

MAutoAlign

MAutoAlign (14.15) L+R Bypass Settings

Group Drumset

LATENCY 0 ms **DELAY** 0 samples

ANALYSIS Zero latency during analysis

LENGTH 4000 ms **MAXIMUM DELAY** 10 ms

High-pass Off Low-pass Off
Phase resolution 4096 Noise floor -80.0 dB

SPECTRAL PHASE COMPENSATION Bypass **Enable**

THRESHOLD -60.0 dB **SMOOTHING** 10.0% **DEPTH** 100.0%

- Disable delay detection
- Optimize delay
- Show wrapped phase

The graph displays phase shift in degrees on the y-axis (ranging from -17.00 to +17.00) against frequency in Hz on a logarithmic x-axis (ranging from 20 to 20k). The phase is flat at 0.00 degrees until approximately 500 Hz, then drops sharply to about -11.33 degrees at 2k Hz, and continues to decrease to a minimum of approximately -15.00 degrees at 5k Hz before leveling off.

Analyse Bypass all A / B

0 samples 0.0 ms 0.0 cm 0.0 inch INV

Overview

Phase cancellation is a constant problem when recording an instrument (or instruments) with multiple microphones. It is caused by the relatively slow speed of sound that allows the same audio signal to be received by each mike at slightly different times depending on the distance of each from the sound source, surrounding walls etc. When these tracks are subsequently mixed, problems often occur. Some frequencies cancel each other out and the audio loses punch resulting in a poor recording. This phenomenon is always present in varying degrees and it cannot be fixed physically. So the first step is to acknowledge the problem and then find a way to fix it.

The solution is to delay each of the recorded tracks by a certain number of samples and most engineers do this manually. However it is not uncommon to have delays of around 500 samples between microphones so even with just 4 microphones this gives us more than 62,500,000,000 possibilities!

MAutoAlign solves this problem in an extremely efficient way. All you need to do is place MAutoAlign on all tracks that need to be aligned, start playback and press the **Analyze button** in one of them. The plugin will then analyze all tracks and determine the correct delay for each track after just a few seconds. MAutoAlign does not alter the tracks in any way, it only delays them and inverts the phase if necessary.

Groups

When multiple instruments are recorded, drums and guitars for example, it's unlikely that any cancellations between the drum and guitar tracks will occur as they essentially contain completely different audio. In fact, trying to fix such a group of tracks would probably do more harm than good. In this case all that is needed is to group the different types of tracks.

For example, if 8 microphones have been used for drums, and 2 for the lead guitar, group **D** could be set for all 8 MAutoAlign instances in the drum tracks and group **G** for both guitar tracks. Pressing Analyse in any of the drum tracks, will cause all 8 to be analysed, while the guitar track settings will remain unchanged. Pressing Analyse in either guitar track will analyze just those two.

Spectral phase compensation system

Since version 11.02 MAutoAlign features most likely the world's first spectral phase compensation system. The delay and phase inversion described above technically do nothing to the actual audio, so they are completely transparent, but they also have very limited power. In a perfect world, where audio instruments would be microscopic points in space, microphones would record perfect audio without any phase shift and no ambience, this would be fine. Unfortunately that's not the case and the reality is that whatever you do, each track will have the phase changed in some way. In practice this means that when you mix multiple signals, some frequencies will combine satisfactorily, others will cancel each other out making the output sound thin and, well, ugly. And although some people try to fix that problem using allpass filters, it's an almost impossible job. MAutoAlign can now solve that in a matter of seconds.

As part of the analysis, MAutoAlign calculates the optimal phase adjustment to minimize cancellations. You can then simply enable the **Spectral phase compensation** module and it will apply the phase adjustment to the input signals. It is disabled by default as it induces latency (hence it cannot be used live) and it may not always work.

