

MCompare



Overview

MCompare is an extremely versatile tool for comparing your audio to external audio files or even to other stages of your processing. It provides a smooth transition between the input and whatever you want to compare it to, with automatic loudness and latency compensation, and extensive metering. Comparing your audio to finished recordings and to different stages of the processing is an essential part of both mixing and mastering, and even though you can do it in your host, MCompare is intended for this task, provides the quickest possible workflow and avoids potential confusion caused by different loudness or latency.

Case 1 - Comparing your audio to finished commercial recordings

This is the most basic use that almost every successful mixing and mastering engineer carried out. In order to get the correct loudness, spectrum & spatial balance, level proportions between different tracks of the recording and other sonic qualities, it is essential to compare your mix/master to actual finished records. The evolution in mixing and mastering throughout many decades is huge and there's no idea worse than reinventing the wheel by starting from scratch. In most music styles a certain sound and/or sonic character is expected and providing it without extensive comparisons with commercial recordings is almost impossible (unless you are very lucky).

MCompare provide multiple slots for external files, called **File 1**, **File 2** etc.. You can load an audio file into each of these slots using the browse button. The plugin can load all important audio file formats (WAV, MP3, FLAC etc.) and resamples the audio to match your current project automatically. By clicking the particular File button you can then switch between the plugin input and the file. In the **audio file view** on top of the plugin you can see the whole waveform, use the green region to control loop points and double click anywhere to jump to that particular position.

Mixing considerations

In the mixing stage we assume the overall spectrum balance, dynamics and loudness is not so important as it will be managed during the mastering. However it's pretty common to use some mild master compressor and limiter during mixing to be able to "anticipate" the resulting sound. In any event the most important thing is to avoid getting fooled by loudness differences. The plugin provides an **automatic loudness compensation (ALC)**, which may be a good starting point. You can either enable the ALC itself, which will compensate the

loudness on-the-fly, or use the **Set** button to let the engine configure the **Gain** parameter, in which case the loudness will approximately match at the moment you press the button, but you may need to use it again when you switch to a different part of the song for example.

We would recommend selecting a certain part of your audio recording, say a chorus, then selecting a sonically similar part of the file (using the loop area in the audio file view) and adjusting the loudness. From that moment onwards you can focus on the balance between tracks, spatial character, spectral collisions etc. without worrying about misjudgement caused by different loudness levels.

Mastering considerations

During mastering it is not recommended to use ALC, because the overall loudness is one of the relevant factors and fluctuations should be avoided. We would therefore again recommend selecting a certain part of the master, selecting a sonically similar part of the file, and using the Set button to compensate for the loudness at the beginning. When the spectral and dynamic balance of your master is provisionally good enough, you should reset the gain parameter to 0dB by right-clicking on it, so that you can start focussing about the output loudness (controlled probably by the master limiter). Of course, you may still readjust the gain, but it is recommended doing that only slightly, because the comparison file serves as the loudness reference too of course. In any case we would advise setting the gain below 0dB, hence lowering the reference loudness, so that you avoid the typical over-processing and can ignore the "loudness war", which sucks life out of many modern recordings.

Case 2 - Comparing your audio to a different stage of your mastering chain

Mastering typically contains just one set of processors in series, let's say compressor -> equalizer -> limiter. Besides comparing your master to some commercial recordings it is almost always a good thing to compare your processed master to the original mix, simply to ensure your processing made some improvement and didn't mess things up. There are 2 problems - mastering processors often induce latency, which makes the comparison rather problematic, and the overall loudness almost always changes during mastering, it's one of the purposes of the whole process after all. Sadly, this makes reasonable comparison almost impossible as it is extremely hard to focus on the sonic qualities without being fooled by the higher loudness of the master. MCompare provides with a solution to both problems.

First you need to setup a new mastering chain: MCompare A -> compressor -> equalizer -> limiter -> MCompare B. Instance A is called a **source generator**, or just source. It does literally nothing to your audio, but it sends it somewhere into the depths of your computer from where the other instance can get it. You simply insert MCompare and click the **Listening** button at the top and select, for example, Generate source 1. Then you can close the GUI, there's really nothing more you'll probably need. Instance B is called the **monitoring instance** as this is the one you will be using to listen and compare. And here you simply press **Source 1** button compare your mastered audio with the original mix.

