MRotary

Easy screen vs. Edit screen

The plugin provides 2 user interfaces - an **easy screen** and an **edit screen**. Use the Edit button to switch between the two.

By default most plugins open on the **easy screen** (edit button released). This screen is a simplified view of the plugin which provides just a few controls. On the left hand side of the plugin you can see the list of available **devices / instruments** (previously called 'active presets'), that is, presets with controls. These controls are actually nothing more than multiparameters (single knobs that can control one or more of the plug-in's parameters and sometimes known as Macro controls in other plug-ins) and are described in more detail later. Each device may provide different controls and usually is intended for a specific purpose. The easy screen is designed for you to be able to perform common tasks, quickly and easily, without the need to use the advanced settings (that is, those available on the Edit screen).

In most cases the devices are highlighted using different text colors. In some cases the colors only mark different types of processing, but in most cases the general rule is that **black/white devices** are the essential ones designed for general use. **Green devices** are designed for a specific task or audio materials, e.g. de-essing or processing vocals in a compressor plugin. **Red devices** usually provide some very special processing or some extreme or creative settings. In a distortion plugin, for example, these may produce an extremely distorted output. **Blue devices** require an additional input, a side-chain or MIDI input usually. Without these additional inputs these **Blue** presets usually do not function as intended. Please check your host's documentation about routing side-chain and MIDI into an effect plugin.

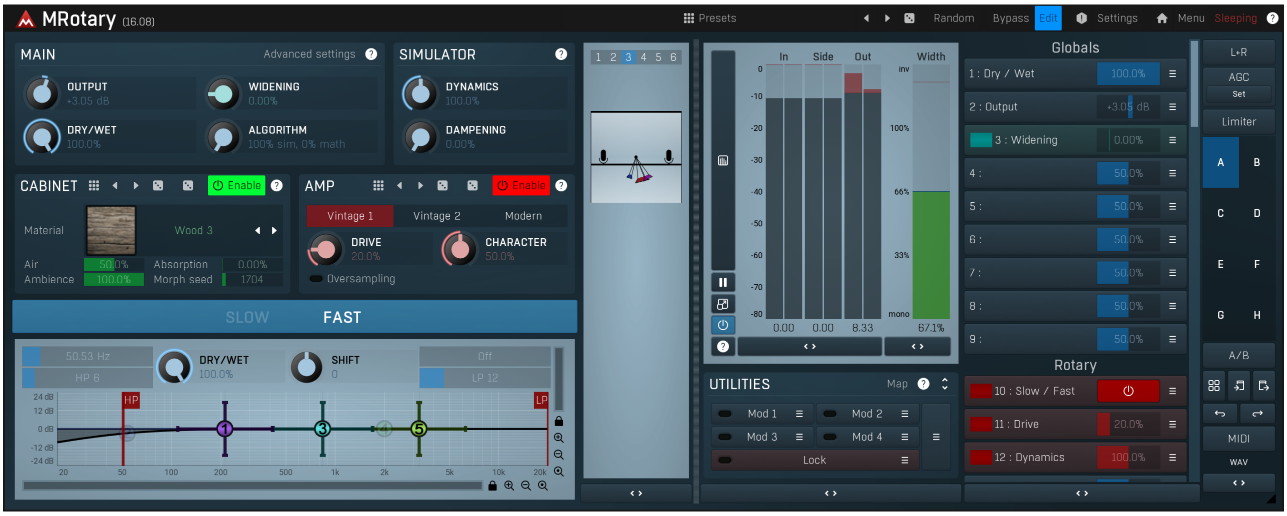
To the right of the controls are the meters or time-graphs for the plugin; the standard plugin Toolbar may be to the right of these or at the bottom of the plugin.

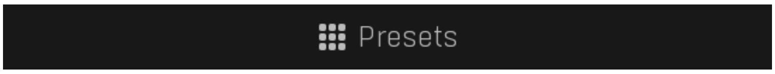
By clicking the **Edit button** you can switch the plugin to **edit mode** (edit button pushed). This mode provides all the of the features that the plugin offers. You lose no settings by toggling between edit mode and the easy screen unless you actually change something. This way you can easily check what is "under the hood" for each device, or start with an device and then tweak the plugin settings further.

Devices are factory specified and cannot be modified directly by users, however you can still make your own and store them as normal presets. To do so, configure the plugin as desired, then define each multiparameter and specify its name in its settings. You can then switch to the easy screen and check the user interface that you have created. Once you are satisfied with it, save it as a normal preset while you are on the easy screen. Although your preset will not be displayed or selected in the list of available devices, the functionality will be exactly the same. For more information about multiparameters and devices please check the **online video tutorials**.

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

Edit mode





Presets

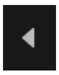
Presets button shows a window with all available presets. A preset can be loaded from the preset window by double-clicking on it, selecting via the buttons or by using your keyboard. You can also manage the directory structure, store new presets, replace existing ones etc. Presets are global, so a preset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

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

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

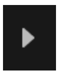
Files are named based on the name of the plugin like this: "{pluginname}.presets", so for example MAutopan.presets or MDynamics.presets. If the directory cannot be found on your computer for some reason, you can just search for the particular file.

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



Left arrow

Left arrow button loads the previous preset.

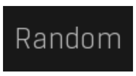


Right arrow

Right arrow button loads the next preset.

Randomize

Randomize button loads a random preset.



Randomize

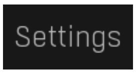
Randomize button (with the text 'Random') generates random settings. Generally, randomization in plug-ins works by selecting random values for all parameters, but rarely achieves satisfactory results, as the more parameters that change the more likely one will cause an unwanted effect. Our plugins employ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

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

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

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

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



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

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

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

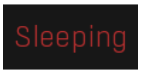
Advanced settings configures several processing options for the plug-in.

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

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

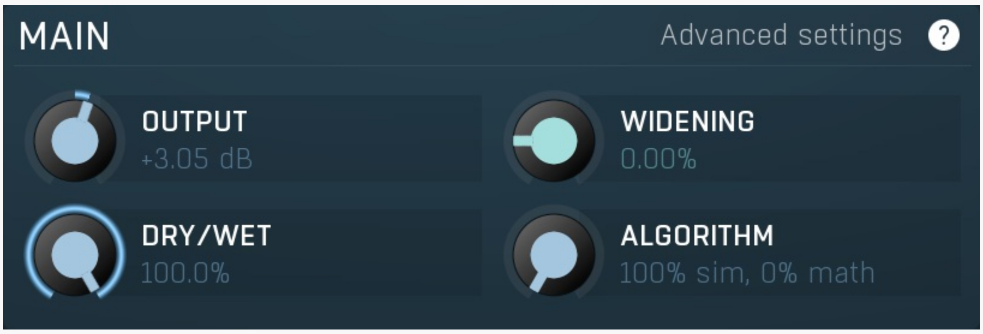
Smart interpolation adjusts the interpolation algorithm used when changing parameter values; the higher the setting the higher the audio quality and the lower the chance of zippering noise, but more CPU will be used.

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

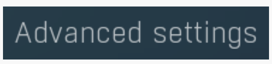


Sleep indicator
Sleep indicator informs whether the plugin is currently active or in sleep mode. The plugin can automatically switch itself off to save CPU, when there is no input signal and the plugin knows it cannot produce any signal on its own and it generally makes sense. You can disable this in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

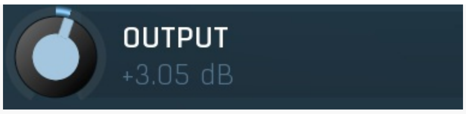
Main panel



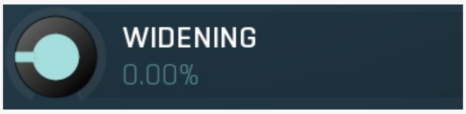
Main panel contains the most basic settings.



Advanced
Advanced button displays additional settings.



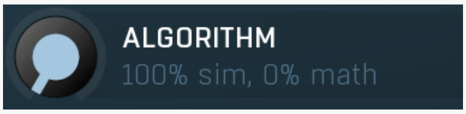
Output gain
Output gain defines the power modification applied to the output signal.
Range: -24.00 dB to +24.00 dB, default 0.00 dB



Widening
Widening defines the broad-band stereo field widening depth. The algorithm is fully mono-compatible as it only extends the existing stereo field and no new signal is added. This parameter should only be used to control the existing stereo field.

