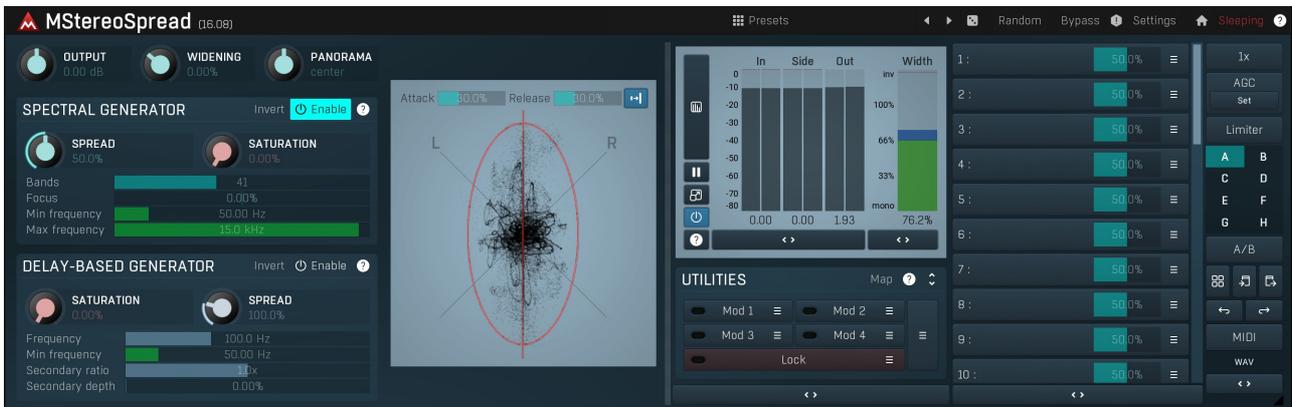# MStereoSpread

MStereoSpread is a unique artificial stereo generator. It implements 2 algorithms, the first a high-end spectral algorithm, which is preferred and provides unbeatable audio quality, and the second a fairly common algorithm based on comb-filtering. You can use either one or both of them at the same time. The plugin also provides additional controls such as a stereo width knob, and a stereo field analyzer.

Basics of stereo field psychoacoustics

A healthy human possesses two ears and since birth, our brain has been using them to learn to identify different sounds and their origins. After years of practice we are able to identify sounds inside whole mixes and estimate the direction and distance to the sound origin with surprising accuracy. We are even able to guess the size and type of the room we are in. All of this done using just our two ears, both of which are essentially just one dimensional.

The invention of stereo audio gives a mixing engineer unprecedented possibilities, which are way beyond the basic panorama. The fact that our brain can extract different parts of a mix gives us another two dimensions which we can use to make a mix "clear". A mixing engineer's task is to make the listener understand the mix easily and let them focus on what is important without actually thinking about it and analysing the audio. Here we'll discuss the basic clues that our brain uses to identify the origin and explain why and how we can take advantage of these clues.

Left or right

When an audio source is on the left side, it is natural that you will hear it first using your left ear, then you will probably get some reflections and resonances from your skull to your right ear. Note that this makes perfect sense, yet most people still use a panorama control to make a sound come from the left side! In the real world that's something that would only happen in an anechoic chamber with a dampening device attached to your head! It may sound acceptable when using monitors, because the environment actually creates the reflections for you and the skull resonances are again present, but it just doesn't sound natural with headphones. However the majority of us use headphones every day so it is essential to learn to simulate direction using more advanced techniques.

To summarise: Sounds coming from our left side occur first in our left ear and then get slightly delayed, lowered/attenuated, filtered in our right ear and are followed by a set of reflections. It is usually not necessary to simulate the reflections and the filtering cannot actually be simulated, because every skull is different. So all that is needed is to delay the right channel and attenuate it slightly. This is usually called a Haas delay.