After that it would be good to compensate for the latency, if needed. To do that you simply start playback and press the **Detect** button located next to the **Delay** parameter. It will do some processing for a few seconds and then the new delay value should be configured. If you feel like it's not working well, press the panic button '!' at the top of the plugin, restart playback and try again.

Finally it may be useful to adjust the loudness. The idea here is to increase the loudness of the source, so that you can focus on the actual sonic character. It is recommended to use the Set button; this works in exactly the same way as it does for the **File** buttons. You may also need to set the output gain in your mixer to some lower value for a time, because otherwise your master will be fine, but the source won't be protected by the mastering limiter, hence probably clipping. So you can, for example, set the **Volume** to -10dB and increase the volume knob in your audio interface accordingly. Then when comparing your master with your mix, just press the source button repeatedly. Then pushed, the source is playing and the output is likely to exceed 0dB. When the button is released your master is playing and the output will NOT exceed 0dB, since it is protected by the limiter.

Now since MCompare provides multiple source slots, you can actually monitor multiple stages of your mastering chain! In that case the chain could look for example like this: MCompare Generate source 1 -> compressor -> MCompare Generate source 2 -> equalizer -> MCompare Generate source 3 -> limiter -> MCompare B. And now you can monitor all stages of your mastering chain without being fooled by latency or loudness changes!

Case 3 - Comparing your master in sync with an exported file

It is pretty common that you master a track, export it to say FLAC file and a week later you come back and want to try again, but of course you need to check if your new master is better or not. You can load the FLAC file into your host, but then you have the same typical trouble with A/Bing it there. Or you can load it to a MCompare File slot, but then it would be nice if you could keep it in sync with your master, so that the comparison is really effective.

Fortunately MCompare makes this very simple as well. Just enable the **Sync** button in the File slot. The plugin will then follow your host, so when you rewind in your host, the plugin will rewind as well. The last thing to do is to make both exactly in sync. To do that first use the **Start** parameter to make it approximately in sync. If your project starts in the beginning, then there's probably nothing to do really. And to get an exact synchronization, use the same approach as in Case 2 with sources - enable playback and press the **Detect** button. The plugin will work a few seconds and then compute the ideal delay.

Case 4 - The ultimate mastering comparison method

Now let us take all the knowledge from above and create one ultimate method to compare your master to commercial recordings, original mix and even different versions of your master. Let's take it one by one. You should always use at least one commercial recording for comparison. You have multiple file slots (see Case 1) for that, use them. Next the original mix - that has been described in Case 2. Note that you could also use the mix as an audio file, but in that case the plugin couldn't synchronize the master with the mix, so the process wouldn't be very effective.

Now we get to the interesting part - at one point you could say that your master is pretty good but could be better. You'd like to try again something else. So you could export your current master, save the project as a new one and there use MCompare to compare your new master, which you are going to "try", with the exported master and use a File button for that. Well, yes, that could work, but it's a lot of work and again, these 2 masters would be out of sync, so the comparison wouldn't be effective.

So there's another solution - let's say you master in Cubase (or any other multitrack host for that matter). Instead of having the mastering chain on the master channel, as you would probably do, mainly because of its name :), you do everything in an audio track, just keep the main MCompare instance on the master track. Then when you are want to try another mastering chain, you simply duplicate the track, put another MCompare instance on the old one's end, let it generate a source and set the volume to silence, so that the track is processing, but instead of sending the output to your host's mix, it sends it to MCompare as a source and mutes it in the host. So it would look like this:

Track 1 = The first mastering chain: MCompare Generate source 1 (dry mix) -> mastering chain -> MCompare Generate source 2 (master 1), with Volume set to silence

Track 2 = The new mastering chain: mastering chain

Master track = MCompare monitoring instance

So now you have dry mix in Source 1, first master in Source 2 and your new master you are working on now is the main MCompare instance input. And at this moment you are one click away from comparing your new master to your old master, dry mix, and all commercial recordings, all of them with loudness compensation, so you can be sure your ears are working the way they should.

