

# MSoundFactory



## Overview

MSoundFactory is a modular instrument featuring 2 separate sections, the **Generator** and the **FX**, together with a general **Globals** tab.

## Generator and Globals

The **Generator section** is a modular voice generator. In this context a "voice" is the audio produced when a MIDI note is received; a 3-note chord will result in three separate voices. You can use various synthesizers, samplers, filters and even per-voice effects here and this part is processing each voice separately. The processing of each voice can result in 1 to 6 output streams, depending of the routing between the processing modules. Each of these 6 outputs (for all current voices) is routed separately to the FX section. When all voices are rendered (that is output from MSoundFactory), they are all mixed together. You can use the single output edition of MSoundFactory, or the 6-output one (which would enable you to apply different external effects to the different parts of the voices).

Before we explain the FX section, we also need to discuss the main MSoundFactory tab, the **Globals**. While global pitch, volume, panorama and quality are pretty self-explanatory, there are 2 vital parts - voicing mode and global envelope - to discuss. **Voicing mode** controls whether the instrument is polyphonic or monophonic, or operates as a (one-shot) trigger which is useful for percussive sounds. The **global envelope** has 2 effects - firstly it is applied on each of the 6 Generator outputs of each voice before they are mixed and routed to that input in the FX section. And secondly, it defines when a voice ends - once the envelope ends, the voice is finished. The envelope is a very flexible DAHDSR envelope, with several parameters that can be modulated per-voice. Also note that, in the trigger voicing mode, the Sustain section is ignored, with the envelope going from the Decay stage to the Release stage. This is important to understand in relation to the per-voice effects, see below.

## FX

The **FX section** has 6 inputs, matching the 6 outputs of the Generator. The FX section is therefore always processing all voices mixed together, but the 6 audio streams let you process parts of the voice output differently. For example, you may place a drum sampler module in the Generator's lane 1 for initial sound attack and a wavetable module in lane 2. The voice is the audio from both lanes. Then placing a reverb on lane 1 in the FX section would be processing only the drum sampler output.

## Per-voice effects

There are effects in the Generator section as well, these are the per-voice FX. Such a feature is rarely available in instrument plug-ins and it is important to understand what is this for and what the differences are between global FX and per-voice FX.

First of all, per-voice effects process each voice individually. As a consequence the CPU consumption is higher (unless only one voice is played) compared with using the same FX in the FX section. For our 3-note chord, there would effectively be 3 instances of the per-voice effect compared with one in the FX section.

Some of their parameters are available for per-voice modulation (see below). Per-voice FX are inserted on a lane (one of the six audio streams), they can be applied in parallel to the generated audio and produce quite different outputs. And finally, their results may be quite different compared with using the global effects, whenever multiple notes are played at the same time.

Some effects won't make a difference - all equalizers, static filters and non-modulated reverbs for example. These effects are not creating any new frequency content and to spare you complex mathematical details, let's just say that it doesn't matter in which order you use them. In other words EQ->reverb will produce the same output as reverb->EQ. The same is true for mixing, it doesn't matter if you mix track 1 with track 2 and then mix the result with track 3 or do it in a different order. The conclusion here is that if you only want to use a static filter without any modulation at the end of the Generator pipeline for example, then it is better to put it into the FX section, because the result will be the same, but it would require less CPU. It's not true for a reverb though, because the global envelope is applied AFTER the generator. So placing a reverb into the Generator section could result in the reverb tail being cut off by the global envelope.

Most effects are not like that; they produce a new spectral content and it does matter in which order you process them. The most obvious one is distortion, of any kind. Processing a sine wave using a distortion doesn't result in a sine wave anymore, but in a more complex spectrum. But there's a problem - processing 2 sine waves (s1 and s2) mixed together using a distortion doesn't produce the same thing as processing 2 sine waves through a distortion then mixing the results.

*In fact, distorting a mixed signal can quickly get out of hand. In our case with 2 mixed sine waves processing it with a distortion would indeed produce the distorted s1 and distorted s2, but also some complex output featuring additional frequencies such as s1+s2 and s1-s2. With signals more complex than just sine waves the results can be quite brutal. This is why playing complex chords on highly distorted electric guitars is rarely possible, since the output would quickly become dissonant. And that's why the so-called powerchords (a fifth) are exploited so much on electric guitar - the fifth nicely aligns most harmonics, so the complex output is still not dissonant, for the most part.*

A lesson here is that using a distortion on the FX tab may sound good, but it is likely to become ugly and dissonant once you play more than a single note. But distortion is an extremely handy tool for sound design, so the per-voice FX give you a solution - these let you distort each note separately, avoiding all this madness. Unless, of course, that is what you're aiming for.

Of course the per-voice FX let you do quite a lot more than that. For instance, you can modulate the amount of flanging by an envelope. Or you can process an oscillator by one effect and then merge the results using frequency modulation with another oscillator. And... well there are pretty much an infinite number of possibilities.

## Dual modulation system

MSoundFactory features a dual modulation system, a **global** and a **per-voice** one. Each of them is useful in different situations, but for most sound design situations you are likely to exploit the per-voice modulation system.

**Global modulation** system is available via the classic modulators that you know from other MeldaProduction plugins. These can be applied to nearly any parameter in the Globals, Generator or FX sections, and the changes affect all voices identically. Each of the modulators can modulate several parameters.

**Per-voice modulation** is available only for some of the parameters in the Generator section and in the global envelope, but these are processed individually for each voice. For example, you can use these to evolve a filter frequency over time. At the bottom of the **Generator** tab you can see the Per-voice modulation sources panel, where you can configure some of these sources; the other per voice modulation sources are not available here, because there is nothing to configure for them (for example, MIDI controllers, MIDI note properties etc).

Each parameter in the generator plug-in modules that can be modulated has a button just to the left of it. It displays the modulation range and the current value (which works well only if there is just a single voice playing of course). You can simply drag & drop the arrow pointer on any modulation source onto that modulation button. But there's more. Clicking the left-hand half of the modulation button opens a modulation settings window, which contains a 4x4 modulation grid. So every parameter can be controlled by up to 16 modulation sources. Sources on each row in the modulation grid are combined using selectable operators and the rows are summed together to form the resulting modulation value. The depth of each modulation source and the overall modulation Depth can also be set.

That sounds a bit techy, but it's actually pretty simple, let's see it in action in an example: Click on the modulation button of any parameter you'd like to modulate. In the modulation settings only one item of the grid is available, the top left one (also known as "A1"), it is labelled "Off". Click on it and select LFO 1. Increase the overall modulation depth, nothing happens! Now, play a note on your MIDI keyboard so that it starts doing something. You'll hear a sound, see the horizontal bars move in the modulation settings window and see the modulation amount change on the modulation button.

Now the parameter is modulated by the LFO 1. But also 2 new grid items are now available, the second one on the first row ("A2") and the first one of the second row ("B1"). Click on the first item of the 2nd row ("B1") and select Env 1. Now the parameter is modulated by LFO 1 and Env 1 and the outputs of each are summed, which means they do not affect each other. From the beginning of every voice the LFO is modulating the parameter fully, same for Env 1.

Now click on the Env 1 field again and select "Off" to remove the modulation source. Then select Env 1 for the 2nd item on the 1st row instead. Now the parameter is once again controlled by both modulation sources, but not in quite the same way - to see it, well, increase the attack of the envelope Env 1. Press a key and see how initially the LFO is not doing much and in time its effect is strengthening. That's the case when the sources are multiplied. The first source on a row defines the modulation, and the remaining sources on that row say "how much". You can change the multiplication of the two sources to addition, subtraction, maximum or minimum too. Besides the obvious evolution over time that you can get from this, you can also control the amount of modulation by key pressure, and other controllers for example. The available sources are all listed in the modulation source selection window.

Finally, the horizontal blue bars below each modulation source in the grid show the value of that source, red bars below the row show that

combined value of the active sources in that row and the green bar just above the 4x4 grid shows the overall value (and that is the one that actually modulates the parameter).

## Per-voice modulation sources

In this section we are going to explore the various modulation sources available in the Generator. Click any modulation button to see the pre-voice modulation matrix, then click any available modulator button in the 4x4 grid to see the Select modulation source window.

## Controllers section

Controllers are essential for expressive playing, now even more with "3D" (MIDI Polyphonic Expression, or "MPE") instruments such as Linnstrument or Seaboard. By default only the note velocity and pitch-bend are used by the instrument, so every other MIDI controller needs to be mapped in some way if you want to use it. In MSoundFactory we decided to make some of the controllers mandatory for all devices, simply to make everything MPE compliant, and so that everyone with any MPE controller can take advantage of the modern expressive playing.

Note that although the following list doesn't include all the controllers, it does include all the standard per-voice controllers. You can also use global MIDI settings in MSoundFactory to map other controllers to any parameters as well. It wouldn't be "per-voice", nor would most of these controllers. Having said that, **Main controllers** can be defined in MIDI settings and marked as "MPE".

### Aftertouch & Pressure (mandatory)

While velocity defines how quickly the user initially presses a key, pressure is sent by a MIDI controller afterwards and depends on how strongly the user presses the pressure pads under each key. This enables somewhat expressive playing when the user varies the pressure under each key.

### Breath CC#2

Breath controllers are not that popular these days, but the principle is self-explanatory - you control it by breathing into a tube attached to your instrument. It's often used by MIDI controllers simulating wind instruments such as a saxophone or flute, where you literally breathe into them to make them sound, just like with the real deal.

### Expression CC#11

Expression is usually available as a dedicated pedal attached to the MIDI controller. Imagine a guitarist's wah-wah pedal for example.

### Modulation wheel CC#1 (mandatory)

Modulation wheel is available in nearly all MIDI keyboards and hence it is often considered mandatory for expressive playing. Some MPE instruments can also send this as one of the 3D clues.

### Pitch-bend

Pitch-bend usually modifies pitch and in MPE controllers it is actually used for gliding between notes. It is usually not a good idea to remove the default pitch-bending effect by setting the **Pitch-bend range** to 0, but you can use this controller to make the instrument do more than just pitch change.

### Timbre CC#74 (mandatory)

Timbre is used by MPE controllers to send information about vertical movement on the player's finger on the key.

## Envelope section

Envelope section contains several custom envelopes and the global envelope. A general rule of thumb - use as few of them as possible. These envelopes are "calculated" only if they are attached to some parameters (except for the global envelope, which needs to be calculated every time of course), so in order to save CPU it is better to avoid using several identical envelopes for example.

## LFO section

LFO section contains several LFOs (low-frequency oscillators) and Random sources. Both kinds have various speed settings and they mainly

differ in the fact that LFOs are oscillators, and are repeating the same shape all the time, while Random sources are generating a pseudo-random flow, so they are useful for mimicking analog behaviour for example.

**LFOs** have quite a huge range of up to 20 kHz, so these are technically not LFOs anymore, but general oscillators. You can therefore use them as classic LFOs, or they can follow the key frequency for example. That can be useful for some creative sound design applications. However be careful, sound-quality-wise it is always better to use a dedicated module for specific tasks. For example, you can use an Oscillator module, or you can use a Set module and modulate its output value by the LFO. In this case it is obvious, that the Oscillator module will produce a much better audio quality as it is designed for this specific purpose and will completely avoid aliasing, while the Set module wouldn't. Global oversampling could help though.

## Main Controllers section

Main controllers section is similar to the **Controllers section**. However you can consider the 16 Main controllers to be virtual controllers, they are mapped to real MIDI CC controller numbers in MIDI settings. These are sort of proxies to the actual MIDI controllers, so that users can set the Main controllers depending on the MIDI controller they are using and changing map them to the target parameters using the Main controllers. Then, when a user changes the MIDI controller for example, all he needs to do is change the Main controllers.

Main controllers can also be MPE. This means that MSoundFactory will recognize different values on each MIDI channel for that controller. By selecting such a Main controller in the per-voice modulation, each voice can be controlled differently. This option should be used **ONLY** for custom MPE controllers which have the per-voice controllers available.

## Note section

Note section contains many parameters of the note being played or even some specific sources, which are different for each note.

### Note frequency

Note frequency follows the frequency of the voice, including glide. Since modulation always requires a range of 0 to 1, this controller follows the path of highest compatibility by transforming the logarithmic range of 20 Hz to 20 kHz into the 0 to 1 range. Since most frequency parameters follow the same range, this should make everything nicely compatible with each other.

### Note key

Note key is another parameter following pretty much the same thing. It follows the key number (0 to 127 by MIDI standard) and transforms it into the 0 to 1 modulation range.

### Note random 1 & 2

Both produce a random value for each note. They are good for making each voice slightly different, just like the analog devices would. If you set say Note random 1 to modulate multiple parameters, they will all be modulated by exactly the same amount each time.

### Note sequence 1 & 2

Both produce a random sequence, which is the same for every note received. This is useful when using the Unison module for example, which creates several voices with varying pitch, but its main purpose is to make each voice different by more than just the pitch. That's where these 2 sources come in handy. By assigning them to modulate various parameters you will make those parameters different for each unison voice, but overall the main voice will be the same every time.

### Note true random

This one is similar to **Note random**, however it generates a different value for every parameter to which you assign it.

### Note velocity

Velocity may do more than just control volume. It is defined by how quickly the user presses the key.

### Note velocity off

This one is initially 0 and as soon as user releases the key, it becomes defined by how quickly he/she does that. If the controller supports it of course.

## Special section

Here you can find some more modulation sources, which don't really have a specific category.

## Gate

Gate is 1 initially and becomes 0 as soon as the user releases the key.

## Noise 1 & 2

Both produce a noise signal, which can be used creatively, usually to produce some sort of distortion. If you set say Noise 1 to modulate multiple parameters, they will be modulated in exactly the same way.

## Random on note

This is not strictly a per-voice modulation source, since it has the same value for all voices. It holds a random value, which gets changed every time the user presses a key. Be aware that this makes the amount of modulation identical for all voices.

## True noise

Produces a noise signal, which (unlike **Noise 1 & 2**) is different for every parameter to which you assign it.

## Advanced settings

If you feel that a classic feature on the main plugin window is missing, you will probably find it in the Advanced settings available by clicking the **Show advanced settings** button just above the global envelope in the Globals section. There you can find settings controlling how the instrument assigns voices, whether it follows sustain and other controllers, you can configure the glide (portamento) feature, transform the velocity and other controllers, and finally exploit the harmony feature, which lets you play chords by pressing a single key.

## 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**.

If you are an advanced designer, you can also view both the easy and edit screens at the same time. To do that, hold **Ctrl** key and press the Edit button.

# 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, selecting via the buttons or by using your keyboard. You can also manage the directory structure, store new presets, replace existing ones etc. Presets are global, so a preset saved from one project, can easily be used in another. The arrow buttons next to the preset button can be used to switch between presets easily.

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

Presets can be backed up by 3 different methods:

A) Using "Backup" and "Restore" buttons in each preset window, which produces a single archive of all presets on the computer.

B) Using "Export/Import" buttons, which export a single folder of presets for one plugin.

C) By saving the actual preset files, which are found in the following directories (not recommended):

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

Mac OS X: /Library/Application support/MeldaProduction

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

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



### Left arrow

Left arrow button loads the previous preset.



### Right arrow

Right arrow button loads the next preset.



### 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.



## Sound

Sound button generates a note for the synthesizer to play. One click produces a note-on, the next one is a note-off. This can be useful to audition the current settings without using a MIDI keyboard or any other MIDI source to control the instrument.



## 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.

Multi-core

## MultiCore

MultiCore switch lets the plugin exploit the full potential of your CPU by parallelizing rendering of audio for different voices. Note that in certain cases this may actually lower the performance depending on the host and other plugins. The performance can be especially degraded when small processing buffers hence small latency is used. On the other hand the higher number of voices is used, the bigger the gain from the parallelization can be.

Settings

## Settings

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

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

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

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

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

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

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



## WWW

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

Sleeping

## Sleep indicator

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

GLOBALS

GENERATOR

FX

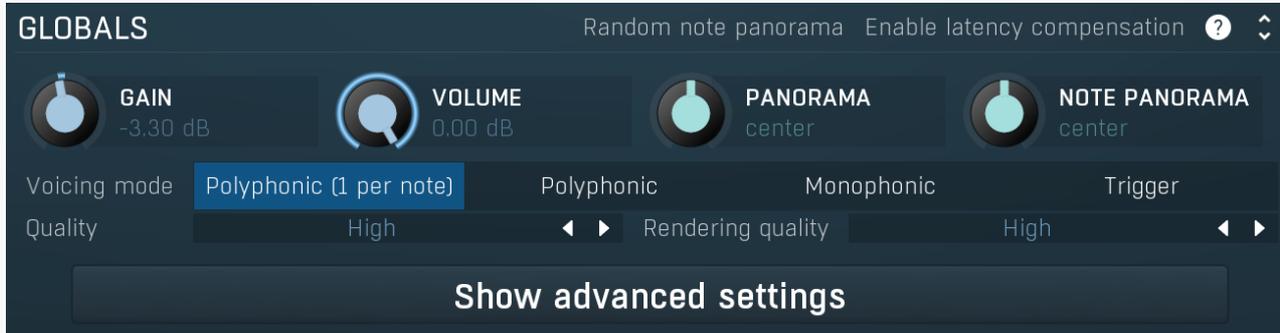


Tab

## selector

Tab selector switches between subsections.

# Globals panel



Globals panel contains some basic global settings, such as output volume or panorama

### Random note panorama

#### Random note panorama

Random note panorama switch changes the behaviour of **Note panorama**. Normally that parameter sets the panorama directly for each note, so you can for example use a modulator to control it. When this switch is enabled, Note panorama is merely a range and the initial panorama of each note will be randomly selected from the range -Note panorama to +Note panorama. Hence it also doesn't matter if the current Note panorama is right or left. The actual panorama of the sound is determined by Note panorama at the point of the note being pressed and by Panorama while the sound is generated.

### Enable latency compensation

#### Enable latency compensation

Enable latency compensation switch enables latency compensation for FX modules. Note that if you use such modules and enable this option, the plugin won't be usable live, since there would be a delay. With this option disabled you can still play live if the latency introducing modules are on a parallel lane. That can still be useful for instance to process delays of the original sound, where the exact timing may not matter.



#### Gain

Gain defines the output gain. It is applied after all the generators and effects are processed.

Range: -48.00 dB to +48.00 dB, default 0.00 dB



#### Volume

Volume defines the output volume adjustment, which is basically an alternative to gain with a different range. It is applied after all the generators and effects are processed.

Range: silence to 0.00 dB, default 0.00 dB



#### Panorama

Panorama defines the output panorama. It is applied after all the generators and effects are processed.

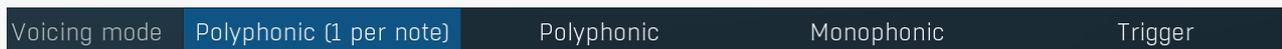
Range: 100% left to 100% right, default center



#### Note panorama

Note panorama defines the panorama-per-note. It is applied after all the generators and effects are performed, but unlike global **Panorama**, this may be different for each note. The value is assigned at the moment the note is pressed and isn't changed later.

