MCabinet

MCabinet is a revolutionary solution for simulating eletric guitar cabinets, eletric bass cabinets, and technically any other cabinet for that matter. Many musicians agree that these are half of the resulting sound. As it turns out, MCabinet can also produce various creative tones, a very natural sounding stereo expansion and even double tracking and short reverberation.

MCabinet's main core is based on a statistical analysis of measurements of actual physical cabinets, impulse responses, resulting in the so-called profiles. Many are already provided as presets. Guitarists are used to browsing thousands of IR files and using one or a few of them in parallel via convolution. While this is certainly possible, it is extremely clumsy, requires having tons of tiny files (usually commercial) on your computer, and selecting the right one is kind of a hit and miss process taking lots of precious time.

A profile can be made by analyzing a single IR file, but the true magic is made by analysing thousands of them. MCabinet extracts what's "common" between all the recorded cabinets as well as how much they differ from each other. The profile itself can often sound better than the original IRs, since it avoids the intermediary faulty resonances, but the true fun comes from using the MCabinet's parameters. First you can alter the cabinet response by various processes such as spectral smoothing and tilting. But the actual creative power is brought by the resonators and wideners. As a result, MCabinet is able to produce pretty much infinite number of cabinets and you can control various aspects of these cabinets directly instead of just browsing thousands of files. There is also a dynamic equalizer to fine-tune the resulting sound.

Resonators

MCabinet's resonators are based on physical processes that occur within real cabinets, taken to an extreme. These generate the actual variations in the response of the cabinet, taking into account the actual profile. By analyzing a profile of thousands of cabinet IRs and using the resonators, MCabinet can produce variations of the original cabinets, ranging from realistic stuff that does or doesn't exist, up to a completely creative madness.

When recording guitars through real cabinets, it is kind of a hit and miss process, since the tone and resonances depend on the actual cabinet, microphone, position of the mike and recording room at the same time. In most cases this produces unnatural sounding resonances, which are then attenuated by mixing multiple microphones (or IRs in the box) etc. MCabinet on the other hand profiles the cabinets without these unnatural resonances, or can intentionally remove them, and then it can give the sound a character based on the parameters you like. It can produce the unnatural resonances you are used to if you like them for some reason. Or it can give it more natural character you are used as a listener when a real cabinet isn't recorded yet.

You are going to enjoy MCabinet even more when you start using stereo processing. Guitarists are traditionally searching for matching IRs, one for left and one for right channel, especially if they are not double tracking the guitar performances. This is an enormously time consuming task. MCabinet makes that extremely quick and simple by creating stereo responses, which are matched naturally.

Whether you are using mono or stereo processing, the sound of each resonator is defined by its few parameters and the 3 seeds, one for bass, one for mids and one for treble. Seeds are basically random numbers, that somehow produce the response. All you need to do is press the seed buttons until you find a sound you like. The main resonator controls are **Depth**, which controls how much the resonator does, and **Smoothing**, which basically defines how many spectral hills and valleys there will be. The higher the value, the more natural the response

is and it can be called "tone". With low values, steep hills and valleys start occuring, producing more distinct resonances. With very low values you may experience ringing and a sort of ambience or short reverberation. It is often useful to use mutliple resonators, with different smoothing values, each controlling a different aspect of the audio.

Wideners

Wideners serve as a supplement to the resonators with main purpose - create artificial stereo. Whereas resonators mainly change the spectral level profile, wideners produce spectral delay profile. So in a way wideners delay frequencies differently between left and right channels, which is a similar process to what happens when doing a manual double tracking.

Wideners have pretty much the same parameters as the resonators and work similarly, but do a very different thing. With high enough smoothing they can provide a very natural stereo expansion, almost like a double tracking, especially when inserted before a stereo amp. With low smoothing you can again expect audible reverberation effects.

Using MCabinet and performance considerations

To use MCabinet, it's best to simply start with one of the global presets or profiles presets. Then use one or more resonators to get the specific sound you want. Finally if you are into stereo processing, you can use a stereo resonator, but usually more effective is to use a widener, which is designed for the job. On the other hand, if you are processing a mono signal and expect a mono output, it would be desirable not to use any of the stereo tools, and potentially set the global Channel mode to M (as mid or mono) to save some CPU if your DAW doesn't tell the plugin that it is processing a mono signal.

Flat profile and creative processing

MCabinet may be quite useful for general sound design as well. The classic profiles produce the specific sound character typical for say guitar cabinets. But the default Flat profile doesn't initially alter the spectrum levels, so it becomes pretty natural and you can exploit the resonators and wideners to create entirely original effects. Also consider producing profiles by analyzing samples that aren't really cabinet IRs. You may also exploit the included modulators, but be aware that modulating any of the resonator & widener parameters may require lots of CPU and since recalculation of the processing structures is performed in background, you may need to perform "live" rendering in your DAW.

