

# MUnison



## Overview

MUnison is a powerful plugin, that generates clones of the input signal, slightly modified, and can even generate harmonies from them. The origin of the plugin is in the classic effect of double tracking - recording the same performance twice (or more), which certainly is always slightly different. Playing these back together results in a very wide and powerful sound. MUnison extends this to up to 50 voices.

To make the voices different, it varies several parameters - pitch, stereo placement, formants and it also delays each voice. These parameter changes are generated automatically for each voice for easier workflow; after all, who would (want to) edit parameters of 50 voices. It is also equipped with a simple automatic tuner available via **Auto-speed** parameter.

By default the plugin generates clones of the main voice at approximately the same pitch. However it also provides a smart harmonizer, which lets it generate harmonies in the specified scale. It can even be used live and you can control the scale and requested harmonies by MIDI and automation. For the harmonizer to work, the plugin requires to be able to detect input pitch, hence it works properly only with monophonic signals.

## Easy screen vs. Edit screen

The plugin provides 2 user interfaces - an **easy screen** and an **edit screen**. Use the Edit button to switch between the two.

By default most plugins open on the **easy screen** (edit button released). This screen is a simplified view of the plugin which provides just a few controls. On the left hand side of the plugin you can see the list of available **devices / instruments** (previously called 'active presets'), that is, presets with controls. These controls are actually nothing more than multiparameters (single knobs that can control one or more of the plug-in's parameters and sometimes known as Macro controls in other plug-ins) and are described in more detail later. Each device may provide different controls and usually is intended for a specific purpose. The easy screen is designed for you to be able to perform common tasks, quickly and easily, without the need to use the advanced settings (that is, those available on the Edit screen).

In most cases the devices are highlighted using different text colors. In some cases the colors only mark different types of processing, but in most cases the general rule is that **black/white devices** are the essential ones designed for general use. **Green devices** are designed for a specific task or audio materials, e.g. de-essing or processing vocals in a compressor plugin. **Red devices** usually provide some very special processing or some extreme or creative settings. In a distortion plugin, for example, these may produce an extremely distorted output. **Blue devices** require an additional input, a side-chain or MIDI input usually. Without these additional inputs these **Blue** presets usually do not function as intended. Please check your host's documentation about routing side-chain and MIDI into an effect plugin.

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

By clicking the **Edit button** you can switch the plugin to **edit mode** (edit button pushed). This mode provides all the of the features that the plugin offers. You lose no settings by toggling between edit mode and the easy screen unless you actually change something. This way you can easily check what is "under the hood" for each device, or start with an device and then tweak the plugin settings further.

Devices are factory specified and cannot be modified directly by users, however you can still make your own and store them as normal presets. To do so, configure the plugin as desired, then define each multiparameter and specify its name in its settings. You can then switch to the easy screen and check the user interface that you have created. Once you are satisfied with it, save it as a normal preset while you are on the easy screen. Although your preset will not be displayed or selected in the list of available devices, the functionality will be exactly the same. For more information about multiparameters and devices please check the [online video tutorials](#).

If you are an advanced designer, you can also view both the easy and edit screens at the same time. To do that, hold **Ctrl** key and press the Edit button.

# Edit mode



## Presets

### Presets

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:

- Using "Backup" and "Restore" buttons in each preset window, which produces a single archive of all presets on the computer.
- Using "Export/Import" buttons, which export a single folder of presets for one plugin.
- 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.

## Random

### Randomize

Randomize button (with the text 'Random') generates random settings. Generally, randomization in plug-ins works by selecting random values for all parameters, but rarely achieves satisfactory results, as the more parameters that change the more likely one will cause an unwanted effect. Our plugins employ a smart randomization engine that learns which settings are suitable for randomization (using the existing presets) and so is much more likely to create successful changes.

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

Holding **Ctrl** while clicking the button constrains the randomization engine so that parameters are only modified slightly rather than completely randomized. This is suitable to create small variations of existing interesting settings.

Holding **Alt** while clicking the button will force the engine to use full randomization, which sets random values for all reasonable automatable parameters. This can often result in "extreme" settings. Please note that some parameters cannot be randomized this way.