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



#### Voicing mode

Voicing mode defines the way in which the instrument works when multiple notes are played.

**Polyphonic** mode is the default mode, which enables the instrument to create as many simultaneous voices as necessary, up to a limit of 256, after that the earliest voice is dropped.

**Polyphonic (single voice per note)** is similar to polyphonic, but ensures that only one voice will be created at any time for each MIDI key played. If sustain is pressed (or other conditions cause multiple notes of the same key), the previous one will be terminated in the same way as when you release the key. This prevents voices stacking up when holding the sustain pedal, however it may cause unexpected results when harmonies are used, as pressing one key simulates pressing multiple keys.

**Monophonic** mode makes the instrument play only one note at a time. When you hold one note and press another one the existing previous voice pitch is changed according to the new note, potentially with gliding. No attack stage or any kind of restart happens, so it sounds simply like the pitch has been changed.

**Trigger** mode makes the instrument completely ignore note release events (Note Offs) and makes the notes be triggered on and off immediately providing a sort of staccato behaviour. Effectively it uses the main ADSR envelope shape, without holding at the Sustain stage (the other modes do hold at this stage). Accordingly, the sounds can still be shaped using this envelope. This is especially useful

for percussive sounds and when used in conjunction with another instrument to form special attack sounds to accompany the sound produced by that other instrument, emphasising transients, for example.

## Quality High

Quality controls the ratio between audio quality and CPU requirements used by the signal generator. In almost all instruments the signal generators are the weakest point, because generating digital signals is extremely CPU demanding if it shouldn't be prone to aliasing and digital distortion. Therefore almost all synthesizers on the market provide inferior quality, sometimes masking it by adding noise and claiming it to be analog simulations. Our generators on the other hand are the finest on the market and in most cases Medium mode will provide nearly perfect sound quality. If you are using saw-tooth waves, frequency modulation and other features creating extremely harmonically rich signals, you may consider using High or even Highest quality.

**Low** quality requires the least amount of CPU time, but you can experience aliasing (which may be used creatively of course). This is especially significant when using harmonically rich generator shapes, such as saw-tooth or square waves.

**Medium** activates an advanced generator, which requires slightly more CPU and memory resources, but minimizes aliasing. This mode is sufficient in most cases.

**High** improves the quality using more advanced interpolation.

**Highest** activates oversampling to at least 80kHz. This usually removes all aliasing and provides a true analog sound, in fact it provides a better than analog sound as it doesn't suffer from either analog or digital artifacts.

**Extreme** additionally uses extremely steep filters for the oversampling to minimize aliasing way below the limits of human hearing. This mode provides technically perfect generators. Its modifications such as **Extreme 16x** in addition increase the oversampling factor, which may be useful to remove distortion & aliasing when using transformations and additional mixing options such as frequency modulation. Please note that you can use the global oversampling as well, but this one is per-voice and may save CPU power as it is not oversampling the filters and FX section.

## Rendering quality High

Rendering controls the quality for rendering. If it is better than the **Quality** parameter, it will be used when rendering your project offline (if your host informs the plugin about it). You can use this option if you want to save CPU power, but still want to maximize the audio quality for the final rendering. Be aware that when different settings are used, the output may sound significantly different.

## Show advanced settings

### Advanced settings

Advanced settings button displays additional settings.

## Global pitch panel



Global pitch panel lets you shift all generators and filters upwards or downwards.

## Octaves 0

Octaves defines the global pitch change in octaves.

Range: -8 to +8, default 0

## Semitones 0

Semitones defines the global pitch change in semitones.

Range: -24.00 to +24.00, default 0

## Pitch-bend range 48.00

Pitch-bend range controls the range in semitones up or down of the global pitch change caused by the MIDI pitch-bend controller. For example, a value of 48 is a range of +/-48 semitones (that is +/- 2 octaves). This is especially useful with MPE controllers, which tend to have a predefined range letting the users slide through the entire range of the controller.

Range: 0.00 to 48.00, default 48.00

## Tuning 440.0 Hz

Tuning controls the global tuning, also called pitch standard, being the frequency of A above middle C.

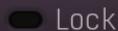
Range: 400.0 Hz to 480.0 Hz, default 440.0 Hz

Default tuning

### Default tuning

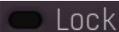
Default tuning button restores the default equal temperament tuning.

Load tuning



### Load tuning

Load tuning button lets you load TUN files containing custom micro-tuning, which will replace the default equal temperament tuning (in which the logarithmic distance between every 2 semitones is exactly the same).



Lock

### Lock tuning

Lock tuning switch makes the engine keep the microtuning when browsing presets.

## Analog panel



Analog panel controls parameters for simulating inaccuracies found in analog devices. To mimic a true analog synth, you need 3 essential elements :

1. Nonlinearities, which you can introduce using various distortion modules such as Saturator.
2. Noise, which you can generate using NoiseGenerator module.
3. Introduce inaccuracies and make various parameters move in time. Especially frequencies of oscillators, filters etc. This is the key element, which makes every voice slightly different giving it the lush 'alive' character. It can be emulated using the Random per-voice modulation, but using the analog drift is much easier. This panel is the master control for all parameters this applies to - all parameters, which can be modulated. Each of these parameters has its own analog drift control in its modulation matrix so you can also control how much each parameter is affected. Many of these controls (e.g. detuning, frequencies, envelope times...) already have set the default analog drift, so in most cases you can simply enable this feature and you are done.



DEPTH

10.0%

### Depth

Depth controls the total amount of analog drifting for all affected parameters. Note that we intentionally made the range very high, from no drifting at all to extreme movement useful for sound effects and experiments.

Range: 0.00% to 100.0%, default 10.0%



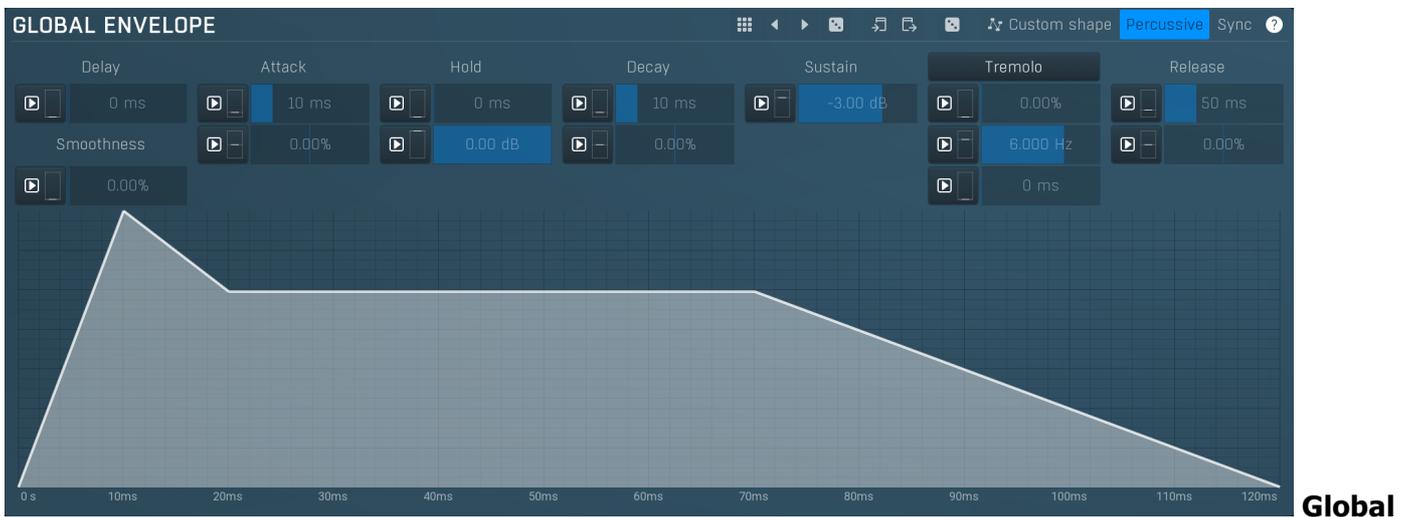
SPEED

0.00%

### Speed

Speed controls the speed of the analog drift, hence how quickly are the parameters randomly changing.

Range: -100.0% to 100.0%, default 0.00%



## envelope graph

Global envelope graph controls the global envelope of the generated voices. Please note that additional oscillators, noise generators and filters can have their own ADSRs too. Our ADSR envelopes are much more sophisticated than classic attack-decay-sustain-release envelopes. Besides these common parameters they also let you control the curvature of each stage. Additionally, there are hold and delay sections ("DAHDSR"), global smoothing and tremolo. You can even use the custom shape mode to define your own attack/release curves.



### 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.



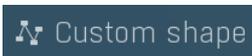
### Paste

Paste button loads the settings from the system clipboard.



### Random

Random button generates random settings using the existing presets.



### Custom shape

Custom shape 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 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 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

Edit    Edit

## 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

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.

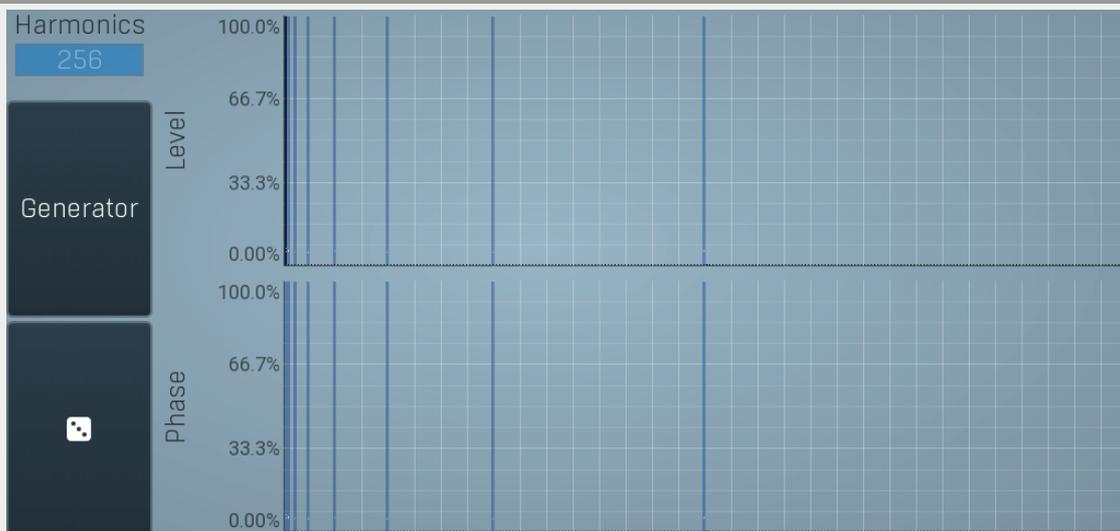


Advanced

### 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

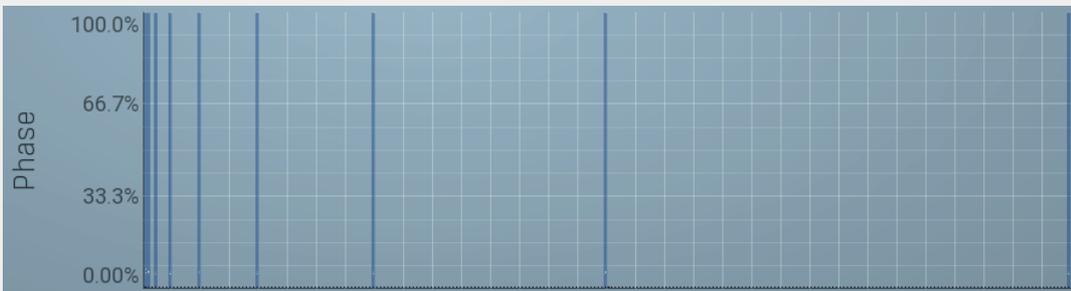
Randomize button selects random parameters for the harmonics generator, so you can use it to get a random sound character instantly. Hold **Ctrl** to slightly modify existing generator settings instead of completely changing them.



**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.

10 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.

-3.00 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.

50 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

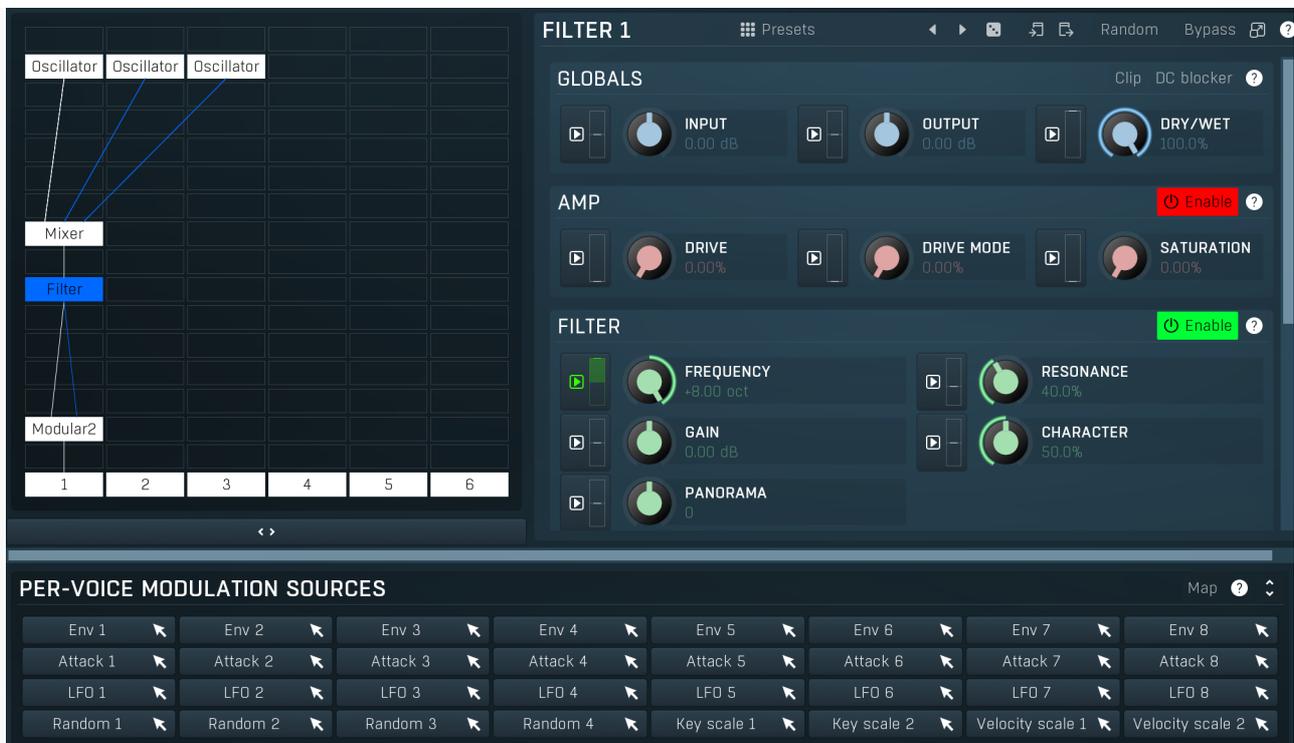
Smoothing lets you smoothen the entire envelope avoiding abrupt jumps. Note that in some cases involving short jumps the results may be a bit obscure.

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.

## Generator panel



Generator panel controls the generator engine, including all oscillators, filters and other per-voice modules.

## Presets

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

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

Presets can be backed up by 3 different methods:

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

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

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

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

## Left arrow

Left arrow button loads the previous preset.

## Right arrow

Right arrow button loads the next preset.

## 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.

## 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.

## Modulation sources panel

### PER-VOICE MODULATION SOURCES

Env 1	Env 2	Env 3	Env 4	Env 5	Env 6	Env 7	Env 8
Attack 1	Attack 2	Attack 3	Attack 4	Attack 5	Attack 6	Attack 7	Attack 8
LFO 1	LFO 2	LFO 3	LFO 4	LFO 5	LFO 6	LFO 7	LFO 8
Random 1	Random 2	Random 3	Random 4	Key scale 1	Key scale 2	Velocity scale 1	Velocity scale 2

Modulation sources panel contains 8 envelopes, 8 LFOs and 4 random sources, that can be used to modulate many of the parameters of the modules in the Generator. Clicking any of the buttons will open the editor for that modulation source. And dragging an arrow pointer directly onto a parameter's **Modulation** button directly assigns that source to that target parameter. Once mapped the modulation can be edited by clicking the target's parameter's **Modulation** button.

### Map **Map**

Map button displays all current mappings of per-voice modulation.

Env 1

### Envelope

Envelope modulation source provides a versatile attack/decay/sustain/release envelope. Unlike traditional ADSR envelopes, this also contains features such as delay, hold, tremolo, custom shapes... Envelopes are used to create some sort of movement in the sound.

### Drag

Drag button lets you quickly associate this modulation source to a parameter by drag & dropping it onto the parameter's modulation button.

Attack 1

### Attack

Attack modulation source provides a simple freeform envelope. While you can do the same thing using envelopes in custom mode, this one is usually easier to use and less resource demanding.

### Drag

Drag button lets you quickly associate this modulation source to a parameter by drag & dropping it onto the parameter's modulation button.

LFO 1

### LFO

LFO is a versatile low-frequency oscillator. Despite it is called LFO, it can actually range to extremely high frequencies. Using very low frequencies combined with some complex oscillator shapes these can be used to produce some sort of repeating semirandom movement as well.

### Drag

Drag button lets you quickly associate this modulation source to a parameter by drag & dropping it onto the parameter's modulation button.

Random 1

### Random

Random source provides a per-voice random signal generator, which is useful to bring unpredictable movement to the sound. If used mildly it can be used to mimic analogue instruments.

### Drag

Drag button lets you quickly associate this modulation source to a parameter by drag & dropping it onto the parameter's modulation button.

Key scale 1

### Key scale

Key scale lets you respond to which key is being pressed. By default lowest of the 128 MIDI keys produces value of 0%, the highest produces 100%, but you can adjust the response freely using the transformation curve.

*You may, for example, want to increase a filter frequency for higher keys. While the filters can generally follow the keys directly,*

you may need some sort of limited/specialized response, where the filter frequency is not constant nor following the keys 1:1.

### Drag

Drag button lets you quickly associate this modulation source to a parameter by drag & dropping it onto the parameter's modulation button.