Widening converts the audio into its mid (mono) and side channels, leaving the mid intact and applying a gain to the side channel, then converts the signal back to left and right channels. As a result the stereo image becomes wider (for widening above 0%) or narrower (for widening below 0%). This method of widening the stereo image may initially sound pleasing, however it can quickly become fatiguing on the ear and often sounds unnatural, especially for larger amounts of widening. Use this parameter to control the existing stereo field and as a special effect. Use it to increase width only with caution.
Range: Mono to 400.0%, default 0.00%

Dry/Wet
Dry/Wet defines ratio between dry and wet signals. 100% means fully processed, 0% means no processing at all.
Range: 0.00% to 100.0%, default 100.0%

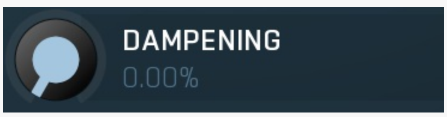


Algorithm
Algorithm controls the ratio between the 2 algorithms that the plugin provides.
Minimum value enables only the **simulator algorithm**, which closely resembles the vintage rotary cabinets. In this mode several parameters of the cabinet geometry do not make any difference. Only distance and width are relevant.
Maximum value enables only the **mathematical algorithm**, which is based on purely mathematical operations according to the physical properties of a virtual cabinet. All parameters of the cabinet geometry are relevant here.
Any value in-between enables both algorithms and blends between the two, hence it requires more CPU power. Please note that you can modify this value for each speaker of the virtual cabinet.
Range: 100% sim, 0% math to 0% sim, 100% math, default 0% sim, 100% math

Simulator panel

Simulator panel contains parameters of the **simulator algorithm**. The amount of signal provided by this algorithm is controlled by the **Algorithm** parameter.

Dynamics
Dynamics controls the amount of dynamic behaviour. With 100% the plugin closely mimics the real cabinet. Lower values diminish the dynamic behaviour, which usually sounds as if the natural tremolo is being removed.
Range: 0.00% to 200.0%, default 100.0%



Dampening

Dampening simulates filling the cabinet with an absorption material hence shortening the response. Please note that this value cannot be modified dynamically without artifacts, therefore it should not be modulated/automated.
Range: 0.00% to 100.0%, default 0.00%

Cabinet panel

Cabinet panel controls the virtual cabinet simulator. This powerful device emulates several phenomena which occur in natural real-world cabinets and let you simulate boxes built from different materials with different sound absorption qualities etc. This emulator can be completely disabled leading to more transparent results. The cabinet configuration is mostly applied in the **mathematical algorithm**, but it is relevant for the ambience in **simulator algorithm** as well.

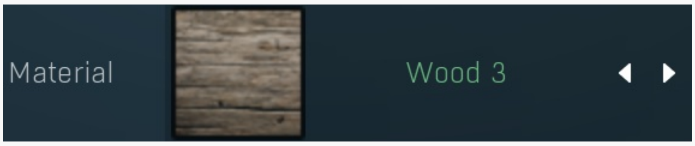
Presets
Presets button displays a window where you can load and manage available presets. Hold **Ctrl** when clicking to load a random preset instead.

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.

Random
Random button generates random settings using the existing presets.



Material
Material controls the material from which the virtual cabinet is made. Different materials have different spectrum and time qualities and produce different resonances. Notice the **Morph** parameter, which creates a cabinet from different materials and the structure is controlled by the **Morph seed** parameter.

Air
Air controls the amount of reflections in the cabinet. Essentially this is a form of varying reverberator. It also affects the ability of the cabinet to absorb or resonate with specific frequencies. The higher the value, the longer the reflections inside the cabinet will be and the more the cabinet can resonate. You will probably want to control this value according to the current mixing situation - while a higher value usually leads to fuller sound, it may interfere with other instruments in the mix. Also the shorter the response, the tighter the output usually is; which is typical for modern mixes.
Range: 0.00% to 100.0%, default 50.0%

Absorption
Absorption controls amount of the absorption of the material and is inspired by the acoustic insulation materials, which have kind of a spongy surface which lets them absorb acoustic energy better. Here however, the main effect is to support the rotation of the speakers. While the speaker cones move towards or away from the microphones, different amounts of energy are absorbed by the cabinet. By increasing the absorption, you essentially make this more apparent and the effect stronger. Please note that as with most effects this can lead to unnatural sound when overused.
Range: 0.00% to 100.0%, default 0.00%

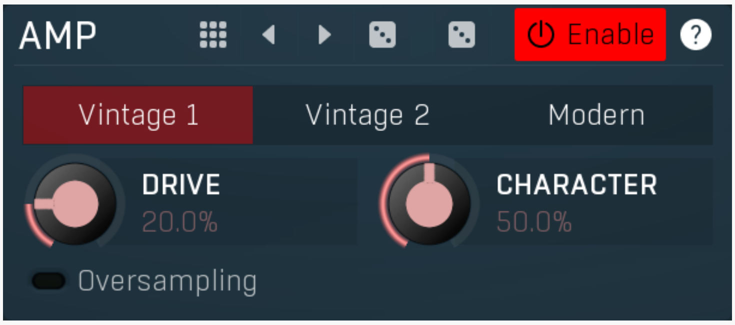
Ambience

Ambience controls additional ambience processing, which can glue the sound together and provide some sense of depth.
Range: 0.00% to 100.0%, default 100.0%

Morph seed

Morph seed controls the cabinet structure when **Morph** material is used. Essentially such a cabinet is created from several different materials and instead of letting you control each part of the cabinet manually, you use just this single parameter. By changing this value, the whole cabinet is changed. This is especially useful for creative purposes. However please note that automating or modulating this parameter is not advised.
Range: 0 to 32767, default 0

Amp panel



Amp panel controls the amp simulator providing a smooth distortion recognisable from the vintage rotary cabinets.

Presets

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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

Random button generates random settings using the existing presets.

Type

Type defines the sound character, which basically controls the amount and dirtiness of the distortion.

Drive

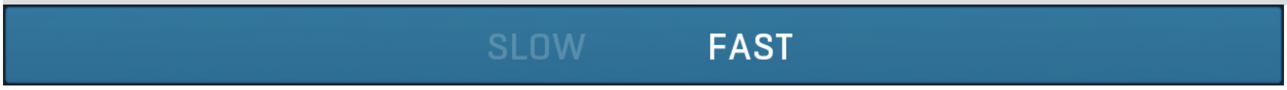
Drive controls the input gain resulting in the amount of distortion. Since the distortion may increase loudness in general, you can use the main plugin gain to compensate for it.
Range: 0.00% to 100.0%, default 20.0%

Character

Character controls the distribution of higher harmonics, hence, again, sound character.
Range: 0.00% to 100.0%, default 50.0%

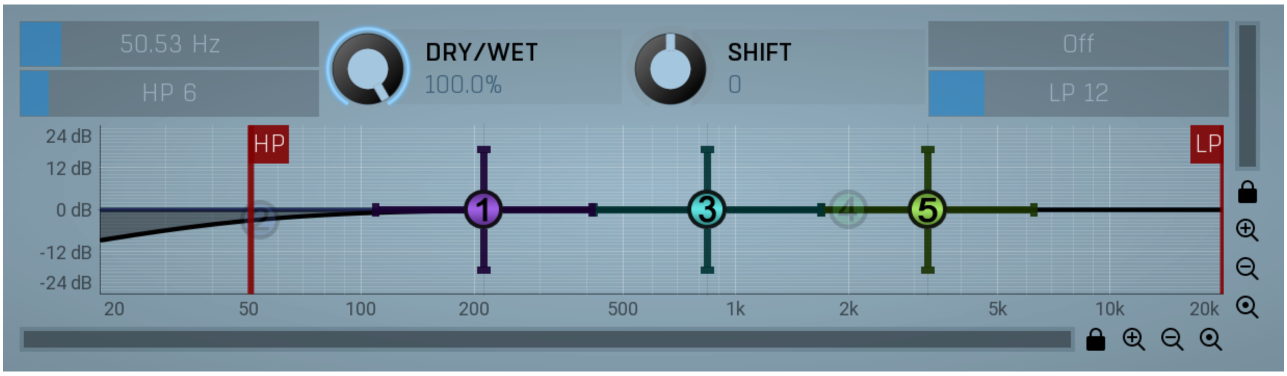
Oversampling

Oversampling activates the integrated oversampler, which performs the distortion in higher sampling rate and avoids aliasing, which may be especially noticeable for higher drive values. However note that this increases the CPU requirements extensively.



Slow/fast

Slow/fast switch switches between slow and fast modes. You can configure the speeds and acceleration times in the Advanced settings.



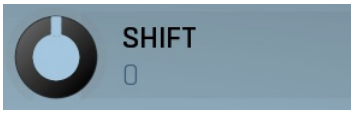
Equalizer shape graph

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

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

Dry/Wet

Dry/Wet defines ratio between dry and wet signals. 100% means fully processed, 0% means no processing at all. In normal mode only peak and shelf filters are affected correctly, other filters are left at 100% unless the ratio is set to 0%, in which case the equalizer is bypassed.
Range: 0.00% to 100.0%, default 100.0%

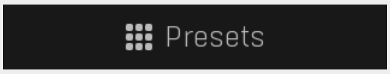


Shift

Shift lets you pitch shift all bands by specified number of semitones. It doesn't change the actual band points, but changes the resulting EQ shape appropriately.
Range: -24.00 to +24.00, default 0

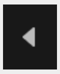
Band settings window

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



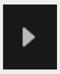
Presets

Presets button displays a window where you can load and manage available presets. Hold **Ctrl** when clicking to load a random preset instead.



