MDrumReplacer

MDrumReplacer is perhaps the most powerful drum replacing plugin on the market. It's purpose is to detect hits in acoustic drums and percussion recordings and synthesize it's own drum performance based on that, usually using samples of well recorded acoustic drums, which is then mixed with the original audio or even replaces it. Originally these tools have been used for cases when the acoustic drums haven't been recorded well, but it is commonly used to improve any kind of percussion tracks these days, for example by layering a synthetic kick over an acoustic kick recorded by the drummer.

Sound generator in MDrumReplacer

The plugin features the synthesis unit from MDrummer, which can produce anything from synthesized drums to high quality multisampled acoustic drums. The plugin shares drum data with MDrummer, so if you use that one, you won't need to install any further factory content, and it will also be quite easy for you to grasp the sound engine. If you do not use MDrummer, then you will need to install the drum packs from our website. You will be guided there when you open the plugin for the first time.

In most cases you will simply select a drum from the **Drum library** tab. The components in there are shared with the components from MDrummer. However you may also edit every aspect of the generated sound the same way you would do in MDrummer via the **Drum editor** tab.

Note detection in MDrumReplacer

The plugin features powerful detection unit from MDrumLeveler including all its leveling features. It's important to understand how the unit works and most is visualised in the **Detection** tab.

The input signal level is measured (potentially prefiltered), and based on its levels notes (drum hits) are detected. If you look at an average drum hit waveform, you can visually estimate its velocity by the highest peak. Something like that is done by the plugin, but since the hit actually starts earlier, in order to detect velocities properly, the plugin needs to "look into the future" - introduce latency. You can control the latency by the **Latency** parameter. The higher it is the better, but there's rarely a need to go above say 50ms.

The plugin needs to determine what is hit and what is not and for you the task is usually simple - just set the **Threshold** in the graph properly. For some specific materials, especially those with lots of bleed between drums the work can get problematic, we will investigate these below.

The detected hits and the levels are displayed as red dots in the graph. But there are also black dots, which represent the leveled hits. Drum leveling is based on the **Velocity markers** and lets you improve the fluctuations of the performance by trying to get closer to the nearest velocity marker. That's the algorithm used by MDrumLeveler, which is designed to improve drum performances and is available here too via the **Leveling** parameter. It's like an automatic tuner for vocals.

The black dots represent levels of each hit, but now its needed to transform that to actual note velocities for the virtual drummer to play. The plugin simply converts the levels to velocities as percents (0% - 100%). In many cases this could lead to make the synthesized drum hits too soft or hard. That's when the **Velocity shift** and **Velocity shape** parameters come handy. If that's not enough, you can arbitrarily transform all hits using the **Custom velocity transform** feature.

The typical scenario

Let's say you are not happy with the kick in your drum recording, this is what you do in most cases:

1. Insert the plugin on the bass drum track.
2. Set the threshold marker so that only the events you want to process are detected.
3. Select a drum sound - just choose something from the **Drum library** tab.
4. Use drum leveling if desired - increase the **Leveling** and set the **Velocity markers**, so that hits become as consistent as you'd like them to be.
5. Control the velocity detection using the **Velocity shift** and **Velocity shape** parameters, so that the virtual drummer plays nicely with the original.
6. Set the amount of dry and wet signals - in most cases you will probably mix a little bit of the synthesized signal to your recorded performances, but in contemporary music its not uncommon to completely discard to original audio and replace it with the generated signal entirely (dry 0%, wet 100%).

Targeting the processor to specific drum hits

When your drum kit is well recorded without much leakage, the job is easy. But recording drums is generally the most problematic exactly because it is never perfect. And to make the plugin work well, you need to make it detect the right hits. So let's go through some typical scenarios.

Bass drum

Bass drum is usually easy enough. A microphone is placed from the back (not the beater), usually inside. That avoids most leakage. If there is some, use a **band-pass** filter in the (**Prefiltering** panel) with some pretty low frequency, say 50Hz and rather a low Q. Most other drums are not emitting anything below 100Hz, so this should do the trick just fine. If you need it, you can use a higher Q, but it may resonate too slowly, so the detector signal may be too lazy. Therefore always try to use lower Q if possible.

If you used some nonstandard approach or used several microphones, e.g. one from the front, which can pick-up a lot of snare sound, make at least one of the microphones work well and use this side-chain trick for those which cannot be fixed : All of the microphones are recording the same performance, so first use MAutoAlign to get them in phase (you should do that anyway). Then send the drum leaking to your microphone to the plugin's sidechain and enable **Subtract side-chain** feature, which will suppress the hits if the side-chained signal contains them.

Toms

Tom microphones often record lots of leakage from literally everything. The good thing is that they are not played that much, so in the worst case you can automate some parameters, e.g. threshold, but also that the toms are often quite "tuned". Actually a good practice is to tune the toms to the scale of the song, but not many musicians do that probably. Anyway, the fact they are tuned makes it possible to use a band-pass filter and find the exact frequency of the drum.

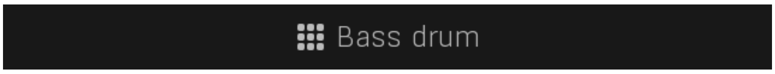
Try a sonogram (MAnalyzer, MAutoDynamicEq and all our equalizers etc.), it makes it easy to spot the frequency easily. Classic analyzers don't work that well, because all of the drum hits everywhere filling the spectrum. The sweet spot is usually between 100Hz and 200Hz. Then use a higher Q (the lower the better but...) so you can set the threshold appropriately. The toms usually don't have a very quick transient anyway, so smoothing it even more with a resonant band-pass filter doesn't harm that much. If it does, try lowering the **Transients** parameter to make the plugin focus less on the transients that are not there anymore. It can especially help with leakage from the snare drum.

In extreme cases a bass drum (or snare drum) can interfere a lot with the toms and from brief listening to the unprocessed track you may be able to tell that the leakage is just too high. In that case you can again use the side-chain trick mentioned above.

Snare drum

Snare drum is probably the most problematic yet the most important (along with the bass drum). There is usually lots of leakage from the hi-hat and just about any other part of the drum kit. It is often recorded using multiple microphones from top, bottom and in rare cases even side.

Again we need to remove leakage. Snare drums may be tuned, but unless the snares are loosened, there is no specific pitch other than what the metal rim produces and that's rarely useful. So try finding some frequency that is present a lot in the drum hit, but not in other parts of the kit. A sonogram can be a huge help here again. The spot will usually be somewhere between 200Hz and 600Hz. If the drummer uses a rim shot, that one may have a completely different pitch defined by the drum, the stick and even the position where the back end of the stick was touching the drum head. But there usually is some form of pitch if the drummer is good enough, so you can use the second band-pass filter to target this frequency as well. The filters are in parallel, so this way you can make one plugin instance tuned to both the normal hits and the rim shots.



Presets

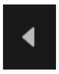
Presets button shows a window with all available presets. A preset can be loaded from the preset window by double-clicking on it, selecting via the buttons or by using your keyboard. You can also manage the directory structure, store new presets, replace existing ones etc. Presets are global, so a preset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

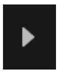
Files are named based on the name of the plugin like this: "{pluginname}.presets", so for example MAutopan.presets or MDynamics.presets. If the directory cannot be found on your computer for some reason, you can just search for the particular file.

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

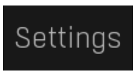
Right arrow button loads the next preset.

Randomize

Randomize button loads a random preset.

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.

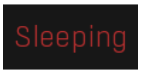
Global system settings contains some settings for all MeldaProduction plugins. Once you change any of them, restart your DAW if needed, and it will affect all MeldaProduction plugins.