Velocity scale 1 

### Velocity scale

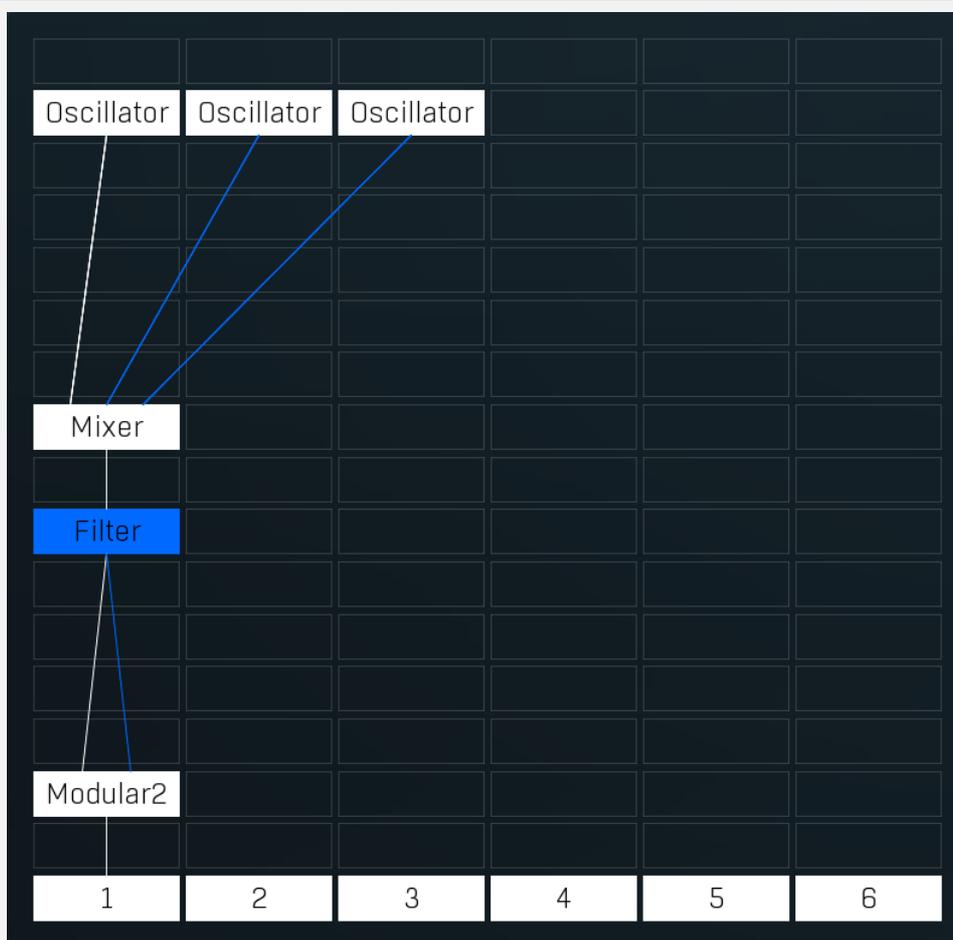
Velocity scale lets you respond to the velocity of the key being pressed. Velocity is generally defined as 'how quickly the key has been pressed'. By default minimum velocity produces 0% and maximum velocity produces 100%, but you can adjust the response freely using the transformation curve.

*You may, for example, want to control a filter frequency depending on the velocity.*

### Drag

Drag button lets you quickly associate this modulation source to a parameter by drag & dropping it onto the parameter's modulation button.

## 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.

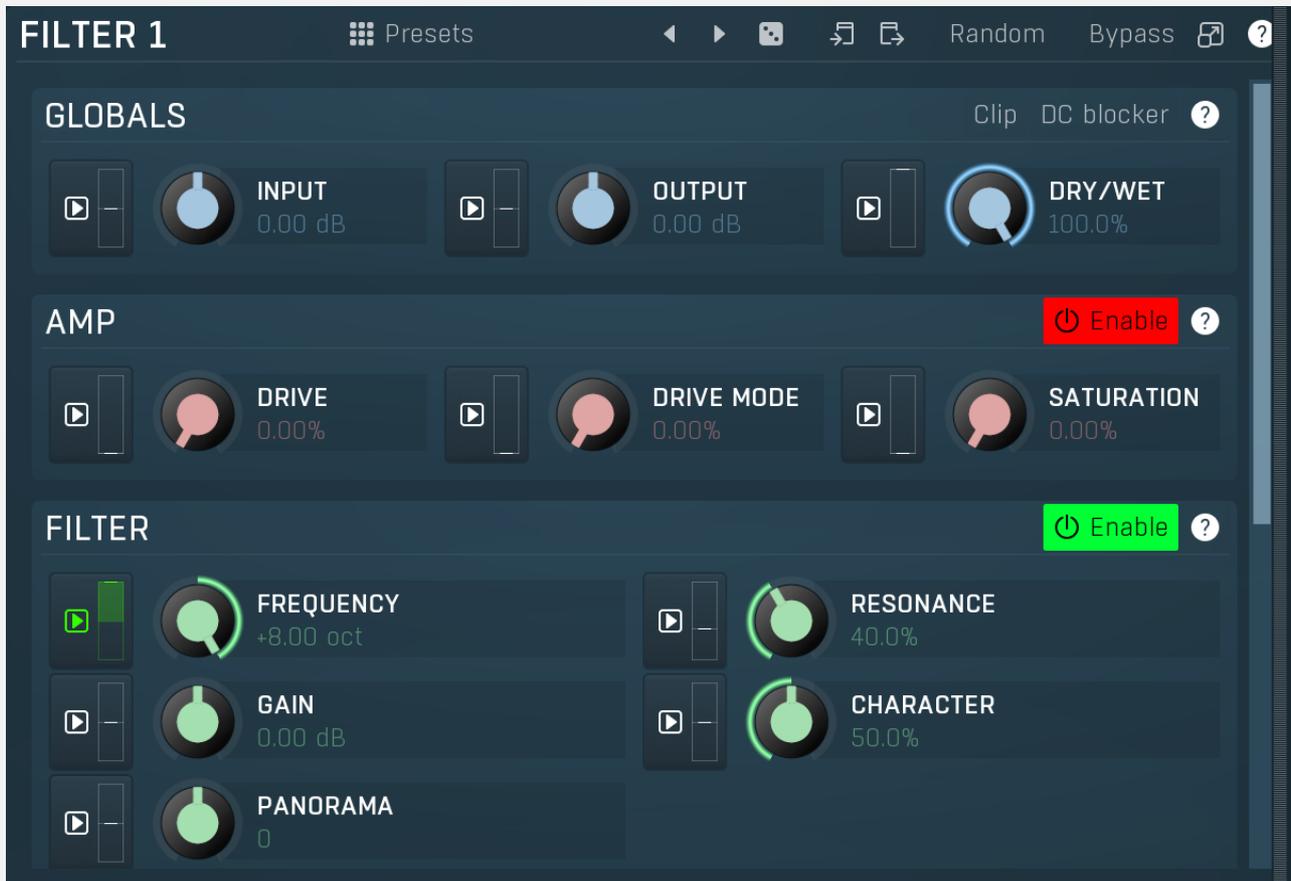
Hold **Shift** and click on the plugin to show its editor in a pop-up window.



**Collapse**

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

## Filter 1



Presets

### Presets

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#### Left arrow

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#### Right arrow

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## Randomize

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## Copy

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## Paste

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## 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.

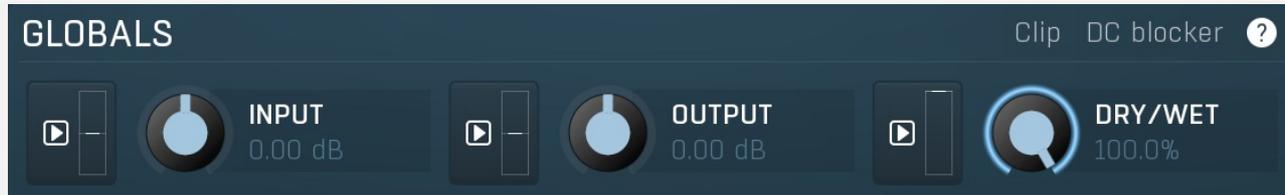
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Holding **Alt** while clicking the button will force the engine to use full randomization, which sets random values for all reasonable automatable parameters. This can often result in "extreme" settings. Please note that some parameters cannot be randomized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

## 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.

## Globals panel



Globals panel contains settings for the overall behaviour.

## Clip **Clip**

Clip enables an optional clipper placed after the filter's output gain. It can be used to remove potential peaks that the filter may cause especially with wild settings. As a hard clipper it can also be used as another distortion stage, however be cautious with it.

## DC blocker **DC blocker**

DC blocker activates the filter DC filter. It will remove content below 20Hz, which isn't audible but is interfering with many processing algorithms. Please note, that when the DC filter is use, the rendered oscillator shapes may start looking significantly different - square waves won't look like square waves at all. This is not an error, the generators will still sound the same, just the content below 20Hz will be removed, which is what causes the signal shape difference.



### Input gain

Input gain defines the input gain that is applied before the amp and the filter. This could be useful for controlling the input level, which, when too high, may cause problems especially with the amp. When you feed a 0dBFS saw wave into the amp for example, there's not much to add into the signal to enrich it, which is what an amp does, so the output signal becomes only some dirty mess and the amp may even start destroying the fundamental. This is the moment when you should probably lower the input gain to get the most out of the amp.

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



### Output gain

Output gain defines the output gain that is applied after the filter. This could be useful for controlling the input to the next stages - the next filter, global saturation. The rule of thumb is - the higher the **Drive** or filter **Gain**, the lower this output gain should be to compensate. For example, you may set a high drive followed by a Sub-X with high gain in the first filter. Its input distortion will

generate lots of higher harmonics as well as change the output level, and the filter would increase the level even more. This could easily be more than +20dB, which when fed to the following filter's distortion or global saturation, may be unusable as each of these nonlinear processors would be immediately overdriven. Just use the filter's output gain to compensate for this by dropping it down.

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

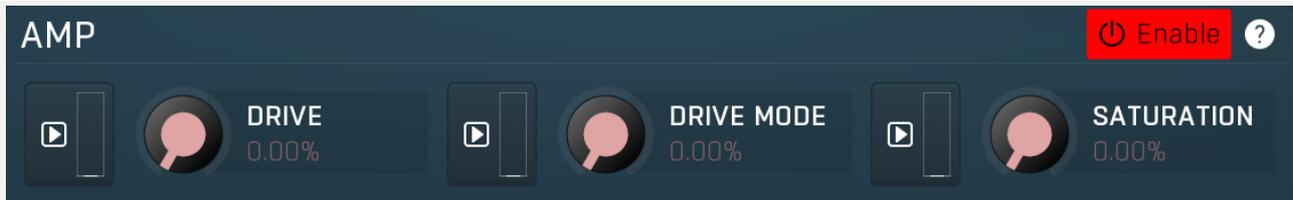


### Dry/Wet

Dry/Wet controls the filter dry/wet ratio. Please note that since most filters are altering phase, you may experience various frequency level changes for values other than 0% and 100%.

Range: 0.00% to 100.0%, default 100.0%

## Amp panel



Amp panel contains settings of the input/output amp.



### Drive

Drive controls the input distortion of the filter. This creates higher harmonics, which are then processed together with the original signal through the filter. Usually the drier the input signal is, the more drive may be used to make the signal richer before any filtering occurs. When applied to an already-rich signal, the results may simply be too dirty. However when applied to a rich yet harmonic signal (such as a saw-tooth wave), only existing harmonics will be added, so the effect won't be creating additional harmonics but rather changing their levels resulting in a different spectrum. It is highly advised to use the plugin's oversampling feature in order to minimize disharmonic components created by aliasing.

Range: 0.00% to 100.0%, default 0.00%



### Drive mode

Drive mode controls the input distortion character. Essentially this controls the levels of different harmonics.

Range: 0.00% to 100.0%, default 0.00%

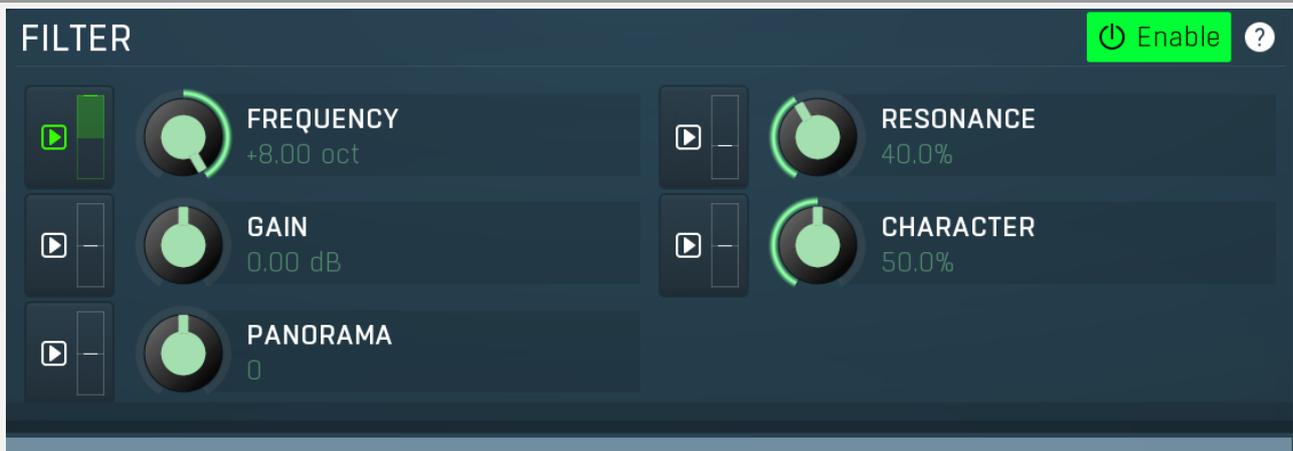


### Saturation

Saturation controls the output saturation that is performed before the output gain. This provides further enrichment performed after the filtering. *For example, you may have a simple sine wave on the input, processed through the input distortion, which adds several harmonics. A filtering using a low-pass filter may then remove most of the higher harmonics content. Saturation may then be used to generate some of the harmonics back again.*

Range: 0.00% to 100.0%, default 0.00%

## Filter panel



Filter panel contains settings of the main filter processor.

 **FREQUENCY**  
+8.00 oct

### Frequency

Frequency controls the filter minimum frequency.  
Range: -8.00 oct to +8.00 oct, default 0.00 oct

 **RESONANCE**  
40.0%

### Resonance

Resonance controls the filter central resonance. Please note that it is used only for some filter types.  
Range: 0.00% to 100.0%, default 40.0%

 **GAIN**  
0.00 dB

### Gain

Gain controls the central gain of the filter. Please note that it is used only for some filter types.  
Range: -48.00 dB to +48.00 dB, default 0.00 dB

 **CHARACTER**  
50.0%

### Character

Character controls the central character of the filter. Please note that it is used only for some filter types. Character affects some additional filter specific features, such as dispersion of harmonics. For polymorph filters character actually controls the internal structure of the filter and any change to this value completely changes the algorithm providing maximum unique sound combinations. Therefore character modulation is not available for polymorph filters, as modulation would create extreme changes.  
Range: 0.00% to 100.0%, default 50.0%

 **PANORAMA**  
0

### Panorama

Panorama lets you shift the filter frequency between channels. Left channel's frequency is shifted down by specified amount, right channel is shifted up, third down etc.  
Range: -4.00 oct to +4.00 oct, default 0

### Type

Type defines the type of filter. Note that different filters may consume different amounts of CPU. By definition a filter does not produce any frequencies which are not already in the signal, hence the name "filter". The difference between the types is how each filter modifies the levels of each frequency. Some filters completely remove certain frequencies, others just change the levels of certain frequencies. If you wish to make the signal richer by creating additional frequencies which are NOT in the signal yet, use a distortion or saturation plugin.

**Low-pass, high-pass, band-pass and notch** filter out some frequencies completely. Low-pass filter, for example, lets all frequencies below a certain limit pass and removes everything above. This is possible only in theory though, so you might say that the higher the frequency is above the filter frequency, the more it is attenuated. The higher the slope is, the steeper the filter is, hence it removes more of the unwanted frequencies. Traditional low-pass filters have a 12dB/octave slope, which means that, for example, if you have that filter set at 1kHz and the Q is configured so that at 1kHz the gain is -3dB (which is usually the default, technical reasons), then at 2kHz (+1 octave) it is -15dB, at 4kHz (+2 octaves) it is -27dB etc. Our filters can provide up to 120dB/octave slope, so it can pretty much kill everything above it within a single octave. High-pass filter works the same way, but kills everything below its frequency. Notch kills everything at the filter frequency plus some adjacent frequency range (determined by the filter's Q value), while band-pass works the other way around - it only lets through the filter frequency and the adjacent frequency range.

**Peak and shelf** filters are similar to those used in equalizers.

**Fade** filters provide cross-fades between low-pass and high-pass filters and other combinations. Use the **Character** parameter to control how much LP and how much HP is used then.

**Harmonics** filters are complex combinations of peak filters designed to process multiple harmonics of the base frequency. Basically if you configure a harmonic filter at say 100Hz, then there will be series of peak filters at 100Hz, 200Hz, 400Hz etc. or (100Hz, 200Hz, 300Hz... if the linear version is used). The **character** parameter controls the level of succeeding harmonics. For example, if character is 0%, then it is basically just an ordinary peak filter. If it is 100%, then there is one filter for all available harmonics, each with the same gain. For something in between, the gain for each higher harmonic is lower than the previous one.

**Linear** harmonics filters affect linear multiples of the base frequency, while normal harmonics filters only affect power-2 multiples, hence octaves above the base. **Swap** versions cause inverted gain for odd and even harmonics.

**Sub-X, over-X and band-X** filters are specialized complex combinations of other filters originally designed for wobbling basses. These mainly combine LP/HP/BP filters with harmonic filters. The **character** parameter controls the distribution of harmonics and should be used simply by trial-and-error.

**Formant** filters are filters emphasizing vowel sounds. There are filters for each vowel and the newest filter, called **Formant A-E-I-O-U** cross-fades between these 5 vowels, depending on the **character** parameter. To get reasonable "talkbox" sounds, it is recommended to use a rich audio signal (e.g. saw wave).

**Comb and diffuser** filters are complex comb filtering processors with pretty wild and fat responses. These range from simple comb filtering to complex almost ambient responses. Each filter uses a different kernel, so it shall be selected by trial-and-error approach. The **Character** parameter controls the internal feedback of the filter.

**Polymorph** filters are generic polymorphic filters, which change its internal structure according to the **Character** parameter and provide a virtually limitless number of unique sound combinations. However, these are usually also the most computationally demanding.

## Pitch mode

Pitch mode controls the way the current pitch is determined.

**Normal** mode follows the current pitch of each voice, including glide.

**Constant** mode ignores the note pitch and behaves as if an A4 (440Hz) note is received each time.

**Ignore glide** mode follows the current pitch of each voice, but ignores glide.



## Analyzer view

Analyzer view contains the different forms of analysis that you can display at once. This mainly consists of a powerful spectrum analyzer and sonogram.

### 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 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.

### Analyzer Rainbow Colors

Analyzer Rainbow Colors lets you see the analyzed sound spectrum in beautiful colors, following the same style as visible light. It ranges from infra-red colors for the lowest frequencies to ultra-violet colors for the highest frequencies in the analyzed audio. If rainbow colors are disabled, the analyzer and graph will be single-colored, following the setup from Settings/Graphs.

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

### Pause

Pause button stops the analyzer temporarily.