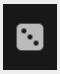
Left arrow

Left arrow button loads the previous preset.



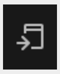
Right arrow

Right arrow button loads the next preset.



Randomize

Randomize button loads a random preset.

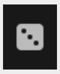


Copy

Copy button copies the settings onto the system clipboard.

Paste

Paste button loads the settings from the system clipboard.



Random

Random button generates random settings using the existing presets.

General panel

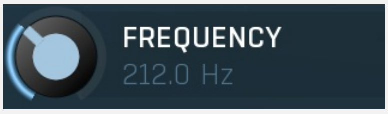
General panel contains standard filter settings such as frequency or Q. Most of these values are available directly from the band graph, but it may be necessary to use these controls for more accurate or textual access.

Invert gain

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

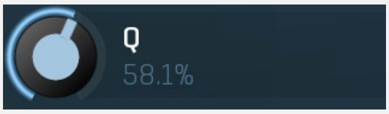
Swap gains

Swap gains button swaps values between gain and dynamics gain.



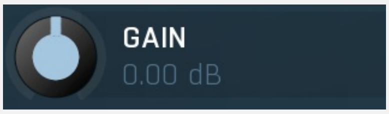
Frequency

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



Q

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



Gain

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

Slope

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

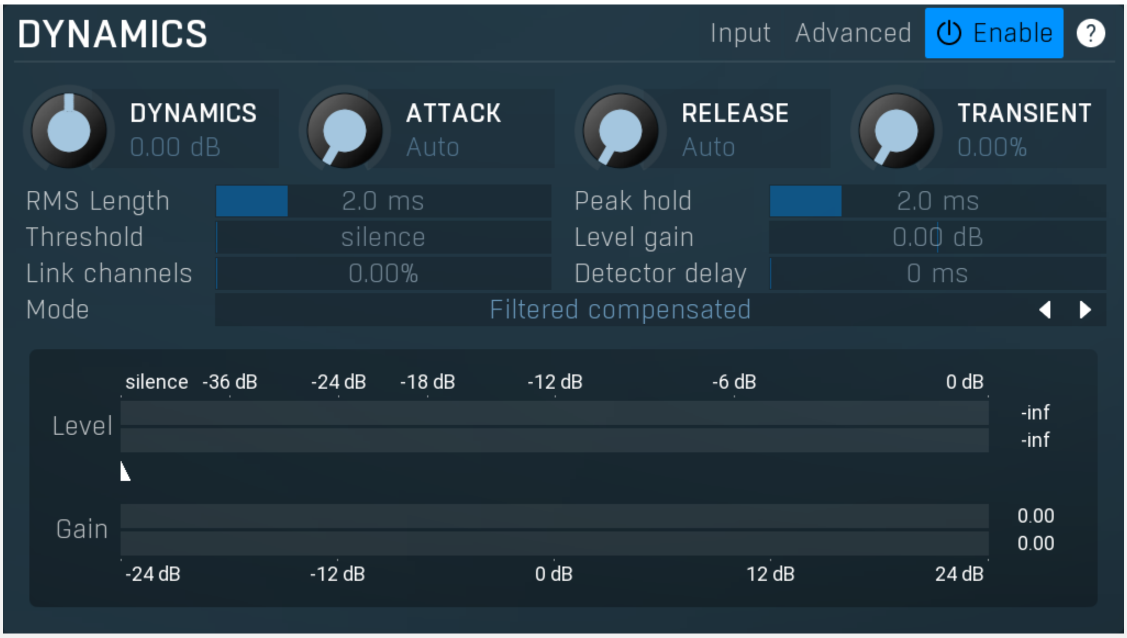
Channels

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

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

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

Dynamics panel



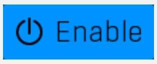
Dynamics panel contains settings of the dynamics processing which control how the filter behaves depending on input signal. Normal filters are static, meaning they don't change any features depending on the input signal. If you enable dynamic properties, by making the **dynamic gain** nonzero, the filter will start listening to the level of the input signal. This requires more CPU of course, as such a band is essentially an extremely complex generalized compressor, but the algorithms used are as efficient as it is technically possible.
A dynamic band varies the gain according to the input level. It can listen to the whole spectrum or to just part of it. By default it is driven by the partial spectrum, which it modifies itself, so, for example, when you have a high shelf, it is essentially listening to a high part of the spectrum. You can do many things with such a dynamic processor, but essentially it can work as a compressor or expander. There are many more advanced ideas that you can do and the full power hasn't really been explored yet.

Input

Input switch makes the band measure the input level instead of current level in the chain of bands. When this is disabled (default) and the equalizer is processing the bands serially, which means that each band is processing the output from the previous stage, including level measurement. If you enable this switch however, the dynamic processing will be driven by the original input signal instead.
Please note that when **Side-chain** is on, this switch has no meaning, since side-chain has priority.

Advanced

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

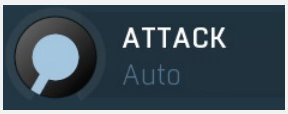


Enable

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

Dynamics

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



Attack

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

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

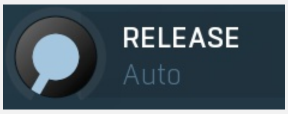
To understand the working of a level detector, it is best to cover the typical cases:

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

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

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

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



Release

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

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

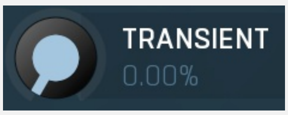
To understand the working of a level detector, it is best to cover the typical cases:

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

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

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

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



Transient

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

RMS length

RMS length smoothes out the values of the input levels (not the input itself), such that the level detector receives the pre-processed signal without so many fluctuations. When set to its minimum value the detector becomes a so-called "peak detector", otherwise it is an "RMS detector".

When you look at a typical waveform in any editor, you can see that the signal is constantly changing and contains various transient bursts and separate peaks. This is especially noticeable with rhythmical signals, such as drums. Trying to imagine how a typical attack/release detector works with such a wild signal may be complex, at least. RMS essentially takes the surrounding samples and averages them. The result is a much smoother signal with fewer individual peaks and short noise bursts.

RMS length controls how many samples are taken to calculate the average. It stabilizes the levels, but it also causes a slower response time. As such it is great for mastering, when you want to lower the dynamic range in a very subtle way without any instabilities. However, it is not really desirable for processing drums, for example, where the transient bursts may actually be individual drum hits, hence it is usually recommended to use peak detectors for percussive instruments.

Note that the RMS detector has 2 modes - a simplified approximation is used by default, and a true RMS is processor can be enabled from the advanced settings (if provided). Both respond differently, neither of them is better than the other, they are simply different.

Peak hold

Peak hold defines the time that signal level detector holds its maximum before the release stage is allowed to start. As an example, you can imagine that when an attack stage ends there can be an additional peak hold stage and the level is not yet falling, before the release stage starts. This is true only when **true peak** mode is enabled (check the advanced detector settings if available).

It is often used in **gates** to avoid the gated level falling below the threshold too quickly, while having short release times. If you want the gate to close quickly, you need a short release time. But in that case the ending may be too abrupt and even cause some distortion. So you use the peak hold to delay the release stage.

It is also used along with **look-ahead** to avoid distortion in **limiters and compressors**. If you need a very short attack, the attack stage may be too quick and cause distortions. In limiters this attack time is often 0ms, in which case it becomes a clipper. Setting look-ahead and peak hold to the same value will make the detector move ahead in time, so that it can react to attack stages before they actually occur and yet hold the levels for the actual signal to come.

Threshold

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

Level gain

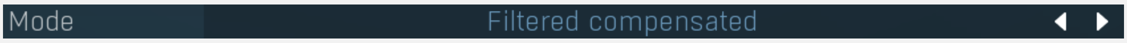
Level gain controls the gain applied to the detector, which can be used for example when the input level is too low, so that dynamic processing would be negligible, unless the level is boosted.

Link channels

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

Detector delay

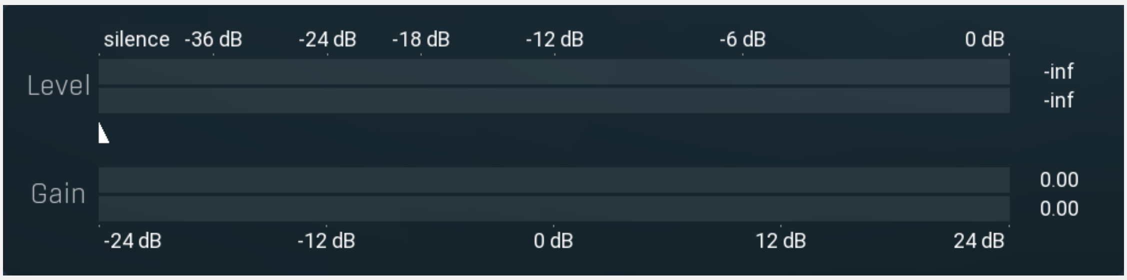
Detector delay lets you delay the detector input, hence the band will react later than the actual input signal.



