MMorph

MMorph allows seamless morphing from one signal to another. Send one signal to the main input and another to the side chain, MMorph then allows you to transition frequency characteristics smoothly between the two. MMorph also includes a host of pre and post morphing processes to prepare and optimize the signals, allowing you to tailor the morphed output beyond your imagination.

When using MMorph note that the processing is asymmetric - it does matter which signal is A and which is B, so if the results do not sound too good, just use the **Swap A and B** to try the other routing. By default, the main input signal is A and the side-chain input is B. When processing vocals it is often useful to use the vocal as the side-chain (B by default). But it is always worth experimenting with both ways.

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.

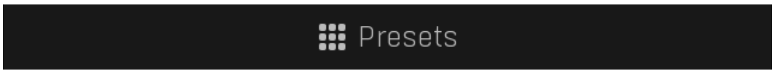
To the right of the controls are the meters or time-graphs for the plugin; the standard plugin Toolbar may be to the right of these or at the bottom of the plugin.

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

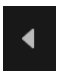
Presets button shows a window with all available presets. A preset can be loaded from the preset window by double-clicking on it, selecting via the buttons or by using your keyboard. You can also manage the directory structure, store new presets, replace existing ones etc. Presets are global, so a preset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

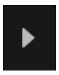
Files are named based on the name of the plugin like this: "{pluginname}.presets", so for example MAutopan.presets or MDynamics.presets. If the directory cannot be found on your computer for some reason, you can just search for the particular file.

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.*



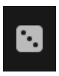
Left arrow

Left arrow button loads the previous preset.



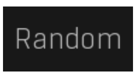
Right arrow

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.

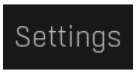
In addition, there are some mouse modifiers that assist this process. The smart randomization engine is used by default if no modifier keys are held.

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.

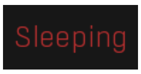
Global system settings contains some settings for all MeldaProduction plugins. Once you change any of them, restart your DAW if needed, and it will affect all MeldaProduction plugins.

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

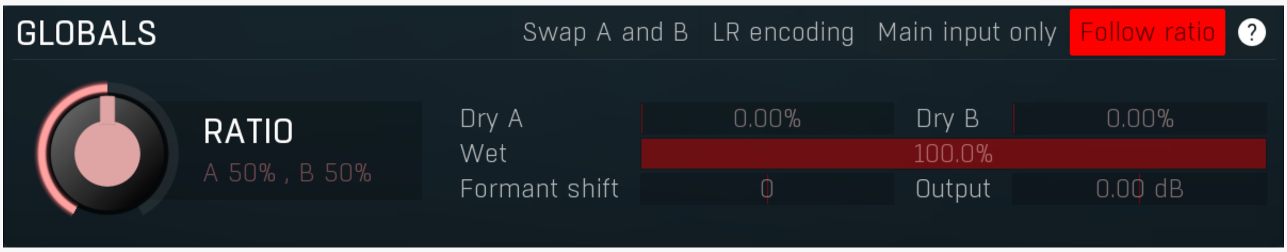
WWW button shows a menu with additional information about the plugin. You can check for updates, get easy access to support, MeldaProduction web page, video tutorials, Facebook/Twitter/YouTube channels and more.



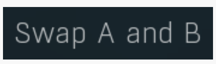
Sleep indicator

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.

Globals panel



Globals panel contains the most important parameters.



Swap A and B

Swap A and B swaps the A and B signals. A is taken from the main input and B from the side-chain by default, but when this switch is enabled, it is the other way around.

LR encoding

LR encoding exchanges the standard side-chain input for stereo input - the plugin starts morphing between the left and right channels. Signal A is the main input's left channel and Signal B is the main inputs right channel.

Main input only

Main input only makes the plugin ignore the sidechain and use main input for both signals A and B. In this scenario the plugin actually doesn't morph, since both signals are the same, but it can still extract the dominant features and you can use the additional parameters to process the sound. The plugin then becomes a powerful filter.

Follow ratio

Follow ratio switch determines whether the Ratio should affect other settings. For example, the **Release** parameter can strongly affect the output sound. With this option disabled the release processor remains the same regardless of the ratio setting, which might sound weird with values close to 0% or 100%.

