing legato, and you want some kind of effect at the beginning of the note (in case of **Follower** mode) or you just want to restart some kind of filter at the beginning of each note (in case of **Envelope** mode). But since the performance is legato, meaning there are no gaps in between the notes, the level graph is just a steady horizontal line, which is pretty much useless for us. The pitch modulation lets you replace this horizontal line with something much more useful - the pitch. While the level isn't changing much at all, the pitch is changing. The plugin then takes the actual pitch as the input signal, so you can let the plugin follow it in some way, start envelopes when a certain pitch is exceeded, or using **Transformation** you can even let the plugin restart an envelope every time the pitch changes. By setting the pitch modulation half-way you can let the plugin react to both properties, the level and the pitch.



### Transient modulation

Transient modulation lets you detect transients and blend that detection with the control signal. This way you may let the modulator be controlled not by level (alone or in combination with pitch for example), but also by transients detected in either of these properties - level or pitch.



### Transient mode

Transient mode controls the way in which the transients are detected. These simply provide different results, so you should just try the alternative modes if the default one doesn't suit your audio material. **Attack only modes** ignore sustain transients - those

moments when the level decreases.

## Release mode Manual ◀ ▶ Release mode

Release mode defines how the plug-in performs when decreasing level. In **manual mode** this is based only on the **release time**, which is suitable for most cases when the signal has constant characteristics. Automatic release modes can adapt to signals with unstable characteristics.

**Automatic** and **Automatic fast** modes: the longer the level stays above the threshold, the longer the release time will be and thus, the longer it will take to move below the threshold and end the release stage. The idea is that if the input is loud for some time, it will most likely stay that way for some more time, hence it should be stabilized to avoid unnecessary temporary fluctuations, which could result in pumping.

Both automatic modes increase the release time when the input signal is above the threshold and vice versa. The speed of the increase depends on the **Auto speed** parameter. Automatic fast mode uses full speed immediately after crossing the threshold, automatic mode varies the speed according to the current signal level.

*For example, when a guitarist plays softly, the level is low and fluctuates around the threshold and the release time gets slower. So the processor quickly responds to sudden changes. However, when the guitarist starts playing a solo, the level rises and, the longer the solo is, the longer the release time becomes, hence the response becomes slower avoiding unnecessary fluctuations (pumping) when the solo contains small silent sections.*

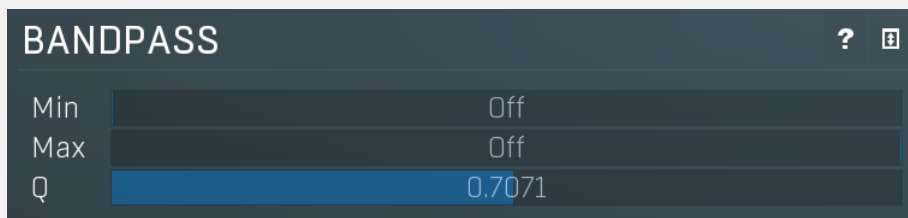
**Linear 1** and **Linear 2** modes: the higher the level is, the longer the release. The idea is that if the input is very loud, it will probably stay that way for some time, so it is wise to keep the levels up too. This is similar to the automatic modes, however the main factor is not how long the level is high, but how high it is.

Below the threshold the release time is the same as the attack time, above the threshold the release time rises from the attack time up to the specified release time parameter. Linear 1 mode usually provides higher release times than does Linear 2.

**Opto** mode: the higher the level is, the shorter the release. So this is kind of the opposite of linear modes. The idea is, that you are expecting short transients, which you wish to deal with. Normally the higher the level would get in such a transient, the longer it would take to get the level below the threshold, so, when used in a compressor for example, these transients would cause unnecessary compression in the sustain stage. The opto detector lowers the level quickly, minimizing the amount of compression in the sustain stage.

*For example, let's say you are compressing a full drumset, but there is a very dominant sharp and short hi-hat sound, so it is appropriate to have short release times. You would use **Opto** mode. But the rest of the drumset deserves a softer treatment, so you want to keep longer release times. Use one of the other modes.*

## Band-pass panel



Band-pass panel contains parameters of the envelope detector band-pass. Using this feature you can make the envelope detect level of just part of the spectrum instead of all frequencies. For example, when using a band-pass from 20Hz to 100Hz the modulator will react mainly to a bass or bass-drum.

### Min Off **Minimum**

Minimum defines the high-pass filter cut-off frequency. The bandpass is disabled if both frequencies are set to their limits, thus from 20Hz to 20kHz.

### Max Off **Maximum**

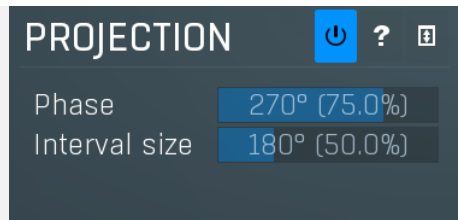
Maximum defines the low-pass filter cut-off frequency. The bandpass is disabled if both frequencies are set to their limits, thus from 20Hz to 20kHz.

### Q 0.7071 **Q**

Q defines the bandwidth for the high-pass and low-pass filters.

## Projection panel





Projection panel contains parameters of projection onto the LFO oscillator shape, which takes the value generated by the modulator and puts it onto the LFO oscillator shape. This features is useful for several creative effects.



### Enable

Enable button enables or disables the projection onto the LFO oscillator shape.

Phase

270° (75.0%)

### Phase

Phase defines the offset from zero of the signal curve. By default it is 75%, because when you look at common oscillator shapes, such as a sine or triangle, at position 75% its value is minimal. Then when you look at the right side, the value is growing up to the 25%, where it becomes the maximum.

Interval size

180° (50.0%)

### Interval

Interval defines the size of the interval from the oscillator shape in addition to **Phase**. As a result, phase defines where you start on the shape and interval specifies size of the window on the shape. Default value is 50% as for example sine grows from minimum to maximum in 50% of the period.



### 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.



### Random

Random button generates random settings using the existing presets.



Custom shape

### Custom shape

Custom shape button enables custom shape mode, which lets you draw your own attack and release stages using the envelope system. Both stages are then automatically connected to form the resulting envelope.

Percussive

### Percussive

Percussive button activates the immediate release mode in which case the note-off causes an immediate switch to the release stage. If this is disabled, the release stage does not occur until the whole attack/decay stage finishes.

Sync

### Sync

Sync button controls the ADSR tempo sync feature. By default this is disabled and means that all times are followed exactly, meaning that if **Attack** is say 100ms, then it will be 100ms indeed. Tempo sync lets the plugin adjust the times to ensure it will be always in sync with the host tempo. In this case 100ms may become say 125ms if the tempo is 120bpm, because 125ms is the length of a 16th note. This makes it extremely simple to convert any envelope to a tempo-synced one. The plugin always chooses the nearest longer note, in other words it always round up.

**Straight** and **Triplets** modes automatically find 'nice' values.

*For example, if a 16th note takes 100ms, the attack time is 550ms, and the sync mode is straight, then the plugin checks for 100ms, find out that it is too low, so it checks 8th note, being 200ms, still too low, then continues with quarter note, which takes 400ms, and*

still not enough, finally 800ms corresponding to a half note is the one, so the resulting time will be 800ms. Triplet cases are more complex, but the principle is the same.

1/16, 1/8 and 1/4 modes choose the nearest higher multiply of the base note length. For example, if a 16th note takes 100ms, the attack time is 550ms, and the sync mode is 1/16, the resulting time will be 600ms.

Tremolo

## Tremolo

Tremolo button displays additional tremolo settings, containing tremolo behaviour and shape.

### Tremolo settings

**TREMULO SETTINGS**

**TREMULO BEHAVIOUR**

DEPTH 0.00%    RATE 6.000 Hz    FADE-IN 0 ms

Tempo sync Off

- Tremolo starts in decay stage
- Tremolo continues in release stage
- Random initial phase
- Follow sustain level

**TREMULO SHAPE**    Presets    Random

Shape Sine    Custom 25.0%    Step 25.0%    Smooth 50.0%    Advanced

Normal

Harmonics

### Tremolo behaviour

**TREMULO BEHAVIOUR**

DEPTH 0.00%    RATE 6.000 Hz    FADE-IN 0 ms

Tempo sync Off

- Tremolo starts in decay stage
- Tremolo continues in release stage
- Random initial phase
- Follow sustain level

DEPTH 0.00%

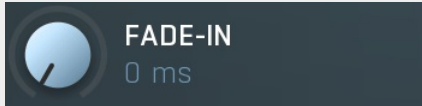
#### Depth

Depth controls the amount of tremolo mixed in the sustain stage (or potentially before).

RATE 6.000 Hz

#### Rate

Rate controls the tremolo rate and is relevant only if tempo sync is not used.

 **FADE-IN**  
0 ms

### Fade-in

Fade-in controls the length of the tremolo fade-in. It is especially useful when you want to use the random initial phase feature to avoid the initial discontinuity when the tremolo kicks in.

Tempo sync Off ◀ ▶ **Tempo**

### sync

Tempo sync lets you synchronize the tremolo to the host's tempo.

Tremolo starts in decay stage **Tremolo**

### starts in decay stage

Tremolo starts in decay stage makes the tremolo start during the decay stage. By default this is disabled and the tremolo starts in the sustain stage. When it is enabled you will most likely have a longer decay and also a longer tremolo fade-in, so that the tremolo slowly comes in as the envelope is decaying.

Tremolo continues in release stage **Tremolo**

### continues in release stage

Tremolo continues in release stage makes the tremolo continue with the tremolo during the release stage. By default this is disabled and the tremolo stops as soon as the release stage starts.

Random initial phase **Random**

### initial phase

Random initial phase makes the tremolo start with a random phase. By default this is disabled and the tremolo starts always starts in the 0 phase, which ensures the tremolo always starts in the same way. However if you play multiple notes at once, the tremolo will be exactly the same, while you may want it to be different for each note and make it sound more 'human'. Enabling this option also activates a short **tremolo fade-in** to avoid initial discontinuity.

Follow sustain level **Follow**

### sustain level

Follow sustain level makes the tremolo level based on sustain level. When this is disabled, the tremolo rarely reaches up to 100% level. However if the sustain level is say -20dB, then the tremolo actually cannot exceed 1% (which is -20dB), so it is clipped. It can however go upwards to 100%. This naturally changes the actual tremolo shape. If you want to avoid that and make sine really be a sine for example, enable this option, and in the case above the tremolo will really go up/down -20dB if set to 100%.

## 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.

 **Random**

Random button generates random settings using the existing presets.

## Normal

Normal button switches the generator into the normal mode, which lets you edit the shape of the oscillator. This is especially advantageous for low-frequency oscillators, where the shape matters even though it doesn't have any physical meaning.



## Convert

Convert button converts the current shape into harmonic-based representation. Please note that since the number of harmonics is limited, the result will not perfectly resemble the original shape.

## Harmonics

Harmonics button switches the generator into the harmonics mode, which lets you edit the levels and phases of individual harmonics. This is especially advantageous for high-frequency oscillators, hence sound generators.

## Signal generator in Normal mode

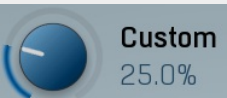


Signal generator in Normal mode works by generating the oscillator shape using a combination of several curves - a predefined set of standard curves, custom shape, step sequencer and custom sample. It also post-processes the shape using several filters including smoothing to custom transformations. This is especially useful when using the oscillator as an LFO (low-frequency-oscillator), where the harmonic contents does not really matter, but the shape does.



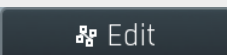
### Shape

Shape controls the main shape used by the signal generator. There are several predefined shapes: exponential, triangle, sine power 8, sine power 4, sine square, sine, harmonics, more harmonics, disharmonics, sine square root, sine 4 root, rectangle, rect-saw, saw, noise and mess. You can choose any of them or interpolate between any 2 adjacent shapes using this control.



### Custom

Custom controls the amount of the custom shape that is blended into the main shape.



### Edit

Edit button shows the custom shape editor.