Mode

Mode controls the way the band reacts to the input signal. It has no meaning if the dynamic gain is 0dB.
Filtered compensated mode is default and it means that the source for measuring input level is a filtered signal with additional compensation. For example, when using a low-shelf filter, the signal is low-passed with a filter with the same settings as the low-shelf, therefore the low-shelf filter is affected only by the signal the low-shelf is actually amplifying or attenuating. Since a low-passed signal with cut-off at 100Hz has usually a much lower level than the one filtered with cut-off at 10 kHz, additional compensation is performed to diminish these differences.
Filtered mode is similar, but the compensation is not performed. This may be advantageous for audio materials that do not contain the full spectrum, e.g. a bass line, where the compensation may make things complicated.
Entire spectrum mode is the simplest - it simply takes the input signal without any further processing. This may be useful for example to attenuate selected frequencies when the input level gets too high.

meters



Threshold

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

Harmonics panel

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

Linear

Linear button enables the linear harmonics spacing. When the main band frequency is say 100Hz and the **Semitones** value is 12, then in the default logarithmic mode the harmonics are 200Hz, 400Hz, 800Hz etc., increasing by 12 semitones (1 octave) each time. This is suitable because the filters themselves are logarithmic.
However harmonics generated by physical instruments are not spaced in this way. Rather, for a **Semitones** value of 12, they increase by a multiple of 12/12 of the main frequency each time. For example, for a base frequency of 100Hz, they will be at 200Hz, 300Hz, 400Hz, 500Hz etc. In linear mode the harmonics work in this way, but please note that then there is only a limited set of harmonics and Q is modified to approximate a reasonable behaviour, which is not always possible.

Dynamics by fundamental

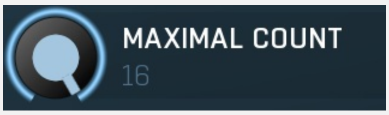
Dynamics by fundamental switch causes each harmonic to be driven by the same detector settings as set for the main band. It is disabled by default, which means that each harmonic is literally a clone of the original filter and has its own dynamics detector depending on its own frequency.
Please note that if you want each harmonic to behave in exactly the same way as the main band, you also need to switch on the Input (at the top of the Dynamics panel), otherwise the harmonics would be measuring the signal processed by the main band.

Harmonics

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

Semitones

Semitones defines the frequency interval of the harmonics. For example, if the band is at 100Hz and the number of semitones is 12 (default), then the first harmonic will be at 200Hz (12 semitones higher), second at 400Hz etc., increasing by 12 semitones (1 octave) each time. Thus they are logarithmically-spaced harmonics. When linearly-spaced harmonics are enabled, this merely changes the ratio between them. In this mode, 100Hz is followed by 200Hz, 300Hz, 400Hz, 500Hz etc, that is, increasing by a multiple of 12/12 of the main frequency each time.
For a value of 7 (a perfect fifth), the logarithmic harmonics would be at 150Hz, 225Hz, 337.5Hz, 506.25Hz etc, increasing by 7 semitones (= 50%, as 1.05946 ^ 7 = 1.498) each time and the linear harmonics would be at 158Hz, 251Hz, 397Hz, 628Hz etc, increasing by 7/12 each time.



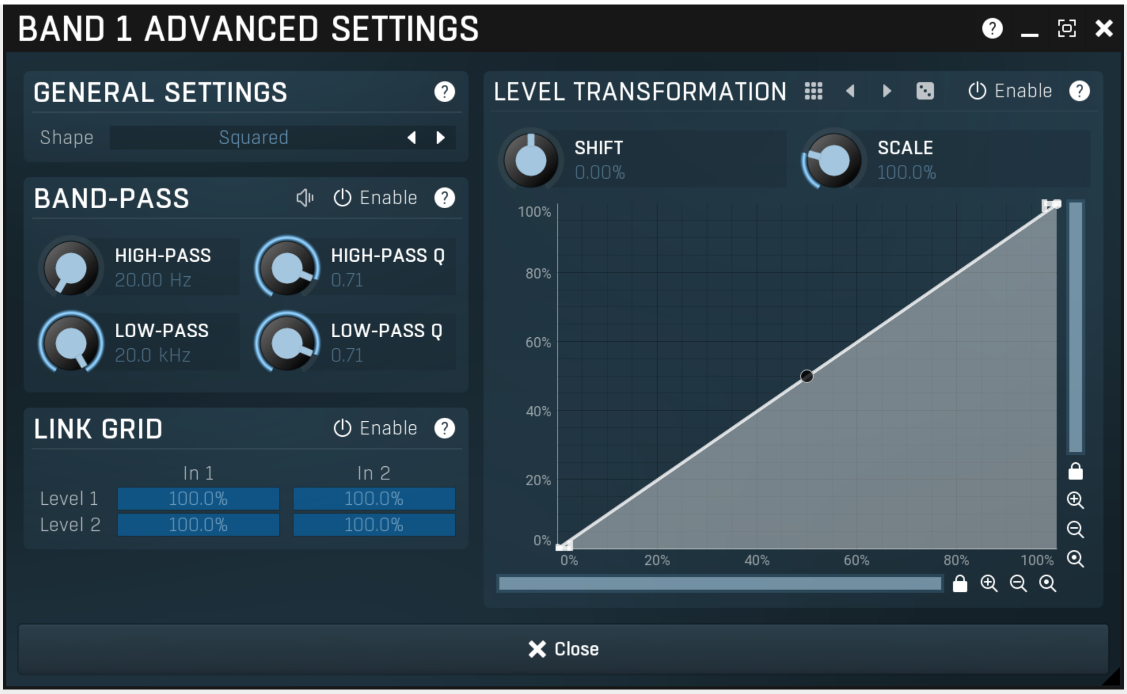
Maximal count

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

Harmonics grid

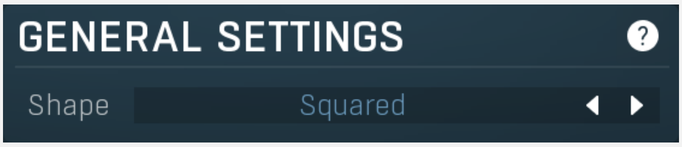
Harmonics grid is useful to turn on/off particular harmonics manually. Click any one to enable / disable it.

Band advanced settings

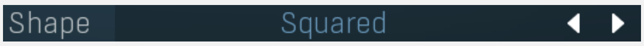


Band advanced settings contains additional settings for the band. These contain some more esoteric features, such as a dynamic transformation shape. It can be displayed by clicking the right mouse button on a band while holding **Ctrl**, from the basic band settings window, or from the band list if provided.

General settings panel



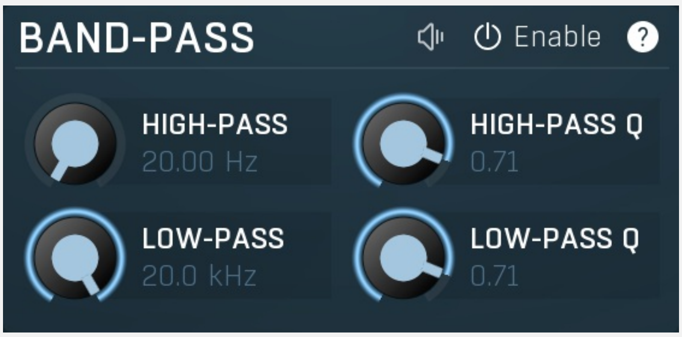
General settings panel contains additional parameters, which are too scientific to be available from the main band settings.



