MWobbler

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.

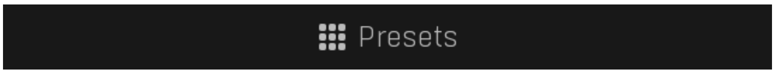
To the right of the controls are the meters or time-graphs for the plugin; the standard plugin Toolbar may be to the right of these or at the bottom of the plugin.

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



Presets

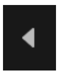
Presets button shows a window with all available presets. A preset can be loaded from the preset window by double-clicking on it, selecting via the buttons or by using your keyboard. You can also manage the directory structure, store new presets, replace existing ones etc. Presets are global, so a preset 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

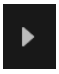
Files are named based on the name of the plugin like this: "{pluginname}.presets", so for example MAutopan.presets or MDynamics.presets. If the directory cannot be found on your computer for some reason, you can just search for the particular file.

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.*



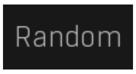
Left arrow

Left arrow button loads the previous preset.



Right arrow
Right arrow button loads the next preset.

Randomize
Randomize button loads a random preset.



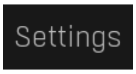
Randomize
Randomize button (with the text 'Random') generates random settings. Generally, randomization in plug-ins works by selecting random values for all parameters, but rarely achieves satisfactory results, as the more parameters that change the more likely one will cause an unwanted effect. Our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

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

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

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

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



Settings
Settings 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).

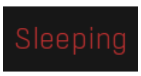
Advanced settings configures several processing options for the plug-in.

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.

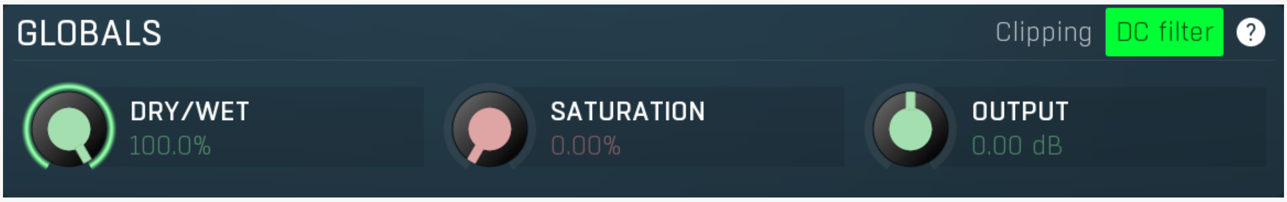
Smart interpolation adjusts the interpolation algorithm used when changing parameter values; the higher the setting the higher the audio quality and the lower the chance of zippering noise, but more CPU will be used.

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.

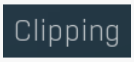


Sleep indicator
Sleep indicator informs whether the plugin is currently active or in sleep mode. The plugin can automatically switch itself off to save CPU, when there is no input signal and the plugin knows it cannot produce any signal on its own and it generally makes sense. You can disable this in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

Globals



Globals contains the main plugin parameters.



Clipping

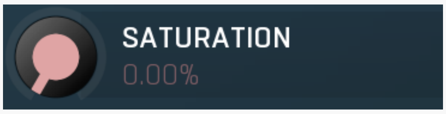
Clipping enables an optional clipper being the very last item in the chain. It can be used to remove potential peaks that the filter may cause especially with wild settings. As a hard clipper it can also be used as another distortion stage, however be cautious with it.

DC filter for filters

DC filter for filters activates the DC filter for output of each filter. It will remove content below 20Hz, which isn't audible but is interfering with many processing algorithms. Such content is often created by the distortion algorithms used by the filters.

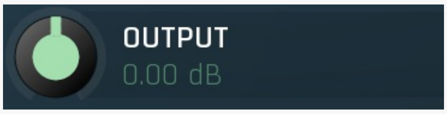
Dry/Wet

Dry/Wet defines ratio between dry and wet signals. 100% means fully processed, 0% means no processing at all. Please note that values in between may causes some phase shifting effects.
Range: 0.00% to 100.0%, default 100.0%



Saturation

Saturation controls the output saturation which is performed before the output gain. This saturation algorithm is different from the algorithm used in the filter output section. This way you can combine both filter saturation and output saturation to get an even dirtier sound.
Range: 0.00% to 100.0%, default 0.00%

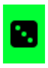


Output gain

Output gain defines the output gain being applied after saturation and before the output clipper. This parameter should be used to adjust the output level; however since the output clipper is located after this stage, it can also be used to control the level of output clipping (if required at all).
Range: -24.00 dB to +24.00 dB, default 0.00 dB

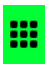
Tab selector

Tab selector switches between subsections.