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

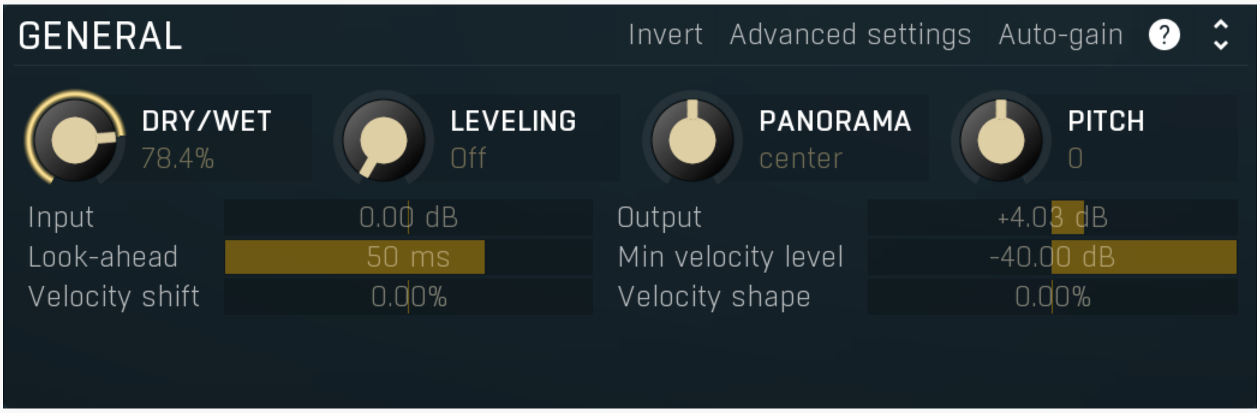
WWW button shows a menu with additional information about the plugin. You can check for updates, get easy access to support, MeldaProduction web page, video tutorials, Facebook/Twitter/YouTube channels and more.



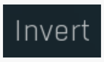
Sleep indicator

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.

General panel

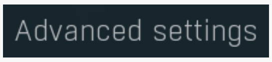


General panel contains the main parameters controlling the plugin behaviour.



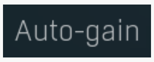
Invert

Invert inverts the phase of the generated signal. Use this if you hear phase cancellation between the original and generated signals.



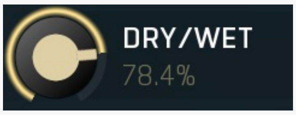
Advanced

Advanced button displays additional settings.



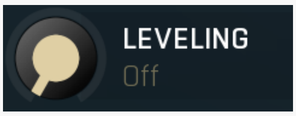
Auto-gain button

Auto-gain button analyzes the drum sound and sets the drum gains for optimal processing levels and to maintain similar loudness to other drums/presets.



Dry/Wet

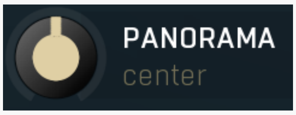
Dry/Wet controls the ratio between the dry input and the synthesized signal. Note that when the **Leveler** is used, its output is counted as dry signal, so you need to control the amount of leveling using **Leveling** and **Gate** parameters directly.
Range: 0.00% to 100.0%, default 50.0%



Levelling

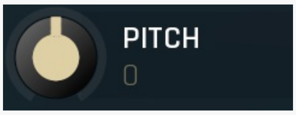
Levelling controls velocity processing in the synthesized drum signal and if **Leveler** panel is enabled, it also controls the amount of drum leveling applied to the input signal.
Setting this to minimum (off) makes the drum replacer closely follow the levels of the input hits to determine the velocities of the hits the plugin plays. In this case, enabling the **Leveler** still makes sense, since the **Gate** still works even if velocities are untouched.
Using the leveler is often advantageous to improve the performance however. This parameter controls how close the output levels should move towards the closest of the 4 velocity level markers that the plugin provides. The higher the value the more the output level moves towards the velocity marker.
For example, if the detector finds an event with level say -30dB, your nearest velocity marker is at say -20dB, and you set the leveling to

100%, the plugin will behave like the hit was actually at -20dB. This makes the performance more 'perfect', potentially even 'too good', almost robotic, so you may want to lower this value.
Range: Off to 100.0%, default Off



Panorama

Panorama controls the panorama applied on the synthesized drum signal.
Range: 100% left to 100% right, default center



Pitch

Pitch controls the pitch of the synthesized drum signal.
Range: -24.00 to +24.00, default 0

Input gain

Input gain defines the power modification applied to the input signal.
Range: -24.00 dB to +24.00 dB, default 0.00 dB

Output gain

Output gain defines the power modification applied to the output signal.
Range: -24.00 dB to +24.00 dB, default 0.00 dB

Look-ahead

Look-ahead controls how far ahead in time the plugin looks. Since nothing can really look into the future, this in effect causes a delay, which is reported to your host as latency, so a properly functional host would compensate for it and it would cause no problems during mixing or mastering. However you may need to address it when using the plugin real-time. The bigger the look-ahead is, the higher the latency of course.
The further it can look into the future, the bigger the chance that it finds the highest peak of each event. With 0ms look-ahead the plugin can still detect events, but it cannot know their level, because when it detects them, the input level is still very low. But if it can look into the future, then it can wait for the highest peak.
So how much should you set? If you look at a drum hit in a wave editor, the typical time between the start of the hit and the moment with maximum level (the length of the initial 'hill') is usually less than 4ms, but with bass drums it can often reach 10ms. If you are using it real-time, you want the delay to be as low as possible. It is recommended to keep it at least 2ms.
Range: 0 ms to 100 ms, default 50 ms

Min velocity level

Min velocity level controls the event level to be considered minimum velocity. By using it you activate the logarithmic scaling, meaning that the vertical scale in the analyzer is indeed transformed as you see it into velocities. For instance, assuming the default value of -40dB, an event at -40dB will be considered velocity 0, -20dB velocity 0.5 and 0dB velocity 0. Set it to maximum, to use the normal peak level transformation instead, which doesn't see to work too naturally in most cases though.
Range: -80.00 dB to Off, default -40.00 dB

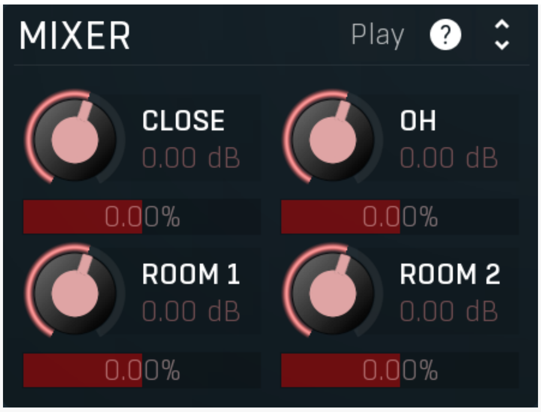
Velocity shift

Velocity shift lets you arbitrarily increase or decrease the velocities of the detected notes and comes handy when the synthesized hits are too soft or loud. You can also arbitrarily transform the velocities using the **Custom velocity transform**.
Range: -100.0% to +100.0%, default 0.00%

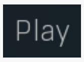
Velocity shape

Velocity shape lets you shape the velocities of the detected notes and comes handy when the synthesized hits tend to be mostly soft or mostly loud. It is an equivalent of the velocity curve you know from most synthesizers and other instruments. By using values above 0% you make the plugin produce higher velocities in average and vice versa. You can also arbitrarily transform the velocities using the **Custom velocity transform**.
Range: -100.0% to +100.0%, default 0.00%

Mixer panel

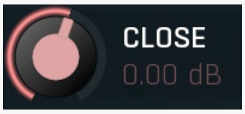


Mixer panel contains the integrated mixer useful for multisampled drums. Most parameters have effect only of the multisample supports it. Only the **Close** parameter always makes a difference. When you are using a simple Sampler for example, only this parameter is useful and controls the sample volume.



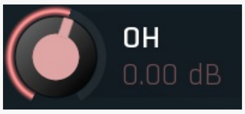
Play

Play button plays an event and may get handy when searching for the right sound.



