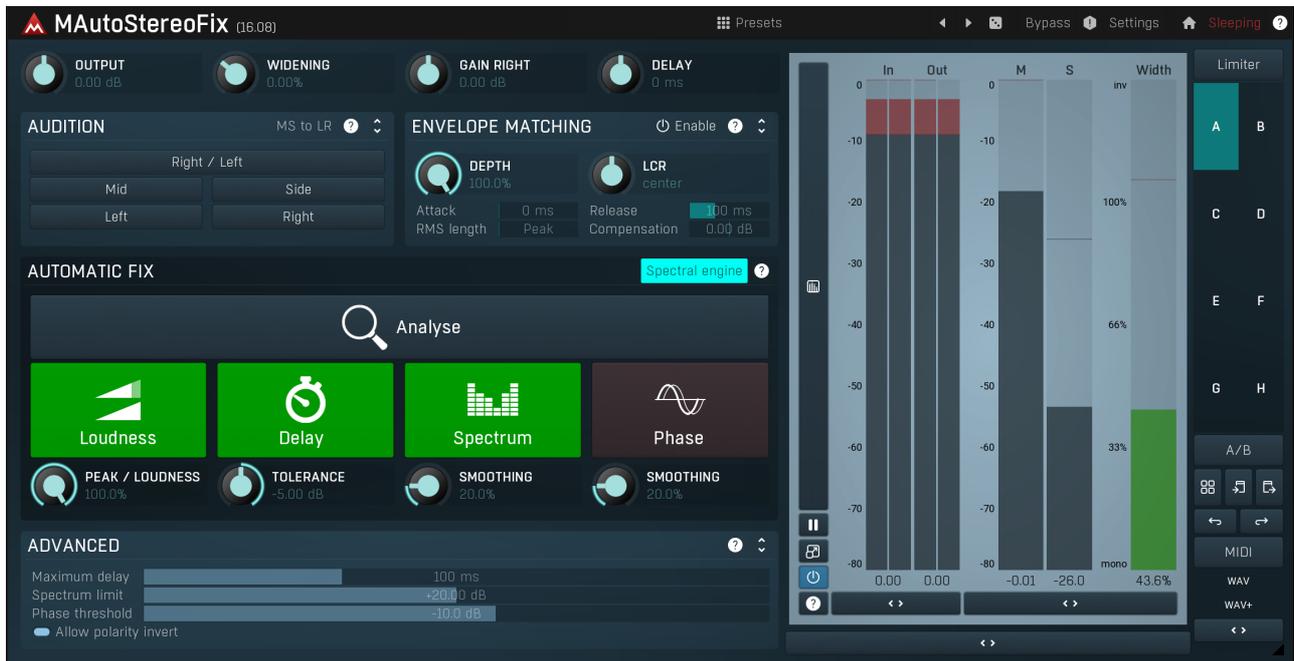


# MAutoStereoFix



## Overview

MAutoStereoFix is an essential tool for processing stereo tracks, whether these are drum overheads, rooms or any kind of stereo signals including master tracks. It lets you first analyze the audio and then apply several "fixes" to improve the stereo field and make it sound centered as it should be. We would recommend using headphones when analyzing the stereo field, since doing that with monitors is nearly impossible even in the best mixing rooms.

When using 2 microphones to record a certain instrument (or instruments), there is nearly no chance that the stereo field will be balanced. That would be possible only for an infinitely small instrument recorded with a pair of identical microphones placed at the exact same distance from the instrument, and recorded using a perfectly flat interface with no phase alteration. None of these requirements are even remotely realistic.

Most engineers adjust manually the level between the 2 microphones and that's pretty much it. Unfortunately that's not enough to keep the stereo field balanced. For example, if one microphone is further away from the instrument (or its prominent part - snare drum in case of drum overheads for example), the sound it will record is going to be delayed and the track will psychoacoustically appear "sounding from the other side". Trying to fix that by increasing the volume of that microphone would then only make things worse and sadly that's the usual way to deal with this problem. Delaying one of the microphones would solve the problem, but manually searching for a perfect delay would take a tremendous amount of time. Even searching for a good volume balance takes a lot of time, adding delay to the equation is nearly impossible even for a skilled engineer.