Randomize

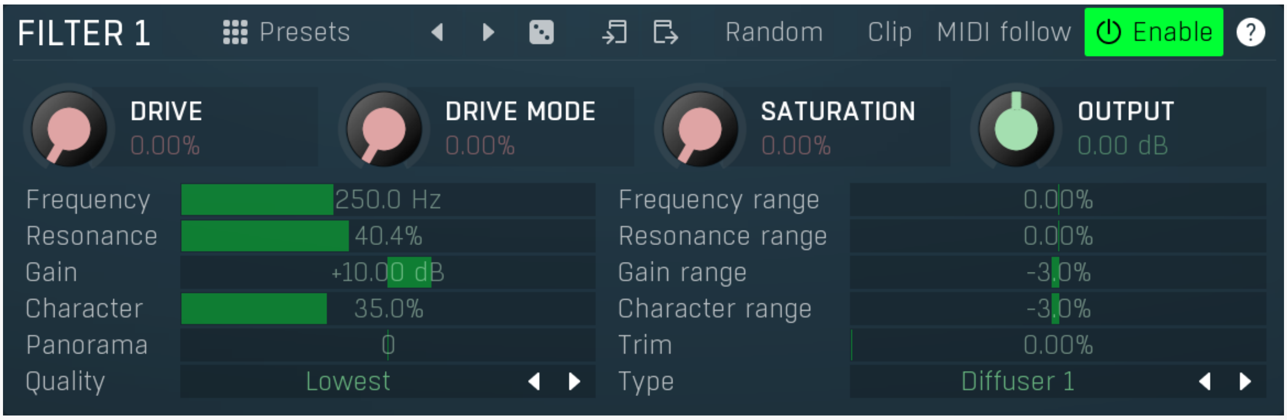
Randomize button generates random settings for the tab.



Presets

Presets button chooses a random preset for the tab.

Filter panel



Filter panel contains settings for the particular 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.

Copy

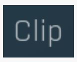
Copy button copies the settings onto the system clipboard.

Paste

Paste button loads the settings from the system clipboard.

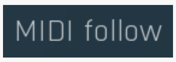
Random

Random button generates random settings using the existing presets.



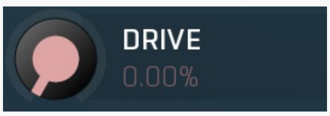
Clip

Clip enables an optional clipper placed after the filter's output gain. It can be used to remove potential peaks 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.



MIDI follow

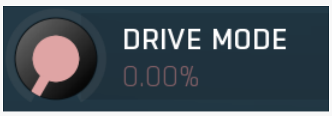
MIDI follow button shows MIDI follow settings, which you can use to make the filter listen to input MIDI notes and adjust its frequency accordingly.



Drive

Drive controls input distortion of the filter. This creates higher harmonics, which are then processed 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 already rich signal, the results may simply be too dirty. However when applied to a rich yet harmonic signal (such as a sawtooth wave), only existing harmonics will be added, so the effect won't be in creating additional harmonics and 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.

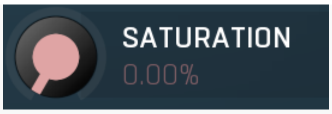
Range: 0.00% to 100.0%, default 0.00%



Drive mode

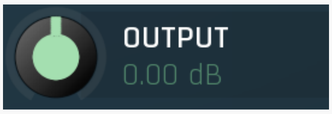
Drive mode controls input distortion character. Essentially this controls the levels of different harmonics.

