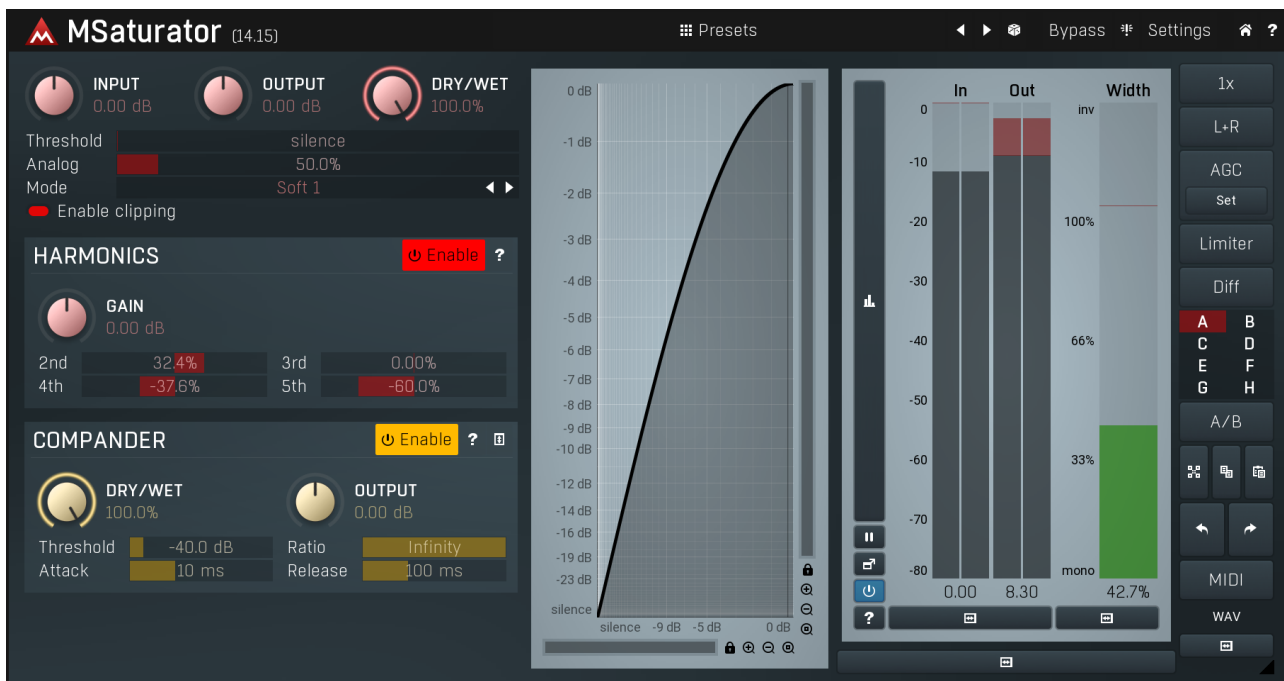


# MSaturator



## 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, using the arrow buttons or by using a combination of the arrow keys and Enter on your keyboard. You can also manage the directory structure, store new presets, replace existing ones etc. Presets are global, so a preset saved from one project, can easily be used in another.

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

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

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

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



### Left arrow

Left arrow button loads the previous preset.



### Right arrow

Right arrow button loads the next preset.



### Randomize

Randomize button loads a random preset.



### Panic

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

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

will start again.

## Settings

### Settings

Settings button shows a menu with additional settings of the plugin. Here is a brief description of the separate items.

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

There are 4 groups of settings, each section has its own detailed help information: **GUI & Style** enables you to pick the GUI style for the plug-in and the main colours used for the background, the title bars of the windows and panels, the text and graphs area and the highlighting (used for enabled buttons, sliders, knobs etc).

**Advanced settings** configures several processing options for the plug-in.

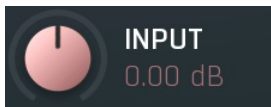
**Dry/wet affects** determines, for Multiband plug-ins, which multiband parameters are affected by the Global dry/wet control.

**Smart interpolation** adjusts the interpolation algorithm used when changing parameter values; the higher the setting the higher the audio quality and the lower the chance of zippering noise, but more CPU will be used.



### WWW

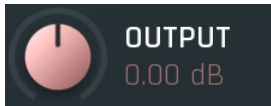
WWW button shows a menu with additional information about the plugin. You can check for updates, get easy access to support, MeldaProduction web page, video tutorials, Facebook/Twitter/YouTube channels and more.



### Input gain

Input gain defines the input gain.

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



### Output gain

Output gain defines the gain applied to the output.

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



### Dry/wet

Dry/wet defines ratio between dry and wet signals. 100% means fully processed, 0% means no processing at all.

Range: 0.00% to 100.0%, default 100.0%



### Threshold

Threshold determines the minimal signal level above which the effect starts to apply. By lowering the threshold you increase loudness and also distortion.

Range: silence to 0.00 dB, default silence



### Analog

Analog determines the amount of even harmonics added to the signal in addition to the main saturation, which is typical for analogue saturation. The amount of the harmonics is dependent on the saturation signal, unlike the full harmonic control in the **Harmonics** panel, which is completely independent of the actual saturation processing.

