# MCharacter



## Overview

MCharacter is a unique tool, that lets you directly access the harmonics of your monophonic audio material. It can be used on vocals, bass, trumpet, saxophone... just about any instrument that is monophonic. It opens a range of audio processing and creative possibilities.

## Leveling harmonics and changing character

The original idea was about mixing however - in many cases equalization just doesn't fill the needs, because the audio being processed is constantly moving in the frequency spectrum. Sure you can automate the whole thing, but that would be extremely time consuming and results inaccurate. Hardware mixing consoles have so many limitations that anything like this was simply unimaginable; the digital world gives you yet another marvellous possibility - moving the equalizer automatically depending on the pitch. You can do that with a combination of MAutoDynamicEq and MFilter. However it is still not a perfect solution because it needs to mix 2 worlds - spectrum domain for pitch detection and time domain for the actual processing. That in most cases makes following the pitch quite inaccurate and problematic.

Accessing the harmonics themselves is yet another problem, because the parametric equalizers, no matter how sophisticated, don't make this very unnatural and scientific operation very simple.

And that's exactly what MCharacter is about. Most audio instruments have a distinctive character defined by their harmonics and formants. While formants are pretty much static in the spectrum, so you can change them directly with an equalizer, harmonics are more problematic, because they follow the pitch. MCharacter lets you control the levels of first 32 harmonics in a very simple and effective way - with the **Levels graph**. By working directly in spectrum domain the plugin's ability to detect pitch is outstanding and the changes are instant, which makes it outperform traditional equalizers in just about every way.

What exactly are harmonics? Let's say you will be singing a tone with pitch of 100Hz. The 100Hz is called fundamental frequency and in most cases is dominant, or at least one of the most prominent frequencies. Your voice then also generates 200Hz, 300Hz, 400Hz... These are called harmonics and the levels of them define considerable part of the instrument's character. 200Hz is 1 octave above the pitch, 400Hz is 2 octaves above, 800Hz is 3 octaves etc.

## Creative potential

The plugin also contains 2 synthesis methods - Harmonic synthesis via the **Synthesis graph** and an actual note synthesis via the **Notes graph**. These actually work pretty much the same way, but their range is different. The idea is, that the plugin can actually synthesize any frequency related to the pitch.

For example, if your audio material doesn't contain 8th harmonic at all, or is very low in level, use the Synthesis graph to generate it. The plugin essentially takes each different frequency that exists inside your audio material and adds its shifted version into the signal again. That makes it closely follow the character of the audio material making the results quite natural.

Notes graph serves a different purpose - while the synthesis algorithm is exactly the same, it doesn't let you create frequencies at locations of harmonics, but instead it uses transposition - it lets you generate frequencies at -2 octaves to +2 octaves from the fundamental. So for

example, if you want a high quality octaver producing frequency 1 octave below the pitch, this is the way to go ;).

There are also many possibilities using the plugin's modulators. For example, you can randomly change the harmonics in time giving a living character to otherwise dead synthetic lead. Or change the character depending on the input level. The only limit is your imagination ;).

### ⠿ More Head
**Presets**

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

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

Presets can be backed up by 3 different methods:
A) Using "Backup" and "Restore" buttons in each preset window, which produces a single archive of all presets on the computer.
B) Using "Export/Import" buttons, which export a single folder of presets for one plugin.
C) By saving the actual preset files, which are found in the following directories (not recommended):
Windows: C:\Users\{username}\AppData\Roaming\MeldaProduction
Mac OS X: /Library/Application support/MeldaProduction

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

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

### ◀ Left arrow
Left arrow button loads the previous preset.

### ▶ Right arrow
Right arrow button loads the next preset.

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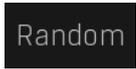
### Random Randomize
Randomize button (with the text 'Random') generates random settings. Generally, randomization in plug-ins works by selecting random values for all parameters, but rarely achieves satisfactory results, as the more parameters that change the more likely one will cause an unwanted effect.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

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

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

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

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

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

**Licence manager** lets you activate/deactivate the plugins and manage subscriptions. While you can simply drag & drop a licence file onto

the plugin, in some cases there may be a faster way. For instance, you can enter your user account name and password and the plugin will do all the activating for you.

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

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

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

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

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

### 🏠 WWW

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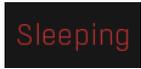
### Sleeping  Sleep indicator

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

## Globals panel

Globals panel controls the main parameters of the plugin.