Range: 0.00% to 100.0%, default 0.00%



Saturation

Saturation controls the output saturation performed before the output gain. This provides another enrichment performed after the filtering.*For example, you may have a simple sine wave on the input, processed through the input distorting, 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.*
Range: 0.00% to 100.0%, default 0.00%



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

Frequency

Frequency controls the filter central frequency.
Range: 20.00 Hz to 20.0 kHz, default 800.0 Hz

Frequency range

Frequency range controls to which extent the frequency is modulated. With 0% no modulation occurs and the filter frequency is defined by **Frequency** parameter only. If you increase the range however, the filter frequency will move away from the central frequency according to the control signal (LFO/follower). Values above 0% change the filter frequency to be ABOVE its center when the control signal is above 0 and vice versa. Values below 0% change the filter frequency to be BELOW its center when the control signal is above 0 and vice versa. This makes it easy to modulate each parameter differently.
Range: -100.0% to 100.0%, default 0.00%

Resonance

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

Resonance range

Resonance range controls the extent to which the resonance is modulated. With 0% no modulation occurs and the filter resonance is defined by **Resonance** parameter only. If you increase the range however, the filter resonance will move away from the central resonance according to the control signal (LFO/follower). Values above 0% change the filter resonance to be ABOVE its center when the control signal is above 0 and vice versa. Values below 0% change the filter resonance to be BELOW its center when the control signal is above 0 and vice versa. This makes it easy to modulate each parameter differently.
Range: -100.0% to 100.0%, default 0.00%

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

Gain range

Gain range controls to which extent the gain is modulated. With 0% no modulation occurs and the filter gain is defined by **Gain** parameter only. If you increase the range however, the filter gain will move away from the central gain according to the control signal (LFO/follower). Values above 0% change the filter gain to be ABOVE its center when the control signal is above 0 and vice versa. Values below 0% change the filter gain to be BELOW its center when the control signal is above 0 and vice versa. This makes it easy to modulate each parameter differently.
Range: -100.0% to 100.0%, default 0.00%

Character

Character controls the central character of the filter. Please note that it is used only for some filters. 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.
Range: 0.00% to 100.0%, default 50.0%

Character range

Character range controls to which extent the character is modulated. With 0% no modulation occurs and the filter character is defined by **Character** parameter only. If you increase the range however, the filter character will move away from the central character according to the control signal (LFO/follower). Values above 0% change the filter character to be ABOVE its center when the control signal is above 0 and vice versa. Values below 0% change the filter character to be BELOW its center when the control signal is above 0 and vice versa. This makes it easy to modulate each parameter differently.
Range: -100.0% to 100.0%, default 0.00%

Panorama

Panorama lets you shift the filter frequency between channels. Left channel's frequency is shifted down by specified amount of semitones, right channel is shifted up, third down etc.
Range: -48.00 to +48.00, default 0

Trim

Trim defines minimum of the control signal (LFO/follower), which will set the minimum filter values. This is a very specific feature, which essentially modifies the processing and finds its use in several applications, such as wobbling basses. *For example, set a full saw shape in the LFO and let it modulate the filter frequency of a LP filter. This makes the filter change create the obvious saw low-pass filtering. However it turns out that keeping the minimum filter frequency longer isn't a bad idea at all. To do this just increase the trim parameter. You can picture that it cuts off the bottom of each saw.*
Range: 0.00% to 100.0%, default 0.00%

Quality

Quality controls the ratio between audio quality and CPU requirements.

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.

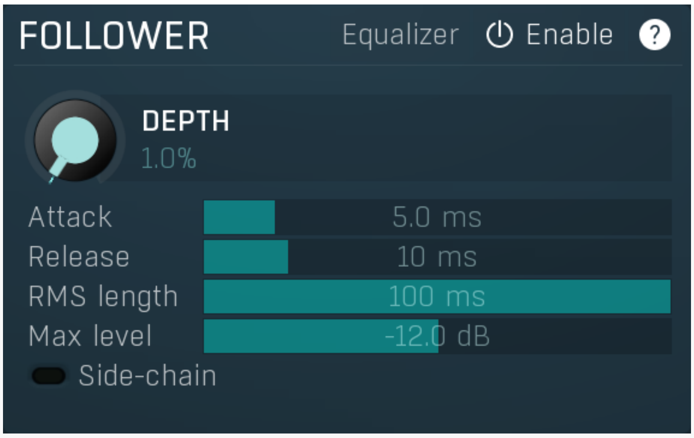
Sub-X, over-X and band-X filters are specialized complex combinations of other filters originally designed for wobbling basses. These mainly combine LP/HP/BP filters with harmonic filters. The **character** parameter controls the distribution of harmonics and should be used simply by trial-and-error.