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.

Panic

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

Settings

Settings

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

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

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

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

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

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

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

WWW

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

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.
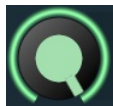
Input

Input defines the input gain, the power modification applied to the input signal. While it doesn't have any effect on the cabinet simulation itself, it may have effect on the dynamic equalizer.
Range: -24.00 dB to +24.00 dB, default 0.00 dB

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%

Output

Output defines the power modification applied to the output signal. It is performed after the dynamics processing, hence its main purpose is to fix the output level so that the **Dry/Wet** parameter is easier to use.
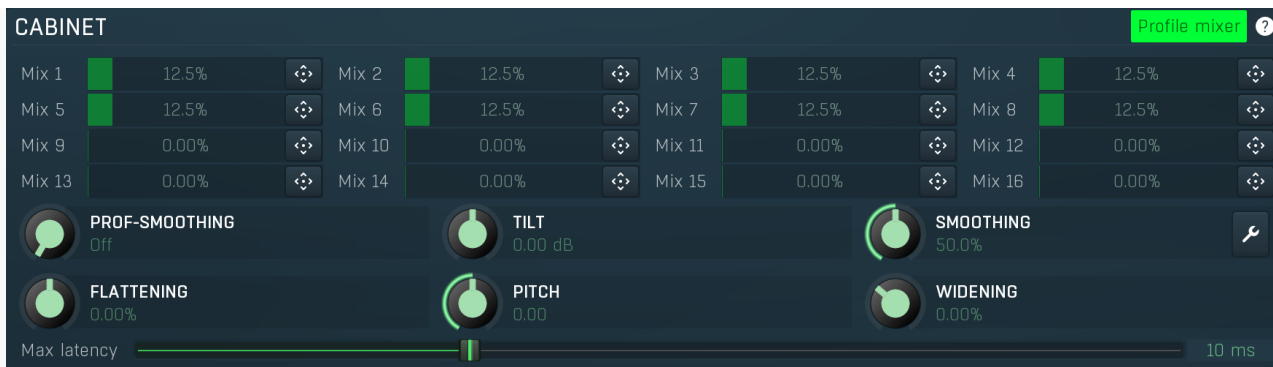Range: -24.00 dB to +24.00 dB, default 0.00 dB

Tab selector

Tab selector switches between subsections.

Cabinet panel

Cabinet panel contains the parameters of the actual virtual cabinet generator.

Profile mixer

Profile mixer

Profile mixer lets you mix multiple profiles instead of just selecting one. This lets you not only use resonators to generate custom tones, but start with a profile created as mix of other profiles.
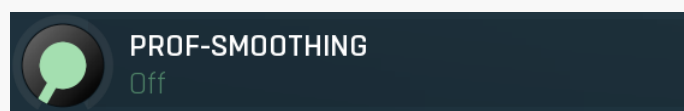
Prof-smoothing

Prof-smoothing post-processes the profile by making it smoother. In practice this lowers the amount of resonances. Unlike **Smoothing**, which processes the final output including the resonators, this one makes the input profile smooth only. That way you can remove details in the profile itself and use is as tonal balance only, then use the resonators to produce the resonances. It is especially useful when using a single IR for analysis.
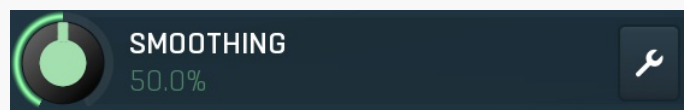Range: Off to 100.0%, default Off

Tilt

Tilt applies a linear equalization to the profile making it more trebly or more bassy.
Range: -24.00 dB to +24.00 dB, default 0.00 dB

Smoothing

Smoothing post-processes the output spectrum by making it smoother. In practice this lowers the amount of resonances.
Range: Off to 100.0%, default Off

Flattening

Flattening post-processes the spectrum profile by making it flatter. For example, most cabinets have very poor high frequency response, which is part of the tone, but the flattening feature lets you restore the high frequencies without destroying the resonances and character. With negative values it can actually do the opposite and make the overall character even more noticeable.
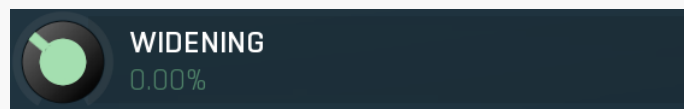Range: -100.0% to 100.0%, default 0.00%

Pitch

Pitch shifts the pitch of the cabinet response.
Range: -24.00 to +24.00, default 0.00