Ratio

Ratio is the main parameter, which controls the ratio between the A and B signals. Unlike simple cross-fading, the plugin analyses both signals, extracts features that they have in common and produces a result somewhere between both signals depending on the ratio parameter. The morphing process is not symmetrical, which means that it makes a significant difference which signal is A and which is B. If the results sound weird or unusable or you just don't like it, try switching the **Swap A and B**, so that the A and B signals are swapped.
Range: A 100% , B 0% to A 0% , B 100%, default A 50% , B 50%

Dry A

Dry A defines the amount of the unprocessed input A signal in the output. Signal A comes from the main input, unless **Swap A and B** is enabled, in which case it is taken from the side-chain input.
Range: 0.00% to 100.0%, default 0.00%

Dry B

Dry B defines the amount of the unprocessed input B signal in the output. Signal B comes from the side-chain input, unless **Swap A and B** is enabled, in which case it is taken from the main input.
Range: 0.00% to 100.0%, default 0.00%

Wet

Wet controls the amount of the processed signal in the output.
Range: 0.00% to 100.0%, default 100.0%

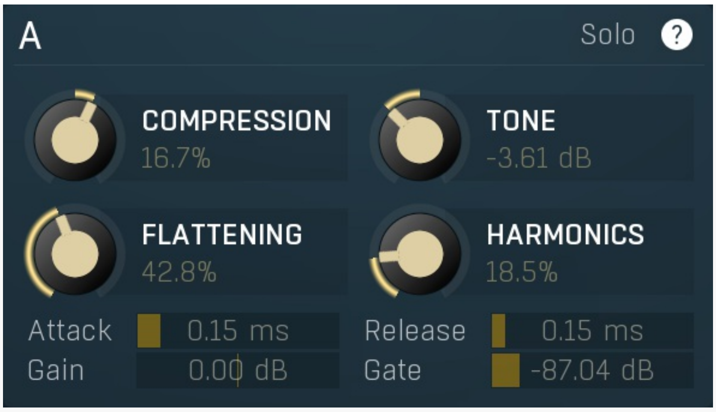
Formant shift

Formant shift lets you shift the pitch of the B signal resulting in formant shift of the output. That may be useful to make A and B similar in spectrum, so that the plugin can find similar features in both.
Range: -12.00 to +12.00, default 0

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

Panel A



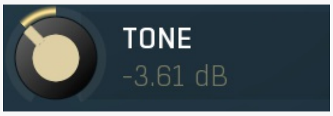
Panel A contains settings for signal A. This includes several processors that may be used to tailor your audio signals for the processing, or be used creatively. Since the morphing processing is not symmetrical, the settings for the A and B signals are slightly different, including the defaults.

Solo

Solo lets you audition just the A signal. You can do the same thing by setting wet and dry B to 0% and dry A to 100%, but this is just more convenient.

Compression

Compression controls the spectral compression/expansion. Despite its name, it doesn't actually control dynamics, but rather changes the proportions between frequencies. If the value is above 0%, spectral compression is performed, which means that the difference in levels between frequencies is lowered. In other words, it amplifies quiet frequencies and attenuates loud frequencies. If the value is below 0%, spectral expansion occurs and the effect is the opposite. This feature is often useful to make the audio signals more similar, as a result the later stages of the plugin can find similar frequencies and produce more useful output. For example, basses often have very little higher frequencies; vocals on the other hand have almost no low frequencies. As a result it could be useful to perform spectral compression for the bass, so that the little high frequency content there (which exists in abundance in the vocal) is amplified and the plugin can find something to work with. Similarly it is often useful to perform spectral expansion on vocals, because their spectral content is usually very complex, making the output somewhat distorted; spectral expansion then might be useful to amplify the most distinctive features and hide the less relevant ones.
Range: -100.0% to 100.0%, default 25.0%

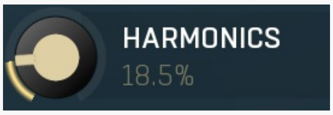


Tone

Tone controls the spectral slope, in other words a sort of linear high-shelf filter. Positive values amplify the high frequencies and vice versa. This filter is placed at the beginning of the processing chain, so you might reduce its effect using other parameters, but that's intentional, as you can use this parameter as a sort of input filter, to "prepare" the audio signal for the rest of the processing. For example, if the input audio has very strong high frequency content, the remaining parameters may amplify it even more making the output hard to listen to. You might then lower the tone, so that it is removed at the start of processing.
Range: -12.00 dB to +12.00 dB, default -6.00 dB