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.



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.



Copy

Copy button copies the current settings to the system clipboard. Other presets, oversampling, channel mode and other global settings are not copied.

Hold **Ctrl** to save the settings as a file instead. That may be necessary for complex settings, which may be too long for system clipboard to handle. It may also be advantageous when you want to send the settings via email. You can load the settings by drag & dropping them to a plugin or holding **Ctrl** and clicking **Paste**.



Paste

Paste button pastes settings from the system clipboard into the current preset. Hold **Ctrl** to load the settings from a file instead. Hold **Shift** to paste the settings to all of the A-H slots at once.



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.

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.

Left

Left

Left button lets you audition the left channel signal only.

Right

Right

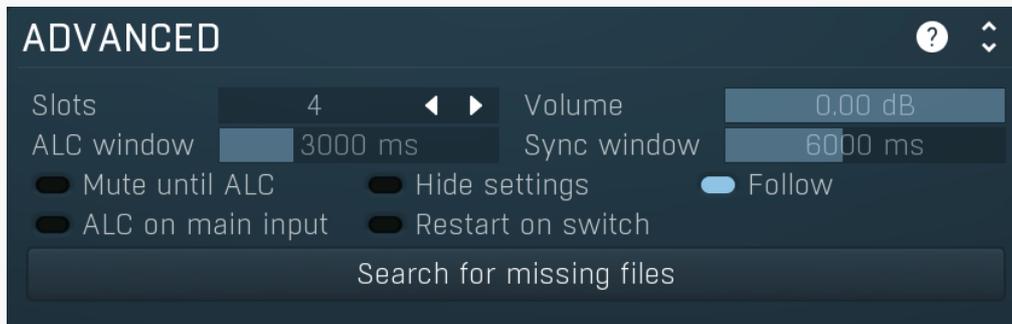
Right button lets you audition the right channel signal only.

Diff

Diff

Diff button lets you audition the difference between the input signal and the comparison. Please note that this works properly if and only if there is no delay nor phase shift between the input and the comparison signals. Otherwise the result will be some undefined mixture of both signals. Therefore it is necessary to make all intervening processors linear-phase, this includes equalizers, crossovers etc. Broadband dynamics processors such as compressors or expanders should be fine unless some non-linear-phase filtering in part of the algorithm. Also note that when this is enabled, the latency detection is not functional. Please use this function ONLY if you are an expert and you know exactly what each processor you are using does.

Advanced panel



Advanced panel contains more advanced settings.

Slots

4

Slots

Slots controls the number of visible slots.

Range: 1 to 16, default 4

Volume

0.00 dB

Volume

Volume defines output volume. Use it to ensure the output won't clip. For example, if you compare your master to the dry mix, the master will most likely be much louder, so when you use ALC or just use the gain, the plugin will increase the gain of the dry mix and that will start peaking above 0dB. So you can simply set the volume to say -10dB as a precaution and adjust the volume on your audio interface/monitors to ensure nothing will be peaking and you can still use the loudness matching.

Range: silence to 0.00 dB, default 0.00 dB

ALC window

3000 ms

ALC window

ALC window controls the length of the loudness calculation window. The longer it is, the slower the response to loudness changes, but it also avoids quick fluctuations, smoothing them out. Set it to a longer value if the playback seems fluttering, or contains unnatural level changes.

Range: 1000 ms to 30000 ms, default 3000 ms

Sync window

6000 ms

Sync window

Sync window controls the length of the analysis window used to auto-sync the instance to another instance or to another file. The longer it is, the longer the analysis will take and the more resources it will need. It is also the maximum delay that the plugin will be able to auto-detect. If, for example, you suspect that the delay between the 2 sources is about 5 seconds, the sync window should be more than 5 seconds, ideally much more, e.g. 8 seconds.