### Panic

Panic button resets the plugin state. You can use it to force the plugin to report latency to the host again and to avoid any audio problems. For example, some plugins, having a look-ahead feature, report the size of the look-ahead delay as latency, but it is inconvenient to do that every time the look-ahead changes as it usually causes the playback to stop. After you tweak the latency to the correct value, just click this button to sync the track in time with the others, minimizing phasing artifacts caused by the look-ahead delay mixing with undelayed audio signals in your host. It may also be necessary to restart playback in your host.

Another example is if some malfunctioning plugin generates extremely high values for the input of this plugin. A potential filter may start generating very high values as well and as a result the playback will stop. You can just click this button to reset the plugin and the playback will start again.

## Settings

### Settings

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 zipper 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.

## Sleeping

### Sleep indicator

Sleep indicator informs whether the plugin is currently active or in sleep mode. The plugin can automatically switch itself off to save CPU, when there is no input signal and the plugin knows it cannot produce any signal on its own and it generally makes sense. You can disable this in Settings / **Intelligent sleep on silence** both for individual instances and globally for all plugins on the system.

# Globals panel



Globals panel controls the main parameters of the plugin.

### Keep below pitch **Keep below pitch**

Keep below pitch option makes sure that nothing below the current pitch gets processed. This is known to vastly improve pitch shifting for monophonic instruments such as vocals, where there is often lots of rumble below the pitch, which only produce artifacts when shifted.

### Kill below pitch **Kill below pitch**

Kill below pitch is a strongest alternative to **Keep below pitch**, which doesn't let anything below the fundamental frequency pass at all. In essence this should clear any potential low-frequency rumble.

### Process dry **Process dry**

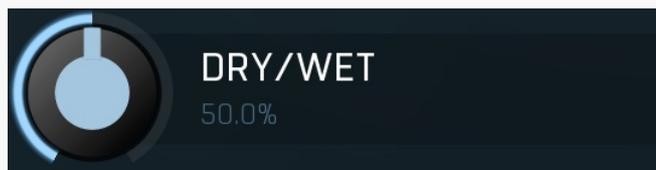
Process dry option makes the plugin apply the formant shifting and auto-tuning to the dry signal as well. You may use it if you want to exploit the automatic tuner for the main vocal for example.



#### Voices

Voices controls the number of artificial voices that the plugin creates. The higher the number, the more CPU power the plugin requires of course. If **Follow voices** switch is enabled, changing voices also varies other parameters in the background to make the sound character appropriate.

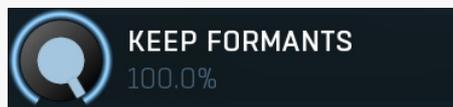
Range: 0.00 to 50.00, default 4.00



#### Dry/Wet

Dry/Wet defines how powerful the effect is, thus this is the ratio between dry and wet signals.

Range: 0.00% to 100.0%, default 50.0%



#### Keep formants

Keep formants makes the algorithm try to reduce the altering of the spectral envelope. Most natural instruments including the human voice do not change the complete spectrum when changing pitch. However the pitch shifting algorithm does that since it does not follow the complex physical laws, which results in the typical mickey-mouse effect etc. Keep formants tries to approximate these physical laws and usually results in more natural results.

Range: 0.00% to 100.0%, default 100.0%



#### Formant shift

Formant shift lets you manually alter the formant information (in semitones), which generally results in no pitch shifting, but can create the mickey-mouse effect for example.

Range: -12.00 to +12.00, default 0



**AUTO SPEED**  
Off

### Auto speed

Auto speed controls the speed of the integrated automatic tuner. It is applied to the artificial voices only, the dry input stays intact. When the harmonizer is enabled, it follows the scale in it; otherwise it assumes a chromatic scale.

Range: Off to 100.0%, default Off



**FOLLOW VOICES**  
2.50x

### Follow voices

Follow voices controls how much certain parameters follow the number of voices. The purpose of the plugin is to create artificial voices that sound similar to the original (unless harmonized), yet sort of different, so that the human brain recognizes them as separate voices. It does that by detuning them, changing the formants etc., but these values usually need to be different for different number of voices. For example, if you have just 4 voices, they all need to be fairly in tune, otherwise one would say that some of them sing really badly. But if you have 40 voices, which are all in tune, they could just sound too similar. You can then compensate for that by increasing the pitch depth, or let the plugin do that for you automatically.