Flattening

Flattening controls the amount of spectral flattening, which essentially amplifies those quiet frequencies that are near to loud ones. It is similar to Compression, but flattening takes into account the surrounding frequencies; hence it sounds more natural and provides more of an overall spectral balance rather than trying to increase/decrease differences between frequencies.
Range: 0.00% to 100.0%, default 25.0%



Harmonics

Harmonics defines the depth of the harmonic generator, which can produce higher harmonics that don't actually exist in the signal. You can use it to increase the chance that there will be similarities between both input signals.
Range: 0.00% to 100.0%, default 0.00%

Attack

Attack controls the length of the attack stage of the level follower, which can smooth the signal in time and provide a sort of attack smoothing, which is especially useful for creative processing with rhythmical materials.
Range: 0 ms to 1000 ms, default 0 ms

Release

Release controls the length of the release stage of the level follower, which can smooth the signal in time and result in a sort of resonant reverberation, which may be useful for creative processing.

Range: 0 ms to 10000 ms, default 0 ms

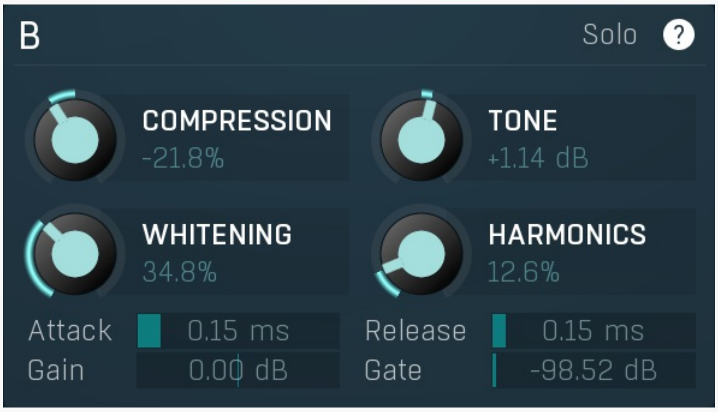
Gain

Gain defines the power modification applied to the input. You can use it to make sure the input signal has a reasonable level.
Range: -24.00 dB to +24.00 dB, default 0.00 dB

Gate

Gate controls the threshold for the gate, which you can use to remove any input below certain level.
Range: -100.00 dB to 0.00 dB, default -100.00 dB

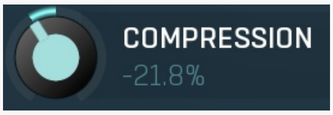
Panel B



Panel B contains settings for signal B. This includes several processors that may be used to tailor your audio signals for the processing, or to be used creatively. Since the morphing processing is not symmetrical, the settings for the A and B signals are slightly different, including the defaults.

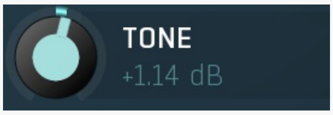
Solo

Solo lets you audition just the B signal. You can do the same thing by setting wet and dry A to 0% and dry B to 100%, but this is just more convenient.



Compression

Compression controls the spectral compression/expansion. Despite its name, it doesn't actually control dynamics, but rather changes the proportions between frequencies. If the value is above 0%, spectral compression is performed, which means that the difference in levels between frequencies is lowered. In other words, it amplifies quiet frequencies and attenuates loud frequencies. If the value is below 0%, spectral expansion occurs and the effect is the opposite. This feature is often useful to make the audio signals more similar, as a result the plugin can find similar frequencies and produce more useful output. For example, basses often have very little higher frequencies, vocals on the other hand have almost no low frequencies. As a result it could be useful to perform spectral compression for the bass, so that the little high frequency content, which exists in the vocal, is amplified and the plugin can find something to work with. Similarly it is often useful to perform spectral expansion on vocals, because their spectral content is usually very complex, making the output sort of distorted, spectral expansion then might be useful to amplify the most distinctive features and hide the less relevant ones.
Range: -100.0% to 100.0%, default -25.0%