Range: 1000 ms to 30000 ms, default 6000 ms

Mute until ALC

Mute until ALC button silences the output until a loudness estimate becomes available. This is disabled by default and means, that until the plugin successfully estimates loudness of the input & comparison tracks, it simply doesn't change the level at all.

A problem arises if you try to compare your very silent audio to a very loud comparison audio, because the plugin cannot prevent the high loudness comparison until it knows that the audio actually is loud, which may take a few seconds depending on the **ALC window**. This option avoids the problem by completely muting the audio output until the level is available.

Hide settings

Hide settings switch shows/hides the settings inside each File and Source button, which makes it easier to compare them without worrying about clicking on individual settings.

Follow

Follow button makes the plugin enable playback only if the host has enabled playback, which is the default. You may however disable this in which case you can use MCompare as a simple audio file player, for live performances for example.

ALC on main input

ALC on main input switch changes the way in which automatic loudness compensation works. When this is disabled (the default), the main input is not affected at all and the loudness of any comparison file or source is matched to the main input. This is ideal for most cases, however in some special scenarios you may activate this switch, which makes the loudness of the main input be matched to the comparison instead, hence it works the other way around.

It may be useful when examining some processing chains for instance: MCompare send -> EQ -> MCompare. This lets you test whether the EQ is doing something actually beneficial and ALC helps you avoid any loudness trickery. By default it ensures that the send signal (that is, before the EQ) loudness is matched to the main input (after the EQ). However this means that you won't be listening to the exact original signal. This switch lets you compare the true original signal with the loudness matched version.

Please note that this feature works properly only for Sources, not for Files; in the latter case the file loudness (being the signal to which the main input is matched) is updated only when the file is actually being played.

Restart on switch

Restart on switch enables a retrigger mode - if sync for the File you are using is not activated, then every time you switch to this file its playback position will be reset to the left locator. That way you can audition a specific portion of the file in detail.

Search for missing files

Search for missing files lets you quickly remap all the missing files, if any. It lets you select the root folder and then tries to find every missing file inside it or any subfolder of any level. It is useful if you move your audio files somewhere and don't want to remap all manually.

Filter panel



Filter panel contains the HP & LP filters which may be used to compare a specific part of the spectrum.

HIGH-PASS Off

High-pass defines the high-pass filter cut-off frequency which may be used to compare a specific part of the spectrum.
Range: Off to 20.0 kHz, default Off



LOW-PASS
Off

Low-pass

Low-pass defines the low-pass filter cut-off frequency which may be used to compare a specific part of the spectrum.

Range: 20.00 Hz to Off, default Off

COMPARE

BLIND TEST

GENERATE SOURCE

Main

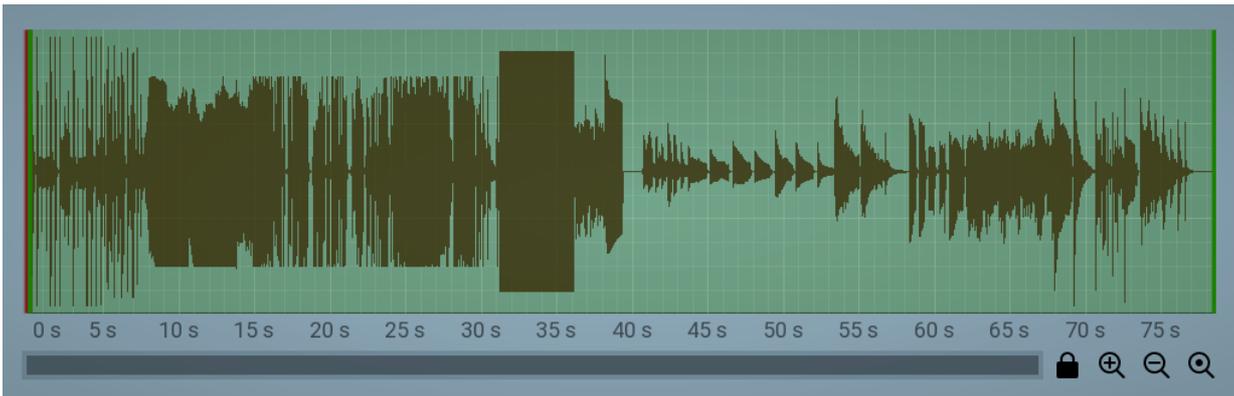
tabset

Main tabset lets you choose between 3 main functionalities of the plugin.