Shape

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

Band-pass panel



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

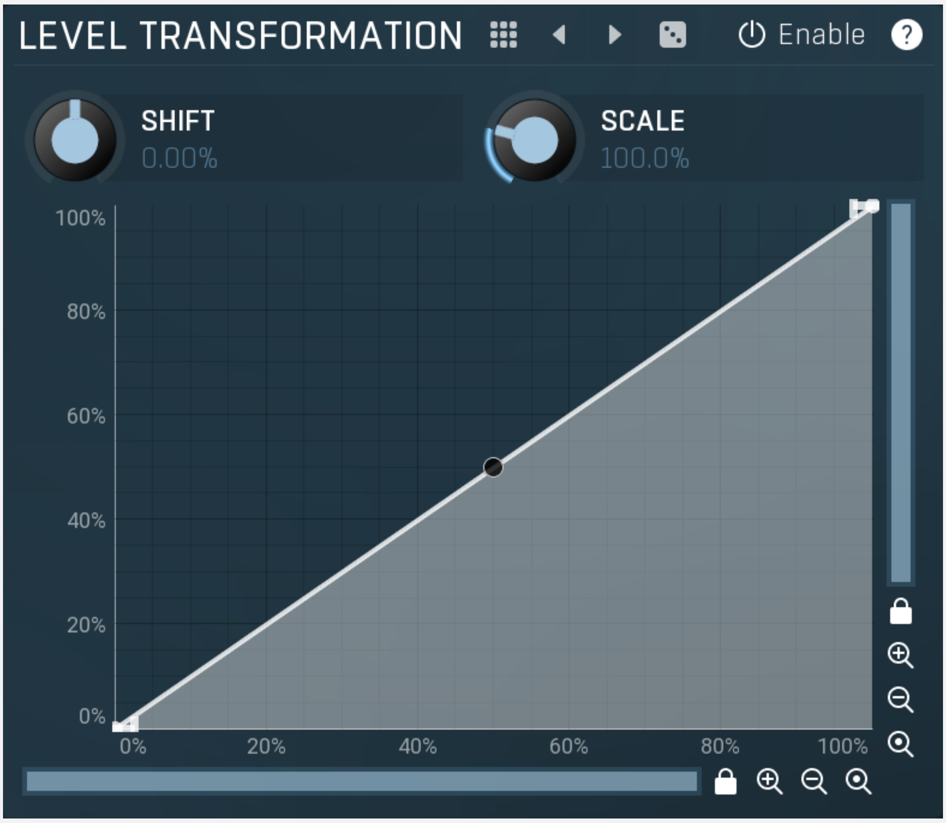
Play

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

Enable

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

Level transformation



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

Presets

Presets button displays a window where you can load and manage available presets. Hold **Ctrl** when clicking to load a random preset instead.

Left arrow

Left arrow button loads the previous preset.

Right arrow

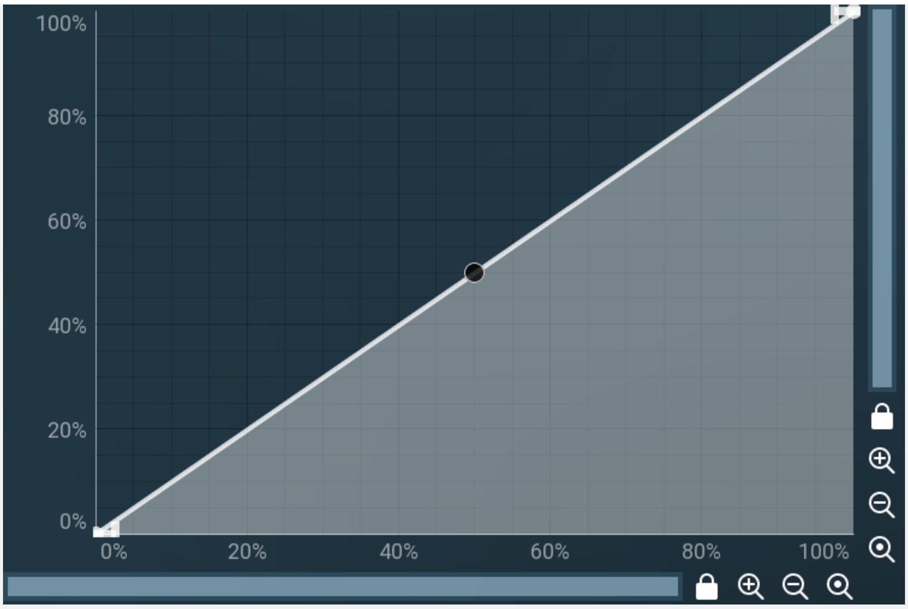
Right arrow button loads the next preset.

Randomize

Randomize button loads a random preset.

Enable

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



EnvelopeEditorGraph

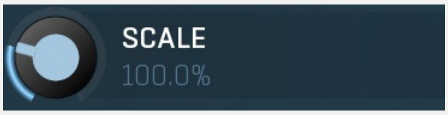
Envelope graph

Envelope graph provides an extremely advanced way to edit any kind of shape that you can imagine. An envelope has a potentially unlimited number of points, connected by several types of curves with adjustable curvature (drag the dot in the middle of each arc) and the surroundings of each point can also be automatically smoothed using the smoothness (horizontal pull rod) control. You can also literally draw the shape in drawing mode (available via the main context menu).

- **Left mouse button** can be used to select points. If there is a *point*, you can move it (or the entire selection) by dragging it. If there is a *curvature circle*, you can set up its tension by dragging it. If there is a *line*, you can drag both edge points of it. If there is a *smoothing controller*, you can drag its size. Hold **Shift** to drag more precisely. Hold **Ctrl** to create a new point and to remove any points above or below.
- **Left mouse button double click** can be used to create a new point. If there is a *point*, it will be removed instead. If there is a *curvature circle*, zero tension will be set. If there is a *smoothing controller*, zero size will be set.
- **Right mouse button** shows a context menu relevant to the object under the cursor or to the entire selection. Hold **Ctrl** to create or remove any points above or below.
- **Middle mouse button** drag creates a new point and removes any points above or below. It is the same as holding Ctrl and dragging using left mouse button.
- **Mouse wheel** over a point modifies its smoothing controller. If no point is selected, then all points are modified.
- **Ctrl+A** selects all points. **Delete** deletes all selected points.

Shift

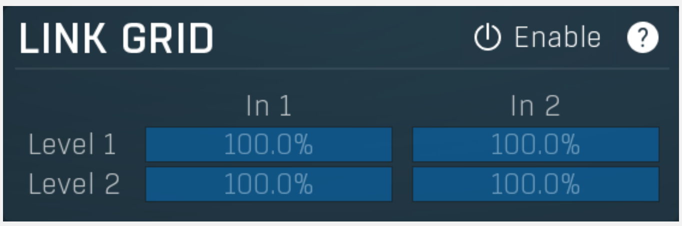
Shift lets you virtually shift the whole graph vertically. This basically shifts the dynamic gain.



Scale

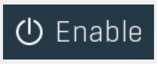
Scale lets you virtually scale the whole graph vertically. This basically scales the dynamic gain.

