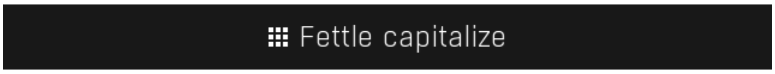
MFreqShifter



Presets

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

Holding **Ctrl** while pressing the button loads an existing preset, selected at random.

Presets can be backed up by using either the Export button, or by saving the actual preset files, which are found in the following directories:
Windows: C:\Users\{username}\AppData\Roaming\MeldaProduction
Mac OS X: ~/Library/Application support/MeldaProduction

Exported preset files can be loaded into the plug-in's preset store using the Import button. Or the preset files themselves can be copied into the directories named above.

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

Left arrow

Left arrow button loads the previous preset.

Right arrow

Right arrow button loads the next preset.

Randomize

Randomize button loads a random preset.

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.

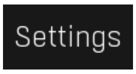
In addition, there are some mouse modifiers that assist this process. The smart randomization engine is used by default if no modifier keys are held.

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.

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

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

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

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

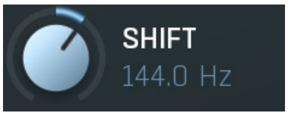
Smart interpolation adjusts the interpolation algorithm used when changing parameter values; the higher the setting the higher the audio quality and the lower the chance of zippering noise, but more CPU will be used.

WWW

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

Depth

Depth determines how powerful the effect is, thus this is the ratio between dry and wet signals.
Range: 0.00% to 100.0%, default 100.0%

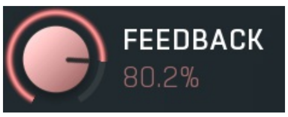


Shift

Shift defines the amount by which each frequency is shifted. The more you move each frequency, the more they will lose harmonic relationship and start sounding dirty and distorted.
Range: -10.0 kHz to 10.0 kHz, default 0 Hz

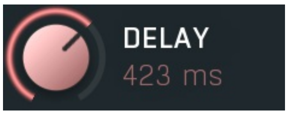
Width

Width controls the frequency shift between channels. It is applied as multiplicative coefficient, which means that if first channel is shifted by S, then second channel is shifted by S*Width. If you are using a multichannel setup, then the next channel will be S*Width*Width etc. For example, if you set shift to 100Hz and width to -100% in stereo setup, then left channel is shifted by +100Hz and right one by -100Hz. Any value other than 100% makes each channel shifted by a different amount, which can provide more sound width.
Range: -200.0% to 200.0%, default 100.0%



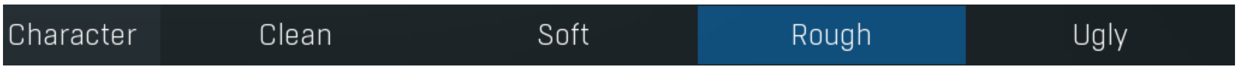
Feedback

Feedback controls how much of the output signal is sent back with some delay. As a result such signal is shifted every time it is fed back. This results in sequences with a constantly changing spectrum.
Range: 0.00% to 99.0%, default 0.00%



Delay

Delay determines the feedback delay. You can also use synchronization below to make the delay adjust dynamically to the song's tempo.
Range: 0 ms to 1000 ms, default 10 ms



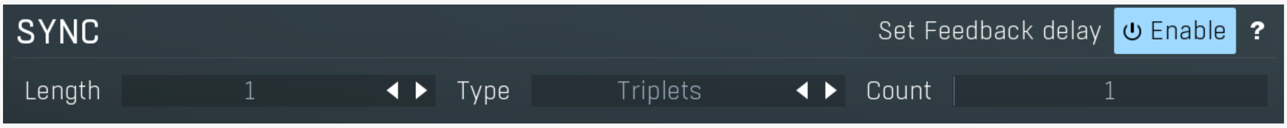
Character

Character controls a filtering system used in the plugin and determines the output sound character. There is no comparison in sound quality, you have to choose the one you like for any particular audio.