The spectral phase compensation works when there is a stable phase difference. Hence it mostly works when there is not too much noise or ambience, ideally for close mics. Ambience and noise are very unpredictable signals with no stability whatsoever, which makes the plugin try to come up with at least something, but the results are usually not helpful. If that's the case, try setting up a longer analysis length, which can make the noise and ambience cancel itself out in time making the useful signal more pronounced.

Spectral phase compensation system for mixing dry & wet signals

Imagine that you take some signal, let it be a piano, and process it using some equalizer or analog device for instance. This processing most likely has caused some phase shifting to the signal. Now you cannot really mix the processed version with the original anymore, because some frequencies would just cancel each other out and results would be awful, or at least not exactly ideal. The spectral phase compensation system can once again fix this for you - it will analyze the phase difference between the 2 signals and try to either revert the phase alteration in the processed signal, or cause a similar phase alteration in the dry signal. Either way it will hopefully minimize phase cancellation.

This also means, that for the first time you can now equalize the tracks before aligning them and MAutoAlign will not only fix the time differences between the tracks, but also correct the phases of each track, whether it was caused by the processing, the microphone, or anything else on the way.

Where to place MAutoAlign in the effect chain?

MAutoAlign should be placed in the insert effects chain of your host and if you don't use any plugins causing latency, then it is only a matter of personal preference. Making MAutoAlign the first item in any chain will usually result in a more successful analysis as this will likely be free of other effects and be the most natural input. Making it the last item lets it fix the correct the phase alterations caused by any processors before it, such as equalizers or delays.

Problems may arise when you use latency-inducing plugins. The plugin uses the host's timestamps to synchronize all instances, unfortunately many hosts are rather problematic in that aspect. If you exhibit erratic behaviour, where each analysis produces in different results despite analyzing the same portion of the audio, try disabling all latency-inducing plugins temporarily. And let your host's developer know about the problem.

How to group tracks of an acoustic drum kit?

Acoustic drums are typically recorded using several close microphones, together with overheads and room microphones. Let's start with the "close mics".

The question is, should you group all close microphones together? There is no definitive answer, but it is customary to try it this way first. If it doesn't provide satisfactory results, which is often the case if multiple drums are recorded using multiple microphones, you should try grouping the individual drum kit pieces together. In most cases, the original setting with all close mics grouped together works just fine.

Although the close mics are placed very close to the individual drums, they will often capture the sound of other drums as well. This so-called bleed usually doesn't need to be handled specifically, unless it is too loud. Drums recorded with multiple microphones, e.g. snare top and snare bottom, should always be aligned together.

Make sure that you use a reasonable part of the recording for analysis - and that all microphones are actually capturing the performance. If some of the tracks are silent, it is unlikely these tracks will be aligned correctly. Be especially careful when you use MAutoAlign as the last item in the chain for each track and you also use a gate before it. By definition, the gate will be cutting out some parts of the track.

Overheads are also quite close to the drum kit, but they capture lots of ambience too. Experience tells us that this ambience usually makes aligning pointless, however you should try anyway. Sometimes overhead microphones are placed some distance from each other. In this case it may be necessary to align them with each other as there may be phase collision between them and alignment would improve the stereo image.

You can also try aligning the overheads with the close mics. Your first step is to listen to discover if they are actually colliding with the close microphones. If so, first mix all the close microphones into a bus (group) track, let's call it "closedrums" and then align the overheads with that track. This grouping is done because aligning overheads with the separate close microphones, can confuse the analysis as the phasing problems are minimal and the overheads contain too much ambience. So it is usually better to work with the close microphones together, after all these are the key for a tight mix, overheads only fill the gaps, glue it together and provide some natural ambience.

Room microphones are usually placed some distance away from the whole performance and as the name suggests, are there to capture the room. With that much ambience it is difficult to find and fix phasing issues, but it is often worth trying. As with the overheads, you may use MAutoAlign to fix phasing issues between multiple room microphones if they have been placed far enough away from each other. The good news is that ambience in a way fights phase cancellations, so in most cases there actually isn't any problem in the first place.