### Level transformation  Level transformation

Level transformation lets you perform arbitrary transformation of the harmonics based on their current level. You can for example compress the levels of the harmonics leading to less difference between their levels making the output richer.
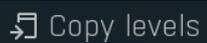
### Copy levels  Copy levels

Copy levels button copies the harmonics of the currently processed sound to the system clipboard. You can then use the **Paste levels** button to impose the harmonic structure of one sound to another.

### Paste levels  Paste levels

Paste levels button pastes the harmonics from the system clipboard by first analyzing the harmonics of the currently processed sound and then setting up the levels so that the harmonics have the same levels as those from the clipboard. This way you can impose harmonic structure of one sound onto the one being processed.

### DRY  Dry

Dry controls the amount of non-synthesized signal on the output. This includes the input signal with its processed harmonics levels. You can lower this value to remove the original audio and leave only the completely synthesized signal in the output.
Range: 0.00% to 100.0%, default 100.0%

### LEVELS  Levels

Levels controls the amount of level transformation. It is available via the first graph, which defines the level adjustment for each harmonic. The plugin first detects the pitch of the sound, say 100Hz. You can then adjust the gain for each harmonic. The first harmonic is the fundamental - the 100Hz, 2nd is 200Hz, 3rd 300Hz etc. The power-of-2 harmonics are highlighted in the graph because these correspond to octaves (the most natural harmonics).
Range: 0.00% to 100.0%, default 100.0%

### Synthesis

Synthesis controls the amount of synthesized signal. It is available via the second graph, which defines the level for each synthesized harmonic. The plugin first detects the pitch of the sound, say 100Hz. You can then adjust the level for each harmonic to be synthesized. The first harmonic is the fundamental - the 100Hz, 2nd is 200Hz, 3rd 300Hz etc. The power-of-2 harmonics are highlighted in the graph because these correspond to octaves (the most natural harmonics).
Range: 0.00% to 100.0%, default 100.0%
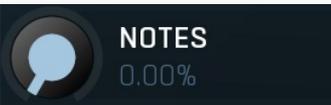
### Notes

Notes controls the amount of synthesized signal for pitch shifted frequencies. It is available via the third graph, which defines the level for each synthesized frequency. The plugin first detects the pitch of the sound, say 100Hz. You can then adjust the level for each frequency to be synthesized. The highlighted frequency in the middle corresponds to the current pitch, the fundamental frequency. Then there are 2 octaves below and above, octaves are highlighted. It can be used for example as a very high quality octaver / subharmonic generator.
Range: 0.00% to 100.0%, default 100.0%

### Output gain

Output gain defines the power modification applied to the output signal. You can use it to make the processed signal similar in loudness if needed, hence making dry/wet easier to use.
Range: -24.00 dB to +24.00 dB, default 0.00 dB

# Detector panel

Detector panel contains advanced settings for the pitch detector that the plugin uses. Use the parameters to adjust it to your audio material.

### Side-chain

Side-chain button activates the side-chain input. When enabled, the processor doesn't analyze the pitch of the regular input and uses the secondary input instead.

### MIDI

MIDI button activates the MIDI input. When enabled, the processor allows any pitch detection to be overridden using MIDI input on channel 1. In that case whenever you press a MIDI note, its frequency is used instead of detecting pitch of any input signal.

### Threshold

Threshold determines the minimal signal level above which the effect is applied. Many natural instruments, such as vocals, contain noise-like gaps, which could confuse the pitch detector. These should also not be processed at all, because they in fact do not have a pitch, otherwise the processing could lead to sort of alien sounds. Since these sounds are usually quite low in level, you can simply use this parameter to exclude these sections from the processing.
Range: silence to 0.00 dB, default -40.0 dB
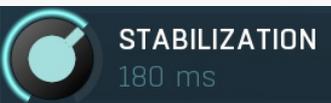
### Stabilization

Stabilization specifies how quickly can the pitch make bigger changes. This can be useful for more complicated material, such as voice, which often contains short pieces of inharmonic material, which would normally make the detector jump too quickly.
Range: 0 ms to 1000 ms, default 10 ms

### Speed

Speed specifies how quickly the pitch can change. By lowering this value the pitch won't be able to change quickly, which can help avoiding unwanted vibratos for example. It can also be used creatively.
Range: 0.00% to 100.0%, default 100.0%

### Min F

Min F controls the minimum frequency that the pitch detector can calculate. Most signals contain lots of noise, low-frequency rumble, high frequency harmonics and other components, which may make the detector report an incorrect fundamental pitch. This limit helps the detector ignore the irrelevant components and ensure the processor works quickly with maximum accuracy with all reasonable signals.
Range: 20.00 Hz to 20.0 kHz, default 20.00 Hz