It gets worse when it comes to spectrum. For example, one of the microphones may record a little less high frequency content due to a different high frequency response, location close to a dampening surface, or even the room acoustics. The instrument will then appear closer to the other side again, because high frequencies are absorbed by the air and our brains know that. Fixing that using a linear-phase equalizer would be even worse than setting up delays and volumes.

And the final stage is phase. Any of the devices in the recording chain can alter the phase, whether it is the microphone, AD converters and especially if you are using an analog deck. Fixing phase manually would probably mean using an equalizer featuring allpass filters and that would take forever.

The worst part is that a phase or spectrum difference would not only make the stereo field appear out of the center, it would also sound confusing, because the clues picked by the brain would just don't make sense. After all stereo recording is generally everything but natural.

**MAutoStereoFix solves all of these problems automatically.** It analyzes the audio and calculates the optimal levels, delays, polarity inversion, equalization and spectral phase correct, so that the stereo field becomes as centered as possible. Please note however that the plugin cannot magically fix poor recording conditions such as room acoustics or crazy microphone placement.

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.



### 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.

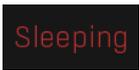
**Dry/Wet affects** determines, for Multiband plug-ins, which multiband parameters are affected by the Global dry/wet control.

**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

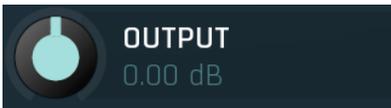
WWW button shows a menu with additional information about the plugin. You can check for updates, get easy access to support, MeldaProduction web page, video tutorials, Facebook/Twitter/YouTube channels and more.



### 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.



### Output gain

Output gain defines the power modification applied to the output signal.

Range: -24.00 dB to +24.00 dB, default 0.00 dB

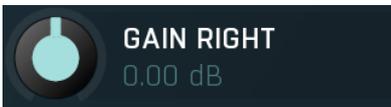


### Widening

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

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

Range: Mono to 200.0%, default 0.00%



### Gain right

Gain right defines the power modification applied to the right channel of the output signal. You can use it to manually balance the stereo field if needed.

Range: -10.00 dB to +10.00 dB, default 0.00 dB

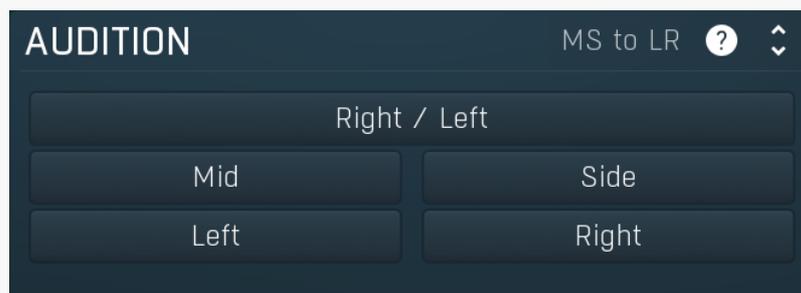


### Delay

Delay controls the delay applied on the left channel of the output signal. If it is negative, it is applied on the right channel instead. This way it in a way works as a panorama control. You can use it to manually balance the stereo field if needed. It is recommended to use this only if you have a creative reason to do that, or to adjust the delay between the channels if you feel that the automatic detection didn't produce ideal results.

Range: -25 ms to 25 ms, default 0 ms

## Audition panel



Audition panel contains multiple buttons, which let you audition specific part of the stereo signal. This can be used to understand the stereo field properly and check if the output signal is well balanced.

### MS to LR

MS to LR lets you perform Mid/Side to Left/Right encoding. In certain recording techniques this is a necessity as one of the microphones is design as mid and the other is side, however it turns out that even with classic recording techniques it can be advantageous to use this feature. If you turn this on, you will usually notice wider stereo image, which is however perfectly mono compatible, because the right channel (now serving as side signal) will disappear in mono. Another positive aspect of this is that there won't be any phase cancellation when playing the signal as mono. Be careful though that this increased width may often sound unnatural and fatiguing, therefore it is advised to check repeatedly whether this feature generates an actual improvement.

### Right / Left

### Right / Left

Right / Left swaps samples between the left and right channels of the output signal, so that you listen right / left instead of left / right. You can use it to check whether the stereo is ballanced. If it is not, you will hear a serious unnatural change of the stereo field when swapping the channels. In an ideal case you should hear the difference, but it shouldn't sound "strange" and unpleasant.