Presets

Presets button shows a window with all available presets. A preset can be loaded from the preset window by double-clicking on it, using the arrow buttons or by using a combination of the arrow keys and Enter on 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.

Holding **Ctrl** while pressing the button loads an existing preset, selected at random.

Presets can be backed up by using either the Export button, or by saving the actual preset files, which are found in the following directories:
Windows: C:\Users\{username}\AppData\Roaming\MeldaProduction
Mac OS X: ~/Library/Application support/MeldaProduction

Exported preset files can be loaded into the plug-in's preset store using the Import button. Or the preset files themselves can be copied into the directories named above.

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



Left arrow

Left arrow button loads the previous preset.



Right arrow

Right arrow button loads the next preset.



Channel mode

Channel mode button shows the current processing channel mode, e.g. **Left+Right (L+R)** indicates the processing of left and right channels. This is the default mode for mono and stereo audio material and effectively processes the incoming signal as expected. However the plugin also provides additional modes, of which you may take advantage as described below. Mastering this feature will give you unbelievable options for controlling the stereo field.

Note that this is not relevant for mono audio tracks, because the host supplies only one input and output channel.

Left (L) mode and Right (R) mode allow the plugin to process just one channel, only the left or only the right. This feature has a number of simple uses. Equalizing only one channel allows you to fix spectral inconsistencies, when mids are lower in one channel for example. A kind of stereo expander can be produced by equalizing each side differently. Stereo expansion could also be produced by using a modulation effect, such as a vibrato or flanger, on one of these channels. Note however that the results would not be fully mono compatible.

Left and right channels can be processed separately with different settings, by creating two instances of the plugin in series, one set to 'L' mode and the other to 'R' mode. The instance in 'L' mode will not touch the right channel and vice versa. This approach is perfectly safe and is even advantageous, as both sides can be configured completely independently with both settings visible next to each other.

Mid (M) mode allows the plugin to process the so-called mid (or mono) signal. Any stereo signal can be transformed from left and right, to mid and side, and back again, with minimal CPU usage and no loss of audio quality. The mid channel contains the mono sum (or centre), which is the signal present in both left and right channels (in phase). The side channel contains the difference between the left and right channels, which is the "stereo" part. In 'M mode' the plugin performs the conversion into mid and side channels, processes mid, leaves side intact and converts the results back into the left and right channels expected by the host.

To understand what a mid signal is, consider using a simple gain feature, available in many plugins. Setting the plugin to M mode and decreasing gain, will actually lower or attenuate the mono content and the signal will appear "wider". There must be some stereo content present, this will not work for monophonic audio material placed in stereo tracks of course. Similarly amplifying the mono content by increasing the gain, will make the mono content dominant and the stereo image will become "narrower".

As well as a simple gain control there are various creative uses for this channel mode.

Using a **compressor** on the mid channel can widen the stereo image, because in louder parts the mid part gets attenuated and the stereo becomes more prominent. This is a good trick to make the listener focus on an instrument whenever it is louder, because a wider stereo image makes the listener feel that the origin of the sound is closer to, or even around them.

A **reverb** on the mid part makes the room appear thin and distant. It is a good way to make the track wide due to the existing stereo content, yet spacey and centered at the same time. Note that since this effect does not occur naturally, the result may sound artificial on its own, however it may help you fit a dominant track into a mix.

An **equalizer** gives many possibilities - for example, the removal of frequencies that are colliding with those on another track. By processing only the mid channel you can keep the problematic frequencies in the stereo channel. This way it is possible to actually fit both tracks into the same part of the spectrum - one occupying the mid (centre) part of the signal, physically appearing further away from the listener, the other occupying the side part of the signal, appearing closer to the listener.

Using various **modulation effects** can vary the mid signal, to make the stereo signal less correlated. This creates a wider stereo image and makes the audio appear closer to the listener.