### Max F

Max F controls the maximum frequency that the pitch detector can calculate. Most signals contain lots of noise, low-frequency rumble, high frequency harmonics and other components, which may make the detector report anincorrect fundamental pitch. This limit helps the detector ignore the irrelevant components and ensure the processor works quickly with maximum accuracy with all reasonable signals.
Range: 20.00 Hz to 20.0 kHz, default 1000 Hz

# Spectral settings panel

Spectral settings panel controls the properties of the spectral transformation the plugin operates it. The input signal you are working with is in so-called time-domain. The problem is, the processing that can be performed in time domain is very limited. So the plugin performs a high-quality transformation to so-called frequency (or spectral) domain, where there are lots of additional possibilities. After the processing the plugin converts the data back to time-domain, so that the output can be played and additionally processed. This panel controls properties of both these transformations.

### Buffer size

Buffer size controls the block size used for processing. This plugin performs processing in the so-called spectral domain. This allows it to access features that are normally unavailable, however in order to do that it requires the audio to be separated into blocks of audio. As a result, the plugin causes latency. This setting controls the latency length. Additionally, the higher it is the more detail the plugin has, which usually provides higher audio quality (but this is not always the case!), at the expense of greater CPU cost and increased latency. Also note that with some settings having too high a buffer size will produce a sort of time-smearing, ambient-like sound quality. Also note that this value is assigned only for sampling rates around 44-48KHz, the engine may readjust it for higher sampling rates in order to get similar audio results.
Range: 256 to 16384, default 2048

### Resolution

Resolution defines how accurately the processor can analyze the audio. The lower the resolution, the more CPU is needed, but also more of the time domain characteristics are preserved, hence potentially higher audio quality.
Range: 1.0 ms to 100 ms, default 25 ms

### M/S processing

M/S processing makes the plugin intentionally process mid/side instead of left/right channels. This usually keeps better stereo coherence. If you disable this, the results usually slowly cumulate error between left and right channels, gradually shifting the stereo field. Thought this can sort of create some artificial stereo, it cannot be controlled and is usually unwanted.

| LEVELS | SYNTHESIS | NOTES |
|--------|-----------|-------|

**Tab selector**

Tab selector switches between subsections.
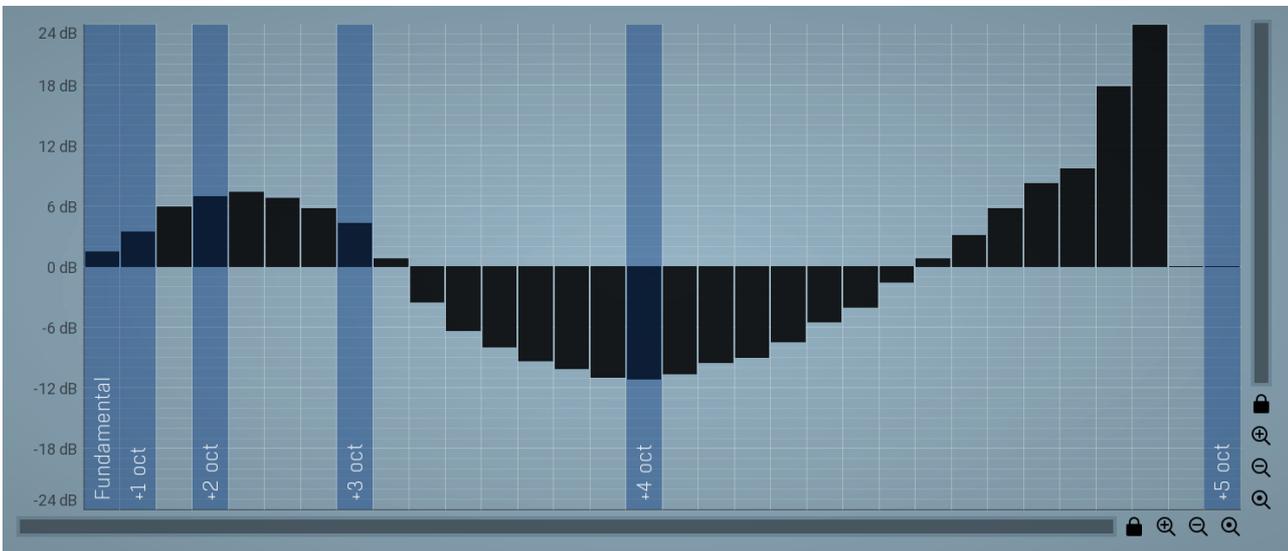
**Level**