This value controls the proportion between 4 and 40 voices. For example, if the **pitch depth** is set to 20% and this value is 3x, then for 4 voices or less the pitch depth will indeed be 20%, but for 40 voices it will be 3x that much, hence 60%.

Range: 1.00x to 5.00x, default 2.50x



**THRESHOLD**  
-80.0 dB

### Threshold

Threshold defines the minimum level at which the plugin generates voices. It is useful to avoid making unison noise.

Range: silence to 0.00 dB, default -80.0 dB



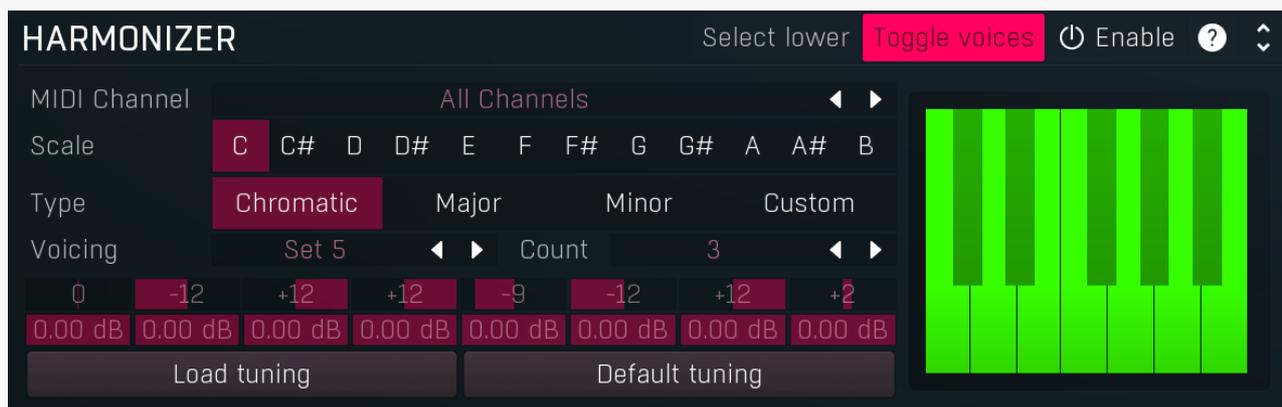
**OUTPUT**  
0.00 dB

### Gain

Gain defines power modification applied to the wet signal. You can use it to make the processed signal similar in loudness if needed, hence making dry/wet easier to use.

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

## Harmonizer panel



**HARMONIZER** Select lower **Toggle voices** Enable ?

MIDI Channel All Channels

Scale C C# D D# E F F# G G# A A# B

Type Chromatic Major Minor Custom

Voicing Set 5 Count 3

0 -12 +12 +12 -9 -12 +12 +2

0.00 dB 0.00 dB

Load tuning Default tuning

Harmonizer panel provides an integrated smart harmonizer, which can distribute the voices into intervals from the original. Normally the harmonizer is disabled, hence also the pitch detector is disabled and the processor creates unison voices. But you can exploit the harmonizer to, for example, create automatic backup voices for your singing. The harmonizer works well only for monophonic audio signal of course.

### Select lower

Select lower switch makes the harmonizer select the lower note if it is not obvious which one to choose. For example, if you select C major scale, enable voice +3 and play C, the harmonizer is instructed to play C + 3, which is D#, but that is not in the C major scale. If this switch is disabled, then it will select E, if enabled, the plugin will play D.

### Toggle voices

Toggle voices switch defines how the harmonizer custom voicing MIDI control works. When this switch is on, pressing a key in the lowest octave enables a key in the custom voicing, pressing it again disables it. That is usually good for live performances, but not so good for arranging in the studio, because when you rewind, you won't know what should be enabled and what shouldn't. So by disabling

this option you make the plugin enable a key when the corresponding key is pressed and disable it when it is released.