Range: 0.00% to 500.0%, default 50.0%



### Mode

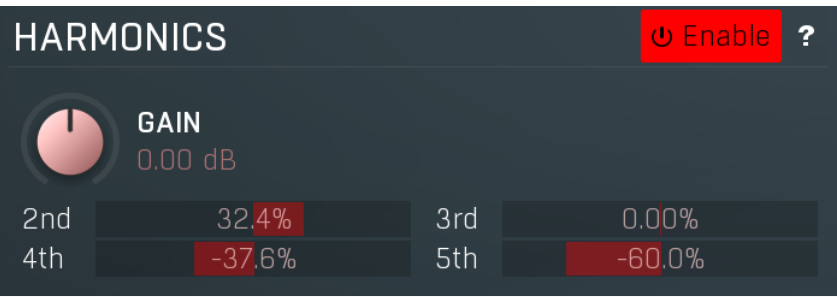
Mode defines the limiting shape. When disabled the whole saturation unit is disabled leaving only the harmonics processor.



### Enable clipping

Enable clipping activates the clipper performed after the saturation and before the harmonics generator.

## Harmonics panel



Harmonics panel provides a generator of additional harmonics. This processing is applied after the saturation. It is well-known phenomenon, that transistors and digital shaping devices generate only odd harmonics (3rd, 5th), while tubes, tapes and other nonlinear analog devices create even harmonics (2nd, 4th) too.



**Gain**

Gain defines the input gain when the harmonics control is enabled. It may come in handy as the character of each harmonic depends on the input level and also the clipper being the last module in the plugin could cause hard digital distortion if overdriven by generated harmonics.

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



**2nd**

2nd controls the amount of 2nd harmonic typical for tubes.

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



**3rd**

3rd controls the amount of 3rd harmonic typical for transistors.

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



**4th**

4th controls the amount of 4th harmonic typical for tubes.

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

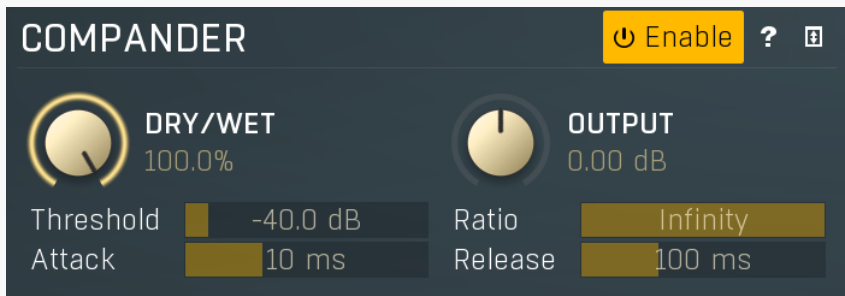


**5th**

5th controls the amount of 5th harmonic typical for transistors.

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

# Compander panel



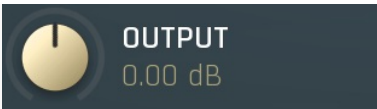
Compander panel controls the integrated compander dynamics section, which works like this: a compressor processes the input so that the levels are more stable for the saturation section and afterwards the compressor effect is undone. This feature greatly improves the saturation stability and produces sort of analogue behaviour, which is generally dynamic.



**Dry/wet**

Dry/wet defines the ratio between dry and wet signals for the compressor. 100% means fully processed, 0% means no processing at all. This feature essentially provides a modern way to do so-called parallel (or 'New York') compression. Essentially there are main 2 approaches to compression - A) set the threshold high, so that it affects everything above it, B) set the threshold low and use dry/wet to actually lower the effect of compression, which provides an easy way to control the amount of compression without too much editing of the more advanced parameters. Please note that lowering ratio does NOT have the same effect as lowering dry/wet in most cases.

Range: 0.00% to 100.0%, default 100.0%



## Output gain

Output gain defines the power modification applied to the output signal. The compressor generally lowers the level, it is reasonable to amplify the output, so that the saturator can produce the desired effect. Since the gain is undone by the compander output, it mostly controls the amount of saturation actually.

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



## Threshold

Threshold determines the minimal signal level above which the effect is applied.

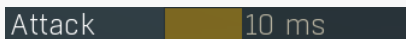
Range: -80.0 dB to 0.00 dB, default -40.0 dB



## Ratio

Ratio defines the compression ratio of the input signal above the threshold.

Range: 1.00 : 1 to Infinity, default Infinity



## Attack

Attack defines the attack time, that is how quickly the level detector increases the measured input level. When the input peak level is higher than the current level measured by the detector, the detector moves into the attack mode, in which the measured level is increased depending on the input signal. The higher the input signal, or the shorter the attack time, the faster the measured level rises. Once the measured level exceeds the **Threshold** then the dynamics processing (compression, limiting, gating) will start.

There must be a reasonable balance between attack and **release** times. If the attack is too long compared to the release, the detector will tend to keep the measured level low, because the release would cause that level to fall too quickly. In most cases you may expect the attack time to be shorter than the release time.