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).

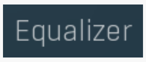
Comb and diffuser filters are complex comb filtering processors with pretty wild and fat responses. These range from simple comb filtering to complex almost ambient responses. Each filter uses a different kernel, so it shall be selected by trial-and-error approach. The**character** parameter controls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

Follower panel

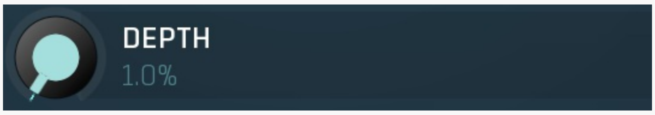


Follower panel contains parameters of the input level follower.



Eq

Eq button shows the settings of the side-chain equalizer. This equalizer does not affect the outgoing signal, but processes the signal entering the level detector. You can use it to target those frequencies to which you want the processor to react.
In most cases you will be using low/high/band-pass filters to remove those parts of the spectrum that you are not interested in utilizing. For example, to make the detector react to a bass drum, you may use a low-pass filter with a frequency of say 100 Hz.
Additionally, the equalizer lets you perform more complicated processing. For example, you may want the detector to react to the whole spectrum, but especially the high end of the spectrum, in which case a high-shelf filter may be the appropriate one to choose.



Depth

Depth defines how much the level follower controls the filters. For 0% the filters are fully LFO based. For 100% the LFO is disabled and only the follower is relevant.
Range: 0.00% to 100.0%, default 0.00%

Attack

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

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

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

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

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

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

In a modulator, the detector is driving other parameters, a filter cut-off frequency for example, and the situation really depends on the target. If you want the detector to react quickly on the input level rising, use a shorter attack time; if you want it to follow the flow of the input signal slowly, use longer attack and release times.
Range: 0 ms to 10000 ms, default 10.0 ms

Release

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

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

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

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

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

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

In a modulator, the detector is driving other parameters, a filter cut-off frequency for example, and the situation really depends on the target. If you want the detector to react quickly on the input level falling, use a shorter release time; if you want it to follow the flow of the input signal slowly, use longer attack and release times.
Range: 0 ms to 10000 ms, default 10.0 ms

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.
Range: 0 ms to 100 ms, default 1.0 ms

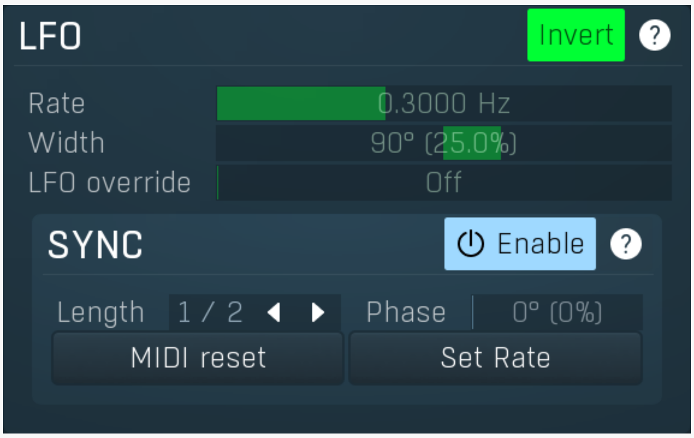
Max level

Max level defines the maximum level assumed on the input above which the filters have maximum frequency, Q and gain.
Range: silence to 0.00 dB, default 0.00 dB

Side-chain

Side-chain button enables the side-chain for the follower. Normally the follower is driven by the same signal which is filtered. By enabling this option you can plug anything into the plugin's sidechain and the follower will be driven by the level of that audio signal.

LFO panel



LFO panel contains parameters of the low-frequency oscillator.

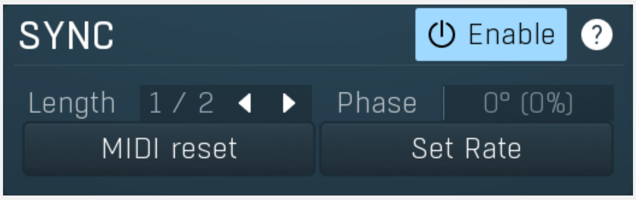
Invert
Invert button inverts the oscillator shape vertically.