MIDI Channel All Channels ◀ ▶ **MIDI Channel**

MIDI Channel selects the channel used to control the harmonizer features. **None** disables the MIDI Control.

Scale C C# D D# E F F# G G# A A# B **Scale**

Scale controls the root key for the current scale, from which the harmonizer is allowed to select notes to play. MIDI input can be used to select the scale - send a MIDI note on any octave (except for the 2 lowest ones, which are used for other purposes) to the plugin.

Type Chromatic Major Minor Custom **Type**

Type defines the type of the scale. You can either use one of the predefined basic types or create a custom one. MIDI input can be used to enable or disable the custom scale - send a MIDI note on the lowest octave to the plugin.

Voicing Set 5 ◀ ▶ **Voicing**

Voicing parameter lets you switch between multiple sets of voicings. Each voicing set is controlled by the list of pitch shifts below. MIDI input can be used to select voicing - send a MIDI note on the 2nd lowest octave to the plugin.

Range: Set 1 to Set 12, default Set 5

Count 3 ◀ ▶ **Count**

Count controls the number of voices in the current voicing set, hence the number of pitch shift fields below.

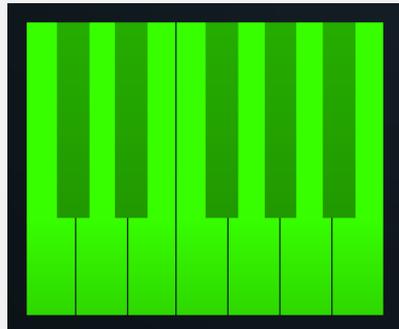
0 **Pitch shift field**

Pitch shift field controls the pitch shift in semitones for the n'th voice.

0.00 dB **Weight field**

Weight field controls the weight of the pitch shift above it corresponding to the n'th voice.

## Piano keyboard



Piano keyboard controls the custom scale and highlights the keys that are allowed to play. Please note that it always displays the case for the C scale.

Load tuning

### Load tuning

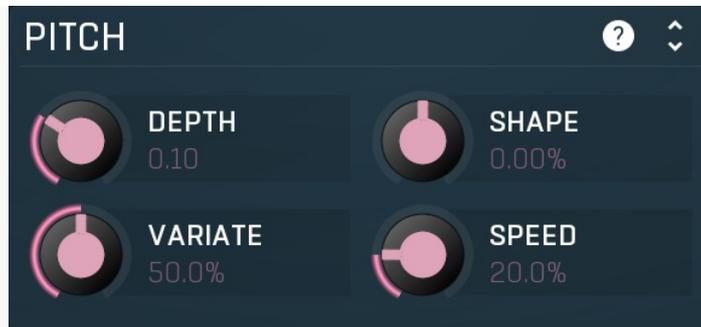
Load tuning button lets you load TUN files containing custom micro-tuning, which will replace the default equal temperament tuning (in which the logarithmic distance between every 2 semitones is exactly the same).

Default tuning

### Default tuning

Default tuning button restores the default equal temperament tuning.

## Pitch panel



Pitch panel controls the pitch of the artificial unison voices.



### Depth

Depth controls the detuning of the artificial voices. It is the most essential parameter that makes the voices different from the original and from each other. 0% makes the voices almost the same producing sort of phasey results. 100% makes them be detuned by a semitone up and down, so such a high value might be used creatively for creating a crowd of shouting people for example, but the results are unlikely to be harmonic anymore.

Range: 0.00 to 1.00, default 0.10



### Shape

Shape controls the shape of the frequencies for each unison voice. At 0% the frequencies will be equally displaced around the base pitch. At +100% most of the frequencies will be placed closely around the base pitch. Conversely, at -100%, they will tend towards the extremes and be as far from the base pitch as possible, controlled by the **Depth** parameter, hence providing more out-of-tune sound.

Range: -100.0% to 100.0%, default 0.00%



### Variate

Variate parameter controls how much the detuning changes in time. No real-world instrumentalist performs in the same way every time, variation tries to simulate it. It randomly changes the parameter slightly in time providing the natural performance that one would get by recording the same instrument multiple times.