Mid

### Mid

Mid button lets you audition the mid signal only, that is the mono part of the output. You can use it to check for potential phase cancellations and to get your ears adjusted to the actual center.

Side

### Side

Side button lets you audition the side signal only, that is the stereo part of the output. You can use it to understand what exactly is the stereo signal created from and if there are some specific problems you need to address only in the side signal.

Left

### Left

Left button lets you audition the left channel only. It is often useful to switch between Left and Right channels, to understand what is in each of them, since in most cases these have been recorded with different microphones. It is pretty common to hear a specific difference in one of them, such as a certain room resonance, longer instrument sustain, different spectrum response etc. Despite the automatic fix should correct many of these problems, it cannot perform miracles by for example changing the sustain in one channel. And these are the things that may make the signal impossible to perfectly align to the center.

Right

### Right

Right button lets you audition the right channel only. It is often useful to switch between Left and Right channels, to understand what is in each of them, since in most cases these have been recorded with different microphones. It is pretty common to hear a specific difference in one of them, such as a certain room resonance, longer instrument sustain, different spectrum response etc. Despite the automatic fix should correct many of these problems, it cannot perform miracles by for example changing the sustain in one channel. And these are the things that may make the signal impossible to perfectly align to the center.

## Envelope matching panel



Envelope matching panel lets you automatically match the level envelope in one channel to the other. In most cases one microphone is picking up more ambience than the other for example. This can cause disturbing differences in audio level between the left and right channels, especially with percussive signals. Envelope matching provides a simple and efficient way to fix that by letting the plugin follow level of one channel with the other.



### Depth

Depth controls the amount of envelope matching. 0% means no processing at all, 100% means full level matching.

Range: 0.00% to 100.0%, default 100.0%



### LCR

LCR controls which channels will be affected.

Range: 100% left to 100% right, default center



### 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.

Range: 0 ms to 100 ms, default 0 ms

Release 100 ms 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.

Range: 0 ms to 10000 ms, default 100 ms

RMS length Peak 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.

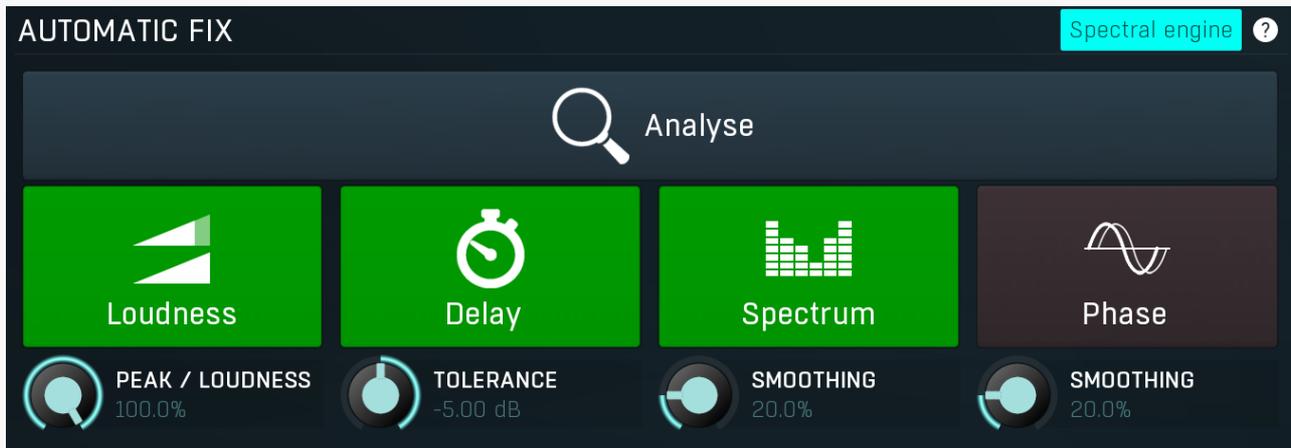
Range: Peak to 1000 ms, default Peak

Compensation 0.00 dB Compensation

Compensation defines the power modification applied to the right channel of the output signal when **Depth** is 100%. You can use it to manually balance the stereo field if needed, since the envelope matching can change it.