Close

Close controls the volume of the close mics and is only useful for multisampled drums featuring multiple microphones. While this can be controlled directly from the multisampler as well, workflow-wise it is more accessible here.
Range: silence to 10.0 dB, default 0.00 dB

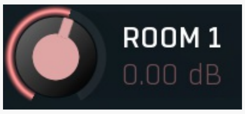


OH

OH controls the volume of the overhead mics and is only useful for multisampled drums featuring multiple microphones. While this can be controlled directly from the multisampler as well, workflow-wise it is more accessible here.
Range: silence to 10.0 dB, default 0.00 dB

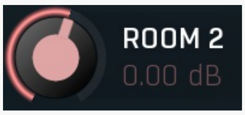
Ratio

Ratio lets you adjust ratio between 2 mics the drum may have available. This only works with certain multisampled drums. In other cases the control has no effect.
Range: -100.0% to +100.0%, default 0.00%



Room 1

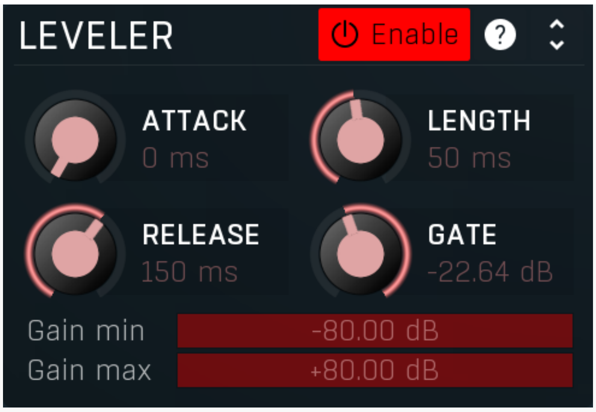
Room 1 controls the volume of the first set of room mics and is only useful for multisampled drums featuring multiple microphones. While this can be controlled directly from the multisampler as well, workflow-wise it is more accessible here.
Range: silence to 10.0 dB, default 0.00 dB



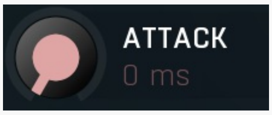
Room 2

Room 2 controls the volume of the first set of room mics and is only useful for multisampled drums featuring multiple microphones. While this can be controlled directly from the multisampler as well, workflow-wise it is more accessible here.
Range: silence to 10.0 dB, default 0.00 dB

Leveler panel



Leveler panel contains the integrated drum leveler processing the input signal.When the detector find an event and computes its correct level, the processor needs to apply the requested gain to the event. To do that it starts an envelope, which initially increases the gain during the **Attack** stage, then keeps the gain at the target value for time of the event **Length** and then decreases it back to 0dB change in the **Release** stage. This ensures the processing sounds natural, smooth and fixes the performance. You may change these parameters to adjust it to your audio material or even for some creative processing.



Attack

Attack controls the length of the attack stage. You rarely need to touch this parameter, because the processor usually detects the beginning of each event very precisely, so that the transition is still smooth. If you hear clicks at the beginning of the events, the detector is not working properly (which often occurs with too low look-ahead) and you may need to adjust the attack.
Range: 0 ms to 50 ms, default 0 ms

Length

Length controls the length of the event during which the maximum (or minimum) gain is applied. This should cover the body of the drum hit, which can be anything from 2ms with hihats to 400ms with toms. You may want to lower it provide a sort of gate effect and remove the unwanted ambience, leakage from other instruments etc.
Range: 0 ms to 1000 ms, default 50 ms

Release

Release controls the length of the release stage during which the gain is coming back to 0dB change. This works hand in hand with the **Length** parameter and has a similar effect. Note however that if you set this too low, an abrupt end may occur after each hit, which would sound as a click. It may be used creatively of course.
Range: 0 ms to 1000 ms, default 150 ms

Gate

Gate controls the level between the events. This gives you an opportunity to create an extremely accurate event-based gate useful mainly to remove leakage from other instruments and to shape the actual drum sound.
Range: -40.00 dB to 0.00 dB, default 0.00 dB

Gain min

Gain min controls the minimum gain that the plugin can perform on each event. You can use it to limit the event processing. For example, if you don't want the plugin to attenuate the events more than -20dB, then this would be your target value.
Range: -80.00 dB to 0.00 dB, default -80.00 dB

Gain max

Gain max controls the maximum gain the plugin can perform on each event. You can use it to limit the even processing. For example, if you don't want the plugin to amplify the events more than +20dB, then this is your target value.
Range: 0.00 dB to +80.00 dB, default +80.00 dB

Tabs

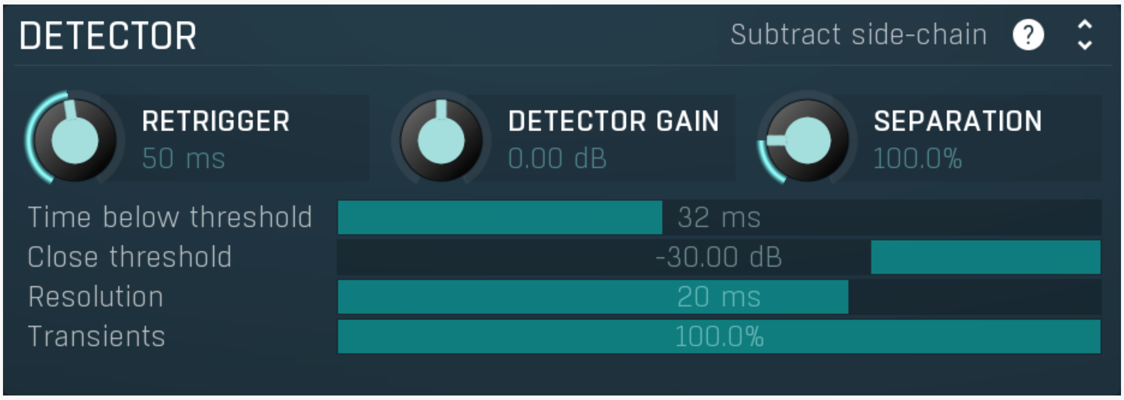
Tabs contain the heart of the plugin. There are 3 tabs:

Detection controls the actual detector and contains only an analyzer displaying the input signal levels with detected hits. You will mainly want to use the **Threshold** control and if you intend to use the **Leveling** feature, you will also need the **Velocity** controls.

Drum library contains a file browser for predefined drum components available from MDrummer libraries. Simply click on a sound you want and the plugin will load it and use it immediately.

Drum editor lets you tweak the details of the drum sound you loaded in the **Drum library**. It is nearly identical to MDrummer's Drumset editor, except it allows only a single drum.

Detector panel



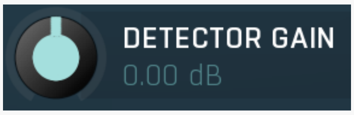
Detector panel contains some advanced detector parameters controlling how the events are detected.

Subtract side-chain

Subtract side-chain switch lets you solve difficult situation involving leakage from other microphones. If you are, for example, replacing a snare drum with too much leakage from bass drum and using the prefiltering doesn't help, you may solve the situation by sending the actual bass drum to the side-chain input and activating this feature. The plugin will then subtract the levels of the side-chain (bass drum) from the actual input levels, so when the bass drum hit occurs, the side-chain will cancel it from the main input and the hit won't be detected.

Retrigger

Retrigger defines the minimum time between 2 successive events. You may want to increase it if the detector finds more events in a row then there actually are. Many drummers for example play in too stiff a way, pressing the sticks against the drum head, which often makes the stick jump away and hit the head again. This creates 2 hits close to each other and to avoid processing both of them you may need to adjust the retrigger time.
Range: 1.0 ms to 1000 ms, default 50 ms



Detector gain

Detector gain defines the power modification applied to the detector signal, displayed as a black graph. You can use it if the input signal is too high or low, for example when pre-filtering with high resonance caused the level to drop too much. It would be too inconvenient to use the global **Input gain** for it, so this is an easier method which affects only the detector signal, the actual audio is not changed at all.
Range: -40.00 dB to +40.00 dB, default 0.00 dB

Separation

Separation controls how much the main level affects the detected transients. With 0% the transients won't be affected by level at all, which is rarely useful, since it detects transients regardless of the level. The higher the value, the more the actual level will be relevant making the transients sort of separated. Increase this value if there are lots of ghost notes leaking from other drums you want to avoid.
Range: 0.00% to 500.0%, default 100.0%

Time below threshold

Time below threshold controls the time the level needs to be below the threshold in order to trigger another even. This serves as a protection from producing multiple notes from a single event containing a short gap.
Range: 0 ms to 1000 ms, default 0 ms

Close threshold

Close threshold controls the threshold closing the detector relative to the main threshold, which starts a new event. This serves as a protection from producing multiple notes from a single event containing a short gap.
Range: -100.00 dB to 0.00 dB, default -30.00 dB