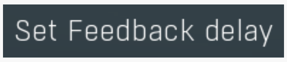
Prefiltering

Prefiltering enables the input filter, which removes those frequencies which cannot be shifted, because they are too high or too low to be represented and reproduced. If these frequencies are not removed using this filter, they will present themselves as aliasing, which may or may not be desired. For cleaner results you should keep the filter enabled.

Synchronization panel

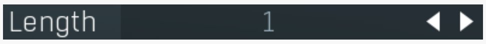


Synchronization panel contains parameters for the to-host synchronization.



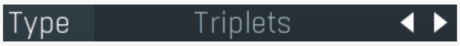
Set Feedback delay

Set Feedback delay button sets the **Feedback delay** parameter used when sync is disabled according to current sync speed. This is useful when you want to leave the processor nonsynced, however you want to start with current synced state.



Length

Length defines the note length to be used.



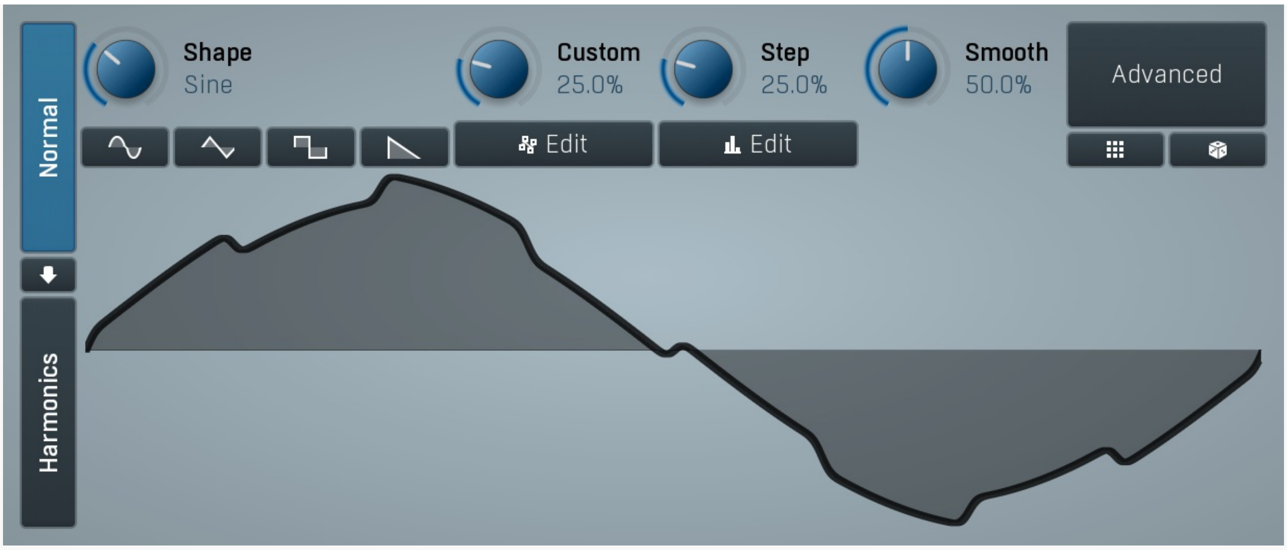
Type

Type defines the note type, such as straight notes or triplets, to be used. Together the **Length** and **Type** determine the actual time/delay.
Example: '1/4 Straight' at 120 bpm = a delay of 500 ms, '1/4 Triplet' at 160 bpm = a delay of 281.25 ms.

Count

Count defines the number of the units, hence multiplies of the sync length.
Range: 1 to 64, default 1

Signal editor



Signal editor 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

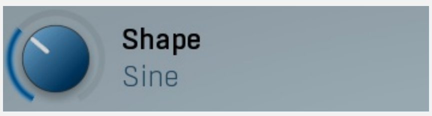
Convert button converts the current shape into harmonic-based representation. Please note that since the number of harmonics is limited, the result will not perfectly resemble the original shape.

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.