Link grid panel



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

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

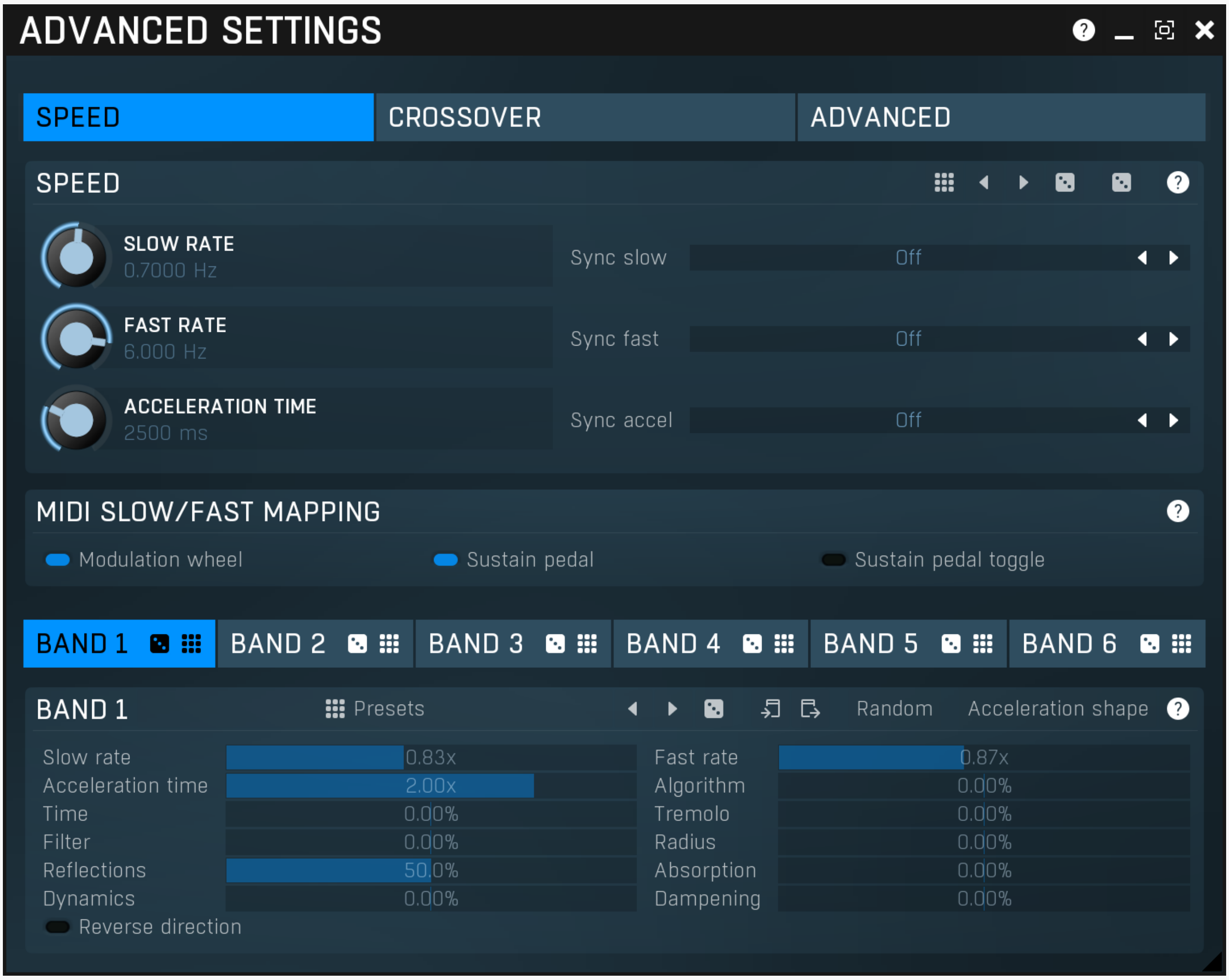


Enable
Enable button enables the link-grid module. It is off by default to save CPU resources.

Bands
Bands defines how many speakers (hence also crossover) bands are used. Standard rotary cabinets have 2 speakers, however you here can have up to 6 of them with different settings for each one.


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

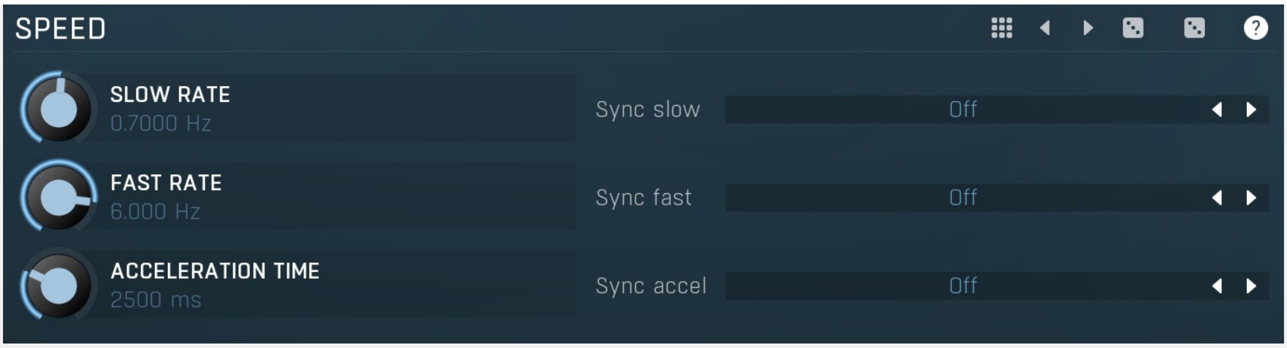
MRotaryAdvanced



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

Tab selector
Tab selector switches between subsections.

Speed panel



Speed panel contains the speed settings of the rotary speakers - rates in slow and fast modes and acceleration times. It is possible to synchronize each of these values to the host's tempo.

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

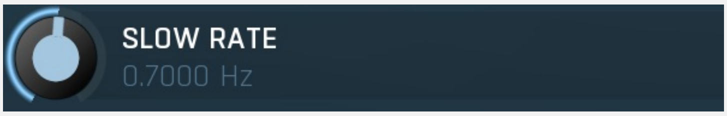
Right arrow button loads the next preset.

Randomize

Randomize button loads a random preset.

Random

Random button generates random settings using the existing presets.

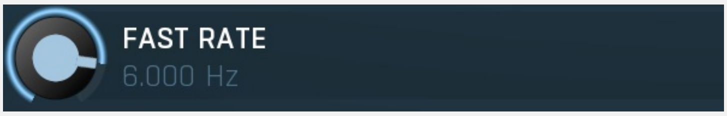


Slow rate

Slow rate defines the rotary speed in slow mode. Please note that each of the speakers can rotate at a different speed; you can configure these in the Advanced settings.
Range: 0.0200 Hz to 20.00 Hz, default 0.7000 Hz

Sync slow

Sync slow controls synchronization to host tempo.

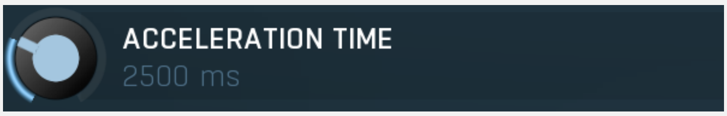


Fast rate

Fast rate defines the rotary speed in fast mode. Please note that each of the speakers can rotate at a different speed; you can configure these in the Advanced settings.
Range: 0.0200 Hz to 20.00 Hz, default 6.000 Hz

Sync fast

Sync fast controls synchronization to host tempo.



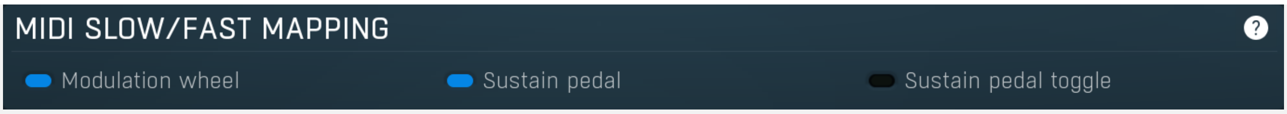
Acceleration time

Acceleration time controls how quickly each speaker can speed up or slow down. As usual, each speaker can have a different acceleration time and you can configure these in the Advanced settings.
Range: 100 ms to 30000 ms, default 2500 ms

Sync accel

Sync accel controls synchronization to host tempo.

MIDI slow/fast mapping panel



MIDI slow/fast mapping panel is useful when you want to map standard controllers to the slow/fast switch. Although you can map almost anything using the main plugin MIDI settings, this features is provided as it is easier to use.

Modulation wheel

Modulation wheel maps a MIDI modulation wheel to the slow/fast switch.

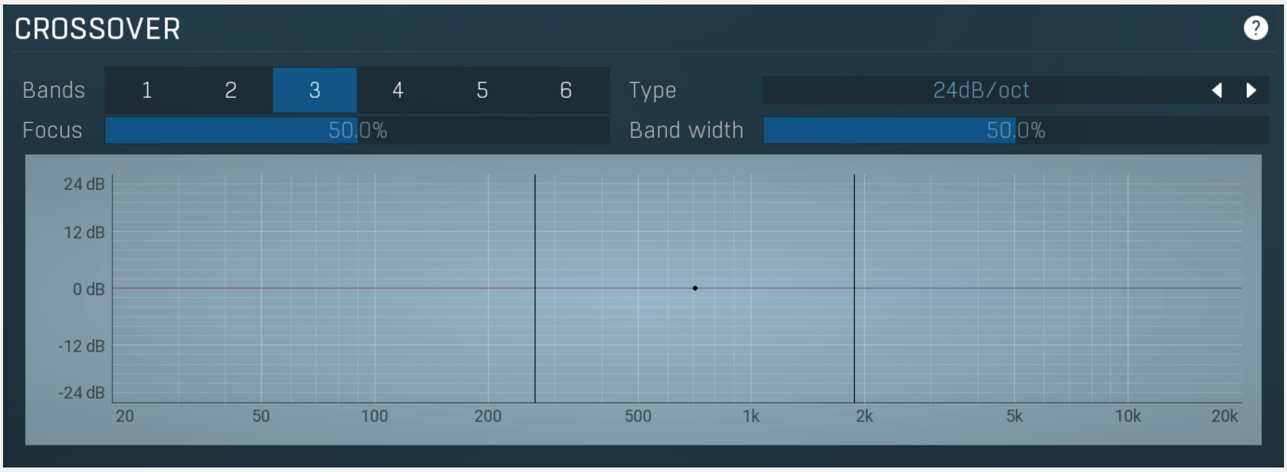
Sustain pedal

Sustain pedal maps a MIDI sustain pedal to the slow/fast switch.

Sustain pedal toggle

Sustain pedal toggle makes the sustain pedal toggle between the slow & fast modes, rather than the default (where sustain on switches to fast mode and sustain off switches to slow mode).

Crossover panel



Crossover panel controls the number of speakers and the crossover used to send different frequencies to each of them.