Step  
25.0%

### Step

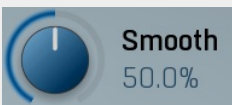
Step controls the amount of the step sequencer shape that is blended into the main shape (which has already been blended with the custom shape).



Edit

### Edit

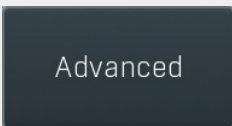
Edit button shows the step sequencer editor.



Smooth  
50.0%

### Smooth

Smooth controls the amount of smoothing. Many shapes, especially those produced by the step sequencer, have rough jagged edges, which may be advantageous, but when used to modulate certain parameters, the output may be clicking or causing other artifacts. Smoothness helps it by smoothing the whole signal shape out and removing these rough edges.

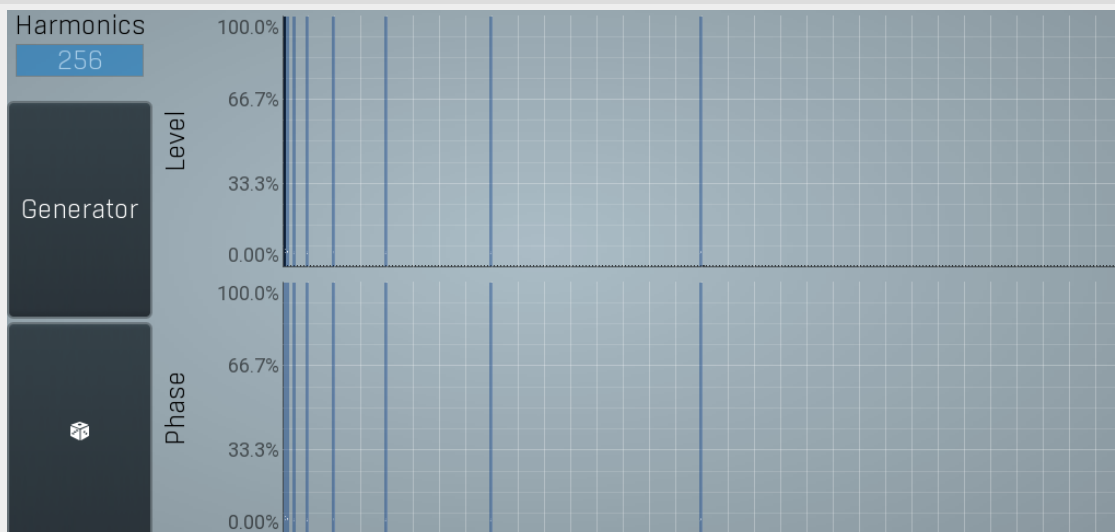


Advanced

### Advanced

Advanced button displays an additional window with more advanced settings for post-processing the signal shape, such as harmonics or custom transformations.

## Signal generator in Harmonics mode



Signal generator in Harmonics mode works by generating the oscillator shape using individual harmonics. Essentially a harmonic is a sine wave. The first harmonic, known as the fundamental, fits once in the oscillator time period, hence it is the same as selecting sine wave in the **Normal mode**. The second harmonic fits twice, the third three times etc. In theory, any shape you create in normal mode can be converted into harmonics. However, this approach to signal generation needs an enormous number of harmonics, which is both inefficient to calculate and mostly hard to edit. Therefore, the harmonic mode can process up to 256 harmonics, which is enough for very complex spectrums, however it is still not enough to generate an accurate square wave for example. If your goal is to create basic shapes, it is better to use the normal mode.

It is nearly impossible to say how a particular curve will sound when used as a high-frequency oscillator in a synthesizer, just by looking at its shape. Harmonics mode, on the other hand, is directly related to human hearing and makes this process very simple. In general, the more harmonics you add, the richer the sound will be. The higher the harmonic, the higher the tone. Usually, one leaves the first harmonic enabled too, as this is the fundamental tone, however you may experiment with more dissonant sounds without it.

Editing harmonics can be time consuming unless you hear what you want, so a signal generator is also available. This great tool lets you generate a random spectrum by a single click. You can also open the **Generator** settings and edit its parameters, which basically control the audio properties in a more natural way - using parameters such as complexity, harmonicity etc.



Generator

### Generator

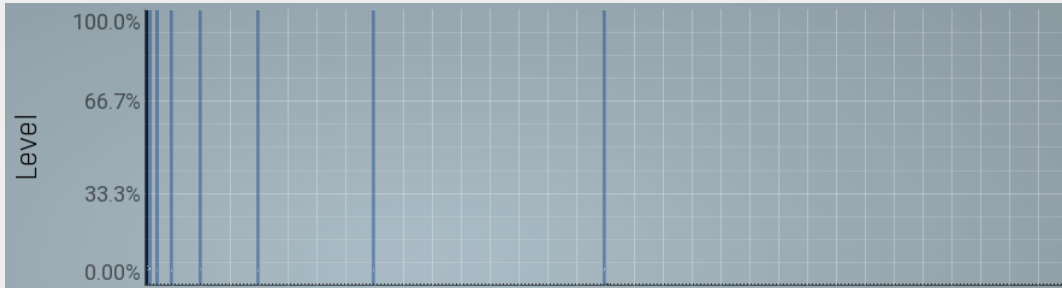
Generator button shows a powerful harmonics generator, which can create unlimited number of various timbres and even

analyze a sample and extract harmonics from it.



### Randomize

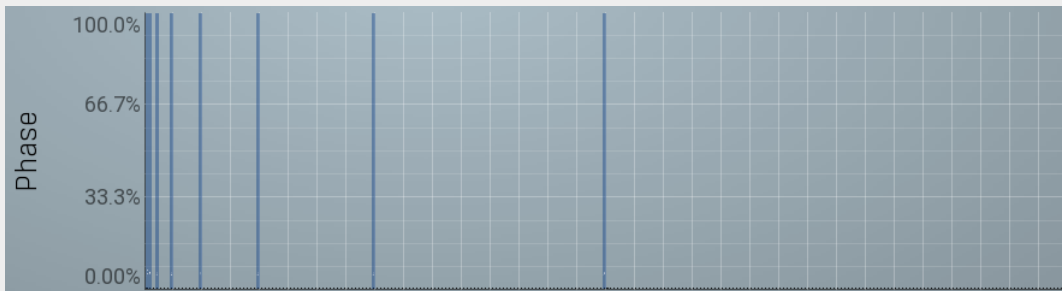
Randomize button selects random parameters for the harmonics generator, so you can use it to get a random sound character instantly. Hold **Ctrl** to slightly modify existing generator settings instead of completely changing them.



**Magnitudes**

### graph

Magnitudes graph contains the levels of the individual harmonics. The highlighted bars are octaves, thus the 1st, 2nd, 4th, 8th harmonic etc.



**Phases graph**

Phases graph contains the phases of the individual harmonics. The highlighted bars are octaves, thus the 1st, 2nd, 4th, 8th harmonic etc.

0 ms **Delay**

Delay lets you shift the entire envelope forwards in time. While this doesn't make much sense for a global instrument envelope for instance, it may be well useful to control characteristics of evolving sounds.

20 ms **Attack**

Attack controls the length of the initial stage of the envelope. It is one of the most important parameters controlling how quick the initial transient is. For most instruments the length is quick short, but for pads and other slowly evolving sounds it is quite common to set this to several seconds.

0 ms **Hold**

Hold specifies the time the level stays at maximum after the attack stage.

10 ms **Decay**

Decay controls the time it takes for the level to drop from the maximum to the **Sustain**. If the sustain is 0dB, then this parameter has no effect, because in a way the sustain stage starts immediately after the attack.

0.00 dB **Sustain**

Sustain controls the sustain level. For most sounds the initial attack transient is the highest point of the entire sound. Imagine playing a string instrument, such as a guitar, the initial hit to the strings is represented by the attack+hold+decay sections and is the most prominent. After that the level drops to the sustain stage, where it holds for most of the time.

0.00% **Tremolo**

Tremolo defines the amount of the tremolo effect that is engaged in the sustain, or even in the decay section and continues until the envelope ends. While this is a rather unusual feature for an envelope to have, it is very handy for simulating various effects human players do when performing on real instruments, such as the tremolo or vibrato.

200 ms **Release**

Release controls the length of the release section, which usually starts when a note is released.

0.00% **Attack shape**

Attack shape controls the shape of the attack section and defines its sound character.

### 0.00 dB **Hold level**

Hold level controls the level of the hold section. By default it equals maximum meaning that the hold section actually holds the maximum level. However by making it lower you can sort of simulate 2 separate decay sections, first going from maximum to hold level, second going from hold level to sustain.

### 0.00% **Decay shape**

Decay shape controls the shape of the decay section and defines its sound character.

### 6.000 Hz **Tremolo rate**

Tremolo rate controls the speed of the tremolo. In the tremolo settings it is possible to control additional characteristics including tempo sync.

### 0.00% **Release shape**

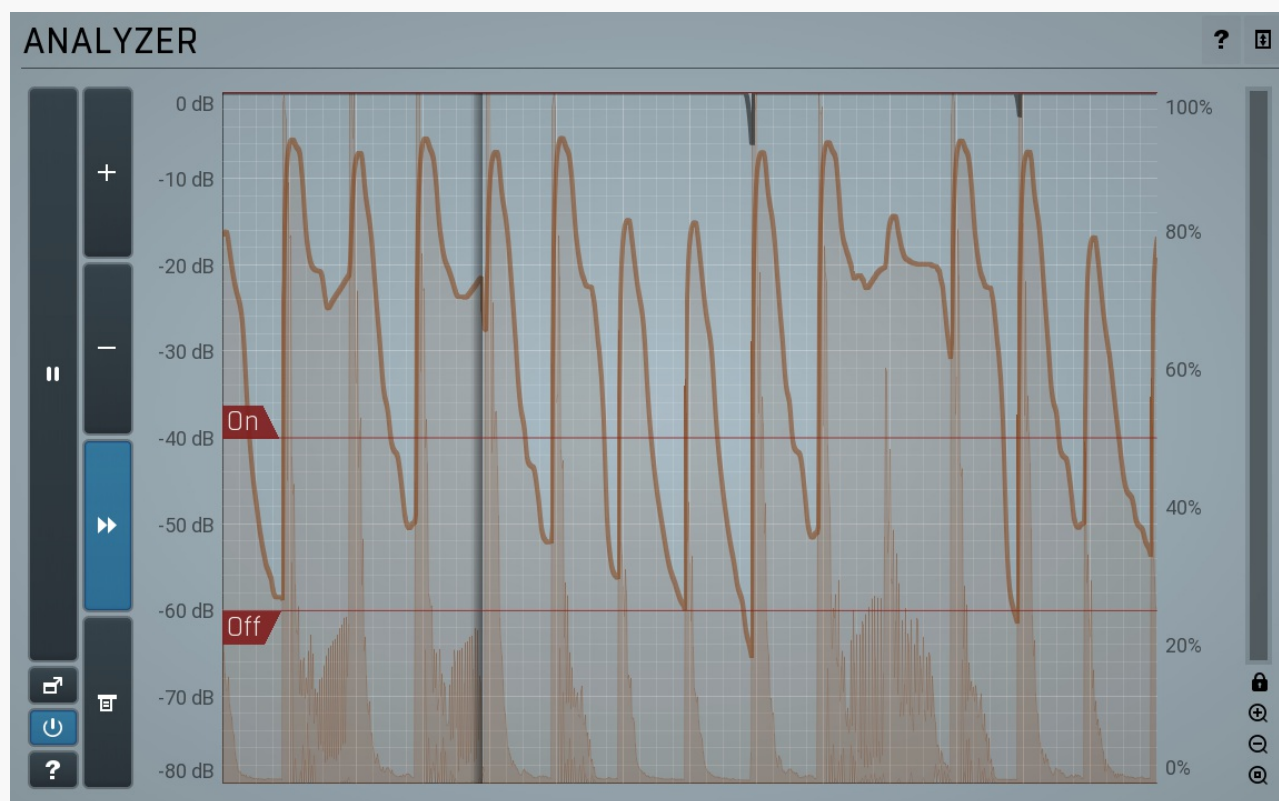
Release shape controls the shape of the release section and defines its sound character.

