MQ57 v1.5
PARAMETRIC MASTERING EQUALIZER

INTRODUCTION

MQ57 is parametric equalizer designed for mastering. It is powerful, but quite easy to use after you understand what each knob and button does, so learning curve should be very mild for everyone who grasps basic principles of equalization. The main difference from other EQs is that MQ57 has all controls and functions optimized for mastering needs – it has relatively wide Q factors that focus on overall balance rather than narrow cuts and boosts, moderate gain ranges for fine-tuning and low and high-cut filters that tame spectrum extremes. There is no hidden processing going on beyond what you dial in – what you set is what you get.

- 7 fully parametric minimum-phase filters per channel: 5 of them are peak, 2 are shelf;
- 2 additional cut filters per channel with two selectable slopes;
- Separated or linked Mid/Side operation for each filter;
- Functions and ranges optimized for subtle mastering equalization tasks;
- Band-pass frequency monitoring for 5 peak filters;
- 32-bit internal precision with zero latency and reasonable CPU consumption;
- No introduced harmonic or dynamic distortion;

MASTER SECTION

Master section is located in upper-left part of GUI. Set the input gain so that subjective wet signal level matches dry for honest comparation between them. Next to it are buttons labeled Mid, Solo and Inv. Mid/Solo button chooses which part of stereo signal you're adjusting with all the knobs on the right. Solo button isolates only the active channel (in solo mode side channel is temporarily boosted 6dB to more closely match usual Mid signal levels). Inv button flips the output phase of R channel. LED clip indicator lights orange at -3 dBFS and red at 0 dBFS. Monitor button activates band-pass frequency monitoring for peak filters. When it's active you can preview isolated frequency of filter while you adjust orange-colored knobs. Shelf filters (LSF and HSF) can't be monitored.

LOW-CUT AND HIGH-CUT FILTERS

These are butterworth cut/pass filters and they have two selectable slopes: 12 and 18 dB/octave for low-cut, 18 and 24 dB/o for high-cut. Use them to remove low-end rumble or DC signals and to tame top-end harshness. LCF has different ranges for Mid and Side channels, because often you'll want to remove more bass from side channel to achieve more cohesive stereo in that part of spectrum. On/off and link switches work the same as for other filters.

PEAK AND SHELF FILTERS

The larger part of interface consists of seven modules (labeled LSF, LF, LMF, etc.). The first and the last are low-shelf and high-shelf, while inner five are peak filters. They all have ±6dB operating range. Frequency controls are divided into seven overlapping zones covering every part of audio spectrum. LSF and LF as well as HF and HSF share those zones, but since they are different types you'll use them in different manner. Above them are displays with numerical readouts showing current frequency in Hertzs or gain amount while you change it. Q factors are controlled with lower row of blue-colored knobs. Shelf and peak filters have Q factors that do not act the same, although they share the same principle of higher Q meaning steeper curve. On the bottom there are two buttons for each filter. First is standard on/off switch. Second button links all controls on that filter for both channels (Mid+Side), including the on/off switch.

GENERAL TIPS

- To reset knobs to their default position double-click on them. For precise knob movement use CTRL.
- For the best results use higher sample rates (88.2kHz, 96kHz, etc.) on any high frequency equalization, especially when using HCF filter.
- Spend enough time analyzing mixdown you're working with, write down or just remember where do you want to take it and how to do it. Don't automatically start with low+high boost or any predefined measures. That's why this plugin doesn't have any presets – there is no "one size fits all" curve.
- Bypass should be the most used button here: always check whether you're improving, just changing or degrading dry signal. It goes without saying that it should be done with matched dry and wet levels.
- It's very helpful to have reference track(s) that is already mastered and that resembles what you're trying to achieve. For less experienced users spectrum analyzer can be helpful. Depending on the genre, try to get reasonably leveled frequency response (without drastic peaks and dips) using 3 – 6 dB/o spectrum slope, between pink and red noise.
- If something really needs more than ±6dB of equalization you should probably fix it in mixing stage. You positively can't fix a mixing error during mastering without disrupting something else.
- Use monitoring option occasionally, do not rely solely on it for finding every filter's central frequency. Something that sounds harsh isolated can become desired in the context and vice versa.
- To avoid any potential peak clipping reduce input signal into the safe zone. Leave headroom for equalization and you can later push it to the limit with compressor or limiter.
- Do not overwork your ears doing single track. If you have many of them (for an album) do them almost parallelly. One of the main goals of mastering is to create consistent sound on the whole album and it's easy to loose perspective if you keep perfecting just one track at a time.