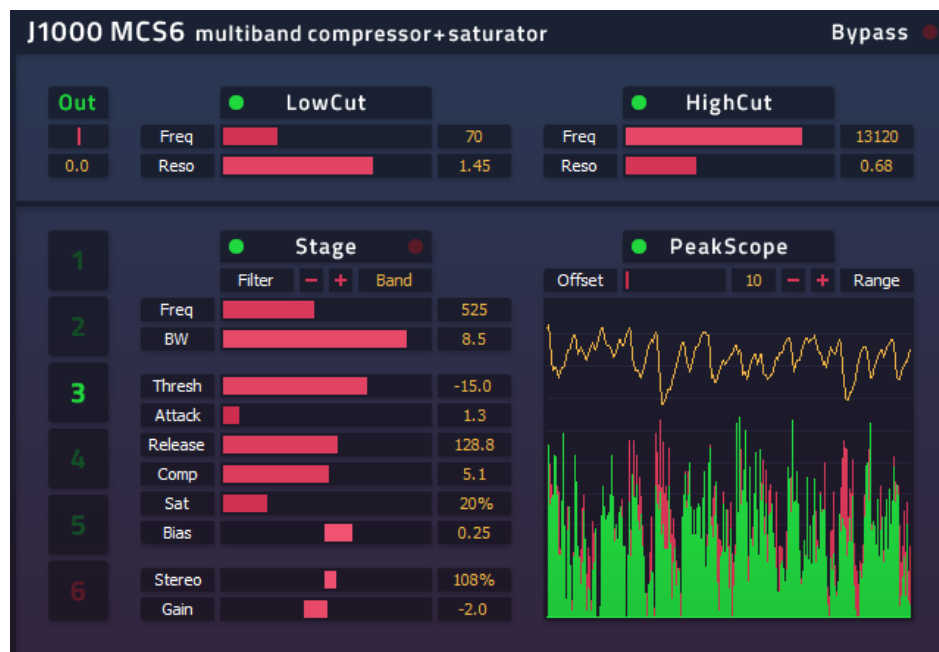


MCS6 v1.0

MULTIBAND COMPRESSOR AND SATURATOR



- Six serial stages for compression and saturation;
- Full range, low-pass, high-pass, band-pass and notch filtering;
- PeakScope display showing I/O peaks and gain reduction;
- Smooth compression envelope with automatic soft-knee response curve;
- Subtle biased (asymmetric) saturation dependent on compression;
- Stereo width and gain control for each stage;
- Main pair of low-cut and high-cut filters with adjustable resonance;

INTRODUCTION

MCS6 is a highly versatile audio processor. It is designed for users who already have decent knowledge about compression and saturation. If you are still getting accustomed with them it's better that you avoid relatively complex processing, because you can easily degrade sound quality instead of improve it. That being said, if you know what you're doing and hear differences between dry and wet sound (including unwanted side effects), MCS6 can be very helpful and powerful audio tool, both for full mixes and individual tracks.

To install it just copy MCS6.DLL file into your VST plugins folder. It is compatible only with 32-bit VST hosts, but there is possibility, through bridging software, to use it also in 64-bit hosts.

The upper, smaller part of interface consists of In/Out gain controls and two cut filters. Clicking on green In/Out label allows separate gain adjustments within +/-10 dB range. Cut filters are two-pole (12dB/oct) with adjustable resonance. The lower part of interface is where the most of action happens. It has six independent and identical sets of controls that are switched by clicking on numbers on the left, highlighted being the current one. Importantly, each stage's output feeds the next stage's input, so the signal flow is serial, not parallel like in classic multiband compressors. If stage is active it's corresponding number is green, if it's inactive number is red, for quick overview how many stages are still unused and available.

The display on the right shows input (red) and output (green) peaks and gain reduction curve (yellow) for each stage. It helps you to visualize what's going on, because sometimes it's hard to hear subtle compression/saturation effects. Still, the final judgement should be based on hearing, not on these visual aids.