### 0.00% **Smoothing**

Smoothing lets you smoothen the entire envelope avoiding abrupt jumps. Note that in some cases involving short jumps the results may be a bit obscure.

### 0 ms **Tremolo fade-in**

Tremolo fade-in defines the time for the tremolo to reach its full level. It is a natural behaviour of human players (on say a saxophone) that they don't start a full tremolo immediately and rather let the modulation rise to maximum over a period of time.



## Analyzer panel

Analyzer panel contains the metering system showing the envelope level. It is indispensable when setting up the envelope.

The **orange graph** (assuming the default color) displays the measured level, which depends on the detector parameters, such as **RMS length**. Its purpose is to smooth the input and to avoid extremely fast fluctuations. The main goal will be to set the **Threshold On** and **Threshold Off** properly, so that the events are well detected and there are no false events. Both parameters can be adjusted directly from the graph. In most cases the threshold on will be placed above the threshold off.

The **white graph** (assuming the default color) displays the modulator values. It includes all the processing that affects the modulator including **LFO modulation** and **Project** features.



### Pause

Pause button pauses the processing.



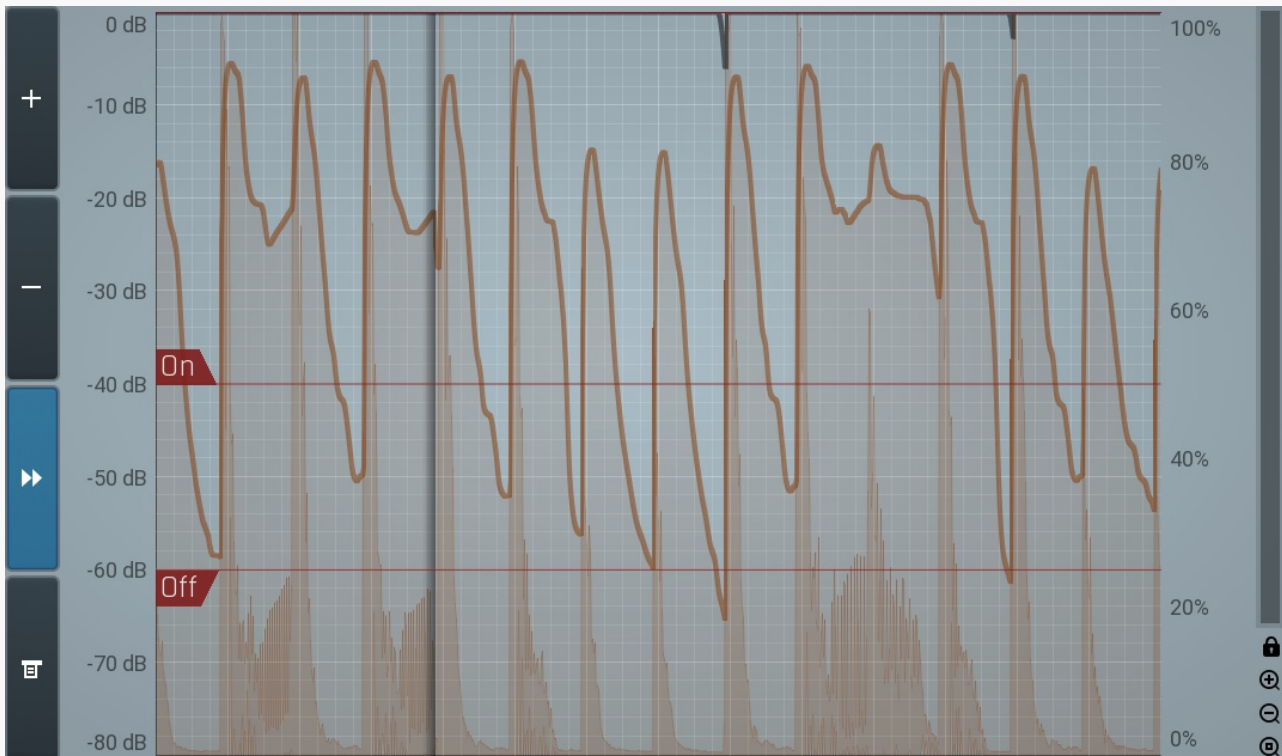
### Popup

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.



### Time-graph view

Time-graph view shows the measurements over a period of time.



### Plus

Plus button increases the time-graph speed (reduces the period that is displayed).





### Minus

Minus button decreases the time-graph speed (increases the period that is displayed).



### Rewind

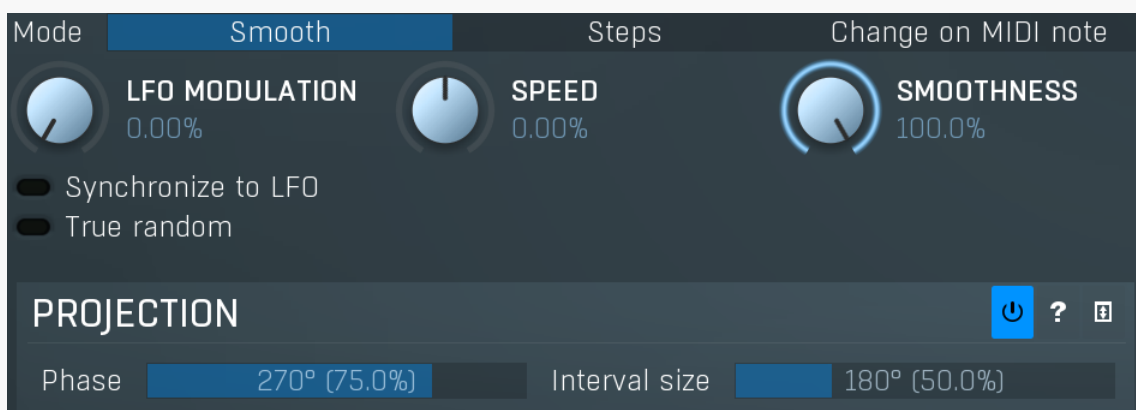
Rewind button enables or disables the time-graph static mode. In static mode the graphs are fixed and the current position cycles from left to right; otherwise the graphs move from right to left and the current position is fixed (at the right-hand side).



### Menu

Menu button displays the time-graph settings. In this window you can control which graphs are displayed, the speed and other relevant parameters.

## Random mode



Random mode makes the modulator generate a pseudorandom sequence. Please note that despite its name, it is created so that it generates the same sequence every time. However the generator is linked to the **Speed** parameter, so if you change it, the whole sequence changes.

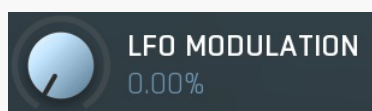


Mode defines the behaviour of the randomizer.

**Smooth** produces a continuous random modulation. **Smoothness** then controls how smooth it will be, where 0% means it will connect distinct values by straight lines, 100% means the modulation will be a completely smooth curve walking through these random points.


**Steps** produces a step change every particular time interval. It can also granularize it to a specified number of possible values according to **Smoothness** value. 100% disables the granularization. Otherwise the number of steps is the number of percentage values, so 3% means there will be 3 possible values, equally distributed over the range, let's call them 0%, 50% and 100%. Since it doesn't make sense to have 0 or 1 steps, the minimum is always 2. 2 steps essentially means the modulator is randomly switching between the minimum and maximum values for all associated parameters.

**Change on MIDI note** generates a random value every time a MIDI note is received by the plugin.



### LFO modulation

LFO modulation defines the amount of LFO modulation applied in addition to the random generator. With 0% the modulator uses only the randomizer; with 100% the modulator does the same job as if the modulator were in **Normal** mode. To set the LFO parameters switch to normal mode temporarily.

 **SPEED**  
0.00%

### Speed

Speed defines the speed of the random changes proportional to the current tempo. 0% means that the speed is the same as your song's tempo.

 **SMOOTHNESS**  
100.0%

### Smoothness

Smoothness defines the amount of smoothing of the randomizer curve in order to minimize abrupt edges.

Synchronize to LFO

### Synchronize to

#### LFO

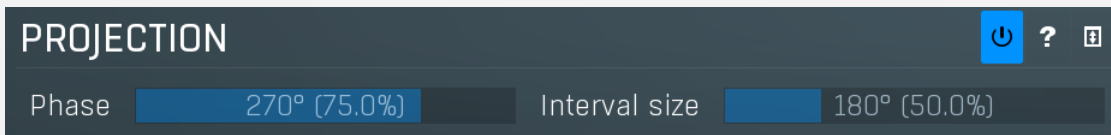
Synchronize to LFO lets you synchronize the speed of the random sequence to LFO (Normal mode), hence also to your host. **Speed** is still applicable and, for example, +100% means 2x speed, +200% means 4x the speed etc.




True random

### True random

True random makes the modulator produce a true pseudo-random sequence independent of the current position within the project. By default this is disabled, so that every time you play your project, it sounds the same. But you might want to enable this option, for live performances for example.

## Projection panel



PROJECTION   

Phase  Interval size

Projection panel contains parameters of projection onto the LFO oscillator shape, which takes the value generated by the modulator and puts it onto the LFO oscillator shape. This features is useful for several creative effects.



### Enable

Enable button enables or disables the projection onto the LFO oscillator shape.

Phase  **Phase**

Phase defines the offset from zero of the signal curve. By default it is 75%, because when you look at common oscillator shapes, such as a sine or triangle, at position 75% its value is minimal. Then when you look at the right side, the value is growing up to the 25%, where it becomes the maximum.

Interval size  **Interval**

Interval defines the size of the interval from the oscillator shape in addition to **Phase**. As a result, phase defines where you start on the shape and interval specifies size of the window on the shape. Default value is 50% as for example sine grows from minimum to maximum in 50% of the period.

## Pitch mode

The screenshot shows a software interface with the following sections:

- LFO MODULATION:** A knob set to 0.00%. Below it, a dropdown menu for "Input from feedback" is set to "Off".
- MIN FREQUENCY:** A knob set to 20.00 Hz.
- MAX FREQUENCY:** A knob set to 20.0 kHz.
- SHIFT:** A section header with a help icon (?).
- SHIFT OCTAVES:** A knob set to 0.
- SHIFT SEMITONES:** A knob set to 0.
- SHIFT CENTS:** A knob set to 0.
- AUTO-TUNE:** A section header with an "Enable" button and a help icon (?).
- AUTO-TUNE SPEED:** A knob set to 50.0%.
- AUTO-TUNE DEPTH:** A knob set to 50.0%.
- DETECTOR:** A section header with a "Side-chain" button, a help icon (?), and a lock icon (🔒).
- Detector Parameters:**

Min recognized	50.00 Hz	Max recognized	1000 Hz
Stabilization	10 ms	Speed	75.0%
Detection accuracy	80.0%	Threshold	-20.0 dB

Pitch mode makes the modulator detect the input pitch.

### LFO modulation

LFO modulation defines the amount of LFO modulation applied in addition to the pitch detector. With 0% the modulator uses only the pitch detector; with 100% the modulator does the same job as if the modulator were in **Normal** mode. To set the LFO parameters switch to normal mode temporarily.

### Min frequency

Min frequency defines the frequency, which will cause the modulated parameters to have their minimum values. This basically manipulates the range of the parameters, but it is based on the frequency rather than on parameter values.

### Max frequency

Max frequency defines the frequency, which will cause the modulated parameters to have their maximum values. This basically manipulates the range of the parameters, but it is based on the frequency rather than on parameter values.

## Shift panel


The screenshot shows the SHIFT panel with three knobs:

- SHIFT OCTAVES:** A knob set to 0.
- SHIFT SEMITONES:** A knob set to 0.
- SHIFT CENTS:** A knob set to 0.

Shift panel lets you shift the detected frequency by specified different amounts. All of its parameters do basically the same thing, but in different units.

### Octaves


Octaves shifts the detected pitch by the specified number of octaves.



SHIFT SEMITONES  
0

### Semitones

Semitones shifts the detected pitch by the specified number of semitones.




SHIFT CENTS  
0

### Cents

Cents shifts the detected pitch by the specified number of cents of a semitone. The actual pitch change is the sum of these 3 control values.