Tone

Tone controls the spectral slope, in other words a sort of linear high-shelf filter. Positive values amplify the high frequencies and vice versa. This filter is placed at the beginning of the processing chain, so you might reduce its effect using other parameters, but that's intentional, as you can use this parameter as a sort of input filter, to "prepare" the audio signal for the rest of the processing. For example, if the input audio has very strong high frequency content, the remaining parameters may amplify it even more making the output hard to listen to. You might then lower the tone, so that it is removed at the start of processing.
Range: -12.00 dB to +12.00 dB, default 0.00 dB

Whitening

Whitening controls the amount of spectral whitening, which reduces the difference between frequencies in time. It is essentially similar to **Compression**, however it takes behaviour over time into account. Frequencies that occur only for a short period of time are amplified, which sort of improves the distinctive features of the signal.

Range: 0.00% to 100.0%, default 0.00%

Harmonics

Harmonics defines the depth of the harmonic generator, which can produce higher harmonics that don't actually exist in the signal. You can use it to increase the chance that there will be similarities between both input signals.
Range: 0.00% to 100.0%, default 0.00%

Attack

Attack controls the length of the attack stage of the level follower, which can smooth the signal in time and provide a sort of attack smoothing, which is especially useful for creative processing with rhythmical materials.
Range: 0 ms to 1000 ms, default 0 ms

Release

Release controls the length of the release stage of the level follower, which can smooth the signal in time and result in a sort of resonant reverberation, which may be useful for creative processing.
Range: 0 ms to 10000 ms, default 0 ms

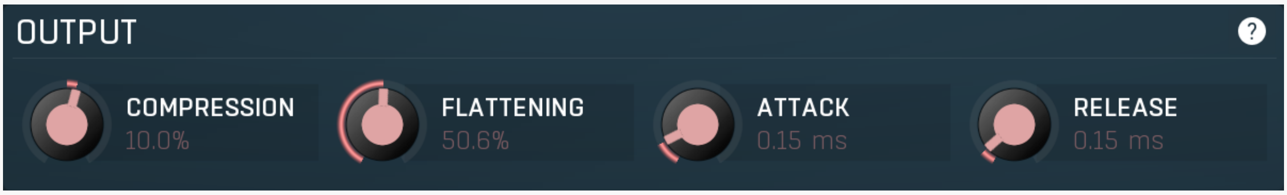
Gain

Gain defines the power modification applied to the input. You can use it to make sure the input signal has a reasonable level.
Range: -24.00 dB to +24.00 dB, default 0.00 dB

Gate

Gate controls the threshold for the gate, which you can use to remove any input below certain level.
Range: -100.00 dB to 0.00 dB, default -100.00 dB

Output panel



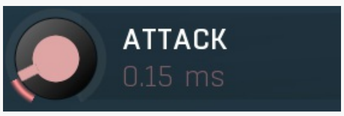
Output panel contains settings for the output signal. This includes several processors that may be used to tailor the output signal for the processing, or to be used creatively.

Compression

Compression controls the spectral compression/expansion performed on the output signal. Despite its name it doesn't actually control dynamics, but rather changes proportions between frequencies. If the value is above 0%, spectral compression is performed, which means that the difference in levels between frequencies is lowered. In other words, it amplifies quiet frequencies and attenuates the loud frequencies. If the value is below 0%, spectral expansion occurs and the effect is opposite. This feature is often useful to make the audio signals more similar, as a result the later stages of the plugin can find similar frequencies and produce more useful output. For example, basses often have very little higher frequencies; vocals on the other hand have almost no low frequencies. As a result it could be useful to perform spectral compression for the bass, so that the little high frequency content there (which exists in abundance in the vocal) is amplified and the plugin can find something to work with. Similarly it is often useful to perform spectral expansion on vocals, because their spectral content is usually very complex, making the output somewhat distorted; spectral expansion then might be useful to amplify the most distinctive features and hide the less relevant ones.
Range: -100.0% to 100.0%, default 25.0%

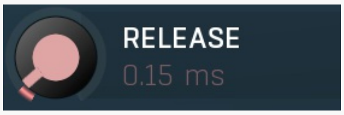
Flattening

Flattening controls the amount of spectral flattening, which essentially amplifies those quiet frequencies that are near loud ones. It might be similar to **Compression**, but flattening takes into account the surrounding frequencies; hence it sounds more natural and provides more of an overall spectral balance rather than trying to increase/decrease the differences between frequencies.
Range: 0.00% to 100.0%, default 50.0%