Rate
Rate defines the speed of the low frequency oscillator. This is available only when synchronization to host is disabled.
Range: 0.0100 Hz to 100.0 Hz, default 4.000 Hz

Width
Width defines the phase difference between particular channels. This is very simple and often practical way to accomplish a kind of stereo expansion.
Range: -360Â° (-100.0%) to 360Â° (100.0%), default 0Â° (0%)

LFO override
LFO override lets you override the LFO and control the modulation value directly. This feature may offer several creative possibilities. You can then either automate it or even better, use the modulators (if the plugin provides any) to follow the input level, pitch, randomize etc. Set this below -1 to disable this feature. Please note that since there is only one parameter, by using it you will lose the possibility of having different values for each channel, hence potential stereoizing capabilities will not be available.
Range: Off to 100.0%, default Off

Synchronization panel



Synchronization panel contains parameters for the to-host synchronization.

Length
Length defines the note length to be used including the note type, such as straight notes or triplets and this determines the actual time/delay.
Example: '1/4 Straight' at 120 bpm = 500 ms, '1/4 Triplet' at 160 bpm = 281.25 ms.

Phase
Phase defines the phase offset of the to-host synchronization.
Range: 0Â° (0%) to 360Â° (100.0%), default 90Â° (25.0%)

MIDI reset
MIDI reset button displays the settings for the MIDI reset feature, which can reset the LFO based on incoming MIDI notes.

Set Rate
Set Rate button sets the **Rate** parameter used when sync is disabled according to current sync speed. This is useful when you want to leave the oscillator unsynchronized, however you want to start with the current synced speed.

Signal graph

Signal graph defines the low frequency oscillator shape.Signal-generator is an incredibly versatile generator of low & high frequency signals. It offers 2 distinct modes - Normal and Harmonics.
Normal mode is appropriate for low-frequency oscillators, where the graphical shape is relevant and is used to drive some form of modulation. For example, a tremolo uses this modulation to change the actual signal level in time. Frequencies for such oscillators usually do not exceed 20Hz as this is a sort of limit above which the frequencies become audible.
Harmonics mode is designed for high-frequency oscillators, where the actual shape is not as important as the harmonic content of the resulting signal, hence it is especially useful for actual audio signals. Please note that since a shape can contain more harmonics than those available from the harmonic generator, the results may not be exactly the same. As an example, a rectangular wave in normal mode may sound fuller than when converted to the harmonic mode.

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

Normal mode

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

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

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

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

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

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

Harmonics mode

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

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

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

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

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

Signal generation fundamentals

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

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

What happens above 20kHz?

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

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

What you see is not always what you get!

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

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

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

How can I generate non-harmonic frequencies?

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

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

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

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

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

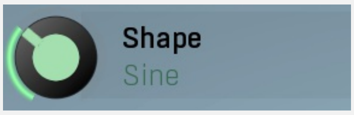
Normal
Normal button switches the generator into the normal mode, which lets you edit the shape of the oscillator. This is especially advantageous for low-frequency oscillators, where the shape matters even though it doesn't have any physical meaning.

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.

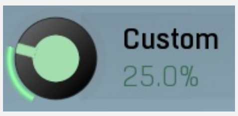
Harmonics
Harmonics button switches the generator into the harmonics mode, which lets you edit the levels and phases of individual harmonics. This is especially advantageous for high-frequency oscillators, hence sound generators.

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

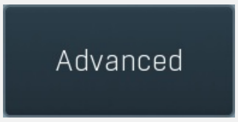
Step controls the amount of the step sequencer shape that is blended into the main shape (which has already been blended with the custom shape).

Edit

Edit button shows the step sequencer editor.