Compare is the default tab which lets you compare your audio material with others.

Blind test also lets you compare, but it initially shuffles all files and sources randomly and selects a random one. Then you can click on different numbers to compare the audio without actually knowing which is which. When you find the best, switch back to the Compare tab to see which it is. Please note that then no files nor sources might be selected, in which case it means you chose the audio input as the best.

Generate source tab is the utility tab which makes the plugin merely a sender. Such an MCompare instance doesn't process the sound in any way and it only sends the incoming audio to the main instance that you will be working with.



Audio file

view

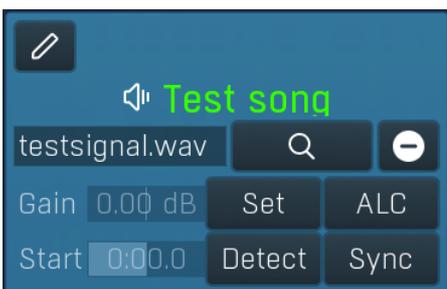
Audio file view displays the selected file waveform. The view is disabled if you don't have a file selected or there's nothing loaded into the file slot. Green range controls the current loop points. You can drag the limits to specify the loop region. Red line shows the current playback position. **Double-click** or **right-click** anywhere in the graph to jump to that position. As with all graph controls you can use mouse-wheel and the bottom scrollbar & zoomer to zoom and move the view.

When sync for the file is enabled, loop points won't make sense anymore, so instead a single line, which helps you setting up the **Start**, is displayed. Despite in most cases the sync feature will be used to compare different versions of the same track, hence Start won't be useful at all, it may be handy when comparing different tracks, especially with modern dance music with fixed tempo. The graph is displayed only when the playback stops, since it depends on the actual position in the host.



Collapse

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



File

File button enables playback from this particular slot into which you can load any supported audio file. When you press this button, the plugin quickly crossfades from whatever is playing now to playback of this file and the button becomes 'pushed', hence selected. When you press it again, the plugin crossfades back to the plugin input. This way you can simply A/B between the plugin input and any external audio file.

 **Rename**

Rename button lets you rename the button. Please note that the button may need to readjust its size to make the name fit, so it's suggested to keep the names short unless you have a high resolution screen at your disposal.

 **Gain**

Gain controls the gain applied to this slot, which you can use to match the input audio level. If the input loudness is different from the loudness of the file being played, you may get fooled by your own ears thinking that the louder one 'sounds better', among other problems. If the loudness is approximately the same however, it is much easier to focus on the sonic qualities only. For that you can use this gain parameter, **ALC**, or both. If ALC is enabled, you can still use gain parameter to adjust the level after the ALC.

 **Set**

Set button sets the **Gain** parameter automatically so that the current slot loudness matches the input loudness. The plugin continually measures loudness of the input and the selected slot, so that it can make both have approximately the same loudness either on-the-fly using **ALC**, or using Gain.

 **ALC**

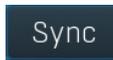
ALC button activates or deactivates the automatic loudness compensation. If the input loudness is different from the loudness of the file being played, you may get fooled by your own ears thinking that the louder one 'sounds better', among other problems. If the loudness is approximately the same however, it is much easier to focus on the sonic qualities only. For that you can use this **Gain** parameter, ALC, or both. The advantage of ALC is that it adjusts the loudness continually. The disadvantage is that it changes over time, obviously. You may also experience temporary glitches - for example, if the slot contains silence and then a quick high level signal, the detector will first measure very low loudness and matches it using high gain, but then it won't react quickly enough for the high level signal, so the temporarily you may experience a short loud burst. For this reason the ALC's gain is limited to -24dB to +24dB. AGC also starts working after about 3 seconds, when the loudness detector has enough information.

 **Start**

Start lets you shift the input file in time so that you can synchronize it with the input. This can be useful if the file is a different version of the same project.