Bands

Bands defines how many speakers (hence also crossover) bands are used. Standard rotary cabinets have 2 speakers, however here you can have up to 6 of them with different settings for each one.
Range: 1 to 6, default 2

Type

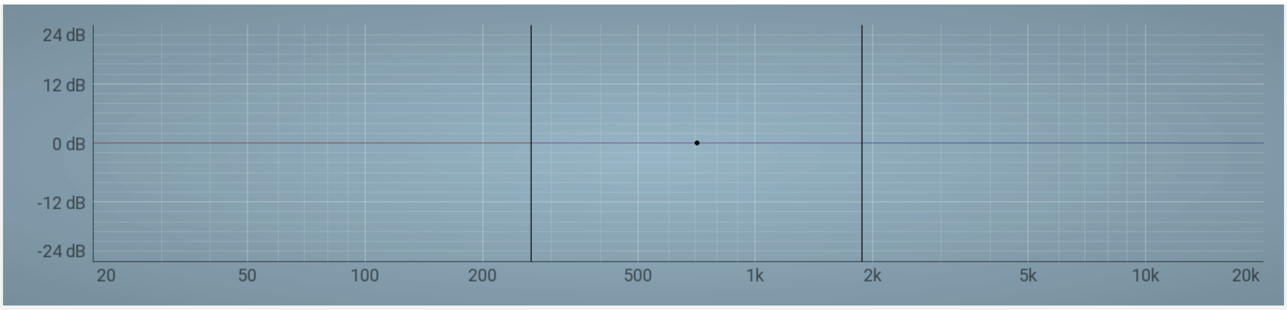
Type controls the type of the crossover (the slope). The higher the slope, the better the separation between them. However note that such separation actually may not be the best thing, because the blending of the frequencies on the borders of each band may produce very nice smooth chorus-like sound.

Focus

Focus controls the band limits. The higher the value, the more the bands are shifted to higher frequencies. Although you can control each limit manually using the graph below, this parameter along with **Band width** can provide faster and more natural band distribution. Also please note that since these parameters actually affect different parameters, you need to be careful when automating or modulating them.
Range: 0.00% to 100.0%, default 50.0%

Band width

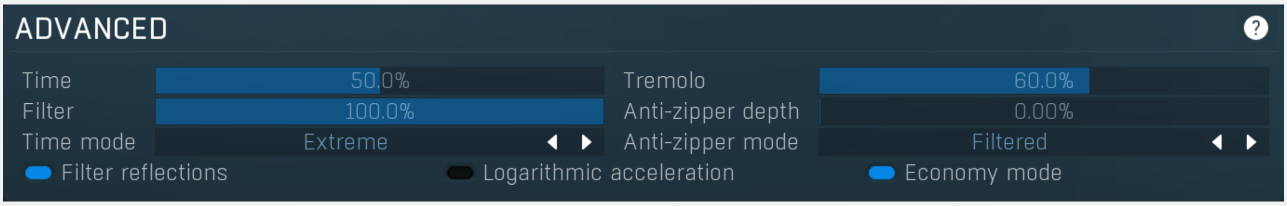
Band width controls the widths of the bands, hence how far the band limits are from each other. The higher the value, the wider these are. Although you can control each limit manually using the graph below, this parameter along with **Focus** can provide faster and more natural band distribution. Also please note that since these parameters actually affect different parameters, you need to be careful when automating or modulating them.
Range: 0.00% to 100.0%, default 50.0%



Bands editor

Bands editor shows the available bands, the cross-over frequencies delimiting them, and the input gains. Use left mouse button to change cross-over points or input gains.

Advanced panel



Advanced panel contains deeper settings, which you may want to adjust if you are seeking more creative sound for example. Most of these parameters are relevant for the **mathematical algorithm** only.

Time

Time defines the amount of time shift the rotary speakers cause. Since each of the speakers is moving away from and towards the microphones, its movement speed is added to the actual speed of sound causing a variable Doppler shift responsible for alterations of pitch. Use this parameter to control the amount of this effect. Higher values usually make the effect deeper, but when overused it may become unnatural, almost vibrato-like.
This parameter is relevant only for the **mathematical algorithm**.
Range: 0.00% to 100.0%, default 50.0%

Tremolo

Tremolo defines the amount of audio level changes. When a speaker cone is oriented towards a microphone, its volume is naturally louder than when it points in another direction. However in some cases the speakers may actually be almost omnidirectional. By increasing this value you make the speaker more directional increasing the differences between speaker pointing towards and away from the microphone.
This parameter is relevant only for the **mathematical algorithm**.
Range: 0.00% to 100.0%, default 60.0%

Filter

Filter controls the amount of additional filtering, which occurs in the virtual cabinet and is closely associated with the cabinet simulator.
This parameter is relevant only for the **mathematical algorithm**.
Range: 0.00% to 100.0%, default 100.0%

Anti-zipper depth

Anti-zipper depth controls amount of the anti-zipper protection. The higher it is, the fewer artifacts you can expect when modulating movements of the microphones for example, however the less accurate the simulation will be. Default value provides enough protection for most cases, even unnatural ones.
This parameter is relevant only for the **mathematical algorithm**.
Range: 0.00% to 100.0%, default 0.00%

Time mode

Time mode controls the way the Doppler shift occurs in different speeds. The **Basic** mode simulates the normal real world situation, where the speed of sound is constant. This is the natural way, however it isn't necessarily the nicest way, because while at low speeds the Doppler shift may not be noticeable at all, at higher speed it may become a vibrato typical for most rotary simulators. Therefore 2 more modes are available. **Extreme** compensates for the mentioned physical problem trying to make the Doppler shift similar in both high and low speeds. That avoids the vibrato effect in high speeds and makes the effects still powerful in low speeds. **Enhanced** mode is a compromise between these 2, being more natural, yet diminishing the physical problems.
This parameter is relevant only for the **mathematical algorithm**.

Anti-zipper mode

Anti-zipper mode avoids digital artifacts caused by Doppler shift and movement of the microphones and other devices in the scenario. The simplest example is when you move a microphone further away from the box. Although this movement may look perfectly natural on the screen, the actual physical movement may be impossible in nature and simulating these would end up with

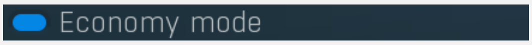
various artifacts. Therefore a precaution in the form of anti-zipper protection is taken. Each mode sounds slightly different, in most cases you will end up with the defaults.
This parameter is relevant only for the **mathematical algorithm**.

Filter reflections

Filter reflections is enabled by default and causes the reflections inside the cabinet to be processed by the rotation filters. This is pretty natural, however you may want to disable it for creative reasons.
This parameter is relevant only for the **mathematical algorithm**.

Logarithmic acceleration

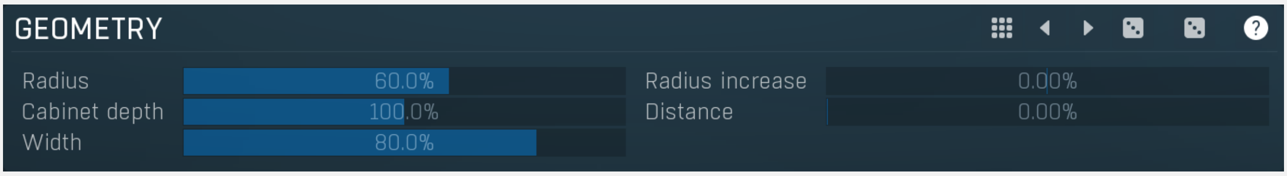
Logarithmic acceleration controls the way in which the rotation speeds up and slows down. By default this is disabled making the speakers increase their speed linearly, which is natural for physical objects exposed to some kind of constant force. Logarithmic scale, which is widely used in audio applications, despite being not natural sounds very good as well. In this case, speeding up from 1Hz to 2Hz takes the same time as for speeding up from 4Hz to 8Hz, or 8Hz to 16Hz for example. This means that the faster an object is, the faster it can accelerate. This is not truth in physical world, but it may be advantageous.



Economy mode

Economy mode saves CPU power when processing in sample rate 80000Hz and higher by temporary downsampling. This is done for the main modulation processing only as processing in higher sampling rates doesn't provide any significant improvement in audio quality and the CPU requirements can get extremely high.

Geometry panel



Geometry panel contains geometrical properties of the virtual cabinet and the virtual recording microphones. These parameters can be configured directly from the editor in the main window by dragging particular components. Please note that only some of these parameters affect the **simulator** algorithm, but all of them are relevant for the **mathematical algorithm**.

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.

Random

Random button generates random settings using the existing presets.