Range: 0.00% to 100.0%, default 50.0%

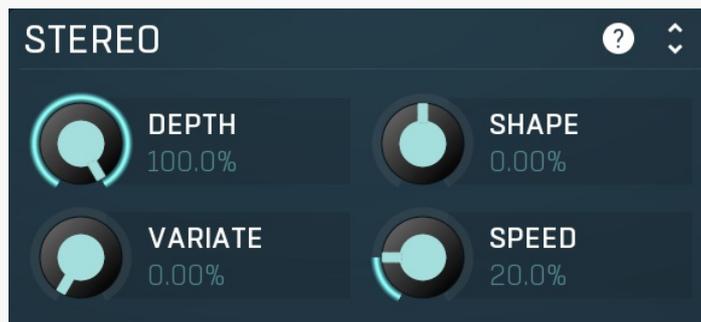


### Speed

Speed controls the speed of the variation, how quickly the detuning changes in time. 0% makes the detuning initially random, yet not changing at all. 100% makes the changes extremely fast, almost noise like. That might be handy for simulating a huge shouting crowd for example.

Range: 0.00% to 100.0%, default 20.0%

## Stereo panel



Stereo panel controls the stereo field of the artificial unison voices.



DEPTH  
100.0%

### Depth

Depth controls the stereo placement of the artificial voices. It is an important parameter, which essentially defines the stereo width. 0% places the voices to the center, so if the source was monophonic, the output will be as well. 100% makes the voices placed evenly from the extreme left to the extreme right resulting in a wide stereo field.

Range: 0.00% to 100.0%, default 100.0%



SHAPE  
0.00%

### Shape

Shape controls the stereo placement of the individual voices. At 0% the voices will be equally displaced from left to right. At -100% most of the voices will be placed closely around the center. Conversely at +100% they will tend towards the extremes resulting in an even wider stereo field.

Range: -100.0% to 100.0%, default 0.00%



VARIATE  
0.00%

### Variate

Variate parameter controls how much the stereo placement changes in time. No real-world instrumentalist performs in the same way every time and is at the same physical place every time, variation tries to simulate it. It randomly changes the parameter slightly in time providing the natural performance that one would get by recording the same instrument multiple times.

Range: 0.00% to 100.0%, default 0.00%



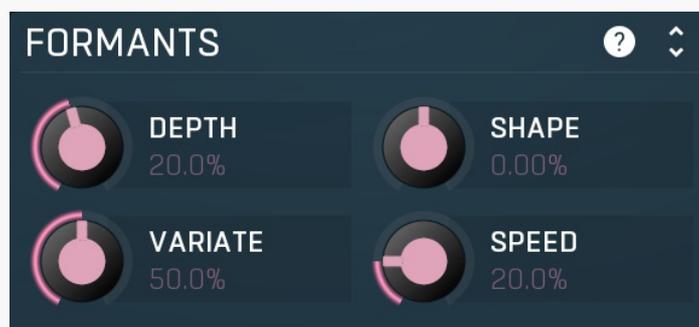
SPEED  
20.0%

### Speed

Speed controls the speed of the variation, how quickly the stereo placement changes in time. 0% makes the placement initially random, yet not changing at all. 100% makes the changes extremely fast, almost noise-like. That might be handy for simulating a huge shouting crowd for example.

Range: 0.00% to 100.0%, default 20.0%

## Formants panel



Formants panel controls the formant changes of the artificial unison voices. Formants essentially define the character of the sound. When processing vocals the formants define if a certain person sounds like a woman, a man or a child for example.



DEPTH  
20.0%

### Depth

Depth controls the formants of the artificial voices. It is commonly used to make each voice sound slightly different. 0% makes each voice sound the same, spectrally, hence if you want to simulate double tracking of a vocalist, you might use 0% as it is very unlikely a single person would have different formant characteristics for each take. 100% produces the maximum formant differences between the voices, in a way you might say it tries to simulate different people when processing vocals for example. However note that formant changes might often sound artificial, so don't overuse this control unless you are after a creative effect.