Direction

Things get more complicated when the audio is not coming from completely left or right, but from any angle around you. This is where the ear filter comes in. When you look at an ear, you may wonder why it has such a weird shape. The interesting thing about this is that it actually looks different from every angle. And it also changes the sound from every direction differently . We can actually simulate this, by placing a tiny microphone into our ear, recording impulse responses from different angles and then use these to convolve any audio with it to make it sound like "it originated from there". Unfortunately tests have shown that each person has different ears and after the years the brain has taken to adjust to them, using a 'different person's ears' doesn't work very well.

So how can we simulate a sound coming from say 45 degrees to your left, and 60 degrees above you? Well, it seems that we cannot do that effectively using just 2 channels. After all it wouldn't work with monitors anyway, only in headphones. Luckily for us, we don't really need vertical placement, and a Haas delay seems to be enough to make a sound originate from in front of us from any horizontal angle.

Distance - how to make a sound appear further away

A sense of distance is much more useful and interesting and it can even be partly applied to monophonic audio too. First, is the air absorption of higher frequencies - sounds originating from a longer distance have substantially lower levels of higher frequencies, because

sound traveling is more energetically demanding for higher frequencies. Secondly, the human brain uses reflections from surrounding objects. The further away the sound source is, the more reflections there are likely to be. It is usually called the ratio between the direct signal and reverberation. Both of these effects can be used in mono.

Now we can finally get to the concept of the stereo image. When the audio source is very far away in front of us, both ears would probably intercept almost the same sound, the difference would only be in the surrounding environment, e.g. a wall on your left side would probably be very audible. This level of detail however is not usually relevant for mixing, so we can simplify it and say that in this case the sound would basically be monophonic. **A rule of thumb - distant objects are nearly monophonic.**

Moving a sound further away is therefore pretty simple - add some reverb, attenuate high frequencies using an equalizer, and collapse the stereo image using the **Widening** controller for example.

Distance - how to make a sound appear closer

When a sound source is very close to your head it is likely that both ears will intercept the sound more or less at the same time, there will be no loss of high frequencies and any reflections would be negligible compared to the direct signal. This may lead to the conclusion, that a monophonic audio track positioned in the center is as close as it can be (as long as it doesn't have too many reflections recorded). But this is not really true.

By making the audio in both ears uncorrelated (or just different) you can fool the brain into thinking that the audio is closer than is physically possible, thus making the sound actually appear inside your head. **A rule of thumb - the closer the object is, the more stereophonic it is.**

This effect is now used extensively in modern tight mixes. The main challenge is how to make it sound natural as this doesn't actually occur in nature. The sound in both ears must be similar enough for the brain to put both pieces together, otherwise they would sound like 2 separate audio signals and that can be very tiring on the ears. But these 2 channels must also be different enough to cause this proximity effect.

What you need is an artificial stereo width. Reverbs are often employed for width, but as you now know they also make the audio sound further away and create a sense of space, which cannot occur if the sound source is "in your head". Another way is to use a chorus or a similar modulation effect, but these modify the sound in a way that is not transparent enough and in most cases they also psycho acoustically shift the sound source further away.

And that's where **MStereoSpread** comes in.

Why do we care about sound origin?

When you listen to a mix that is crowded, it is very fatiguing. It often sounds like all the instruments are at the same location, or everyone is in a different space, or you just cannot say anything about it, because the whole thing just doesn't make sense intuitively. This is often hard to judge, especially for untrained ears, and engineers start playing with volumes and panoramas. However the point is - a good mix must make the listener feel like they are in a specific place and all of the instruments are on stage. That's a natural way for our brain to separate instruments, because in real life it is impossible to have multiple instruments existing at one place.

Good mixing engineers will employ one or more reverbs to make some sense of space and glue the tracks together. They will also apply several other techniques, some of which we have already described, to place each track into a specific spot on stage, whether it is at the end, right next to the listener, or inside their head. And what does the listener focus on? The sound that is closest, of course.

Simply put, the least important tracks should appear far away, hence monophonic, highly reverberated and without many high frequencies. Most important tracks are the exact opposite - as close as possible, so stereophonic, minimum reverberation and with as much original clarity in the high frequencies as possible.

What is artificial stereo?