## graph

Level graph contains levels for each harmonic. The plugin first detects the pitch of the sound, say 100Hz. You can then adjust the gain for each harmonic. The first harmonic is the fundamental - the 100Hz, 2nd is 200Hz, 3rd 300Hz etc. The power-of-2 harmonics are highlighted in the graph, because these correspond to octaves (the most natural harmonics).
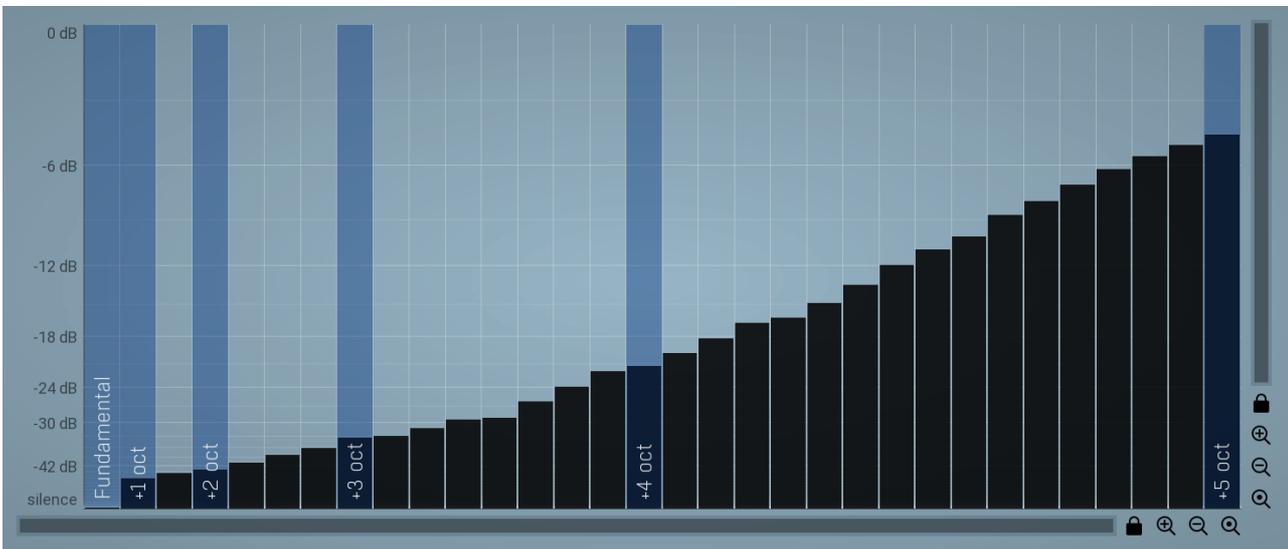


## Synthesis graph

Synthesis graph contains levels for each synthesized harmonic. The plugin first detects the pitch of the sound, say 100Hz. You can then adjust the level for each harmonic to be synthesized. The first harmonic is the fundamental - the 100Hz, 2nd is 200Hz, 3rd 300Hz etc. The power-of-2 harmonics are highlighted in the graph, because these correspond to octaves (the most natural harmonics).
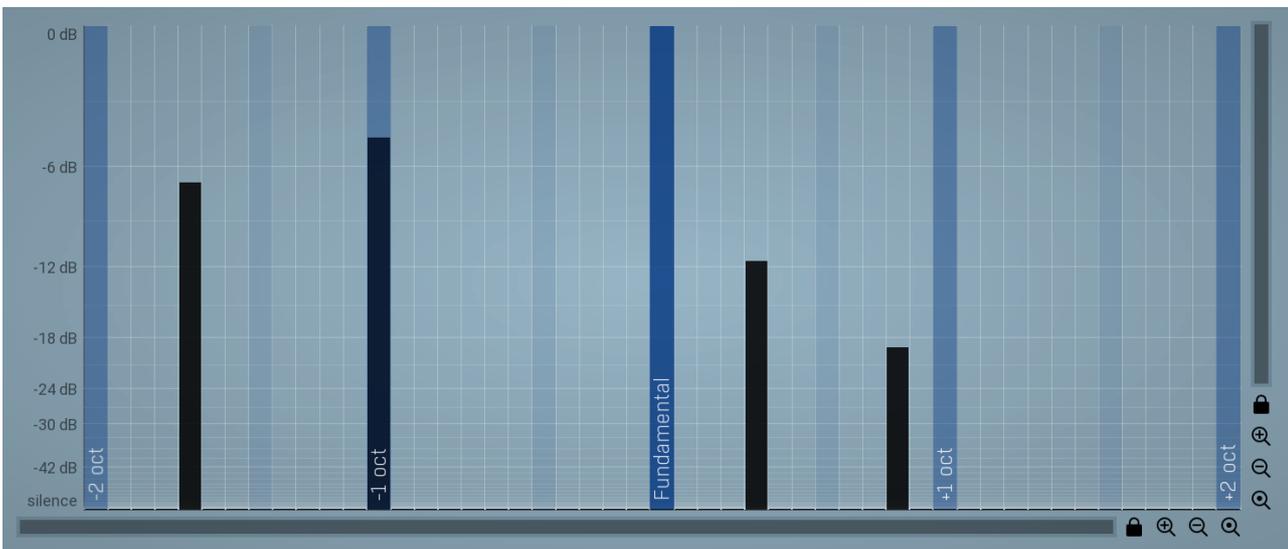