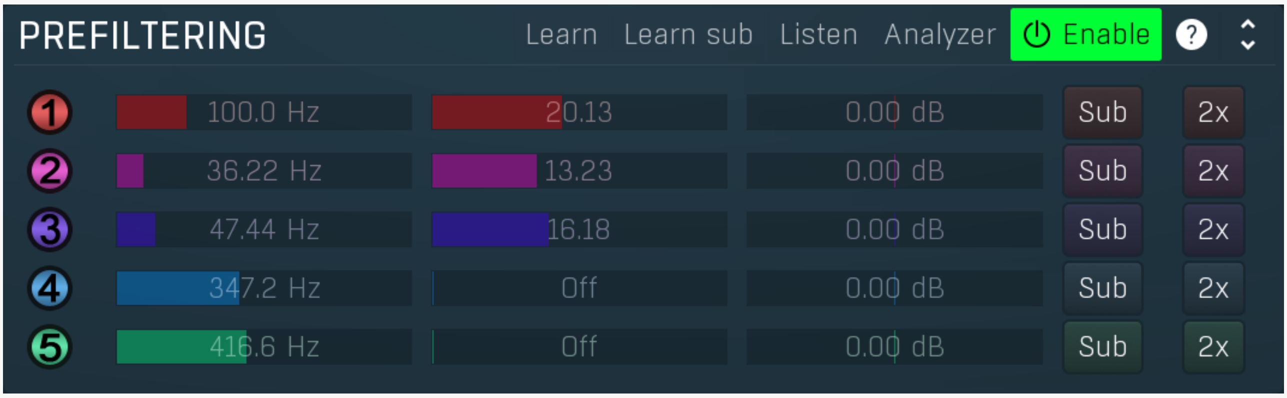
Resolution

Resolution controls the internal detection window time. Change this parameter only if you have a problem adjusting the detector for your audio material.
Range: 0 ms to 100 ms, default 20 ms

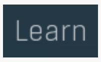
Transients

Transients controls how much the detector uses the information about transients in the signal. In most cases you will leave the default value of 100%, which makes the detector fully employ the transient detection. However in certain situations with lots of leakage you may need to lower this value. A typical example is processing a tom microphone with lots of snare drum leakage. While snare drums have very sharp and strong initial transient, toms usually don't and so with the default 100% value the detector tends to produce higher snare levels than tom levels even though it's not a snare microphone. In that case you may want to lower the value and the detector starts to prefer longer transient hits such as toms.
Range: 0.00% to 100.0%, default 100.0%

Prefiltering panel

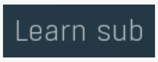


Prefiltering panel contains some filtering parameters that let you preprocess the detection signal. There are mainly several parallel band-pass filters, called resonators, which let you specifically target resonant frequencies in the signal and may come handy if you for example process a signal with lots of leakage, e.g. a hihat microphone which has recorded lots of leakage from the snare drum. Each resonator basically says 'listen to this frequency'. You can combine multiple resonators and you can even subtract resonators from the rest of them, which gets handy again for signals with lots of leakage - you can target the frequencies to listen to and also target frequencies NOT to listen to.



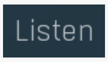
Learn

Learn switch activates learning mode, which analyzes the current signal, identifies the most prominent frequencies and sets the resonators accordingly. To use it, loop a part of the performance with the drum playing solo, or at least being prominent. Single hit could work as well, but it's better to have a more complex performance. Then enable Learn, keep it running for at least say 5 seconds, and press it again. After disabling it, the resonators will be changed. You can also view the actual analysis by clicking **Analyzer** button.



Learn sub

Learn sub switch activates learning mode, which analyzes the current signal, identifies the most prominent frequencies and sets the last 2 resonators to SUBTRACT these frequencies. It works the same way as **Learn**, but instead of making the detector focus on the sound being learned, it lets it ignore it. Use it, if there's a leakage from another instrument too high in level, which triggers notes as well. For example, if you have a hihat too prominent in the snare microphone and its frequencies are unfortunate enough, so that learning the snare resonances doesn't help, you can use this to 'unfocus' the problematic hihat sound.



Listen

Listen lets you audition just the filtered signal, so that you can adjust the filters easily. The entire metering system will still work as normal at that moment.

BP Enable

BP Enable enables or disables a resonator.

BP Frequency

BP Frequency defines the center frequency of the band-pass filter (resonator) used to preprocess the detector signal. This can be used to tune the detector to a specific drum frequency. Please note that the filters are run in parallel, so you can target multiple frequencies.
Range: 20.00 Hz to 20.0 kHz, default 100.0 Hz

BP Q

BP Q controls the resonance of the resonator. The higher the value, the more resonant the filter is and the more tuned it is to the center frequency. Set this to minimum to disable the filter. Please note that if the resonance is too high, the detector signal will start

losing the initial transient so you may want to check the **Transients** parameter.
Range: Off to 100.00, default Off

BP Gain
BP Gain controls the gain of the resonator output. Use this to control how much relevant each frequency is. Increasing this makes the bandpass signal higher in level, hence more important, and vice versa.
Range: -40.00 dB to +40.00 dB, default 0.00 dB

BP Subtract
BP Subtract switch makes the resonance be subtracted from the combined output from other resonators. This basically means 'this is a frequency the detector should not listen to'.

BP 2x
BP 2x switch increases the resonant filter order, which makes it targetting the specific frequency even more.

BP Enable
BP Enable enables or disables a resonator.

BP Frequency
BP Frequency defines the center frequency of the band-pass filter (resonator) used to preprocess the detector signal. This can be used to tune the detector to a specific drum frequency. Please note that the filters are run in parallel, so you can target multiple frequencies.
Range: 20.00 Hz to 20.0 kHz, default 200.0 Hz

BP Q
BP Q controls the resonance of the resonator. The higher the value, the more resonant the filter is and the more tuned it is to the center frequency. Set this to minimum to disable the filter. Please note that if the resonance is too high, the detector signal will start losing the initial transient so you may want to check the **Transients** parameter.
Range: Off to 100.00, default Off

BP Gain
BP Gain controls the gain of the resonator output. Use this to control how much relevant each frequency is. Increasing this makes the bandpass signal higher in level, hence more important, and vice versa.
Range: -40.00 dB to +40.00 dB, default 0.00 dB

BP Subtract
BP Subtract switch makes the resonance be subtracted from the combined output from other resonators. This basically means 'this is a frequency the detector should not listen to'.

BP 2x
BP 2x switch increases the resonant filter order, which makes it targetting the specific frequency even more.

BP Enable
BP Enable enables or disables a resonator.

BP Frequency
BP Frequency defines the center frequency of the band-pass filter (resonator) used to preprocess the detector signal. This can be used to tune the detector to a specific drum frequency. Please note that the filters are run in parallel, so you can target multiple frequencies.
Range: 20.00 Hz to 20.0 kHz, default 400.0 Hz

BP Q
BP Q controls the resonance of the resonator. The higher the value, the more resonant the filter is and the more tuned it is to the center frequency. Set this to minimum to disable the filter. Please note that if the resonance is too high, the detector signal will start losing the initial transient so you may want to check the **Transients** parameter.
Range: Off to 100.00, default Off

BP Gain
BP Gain controls the gain of the resonator output. Use this to control how much relevant each frequency is. Increasing this makes the bandpass signal higher in level, hence more important, and vice versa.
Range: -40.00 dB to +40.00 dB, default 0.00 dB

BP Subtract
BP Subtract switch makes the resonance be subtracted from the combined output from other resonators. This basically means 'this is a frequency the detector should not listen to'.

BP 2x
BP 2x switch increases the resonant filter order, which makes it targetting the specific frequency even more.

BP Enable
BP Enable enables or disables a resonator.

BP Frequency
BP Frequency defines the center frequency of the band-pass filter (resonator) used to preprocess the detector signal. This can be used to tune the detector to a specific drum frequency. Please note that the filters are run in parallel, so you can target multiple frequencies.
Range: 20.00 Hz to 20.0 kHz, default 800.0 Hz

BP Q
BP Q controls the resonance of the resonator. The higher the value, the more resonant the filter is and the more tuned it is to the center frequency. Set this to minimum to disable the filter. Please note that if the resonance is too high, the detector signal will start losing the initial transient so you may want to check the **Transients** parameter.
Range: Off to 100.00, default Off

BP Gain
BP Gain controls the gain of the resonator output. Use this to control how much relevant each frequency is. Increasing this makes the bandpass signal higher in level, hence more important, and vice versa.
Range: -40.00 dB to +40.00 dB, default 0.00 dB

BP Subtract
BP Subtract switch makes the resonance be subtracted from the combined output from other resonators. This basically means 'this is a frequency the detector should not listen to'.