To understand the working of a level detector, it is best to cover the typical cases:

*In a **compressor** the attack time controls how quickly the measured level moves above the threshold and the processor begins compressing. As a result, a very short attack time will compress even the beginning transient of a snare drum for example, hence it would remove the punch. With a very long attack time the measured level may not even reach the threshold, so the compressor may not do anything.*

*In a **limiter** the attack becomes a very sensitive control, defining how much of the signal is limited and how much of it becomes saturated/clipped. If the attack time is very short, limiting starts very quickly and the limiter catches most peaks itself and reduces them, providing lower distortion, but can cause pumping. On the other hand, a higher attack setting (typically above 1ms) will let most peaks through the limiter to the subsequent in-built clipper or saturator, which causes more distortion of the initial transient, but less pumping.*

*In a **gate** the situation is similar to a compressor - the attack time controls how quickly the measured level can rise above the threshold at which point the gate opens. In this case you will usually need very low attack times, so that the gate reacts quickly enough. The inevitable distortion can then be avoided using look-ahead and hold parameters.*

*In a modulator, the detector is driving other parameters, a filter cut-off frequency for example, and the situation really depends on the target. If you want the detector to react quickly on the input level rising, use a shorter attack time; if you want it to follow the flow of the input signal slowly, use longer attack and release times.*

Range: 0 ms to 1000 ms, default 10 ms



## Release

Release defines the release time, that is how quickly the level detector decreases the measured input level. The shorter the release time, the faster the response is. Once the attack stage has been completed, when the input peak level is lower than the current level measured by the detector, the detector moves into the release mode, in which the measured level is decreased depending on the input signal. The lower the input signal, or the shorter the release time, the faster the measured level drops. Once the measured level falls under the **Threshold** then the dynamics processing (compression, limiting, gating) will stop.

There must be a reasonable balance between **attack** and release times. If the attack is too long compared to release, the detector would tend to keep the level low, because release would cause the level to fall too quickly. Hence in most cases you may expect the attack time to be shorter than the release time.

To understand the working of a level detector, it is best to cover the typical cases:

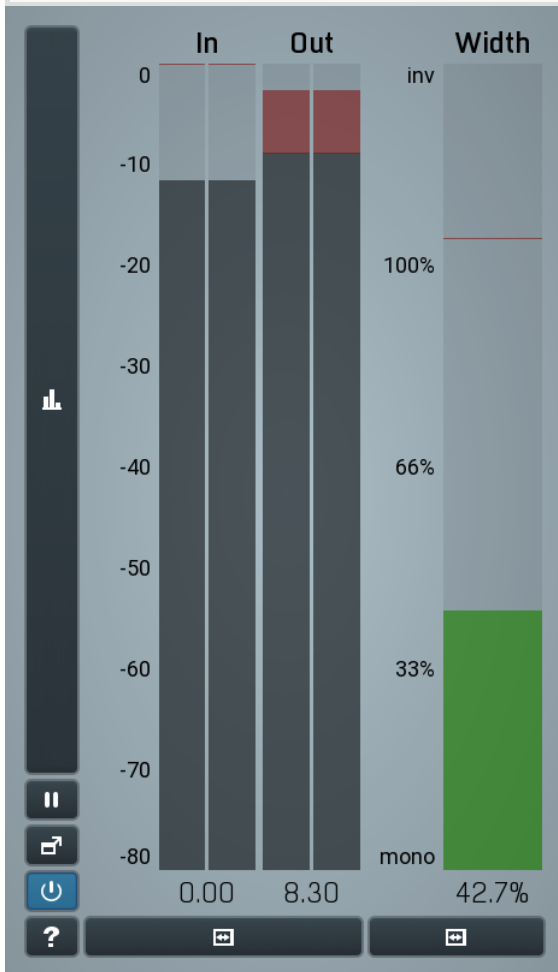
*In a **compressor** the release time controls how quickly the measured level falls below the threshold and the compression stops. As a result a very short release time makes the compressor stop quickly, for example, leaving the sustain of a snare drum intact. On the other hand, a very long release keeps the compression working longer, hence it is useful to stabilize the levels.*

*In a **limiter** the release time keeps the measured level above the limiter threshold causing the gain reduction. Having a very long release time in this case doesn't make sense as the limiter would be working continuously and the effect would be more or less the same as simply decreasing the input gain manually. However too short a release time lets the limiter stop too quickly, which usually causes distortion as the peaks through the limiter to the subsequent in-built clipper or saturator. Hence release time is used to avoid distortion at the expense of decreasing the output level.*

*In a **gate** the situation is similar to a compressor - the release time controls how quickly the measured level can fall below the threshold at which point the gate closes. Having a longer release time in a gate is a perfectly acceptable option. The release time will basically control how much of the sound's sustain will pass.*

*In a modulator, the detector is driving other parameters, a filter cut-off frequency for example, and the situation really depends on the target. If you want the detector to react quickly on the input level falling, use a shorter release time; if you want it to follow the flow of the input signal slowly, use longer attack and release times.*

Range: 1.0 ms to 10000 ms, default 100 ms



### 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 upsampling 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.



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



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