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

Env

## 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.

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## Paste

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

# FX panel

The screenshot displays the FX panel interface. On the left, a modular pipeline is shown with six input slots (In 1 to In 6) and six output slots (1 to 6). A 'Dyn Eq' processor is connected to In 1, and a 'Modular' processor is connected to In 2. On the right, the 'DYNAMICEQ 1' analyzer is active. It features a frequency response graph with a red vertical line at 67.65 Hz. The graph shows five frequency bands with their respective settings: Band 1 (208.8 Hz, 0.71), Band 2 (154.0 Hz, 0.71), Band 3 (595.1 Hz, 0.71), Band 4 (1872 Hz, 0.41), and Band 5 (6325 Hz, 87.5%). The analyzer also includes controls for DRY/WET (100.0%), SHIFT (0), INPUT (0.00 dB), and OUTPUT (0.00 dB). The 'ANALYZER' tab is selected, and the 'BANDS' section is visible at the bottom.

FX panel controls the effects engine. The effects are applied to all voices at once, so for example distortion in the filters reacts very differently from distortion in the effects.

The modular pipeline makes dozens of ultra-high quality effects available and you can connect them together in any way that you want. The signal always flows from top to bottom, left to right through various processors. The engine can also generate complete structures, just in case you need some inspiration. The top row contains the available inputs and the bottom row shows the outputs, which are simply mixed together. You can disable any output by clicking on it (that also disconnects the processors in that chain).

Click on an empty field to add a processor to it. Use **right mouse button** (or **Ctrl + click**) on a processor to get some additional settings, mostly about routing audio and MIDI. There are also a few shortcuts: **double click** enables or disables the processor, **Alt + click** deletes the processor, and **Shift + click** displays the processor in a popup window.

The set of processors includes everything from modulation effects, delays, distortions, dynamics, filters etc., and also contains several building blocks - mixer, crossover, LR <-> MS (de)encoder, LFO, channel matrix and many more. Each processor has some inputs and outputs, in most cases it is just 1 input and 1 output as you might expect. There are also processors with side-chain inputs (e.g. Ratio or Dynamics), processors with multiple outputs (e.g. Crossover), processors with multiple inputs (e.g. Mixer) and even processors with no inputs (e.g. Oscillator, LFO). The inputs and outputs are intelligently mapped by default, in most cases the whole structure "connects", but sometimes you may need to use the **right mouse button** to get the additional routing settings.

Most processors are self-explanatory, but a few of them could use some additional information:

**Modular** processor is actually "itself", the whole modular pipeline inside itself. This is useful for example when the structure is starting to get cluttered. Note that you may run into GUI problems - it is impossible to fit that amount of controls onto a screen without huge resolution, so you may need to use the **shift + click** to get the modular processor displayed in a pop-up window.

**Feedback** processor serves as a feedback source. Normally the feedback would not be possible, because the signal always flows from top to bottom. The feedback processor lets you go around this. It has no inputs, but generates a signal anyway - by taking it from somewhere else, from previous processing to be precise. Any of the processors can "generate feedback" into one of the feedback channels. You can imagine that each processor can not only process the signal somehow and pass the results to the output, it can also store the output in one of the feedback channels, just in case some feedback processor would use it.

The feedback processor does not have many parameters, but 2 of them are very important - feedback and delay. First, feedback is the input gain. If it is too high, reaching 0dB, you may expect the typical infinite feedback screaming. The processor provides a clipper, but it will not save you, so be careful when using it. Second, delay - it is physically impossible to have zero delay feedback, not in the analog world and not in the digital world either. So the delay controls how much time it takes to get back. And here is the catch - for higher delays, say above 5ms, there are no problems, but with lower delays the CPU consumption can rise exponentially. There are technical reasons for that and it can never be improved, think about it as physical limitations.

How to use the feedback then? First, create a feedback processor somewhere, it will use the feedback channel 1 by default. Then right-click on any of the existing processors and set "generate feedback 1", that is all. If you do not have any processors there and you just want to use feedback as some kind of experimental delay generator, use some low-CPU processor, such as Utility.

## Presets

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

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

Presets can be backed up by 3 different methods:

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

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

Mac OS X: /Library/Application support/MeldaProduction

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

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

## Left arrow

Left arrow button loads the previous preset.

## Right arrow

Right arrow button loads the next preset.

## 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.

## 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.

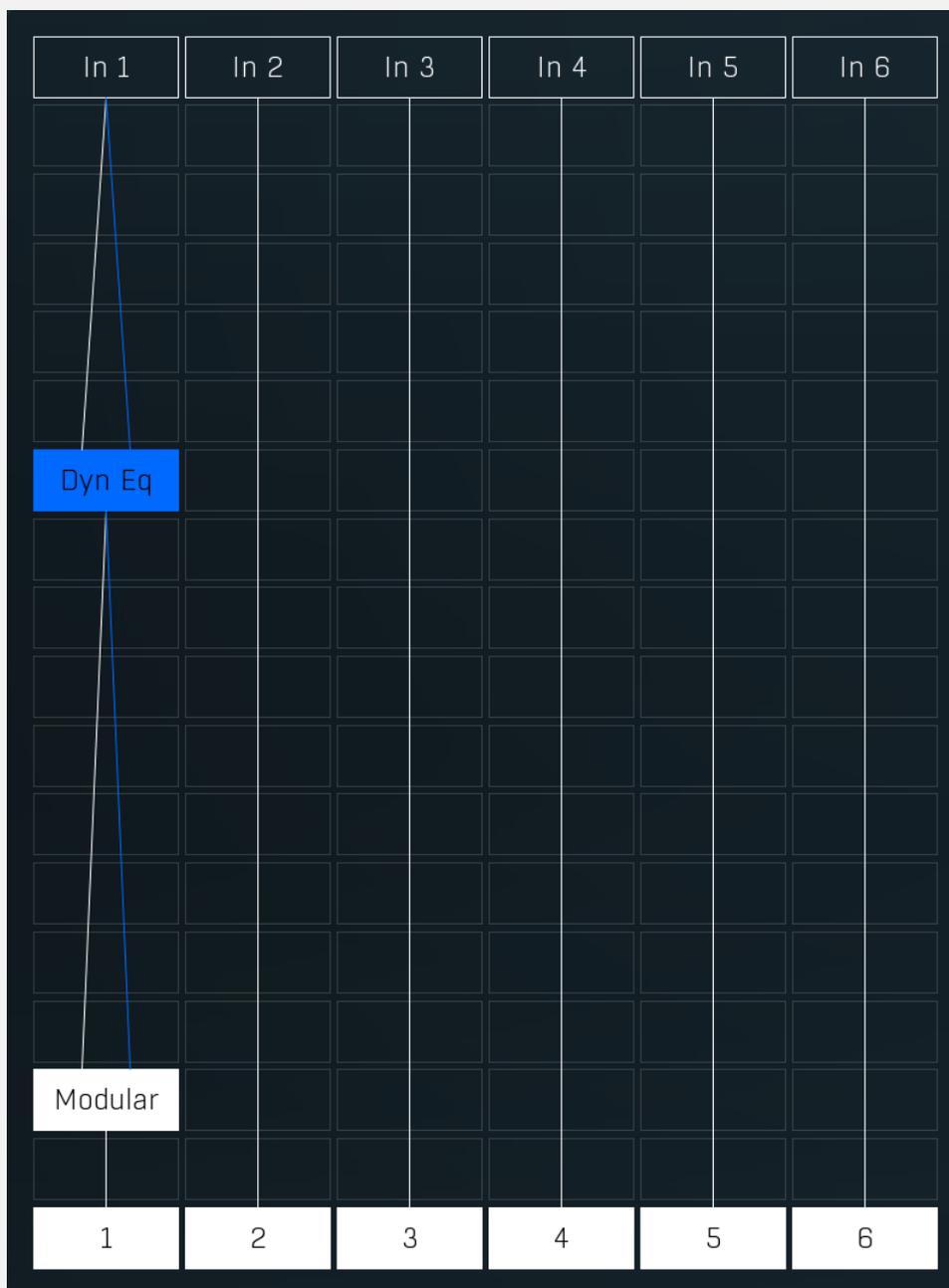
In addition, there are some mouse modifiers that assist this process. The smart randomization engine is used by default if no modifier keys are held.

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.

## 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.

## DynamicEq 1

Filter	1	2	3	4	5
Frequency	208.8 Hz	154.0 Hz	595.1 Hz	1872 Hz	6325 Hz
Q	0.71	0.71	0.71	0.41	87.5%

## Presets

### Presets

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

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Left arrow button loads the previous preset.



### Right arrow

Right arrow button loads the next preset.



## Randomize

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## Paste

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-10

## 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.

## Oversampling

Oversampling 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. 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 oversampling 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 oversampling even they become highly aliased. Now select 16x oversampling and listen to the difference. If you again select 1x oversampling, 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 oversampling, 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 oversampling. 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 oversampling 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.

**Oversampling 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 directly in this popup),

which is not very usable in real time applications. Secondly, oversampling also takes much more CPU power, due to both the processing being performed at a higher sampling rate (for 16x oversampling at 44100 Hz, this equates to 706 kHz!), and the complex filtering. Finally, and most importantly, oversampling 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 oversampling 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 oversampling 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.



## Dry/Wet

Dry/Wet defines ratio between dry and wet signals. 100% means fully processed, 0% means no processing at all. In normal mode only peak and shelf filters are affected correctly, other filters are left at 100% unless the ratio is set to 0%, in which case the equalizer is bypassed.

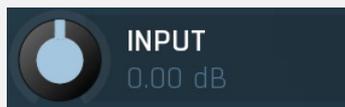
Range: 0.00% to 100.0%, default 100.0%



## Shift

Shift lets you pitch shift all bands by specified number of semitones. It doesn't change the actual band points, but changes the resulting EQ shape appropriately.

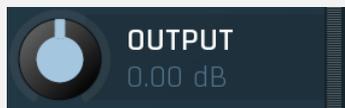
Range: -24.00 to +24.00, default 0



## Input gain

Input gain defines input gain applied before the equalization. Therefore this affects all dynamics-based processes.

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



## Output gain

Output gain defines output gain applied after the equalization. Please note that the real output gain is affected by dry/wet parameter, as opposed to input gain, which is not, because it affects the resulting sound.

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



## Soft saturation

Soft saturation defines amount of saturation simulating analog equalizers.

Range: 0.00% to 100.0%, default 0.00%



## Analog

Analog controls the amount of internal nonlinearities in each filter, typical for analog equalizers. Note that this processing also changes the actual filter shapes, so they won't fully match the displayed graphs anymore when Analog feature is used. It can also require a solid amount of CPU power.

Range: 0.00% to 100.0%, default 0.00%



## Smoothness

Smoothness makes the analyzer smooth out the curve, so it contains less bumping up and down. It approximates the energy in each frequency and the resulting graph should be easier to understand. Also the smoothness affects the automatic equalization. Usually higher value provides more natural results, however you should verify using your ears.

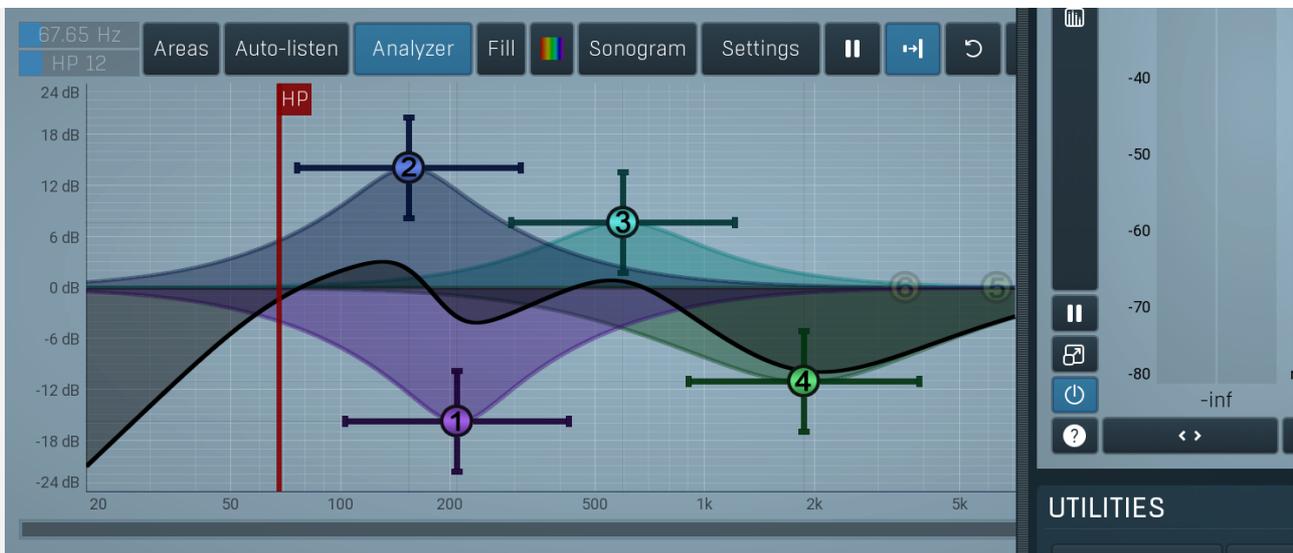
Range: 0.00% to 20.0%, default 5.0%



## Nonlinear gain

Nonlinear gain changes the way the band gain is interpreted. Normally the gain is assumed to be exactly as you set it. But in some cases, e.g. during mastering, you may focus on very small gains and that's exactly what this feature does.

Range: 0.00% to 100.0%, default 0.00%



## Equalizer shape graph

Equalizer shape graph controls and displays the frequency response. There are several bands available, each of them can be enabled/disabled, can be set to a different filter, can have different frequency, Q and other parameters.

Double-click on a band point to enable or disable a band. Drag it to change its frequency and gain. Drag the horizontal nodes to change its Q. Hold **ctrl** key for fine tuning. Click using the right mouse button on it to open a window with additional settings.

The equalizer graph also contains 2 **red vertical lines** on the right and on the left. These are the high-pass and low-pass filters, conveniently placed so that you can perform this often-used task quickly, efficiently and most of all using the highest quality filters available on the market. The high-pass filter also serves as a DC blocker. Slopes for both filters can be adjusted in the title area of the equalizer graph panel.

Typically you want to remove the low frequencies (high-pass filter via the left line) from just about any audio material except for bass and bass drum. Even if the frequencies are not there and are not shown in the sonogram or analyzer views, you may still want to apply a high-pass filter to let the equalization remove any potential low frequency rumble the track might contain. This is always a good practice to clear the resulting mix.

### Areas

#### Areas

Areas button displays settings for the visual areas, which are useful for better visual orientation in the frequency spectrum. These areas are customisable guidelines displayed in the equalizer editor and may contain different octave bands or typical drum frequencies for example. Note that these areas are always only guides, so your particular snare drum may not fit exactly in the very well with the example. In that case it is highly advantageous to use the sonogram or analyzer. Or you can edit your own areas.

### Auto-listen

#### Auto-listen

Auto-listen button enables the auto-listen feature, which temporarily changes the equalizer shape when dragging a band to let you see and hear what that particular band is actually doing. For example, when dragging a peak filter, the equalizer disables the other bands and changes this one to a band pass filter, so that you can focus on the frequencies that the peak filter is modifying.

Also, when this is enabled, you can click anywhere in the band's area (shaded) and the equalizer will let you listen to the frequencies at that position using a band-pass filter. This is great for searching for problematic frequencies for example. Vertical position controls the band-width. You can also hold **shift** to get this feature if auto-listen is not enabled.

### Analyzer

#### 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.



#### Analyzer Rainbow Colors

Analyzer Rainbow Colors lets you see the analyzed sound spectrum in beautiful colors, following the same style as visible light. It

ranges from infra-red colors for the lowest frequencies to ultra-violet colors for the highest frequencies in the analyzed audio. If rainbow colors are disabled, the analyzer and graph will be single-colored, following the setup from Settings/Graphs.

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 button stops the analyzer temporarily.

↔

### 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 analysis

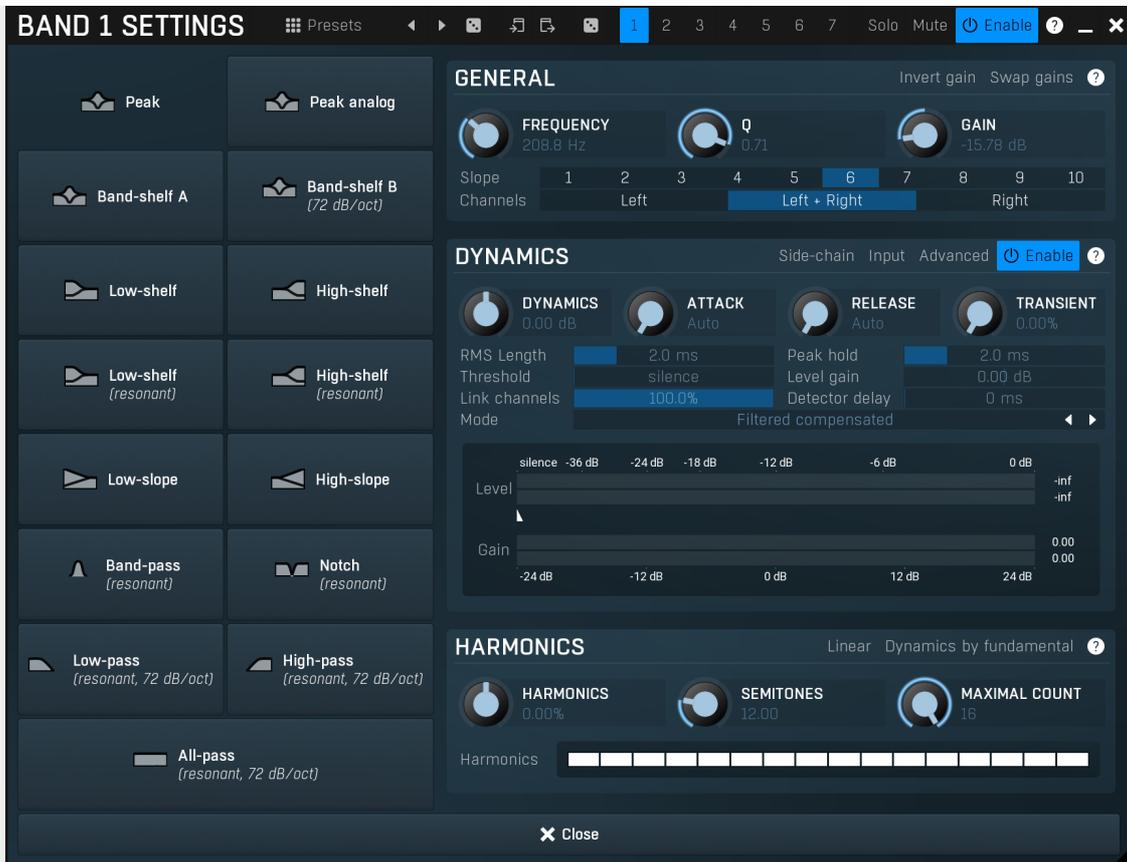
Copy analysis button copies the current state of the analysis into the system clipboard so that you can paste it into another analyzer for comparison. Hold **ctrl** to export the analysis into a CSV file.