Attack

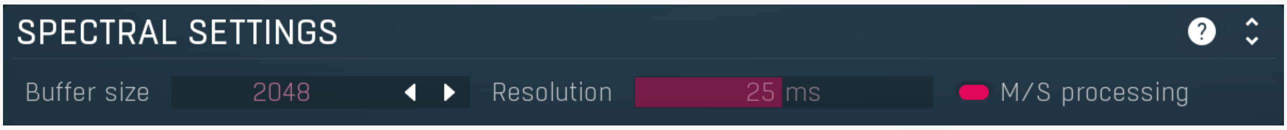
Attack controls the length of the attack stage of the level follower, which can smooth the signal in time and provide sort of attack smoothing which is especially useful for creative processing with rhythmical materials.
Range: 0 ms to 1000 ms, default 0 ms



Release

Release controls the length of the release stage of the level follower, which can smooth the signal in time and result in a sort of resonant reverberation, which may be useful for creative processing.
Range: 0 ms to 10000 ms, default 0 ms

Spectral settings panel



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

Buffer size

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

Resolution

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

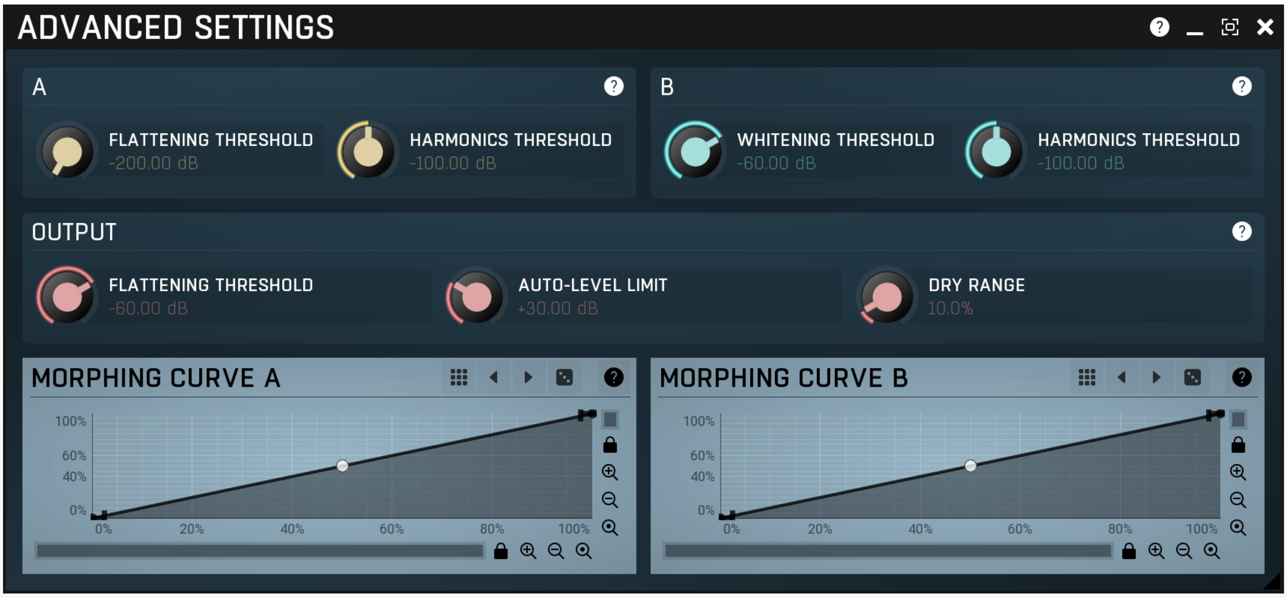
M/S processing

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

Advanced settings

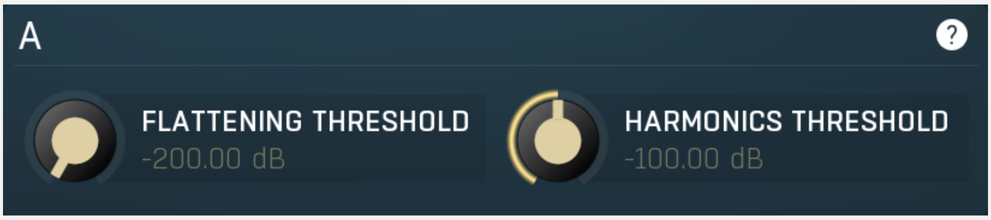
Advanced settings button displays additional settings.