## Auto-tune panel

AUTO-TUNE 🔌 Enable ?



AUTO-TUNE SPEED  
50.0%




AUTO-TUNE DEPTH  
50.0%

Auto-tune panel contains the automatic tuner parameters. When the pitch detector computes the pitch of the input signal, it can further adjust this value in the same way as an automatic tuner plugin, such as MAutoPitch, works. Note that there are no modifications to the input signal, only the pitch is detected differently.

🔌 Enable **Enable**


Enable button enables or disables the auto-tuner.



AUTO-TUNE SPEED  
50.0%

### Speed

Speed defines how quickly the plugin adjusts, when a note has been changed. Higher speed makes the results immediately in tune, but can cause less natural results.









AUTO-TUNE DEPTH  
50.0%

### Depth

Depth defines how accurate the output should be. With 100% depth the output of the detector shall be exactly in tune. With a lower depth the plugin tolerates more deviation.

## Detector panel

DETECTOR Side-chain ? 📄

Min recognized	 50.00 Hz	Max recognized	 1000 Hz
Stabilization	 10 ms	Speed	 75.0%
Detection accuracy	 80.0%	Threshold	 -20.0 dB

Detector panel contains parameters affecting the pitch detection. You can use them to make the detector work well with your audio material.

Side-chain **Side-chain input**

Side-chain input makes the modulator analyze the side-chain input instead of the regular input.

Min recognized  50.00 Hz **Min frequency**

Min frequency defines the minimum recognizable frequency. Any frequency below this value will be considered an error and ignored. For example the fundamental frequency of female vocals rarely goes below 100 Hz, so it may be useful to set this value to this limit to ensure that the detector won't pick up hum or vocal tract noises.

Max recognized  1000 Hz **Max frequency**

Max frequency defines the maximum recognizable frequency. Any frequency above this value will be considered an error and ignored. For example the fundamental frequency of any vocal rarely goes above 1000 Hz, so it may be useful to set this value to this limit to ensure the detector won't pick harmonics as fundamental.

The pitch detector uses a smart search for fundamentals and avoids harmonics. However if you set this value higher than 2200Hz, it's technically impossible to avoid picking harmonics, so this smart search is disabled. This may be useful for scientific audio analysis, but is not desired for common audio processing.

**Stabilization** 10 ms **Stabilization**

Stabilization specifies how quickly can the pitch make bigger changes. This can be useful for more complicated material, such as voice, which often contains short pieces of inharmonic material, which would normally make the detector jump too quickly.

**Speed** 75.0% **Speed**

Speed specifies how quickly the pitch can change. By lowering this value the pitch won't be able to change so quickly, which can improve audio quality when modulating parameters, which are not handling abrupt changes well. It can also be used creatively.

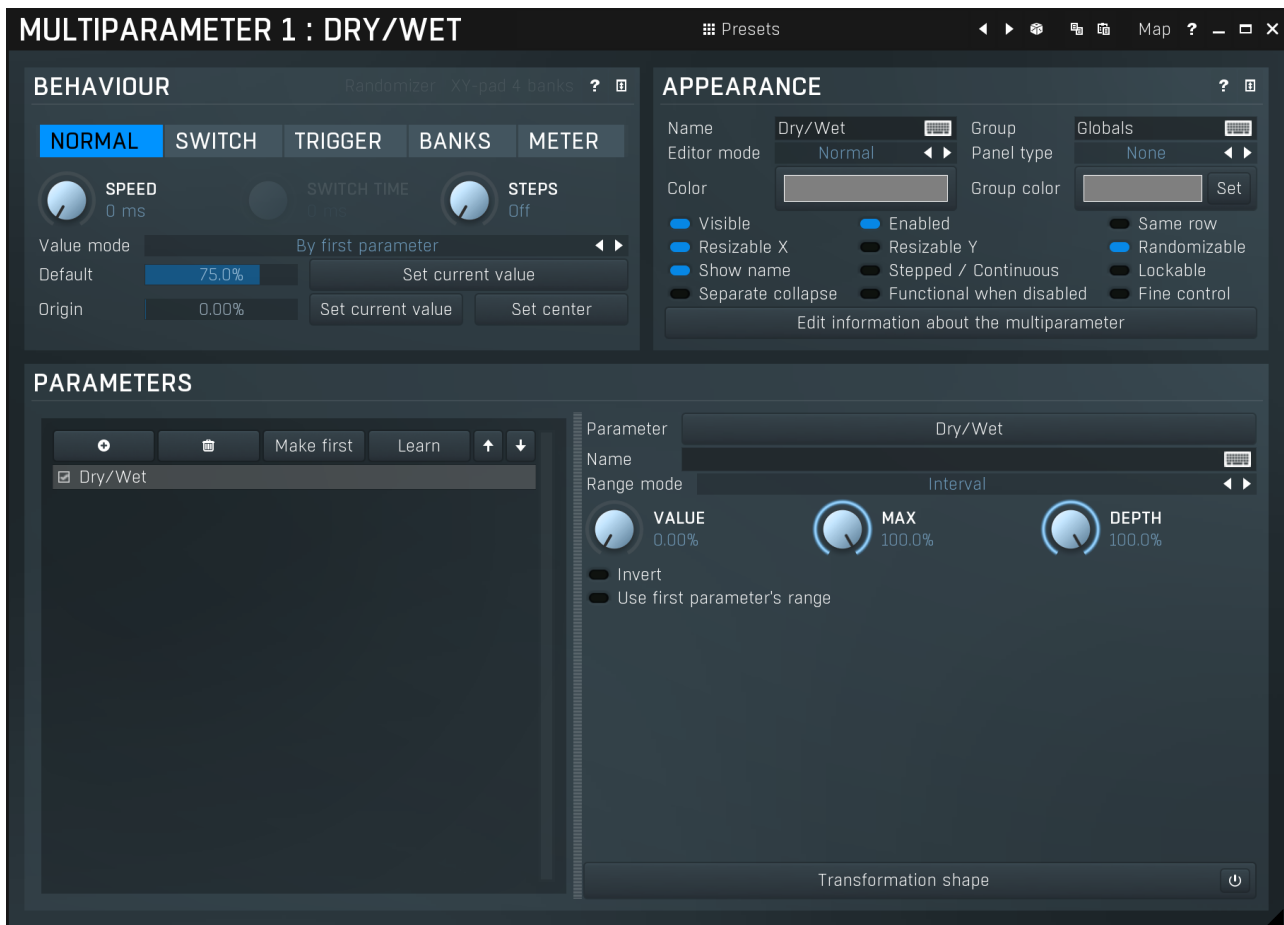
**Detection accuracy** 80.0% **Accuracy**

Accuracy defines how quickly and accurately the detector will work at the expense of higher CPU usage.

**Threshold** -20.0 dB **Threshold**

Threshold controls the minimum input level for the pitch to change. This is provided to minimize artifacts caused by the beginning and ending sections of vocals for example, where the pitch usually fluctuates a lot. It also makes the detector ignore noise in-between actual performances.

# MultiParameter editor



MultiParameter is a powerful structure, which can speed up your workflow significantly and even perform automatic tasks, often useful when performing in real-time for example. Essentially a multiparameter is a controller which controls other parameters, in fact, an unlimited number of them. Each parameter has limits and potentially a transformation curve for more advanced processing. By manually moving the multiparameter (or automating/modulating it) you can control all of the associated parameters at once.

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

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

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

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

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

Now that we have shown you what is possible with multiparameters, it is worth mentioning that they are used extensively for building devices on the easy screens of most Melda plugins. Every multiparameter given a name in the **Information** panel will be shown on the

Easy screen (if the plugin has one). Check our online video tutorials to get more information about **multiparameters and building devices**.

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

⌘ 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

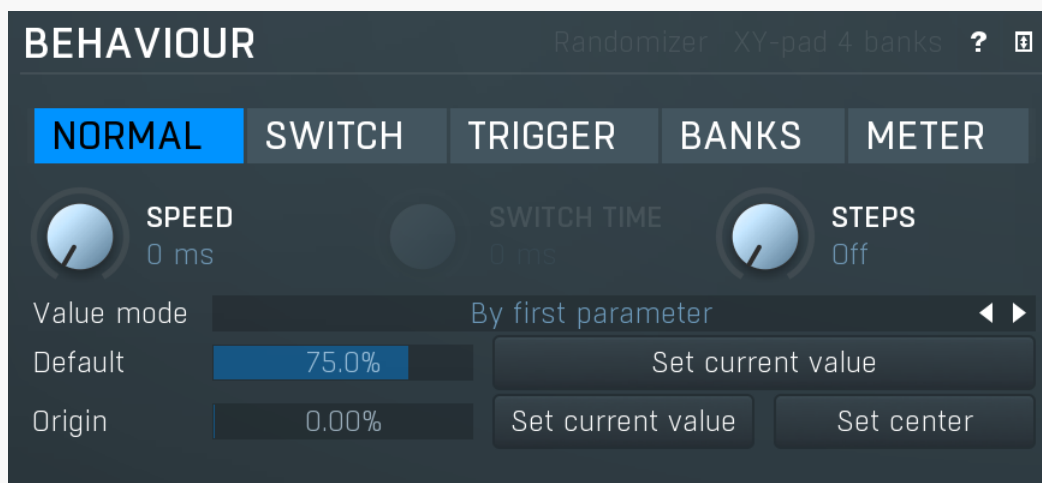
Paste button loads the settings from the system clipboard.



### Map

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

## Behaviour



Randomizer

### Randomizer

Randomizer switch is available only for **Trigger** mode and it makes the multiparameter produce random values for each associated parameters. This is useful to implement some sort of randomization feature, which covers a set of parameters. You usually want to set the **Switch time** to 0, so that the randomization is instant, but longer values may be useful for some creative effects.

XY-pad 4 banks

### XY-pad 4 banks

XY-pad 4 banks switch is available only for **Banks** mode and it lets you create XY pads, that would interpolate between 4 banks you specify. 1st bank belongs to the left top corner, 2nd to the right top, 3rd to left bottom and 4th to the right bottom. Note that in order for this to work, the multiparameter must NOT be the last one and it occupies the next multiparameter as well, so you need to name the next multiparameter and associate it to some parameters, ideally the same ones.

**NORMAL****SWITCH****TRIGGER****BANKS****METER****Mode**

Mode controls the behaviour of the multiparameter.

**Normal** mode makes the multiparameter work like any other control.

**Switch** mode hides the slider and shows a button instead. The button has 2 states. By pushing the button, the multiparameter value starts rising from 0% to 100% over a specified time interval. By pushing it again the value starts falling back to 0%. You could do the same thing having the multiparameter in normal mode and moving the slider from left to right and then back, but mode this performs that automatically and maintains a constant time period.

**Trigger** mode is similar to switch mode, but the button has only a single state and when you push it, the value automatically goes from 0% to 100% and then back without any need to push the button again.

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

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

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

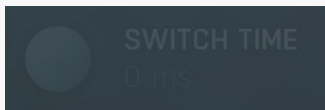
**Meter** mode makes the multiparameter work as a meter. Instead of controlling other parameters it starts following the value of them. You can then use that to implement a simple meter on the easy screen (if the plugin has one).



### Speed

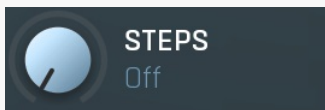
Speed controls the interpolation time. When it is zero and you change the multiparameter value, the associated parameters are adjusted immediately. If this is non-zero however, the actual parameters won't change immediately but will interpolate over time. The speed value is actually the time needed to go from minimum to maximum or vice versa. So if this is 1 second and the current value is say 0% and you click 100%, it will take 1 second for the multiparameter to get there.

This feature is provided mainly because changing some parameter via MIDI or mouse may cause unnecessary zipper noise or inaccuracies due to low MIDI precision. Using the interpolation you can somewhat slow everything down, so that the artifacts become negligible. It can also be used creatively. The default value has been experimentally tested to avoid all artifacts for most parameters.



### Switch time

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



### Steps

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

Value mode

By first parameter



Value mode

Value mode defines the units displayed on the multiparameter.