### Paste

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

## Band settings window



Band settings window contains settings for the particular band and can be displayed by right-clicking on a band or from a band list (if provided). On the left side you can see list of available filters, click on one to select it. On the right side, additional options and features are available.



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.

## General panel



General panel contains standard filter settings such as frequency or Q. Most of these values are available directly from the band graph, but it may be necessary to use these controls for more accurate or textual access.

**Invert gain** **Invert gain**

Invert gain inverts the gain of the band, e.g. makes -6dB from +6dB.

**Swap gains** **Swap gains**

Swap gains button swaps values between gain and dynamics gain.

**FREQUENCY**  
208.8 Hz **Frequency**

Frequency defines the band's central frequency, which has different meaning depending of filter type.

**Q**  
0.71 **Q**

Q defines bandwidth. Please note that Q is an engineering term and the higher it is, the lower the bandwidth. Our implementation is trying to be more user-friendly, and by increasing the value (thus to the right), the bandwidth is increased as well. The editor still displays the Q value correctly.

**GAIN**  
-15.78 dB **Gain**

Gain defines how the particular frequencies are amplified or attenuated. This parameter is used only by peak and shelf filters.

**Slope** 1 2 3 4 5 6 7 8 9 10 **Slope**

Slope can potentially duplicate some of the filters creating steeper ones. By default, the slope is 1 and this usually means 2-pole 12 dB/octave filters. By specifying 2 you can make the plugin uses 4-pole 24 dB/octave filters instead etc. To see the actual slope of each filter look into the filter type list on the left.

**Channels** Left Left + Right Right

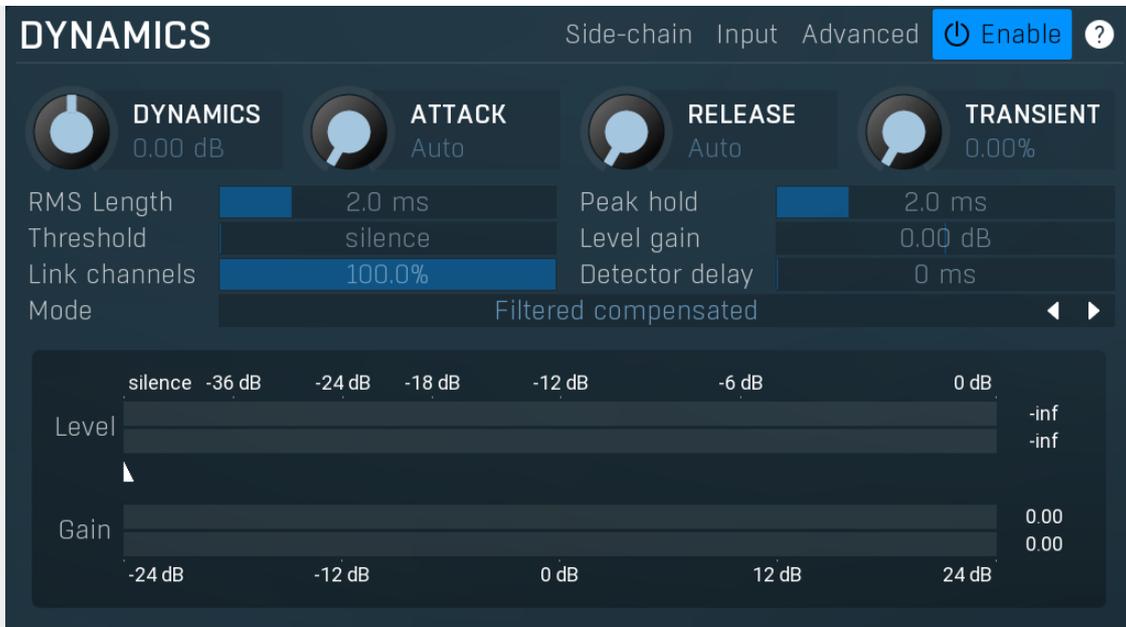
**Channels**

Channels controls which channels the band processes. If the input is stereo (left and right channels, L+R, selected on the toolbar **Channel mode** button), then you can make a band process only the left, only the right, or both channels. Similarly when the plugin is set to M/S channel mode, you can choose between mid, side or both channels.

When one of more bands are set to process a single channel, then 2 EQ curves are displayed, in red for the Left or Mid and in green for the Right or Side. If these are not distinct, then we recommend using a style with a light background for these graphs.

You cannot process left with one band and side with the other, because these are working in different encoding modes. In this case you can easily use 2 instances of the plugin in series, one in L/R mode and the other in M/S.

**Dynamics panel**



Dynamics panel contains settings of the dynamics processing which control how the filter behaves depending on input signal. Normal filters are static, meaning they don't change any features depending on the input signal. If you enable dynamic properties, by making the **dynamic gain** nonzero, the filter will start listening to the level of the input signal. This requires more CPU of course, as such a band is essentially an extremely complex generalized compressor, but the algorithms used are as efficient as it is technically possible.

A dynamic band varies the gain according to the input level. It can listen to the whole spectrum or to just part of it. By default it is driven by the partial spectrum, which it modifies itself, so, for example, when you have a high shelf, it is essentially listening to a high part of the spectrum. You can do many things with such a dynamic processor, but essentially it can work as a compressor or expander. There are many more advanced ideas that you can do and the full power hasn't really been explored yet.

### Side-chain Sidechain

Sidechain switch makes the band measure the input level from the sidechain instead of the input it is processing. This can be used for various techniques, such as avoiding conflicts between bass and bass drum.

### Input Input

Input switch makes the band measure the input level instead of current level in the chain of bands. When this is disabled (default) and the equalizer is processing the bands serially, which means that each band is processing the output from the previous stage, including level measurement. If you enable this switch however, the dynamic processing will be driven by the original input signal instead.

Please note that when **Side-chain** is on, this switch has no meaning, since side-chain has priority.

### Advanced Advanced

Advanced button displays additional settings for this band. These contain some more esoteric features, such as a dynamic transformation shape.

### Enable Enable

Enable button enables the dynamic processing. You can use it to switch between enabled and disabled dynamic processing to check the differences.

### DYNAMICS 0.00 dB Dynamics

Dynamics defines the maximum gain of the filter that could be caused by the input signal. For example, if you set it to -24dB and the input signal contained in the band were very strong, the band will be set to an additional -24dB. This would work similarly to a compressor in that band.

### ATTACK Auto 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.*



### Transient

Transient lets you mix the level follower output with a transient detector output. This lets you follow signal level, transients or both. Note that since transient level is usually lower than level detector's output, **Level gain** is only applied on the level detector's signal, so you can use this to compensate for the difference in level.



### 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.

### Peak hold 2.0 ms 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.

### Threshold silence Threshold

Threshold controls the minimum level above which the dynamic gain actually starts working.

### Level gain 0.00 dB Level gain

Level gain controls the gain applied to the detector, which can be used for example when the input level is too low, so that dynamic processing would be negligible, unless the level is boosted.

### Link channels 100.0% Link channels

Link channels controls how much the signal level for each channel is controlled by the other channels. With 0% the link is disabled and each channel is not affected by the other channels at all. This is suitable to balance stereo channels, for example. With 100% the link is enabled and all channels are controlled by levels of all channels equally (that is the average level of those channels), therefore the processor will apply the same amount of processing on all channels. This is the default in most cases as it preserves relative levels between the channels.

### Detector delay 0 ms Detector delay

Detector delay lets you delay the detector input, hence the band will react later than the actual input signal.

### Mode Filtered compensated Mode

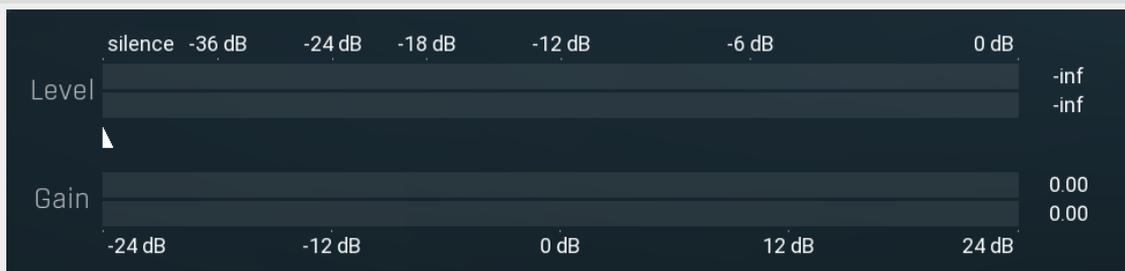
Mode controls the way the band reacts to the input signal. It has no meaning if the dynamic gain is 0dB.

**Filtered compensated** mode is default and it means that the source for measuring input level is a filtered signal with additional compensation. For example, when using a low-shelf filter, the signal is low-passed with a filter with the same settings as the low-shelf, therefore the low-shelf filter is affected only by the signal the low-shelf is actually amplifying or attenuating. Since a low-passed signal with cut-off at 100Hz has usually a much lower level than the one filtered with cut-off at 10 kHz, additional compensation is performed to diminish these differences.

**Filtered** mode is similar, but the compensation is not performed. This may be advantageous for audio materials that do not contain the full spectrum, e.g. a bass line, where the compensation may make things complicated.

**Entire spectrum** mode is the simplest - it simply takes the input signal without any further processing. This may be useful for example to attenuate selected frequencies when the input level gets too high.

## meters



### Threshold

Threshold controls minimum level at which the dynamic gain actually starts working.

## Harmonics panel

# HARMONICS

Linear Dynamics by fundamental ?



Harmonics panel contains parameters of the harmonics - clones of the main band created at higher frequencies derived from the frequency of the main band. This is often useful for removing natural noises, which usually bring some harmonics with them etc.

## Linear **Linear**

Linear button enables the linear harmonics spacing. When the main band frequency is say 100Hz and the **Semitones** value is 12, then in the default logarithmic mode the harmonics are 200Hz, 400Hz, 800Hz etc., increasing by 12 semitones (1 octave) each time. This is suitable because the filters themselves are logarithmic.

However harmonics generated by physical instruments are not spaced in this way. Rather, for a **Semitones** value of 12, they increase by a multiple of 12/12 of the main frequency each time. For example, for a base frequency of 100Hz, they will be at 200Hz, 300Hz, 400Hz, 500Hz etc. In linear mode the harmonics work in this way, but please note that then there is only a limited set of harmonics and Q is modified to approximate a reasonable behaviour, which is not always possible.

## Dynamics by fundamental **Dynamics by fundamental**

Dynamics by fundamental switch causes each harmonic to be driven by the same detector settings as set for the main band. It is disabled by default, which means that each harmonic is literally a clone of the original filter and has its own dynamics detector depending on its own frequency.

Please note that if you want each harmonic to behave in exactly the same way as the main band, you also need to switch on the Input (at the top of the Dynamics panel), otherwise the harmonics would be measuring the signal processed by the main band.



### Harmonics

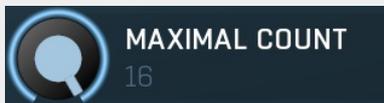
Harmonics defines the gain of the created harmonics. With maximum value (+/- 100%), all harmonics will have the same gain as the main band. A lower value makes the higher harmonics have lower gain. A negative depth will make alternate harmonics have positive and negative gains and is particularly useful for creative effects.



### Semitones

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

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



### Maximal count

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

## Harmonics grid



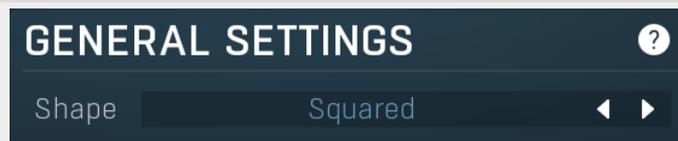
Harmonics grid is useful to turn on/off particular harmonics manually. Click any one to enable / disable it.

## Band advanced settings



Band advanced settings contains additional settings for the band. These contain some more esoteric features, such as a dynamic transformation shape. It can be displayed by clicking the right mouse button on a band while holding **Ctrl**, from the basic band settings window, or from the band list if provided.

### General settings panel



General settings panel contains additional parameters, which are too scientific to be available from the main band settings.



Shape affects the processing shape. The plug-in features specific non-linear transfer shapes which affect the way the level are interpreted. **Logarithmic** mode is the most physical one, increase from, say, -90dB to -80dB and from -10dB to 0dB produces the same difference in the output dynamic gain. However from the nature of it is tends to generate high gains and usually setting a threshold is needed. **Linear** mode on the other hand tends to stay near minimum gains and usually is the most aggressive. **Squared** mode is a compromise between these two. Comparing the three modes, Linear mode requires the least amount of CPU power and Logarithmic requires the most.

### Band-pass panel



Band-pass panel contains parameters of the band pass, which you can use to process the signal that is used measure level of the band additionally. For example, you may want a band at high frequencies to react to bass content; you can do this by placing the band anywhere on the high frequencies and set the low-pass at say 200Hz.

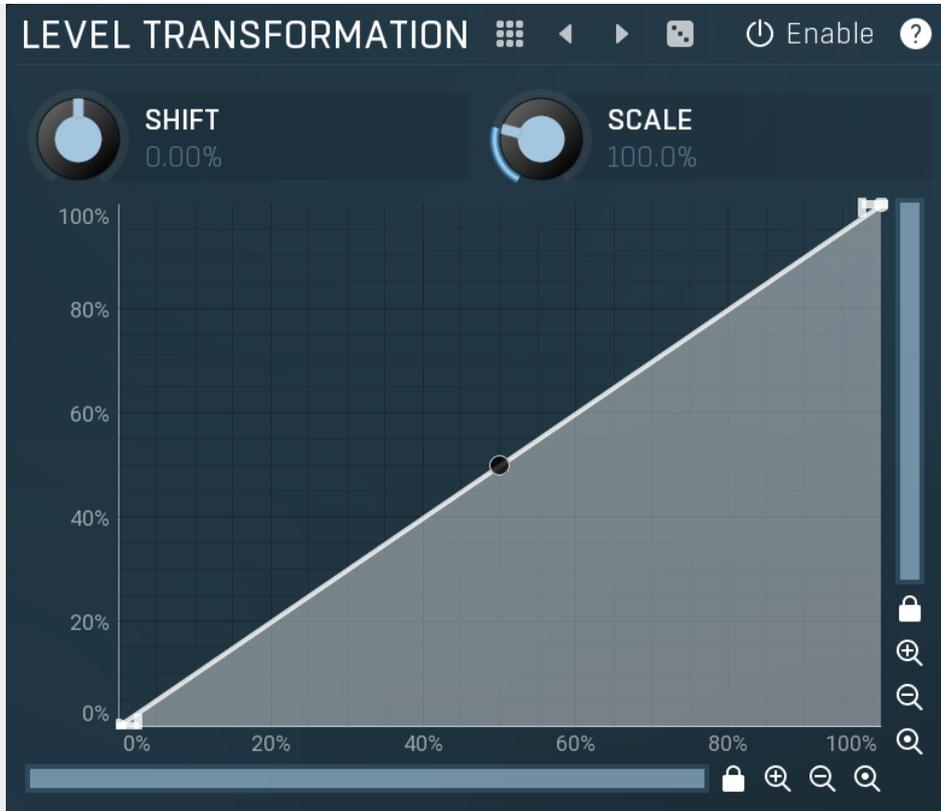


Play button enables the band-pass monitoring and hence could be useful to tweak the band pass.



Enable button enables the band-pass module. It is off by default to save CPU resources.

### Level transformation



Level transformation graph lets you transform the dynamic gain according to the input level. The X axis contains the input level; the Y axis controls the output level, which is then used to set the dynamic gain.



### 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.



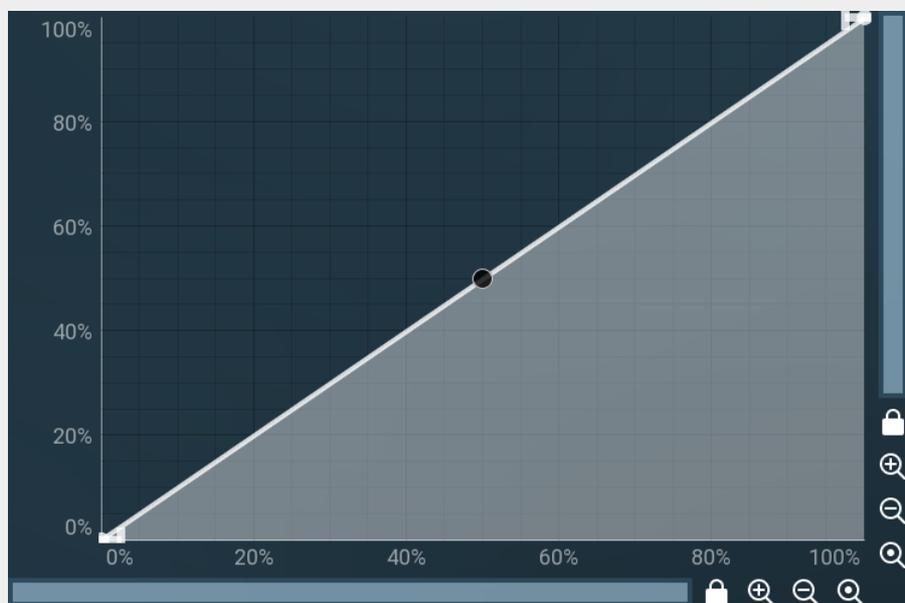
### Randomize

Randomize button loads a random preset.



### Enable

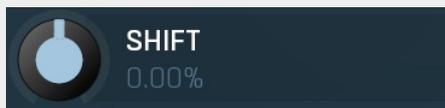
Enable button enables the level transformation module. It is off by default to save CPU resources.



## 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.



### Shift

Shift lets you virtually shift the whole graph vertically. This basically shifts the dynamic gain.



### Scale

Scale lets you virtually scale the whole graph vertically. This basically scales the dynamic gain.

## Link grid panel

	In 1	In 2
Level 1	100.0%	100.0%
Level 2	100.0%	100.0%

Link grid panel controls the linking between the channels; that is, how the input level in each channel affects the levels in the other channels. By default the way channels affect processing in other channels depends solely on the **Link channels** parameter.

Here you can set up a more complicated relationship. For example, you can make the left channel (1) respond to the right channel (2) only and vice versa. Each column in the grid is an input and each row is an output. Each output level is a mix of the factored input levels. For that example above, the values for "Level 1" would be 0% and 100%, and for "Level 2" they would be 100% and 0%.



### Enable

Enable button enables the link-grid module. It is off by default to save CPU resources.

## Bands panel



	1	2	3	4	5
Filter	Low	Low	Low	Low	High
Frequency	208.8 Hz	154.0 Hz	595.1 Hz	1872 Hz	6325 Hz
Q	0.71	0.71	0.71	0.41	87.5%

Bands panel contains the list of available bands along with their basic parameters. You can use it to enable/disable a band, change the parameters and show the band settings window if you do not wish to edit the bands within the equalizer graph panel or if you need to set some values by numeric text entry. The panel is collapsed by default, as it can take a lot of space.