MMorphAdvanced

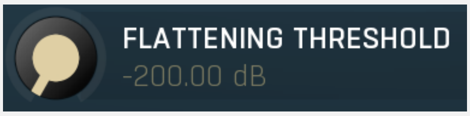


Advanced settings window contains more advanced settings, which are used less often and so are intentionally not shown on the main plugin editor.

Panel A

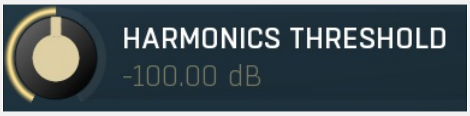


Panel A contains settings for signal A. This includes several processors that may be used to tailor your audio signals for the processing, or be used creatively. Since the morphing processing is not symmetrical, the settings for the A and B signals are slightly different, including the defaults.



Flattening threshold

Flattening threshold defines the minimum level for a frequency to be flattened. The lower the threshold the more the quieter frequencies are amplified, which is fine for clean audio materials, but might be problematic for many recorded sounds. If so, increase the flattening threshold.
Range: -200.00 dB to 0.00 dB, default -200.00 dB

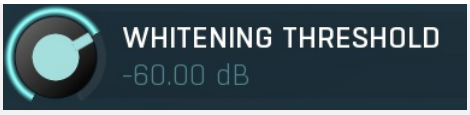


Harmonics threshold

Harmonics threshold controls the minimum level for a frequency to have harmonics generated. Setting this too low may produce too much distortion; it might be used creatively though.
Range: -200.00 dB to 0.00 dB, default -100.00 dB

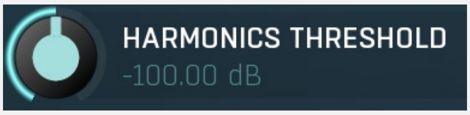
Panel B

Panel B contains settings for signal B. This includes several processors that may be used to tailor your audio signals for the processing, or to be used creatively. Since the morphing processing is not symmetrical, the settings for the A and B signals are slightly different, including the defaults.



Whitening threshold

Whitening threshold controls the minimum level for a frequency be whitened. Setting this too low may amplify irrelevant frequencies, it might be used creatively though.
Range: -200.00 dB to 0.00 dB, default -60.00 dB



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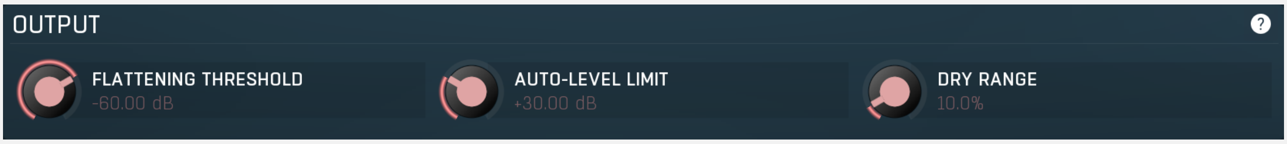
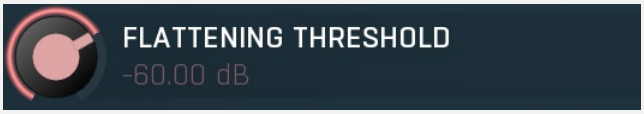
Range: -200.00 dB to 0.00 dB, default -100.00 dB

Output panel

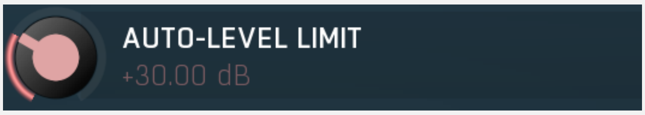


Output panel contains settings for the output signal. This includes several processors that may be used to tailor the output signal for the processing, or to be used creatively.