**Percents** mode lets the multiparameter display percentages between 0% to 100%.

**Percents (-100% to 100%)** displays percentages between -100% to 100%.

**By first parameter** mode uses the current value of the first parameter that is controlled by the multiparameter. For example, if you want to control a plugin gain, but also in addition to the changed gain control other parameters, you may still want to call the multiparameter "gain" and the units should be decibels as usual, not percentages which do not make much sense for such a multiparameter.

**By bank name** displays the name of the nearest bank.

**By bank name interpolated** considers name of all banks numbers. It then interpolates between them and displays the result as a number.

**By bank name interpolated log** is similar, but interpolates the values in logarithmic domain. considers name of all banks numbers. It's useful for units, which are naturally logarithmic, such as frequency.

**By bank number** shows the index of the nearest bank.



Default 75.0% Default

Default controls the default value of the multiparameter. You can edit it directly or just set the MP into its reasonable default and click the **Set current value**. Most GUI components created for the multiparameter respond to right-click by setting the default value in the same way that other parameters do. It is essential for user experience when building your own devices.

Set current value

### Set current value

Set current value stores the current value as the default one for the multiparameter.

Origin 0.00% Origin

Origin informs the GUI engine of the origin of the value. For instance, a default value for panorama is in the center and it is logical that visual elements controlling panorama should somehow highlight the center position. If, for example, you are using a value button to edit the panorama, by default it displays the current value using a bar starting from the left side (being the origin defined as minimum) towards the actual value, but here it is better to display the bar from the center towards the current value, whether it is on the left or right of the center. Therefore the center should be the origin.

Set current value

### Set current value

Set current value stores the current value as the origin for the multiparameter.

Set center

### Set center

Set center sets the center (50%) as the origin for the multiparameter. This is often the case for parameters such as gain and panorama and is the only one supported by knobs, so it deserves a dedicated button.

## Appearance

The screenshot shows the 'APPEARANCE' settings panel for a multiparameter named 'Dry/Wet'. The panel includes the following settings:

- Name: Dry/Wet
- Group: Globals
- Editor mode: Normal
- Panel type: None
- Color: (Color picker)
- Group color: (Color picker) with a 'Set' button
- Visible:
- Enabled:
- Same row:
- Resizable X:
- Resizable Y:
- Randomizable:
- Show name:
- Stepped / Continuous:
- Lockable:
- Separate collapse:
- Functional when disabled:
- Fine control:

At the bottom of the panel is a button labeled 'Edit information about the multiparameter'.

Name Dry/Wet Name

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

Group Globals Group

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

Editor mode Normal Editor mode

Editor mode controls the way the multiparameter are to be displayed on the Easy screen.

**Normal** is the default mode and is represented by a small knob or button.

**Big** mode is similar, but uses a big knob or big button.

**Button** mode displays a value button, which is usually more compact than knobs.

**Check-boxes** makes the multiparameter displayed as a set of checkboxes (also called radio buttons). It is relevant only in **Banks** mode.

**Check-boxes horiz & below** is similar but displays the checkboxes in a single row, hence horizontally. Below mark makes the label underneath the actual checkbox.

**Switcher and Selectors** are useful for selecting a number of discrete values and similarly to check-boxes these are working only in

**Banks** mode.

**Title button** places the control into the title bar of the panel to which it belongs.

**Title enable button** places the control into the title bar of the panel to which it belongs and makes it a standard enable button (which also makes all controls within the panel unavailable if it is itself disabled).

**XY pad** creates a 2 dimensional XY pad control, that edits this multiparameter in the X axis and the next multiparameter in the Y axis. There are multiple versions of this control, all of them differ only by size.

**Spacer** is a helper mode for device design, which doesn't display anything and only keeps empty space.

**Meter** creates a simple meter instead. You will probably want to set the multiparameter to Meter mode as well or attach it to a modulator. Meters don't really control anything and their purpose is purely to get a visual feedback. The meters can be horizontal or vertical and they can be up or down. Up is the usual choice useful for peak meters for example. Down is useful for gain reduction meters.

**Bars start/end** mode creates an editor, similar to step sequencer editor, where each parameter has its own bar. The **Bars start** starts the editor and all multiparameters are then added to it until a multiparameter with **Bars end** mode is found or until there are no remaining multiparameters. Note that this kind of editor doesn't show units and may have several other limitations.

**Order** is a very specific editor for Order modules available in modular systems such as MXXX. It lets you provide an processing order editor on the easy screen. To use it, attach the MP to Order parameter of the Order module and edit the MP information field, so that it contains all the items to be ordered, separated by ';'. The number of items must match the number of items in the Order module, otherwise the order won't work properly. You can also include colors for each item separated by '#'. These can be specified using hexadecimal numbers, or you can even use standard Melda categories of following set: Dynamics, Distortion, Modulation, Stereo, Spectral, Synthesis, Instrument, MDrummer, Reverb, Delay, EQ, Filter, Saturation, Limit, Time, Pitch, FX.

Example of the info: Compressor#Dynamics;EQ#EQ;Limiter#007F7F;Something

Panel type  **Panel type**

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

Color  **Color**

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

For the sake of workflow the colors have predefined meanings. It's highly recommended to follow this standard:

Orange - dynamics  
Green - equalization, filtering  
Brown/yellow - reverb, delay  
Blue - modulation  
Red - limiting, saturation, distortion  
Cyan/yellow - stereo  
Purple/pink - time, pitch, unison...  
Grey - utilities, tools

Group color  Set **Group color**

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

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Red - limiting, saturation, distortion  
Cyan/yellow - stereo  
Purple/pink - time, pitch, unison...  
Grey - utilities, tools

Set **Set**

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

Visible **Visible**

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

example in order to change the GUI somehow.

### Enabled **Enabled**

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

### Same row **Same row**

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

### Resizable X **Resizable X**

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

### Resizable Y **Resizable Y**

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

### Randomizable **Randomizable**

Randomizable option defines if the multiparameter can be randomized on the easy screen. You may want to disable this for input/output gain for example.

### Show name **Show name**

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

### Stepped / Continuous **Stepped / Continuous**

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

### Lockable **Lockable**

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

When the parameter is first locked on the Easy screen it is added to the set of lockable parameters (which are listed in the Global Lock window).

### Separate collapse **Separate collapse**

Separate collapse checkbox makes the panel collapsible separately on the Easy screen. By default it is disabled and that makes the engine find all panels on the same row and collapse all of them or none of them.

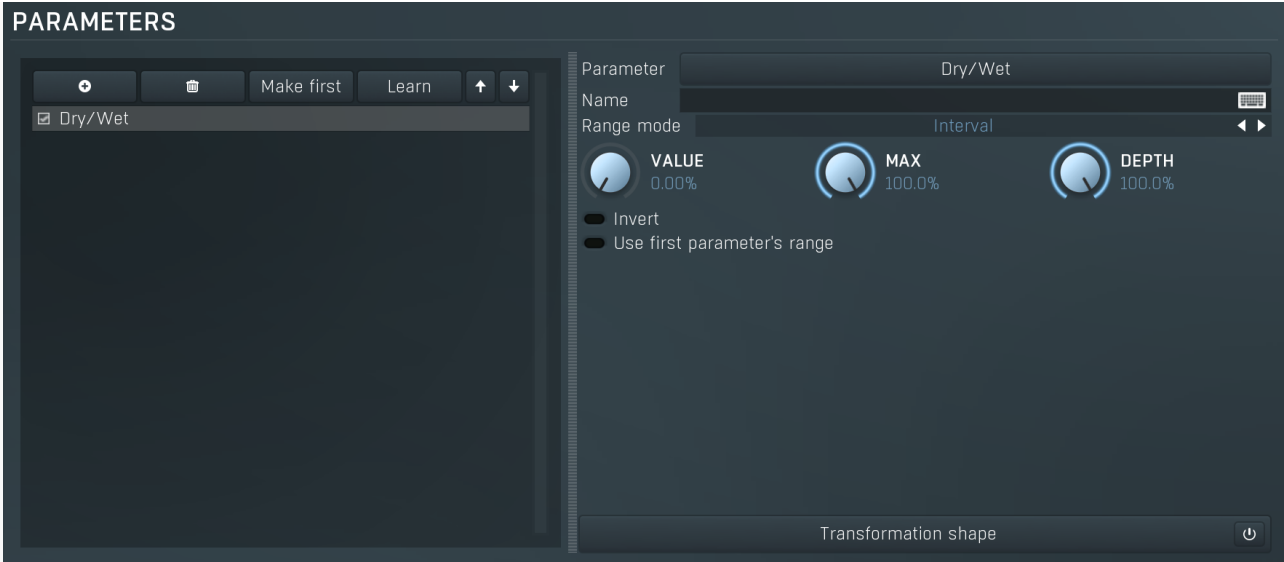
### Functional when disabled **Functional when disabled**

Functional when disabled switch makes the multiparameter work even when disabled. This may be useful in some complex scenarios, where you need to make the MP control the target parameters and only use the **Enabled** flag to grey out the controls on the easy screen.

### Fine control **Fine control**

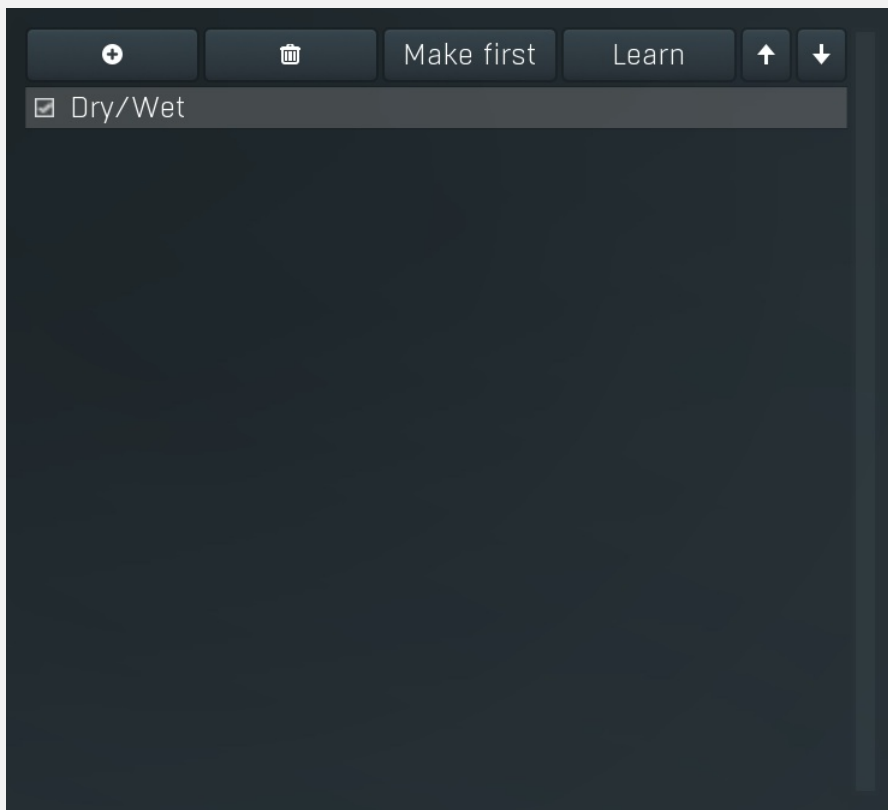
Fine control switch makes the multiparameter editor steps extra small, which is useful, when you need very high precision. This is often handy when using banks mode with many banks interpolating.

## Parameters panel



Parameters panel configures how the multiparameter assigns values to the target parameters.

## ParameterList



### Add

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



### Delete

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

Make first

### Make first

Make first button moves the selected parameter to the first item in the list. This is useful for sake of the **By first parameter** value mode, which makes the multiparameter show the units of the first parameter in the list. Please note that if you have some other multiparameter, modulator or another subsystem access the ranges of individual parameters, this function will reorder them, so these connections will no longer be correct.

Learn

### Learn

Learn button starts or stops the learning. Click it, then move some parameters in the plugin, then click it again. Learning can also be accessed from the global multiparameter menu.