Range: -10.00 dB to +10.00 dB, default 0.00 dB

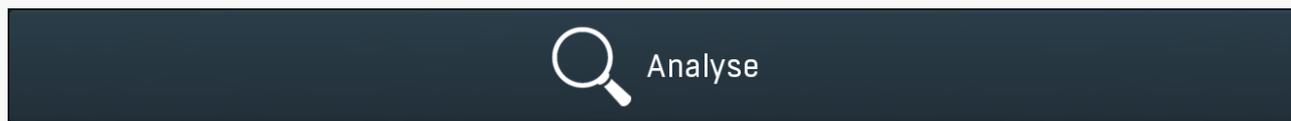
# Automatic fix panel



Automatic fix panel contains the core of the plugin - the automatic detection and fixing of various problems. First enable the playback and use the **Analyse** button to analyse your audio. Use the remaining buttons to enable or disable all each of the fixes that the plugin offers to improve the stereo field of your audio material.

## Spectral engine **Spectral engine**

Spectral engine switch enables the spectral engine; this is required for the **Spectrum fix** and **Phase fix**. Please note that this makes the plugin introduce latency.



### Analyse

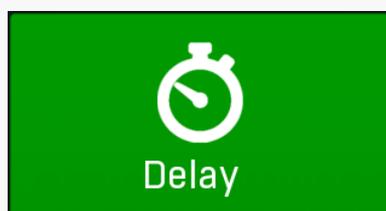
Analyse button starts or stops the analysis. First start playback in your host so that the plugin receives your audio material. Then press this button and a percentage display will appear and will count from 0% to 100%, where 100% is the reasonable minimum required for the actual analysis algorithm. After reaching 100% the counter stops but it is advised to keep the analysis going. In theory the longer the analysis, the more accurate the results will be, but in most cases it is more than enough to analyse say 10 to 20 seconds of your material. The algorithm will actually let you proceed at any point after 20%, but the results may be much less accurate. Press the button again afterwards to finish the analysis. It is essential to choose a representative part of your audio material for analysis. Different sections of the recording may produce very different analysis results. We would advise choosing a medium part, where all instruments being recorded are audible, but there is no extreme signal.

For instance it is quite common when recording a drumset that one of the microphones is placed close to a ride cymbal. When the drummer starts playing this cymbal, the sound from it is too prominent and is effectively killing everything else. Analysing such a part of the song may misguide the plugin to think that this microphone produces very specific loud "cymbal" sound. The results could actually be working very well for that part of the song where the cymbal is indeed playing, but the rest of the song would then sound bad, as the same fix would be applied through the song.



### Loudness

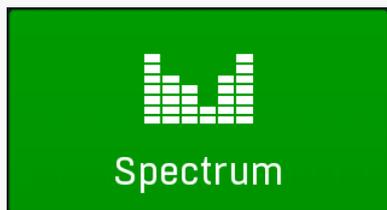
Loudness button activates the loudness fix. It ensures that the loudness of the left channel is the same as loudness of the right channel, which is an essential requirement for a good stereo image. Please note that the output peak meters may not show the same values for both channels, which is entirely correct, since the peak meters don't say too much about the actual perceived loudness. This feature only performs a gain on the right channel, so it is technically fully transparent.



### Delay

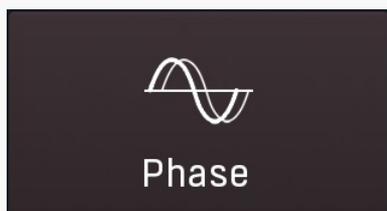
Delay button activates the delay and polarity fix. It can delay the left or right channels and invert polarity of one of them if needed, to ensure that the correlation of both channels is as high as possible. In less scientific words, it fixes any potential time error between the

channels. This feature only performs a delay and potential polarity inversion on left/right channels, so it is technically fully transparent. Time error occurs in most cases since the microphones are rarely exactly the same distance from the instruments being recorded. That is rarely even possible anyway due to the physical nature of the instruments and their surroundings. As a result the prominent signal (the loudest instrument usually) arrives at each microphone at a slightly different time and that shifts the stereo field (As perceived by your ears and brain) to the left or to the right. The plugin fixes that by shifting the channels in time, so that the prominent instrument sounds at the same moment in both channels and that makes the stereo field more centered.