Side (S) mode is complementary to M mode, and allows processing of only the side (stereo) part of the signal leaving the mid intact. The same techniques as described for M mode can also be applied here, giving the opposite results.

Using a **gain** control with positive gain will increase the width of the stereo image.

A **compressor** can attenuate the side part in louder sections making it more monophonic and centered, placing the origin a little further away and in front of the listener.

A **reverb** may extend the stereo width and provide some natural space without affecting the mid content. This creates an interesting side-effect - the reverb gets completely cancelled out when played on a monophonic device (on a mono radio for example). With stereo processing you have much more space to place different sounds in the mix. However when the audio is played on a monophonic system it becomes too crowded, because what was originally in two channels is now in just one and mono has a very limited capability for 2D placement. Therefore getting rid of the reverb in mono may be advantageous, because it frees some space for other instruments.

An **equalizer** can amplify some frequencies in the stereo content making them more apparent and since they psycho acoustically become closer to the listener, the listener will be focused on them. Conversely, frequencies can be removed to free space for other instruments in stereo.

A **saturation / exciter** may make the stereo richer and more appealing by creating higher harmonics without affecting the mid channel, which could otherwise become crowded.

Modulation effects can achieve the same results as in mid mode, but this will vary a lot depending on the effect and the audio material. It can be used in a wide variety of creative ways.

Mid+Side (M+S) lets the plugin process both mid and side channels together using the same settings. In many cases there is no difference to L+R mode, but there are exceptions.

A **reverb** applied in M+S mode will result in minimal changes to the width of the stereo field (unless it is true-stereo, in which case mid will affect side and vice versa), it can be used therefore, to add depth without altering the width.

A **compressor** in M+S mode can be a little harder to understand. It basically stabilizes the levels of the mid and side channels. When channel linking is disabled in the compressor, you can expect some variations in the sound field, because the compressor will attenuate the louder channel (usually the mid), changing the stereo width depending on the audio level. When channel linking is enabled, a compressor will usually react similarly to the L+R channel mode.

Exciters or saturators are both nonlinear processors, their outputs depend on the level of the input, so the dominant channel (usually mid) will be saturated more. This will usually make the stereo image slightly thinner and can be used as a creative effect.

How to modify mid and side with different settings? The answer is the same as for the L and R channels. Use two instances of the plugin one after another, one in M mode, the other in S mode. The instance in M mode will not change the side channel and vice versa.

Left+Right(neg) (L+R-) mode is the same as L+R mode, but the the right channel's phase will be inverted. This may come in handy if the L and R channels seem out of phase. When used on a normal track, it will force the channels out of phase. This may sound like an extreme stereo expansion, but is usually extremely fatiguing on the ears. It is also not mono compatible - on a mono device the track will probably become almost silent. Therefore be advised to use this only if the channels are actually out of phase or if you have some creative intent.

There are also 4 subsidiary modes: **Left & zero Right (L(R0))**, **Right & zero Left (R(L0))**, **Mid & zero Side (M(S0))** and **Side & zero Mid (S(M0))**. Each of these processes one channel and silences the other.

Surround mode is not related to stereo processing but lets the plugin process up to 8 channels, depending on how many the host supplies. For VST2 plugins you have to first activate surround processing using the **Activate surround** item in the bottom. This is a global switch for all MeldaProduction plugins, which configures them to report 8in-8out capabilities to the host, on loading. It is disabled by default, because some hosts have trouble dealing with such plugins. After activation, restart your host to start using the surround capabilities of the plugins. Deactivation is done in the same way. Please note that all input and output busses will be multi-channel, that includes side-chain for example. For VST3/AU/AAX plugins the activation is not necessary.