MStereoSpread modifies both channels, left and right, differently so that when summed together they recreate the original signal, just the phase is altered but in a very natural way. This way the brain can put both channels together, yet it simulates the proximity effect making the audio materials sound highly stereophonic, close, but still natural.

The plugin uses 2 algorithms. The first, a delay-based generator which is a well-known concept used in several other plugins and is provided here only for comparison as it has several flaws. The second, a spectral generator that provides unbelievable clarity and naturality, is therefore always recommended.

⣿ Presets

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.*
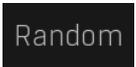
Left arrow
Left arrow button loads the previous preset.

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.

Gain

Gain defines the power modification applied to the output signal. Stereo expansion algorithms in general work by altering the side-signal, which however means they can modify the output level. You can use gain to compensate for this change in loudness caused by the plugin. However please note that the increase in loudness is partly psycho-acoustic, because expanding the stereo signal brings the source virtually closer to the listener, which can make the source appear more apparent and louder.
Range: -24.00 dB to +24.00 dB, default 0.00 dB

Widening

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

Widening converts the audio into its mid (mono) and side channels, leaving the mid intact and applying a gain to the side channel, then converts the signal back to left and right channels. As a result the stereo image becomes wider (for widening above 0%) or narrower (for widening below 0%). This method of widening the stereo image may initially sound pleasing, however it can quickly become fatiguing on the ear and often sounds unnatural, especially for larger amounts of widening. Use this parameter to control the existing stereo field and as a special effect. Use it to increase width only with caution.
Range: Mono to 200.0%, default 0.00%

Panorama

Panorama defines the panorama applied to the output signal. You may use it to compensate for the signal being psycho-acoustically out of the center. However please note that the point of stereo expansion is to create a signal that is highly stereophonic and appears closer to the listener, so it is kind of contradictory to apply high amounts of panorama to it. Anyway since both stereo expansion algorithms are fully mono compatible it is perfectly safe to use a panorama, whether using this parameter or in your host or any further plugin.
Range: 100% left to 100% right, default center

Spectral generator panel

Spectral generator panel configures the first essential part of the stereo spreading processor. Spectral stereo generator is suitable for all audio materials therefore it is recommended to use this one mainly. The algorithm is fully mono compatible and doesn't use any delay-based processing. It is a very complicated IIR filter, whose only is a phase alteration.
The basic idea behind spectral stereo expansion is to remove some frequencies from the left channel and others from the right channel in such a way, that the mono output will give the original signal again. The main difficulty is to make this approach sound natural and to avoid moving the stereo position out of the center. The plugin implements a unique algorithm which provides very high transparency, very natural sound and it keeps the stereo location in the center as much as possible.
The main problem with spectral stereo generation is that since some frequencies are kept in one ear and the rest of them in the other one, the fundamental frequency of monophonic (or even polyphonic) instruments may be heard only in one ear making it appear to be on the left or right side. The plugin fights with this by spreading the harmonics differently, which psycho-acoustically keeps the stereo

image in the center as much as possible.

Invert **Invert**

Invert button inverts the stereo-expansion signals for left & right channels, so the frequencies prevailing in the left channel will now be in the right channel instead and vice versa. This is very useful when expanding multiple tracks, especially with similar settings. In this case the same frequencies would be left in each channel from both tracks. By inverting one of them you can psycho-acoustically free some space in the mix, because each channel will contain both tracks, but the frequencies won't be overlapping, so you would actually have multiple close tracks, which would normally collide, but they are not colliding anymore and the brain is still able decode each track from the sound received in both ears.