**Notes**

## graph

Notes graph contains the level for each synthesized frequency. The plugin first detects the pitch of the sound, say 100Hz. You can then adjust the level for each frequency to be synthesized. The highlighted frequency in the middle corresponds to current pitch, the fundamental frequency. Then there are 2 octaves below and above, octaves are highlighted. It can be used for example as a very high quality octaver / subharmonic generator.
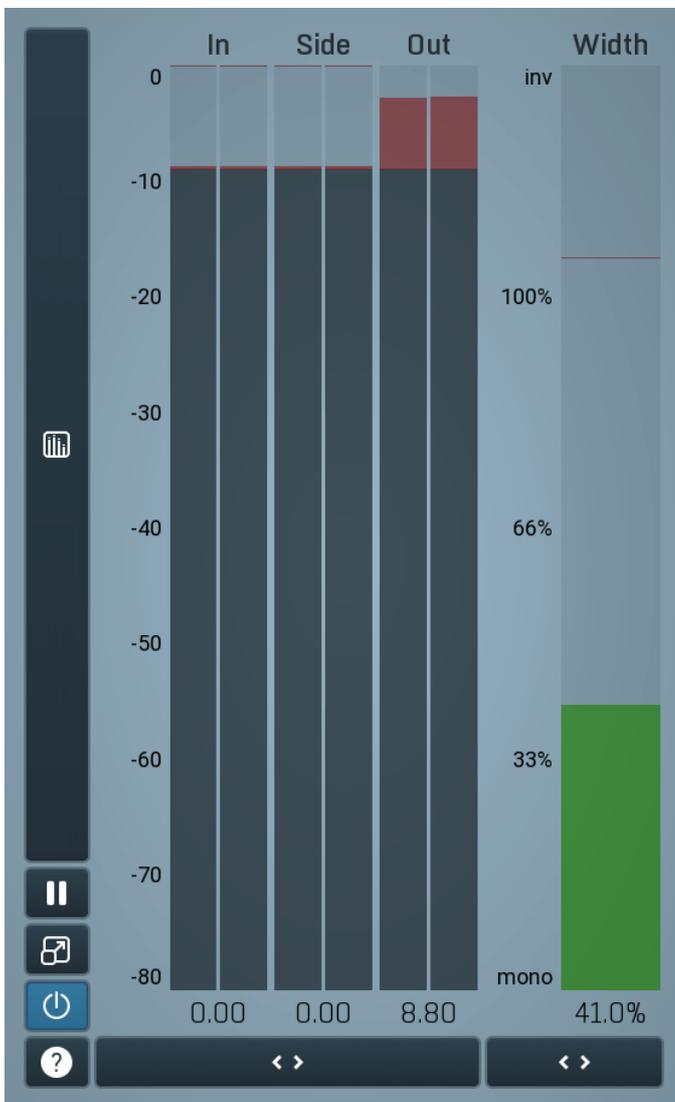
## Global meter view

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
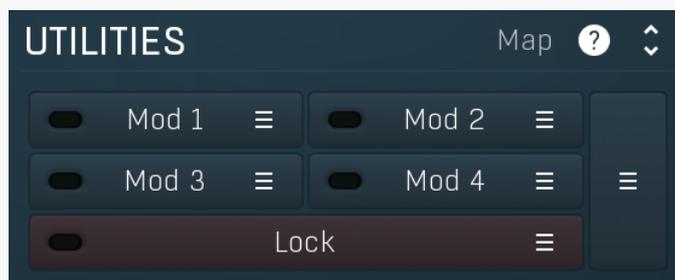
**Map**

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.
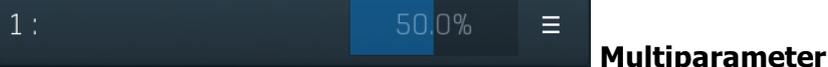
**Multiparameter**

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.