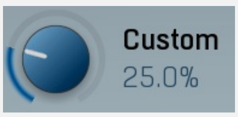
Signal generator in Normal mode

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.



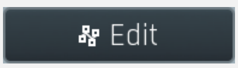
Shape

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

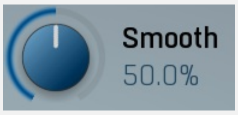
Edit button shows the custom shape editor.

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).

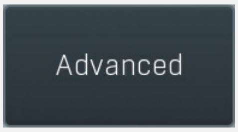
Edit

Edit button shows the step sequencer editor.



Smooth

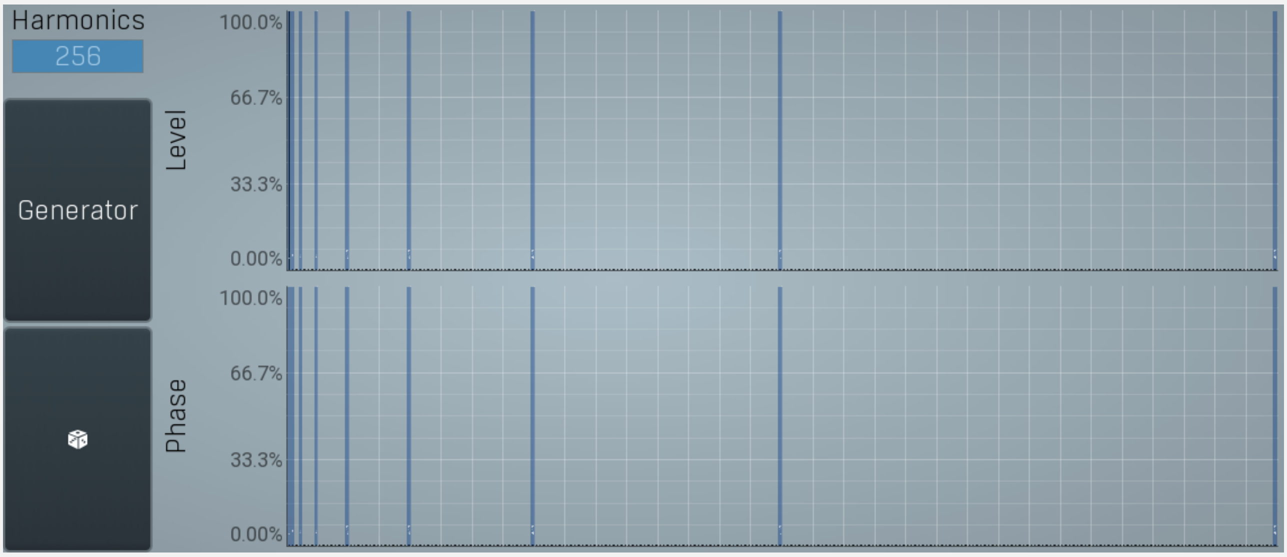
Smooth controls the amount of smoothing. Many shapes, especially those produced by the step sequencer, have rough jagged edges, which may be advantageous, but when used to modulate certain parameters, the output may be clicking or causing other artifacts. Smoothness helps it by smoothing the whole signal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

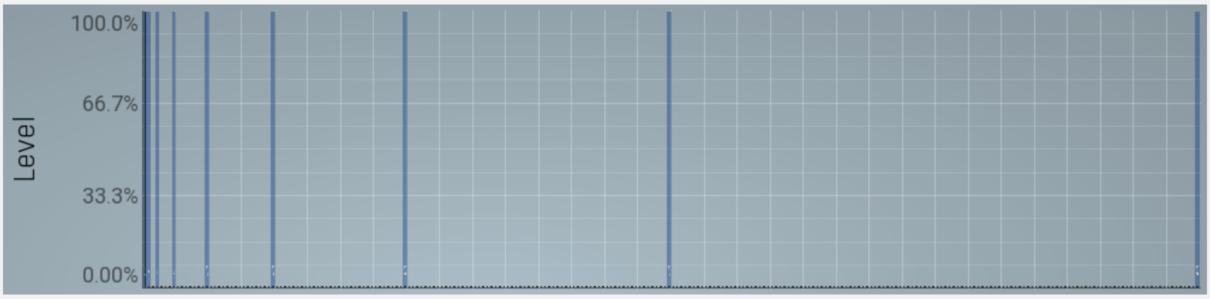
Editing harmonics can be time consuming unless you hear what you want, so a signal generator is also available. This great tool lets you generate a random spectrum by a single click. You can also open the **Generator** settings and edit its parameters, which basically control the audio properties in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