### Spectrum

Spectrum button activates the spectrum fix. It takes the spectrum analysis of the left and right channels and applies any necessary linear-phase equalization on the right signal to match the left channel. This is necessary, since in most cases the spectrum recorded by one microphone is different from the other. It is caused by differences between the microphones, recording equipment, position and surroundings etc. Please note that since this feature is actually changing the spectrum, it is not fully transparent. It also introduces latency.



### Phase

Phase button activates the phase fix. It takes the spectrum analysis of the left and right channels and applies the ideal spectral phase compensation to minimize phase cancellations between them. This is advantageous not only for mono compatibility, but the human brain also picks up phase differences of individual frequencies in the signal and uses them as clues for 3D positioning. Unfortunately since various elements from recording equipment to microphone position affect the phase, the resulting clues are usually very confusing for the brain and this feature can help minimizing them and make the stereo field tighter and centered. Please note that since this feature is actually changing the spectrum, it is not fully transparent. It also introduces latency. Please also note that this feature may sometimes have severe side-effects, especially when it comes to transients. Use the **Phase threshold** and **Phase smoothing** parameters to minimize that if needed.



### Peak / Loudness

Peak / Loudness controls how much the analyzer uses peak level (maximum when the control is at 0%) and how much it uses loudness estimate (maximum when at 100%). In most cases the loudness measure is mainly relevant, but for some percussive instruments the peak levels sometimes differ too much between the channels, which causes a sort of a psychoacoustic confusion. In these cases it may be worth trying to lower this value before analysis. Please note that this parameter is relevant only during analysis.

Range: 0.00% to 100.0%, default 100.0%



### Delay tolerance

Delay tolerance sets the limit for searching for minimum delay. There is rarely a one perfect delay time between the channels. Instead there are several possible delays, which all reasonably avoid phase cancellation, but while this scientifically seems good, it may not be so psychoacoustically. The analysis may tend to select delays that are too long, because they are technically better, but they are in fact worse in practice, since they may make the output not rhythmical.

When analysing the audio material each possible delay is investigated and marked by a quality factor. The delay with the highest quality is then selected. The delay tolerance then specifies how much lower the quality may be compared with the scientifically best result, so the plugin may try to search for a shorter delay, which could then improve the psychoacoustic results while still providing good scientific quality. Please note that this parameter is relevant only during analysis.

Range: -10.00 dB to 0.00 dB, default -5.00 dB



### Spectrum smoothing

Spectrum smoothing lets you smooth the spectrum compensation that is applied on the right channel signal. It could be helpful if the automatic spectrum fix is too brutal, causing various artefacts such as transient smearing for percussive signals. However, please note that by using this parameter you also get further away from the theoretically ideal spectrum processing, so it technically lowers the effect of the compensation itself.

Range: 0.00% to 100.0%, default 20.0%



### Phase smoothing

Phase smoothing lets you smooth the phase envelope used by the phase fix. It could be helpful if the processing is too brutal, causing various artefacts such as transient smearing for percussive signals. However, please note that by using this parameter you also get

further away from the theoretically ideal phase processing, so it technically lowers the effect of the compensation itself.  
Range: 0.00% to 100.0%, default 20.0%

## Advanced panel



Advanced panel contains advanced parameters, which you may need to use only in extreme cases.

Maximum delay 100 ms

### Maximum delay

Maximum delay defines maximum time offset to be detected for the delay fix. You may decrease this value if you feel the plugin is actually introducing delay. And conversely you can increase it if there's a bigger delay than the plugin can detect. Please note that this parameter is relevant only during analysis.

Range: 0 ms to 1000 ms, default 100 ms

Spectrum limit +20.00 dB

### Spectrum limit

Spectrum limit lets you limit the gain performed for each frequency by the spectrum fix. It is an alternative to **Spectrum smoothing**, which also reduces the effect and can therefore be used if the spectrum fix is too brutal.