Smooth

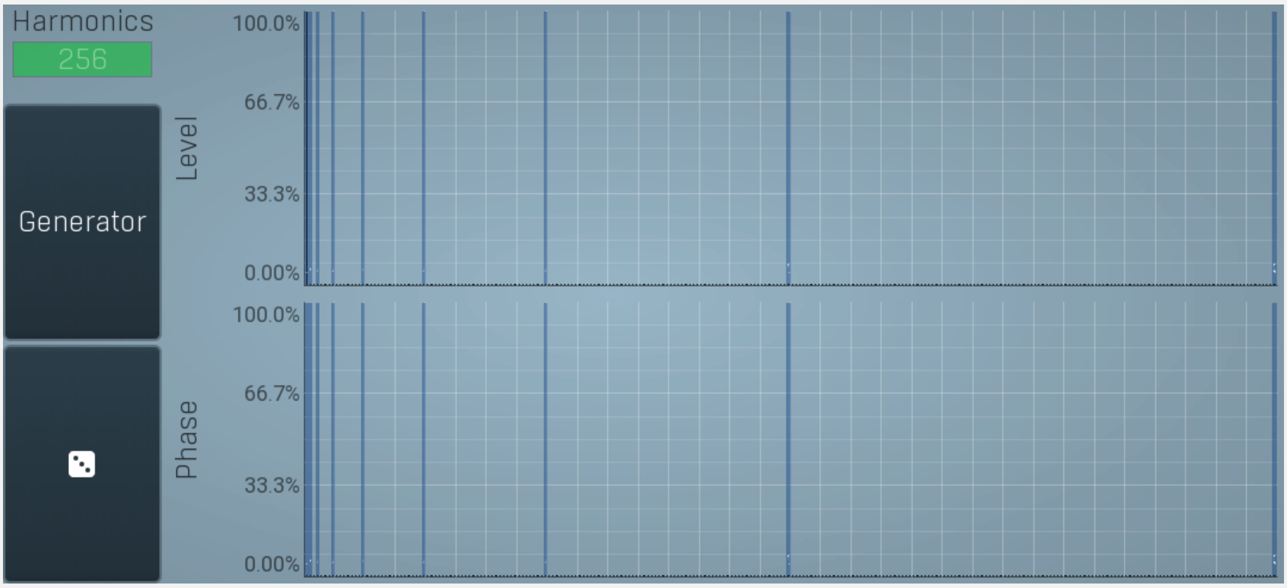
Smooth controls the amount of smoothing. Many shapes, especially those produced by the step sequencer, have rough jagged edges, which may be advantageous, but when used to modulate certain parameters, the output may be clicking or causing other artifacts. Smoothness helps it by smoothing the whole signal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



Advanced

Advanced 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.

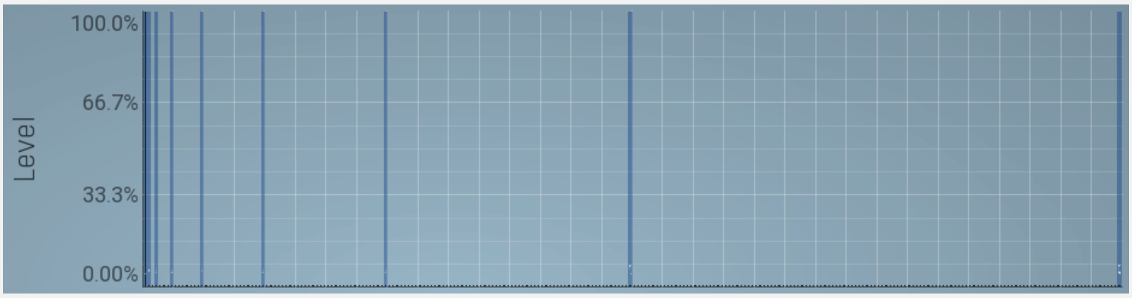
Editing harmonics can be time consuming unless you hear what you want, so a signal generator is also available. This great tool lets you generate a random spectrum by a single click. You can also open the **Generator** settings and edit its parameters, which basically control the audio properties in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

Generator

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

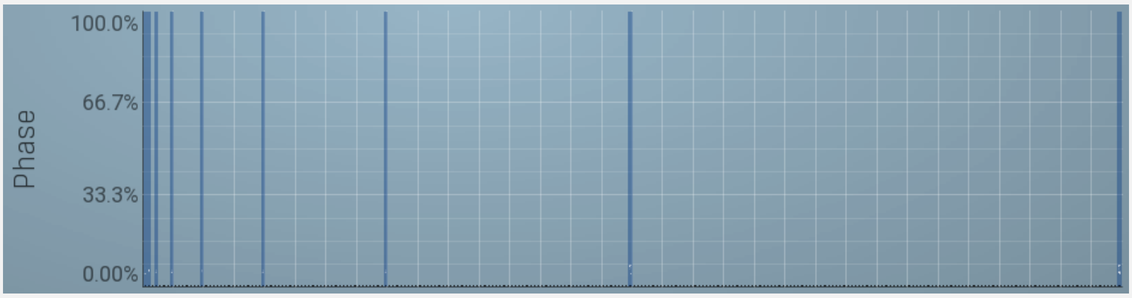
Randomize

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.