BP 2x
BP 2x switch increases the resonant filter order, which makes it targetting the specific frequency even more.

BP Enable
BP Enable enables or disables a resonator.

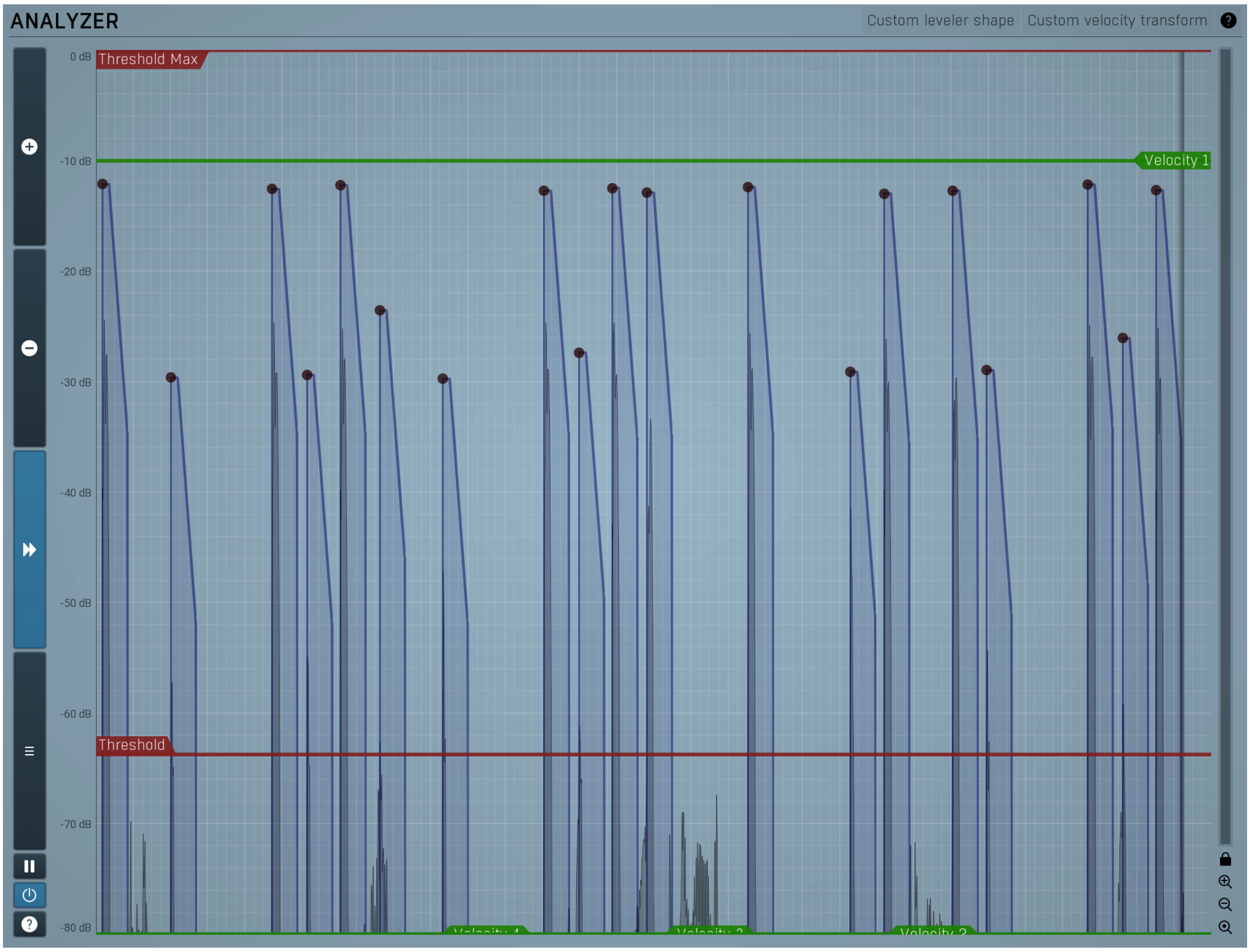
BP Frequency
BP Frequency defines the center frequency of the band-pass filter (resonator) used to preprocess the detector signal. This can be used to tune the detector to a specific drum frequency. Please note that the filters are run in parallel, so you can target multiple frequencies.
Range: 20.00 Hz to 20.0 kHz, default 1600 Hz

BP Q
BP Q controls the resonance of the resonator. The higher the value, the more resonant the filter is and the more tuned it is to the center frequency. Set this to minimum to disable the filter. Please note that if the resonance is too high, the detector signal will start losing the initial transient so you may want to check the **Transients** parameter.
Range: Off to 100.00, default Off

BP Gain
BP Gain controls the gain of the resonator output. Use this to control how much relevant each frequency is. Increasing this makes the bandpass signal higher in level, hence more important, and vice versa.
Range: -40.00 dB to +40.00 dB, default 0.00 dB

BP Subtract
BP Subtract switch makes the resonance be subtracted from the combined output from other resonators. This basically means 'this is a frequency the detector should not listen to'.

BP 2x
BP 2x switch increases the resonant filter order, which makes it targetting the specific frequency even more.



Analyzer

Analyzer displays a continual analysis of the input signal and lets you control the most important parameters easily.

Please note that this description assumes that the default colour palette is used for the time-graph (click the menu button on the left to change the colours) and the graph background is light, the colours may vary for different styles too. The **black graph** displays the **Input detector** level. It will most likely like look like a sequence of hills and valleys. Each hill represents a potential event which may be processed and tags a transient in the input signal including its detected level. Please note that these detector levels are NOT the same as levels of the events themselves (shown as red dots) - while an event level is basically its loudness, the detector signal is pre-processed in a specific way that makes it easy to recognize individual events and set the threshold properly. As such this graph is highly dependent on the detector settings, which you will usually keep at default, but if the graph looks strangely unpredictable, then you might need to adjust the detector settings or input pre-filtering.

There are also coloured dots. The **red dots** represent the detected events above the threshold together with their input levels. For every red dot, there is a corresponding **black dot** placed at the event's output level. In other words, red dots are the input events, black dots are the output events. The input and output levels for an event will differ only if the **Levelling** parameter is turned up. The vertical position of each dot defines the event level (more or less representing the velocity of the hit).

The red **Threshold guideline** controls the input detection threshold, the minimum level of a 'hill' which is to be considered as an event. You should set it so that the hills of events that you want to detect are above it and the hills of events that you want to ignore are below it. In most cases the detector pre-processes the levels (the black graph) extremely well and there will be a significant gap between the real events and the background rumble and leakage from other instruments, so all you need to do is to set the threshold slightly below the level of the smallest (quietest) wanted event.

The red **Threshold Max** guideline controls the maximum input detection threshold, the maximum level of each event. Events with levels above this threshold are ignored. It can be useful to process events only falling within a certain level range, e.g. only ghost notes, without affecting the loud hits. The Threshold Max is disabled when set to the maximum level (0dB).

There are 4 green **Velocity markers** relevant for the **Leveling** feature, which controls the desired output levels. They are essentially exactly what the name implies. They mark target velocities. For example, in most pop and heavy music there are no 'ghost' notes on the snare drum and more or less everything sounds accented. So there is just one velocity level and the plugin will try to fix the drummer's performance and assume he wanted to play at the same velocity every time. If you assume the drummer plays more dynamically however, you may use multiple velocity markers; say one for silent ghost notes, one for normal hits and one for ultra-high level accents.

If a marker is at the bottom of the analyzer (at -80dB), it is disabled and ignored. If there is no active marker, the leveler does nothing. Let's now assume Leveling is 100% (i.e. maximum processing), which will probably result in quite a robotic performance. For each input event (the red dots) the plugin selects a velocity marker, which defines the desired output level for that event (the black dots). So there will be one black dot for each red dot and each of these black dots will lie on a velocity marker. The lower the Leveling is, the further away the black dots will be from the velocity markers and the closer to the input level.