Range: 0.00% to 100.0%, default 20.0%



SHAPE  
0.00%

### Shape

Shape controls the formants of the individual voices. At 0% the voices will be equally displaced lowest and highest formant changes. At -100% most of the voices will be placed closely around the center, hence sounding more like the original. Conversely at +100% they will

tend towards the extremes, hence sounding more different from the original.

Range: -100.0% to 100.0%, default 0.00%



### Variate

Variate parameter controls how much the formants change in time. Although formant characteristics do not generally change for realworld instruments, varying formants in time supports the need for them to sound different, hence it is nonzero by default anyway.

Range: 0.00% to 100.0%, default 50.0%

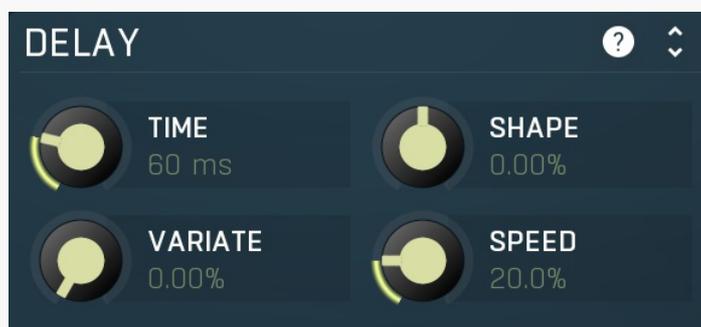


### Speed

Speed controls the speed of the variation, how quickly the formants change in time. 0% makes the placement initially random, yet not changing at all. 100% makes the changes extremely fast, almost noise-like. That might be handy for simulating a huge shouting crowd for example.

Range: 0.00% to 100.0%, default 20.0%

## Delay panel



Delay panel controls the delay for each of the artificial unison voices.



### Time

Time controls the maximum delay for each voice. It is an important parameter, which essentially defines the displacement in time and in extreme case artificial reverberation. 0ms disables any delay for each voice, making them sound at the same time. That is essentially unnatural as it would happen only if a singer, when processing voice for example, could perform at exactly the same time in each take, which never happens. As a result this supports phasing artifacts for example. Increasing this value gradually makes the sound bigger and bigger until a point when the ear is able to separate each voice as a separate melody, which not useful normally, but can be used for creative effects.

Range: 0 ms to 1000 ms, default 40 ms



### Shape

Shape controls the delay of the individual voices. At 0% the voices will be equally displaced from 0ms to the **Time**. At -100% most of the voices will be placed closely around the minimum. Conversely at +100% they will tend towards the maximum resulting in more roomy sound.

Range: -100.0% to 100.0%, default 0.00%



### Variate

Variate parameter controls how much the delay changes in time. No realworld instrumentalist performs in the same way every time, variation tries to simulate it. It randomly changes the parameter slightly in time providing the natural performance one would get by recording the same instrument multiple times. For technical reasons variation to delay provides only microscopic changes, which can however make a huge difference sound-wise.

Range: 0.00% to 100.0%, default 0.00%



SPEED

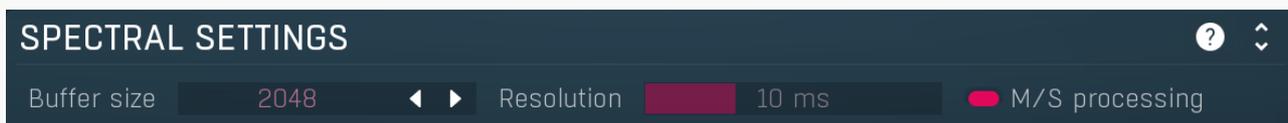
20.0%

### Speed

Speed controls the speed of the variation, how quickly the delay changes in time. 0% makes the delay initially random, yet not changing at all. 100% makes the changes extremely fast, almost noise-like. That might be handy for extreme sound effects.

Range: 0.00% to 100.0%, default 20.0%

## Spectral settings panel



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



Buffer size

2048



### Buffer size

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

Range: 256 to 16384, default 2048



Resolution

10 ms

### Resolution

Resolution defines how accurately the processor can analyze the audio. The lower the resolution, the more CPU is needed, but also more of the time domain characteristics are preserved, hence potentially higher audio quality.