First place the plugin on a surround track - a track that has more than 2 channels. Then select **Surround** from the plug-in's Channel Mode menu. The plugins will regard this mode as a natural extension of 2 channel processing. For example, a compressor will process each channel separately or measure the level by combining the levels of all of the inputs provided. Further surround processing properties, to

enable/disable each channel or adjust its level, can be accessed via the **Surround settings** in the menu.

Ambisonics mode provides support for the modern 3D systems (mostly cinema and VR) with up to 64 channels (ambisonics 7th order). Support for this is still quite rare among the DAWs, so this needs to be activated in all DAWs using the **Activate ambisonics** item in the bottom. This is a global switch for all MeldaProduction plugins, which configures them to report 64in-64out capabilities to the host, on loading. After activation, restart your host to start using the ambisonics capabilities of the plugins. Deactivation is done in the same way. Please note that all input and output busses will be multi-channel, that includes side-chain for example.

First place the plugin on an ambisonics track, supported are all orders from 1st (4 channels) to 7th (64 channels). Then select **Ambisonics** from the plug-in's Channel Mode menu. Finally select the **Ambisonics settings** in the menu and configure the Ambisonics order and other settings if needed. The plugins will regard this mode as a natural extension of 2 channel processing. For example, a compressor will process each channel separately or measure the level by combining the levels of all of the inputs provided.



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.

Activate lets you activate the plugin if the drag & drop activation method does not work in your host. In this case either click this button and browse to the licence file on your computer and select it. Or open the licence file in any text editor, copy its contents to the system clipboard and click this button. The plugin will then perform the activation using the data in the clipboard, if possible.

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.

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.

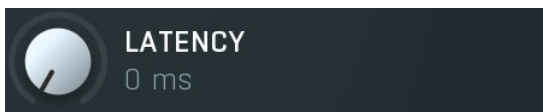


WWW

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.

Group Drumset Group

Group lets you select multiple tracks to synchronize with each other. All tracks with the same group are analyzed together.

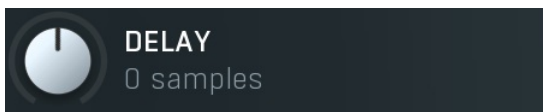


LATENCY
0 ms

Latency

Latency lets all of the instances in the group report this latency. Since the process of aligning delays the tracks, they slightly shift in time, but always forwards. You may therefore experience a feeling of different timing. To compensate for it, the plugin can report latency, for which the host compensates (Plugin Delay Compensation, or PDC), which in effect moves all the audio backwards in time. It is usually necessary to restart playback after changing this value.

Range: 0 ms to 100 ms, default 0 ms



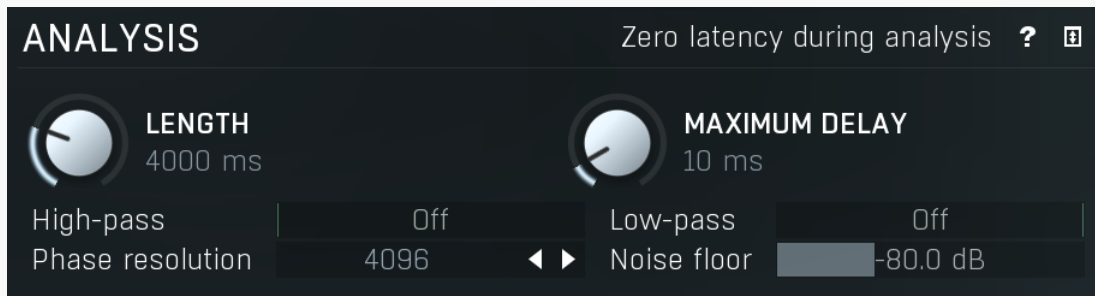
DELAY
0 samples

Delay

Delay lets you manually move the computed offset. You can use this to adjust the results if you are not satisfied with them. What you do is immediately updated in the log. You cannot have negative delay, since it is impossible without inducing latency. But note that the meters showing better values may not mean the results are actually better. You should always trust your ears.