SPREAD 50.0% **Spread**

Spread defines the amount of stereo generated by the algorithm. With 0% the original signal is kept intact. By increasing this parameter you get more stereo effect, however be advised not to overuse it, as too much stereo may be fatiguing and unnatural as it Doesn't occur in the natural world.
Range: 0.00% to 100.0%, default 50.0%

SATURATION 0.00% **Saturation**

Saturation defines the amount of saturation applied to the stereo expanded signal. Saturation naturally generates higher harmonics, which makes the signal richer and exciting. As a secondary result, this may cause the stereo field to appear even wider. However be cautious with this, as saturation is technically a form of distortion.
Range: 0.00% to 100.0%, default 0.00%

Bands **41** **Bands**

Bands parameter defines the number of filters used by the generator. You can imagine that the spectrum is divided into several blocks, the number is defined by this parameter, and some blocks are in the left channel, the others in the right channel.
A low number of bands provides a lower CPU consumption and possibly high stereo spreading, but also less natural results. Since large parts of the spectrum are moved to be heard in one ear and not in the other, the brain is not able to decode the signal's location anymore making it appear too wide. In extreme cases both channels can get so uncorrelated that they sound like different tracks. Be very cautious as this effect sounds pleasant at first, but can be very fatiguing, since the human brain is not able to understand the audio anymore.
A high number of bands requires more CPU and produces much more natural results. By dividing the spectrum in many smaller blocks you ensure that signal in each ear will make sense on its own and by understanding each side, the brain will also be able to correlate the sounds both ears together. This gives much more natural and pleasant results in most cases. However for a very high number of bands you may experience too high amounts of phase shifting. This is usually inaudible for higher frequencies and is generally irrelevant for many audio materials, such as pads, organs and other steady state signals. However it may be a problem for transient materials, such as drums. In this case you can fix potential issues, which usually happen in very low frequencies, using the **Min frequency** parameter.
In general it is recommended to keep the number of bands higher, as the natural quality is a key for a good mix. Keep it higher, depending on your CPU consumption, and if there are phasing problems, fix them by lowering this value or using the **Min frequency** parameter.
Range: 2 to 100, default 20

Focus **0.00%** **Focus**

Focus controls the distribution of the bands over the spectrum. Values lower than 0% makes the processor expand mainly the lower frequencies. Values higher than 0% make the processor expand mainly the higher frequencies. Since the lower frequency expansion is often dangerous and phase issues can arise, it is natural that expansion should rather be focussed on higher frequencies. You may however choose differently for your audio material. This parameter basically controls the sound character and generally there are no restrictions when it comes to audio quality.
Range: -100.0% to 100.0%, default 0.00%

Min frequency **50.00 Hz** **Min frequency**

Min frequency defines the minimal frequency used for the stereo expansion. You can use this to get rid of widening and phasiness in the low part of spectrum.
Human brains are adjusted to resolve phase below about 1kHz. Frequencies above are just too quick for the brain to catch the time differences accurately. This has the disadvantage, that big changes in phase in lower frequencies can cause dispersion, especially for transient signals. For example acoustic bass drums may sound almost synthetic when passed through the stereo expansion algorithm with a high number of bands. By setting Min frequency to say 100Hz you can usually get rid of these artifacts. This is only relevant if you are looking for the most natural results of course.
It is recommended to set Max frequency always higher than Min frequency to make effect processing sense.
Range: 20.00 Hz to 20.0 kHz, default 50.00 Hz

Max frequency **15.0 kHz** **Max frequency**

Max frequency defines the maximal frequency used for the stereo expansion. You can use this parameter to get rid of widening and phasiness in the high part of spectrum.
Range: 20.00 Hz to 20.0 kHz, default 15.0 kHz

Delay-based generator panel

DELAY-BASED GENERATOR   Invert ⏻ Enable   ❓

SATURATION   0.00%
SPREAD   100.0%

Frequency	100.0 Hz
Min frequency	50.00 Hz
Secondary ratio	1.0x
Secondary depth	0.00%

Delay-based generator panel configures the second part of the stereo spreading processor. Delay-based stereo generating is a well-known approach, which uses so-called comb filtering and is used in several audio products because of its simplicity and efficiency. It unfortunately has several negative side-effects, which will be discussed below. It is fully mono compatible.

The idea is to delay the mid channel of the input signal by a small amount, typically under 50ms, and mixing it with the left and right channels with different phase. It effectively creates a series of harmonically spaced notches, complementary in both channels. In other words, it again removes certain frequencies from the left channel and the remaining ones from the right channel.