This is how it works in detail: When an event is started, the plugin first chooses which Velocity marker this belongs to. If there is just one, then the choice is obvious. If there is more than one, then the marker selected is the one closest above the event, or just the highest one if there are no more markers above he event. Then the plugin selects the velocity for the event, which is based on the level of the marker, factored using the Leveling parameter. If there are no markers enabled or Leveling feature is off, the plugin will ignore them and follow the input levels precisely.

The analyzer can also display input, output and sidechain waveforms, which are disabled by default. You can enable them in the time-graph settings (the menu button on the left).

Custom leveler shape

Custom leveler shape button displays a custom shape editor, which lets you control the levels of each event, which in effect define the velocities. The graph controls the mapping of input event level into its output level. Normally it is generated automatically from Velocity markers in the analyzer, but you may want to specify it manually for special effects. The X axis is the input level that the detector computes for each input event (the red dots). The Y axis is the desired output level (the black dots). So for example a straight line from left bottom [-80dB, -80dB] to right top [0dB, 0dB] means no processing at all, because whatever the input event level is, the desired output level is the same. Please note that the **Leveling** parameter controls the amount of this processing, the same way as when using the velocity markers normally.

Custom velocity transform

Custom velocity transform button displays a custom graph editor which lets you arbitrarily transform the output velocities for the created notes. The X axis shows the input velocities as detected by the plugin; and the Y axis defines the output velocities at which the synthesized notes will be playing.

Plus

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

Minus

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

Rewind

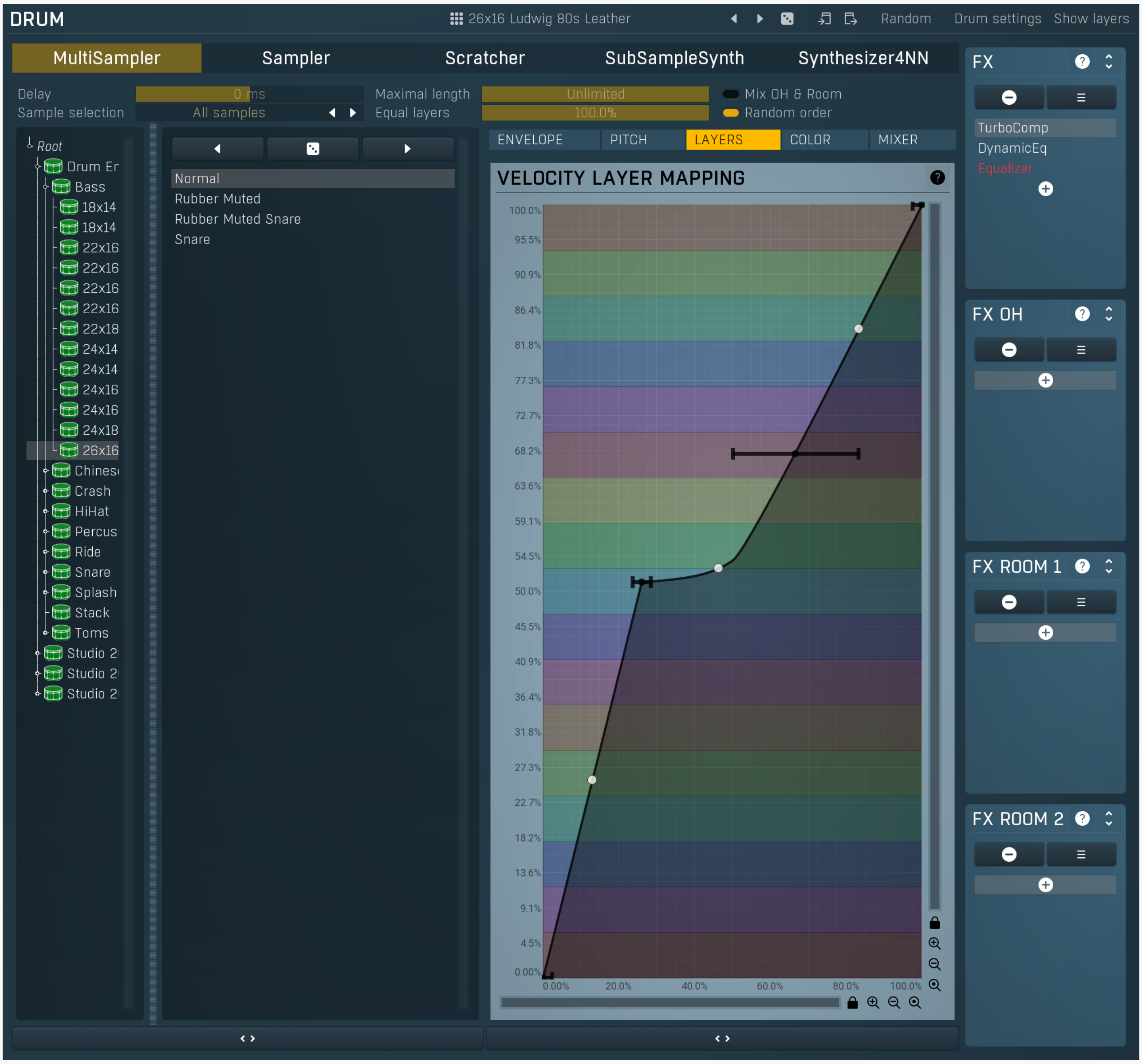
Rewind button enables or disables the time-graph static mode. In static mode the graphs are fixed and the current position cycles from left to right; otherwise the graphs move from right to left and the current position is fixed (at the right-hand side).

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

Pause
Pause button pauses the processing.

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

Drum



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.

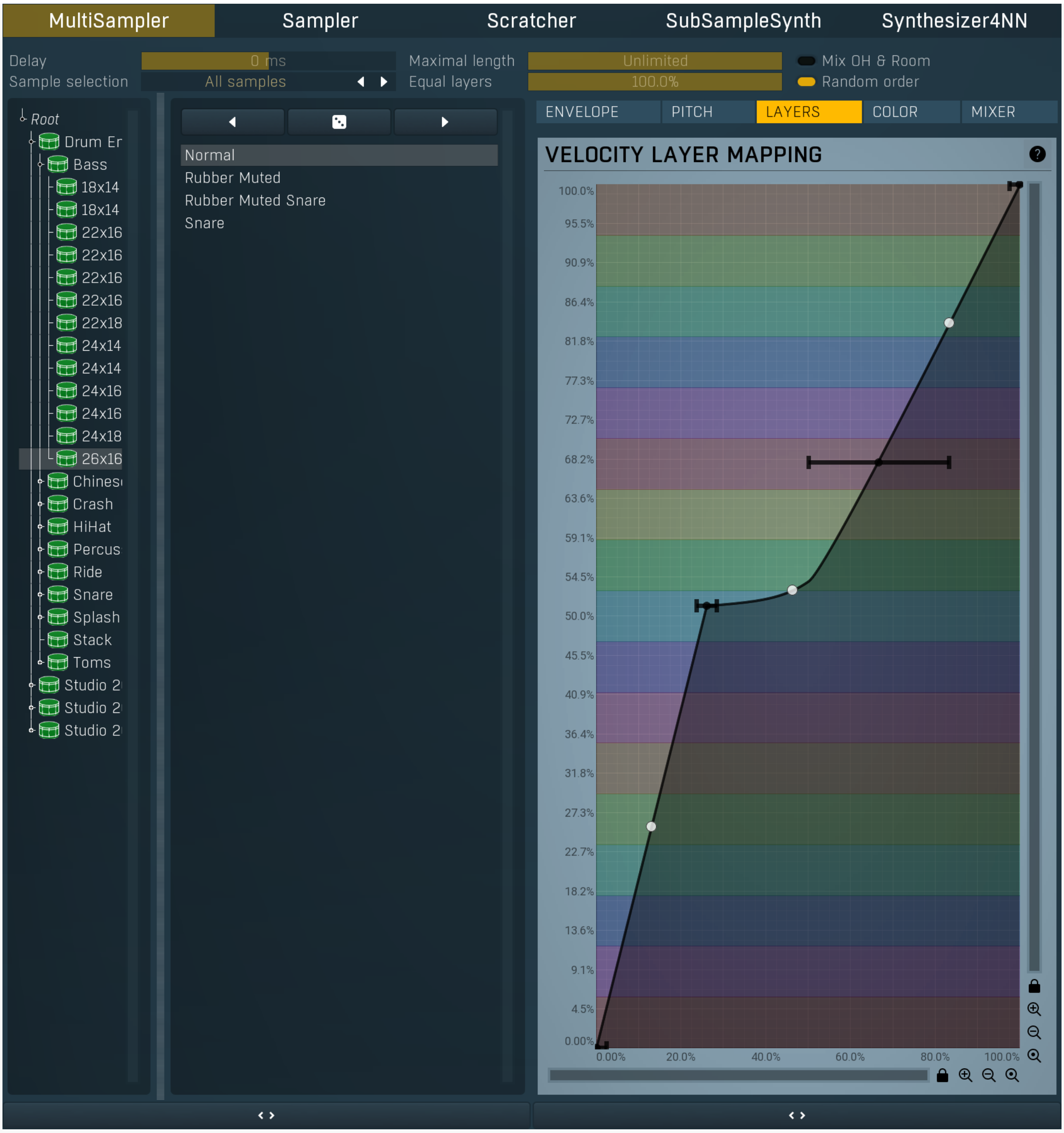
Copy
Copy button copies the settings onto the system clipboard.