 **Detect**

Detect button lets you automatically detect the delay between the file and the input of this plugin and set the **Delay** parameter automatically. To do that, first enable playback and then press this button. A progress window will be displayed for a few seconds recording both the plugin input and the source. When finished the progress window disappears and the delay is changed to match the difference between the file and the input, if possible. Please note that this feature can detect only very small periods, up to say 2 seconds. So first you need to set the delay manually to match the file with the input approximately and use this feature to get sample accurate value, hence perfect synchronization.

 **Sync**

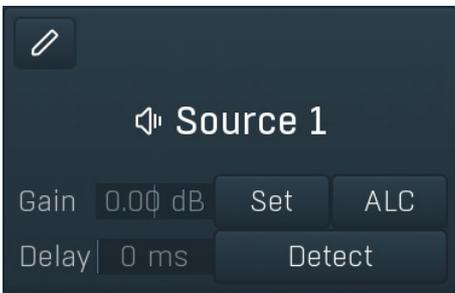
Sync button switches the file slot to synchronized mode. If this is disabled, the plugin's file slot is playing the file between the loop points, the position in the project is not relevant. If this is disabled however, then the file is synchronizing with the project position. This is useful if you want to compare the input with a file containing say a different master of the same recording. The input is technically the same as the file, just processed differently, and you want to compare them. So you enable sync, set the **Delay** to synchronize them approximately and press the **Detect** button to get them perfectly synchronized.

 **Input file path**

Input file path lets you select the input file. Use the browse button on the right. It is an input field, but it is read only, it is provided just in case you need to copy the file name to clipboard for example.

 **Collapse**

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



Source

Source button enables playback from this particular slot to which you can send the signal from any other instance of the plugin. When you press this button, the plugin quickly crossfades from whatever is playing now to this source and the button becomes 'pushed', hence selected. When you press it again, the plugin crossfades back to the plugin input. This way you can simply A/B between the plugin input and any stage of your mix. It is especially useful during mastering to compare the processed signal with a bypassed version. Although you can often use bypass on the individual plugins doing the processing, MCompare provides instant smooth comparison and can compensate for the latency and loudness difference caused by the plugins.



Rename

Rename button lets you rename the button. Please note that the button may need to readjust its size to make the name fit, so it's suggested to keep the names short unless you have a high resolution screen at your disposal.

Gain 0.00 dB

Gain

Gain controls the gain applied to this slot, which you can use to match the input audio level. If the input loudness is different from the loudness of the file being played, you may get fooled by your own ears thinking that the louder one 'sounds better', among other problems. If the loudness is approximately the same however, it is much easier to focus on the sonic qualities only. For that you can use this gain parameter, **ALC**, or both. If ALC is enabled, you can still use gain parameter to adjust the level after the ALC.

Set

Set

Set button sets the **Gain** parameter automatically so that the current slot loudness matches the input loudness. The plugin continually measures loudness of the input and the selected slot, so that it can make both have approximately the same loudness either on-the-fly using **ALC**, or using Gain.

ALC

ALC

ALC button activates or deactivates the automatic loudness compensation. If the input loudness is different than loudness of the file being played, you may get fooled by your own ears thinking that the louder one 'sounds better', among other problems. If the loudness is approximately the same however, it is much easier to focus on the sonic qualities only. For that you can use this **Gain** parameter, ALC, or both. The advantage of ALC is that it adjusts the loudness continually. The disadvantage is that it changes over time, obviously. To prevent accidental short bursts, the ALC's gain is limited to -24dB to +24dB. AGC also starts working after about 3 seconds, when the loudness detector has enough information.

Delay 0 ms

Delay

Delay lets you compensate for the potential latency between the source and the actual plugin input. For example, if you use one instance of MCompare to generate source 1, followed by a linear-phase equalizer (which always has latency) and finally the main MCompare instance for monitoring, the input to this instance will be delayed because of the equalizer's latency. You can still A/B, but the input just won't match the source, so comparison will not be so simple. This delay lets you shift the source in time, so that you can match the latency of the equalizer. Use the **Detect** button to let the plugin calculate the latency automatically.