The first and major disadvantage is that this method basically creates an echo of the original signal. Despite its effect of creating the harmonically spaced notches, it is still just an echo. For high delays you can hear a distinct echo of the original signal, which makes it usable only for steady-state signals and not, for example, for drums for example. For low delays a high degree of coloration occurs. Also, since it is an echo, it can interfere with room acoustics and used reverbs, and can create a sense of depth on its own, which is usually not desirable.

The second disadvantage is the harmonic spacing of the notches, which usually makes the sound appear on one side instead of in the center. The sound of most pitched instruments can be divided into 3 parts - fundamental frequency, harmonics and timbre. The fundamental frequency is the base frequency of the instrument and harmonics are its multiples. For example, if you play A4 being 440Hz, most instruments will also create 880Hz, 1320Hz, 1760Hz etc. Unfortunately when you tune the delay-based stereo expander to 220Hz for example, the notches will be created the same way - at 220Hz, 440Hz, 880Hz etc. This means that the whole fundamental with all its harmonics will be present only in say the left channel. This will make the listener feel like the sound originated from the left side. Then, when the player starts playing say C4 instead, the sound will move to the right channel. As a result, pitched instruments usually move widely in the stereo field, which in effect starts sounding chaotic, difficult for human brain to comprehend and eventually fatiguing.

Considering these disadvantages the delay-based processor should be used as complementary, and the spectral processor should be preferred.

Invert **Invert**

Invert button inverts the stereo-expansion signals for the left & right channels, so the frequencies prevailing in the left channel will now be in the right channel instead and vice versa.

SATURATION   0.00%   **Saturation**

Saturation defines amount of saturation applied to the stereo expanded signal. Saturation naturally generates higher harmonics, which makes the signal richer and exciting. As a secondary result, this may cause the stereo field appear even wider. However be cautions with this, as saturation is technically a form of distortion.

Range: 0.00% to 100.0%, default 0.00%

SPREAD   100.0%   **Spread**

Spread defines the amount of stereo generated by the algorithm. With 0% the original signal is kept intact. By increasing this parameter you get more stereo effect, however be advised not to overuse it, as too much stereo may be fatiguing and unnatural as it Doesn't occur in the natural world.

Range: 0.00% to 400.0%, default 100.0%

Frequency   100.0 Hz   **Frequency**

Frequency controls the delay size. It is specified as frequency as it lets you understand potential artifacts better. For example, when this is set to 100Hz, the delay size becomes 10ms (1 second / 100 Hz). As a result, the notches are created at 100Hz, 200Hz, 300Hz etc. For the most natural results, you will want to set this value to as low a frequency as possible, because the more notches there are, the more evenly the spectrum is spread between the left and right channels. For example, 100Hz creates a notch every 100Hz, but 20Hz creates a notch every 20Hz. Unfortunately 20Hz means 50ms delay, which is a clearly audible echo, so it probably won't be usable for most audio materials. Your goal will usually be to make this value as low as possible before the echo becomes audible. Please note that before the echo becomes obvious, you may first hear some kind of "space", which is usually not wanted either.

Range: 20.00 Hz to 2000 Hz, default 100.0 Hz

Min frequency   50.00 Hz   **Min frequency**

Min frequency defines the minimal frequency used for the stereo expansion. You can use this to get rid of widening and phasiness in the low part of spectrum. By setting Min frequency to say 100Hz you can usually get rid of the artifacts, that occur in the low part of the spectrum.
Range: 20.00 Hz to 20.0 kHz, default 50.00 Hz

| Secondary ratio | 1.0x |

Secondary ratio

Secondary ratio controls the secondary delay size. In order to keep the audio material more stable in the center, the algorithm uses 2 delays. The first one is set by **Frequency** parameter. The second one is the same length by default, which means a ratio of 1x. A ratio of 2x makes the secondary delay twice as long as the first one, for example. Using this delay the algorithm can create a secondary echo with opposite polarity. In other words, what was in the left channel in the first echo will be in the right channel in the second echo and vice versa. This may help to keep the stereo field more centered and in fact it can also provide some additional width.
Range: 0.0x to 2.0x, default 1.0x