Range: -1024 samples to 1024 samples, default 0 samples

Analysis panel



Analysis panel contains the parameters of the standard delay and phase inversion analysis.

Zero latency during analysis

Zero latency during analysis

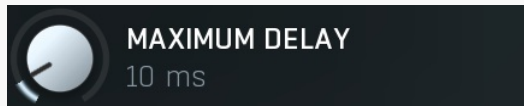
Zero latency during analysis makes all the plugins in the group report 0 latency when the analysis starts and put it back again afterwards. This is useful for hosts, which do not report correct timing info to the plugins when a plugin with latency is used. So basically leave this disabled and if you feel like things are not working properly, turn it on. Note that it makes a difference only if the plugin is actually having latency, typically when spectral phase compensation is used.



Length

Length controls the length of the analysis, hence how much signal needs to be recorded for the analysis. The default should suit most needs, but if you feel like you are processing long and steady signals, you may increase the length for example. Note that increasing the length does not improve the accuracy of analysis and sometimes it can even make things worse.

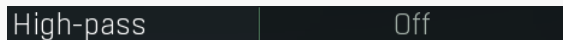
Range: 2000 ms to 30000 ms, default 4000 ms



Maximum delay

Maximum delay defines maximum time offset to be detected. In most cases the delay between microphones is lower than 10ms and making this value higher can make the plugin generate higher delays due repeating sequences in the audio. Please note that this parameter is relevant only when you click the **Analyse button** in.

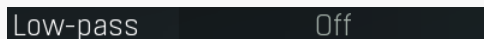
Range: 0.10 ms to 1000 ms, default 10 ms



High-pass

High-pass defines the high-pass filter cut-off frequency which may be used to remove part of the signal for the analysis, for example to avoid drum leakage. Please note that this filter is used only for analysis and does not affect the audio signal itself.

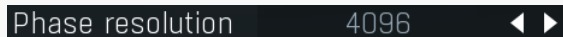
Range: Off to 20.0 kHz, default Off



Low-pass

Low-pass defines the low-pass filter cut-off frequency which may be used to remove part of the signal for the analysis, for example to avoid drum leakage. Please note that this filter is used only for analysis and does not affect the audio signal itself.

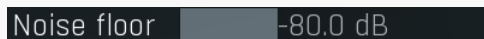
Range: 20.00 Hz to Off, default Off



Phase resolution

Phase resolution controls the block size used for spectral phase compensation analysis and processing. The algorithm is working 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, using the spectral phase compensation 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, at the expense of greater CPU cost and increased latency.