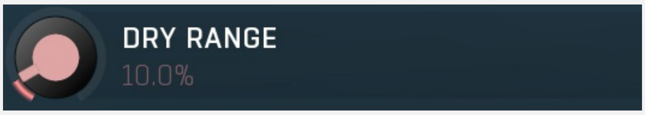
Flattening threshold

Flattening threshold defines the minimum level for a frequency to be flattened. The lower the threshold the more the quieter frequencies are amplified, which is fine for clean audio materials, but might be problematic for many recorded sounds. If so, increase the flattening threshold.
Range: -200.00 dB to 0.00 dB, default -60.00 dB



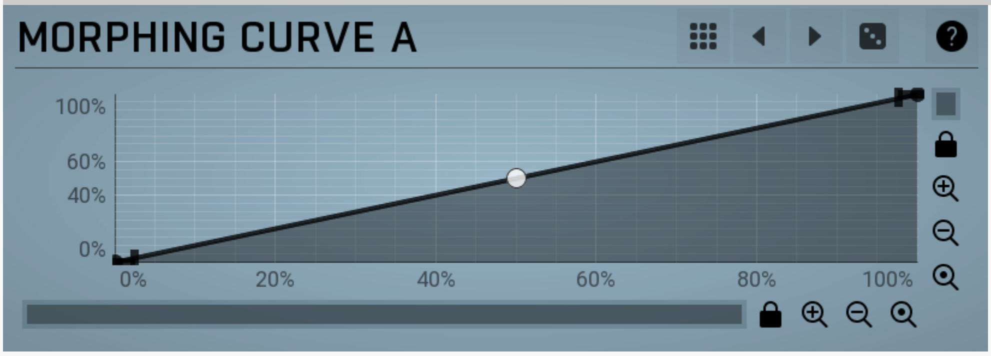
Auto-level limit

Auto-level limit controls the maximum amplification that would be performed to make the output level similar to the input. It is a necessary process, since the plugin may alter the audio level significantly. However also note that too high a limit may cause lower dynamics of the audio, which may or may not be wanted.
Range: 0.00 dB to +100.00 dB, default +30.00 dB



Dry range

Dry range controls how far the ratio must be from 0% for the A signal to disappear and similarly how far from 100% B disappears.
Range: 0.00% to 100.0%, default 10.0%



Morphing curve graph

Morphing curve graph defines the spectral smoothing size for the particular audio signal. In signal A it controls the spectral flattening, in signal B it is used by the algorithm itself and it is a key property of the morphing. By default it is a simple linear 0,0 -> 100,100 graph, but you may use it creatively. The X axis contains the ratio value and the Y axis is the actual smoothing. There is no further logical explanation; you really need to experiment in order to use it.

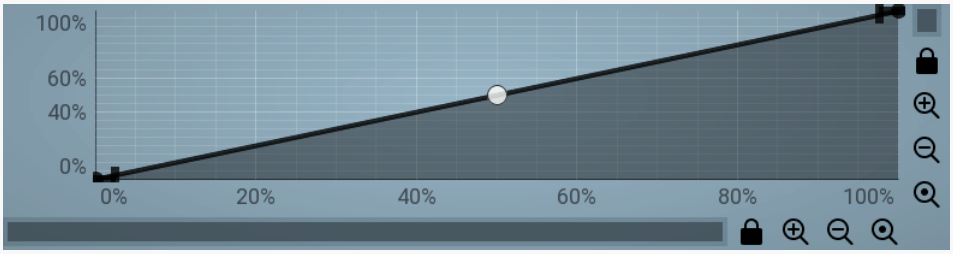
Presets

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.

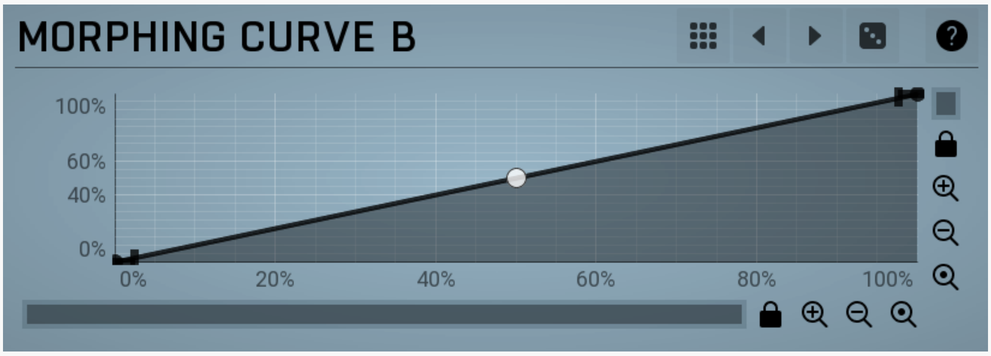


EnvelopeEditorGraph

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.