| Secondary depth | 0.00% |

Secondary depth

Secondary depth defines the amount of the secondary delay. By default this is 0%, which makes the secondary delay disabled. The reason for this is that 2 delays means 2 echoes, which may not be ideal. However increasing this value often makes the signal wider, more stable and centered.
Range: 0.00% to 100.0%, default 0.00%
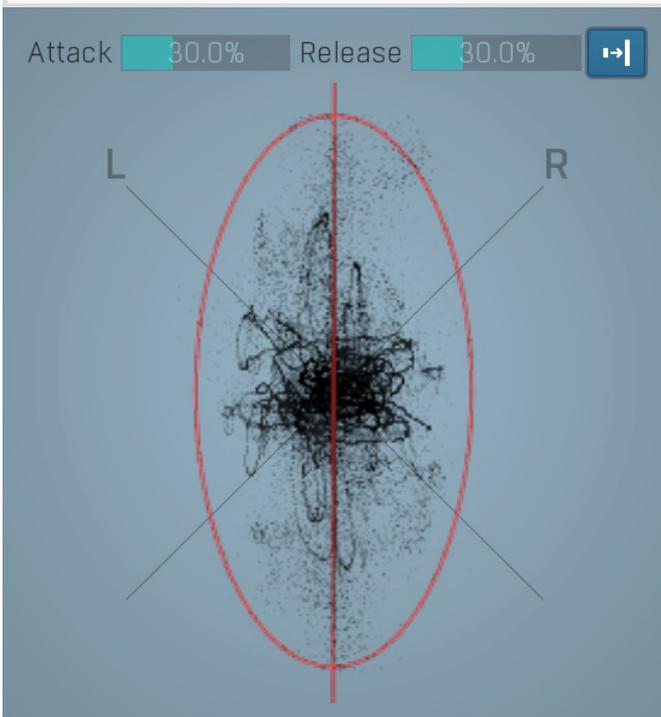
Phase scope

Phase scope shows the current phase distribution. An ideal recording is represented by a "not too thick but not too thin" vertical ellipse. But, trust your ears.
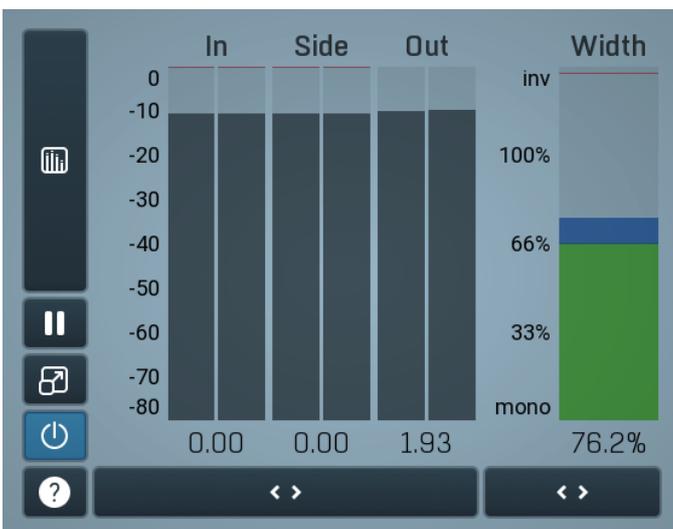
Global meter view

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.
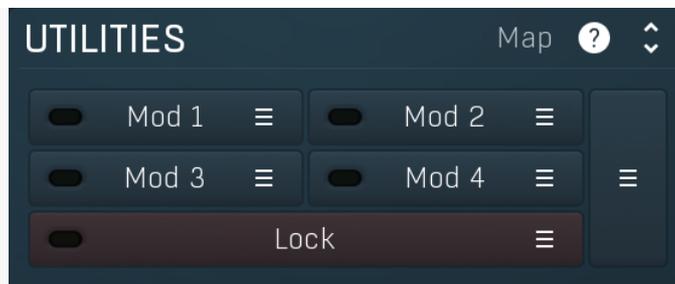
Collapse

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

UTILITIES Map ?

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.