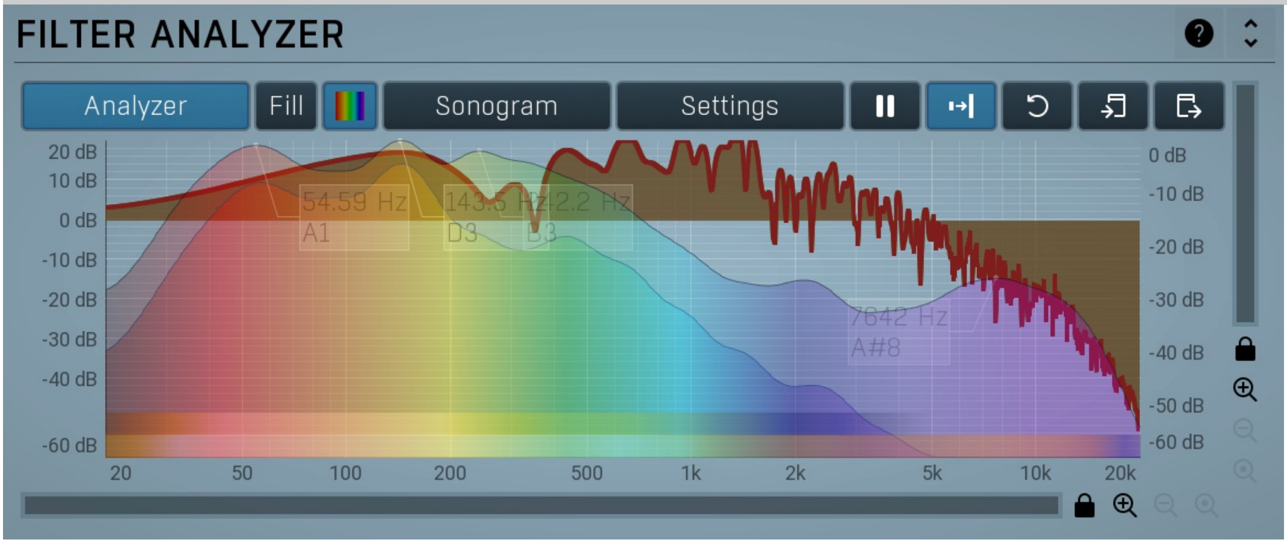
Magnitudes graph

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.