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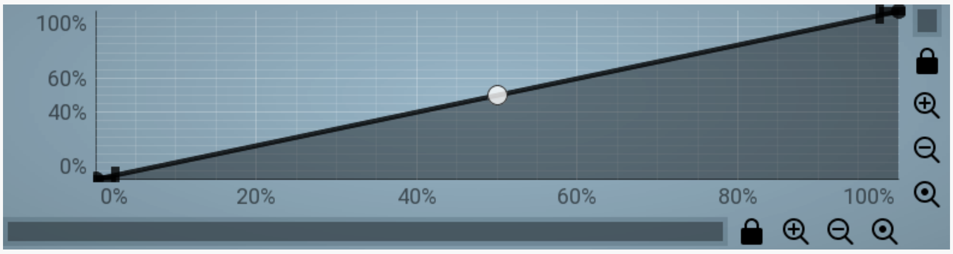

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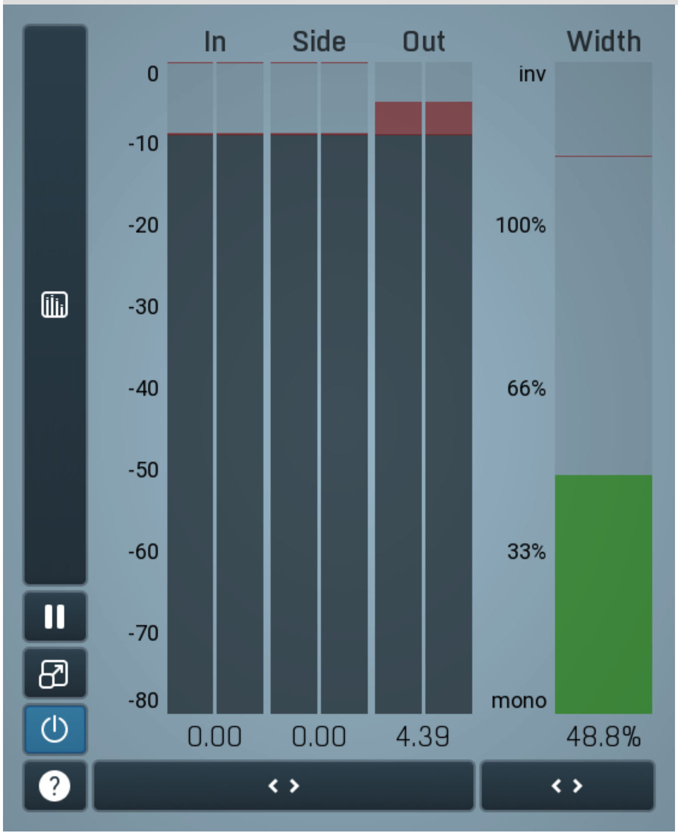

Global meter view

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

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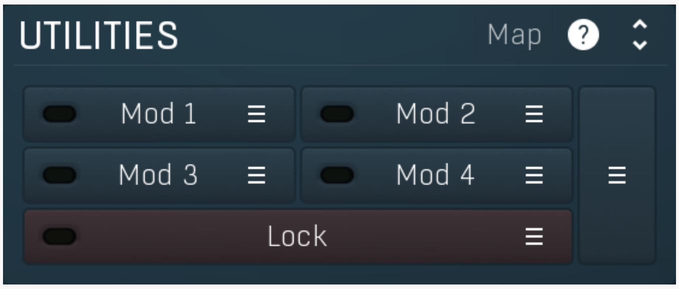
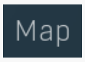
Utilities





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.

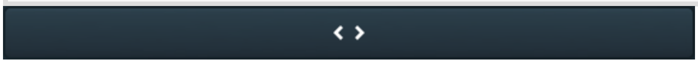
For example, to associate a frequency slider and make a modulator control it from 100Hz to 1KHz, just enable the smart learn mode, click the slider then move it from 100Hz to 1KHz (you can also edit the range later in the modulator window too). Then disable the learning mode by clicking on the button.

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.