Paste
Paste button loads the settings from the system clipboard.

Random
Random button generates random settings using the existing presets.

Drum settings
Drum settings button displays some global settings of the drum.

Show layers
Show layers button displays the velocity layers in the drum editor. Since it is rarely useful, it is disabled by default.



Layer source panel

Layer source panel allows you to set up the drum sound source sample and its parameters for the selected layer.

Copy

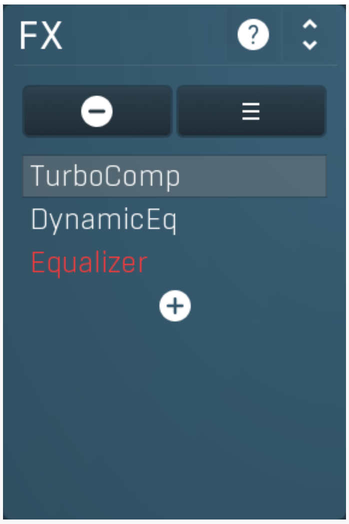
Copy button copies the selected velocity layer onto the system clipboard.

Paste

Paste button pastes the selected velocity layer settings from the system clipboard.

Source sample selector

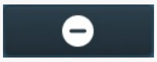
Source sample selector contains the list of installed sound source samples. The currently-loaded one is highlighted. Click on another one to change the sound source for the current layer of this drum. Click using right mouse button to get a context menu containing all of the plugins.



FX list

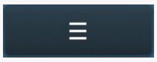
FX list contains the list of effects in the pipeline.
Double click on an effect to display its editor.
Click the plus sign to add a new effect.
Hold alt and click on an effect to delete the effect from the pipeline.
Check/uncheck an effect to toggle the effect's bypass state.
Click and drag an effect up / down to move it, hence changing the order of effect processing.
Click and drag an effect left / right to copy effect with its complete settings to other effect pipeline.
Hold shift and drag an effect left / right to move the effect with its complete settings to other effect pipeline.

FX panel contains the drum effect pipeline, which lets you manipulate the list of effects processing the drum signal. You can have as many of them as you want, but note that each effect consumes some CPU and memory. The plugin generates an audio stream from the drum layers, similar to mixing a single drum using one or more microphones. Then it processes this stream using this effect pipeline. The output of the effect pipeline is then sent to all available sends and mixed with the output.



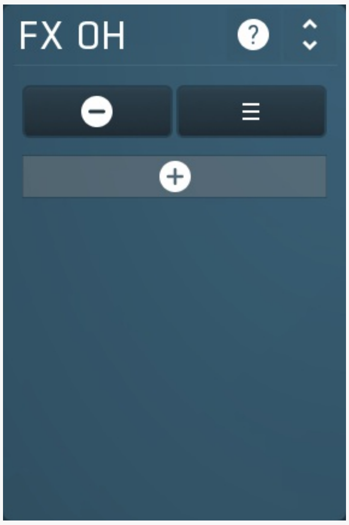
Delete

Delete button deletes the selected effect from the effect pipeline.



Menu

Menu button displays additional functions to randomize/load/save effects etc.



FX list

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Double click on an effect to display its editor.
Click the plus sign to add a new effect.
Hold alt and click on an effect to delete the effect from the pipeline.
Check/uncheck an effect to toggle the effect's bypass state.
Click and drag an effect up / down to move it, hence changing the order of effect processing.
Click and drag an effect left / right to copy effect with its complete settings to other effect pipeline.
Hold shift and drag an effect left / right to move the effect with its complete settings to other effect pipeline.

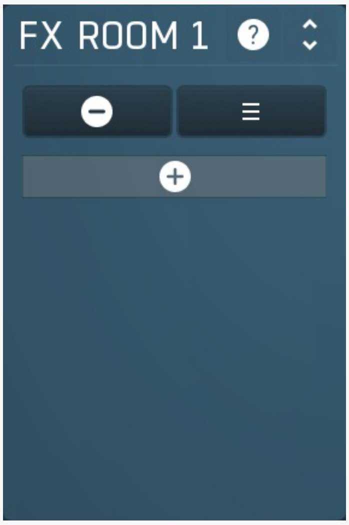
FX OH panel contains the effect pipeline for overheads (if available). Having such a thing is impossible when mixing drums normally, since it requires dedicated signal from overhead microphones containing the individual drums only, but The plugin mixes all drums separately, so this is indeed available. You can use it for example to produce a big snare sound by compressing the overheads, but leaving the rest of the kit intact.

Delete

Delete button deletes the selected effect from the effect pipeline.

Menu

Menu button displays additional functions to randomize/load/save effects etc.



FX list

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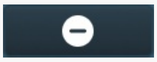
Double click on an effect to display its editor.
Click the plus sign to add a new effect.
Hold alt and click on an effect to delete the effect from the pipeline.
Check/uncheck an effect to toggle the effect's bypass state.
Click and drag an effect up / down to move it, hence changing the order of effect processing.
Click and drag an effect left / right to copy effect with its complete settings to other effect pipeline.
Hold shift and drag an effect left / right to move the effect with its complete settings to other effect pipeline.

FX Room 1 panel contains the effect pipeline for room microphones (if available). Having such a thing is impossible when mixing drums normally, since it requires dedicated signal from room microphones containing the individual drums only, but The plugin mixes all drums separately, so this is indeed available. You can use it for example to produce a big snare sound by compressing the room mics, but leaving the rest of the kit intact.

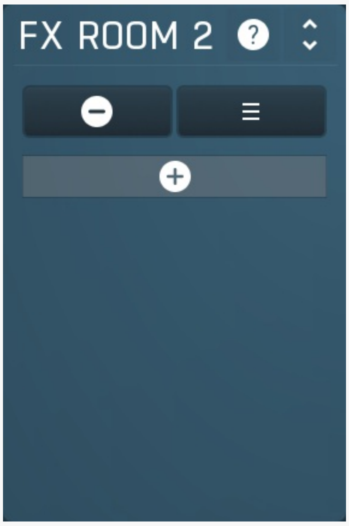


Delete

Delete button deletes the selected effect from the effect pipeline.

Menu

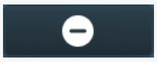
Menu button displays additional functions to randomize/load/save effects etc.



FX list

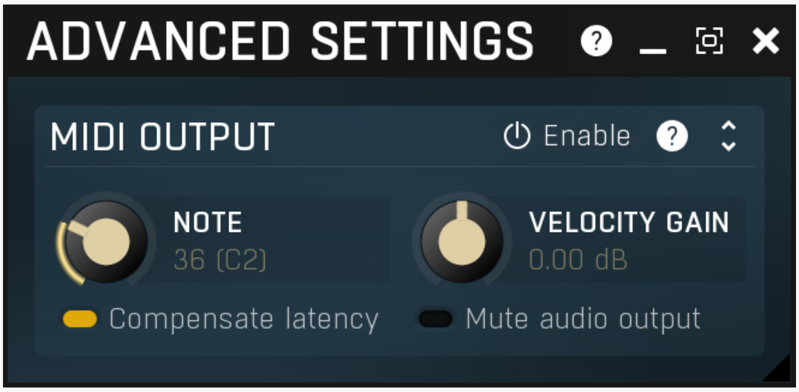
FX list contains the list of effects in the pipeline.
Double click on an effect to display its editor.
Click the plus sign to add a new effect.
Hold alt and click on an effect to delete the effect from the pipeline.
Check/uncheck an effect to toggle the effect's bypass state.
Click and drag an effect up / down to move it, hence changing the order of effect processing.
Click and drag an effect left / right to copy effect with its complete settings to other effect pipeline.
Hold shift and drag an effect left / right to move the effect with its complete settings to other effect pipeline.