Reset

## Reset

Reset button restores the original equalizer settings.

Invert

## Invert

Invert button inverts the gains of all bands.

Swap g

## Swap gains

Swap gains button swaps values between gain and dynamics gain.

Mod 3

## Band graphs

Band graphs button enables/disables display of band graphs.

# Automatic equalizer panel

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

## Analyze source

Analyze source button starts or stops the source analysis, source defines how you want your audio to sound. In your host, route the source audio only to the plug-in and start playback, then press this button to start the analysis. When the graph (shown as a **red line**) stops moving the analysis is finished and you can press the button again to complete the process.

Alternatively, you can analyze an audio file (WAV, MP3, FLAC etc.) offline by clicking the **File** button and browsing to the file or by dragging & dropping the file from your host or Explorer / Finder onto the **Analyze source** button (this latter method may not work in all hosts, especially on Mac). Three other 3 buttons let you save and load an analysis or even draw the desired response manually.

## Analyze target

Analyze target button starts or stops the target analysis, target refers to the audio that you want to process. In your host, route the target audio only to the plug-in and start playback, then press this button to start the analysis. When the graph (shown as a **green line**) stops moving the analysis is finished and you can press the button again to complete the process.

Alternatively, you can analyze an audio file (WAV, MP3, FLAC etc.) offline by dragging & dropping the file from your host or Explorer / Finder onto the **Analyze target** button (this may not work in all hosts, especially on Mac).

This tip may come in handy - instead of playing the whole song back to get an accurate analysis you can render the song and analyse the output file, which will probably be faster.

## Equalize

Equalize button performs automatic equalization - it adjusts the bands to match the source and target analyses as closely as possible. To do that, you need to have the analyses of both the source and target audio first. When you have both analyses ready, you will see the **red** (source) and **green** (target) analyses in the graph area and this button becomes available.

First get the source analysis using the **Analyze source** button. The Source is the reference audio material that you want your track to sound like.

This is most likely a different track, so you have several options to get this analysis. You can move the plugin to the reference track, perform the analysis then move the plugin onto the track you want to process. Or, open another instance of the plugin on the reference track, perform the analysis and copy the analysis (using the copy/paste buttons, below the A-H preset selectors and A|B comparison button) to the plugin on your track. Or, by saving and loading the analysis (using the **Save** and **Load** buttons). Alternatively, you can draw the desired spectrum or analyse an WAV/MP3/FLAC file by dragging & dropping the file onto the **Analyze source** button.

Secondly, use **Analyze target** button to analyse the audio that you are processing (the **green** line in the graph).

Finally press the **Equalize** button to perform the equalization.

Automatic equalizer tries to match the spectral content of the source analysis graph to the target analysis, producing an equalization curve that aims to make the target audio sound tonally more like the source audio. In most cases the result will be too strong so it is worth considering lowering the **Dry/Wet** parameter to say 30-40% to get a more natural output. You can also make use of the **Smoothness** parameter, above the graph area. Increasing the smoothness before pressing **Equalize** will create a less-pronounced equalization curve.



### Separate

Separate button performs automatic separation - adjusting the bands so that the target does not contain those frequencies that are prominent in the source. This is useful, for example, during mixing to avoid collisions between multiple tracks. To do that, you need to have analyses of both the source and target audio first. When you have both analyses ready, you will see **red** (source) and **green** (target) analyses and this button becomes available.

*For example, say you want to avoid collisions between bass and bass drum. One of them will have to be sacrificed and processed by the equalizer, let's choose the bass. In that case you would analyse the bass drum as the source (the **red** line) and put the equalizer into the bass track afterwards and analyse that as the target (the **green** line). **Separate** would then produce an equalization curve that reduces, in the bass track, those frequencies that are prominent in the bass drum track.*

*Another example is typical when your mix is already busy, but you need to put one more track to it. In this case you analyse the whole mix as the source and your new track as the target. The separation will then allow only those frequencies from your new track that are not prominent in the whole mix; in other words, frequencies that are already prominent in the mix are not affected by those same frequencies in the new track. If this separation were not done, then those frequencies would start colliding with the rest of your mix and that could make it sound muddy and crowded.*

When you have both analyses ready, click this button to perform the separation. In most cases the result will be too strong so it is worth considering lowering the **Dry/Wet** parameter to say 30-40% to get more natural output. You can also make use of the **Smoothness** parameter, above the graph area. Increasing the smoothness before pressing **Equalize** will create a less-pronounced equalization curve.



### Reset

Reset button clears both the source and target analyses.



### Remove resonances

Remove resonances button takes the target analysis, tracks peaks in it and creates notch filters to remove them. It is especially useful with various audio materials, such as drums, where resonances of the instruments are too prominent. You simply analyze the part of the audio material, where the resonances are mostly audible, and click this button. Then you can just disable the bands that remove the frequencies that you actually want to keep. It may also be worth trying to play with the Q values of each band to make sure the resonances are completely gone.



### Swap source and target

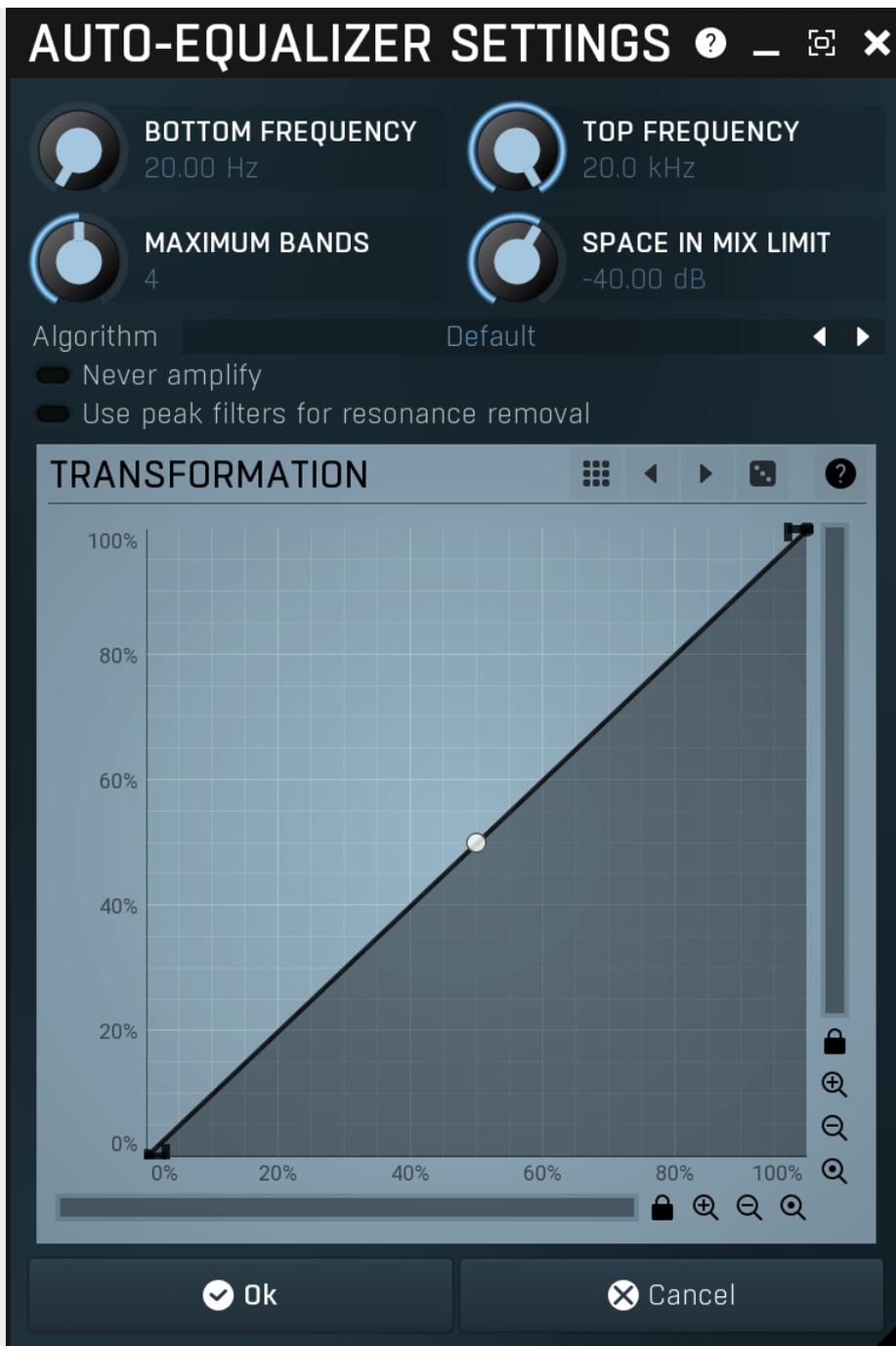
Swap source and target button swaps the analyses of source and target and can be helpful when you want to try equalizing 'the other way around'.



### Settings

Settings button shows additional settings of the automatic equalizer algorithm.

## Auto-equalizer settings

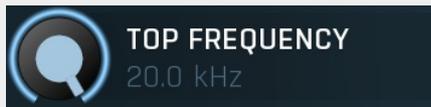


Auto-equalizer settings provides additional settings for the automatic equalization algorithm.



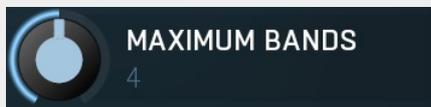
### Bottom frequency

Bottom frequency defines lowest frequency taken into account when performing automatic equalization. Use it to avoid unnecessary processing of the bottom-end.



### Top frequency

Top frequency defines highest frequency taken into account when performing automatic equalization. Use it to avoid unnecessary processing of the high-end.



### Maximum bands

Maximum bands defines maximal number of bands that will be configured during the automatic equalization. Generally more bands provide more accurate match to the source analysis, however this may not always be desired. You can also make use of the **Smoothness** parameter, above the graph area. Increasing the smoothness before pressing **Equalize** will create a less-pronounced equalization curve.



## SPACE IN MIX LIMIT

-40.00 dB

### Space in mix limit

Space in mix limit controls the minimal level below which a particular frequency is considered silent. It is used for the **Separate** feature to determine which frequencies in the source (full mix for example) are candidates for separation and which are not.

For example, if this value is -20dB and the level of the frequency 1000Hz in the source analysis is -20dB, then the separation engine will consider 1000Hz to be "available" in the target audio and will not try to remove this frequency from the target. However if the level in the source were -10dB, then the engine would consider this frequency 50% occupied and may try to use a filter to remove some of the 1000Hz from the target if it contains this frequency as well.

In other words, frequencies in the source with levels louder than the mix limit will be those that would be reduced in the target.

### Algorithm

Default

### Algorithm

Algorithm selector lets you choose which method is used to convert the frequency response into a configuration of the equalizer bands. (This algorithm itself is actually very complex and requires lots of CPU when adjusting the bands during the equalization/separation processes.) Multiple algorithms are available, each of them with a slightly different output.

**Default** provides the most accurate results, but it takes lots of CPU. Note, as mentioned above, that this CPU usage is required only for the pre-computing, the actual audio processing requirements are the same for all algorithms.

**Optimized** algorithm is much faster, however it can produce slightly less accurate results. It may be more suitable because it smoothes out the peaks in the frequency response.

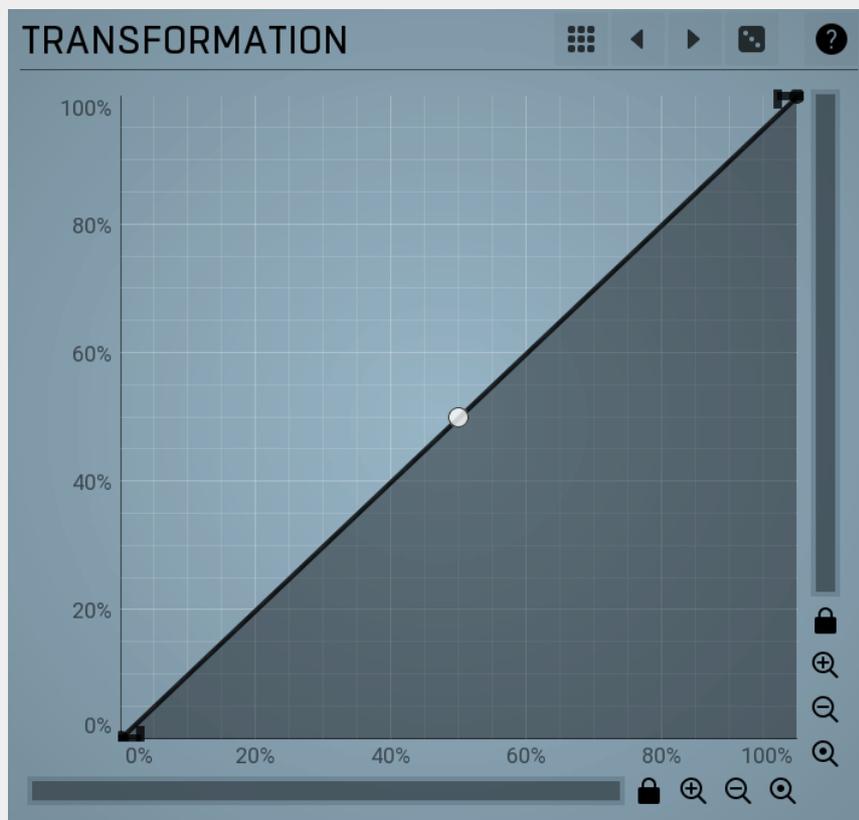
**Super-fast** mode is the fastest of all of them, but it trades speed for accuracy in its results. You may find it useful in specific cases.

### Never amplify **Never amplify**

Never amplify ensures that the automatic equalization only attenuates frequencies so that no amplification occurs. This may be desired, since amplification often increases noise level. On the other hand, due to its nature, it will probably lower the output level. It may be worth considering using AGC to set the output gain afterwards.

### Use peak filters for resonance removal **Use peak filters for resonance removal**

Use peak filters for resonance removal option makes the **Remove resonances** feature use peak filters instead of the default notch filters, which are more effective, but **Dry/Wet** doesn't affect them.



### Transformation

Transformation changes the requested frequency response. The engine first determines the optimum frequency response (the algorithms for equalization and separation are different). Then it computes the optimal response of the equalizer. Before it approximates this response using the equalizer filters, that response can be transformed. The transformation graph shows the original requested response on the X-axis and the new one on the Y-axis. For example you can exaggerate or sharpen the results just using the single point curvature.



### 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.



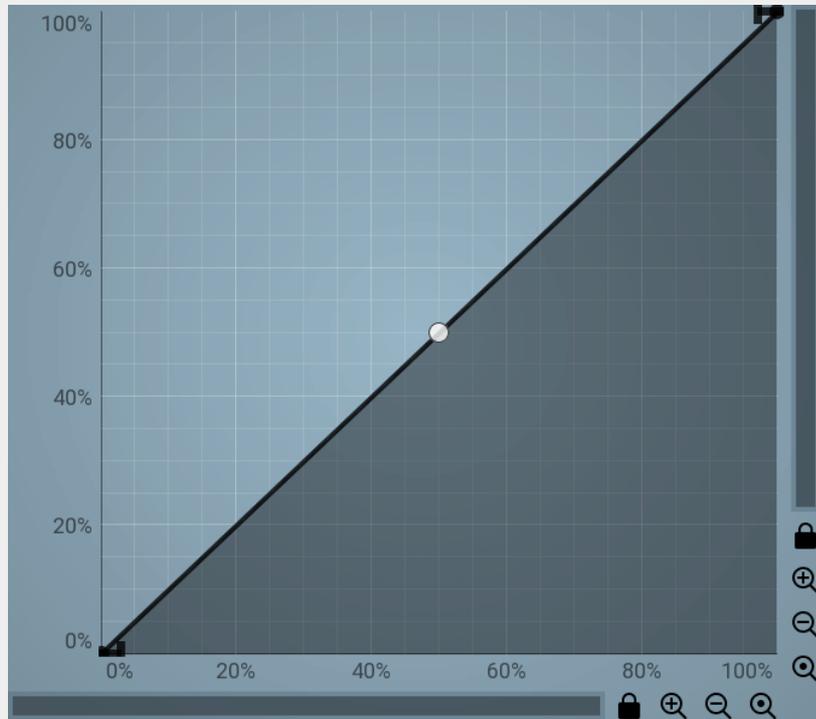
### Right arrow

Right arrow button loads the next preset.



### Randomize

Randomize button loads a random preset.



EnvelopeEditorGraph

## 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.



Advanced settings window contains more advanced settings, which are used less often and so are intentionally not shown on the main plugin editor.



**selector**

Tab selector switches between subsections.

 **Randomize**

Randomize button generates random settings for the tab.

 **Presets**

Presets button chooses a random preset for the tab.

## Voicing panel

**VOICING** Mono toggle Allow multiple monophonic voices ?

Voice restart mode	Smooth	◀ ▶	Voice to kill	Last	◀ ▶
Max voices	128	◀ ▶	Max voices per note	1	◀ ▶
Stop voices when silent	Off		Lowest note	0 (C-1)	
Filter follow mono	10 ms		Highest note	127 (G9)	
Global ADSR use	Volume	◀ ▶	Sustain minimum	50.0%	

Enable sustain pedal   
 Enable sostenuto pedal   
 Enable volume MIDI controller  
 Enable panorama MIDI controller   
 Follow pitchbend after note-off   
 Equal random every time

Voicing panel contains some global settings controlling how new voices are created.

### Mono toggle **Mono toggle**

Mono toggle switch is useful in monophonic modes and if enabled, releasing a new note while holding the previous one makes the pitch of the voice changed back to the note you are holding.

### Allow multiple monophonic voices **Allow multiple monophonic voices**

Allow multiple monophonic voices switch is useful in monophonic modes and if enabled, multiple voices can actually occur at the same time. This has an effect for sounds with long release times. Holding a note and pressing another would change its pitch and do whatever you are used to in normal monophonic modes, but once you release all keys, the release stage starts. Then when you press another key while the previous voice is still in release stage, normally this voice is switched yet again to the new pitch. But if this switch is enabled, pressing a new key (while no other is held) creates a new voice leaving the previous one to decay until finished.

### Voice restart mode **Voice restart mode**

Voice restart mode is useful in monophonic modes and how the new notes are replacing the old ones.

**None** means that the pitch is change, but nothing else happens.

**Smooth** makes all envelopes restarted into the attack stage smoothly connecting them, hence avoiding abrupt changes resulting in clicks.

**Smooth even off** is similar, but the envelopes are restarted into the attack stage even when jumping back from notes being released as a result of **Mono toggle**

**Brutal** makes all envelopes restarted the initial position, hence causing abrupt changes, which may however be desired for certain sounds.

**Restart** mode creates new voices instead of changing the existing ones. The current voices are stopped in the same way as when you release them and new voices are created for the notes being pressed. This way the envelopes are followed and you can get interesting overlaps of the release stages.