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 200.0%, default 0.00%
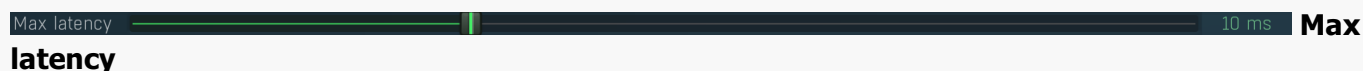
Max latency

Max latency controls the maximum delay of the main transient the plugin is allowed to cause. Like with any real cabinet being miked, there is a certain frequency-dependent delay. This is especially prominent when using the Wideners. While each frequency is delayed differently there's still the main 'peak' in the impulse response, which in way defines the latency of 'most of the spectrum'. This parameter allows you to make the latency smaller if necessary at the expense of lower audio accuracy.
Range: 0 ms to 100 ms, default 10 ms

Resonator panel
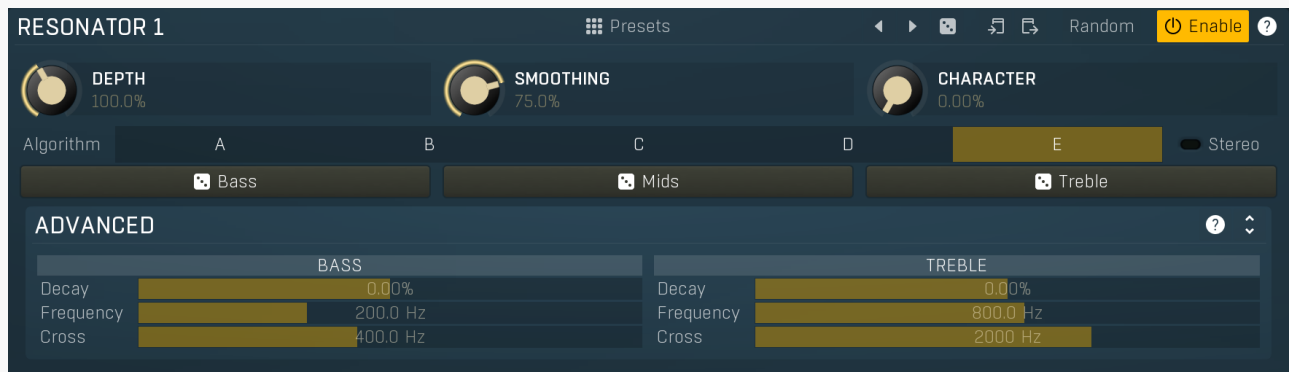
Resonator panel contains controls of the resonator module, which produces some pseudo-random character and is a vital part of the cabinet generator. It follows the profile and based on the parameters and the seed produces changes in the tonal characteristics of the cabinet. By clicking **Seed** you can produce nearly infinite numbers of such cabinets. The plugin features multiple resonators, so that one can generate overall tonal characteristics and the other one resonances, for example.

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
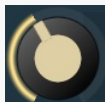Copy button copies the settings onto the system clipboard.

Paste
Paste button loads the settings from the system clipboard.

Random
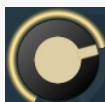Random button generates random settings using the existing presets.

Depth
Depth controls how much the resonator changes the resulting spectrum. With 0% it has no effect at all.
Range: 0.00% to 250.0%, default 100.0%

Smoothing
Smoothing smoothens the output of the resonator and basically controls 'what' the resonator produces. With low values the output is too 'edgy' and can be referred to as resonances. Conversely high values make the resonator output smoother and control the overall tonal characteristics.

Range: 0.00% to 100.0%, default 25.0%

Character

Character

Character controls the way the smoothing is performed and highly affects the resulting tonal character.
Range: 0.00% to 100.0%, default 100.0%

Algorithm	A	B	C	D	E

Algorithm

Algorithm defines which resonator algorithm is used. The plugin provides multiple algorithms for generating resonances each sounding very different. The actual functionality is pretty irrelevant, please use your ears and choose the algorithm you like the most.

Stereo   **Stereo**

Stereo switch makes the resonator produce stereo output instead of mono one. In the old days you would search for multiple cabinets that 'go with each other', one for left channel and the other for right. With this plugin you simply enable this button and the plugin will automatically create left & right channels that are usually compatible. As always if you don't like the character or stereo image, click the **Seed** button to get a new cabinet.

Bass

Bass

Bass controls the pseudorandom value, which changes the bass part of the resonator spectrum. Whenever you click the button, you get a new value and in a way a new cabinet. Your goal is simply to find one you like.
Range: 0 to 2147483647, default 12345678

Mids

Mids

Mids controls the pseudorandom value, which changes the mid part of the resonator spectrum. Whenever you click the button, you get a new value and in a way a new cabinet. Your goal is simply to find one you like.
Range: 0 to 2147483647, default 12345678

Treble

Treble

Treble controls the pseudorandom value, which changes the treble part of the resonator spectrum. Whenever you click the button, you get a new value and in a way a new cabinet. Your goal is simply to find one you like.
Range: 0 to 2147483647, default 12345678

Advanced panel

ADVANCED				? ⇅
BASS			TREBLE	
Decay	0.00%	Decay	0.00%	
Frequency	200.0 Hz	Frequency	800.0 Hz	
Cross	400.0 Hz	Cross	2000 Hz	

Advanced panel contains some more advanced settings of the resonator.