For start it's best to check out the presets that show most of the basic usages. After you get the idea how the signal flows and how to set controls, you are hardly ever going to need presets, so there is no preset manager. Every audio signal has it's own properties and hence different processing requirements.

CONTROL	FUNCTION
In / Out	Input and output gain for the whole plugin, switched by clicking on In/Out label.
Green LED	On/off switch for corresponding element.
Red LED	There are two red (blinking) LEDs that control two different things, but share the same property of being temporary. First one, in the upper-right corner, is global bypass buttons that disables audio processing completely. Second one is located next to the Stage label and activates filter solo mode that outputs only wet signal for that stage, so you can easily hear what's being processed and how much.
Freq	Frequency (Hz) at which filtering operates.
Reso	Resonance of LowCut and HighCut filters. Lower values produce softer curve, higher ones produce noticeable spectrum peak. See graph 1 on next page.
1 2 3 4 5 6	Active stage buttons (green is active, red is inactive, highlighted is the current one).
Filter	Type of filtering applied at input of each stage. There are four types of filter available (graph 2): Band, Notch, Low(pass) and High(pass) or it can be turned off for full range operation. Filters are complementary, so you can switch between them to hear what's being processed and what isn't. Band is opposite of Notch, Low is opposite of High, or mathematically speaking: Band + Notch = Low + High = Full spectrum.
BW	Bandwidth of filter, actually an inverted Q factor. Higher values result in softer and wider curve, lower ones in sharper and narrower curve. The more bandwidth you set, more signal gets passed on to compressor and saturator. Low and High filter types don't have BW control.
Thresh	Standard compressor threshold level adjusted in dBFS.
Attack	Compressor attack time, shown in milliseconds.
Release	Compressor release time. Internal sidechain envelope that controls compression is smoothed, so its artifacts are practically inaudible even at lowest attack and release times.
Comp	Amount of compression being applied above threshold. Unlike classic compression ratio which produces linear transfer function (or with soft knee around threshold), MCS6 compressor is always producing curved response. That's why the amount can't be expressed in usual 1:X format - dynamics are gradually compressed more as the signal passes the threshold and there is no fixed final ratio.
Sat	Saturation amount proportional to compression, shown as a percentage. To avoid static audio distortion, MCS6 uses proportional saturation that is always linked to compression. First signal gets compressed, then it passes through saturator. This method produces more saturation at starting edges of audio signal and less as the compressor kicks in, which results in more transparent and musically pleasing effect.
Bias	Positive/negative symmetry of saturation curve. At default position (0.00) both sides of signal get equally saturated producing only odd harmonics. Positive values saturate more the positive side and negative values the negative side, both introducing even harmonics into the output.
Stereo	Stereo width of each stage. If stage filter is off stereo control affects the whole signal and if filtering is on it affects only the filtered signal. 100% leaves original width, 200% doubles it and 0% produces mono.
Gain	Output gain of each stage. It acts as parametric EQ gain if stage filtering is on.
Offset	Offsets the display level to show lower/quieter peaks i.e. it moves them upwards.
Range	PeakScope display dynamic range, shown in decibels (5, 10, 20 and 40dBs).

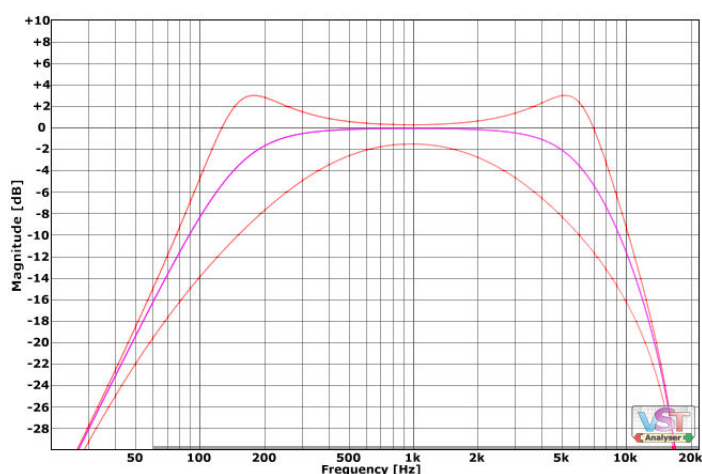
USAGE TIPS

Using MCS6 efficiently takes some time getting used to it's layout and signal flow. Since it has serial instead of parallel structure like classic multiband compressors, it matters what's being done at which stage. Here are some usage tips and suggestions that can speed up the work and get the best results:

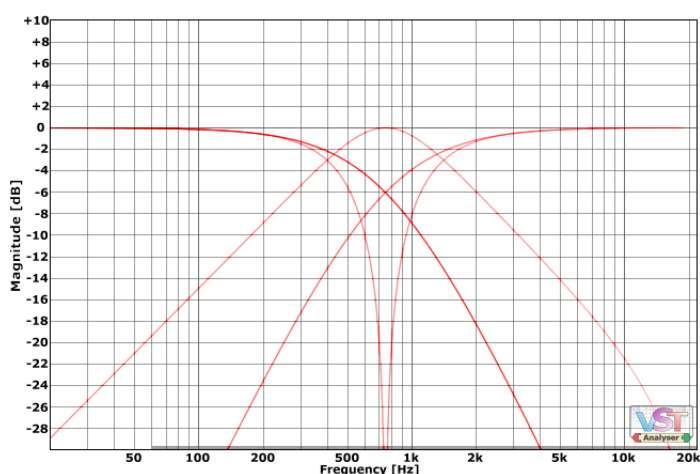
- The whole user interface is laid out so you should work from the top to the bottom.
- Best place to start is to decide if LowCut and HighCut filtering is needed and, if it is, setting it precisely. If you adjust them later they can change audio levels and produce different compression amounts down the signal chain. The same goes for Input gain level – once you set it you should leave it.
- The next step is to single out the most noticeable part of audio that needs to be improved, be it dynamics, frequency response, stereo width etc. Then you should set the most adequate filter type for that task at stage 1 (or turn the filter off).
- Choosing filter type should be relatively easy, but adjusting it's frequency and bandwidth is rather nuanced affair. Helpful technique is to switch between Band/Notch or Low/High filters while solo mode is activated (red blinking button next to Stage). That way you will easily hear and compare dry and wet parts of spectrum until you find just the right spot for the filter.
- Set compression before saturation. Saturation depends on compression and can introduce noticeable distortion if pushed too hard. Still, MCS6 saturation stage is rather subtle if used normally.
- Bias should be used intuitively, when you feel the need to add more “crunchiness” to the sound. It is more suited for use on lower frequencies, where ear is more forgiving on distortion. More often is better to dial at least some amount of bias then to leave at zero.
- Once you set everything at stage 1, proceed to stage 2 if needed and try not to backpedal a lot. Changes in stage 1 spill over to all following stages and can disrupt finely tuned balance.
- You don't have to use all stages, although you can be tempted to do so. Often just one or few stages will get the job done. Overprocessing signal usually reduces sound quality, so try to avoid it.
- Don't work too long with solo button turn on, because it is more important how wet audio blends with dry, than how it sounds isolated.
- Turn off PeakScope to reduce visual distraction if you exactly know what you're doing.
- Golden rule – always compare unprocessed and processed audio at the same perceived loudness.

All sliders are operated by dragging the mouse horizontally and double-clicking resets them to default positions. Numerical values can't be entered and everything is controlled only by mouse. Parameters can be automated, but it's advised not to, because of the interdependent nature of processing.

If you have any questions, suggestions or comments, feel free to send them at mail@jovaniljadica.com.



1. Frequency response of LowCut and HighCut filters. Red curves show resonance at minimum and maximum (0.0 and 2.0), pink curve is default (1.0).



2. Four types of filtering available – BandPass, Notch, LowPass and HighPass. Frequency and bandwidth set at default positions (750Hz and 4.0).