### Up

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



### Down

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

Parameter

Dry/Wet

Parameter

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

Name



Name

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

Transformation shape



### Transformation shape

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

Range mode

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

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



Maximal value

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



Depth

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



Invert

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



Use first

parameter's range

Use first parameter's range makes the parameter display use the same range as the first parameter in the list. This is often useful if want to control the range in some way and apply the range to multiple parameters.

## Cyclic mode

### Cyclic mode

Cyclic mode switches the multiparameter into so-called cyclic mode. If you have say 4 banks, called A, B, C and D, and gradually increase the multiparameter value, it starts with A, then interpolates to B, then to C and finally to D. But after that you cannot interpolate back to A, because D is the last one, the maximum value. In cyclic mode the multiparameter behaves as if there were a clone of A at the end, hence after D is reached, the multiparameter interpolates back to A and creates a full circle A->B->C->D->A. This is handy for example if you use a saw wave modulator to drive the multiparameter and want to repeat the sequence of the banks.

## Interpolate values

### Interpolate values

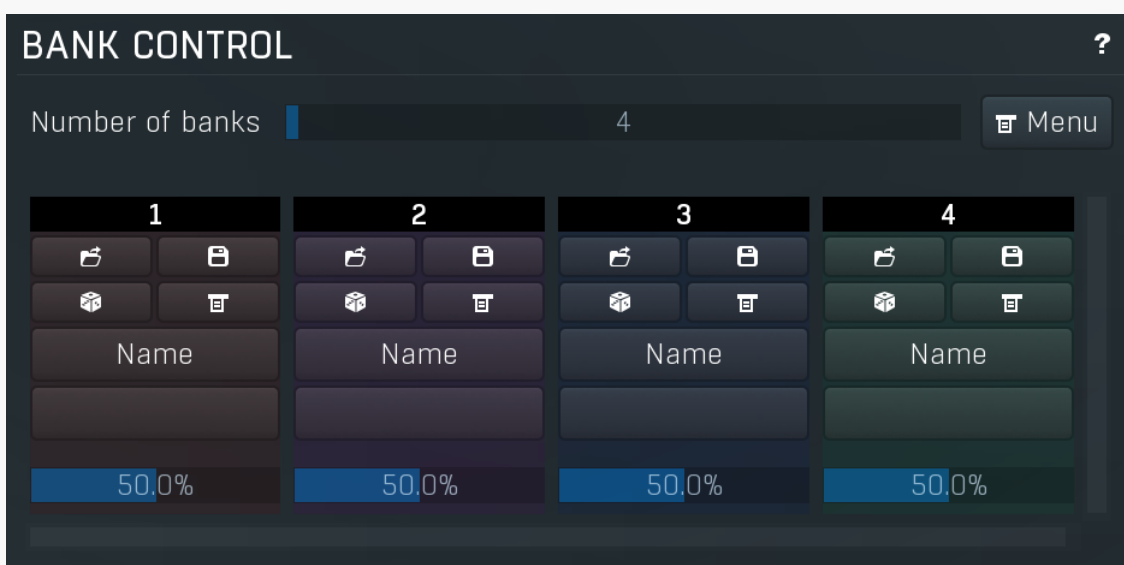
Interpolate values controls if the parameter value is to be interpolated between the bank values or if it will take the value from the nearest bank. For example, when bank A contains the value 0% for the parameter and bank B contains 100% and you set the multiparameter to 30%, then when interpolation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

## Set interpolate to all parameters

### Set interpolate to all parameters buttons

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

## Bank control panel



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

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

## Number of banks

4

### Number of banks

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

## Menu

### Menu

Menu button provides some additional features for processing the entire set of banks.

**Sort banks (up)** reorders the banks so that the values of the selected parameter are in increasing order.

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

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

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

**Auto-gain** (if available) temporarily enables AGC and automatically sets up the main plugin gain to each bank so that all banks provide similar output loudness. To use it, ensure that the main gain parameter is attached to the multiparameter, start playback of your sound

material and press this button. It will take several seconds to complete depending on the number of the banks.

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



### Load

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



### Save

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



### Randomize

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

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

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

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

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

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



### Menu

Menu button provides some additional options related to the bank.

Name

### Name

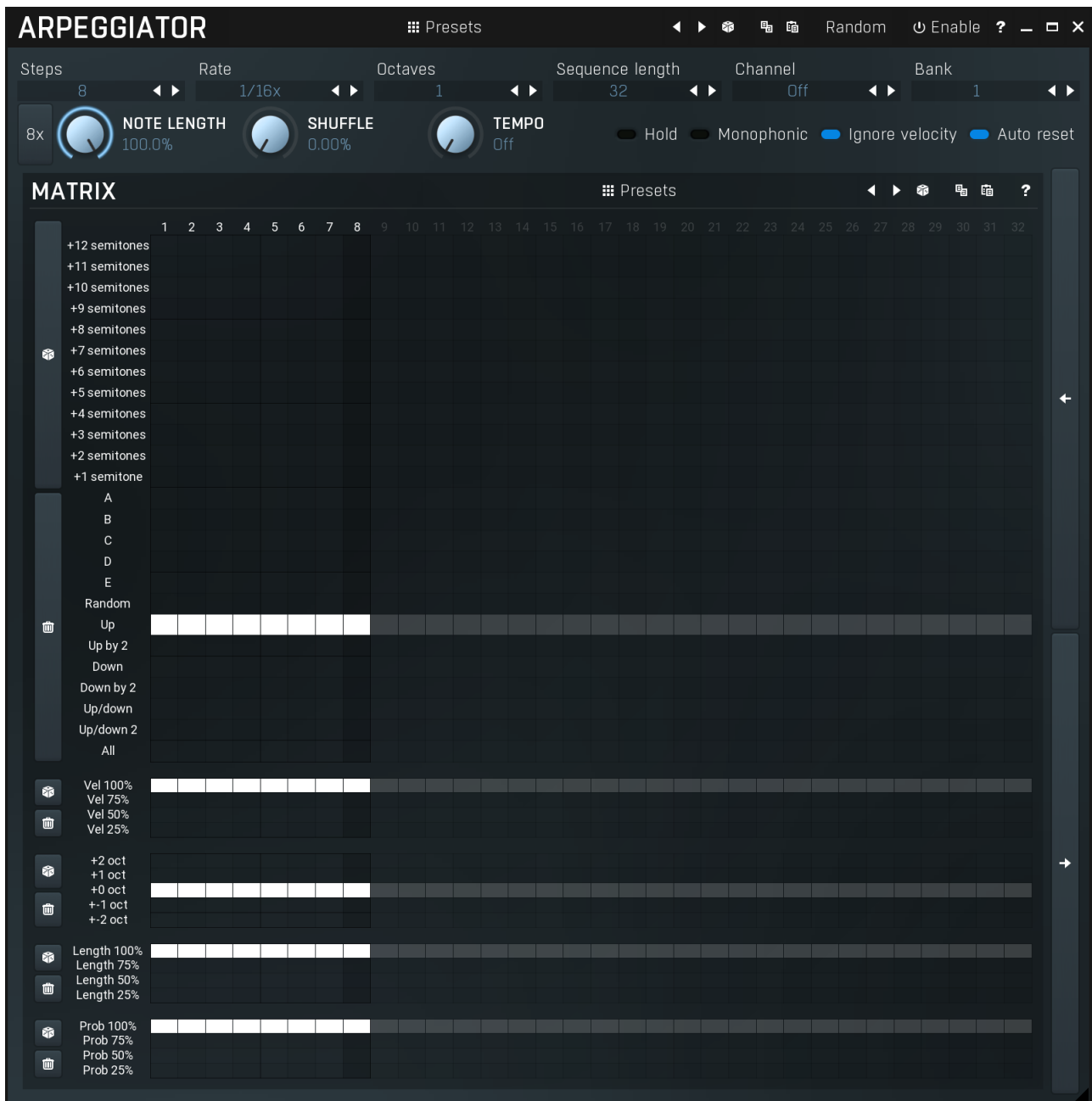
Name button lets you rename the bank.

### Name check

Name check button lets you rename the bank. This is a secondary name used for controls such as checkboxes and selectors if defined.

# Arpeggiator editor





Arpeggiator is a powerful sequencer which transforms the notes that you are playing into potentially long evolving sequences. The idea came from guitar arpeggios, where the left hand is holding the strings, defining the notes to be played, but the right hand is actually playing them, in any order.

This arpeggiator is based on the sequencer matrix, which enables generators in each step. You can imagine a generator is much like the guitarist's finger, but the generator is actually able to play different notes, in fact each generator can play a different note in each step. Now imagine a simple sequence - an empty matrix with just one row and 3 steps, say **Up**, enabled on each step. This would mean that on those steps the **Up** generator is activated and will emit exactly one note, just as a guitarist's finger hitting the string once. And the **Up** generator is playing one note from the chord you are holding, in upwards order. Hence if you hold say C major (C, E, G), the generator is playing C, E, G, C, E, G, ... And of course you may not be holding a chord, but say just a single note, say C. Well, then the sequencer is just repeating C. Of course, the sequence can switch between multiple octaves, say, C0, C1, C2, C0..., depending on the **Octaves** control. Each of the generators works separately and all of them let you create complicated evolving sequences, which may eventually can generate sequences much longer than the 32 steps. In the example above the simple C major chord contains 3 notes, spread between say 4 octaves gets you 12 notes, and you may enable the **Up** generator in just a single step. That way the generator is triggered just once every sequence and each time it chooses the next note. If you enable all the steps then it will take 32 (steps) \* 3 (notes) \* 4 (octaves) = 384 steps before the sequence starts repeating. If you instead use both **Up/Down** and **Up 2/Down 2** generators, again enabled in just a single step, C major chord, 4 octaves, (I'll spare you the boring mathematics) it will take 4,224 steps before the sequence starts repeating. In the default 1/16x synchronization and 120bpm it makes almost 9 minutes! So the arpeggiator lets you build a fully non-repetitive 9 minute song just by pressing 3 tones!

And it can get much more complicated...

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.



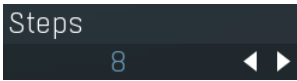
### Paste

Paste button loads the settings from the system clipboard.



### Random

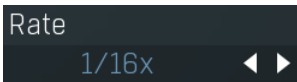
Random button generates random settings using the existing presets.



### Step count

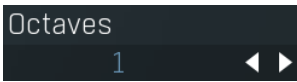
Step count controls the number of steps in the arpeggiator sequencer. Please note that **up/down** and similar generators keep their own order of notes, so for example if you enable **up** for all steps, then it doesn't really matter how many steps you use.

Range: 1 to 32, default 8



### Rate

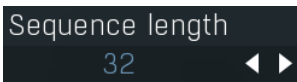
Rate defines the sequencer speed. It is always in sync with the host and this parameter essentially controls the speed. The default value of 1/16 means each step takes 1/16th. So in monophonic mode there would be 1 note emitted every 1/16th, 4 times per quarter note.



### Octaves

Octaves controls how many octaves the arpeggiator can produce. 1 octave means only the notes you are holding can be generated. A higher number makes it possible to emit the same notes in higher octaves.

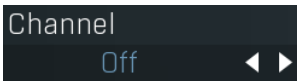
Range: 1 to 5, default 1



### Sequencer length

Sequencer length controls the maximum sequence length for **up, down...** and other generators. It lets you provide some sense of rhythm. For example, if you hold 4 notes and the number of octaves is 3, then 12 notes are generated by say **up** generator before the sequence starts from the beginning. You may want to select a sequencer length of 8, so that it starts repeating every 8 notes, which may be more natural in a typical 4/4 rhythm.

Range: 1 to 32, default 32



### Channel

Channel controls the MIDI channel the arpeggiator produces notes on. Of the value is not 'off', being the minimum, the arp will pass the input notes through and produce the arpeggiator notes on the selected channel. This can be used together with MIDI filtering to make different modules respond to different notes - normal and arp notes.

Range: Off to 16, default Off



### Bank

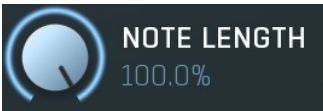
Bank lets you create multiple arpeggiator matrices and switch between using automation, modulation etc.