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

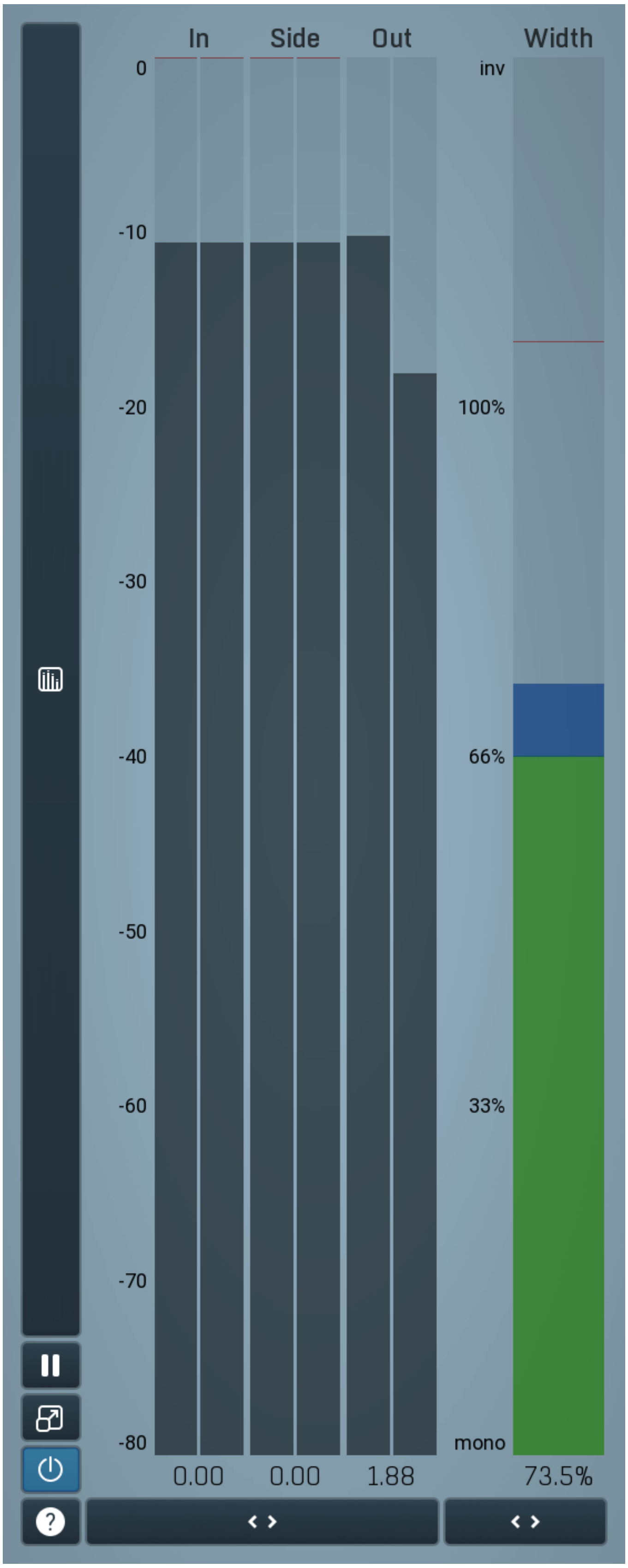
Reset button resets analyzer graphs. This is particularly useful when analyzing infinite average and maximum values.

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.



Global meter view

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

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

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

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

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

Width meter shows the stereo width at the output stage. This meter requires at least 2 channels and therefore does not work in mono mode. Stereo width meter basically shows the difference between the mid and side channels.
When the value is **0%**, the output is monophonic. From 0% to 66% there is a green range, where most audio materials should remain.
From 66% to 100% the audio is very stereophonic and the phase coherence may start causing problems. This range is colored blue. You may still want to use this range for wide materials, such as background pads. It is pretty common for mastered tracks to lie on the edge of green and blue zones.
Above 100% the side signal exceeds the mid signal, therefore it is too monophonic or the signal is out of phase. This is marked using red color. In this case you should consider rotating the phase of the left or right channels or lowering the side signal, otherwise the audio will be highly mono-incompatible and can cause fatigue even when played back in stereo.
For most audio sources the width is fluctuating quickly, so the meter shows a 400ms average. It also shows the temporary maximum above it as a single coloured bar.
If you right click on the meter, you can enable/disable loudness pre-filtering, which uses EBU standard filters to simulate human perception. This may be useful to get a more realistic idea about stereo width. However, since humans perceive the bass spectrum as lower than the treble, this may hide phase problems in that bass spectrum.

Time graph

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

Pause
Pause button pauses the processing.

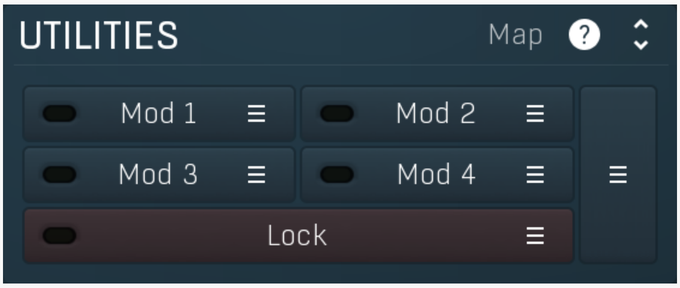
Popup
Popup button shows a pop-up window and moves the whole metering / time-graph system into it. This is especially useful in cases where you cannot enlarge the meters within the main window or such a task is too complicated. The pop-up window can be arbitrarily resized. In metering mode it is useful 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).

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.

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.