**Restart (brutal)** is similar to Restart, but existing voice(s) are immediately brutally stopped.

### Voice to kill **Voice to kill**

Voice to kill defines which voice is removed, when the number of voices exceeds the limit controlled by **Max voices** parameter.

### Max voices **Max voices**

Max voices controls the maximum number of voices that can play at any moment. You can use this to avoid overloading your CPU when too many voices are played. By default the plugin provides unlimited number of voices.

### Max voices per note **Max voices per note**

Max voices per note controls the maximum number of voices on the same note and channel, that can play at any moment in **Polyphonic (single)** voicing mode. You can use this to avoid overloading your CPU when too many voices are played, but still allowing multiple of them, which may be necessary for realistic acoustic instruments such as piano for example.

Range: 1 to 20, default 1

### Stop voices when silent **Stop voices when silent**

Stop voices when silent provides an alternative to the default stopping the voice based on the global ADSR. Normally each voice is stopped when the global ADSR ends, whether it is used to actually control the volume or not. When you enable this however, the voices will be terminated after a specified period of silence. So every time the voice is rendered, the output level is measured and when it is silent (below -120dB), a time measurement is performed and once the time reaches your requested value, the voice will be terminated.

Range: Off to 10000 ms, default Off

### Lowest note **Lowest note**

Lowest note controls the lowest note that will actually be played. This can be used to make the instrument ignore some notes, so that you can use them to control some features for example.

Range: 0 to 127, default 0

### Filter follow mono **Filter follow mono**

Filter follow mono controls the time it takes for the filter to follow note changes in monophonic modes. Use 0ms to make the filter follow instantly, which can cause short glitches. Higher times may provide additional expression, because the filter frequency will highly depend on the first note played.

Range: 0 ms to 10000 ms, default 10.0 ms

### Highest note **Highest note**

Highest note controls the highest note that will actually be played. This can be used to make the instrument ignore some notes, so that you can use them to control some features for example.

Range: 0 to 127, default 127

Global ADSR use

Volume



Global ADSR use

Global ADSR use controls how the global ADSR is applied. Note that it is always used to determine the end of each voice, no matter what this mode is set to.

**Off** makes the engine use it only to detect the end of each note. You can use it manually for modulation however.

**Volume** is the default mode, which makes the engine apply it as volume level on each output of the generator.

**Volume squared** is similar, but makes the volume curve more drastic.

Sustain minimum

50.0%

Sustain minimum

Sustain minimum controls the minimum value of the sustain pedal MIDI controller to be considered 'open'. Above this value all notes being pressed are held until the sustain pedal is released. In some cases, such as simulating an actual piano, you may want to lower this value to simulate partial sustain.

Range: 0.00% to 100.0%, default 50.0%

Enable sustain pedal

Enable sustain pedal

Enable sustain pedal makes the instrument listen to the sustain pedal when processing notes. If this is disabled, the instrument ignores the sustain pedal when processing notes.

Enable sostenuto pedal

Enable sostenuto pedal

Enable sostenuto pedal makes the instrument listen to the sostenuto pedal when processing notes. If this is disabled, the instrument ignores the sostenuto pedal when processing notes.

Enable volume MIDI controller

Enable volume MIDI controller

Enable volume MIDI controller makes the plugin automatically follow the volume MIDI messages. It is disabled by default, since it should be handled manually using global MIDI settings, which let you attach it to the easy screen multiparameters.

Enable panorama MIDI controller

Enable panorama MIDI controller

Enable panorama MIDI controller makes the plugin automatically follow the panorama MIDI messages. You may want to disable it if your MIDI controller malfunctions for example.

Follow pitchbend after note-off

Follow pitchbend after note-off

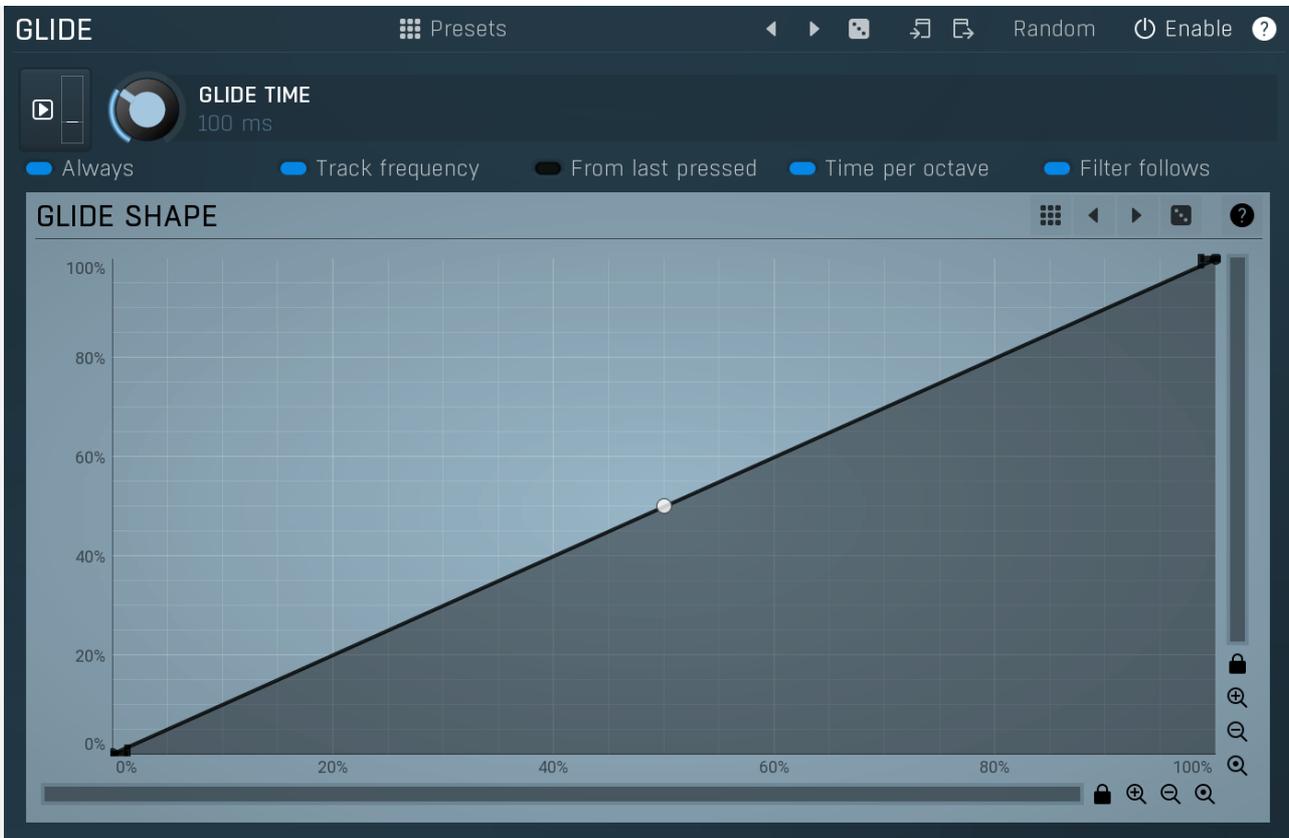
Follow pitchbend after note-off makes the plugin follow any changes to the pitchbend even after the key has been released and the voice is decaying in the release stage. Normally this is disabled, which is necessary for MPE to work properly.

Equal random every time

Equal random every time

Equal random every time controls the way the many random things used in the plugin are initialized when rendering. By keeping this enabled you will get identical (in most cases) performance every time. By disabling this, the plugin will work the same way as most other plugins and hardware, which will sound slightly differently every time.

## Glide panel



Glide panel contains parameters of the glide feature, which lets new notes glide from previous ones.

 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.

#### Always **Always**

Always starts the glide every time you press a note. If this is disabled, the glide is performed only if you currently hold another note, hence the glide is started only if the 2 notes are overlapping.

*For example, if you hold key C and press key G, a glide is always performed. But if you press C, release it and then press G, the glide is started only if this parameter is enabled.*

#### Track frequency **Track frequency**

Track frequency overrides the behaviour when the notes are followed by the glide and instead it keeps track of the current frequency, so it 'starts where the last one ended'. This is useful for long glides. It even works with polyphonic glides, where at any moment the sound can split from the same pitch etc.

*For example, if the glide length is 10 seconds and you press C0 and C4 afterwards for a short time, the glide doesn't have enough time to get to C4. Normally next time the glide would start directly at C4 even though it didn't manage to get to that pitch. With this option enabled, the glide will continue where it finished (a frequency somewhere between C0 and C4).*

## From last pressed **From last pressed**

From last pressed makes the glide start from the last key pressed, which may actually not be the one you are holding right now. For example, if you hold key C, then press and release key E (so there's a short glide from C towards E) and then press key G, there are 2 options. First, the glide may go from C to G as C is currently pressed, and that's what happens if 'from last pressed' is disabled. Second, the glide may go from E to G, because you most recently pressed E, and that's what happens with this option enabled.

## Time per octave **Time per octave**

Time per octave makes the glide time depend on the pitch distance (interval) between the 2 notes. When enabled, the glide time specifies the time needed to glide across one octave. If not enabled, the glide time is the total time to get from the initial note to the target. In that case the actual glide speed differs depending on the 2 notes, because it takes the same amount of time to glide between any 2 notes, no matter if they are 1 semitone or 4 octaves away from each other.

## Filter follows **FromFilter follows**

FromFilter follows makes the filters follow the glide pitch. If this is disabled, the filters use the frequencies of the target notes and ignore the glide, which can be useful for example when you want the note to glide towards the target note, which will resonate using the filter.

## **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**

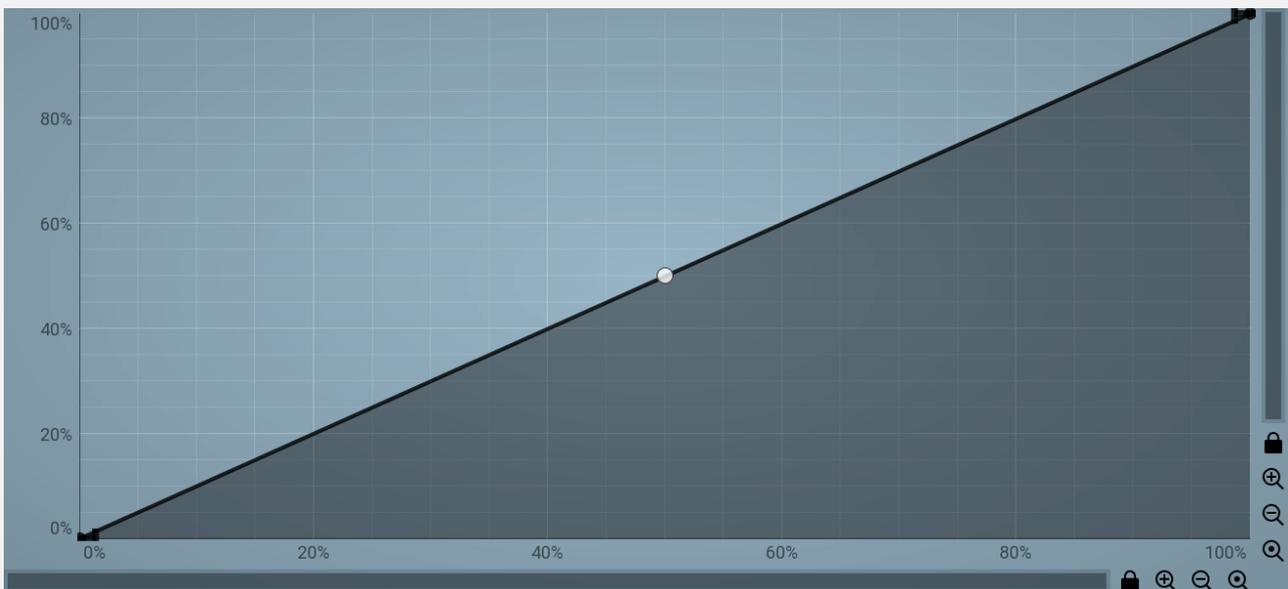
Left arrow button loads the previous preset.

## **Right arrow**

Right arrow button loads the next preset.

## **Randomize**

Randomize button loads a random preset.



## **EnvelopeEditorGraph**

# 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.

## Volume control panel

### VOLUME CONTROL



RANGE  
-30.00 dB

- Velocity interpolation
- Follow velocity off

- Follow velocity
- Follow aftertouch

Volume control panel contains controls defining how voice volume is calculated.



RANGE  
-30.00 dB

#### Velocity range

Velocity range controls the velocity to volume transformation. When set to minimum it becomes "Off" and the volume is calculated not logarithmically, but linearly. Any other value defines the gain of minimum velocity. For example, if you set this to -20dB, than notes with highest velocities will be about 20dB louder than notes with lowest velocities.

Range: Off to 0.00 dB, default -30.00 dB



Velocity interpolation

#### Velocity interpolation

Velocity interpolation activates velocity interpolation for monophonic modes. When this is disabled, only the first note's velocity is relevant and the remaining notes will have the same velocity. If this is enabled however, the velocities of all notes are relevant and a quick transition between them will occur.



Follow velocity

#### Follow velocity

Follow velocity switch makes the plugin initialize the volume of each voice according to the velocity. This is desired on most controllers, but may not be working well with certain types of expression on 3D MPE controllers. It is then useful to disable this option and enable Follow aftertouch instead, which will make the instrument initialize a very small velocity and wait for aftertouch messages.



Follow velocity off

#### Follow velocity off

Follow velocity off switch makes the plugin change the volume of voices when a note off with a velocity specified. Some MIDI controllers (especially MPE controllers) send this value as a signal of 'how quickly the user released the key'. In many cases this is not desired though, so it is disabled by default.



Follow aftertouch

#### Follow aftertouch

Follow aftertouch switch makes the plugin listen to aftertouch MIDI messages and adjust the volume/velocity of each voice accordingly. Usually this is disabled and the aftertouch is only used to modulated some parameters providing the additional feel when playing the instrument.



### 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.



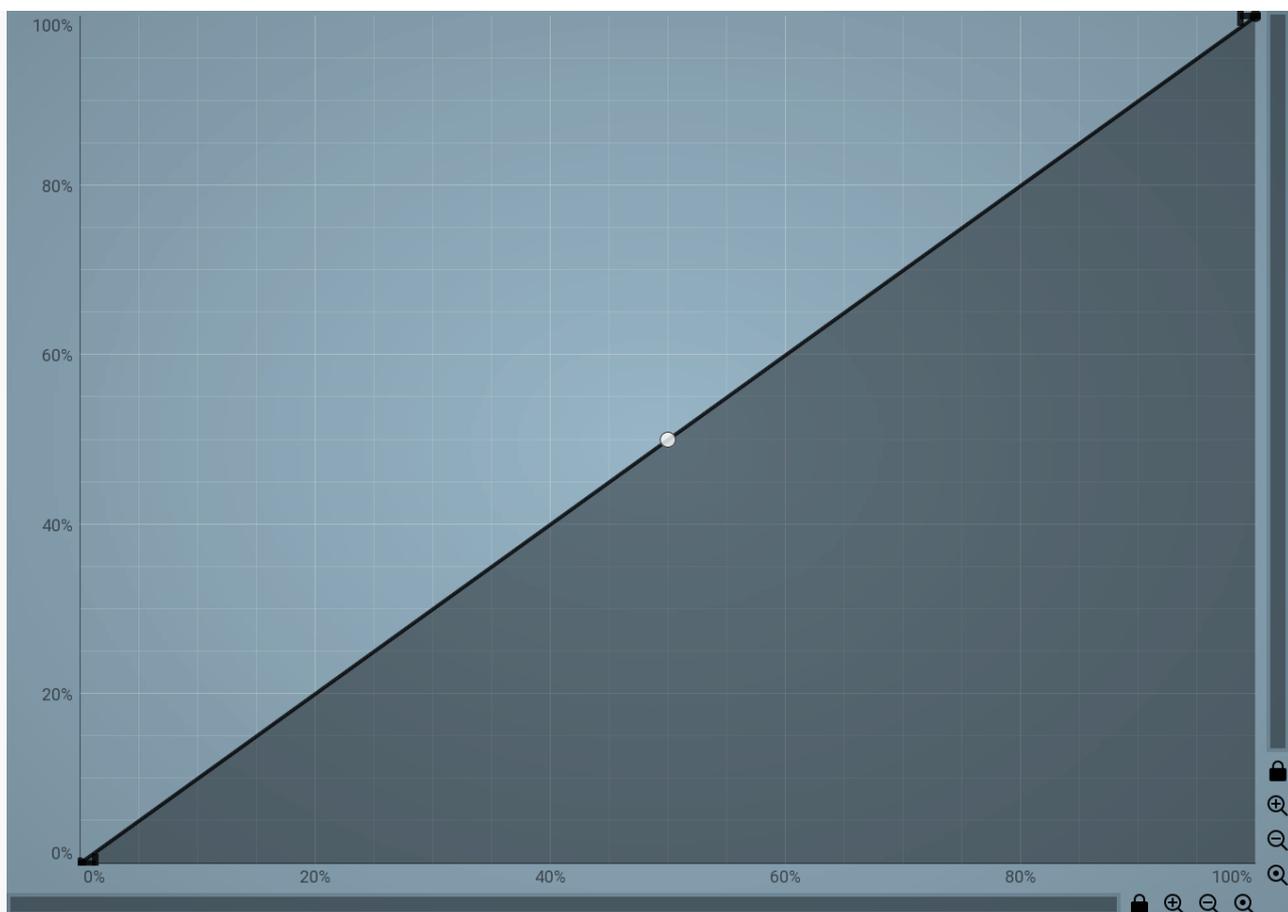
### Right arrow

Right arrow button loads the next preset.



### Randomize

Randomize button loads a random preset.



## EnvelopeEditorGraph

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

#### Modulation wheel is timbre

#### Modulation wheel is timbre

Modulation wheel is timbre switch makes the modulation wheel change the timbre controller as well. This is often useful for various 3D controllers, which should send timbre according to the MPE standard, but send modulation wheel instead, probably because it is more supported.

#### Timbre is bipolar

#### Timbre is bipolar

Timbre is bipolar switch makes the timbre controller bipolar meaning that per-voice modulation will produce values from -100% to +100% as opposed to other controllers, which produce 0% to 100% (hence always in the modulation direction). This is usually advantageous, since the controller itself is bipolar (meaning it's default value is the center), so it simplifies editing sounds featuring this controller.

#### Timbre defaults to minimum

#### Timbre defaults to minimum

Timbre defaults to minimum switch makes the timbre controller default value minimum as opposed to typical middle value. This gets often useful if you want the timbre to work with various controllers, some of which have the timbre, others don't, and control 'amount' of something. With the default behaviour having the timbre 50% would make the modulation running even if there is no timbre controller. Enabling this option would initially make the timbre disable the controller.