Decay	0.00%

Bass decay

Bass decay controls how and the resonator behaves for the lowest frequencies. In nature these frequencies naturally resonate less. Plus they can be quite problematic, producing various reverberation-like sounds. So it is natural to stop them from resonating. However you can disable the decay by setting this to 0% for creative purposes.
Range: -400.0% to 400.0%, default 0.00%

Decay	0.00%

Treble decay

Treble decay controls how and the resonator behaves for the highest frequencies. In nature these frequencies naturally resonate less. Plus they can be quite problematic, producing various reverberation-like sounds. So it is natural to stop them from resonating. However you can disable the decay by setting this to 0% for creative purposes.
Range: -400.0% to 400.0%, default 0.00%

Frequency	200.0 Hz

Bass frequency

Bass frequency controls the frequency below which the effect of the resonator goes down according to **Bass decay** parameter.
Range: 20.00 Hz to 20.0 kHz, default 200.0 Hz

Frequency	800.0 Hz

Treble frequency

Treble frequency controls the frequency above which the effect of the resonator goes down according to **Treble decay** parameter.
Range: 20.00 Hz to 20.0 kHz, default 800.0 Hz

Cross	400.0 Hz

Bass cross

Bass cross controls the frequency below which the effect of the resonator is defined by the seed **Bass**.
Range: 20.00 Hz to 20.0 kHz, default 400.0 Hz

Treble cross

Treble cross controls the frequency above which the effect of the resonator is defined by the seed **Treble**.
Range: 20.00 Hz to 20.0 kHz, default 2000 Hz

Widener panel

Widener panel contains controls of the widener module, which produces some artificial stereo by changing phase of individual frequencies in left & right channels. Its effect may range from a subtle widening to a sort-of short reverberation for small smoothing values. By clicking **Seed** you can produce nearly infinite numbers of different widening characteristics.

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.

Depth

Depth controls how much the widener changes the resulting phase spectrum. With 0% it has no effect at all.
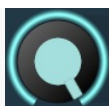Range: 0.00% to 100.0%, default 100.0%

Smoothing

Smoothing smoothens the output of the widener and basically controls 'what' the widener produces. With low values the output is too 'complex' and produces a kind of short reverberation. High values make the widener output smoother and produce a widening targeting different spectrum areas. With very high values it may be hard to find seeds that would even split the spectrum and provide centered stereo field.
Range: 0.00% to 100.0%, default 25.0%

Character

Character controls the way the smoothing is performed and highly affects the resulting character.
Range: 0.00% to 100.0%, default 100.0%

Algorithm

Algorithm defines which widener algorithm is used. The actual functionality is pretty irrelevant, please use your ears and choose the algorithm you like the most.

Seed

Seed controls the pseudorandom value, which changes the widener characteristics. Whenever you click the button, you get a new value and in a way a new stereo field. Your goal is simply to find one you like.
Range: 0 to 2147483647, default 12345678

Advanced panel

Advanced panel contains some more advanced settings of the widener.

Bass decay

Bass decay controls how and the widener behaves for the lowest frequencies. This lets you limit the spectrum being widened.
Range: -400.0% to 400.0%, default 0.00%

Treble decay

Treble decay controls how and the widener behaves for the highest frequencies. This lets you limit the spectrum being widened.
Range: -400.0% to 400.0%, default 0.00%

Bass frequency

Bass frequency controls the frequency below which the effect of the widener goes down according to **Bass decay** parameter.
Range: 20.00 Hz to 20.0 kHz, default 20.00 Hz

Treble frequency

Treble frequency controls the frequency above which the effect of the widener goes down according to **Treble decay** parameter.
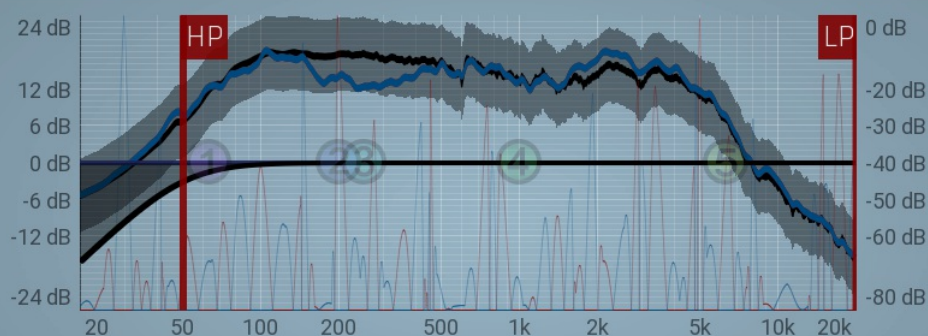Range: 20.00 Hz to 20.0 kHz, default 20.0 kHz

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.