Range: 256 to 65536, default 4096

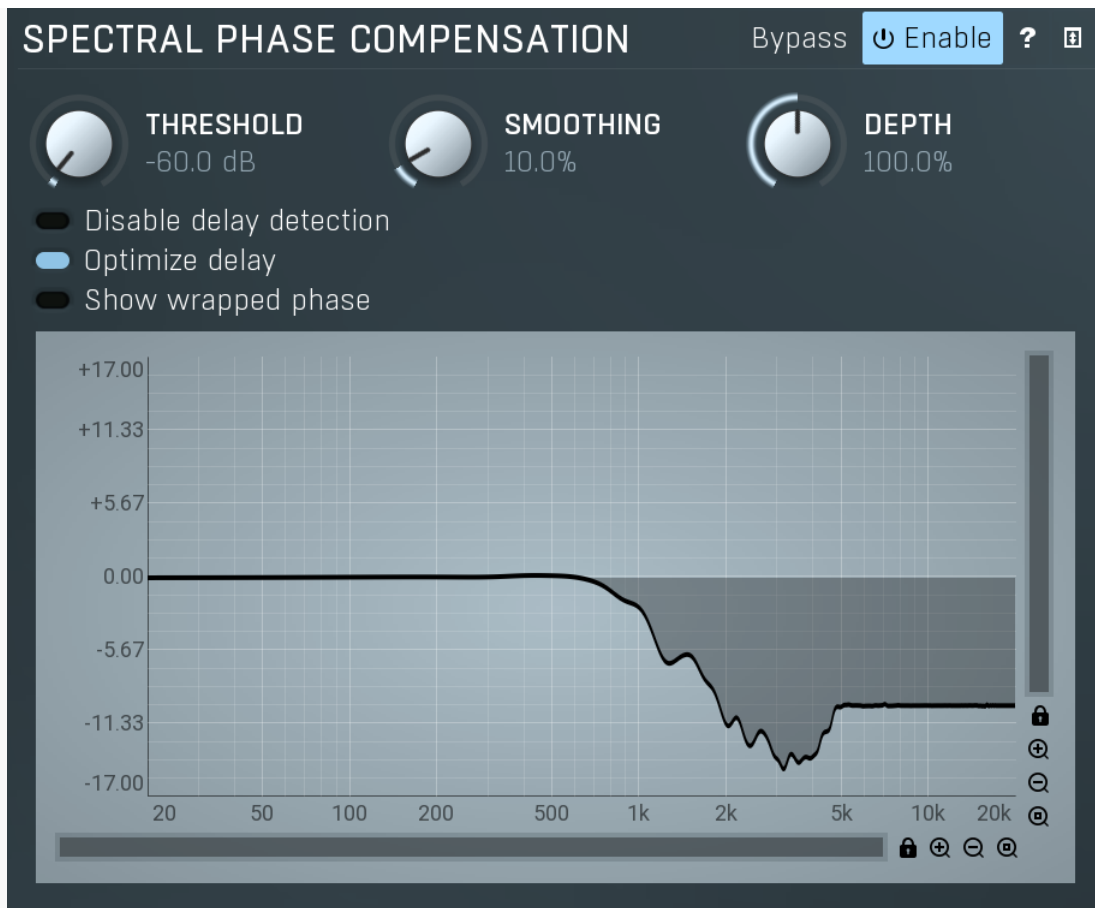


Noise floor

Noise floor controls the input volume considered to be the noise floor, which helps the plugin distinguish between useful signal and noise during analysis. It has no effect on the audio or metering.

Range: silence to 0.00 dB, default -80.0 dB

Spectral phase compensation panel



Spectral phase compensation panel contains parameters of the advanced algorithm, which directly adjusts the phase of all tracks to avoid phase cancellations. Please note that once you enable this panel, all of the instances in the group will start inducing latency.

Bypass

Bypass switch disables the spectral phase compensation without changing the latency, which is what the **Enable** button would do, which makes it hard to use for comparison, since it causes a lag. Bypass is a better choice which lets you properly determine if the phase compensation is worth keeping enabled.

Threshold

Threshold defines the minimum level to be considered relevant for the phase shifting. 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 -60.0 dB

Smoothing

Smoothing lets you smoothe the phase envelope. 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 the phase cancellations can increase again.

Range: 0.00% to 100.0%, default 10.0%

Depth

Depth controls the amount of phase shift being performed. In most cases only 100% makes sense as it performs the exact phase shift determined as ideal. However you may want to change this as an experiment to see if different values could provide better results.

Range: 0.00% to 200.0%, default 100.0%

Disable delay detection

Disable delay

detection

Disable delay detection switch makes the analysis avoid detecting the delay and inversion and focus on phase compensation only. Normally you should leave this disabled as both things work together. However in some cases, such as mixing a dry signal with a wet

one (with altered phase using an equalizer for example), where you know that no delay has been performed, you may try enabling this option to see if it would result in a better analysis.