Range: 1 to 64, default 1



### 8x

8x button increases the note length to be 8x longer. This way the notes may start to overlap. It is useful for arpeggiated pads for example. Please note that because the notes start overlapping, the CPU consumption may rise significantly for some sounds.



### Note length

Note length controls the length of each note. 100% basically produces legato as each note is directly followed by the next one. Lower values makes the emitted notes shorter, leading to staccato.

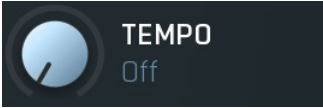
Range: 0.00% to 100.0%, default 100.0%



### Shuffle

Shuffle modifies the timing so that the arpeggiator stops producing straight notes, but rather shuffle (for a value of 100%) or even dotted (for a value of 200%).

Range: 0.00% to 200.0%, default 0.00%



### Tempo

Tempo controls the tempo, which is normally synced to host, but you can unsync the arp and let it flow on its own using this control.

Range: Off to 3200.0000 bpm, default Off



### Hold

Hold switch makes the arp listen only to note-on messages - pressing a key once enables it and pressing it second time disables it.



### Monophonic

Monophonic switch enables or disables the monophonic mode. This doesn't do anything on its own and affects only the editor behaviour. When enabled, any change in the sequence will ensure that the modified steps contain only one note (generator). If disabled, each step may contain multiple notes (generators) and the output would be polyphonic.



### Ignore velocity

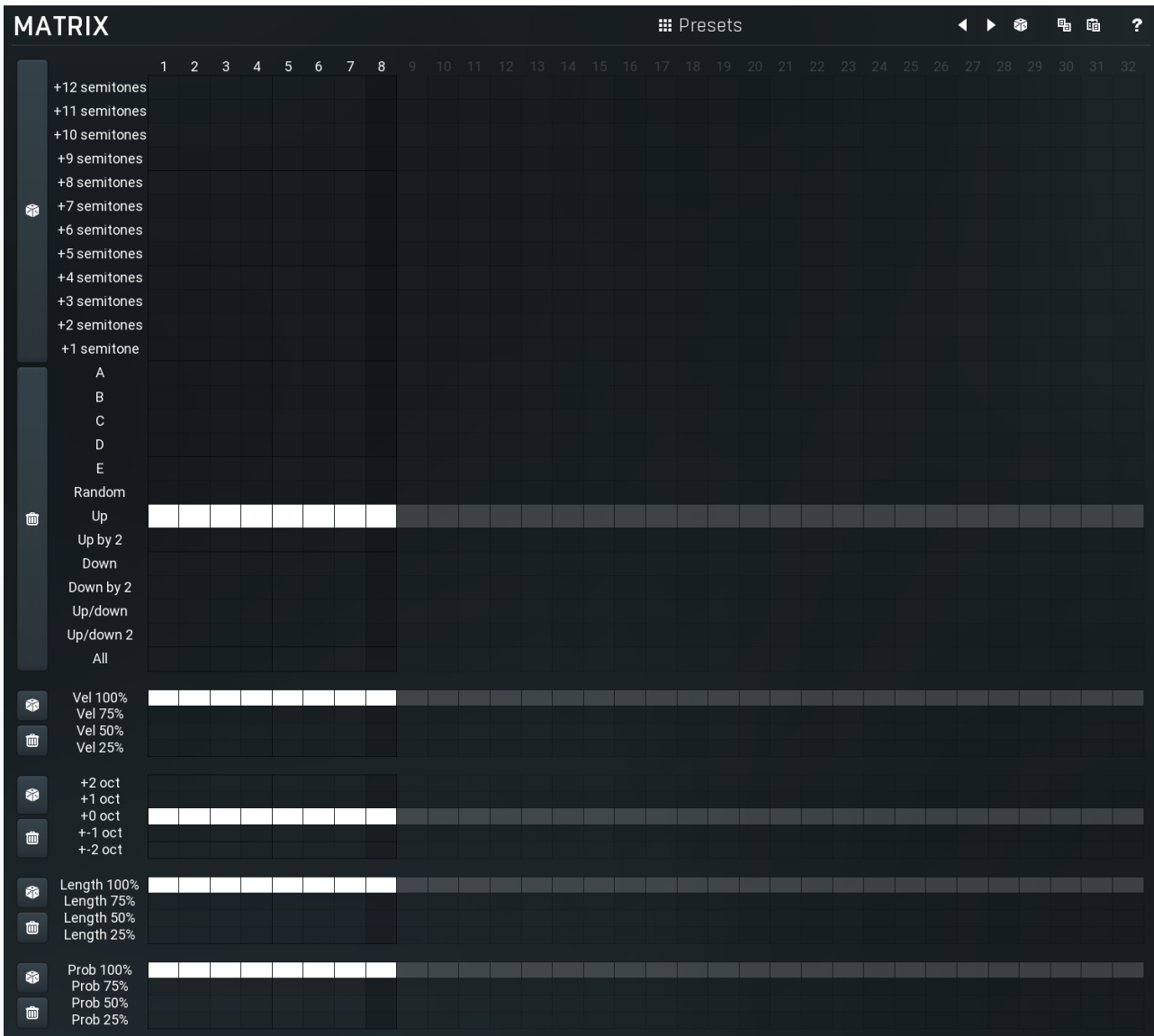
Ignore velocity switch lets the arpeggiator ignore the velocity of the notes that you play. If enabled, the emitted note velocity is defined solely by the velocity matrix. If disabled, the velocity matrix still stands, but the velocity of the input notes still matters.



### Auto-reset

Auto-reset switch makes the arpeggiator reset the synchronization every time a note is pressed.

## Sequencer matrix



Sequencer matrix controls which notes are emitted in each step. Each step is defined in one column and each row enables a particular generator in each step. These generators emit exactly one note in each step; the difference is which note is emitted. In polyphonic mode you can enable more than one generator in each step.

**A, B, C, D, E** generators emit the notes you are playing. A is the lowest note, B the one above it etc. Therefore enabling any of these generators will always produce the same note. You can therefore sequence 5 notes freely. If you are holding less than 5 notes, say 3, then D is the same as A and E is the same as B. These generators ignore settings of the **Octaves** parameter, so the played notes only are emitted.

**+n semitones** generators emit the note transposed upwards from the note A. For example if you are holding C major chord (C, E, G), everything is related to C being the lowest note. The +1 semitone generator then produces C#, +2 semitones is D etc. Therefore enabling any of these generators will always produce the same note. If you have selected A, B, C and +9 semitones in 4 steps, you will hear C, E, G and A (which is 9 semitones up from C). These generators ignore settings of the **Octaves** parameter, so the played notes only are emitted. They are displayed above the **A, B, C, D, E** generators to give a good visual representation of the transpositions.

**Up** generator produces sequence starting from the lowest note up, one by one, including higher octaves. For example, if you are holding C major chord (C0, E0, G0) and the **Octaves** is set to 2, then the generator produces the sequence: C0, E0, G0, C1, E1, G1, C0 ... **Up by 2** is similar, but steps by 2, hence the resulting sequence is: C0, G0, E1, E0, C1, G1, C0 ... **Down** generator is the exact opposite of Up and the resulting sequence is: G1, E1, C1, G0, E0, C0, G1 ... **Down by 2** generator is similar, but again steps by 2, hence the resulting sequence is: G1, C1, E0, E1, G0, C0, G1 ...

**Up/down** generator is a basically a merge of Up and Down generators, going from the lowest note up and then back down, so in the example above the result is: C0, E0, G0, C1, E1, G1, E1, C1, G0, E0, C0... **Up/down 2** generator is similar, but repeats the limit notes, so the sequence is: C0, E0, G0, C1, E1, G1, G1, E1, C1, G0, E0, C0... Combining both Up/down and Up/down 2 generators can produce various evolving sequences.

**Random** generator chooses a random note every time.

**All** generator plays all of the notes every time. It is useful for trance-gate-like effects.

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

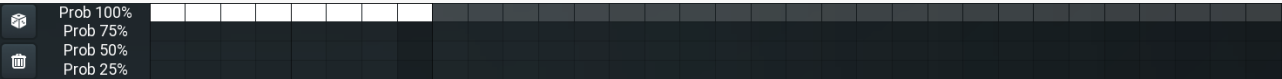
Randomize button loads a random preset.

## Copy

Copy button copies the settings onto the system clipboard.

## Paste

Paste button loads the settings from the system clipboard.



## Probability matrix

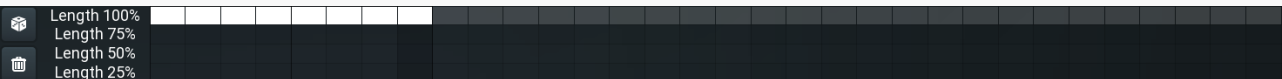
Probability matrix lets you adjust the probability that a note in the step will be emitted. The setting affects all notes in the step, so if you are using polyphonic mode, it is not possible to let different notes in a single step have a different probability.

## Randomize

Randomize button generates a random probability matrix.

## Clear

Clear button sets the probability matrix to its defaults.



## Length matrix

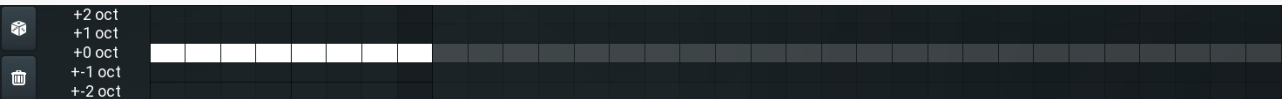
Length matrix lets you adjust the length of each note. This setting works in conjunction with the global **Length** parameter. The setting affects all notes in the step, so if you are using polyphonic mode, it is not possible for different notes in a single step to have different lengths.

## Randomize

Randomize button generates a random length matrix.

## Clear

Clear button sets the length matrix to its defaults.



## Octave matrix

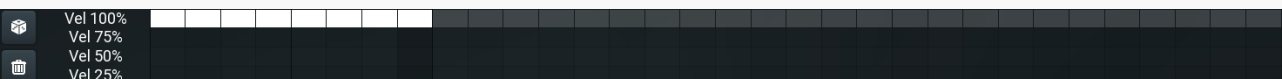
Octave matrix lets you transpose all of the notes in each step by a certain number of octaves. The setting affects all notes in the step, so if you are using polyphonic mode, it is not possible to let different notes in a single step be transposed by different number of octaves.

## Randomize

Randomize button generates a random octave matrix.

## Clear

Clear button sets the octave matrix to its defaults.



## Velocity matrix

Velocity matrix controls the velocity of the notes in each step. The setting affects all notes in the step, so if you are using polyphonic mode, it is not possible to let different notes in a single step have a different velocity.

If **Ignore velocity** is enabled, then the actual velocities of the notes you are holding do not matter and the values in the velocity matrix control the actual velocities of the output notes. These are defined as percentages, but if you are used to the ancient MIDI standard, which uses velocities 0-127, then 100% equals 127, 50% is 63 (or 64) etc.

If **Ignore velocity** is disabled, then the actual velocities are affected by both the velocity of the notes you are holding and the **Velocity matrix**. For example, if you are holding a note with velocity of 60 and the value in the matrix is 50% then the output note velocity will be 30.



## Randomize

Randomize button generates a random velocity matrix.



## Clear

Clear button sets the velocity matrix to its defaults.



## Randomize

Randomize button generates a random sequencer matrix.



## Clear

Clear button sets the sequencer matrix to its defaults.