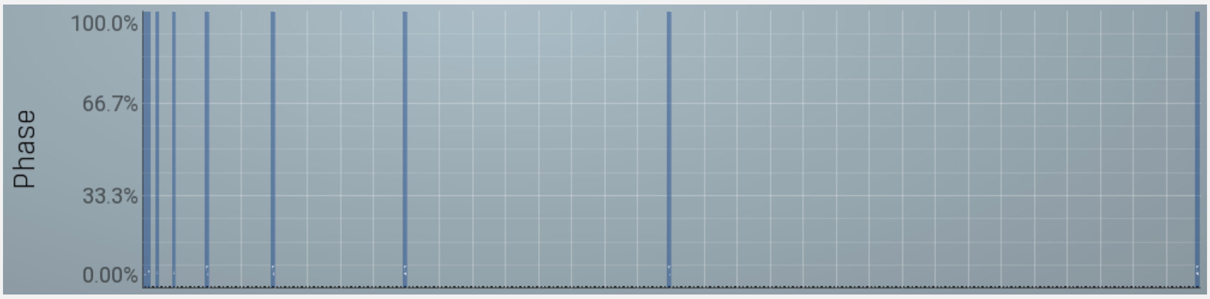
Randomize

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.

Global meter view

Global meter view provides a powerful metering system. If you do not see it in the plug-in, click the **Meters** or **Meters & Utilities** button to the right of the main controls. The display can work as either a classical level indicator or, in time graph mode, show one or more values in

time. Use the first button to the left of the display to switch between the 2 modes and to control additional settings, including pause, disable and pop up the display into a floating window. The meter always shows the actual channels being processed, thus in M/S mode, it shows mid and side channels.
In the classical level indicators mode each of the meters also shows the recent maximum value. Click on any one of these values boxes to reset them all.

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

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

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

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

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

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.

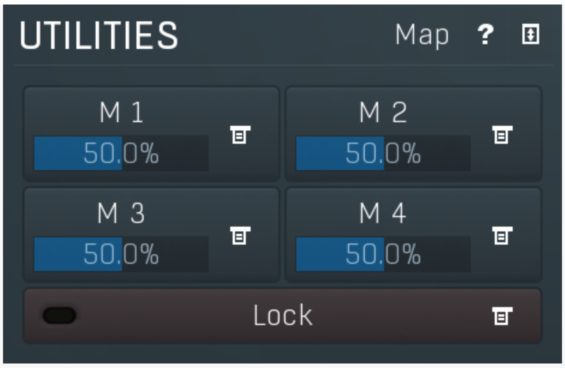
Popup
Popup button shows a pop-up window and moves the whole metering / time-graph system into it. This is especially useful in cases where you cannot enlarge the meters within the main window or such a task is too complicated. The pop-up window can be arbitrarily resized. In metering mode it is useful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

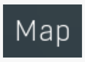
Enable
Enable button enables or disables the metering system. You can disable it to save system resources.

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





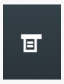
Map

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

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.

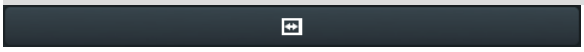
Reorder to ... lets you change the order of the multiparameters. This can be useful when creating active-presets. Please note that this feature can cause problems when one multiparameter controls other multiparameters, as these associations will not be preserved and they will need to be rebuilt.

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.

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.