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.

Menu

Menu button displays a menu with additional features.

Analyse IR lets you analyze a single impulse response file. The deviation will be constant of course. Use **Analyse IR folder** to get more realistic profile from ideally a big set of cabinets. Analysing a single file may be useful for creative processing though, or if you want to alter an existing cabinet IR file somehow.

Analyse IR folder lets you analyze a folder of impulse response files (including all subfolders) and perform a statistical analysis of the existing cabinets represented by these IR files. This is an essential feature, which teaches the plugin how should a certain set of cabinets sound in order to generate them. The plugin will capture the profile based on which it will produce its own cabinets.

Standardize deviation transforms the range of the deviation for the main part of the spectrum. When analyzing cabients the differences are often quite subtle and one then needs to increase the resonator depth too much. This feature analyzes the existing device analysis and changes its range if needed, so that it matches other profiles.

Profile from current creates a profile based on current output of the first channel (hence not stereo). It can be used for creative processing. For example, when you have good sounding settings and you want to perform additional post-processing and creative experiments, it may be desirable to use it as profile and use the resonators from there.

Export IR saves the current output as a WAV file to be used with any plugin or hardware that supports custom convolution IRs.

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

Analog

Analog controls the amount of internal nonlinearities in each filter, typical for analog equalizers. Note that this processing also changes the actual filter shapes, so they won't fully match the displayed graphs anymore when Analog feature is used. It can also require a solid amount of CPU power.
Range: 0.00% to 100.0%, default 0.00%

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

Invert gain   Swap gains   ?

FREQUENCY
63.25 Hz

Q
87.5%

GAIN
0.00 dB

Slope	1	2	3	4	5	6	7	8	9	10

Channels	Left	Left + Right	Right

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

Swap gains

Swap gains
Swap gains button swaps values between gain and dynamics gain.

FREQUENCY
63.25 Hz

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

Q
87.5%

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
0.00 dB

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

Slope	1	2	3	4	5	6	7	8	9	10

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	Left	Left + Right	Right

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.
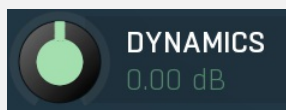
Dynamics

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.
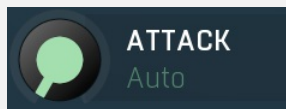
Attack

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.
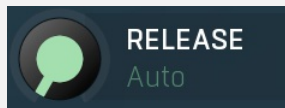
Release

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.
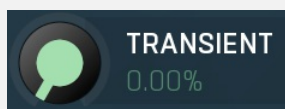
Transient

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 `2.0 ms` **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 `silence` **Threshold**

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

Level gain `0.00 dB` **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 `100.0%` **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 `0 ms` **Detector delay**

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

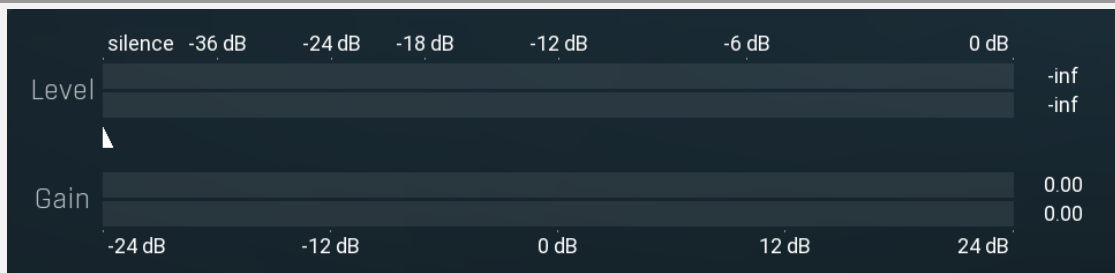**Mode** `Filtered compensated` ◀ ▶ **Mode**

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

silence	-36 dB	-24 dB	-18 dB	-12 dB	-6 dB	0 dB	
Level							-inf
							-inf
Gain							0.00
							0.00
-24 dB		-12 dB		0 dB		12 dB	24 dB