**Left**

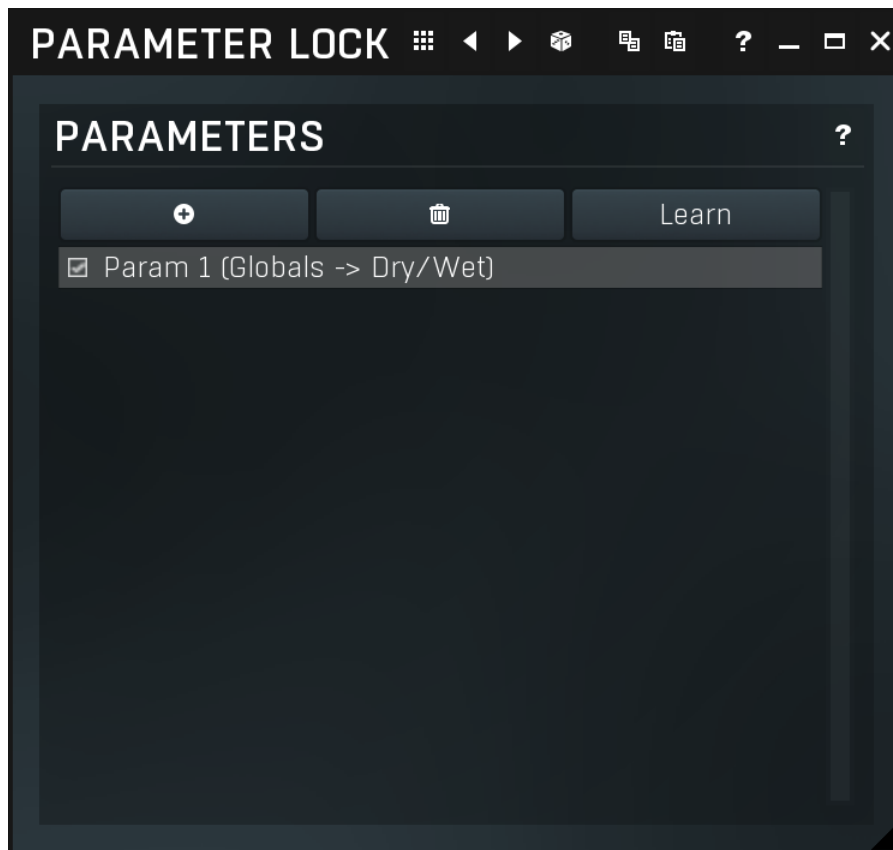
Left button shifts the whole sequence one step to the left.



**Right**

Right button shifts the whole sequence one step to the right.

# Parameter lock editor



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



## Presets

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.

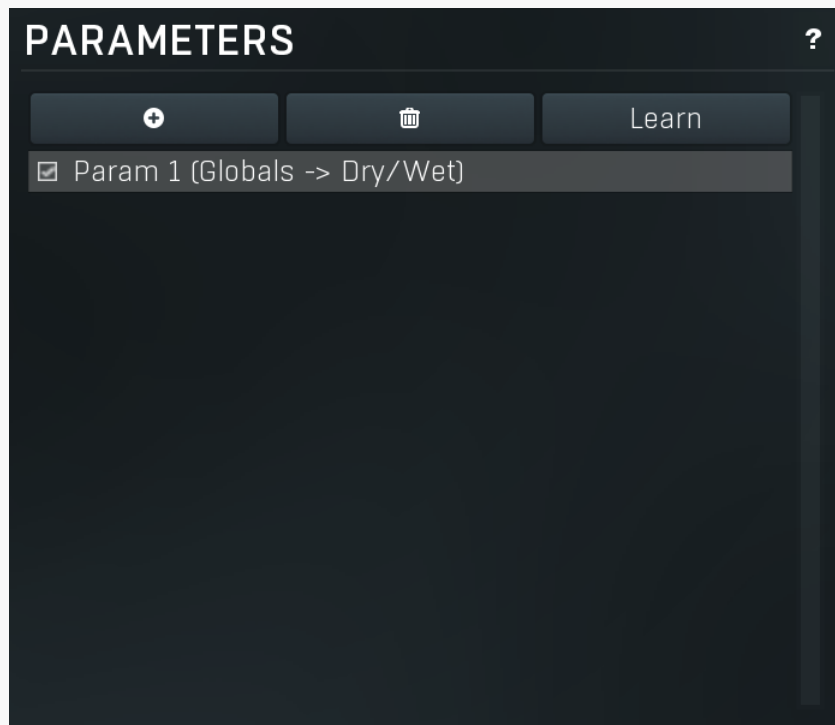


## Paste

Paste button loads the settings from the system clipboard.

## Parameters panel





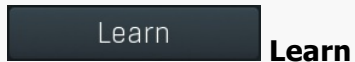
Parameters panel configures the list of the parameters which are locked.



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



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



Learn button starts or stops the learning. Click it, then move some parameters in the plugin, then click it again. Learning can also be accessed from the global parameter lock menu.

# 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

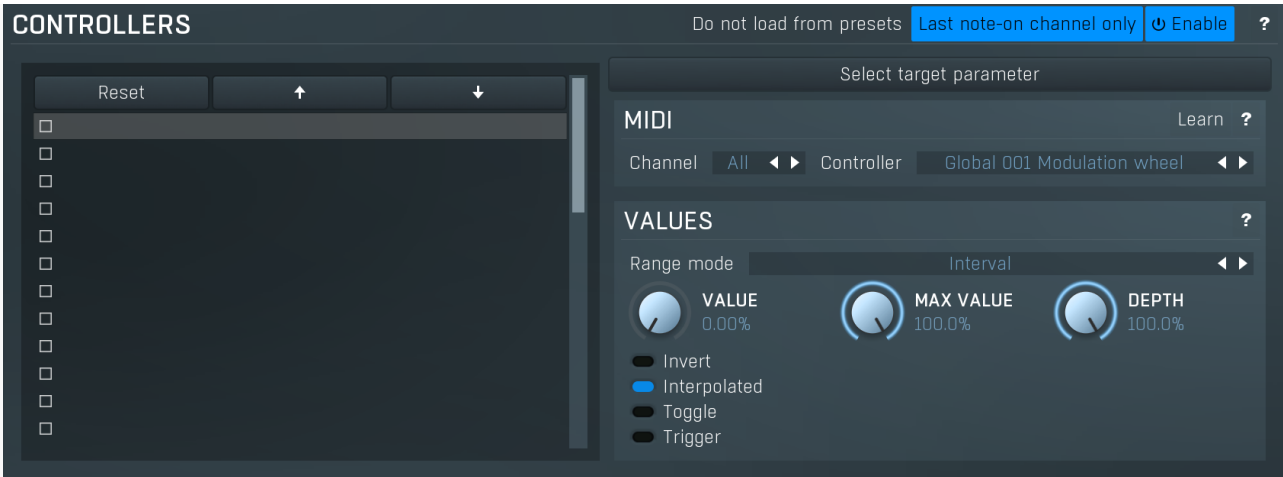
PRESET SWITCH

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

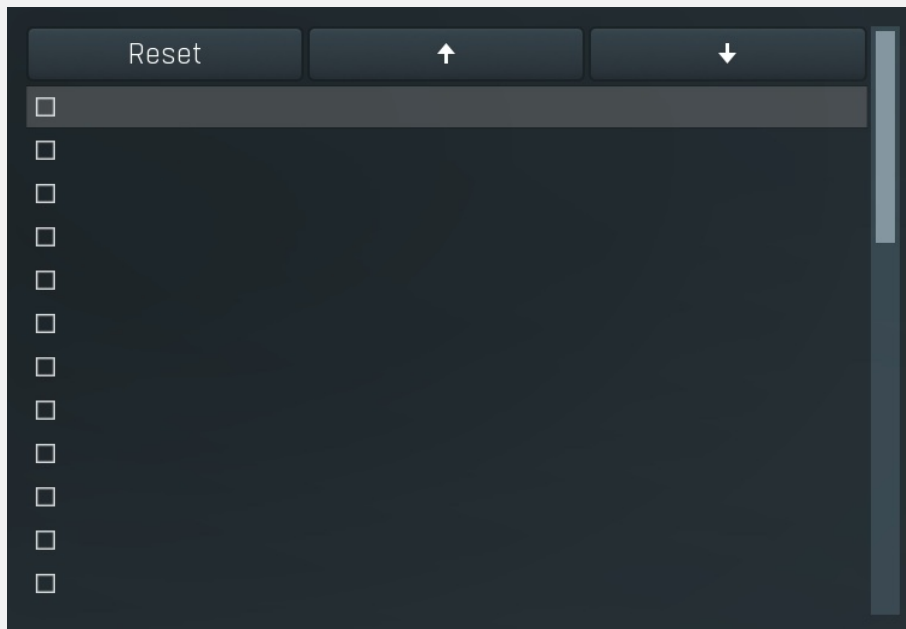
Do not load from presets button disables loading the controllers from presets. This may be handy if you have configured specific MIDI controllers with target parameters and you want to browse the presets without the need to configure them every time. Please note that some presets may rely on specific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

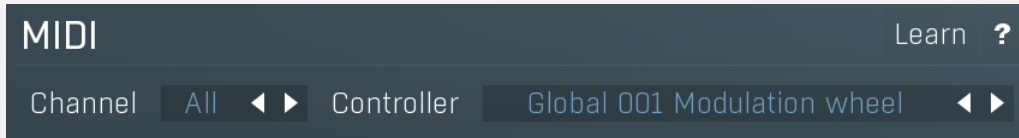
please note that if you have some multiparameter, modulator or another subsystem access the ranges of individual controllers, this function will reorder them, so these connections will no longer be correct.

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.

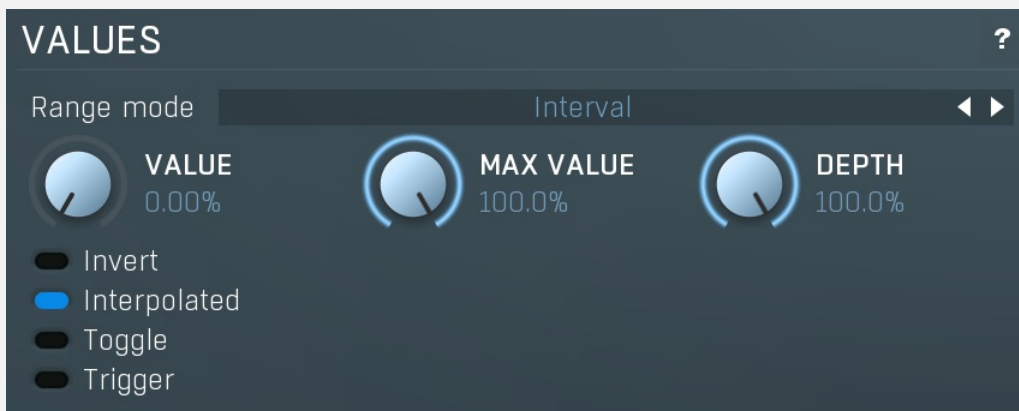
Channel All Channel

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

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

### Maximal value

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

### Depth

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

 Invert

### Invert

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	Off
2	Global 002 Breath controller	Learn	10	Off
3	Global 011 Expression	Learn	11	Off
4	Off	Learn	12	Off
5	Off	Learn	13	Off
6	Off	Learn	14	Off
7	Off	Learn	15	Off
8	Off	Learn	16	Off

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.

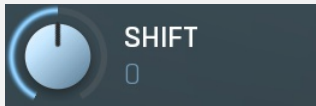
**Linear** converts the notes into frequencies and then transform them into the linear scale from 20Hz to 20kHz.

**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  
Controller

All

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	⌨
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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	⌨
--------	----------	---

### 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.

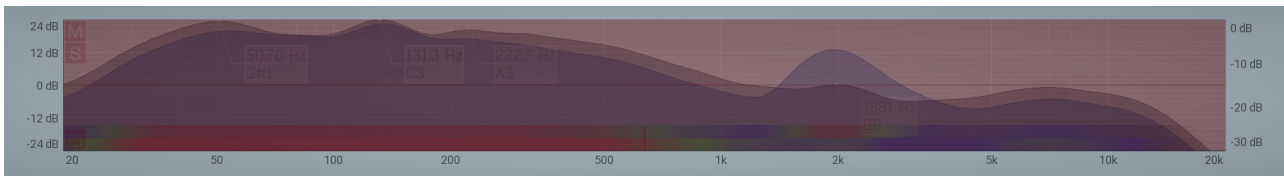
## Value button

Level value -12.0 dB

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.

## Graph editor



Graph editor will show and edit one or more graphs.

# 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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