Radius

Radius defines the radius of the speaker cones. Lower value makes the speakers smaller and higher values make them larger. You can control this value directly from the speaker graph by dragging the speaker axis. As usual, each speaker can have a different acceleration time and you can configure these in the Advanced settings.
Speaker radius is one of the major settings controlling the physical modelling processor. It affects all kinds of properties from Doppler shift and cabinet reflections to rotation volume change. In general, the smaller the cone, the more subtle the effect.
Range: 0.00% to 100.0%, default 60.0%

Radius increase

Radius increase is a simple method to make each of the speaker cones have a different size. If the value is positive, higher speakers will be bigger and vice versa. For more natural results, the value should be lower or equal to zero, because higher frequency speakers are usually smaller in size and can rotate more quickly, which might otherwise have led to extensive Doppler shift (if the cone would be large and fast at the same time). This configuration may however be very well used creatively.
Range: -25.0% to +25.0%, default 0.00%

Cabinet depth
Cabinet depth controls the inner size of the virtual cabinet. This affects mainly the intra-cabinet reflections. You can control this value directly from the speaker graph by dragging the top line.
Range: 0.00% to 200.0%, default 100.0%

Distance
Distance controls the distance of the microphones from the center of the virtual cabinet. This naturally affects the produced depth of space. The higher the distance, the more ambient and thinner the sound. In most cases you will want to place the microphones directly in front of the cabinet (which is the usual way for recording them in the real world) or even inside the cabinet, which is actually something physically hard or impossible to create.
Range: 0.00% to 400.0%, default 0.00%

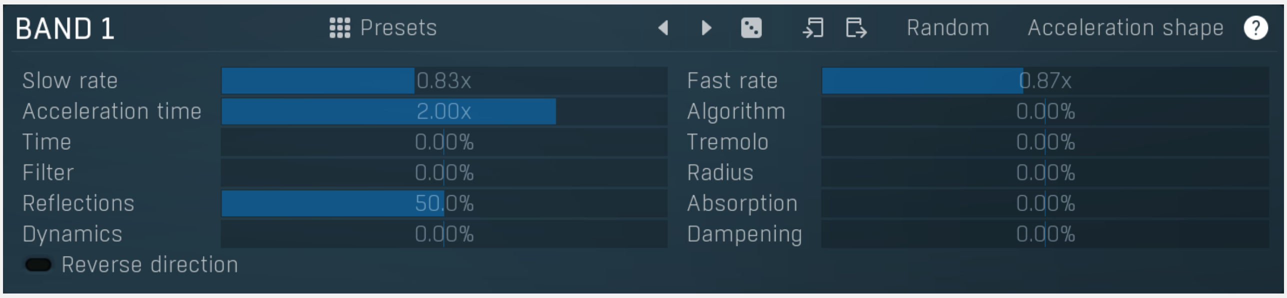
Width
Width defines the distance of the microphones from each other. The further they are from each other, the wider the output usually seems as each microphone is exposed to a different amount of time (Doppler shift), volume and spectral alterations. However note that if the microphones are too close to the sides of the virtual cabinet, the direct sound may start to blend with the initial reflections making the sound thinner. This is one of the reasons the default value is actually slightly smaller.
Range: 0.00% to 100.0%, default 80.0%

Tab selector
Tab selector switches between subsections.

Randomize
Randomize button generates random settings for the tab.

Presets
Presets button chooses a random preset for the tab.

Band panel



Band panel controls the specifics of the single speaker. This includes its speed and acceleration, radius, amount of Doppler shift and much more. You can use this to make each speaker different.

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.

Acceleration shape
Acceleration shape button displays the speed shape for this speaker. The shape controls actual speed when switching from slow to fast and vice versa. You can use this to bend the shape slightly for more accurate physical simulations, or even to implement various creative effects. For example, you can make the speaker speed up and slow down every time the user switches the between slow and fast.

Slow rate 0.83x
Slow rate
Slow rate defines the speaker speed in slow mode. The value is specified as multiplier to the global **Slow rate** parameter.
Range: 0.25x to 4.00x, default 0.83x

Fast rate 0.87x
Fast rate
Fast rate defines the speaker speed in fast mode. The value is specified as a multiplier to the global **Fast rate** parameter.
Range: 0.25x to 4.00x, default 0.87x

Acceleration time 2.00x
Acceleration time
Acceleration time controls how quickly can the speaker speed up and slow down. The value is specified as a multiplier to the global **Acceleration time** parameter.
Range: 0.25x to 4.00x, default 2.00x

Algorithm 0.00%
Algorithm
Algorithm controls the ratio between the 2 algorithms that the plugin provides. The value is specified as the difference from the global **Algorithm** parameter.
Range: -100.0% to +100.0%, default 0.00%

Time 0.00%
Time
Time controls the amount of time shift for this speaker. The value is specified as the difference from the global **Time** parameter. This parameter is relevant only for the **mathematical algorithm**.
Range: -100.0% to +100.0%, default 0.00%

Tremolo 0.00%
Tremolo
Tremolo controls the amount of audio level changes for this speaker. The value is specified as the difference from the global **Tremolo** parameter. This parameter is relevant only for the **mathematical algorithm**.
Range: -100.0% to +100.0%, default 0.00%

Filter 0.00%
Filter
Filter controls the amount of additional filtering for this speaker. The value is specified as the difference from the global **Filter** parameter. This parameter is relevant only for the **mathematical algorithm**.
Range: -100.0% to +100.0%, default 0.00%

Radius 0.00%
Radius
Radius controls the radius of this speaker. The value is specified as the difference from the global **Radius** parameter. For easier handling you may use the global **Radius increase** parameter.
Range: -100.0% to +100.0%, default 0.00%

Reflections 50.0%
Reflections
Reflections controls the character of reflections, thus how well the sound reflects from different parts of the cabinet. It simply affects the sound character. This parameter is relevant only for the **mathematical algorithm**.
Range: 0.00% to 100.0%, default 50.0%

Absorption 0.00%
Absorption
Absorption controls the amount of the absorption of the cabinet material. The value is specified as the difference from the global **Absorption** parameter.
Range: -100.0% to +100.0%, default 0.00%

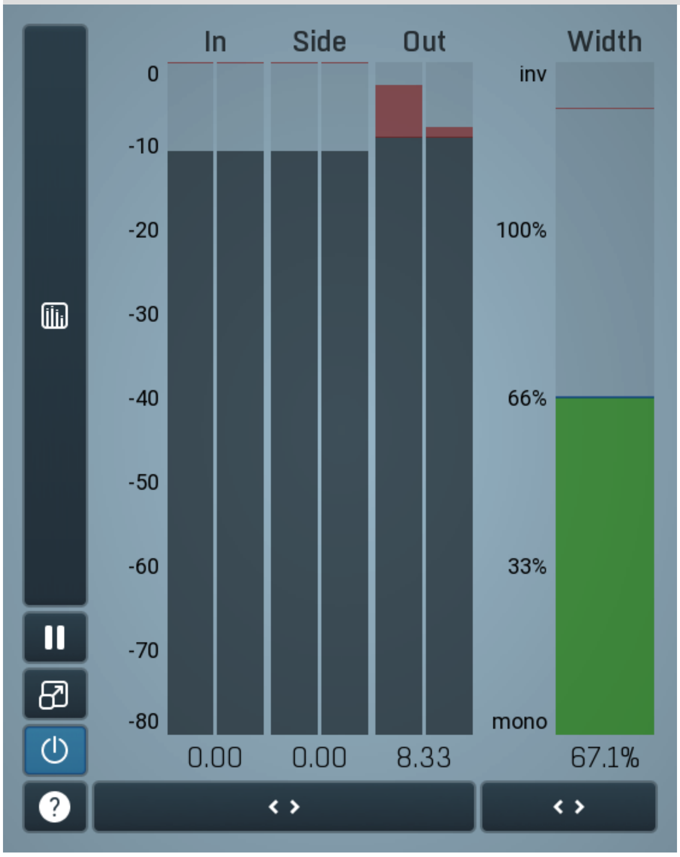
Dynamics 0.00%
Dynamics
Dynamics controls the amount of dynamic behaviour. The value is specified as the difference from the global **Dynamics** parameter. This parameter is relevant only for the **simulator algorithm**.
Range: -200.0% to +200.0%, default 0.00%

Dampening 0.00%
Dampening
Dampening simulates filling the cabinet with an absorption material hence shortening the response The value is specified as the difference from the global **Dampening** parameter. This parameter is relevant only for the **simulator algorithm**.
Range: -100.0% to +100.0%, default 0.00%

Reverse direction

Reverse direction

Reverse direction button makes the speaker rotate clockwise instead of anti-clockwise.



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.

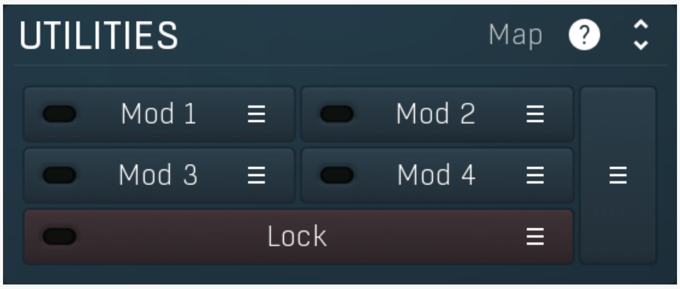
Collapse

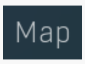
Collapse button minimizes or enlarges the panel to release space for other editors.

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.