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

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

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**.
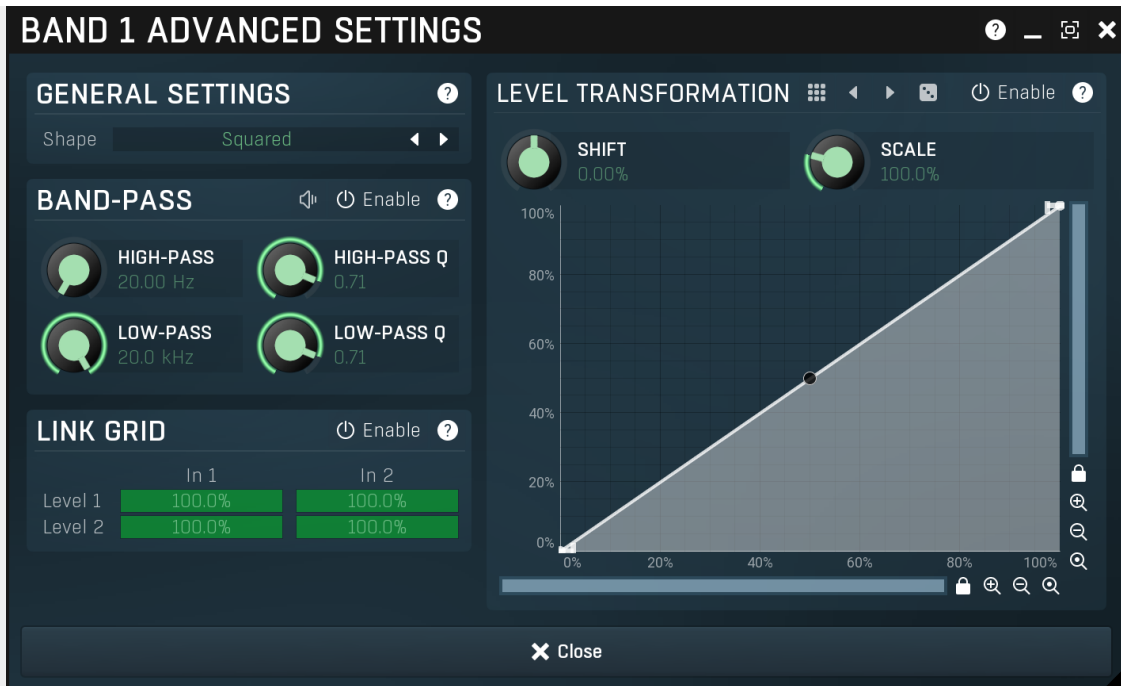
Harmonics grid

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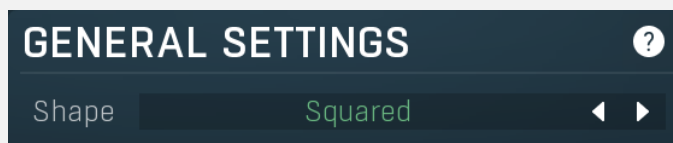
General settings panel contains additional parameters, which are too scientific to be available from the main band settings.

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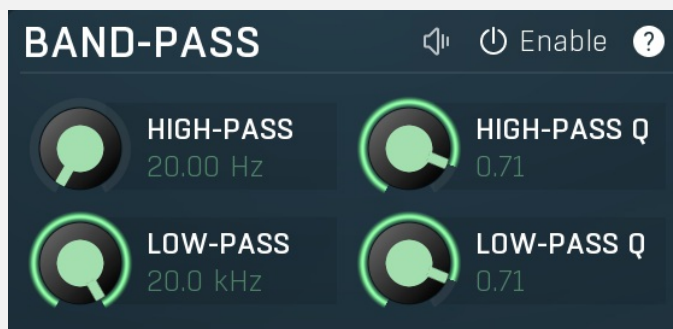
Band-pass panel contains parameters of the band pass, which you can use to process the signal that is used measure level of the band additionally. For example, you may want a band at high frequencies to react to bass content; you can do this by placing the band anywhere on the high frequencies and set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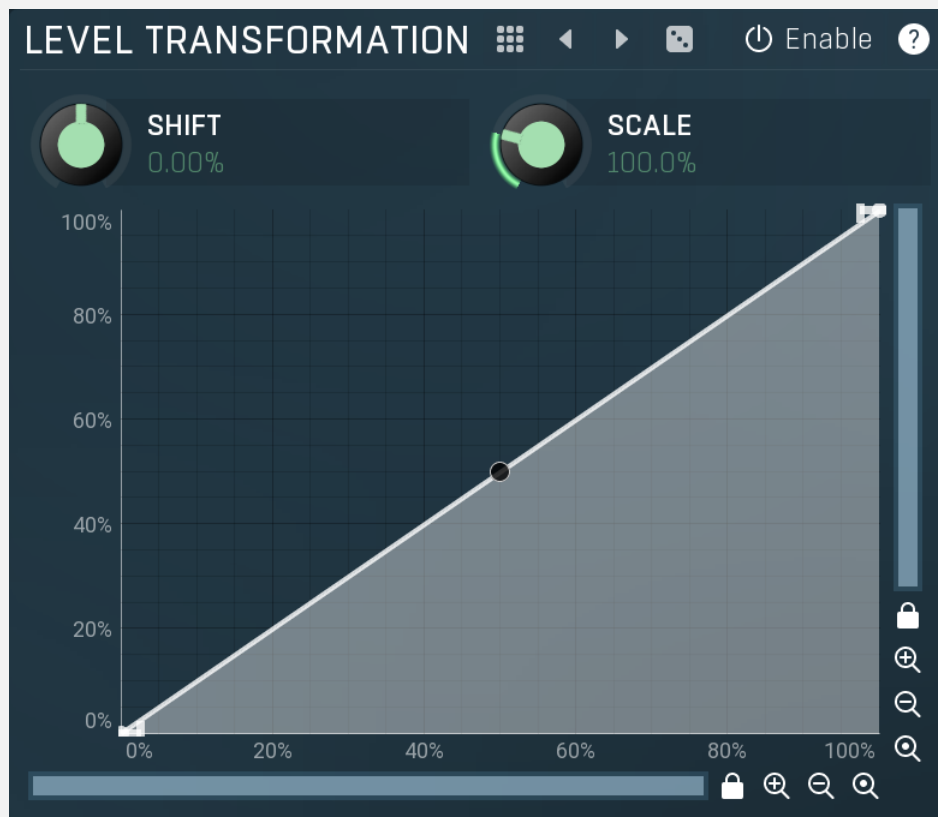
 Play
Play button enables the band-pass monitoring and hence could be useful to tweak the band pass.

