

DDMF The Strip manual



Thank you for demoing or even purchasing this fine DDMF product! To get the most out of it, please try to read this brief manual at least once...

The Strip is a channel strip plugin, meaning to virtually recreate a single channel of a mixing console. With EQ, gate and compression, it entails probably the three effects found most frequently on such consoles. Due to its low CPU usage, it can be applied on every channel of your project without compromising your hardware. You will soon find it to have become an important part of your everyday mixing workflow!

Installation: The Strip is available as a VST/VST3/AAX plugin for Windows and a VST/VST3/AU/AAX plugin for Mac OSX. The installer will automatically place all formats into their default folders, except for the VST format on Windows. For that format you will have to specify your VST plugin directory.

Usage: The Strip consists of an EQ section on the left hand side, a gate section on the upper right side, a compressor section on the right side in the middle and a general input/output control section on the lower right side. There is a collection of LEDs that become green when activated: these are actually buttons to enable the respective section, or, in the case of the EQ, the various subbands. Just click on them to toggle them on or off. The rotary sliders can be moved by a vertical mouse drag. Finetuning mode is entered by holding down the Ctrl button on your keyboard while dragging. Double-clicking a slider snaps it back to its default position. While dragging, the numerical values that are being changed by the slider are being displayed, and faded out again after releasing the slider.

EQ:

The EQ consists (from top to bottom) of a lowpass and a highpass filter with a frequency control rotary slider for each of them, four peaking filters with gain, bandwidth and frequency controls and a low shelf filter with gain and frequency controls. It also applies a very slight harmonic distortion when the audio signal is running through it. This is often perceived to add a certain amount of "analog warmth" even when all EQ gain sliders are set to 0 dB.

Gate:

The gate section consists of a threshold, a range, an attack and a release slider. With the gate you can separate possible low dB noise artefacts from the real signal. Whenever the signal going through the gate falls below the threshold, the signal is attenuated by a fixed amount of dB (which can be chosen via the range slider). A fade-out time given by the release value is used. When the signal crosses the threshold again from below, the attenuation is lifted over a short time span which is given by the attack value. The current amount of attenuation (0..50 dB) is displayed via the LED displays in the center of the gate section.

Compressor:

The compressor has the same set of controls as the gate, but works, roughly speaking, in the opposite direction: when the incoming signal exceeds the threshold, it is being attenuated by a certain amount of decibels, using a fade-in time given by the attack value. After the signal falls below the threshold again, the attenuation is smoothly increased to 0 dB over a time given by the release value. The amount of dB that is being taken off the input signal is given by the ratio: if the ratio is set to a value $x:1$, for every dB the signal is higher than the threshold value, only $1/x$ dB are being transmitted. For instance, setting the threshold to -10 dB and the ratio to 10:1, the whole range from -10 to 0 dB will be compressed to the -10:-9 dB range. Which, in turn, allows you to increase the output volume afterwards by 9 dB, yielding a much louder and powerful signal.

Finally, in the in- and output section we have an input gain slider which lets you change the gain of the incoming signal, and an output gain slider which does the same to the outgoing signal, after it has passed all sections of The Strip. Furthermore, the sign of the signal can be flipped (very useful to check for potential phase issues between different tracks), the order of EQ and compressor can be interchanged, and, last but not least, 2x oversampling can be applied. What this does is it effectively doubles the sample rate of your audio signal inside the plugin. Since there are some nonlinearities in all three sections, this can help to reduce so called aliasing effects, which occur when frequencies that are generated by nonlinear effects are higher than the maximum frequency supported by the sampling rate.

Demo restrictions:

The EQ section emits a short buzz every minute, and settings are not being saved (at least in most hosts...).

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