## Pitch-bend is bipolar

### Pitch-bend is bipolar

Pitch-bend is bipolar switch makes the pitch-bend controller bipolar meaning that per-voice modulation will produce values from -100% to +100% as opposed to other controllers, which produce 0% to 100% (hence always in the modulation direction). This is usually advantageous, since the controller itself is bipolar (meaning it's default value is the center), so it simplifies editing sounds featuring this controller.

## Linear smoothing

### Linear smoothing

Linear smoothing activates the linear smoothing, which provides more accurate controller values, however since it produces equal smoothing speed for any change of the controllers, it requires higher smoothing times than the default exponential smoothing.

## Timbre uses initial-64 mode

### Timbre uses initial-64 mode

Timbre uses initial-64 mode switches on the MPE initial-64 mode for the timbre controller. Enabling this option means the timbre control is always set to value 64 at the note-on, and all subsequent changes of the timbre control relative to its actual initial value are added to that initial value of 64 to produce the actual timbre control value used by the plugin.

## Controller smoothing panel



Controller	Smoothing Time
Pressure	10 ms
Pitch-bend	10 ms
Modulation wheel	10 ms
Expression CC#11	10 ms
Frequency	10 ms
Main 1	10 ms
Main 2	10 ms
Main 3	10 ms
Main 4	10 ms
Main 5	10 ms
Main 6	10 ms
Main 7	10 ms
Main 8	10 ms
Main 9	10 ms
Main 10	10 ms
Main 11	10 ms
Main 12	10 ms
Main 13	10 ms
Main 14	10 ms
Main 15	10 ms
Main 16	10 ms
Timbre CC#74	10 ms
Pitch-bend for pitch	10 ms
Breath CC#2	10 ms
Sustain	10 ms
Velocity off	10 ms

Controller smoothing panel controls smoothing for various MIDI controllers. It avoids abrupt changes caused by low resolution of the MIDI standard and the MIDI controllers themselves. Smoothing is always represented as minimum time it gets for the controller to go from minimum to maximum value.

## 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.

**Pressure** | **Timbre CC#74** | **Pitch-bend** | **Modulation wheel** | **Breath CC#02** | **Expression CC#11** | **Sustain** | **Note-off** | **Tab**

### selector

Tab selector switches between subsections.

## Presets

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### Left arrow

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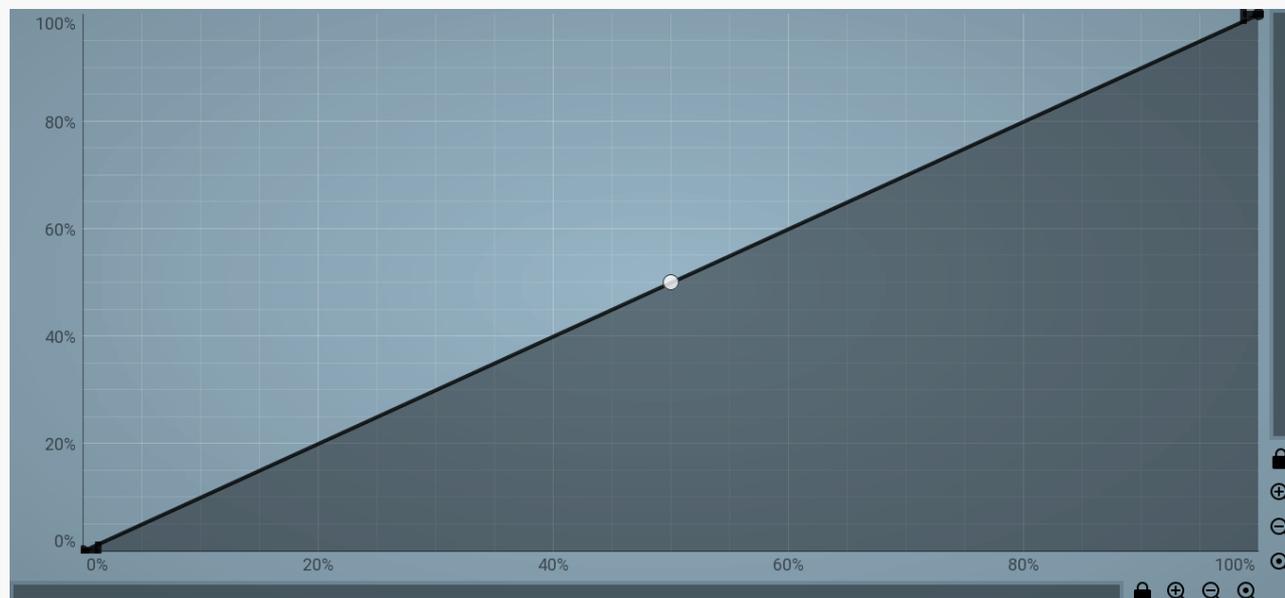
### Right arrow

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### Randomize

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## EnvelopeEditorGraph

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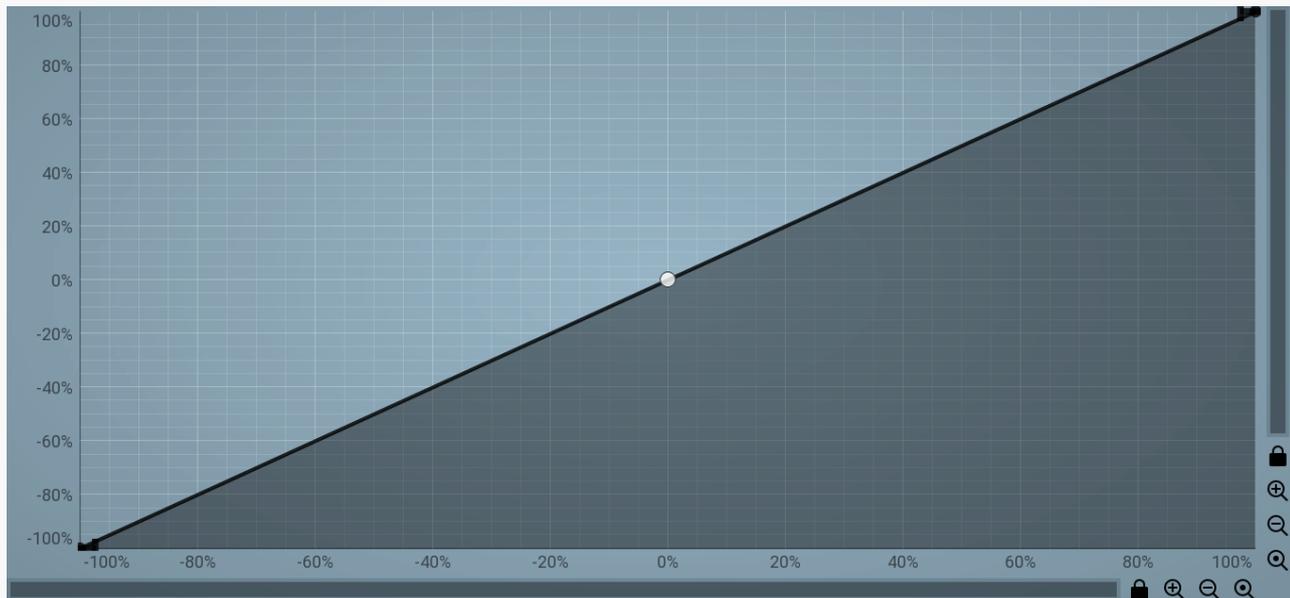
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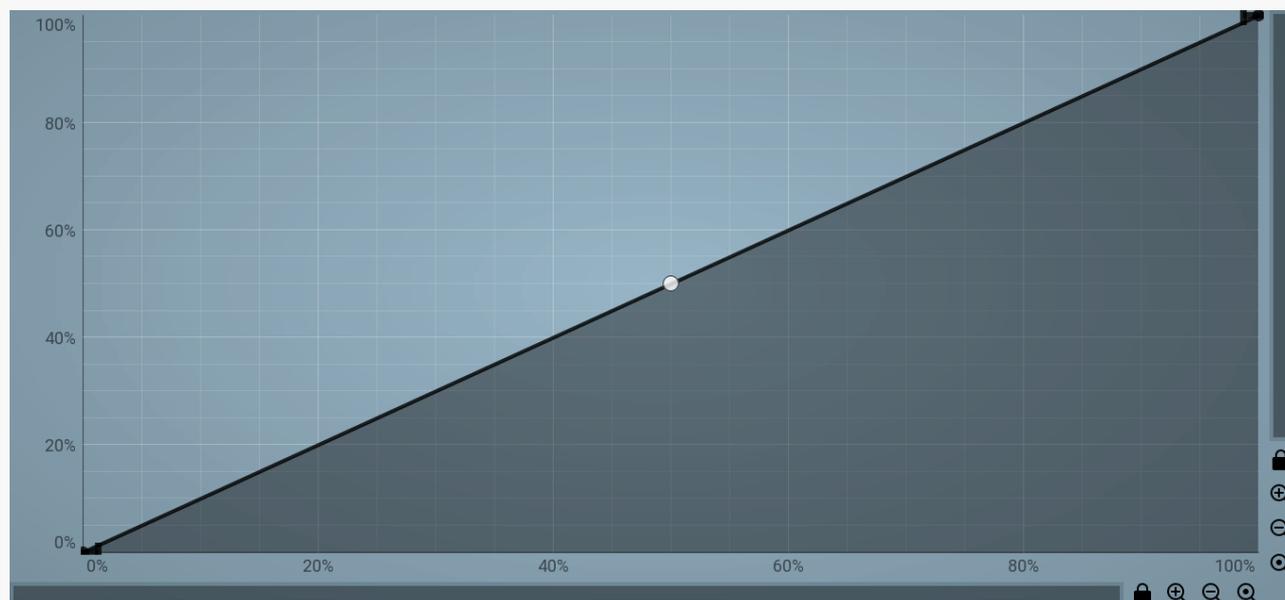
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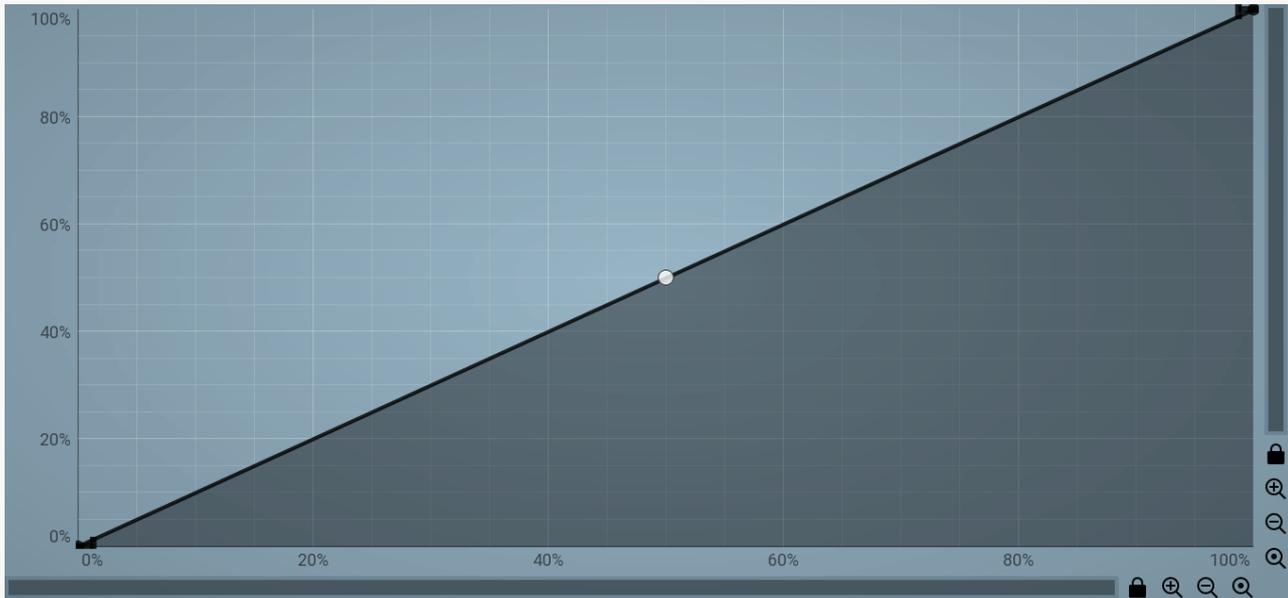
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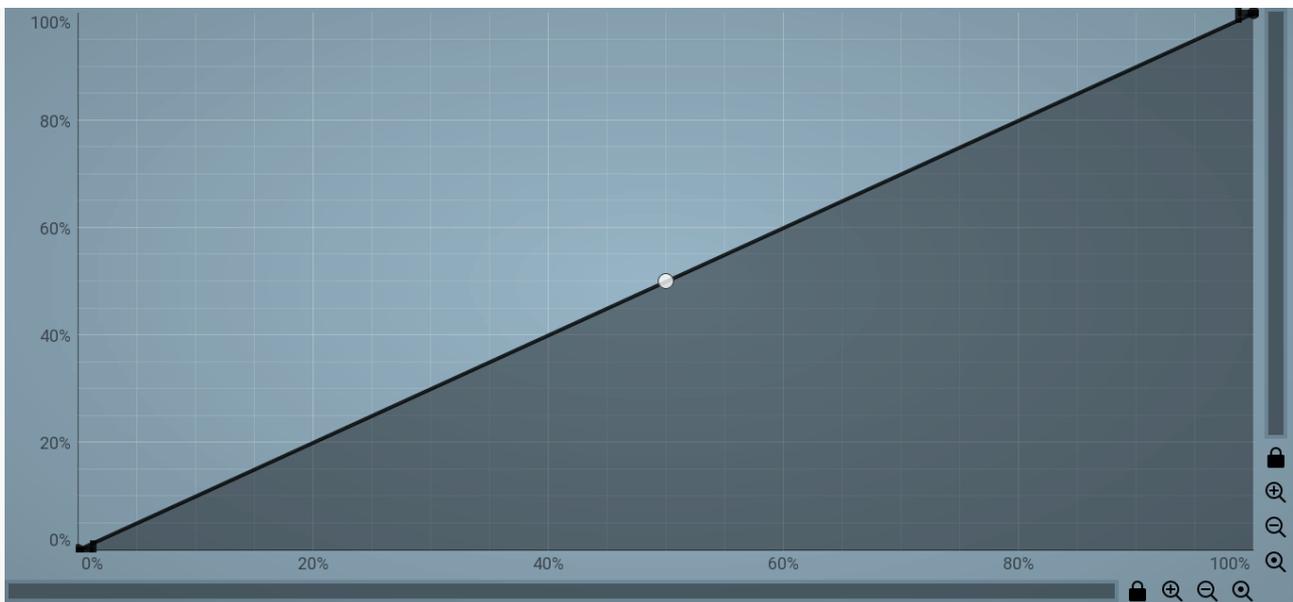
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### Harmony panel



Harmony panel controls additional voices created virtually by the instrument. When you press a note, the plugin may behave as if you had pressed additional notes with different velocities, tuning, delays 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.



### Random

Random button generates random settings using the existing presets.



### Learn

Learn button lets you learn the whole harmony by playing it. Enable the button, then press a chord on your keyboard and the plugin will configure the harmony settings so that the plugin will play the chord when one key is pressed.

## Note panel



Note panel contains parameters of a virtual note. If you enable the note and press a key, the instrument will behave as if you had pressed another key at the same time, with different velocity, tuning, delay etc. Note that the keys cannot be duplicates, so if you for example hold C1 and there is a virtual note 12 semitones up (C2) both are played, but when you then press C2 as well, this one won't generate a new sound and will be ignored as it is already playing, but C3 will be added as well.



### Offset

Offset controls the offset of the virtual note from the original one in semitones.

Range: -48 to +48, default +5



### Detune

Detune controls the detuning of the virtual note in cents. The actual pitch difference is the sum of these two control values.

Range: -100.00 to +100.00, default 0



### Velocity

Velocity controls the difference in velocity of the virtual note from the original one.

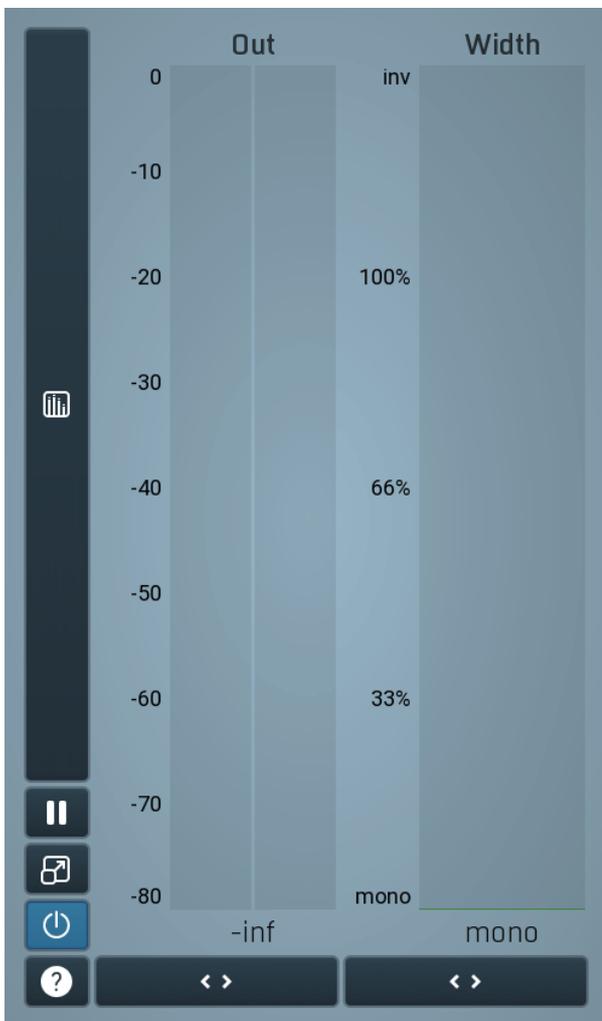
Range: -100.0% to 100.0%, default 0.00%



### Delay

Delay makes the virtual note delayed from the original one and can be used for strumming effects for example.

Range: 0 ms to 1000 ms, default 0 ms



### 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.

**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

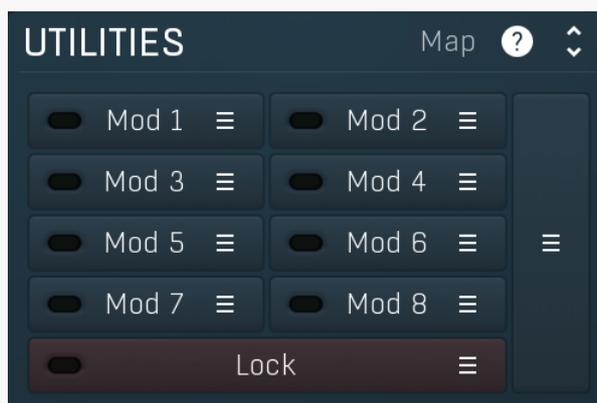
Collapse button minimizes or enlarges the panel to release space for other editors.



### Collapse

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

## Utilities



## Map

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

## Mod 1 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.

## 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.

## 1 : Volume -12.0 dB 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.