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.

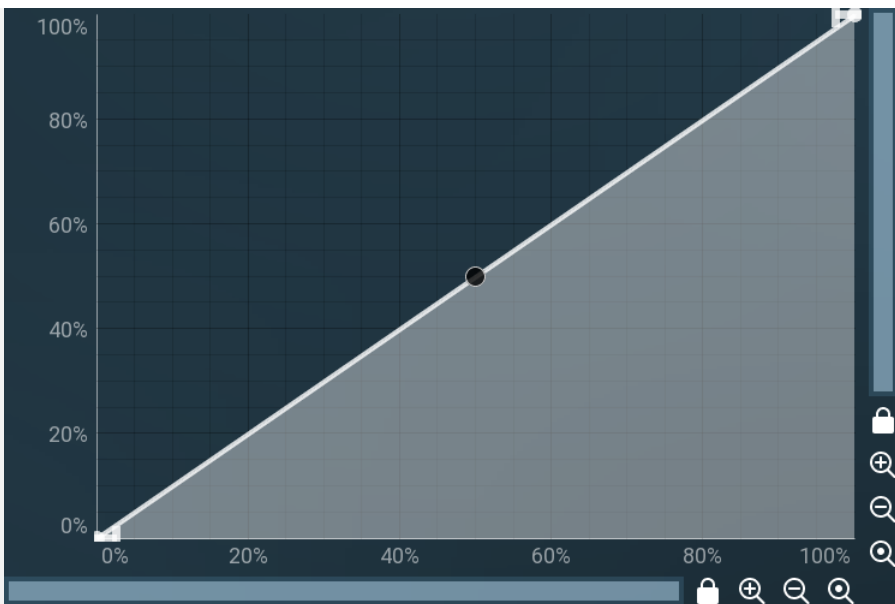**Right arrow**

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.

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.
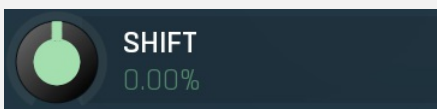
Shift

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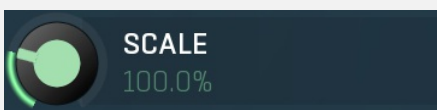
Scale