Range: 0.00 dB to +40.00 dB, default +20.00 dB

Phase threshold -10.0 dB

### Phase threshold

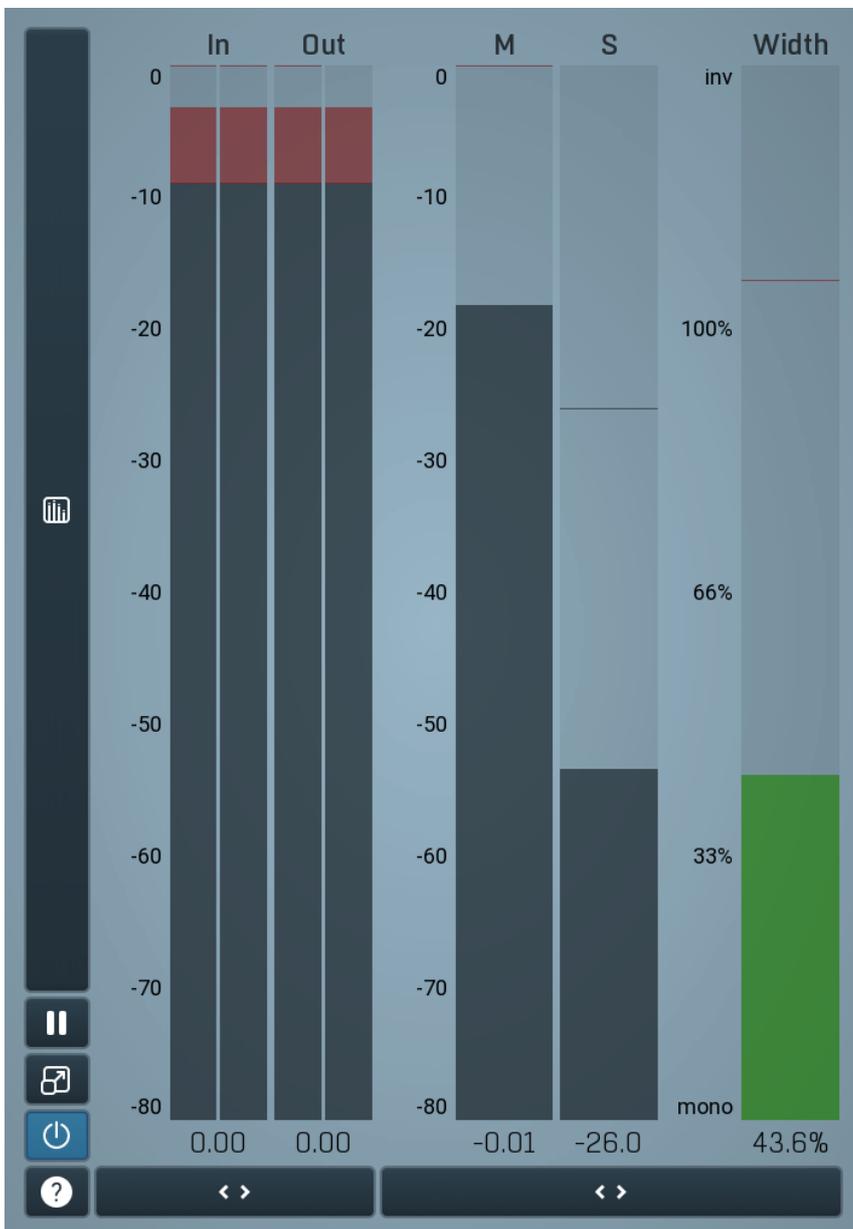
Phase threshold defines the minimum level to be considered relevant for the phase fix. In most signals all frequencies are actually present, but most of them are only noise, distortion and other artefacts. Shifting such signals wouldn't be a problem on its own, but the more the phase is changed, the bigger the chance that something will go wrong. In percussive signals this could lead to smearing of the initial transient for example. This parameter could be used to let the plugin consider only the prominent frequencies relevant for the analysis. All frequencies will then be processed anyway, but this can make the resulting phase envelope much less complex, which in effect improves the phase coherency and minimizes the chance that something goes wrong.

Range: silence to 0.00 dB, default -10.0 dB

Allow polarity invert

### Allow polarity invert

Allow polarity invert switch controls whether the polarity is allowed to be inverted by the delay fix if needed. You can try turning it off if you feel the plugin has performed polarity inversion and made things actually worse.



### 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.

**M & S meters** display mid and side levels. These meters require at least 2 channels, so they do not work in mono mode.



### 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.



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