FX Room 2 panel contains the effect pipeline for room microphones (if available). Having such a thing is impossible when mixing drums normally, since it requires dedicated signal from room microphones containing the individual drums only, but The plugin mixes all drums separately, so this is indeed available. You can use it for example to produce a big snare sound by compressing the room mics, but leaving the rest of the kit intact.


Delete
Delete button deletes the selected effect from the effect pipeline.

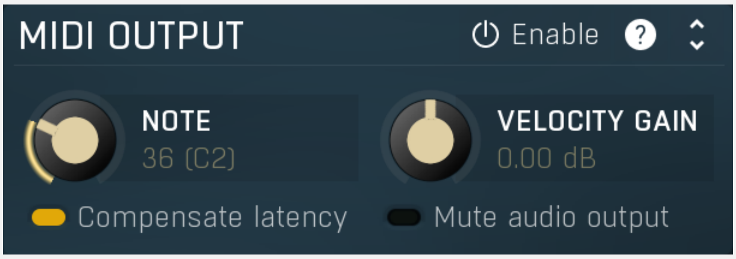

Menu
Menu button displays additional functions to randomize/load/save effects etc.

MDrumReplacerAdvanced

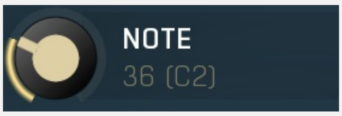


Advanced settings window contains more advanced settings, which are used less often and so are intentionally not shown on the main plugin editor.

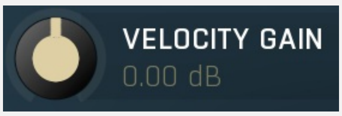
MIDI output panel



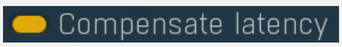
MIDI output panel contains parameters of the optional MIDI output.



Note
Note defines which note should be transmitted for every event. By default it is 'off', meaning that there are no notes produced by the plugin.
Range: 0 to 127, default 36



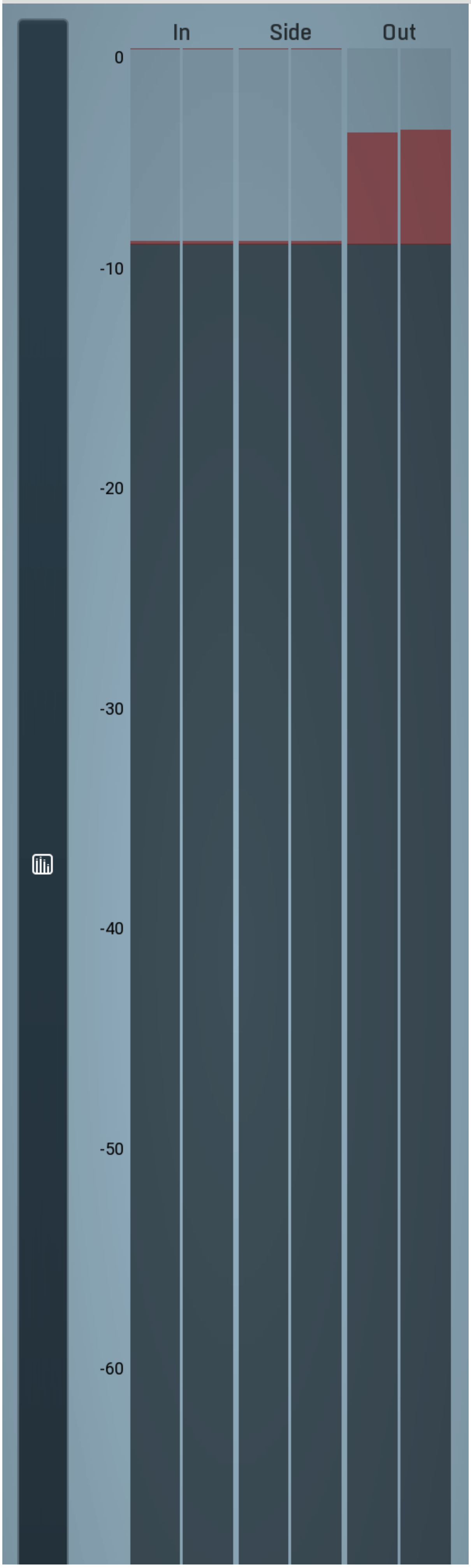
Velocity gain
Velocity gain lets you increase or decrease the level of the detected hits, which in effect transforms the output velocity.
Range: -40.00 dB to +40.00 dB, default 0.00 dB

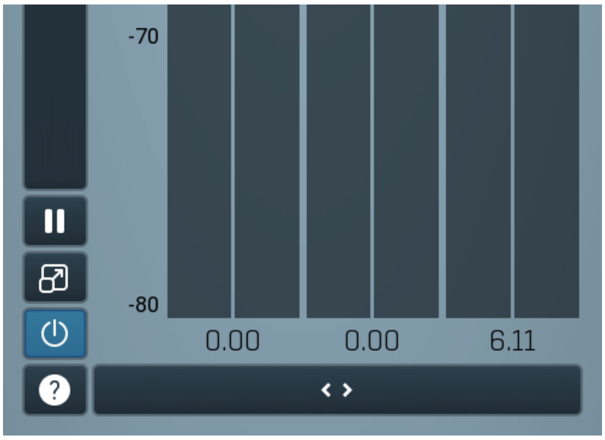


Compensate latency
Compensate latency enables latency compensation for notes generate into the MIDI output. It is provided, because some DAWs compensate latency themselves, but others do not.

Mute audio output
Mute audio output mutes the audio output when the MIDI output is enabled. It is useful in DAWs, where placing a plugin after another automatically sends MIDI from the first to the second, so you can easily put MDrumLeveler in front of a drum sampler for

example.





Global meter view

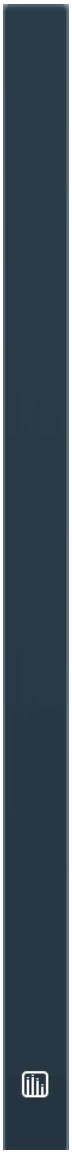
Global meter view provides a powerful metering system. If you do not see it in the plug-in, click the **Meters** or **Meters & Utilities** button to the right of the main controls. The display can work as either a classical level indicator or, in time graph mode, show one or more values in time. Use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

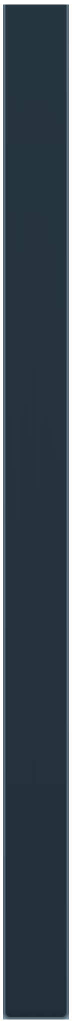
In meter indicates the total input level. The input meter shows the audio level before any specific processing (except potential oversampling and other pre-processing). It is always recommended to keep the input level under 0dB. You may need to adjust the previous processing plugins, track levels or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





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.

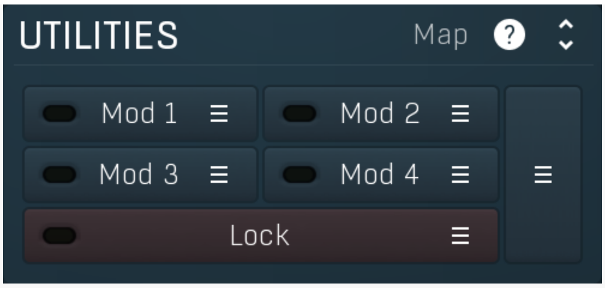
Pause
Pause button pauses the processing.

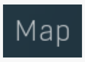
Popup
Popup button shows a pop-up window and moves the whole metering / time-graph system into it. This is especially useful in cases where you cannot enlarge the meters within the main window or such a task is too complicated. The pop-up window can be arbitrarily resized. In metering mode it is useful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

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.