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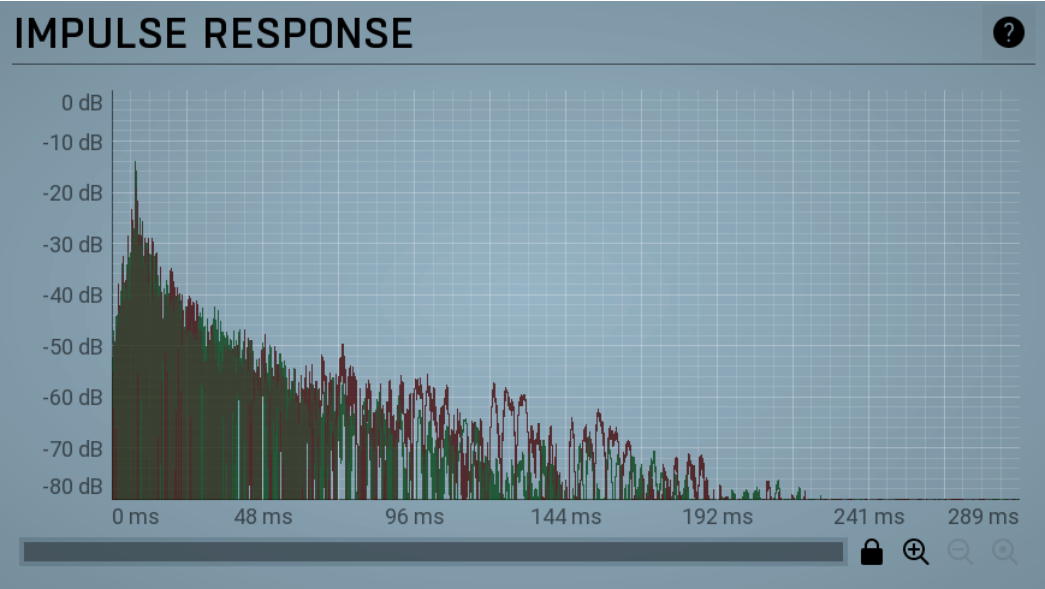
⏻ Enable **Enable**
Enable button enables the link-grid module. It is off by default to save CPU resources.

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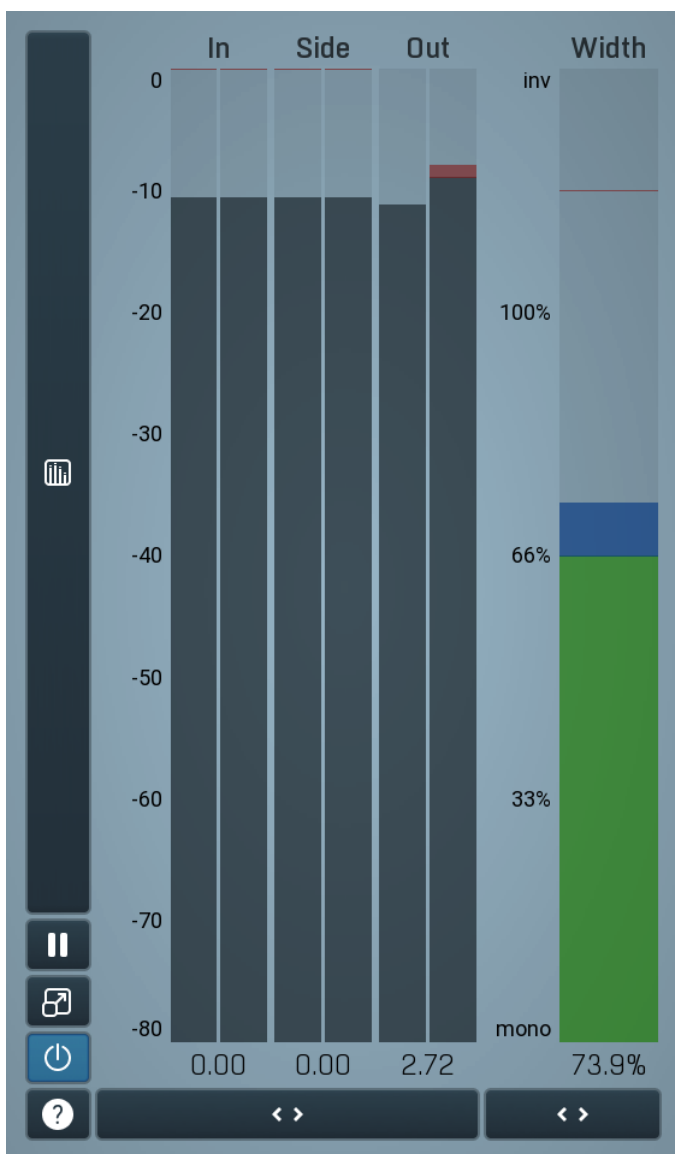

Impulse response graph
Impulse response graph displays the output impulse response.

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

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.
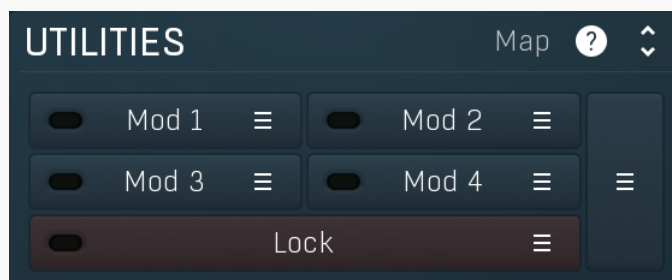
Collapse

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.