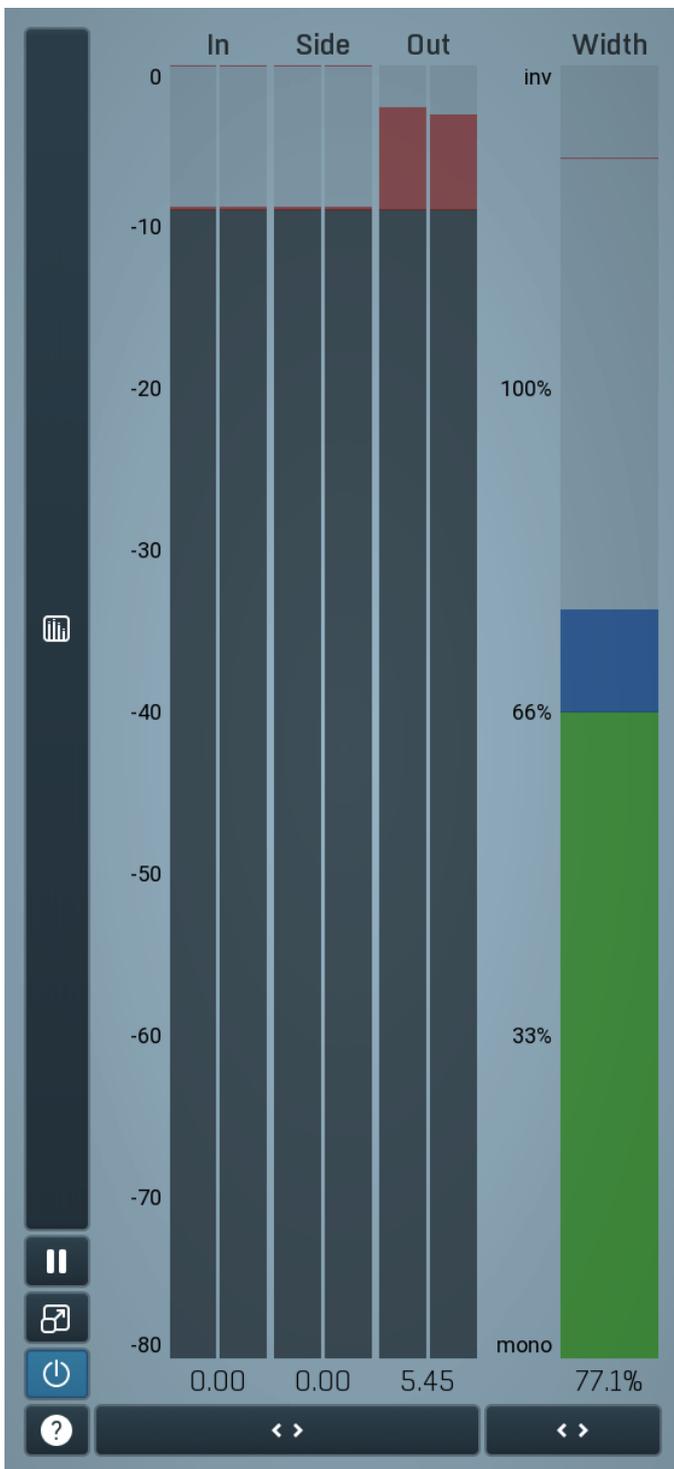
Range: 1.0 ms to 100 ms, default 25 ms



M/S processing

### M/S processing

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



### Global meter view

Global meter view provides a powerful metering system. If you do not see it in the plug-in, click the **Meters** or **Meters & Utilities** button to the right of the main controls. The display can work as either a classical level indicator or, in time graph mode, show one or more values in time. Use the first button to the left of the display to switch between the 2 modes and to control additional settings, including pause, disable and pop up the display into a floating window. The meter always shows the actual channels being processed, thus in M/S mode, it shows mid and side channels.

In the classical level indicators mode each of the meters also shows the recent maximum value. Click on any one of these values boxes to reset them all.

**In meter** indicates the total input level. The input meter shows the audio level before any specific processing (except potential oversampling and other pre-processing). It is always recommended to keep the input level under 0dB. You may need to adjust the previous processing plugins, track levels or gain stages to ensure that it is achieved.