Detect

Detect

Detect button lets you automatically detect the delay between the source and the input of this plugin and set the **Delay** parameter automatically. To do that, first enable playback, then press this button. A progress window will be displayed for a few seconds recording both the plugin input and the source. When finished the progress window disappears and the delay is changed to match the latency, if possible. If you feel something is not correct, it is possible that the host confused the plugin by mismatched processing, in which case just press the **Panic (!) button** and try again.



Collapse

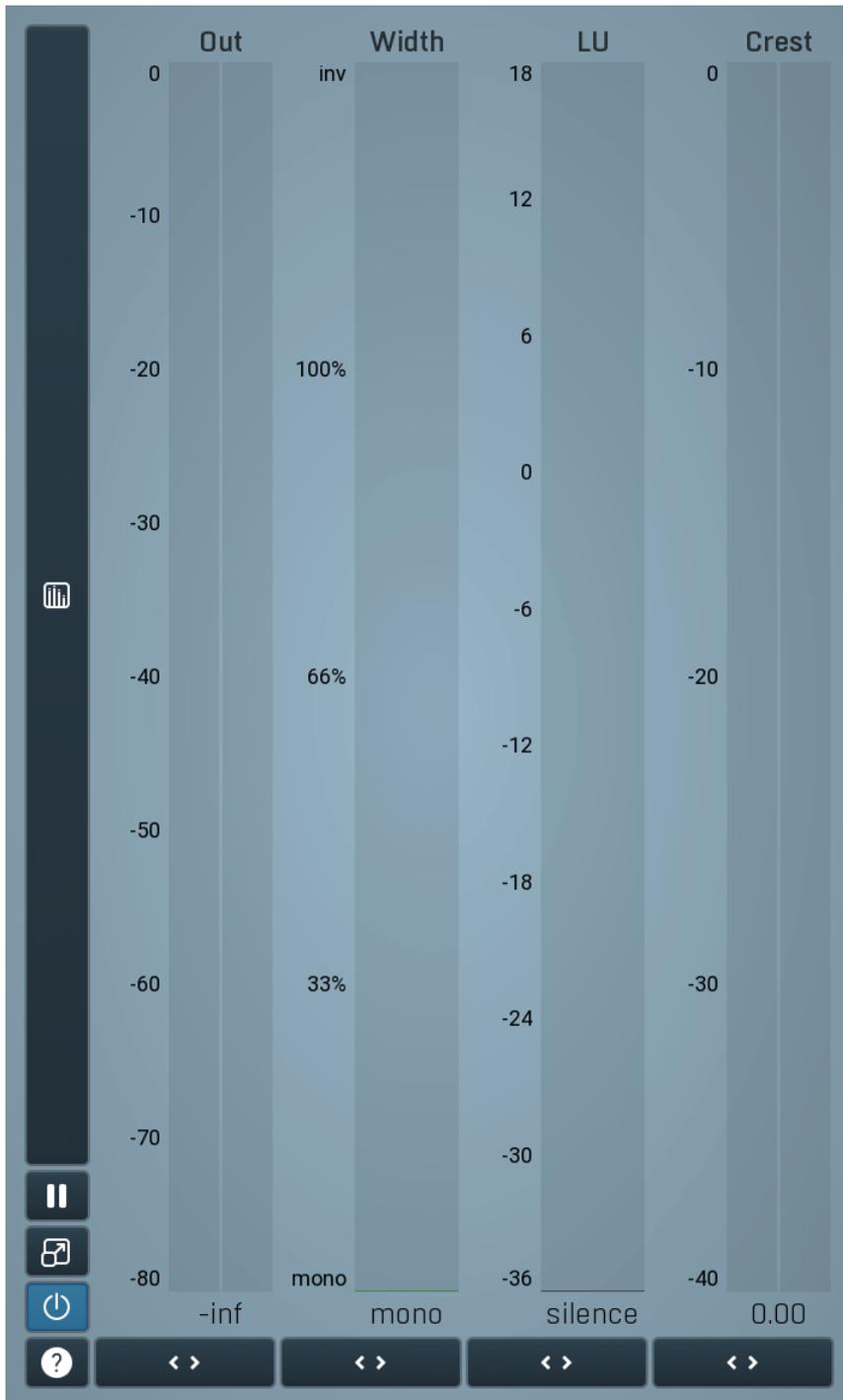
Collapse button minimizes or enlarges the panel to release space for other editors.