Optimize delay

Optimize delay

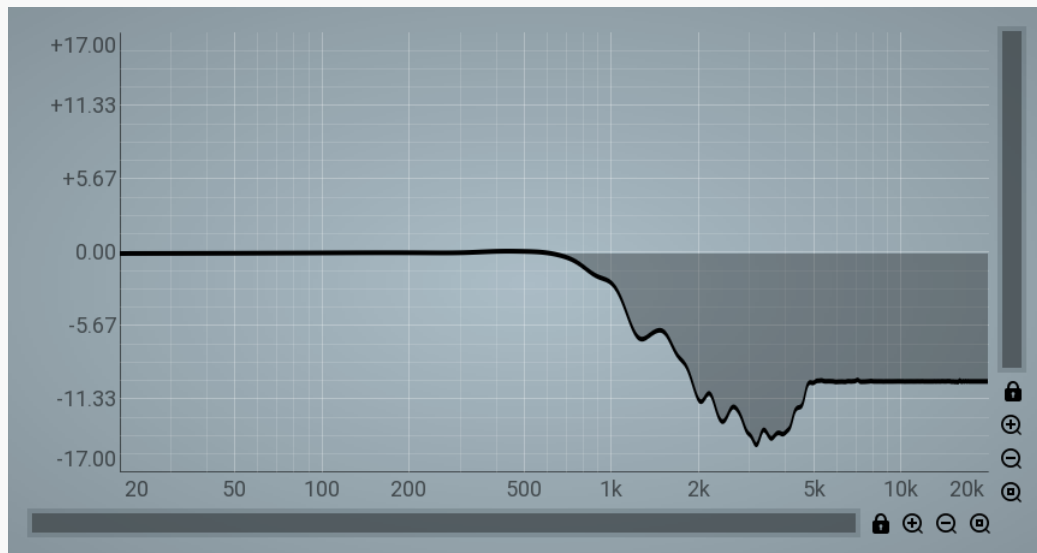
Optimize delay switch enables advanced processing which optimizes both delays and phase compensation at once. It is performed during the analysis, so to make this work, you need to have both spectral phase compensation and optimize delay enabled. It can provide better audio quality.

Show wrapped phase

Show wrapped

phase

Show wrapped phase switch changes the phase graph into a so-called wrapped one. Phase is not a classic measure, it is cyclic, meaning that -180 degrees produces the same effect as +180 degrees. Showing the phase in such a diagram (called wrapped) can then look rather hard to understand, since in many cases the line will be jumping from top to bottom eventually looking like a big mess. Therefore by default the plugin shows so-called unwrapped phase, which adjusts for that and makes the graph look easier to understand for a human.



Phase graph display

Phase graph display shows the phase compensation for this instance after processing by Threshold and other controls. The graph basically shows how much the phase of each frequency is shifted. Note that since the phase adjustment can reach huge numbers (which is not a bad thing due to the fact that phase is cyclic), the vertical units don't show the actual phase, but the number of half-cycles - 1 means 180 degrees, 2 means 360 degrees. In any case these values are useful for your curiosity only.

Analyse

Analyse

Analyse button starts the analysis. Start playback in your host at an averagely loud position and press this button. The processing will take a few seconds and new settings will appear immediately. If you aren't be satisfied with the results, you can just press the button again to proceed with another analysis. The reason is, that in most real cases there is no perfect solution and results highly depend on the actual part of the song being played during the analysis.

Bypass all

Bypass all

Bypass all button bypasses all plugins in the group as opposed to the standard bypass, which overrides just the single instance. Bypassing all lets you check how all tracks sound without the automatic alignment.

A / B

A/B

A/B button lets you switch between 2 analyses. If you switch from A to B, the plugin changes all other plugins in the same group as well. Analysis results are then stored in the particular slot, so you when you have analyzed both, you can easily switch between them using this button and audition which sounds better.

Log view

0 samples

0.0 ms

0.0 cm

0.0 inch

INV

Log view displays calculated delay in samples, milliseconds, and the approximate distance in centimeters and inches.



INV button is highlighted if the phase has been inverted. You can change the state by clicking on it.