As the levels approach 0dB, that part of the meters is displayed with **red** bars. And recent peak levels are indicated by single bars.

**Out meter** indicates the total output level. The output meter is the last item in the processing chain (except potential downsampling and other post-processing). It is always recommended to keep the output under 0dB.

As the levels approach 0dB, that part of the meters is displayed with **red** bars. And recent peak levels are indicated by single bars.

**Width meter** shows the stereo width at the output stage. This meter requires at least 2 channels and therefore does not work in mono mode. Stereo width meter basically shows the difference between the mid and side channels.

When the value is **0%**, the output is monophonic. From 0% to 66% there is a green range, where most audio materials should remain.

**From 66% to 100%** the audio is very stereophonic and the phase coherence may start causing problems. This range is colored blue. You may still want to use this range for wide materials, such as background pads. It is pretty common for mastered tracks to lie on the edge of green and blue zones.

**Above 100%** the side signal exceeds the mid signal, therefore it is too monophonic or the signal is out of phase. This is marked using red color. In this case you should consider rotating the phase of the left or right channels or lowering the side signal, otherwise the audio will be highly mono-incompatible and can cause fatigue even when played back in stereo.

For most audio sources the width is fluctuating quickly, so the meter shows a 400ms average. It also shows the temporary maximum above it as a single coloured bar.

If you right click on the meter, you can enable/disable loudness pre-filtering, which uses EBU standard filters to simulate human perception. This may be useful to get a more realistic idea about stereo width. However, since humans perceive the bass spectrum as lower than the treble, this may hide phase problems in that bass spectrum.



### Time graph

Time graph button switches between the metering view and the time-graphs. The metering view provides an immediate view of the current values including a text representation. The time-graphs provide the same information over a period of time. Since different time-graphs often need different units, only the most important units are provided.



### Pause

Pause button pauses the processing.



### Popup

Popup button shows a pop-up window and moves the whole metering / time-graph system into it. This is especially useful in cases where you cannot enlarge the meters within the main window or such a task is too complicated. The pop-up window can be arbitrarily resized. In metering mode it is useful for easier reading from a distance for example. In time-graph mode it is useful for getting higher accuracy and a longer time perspective.



### Enable

Enable button enables or disables the metering system. You can disable it to save system resources.



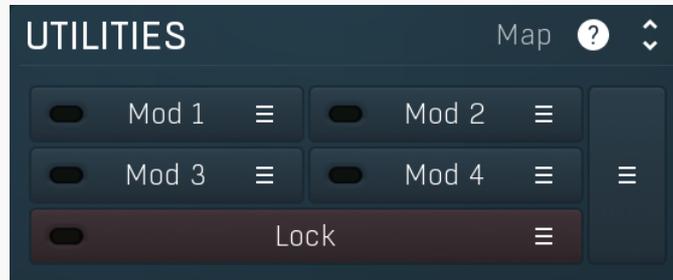
### Collapse

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



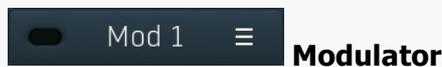
Collapse button minimizes or enlarges the panel to release space for other editors.

## Utilities



### Map

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



Modulator button displays settings of the modulator. It also contains a checkbox, to the left, which you can use to enable or disable the modulator. Click on it using your right mouse button or use the **menu button** to display an additional menu with learning capabilities - as described below.

### Menu

Menu button shows the **smart learn** menu. You can also use the right mouse button anywhere on the modulator button.

**Learn** activates the learning mode and displays "REC" on the button as a reminder, **Clear & Learn** deletes all parameters currently associated with the modulator, then activates the learning mode as above. After that every parameter you touch will be associated to the modulator along with the range that the parameter was changed. Learning mode is ended by clicking the button again.

In smart learn mode the modulator does not operate but rather records your actions. You can still adjust every automatable parameter and use it normally. When you change a parameter, the plugin associates that parameter with the modulator and also records the range of values that you set.

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



Menu button displays additional menu containing features for modulator presets and randomization.



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 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.