Listening

Mute

output

Mute output switch lets you mute the output from this plugin, while sending the input as source to the main instance. This is often useful when comparing different tracks, e.g. different masters placed on different tracks, where each MCompare instance is sending the audio to the main instance you are using to compare the masters.



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.

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.

LU meter shows the output loudness in EBU-18 scale. The loudness metering follows the ITU-R BS.1770-3 and EBU 3341 specifications. The metering units used are LU (Loudness Units) with 0 LU defined as -23 LUFS (LU Full Scale) and you should consider the LU values to be relative - using them to compare the loudness values between different signals. If the difference in loudness between 2 signals is 10 LU, it is approximately 10 dB as well.

Please note that you should still use your ears to judge loudness properly as there is still no accurate model of human loudness perception and every measurement is only an approximation. Loudness perception is also individual.

If you right click on the meter, additional settings will be displayed. Maximum value displays the maximum since the analysis started, rather than the recent maximum. Loudness pre-filtering uses EBU standard filters to simulate human perception. However, you may want to disable this to get more technical measurements.

There are 3 types of loudness measurements, all following the EBU specifications.

Momentary loudness uses an RMS sliding analysis window of 400 milliseconds; therefore it shows quick fluctuations in loudness.

Short-term loudness works in the same way, but uses a window of 3 seconds, therefore it provides more stable loudness measurements.

Integrated loudness shows the overall loudness, hence it is affected by the whole track from the beginning of the playback until you reset it by clicking on the value field. The host may reset it too; it depends on your host.

Please note that the **Integrated loudness** is NOT the same as an averaged loudness, as it ignores quiet passages. Imagine a track which is generally quiet but has a few loud sections. The averaged loudness will be less than the Integrated loudness. Its calculation uses gating to ignore those quiet passages (levels less than 10 LU less than the current ungated level) of the track. Essentially, **Integrated loudness** is a measure of the loudest sections of the track.

Crest meter shows the output crest factor (calculated as RMS divided by peak level), which essentially indicates how extreme the peaks are in the output waveform. The lower the value is, the more peaks there are in the output, the more dynamic it is. If the value reaches 0dB, then the output is over-compressed and flattened and you should consider going easier on the compressor and limiter.

The range from 0dB to -6dB is red and you should prevent your master output from remaining in this range as that would mean it is extremely over-processed. The green range from -6dB to -10dB is the range most that recordings are in, usually jumping below the -10dB.

Please note that others calculate the crest factor as the peak level of the waveform divided by the RMS value of the waveform, and the higher the value, the more peaks there are in the output, the more dynamic it is.



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.



Collapse

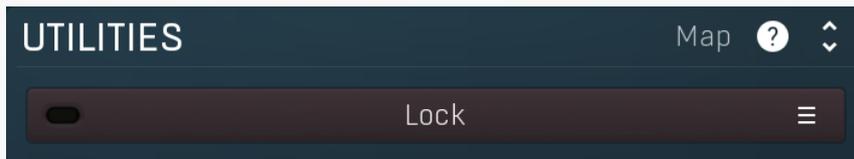
Collapse button minimizes or enlarges the panel to release space for other editors.



Collapse

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

Utilities



Map

Map button displays all current mappings of modulators, multiparameters and MIDI (whichever subsystems the plugin provides).

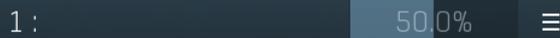


Lock button displays the settings of the global parameter lock. Click on it using your left mouse button to open the Global Parameter Lock window, listing all those parameters that are currently able to be locked. Click on it using your right mouse button or use the **menu button** to display the menu 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.

