LP10v4 Manual

Installation:

Simply run the installer that you should have received alongside with this manual. For Windows, specify your VST plugin directory during installation. After the install, LP10 should show up as an avaiable plugin in your digital audio workstation/host.

Usage: The LP10 is a software equalizer with up to 10 bands and 10 filter types per band to choose from. There are two ways to control the settings of the bands:

a) By direct interaction with the graphical display.

b) By using the controls in the lower left corner of the user interface

Here is how the mouse-display-interaction is organized:

- Clicking on the display adds a new band, represented by a circle, with gain and frequency settings corresponding to the coordinates where the click has occurred. Depending on where the click has occurred, a low- or high pass, a low- or high shelf or a peaking filter are preselected. You can change the filter type by right-clicking on the new circle, which shows all available options.

- Double-clicking on a circle removes it, along with the corresponding filter band.

- Clicking with the mouse cursor on one of the circles will highlight it; also the settings of the band associated with that button will be shown by the band controls. You can move the dot around with your mouse, thereby changing the frequency and the gain settings of this circle's band. The overall frequency response is shown with a thick white line; if there is more than one active band, the individual contribution of the band that is active at the moment is shown as a thin white line.

- You can drag new circles directly from the 0-dB line, offering the fastest way to add new bands.

Note that there are filter types where gain control is disabled (low/high/band/allpass and notch). In this case the circle is restricted to the x axis.

- You can change the ‘Q’ (which is inverse to the width) of a filter by using the mouse wheel.
- You can change the ‘P’ (the phase) of a filter by using the mouse wheel with the ‘Alt’ key pressed.
- Right-clicking onto an already existing circle lets you choose the filter type again from a drop-down menu.
- Left-clicking on a circle while simultaneously pressing the Alt/Option key bypasses the band in question. The circle is shown with an ‘x’ to mark its status. You can reactivate the band again by using the same click-key-combination.
- Ctrl-clicking on a circle activates the autolisten mode: a band pass filter around that circle will be engaged, all other filtering will be bypassed. That way you can quickly hear what's going on the the vicinity of the circle in question. The Q value of the autolisten bandpass filter corresponds to the Q value for the selected filter.

So far for the mouse controls. All bands can also be controlled by the ‘ordinary’ controls in the lower left corner. Choose a band, set its filter type and dial in the appropriate gain, frequency and Q values by moving your mouse vertically over the controls with the left mouse button pressed.

Adjustable phase response: the frequency responses of all filter types of the LP10 are exactly matched onto their analog counterparts (RLC circuits). There is a lot of debate, however, about whether the non-linear phase response of these circuits (“minimum phase”) should also be exactly matched by digital equalizers or whether a linear phase algorithm leads to the best results. Linear phase is often associated with a “cleaner” sound, but it can lead to difficulties at low frequencies (“preringing”). For this reason, the LP10 offers the concept of adjustable phase you can smoothly change the phase response from linear to minimum phase and even to its opposite, inverse minimum phase! A value of -1.0 corresponds to inverse minimum phase, 0.0 means linear phase and +1.0 minimum phase. This gives you full control over the phase behavior of the LP10.
The phase is controlled either via the ‘P’ button for the selected band or by placing the cursor over an active circle in the display and using the mouse wheel with the ‘Alt’ key pressed.

Filter types: The LP10 offers 10 filter types for each of its 10 bands. All filters are calculated in frequency space and thus reproduce the exact frequency response of analog RLC curves. The phase response of each filter can be freely tuned.

The types are:

I. Peak: a peak filter. Use it to amplify or attenuate a specific frequency region which can be specified by choosing “Center” and “Width” accordingly.

II. FlexPeak: a Butterworth-based peaking filter with continuously variable slope from 6 dB/Oct to 60 dB/Oct.

III. FlexNotch: a notch filter to cut steep holes. Disturbing signals which are limited to a narrow frequency band can be completely removed with a notch filter. Note that the “Gain” button is deactivated for notch filters. Again, the slope at the band edges is continuously variable.

IV. LPF12: a low pass filter with a slope of 12 dB/octave (“Gain” deactivated).

V. FlexLP: a Butterworth-based lowpass filter with continuously variable slope.

VI. HPF12: a high pass filter with a slope of 12 dB/octave (“Gain” deactivated).

VII. FlexHP: a Butterworth-based highpass filter with continuously variable slope.

VIII. FlexBP: a Butterworth-based bandpass filter with continuously variable slope.

IX. FlexLS: a Butterworth-based low shelf filter with continuously variable slope.

X. FlexHS: a Butterworth-based high shelf filter with continuously variable slope.

So basically, all filters with 'Flex' in their name have continuously variable slopes which you can tune with the dedicated slider in the lower slider row (third from right).

Further controls:

While you can freely adjust the phase for each of the filters you have created, there is also the possibility to simultaneously set all filters to linear phase (‘Set to LinPh’) or to minimum phase (Set to MinPh’).

Being a high-precision EQ especially designed for mastering, you might find the integrated stereo width control helpful. The stereo width is controlled with the ‘SW’ button and ranges from 0 (collapsed to mono) over 1 (no change to the stereo picture) up to 2 (most extreme stereo widening). This button has no effect on a mono channel.

Right next to the stereo width control you find a button with which the dB resolution can be set. This can be helpful if you want to make very fine adjustments.

On the right of the frequency curve display you find four further controls, which are, from top to bottom:

I: the overall out gain. The overall gain can be set to a value between -80 dB and +24 dB.

II: a global bypass button.

III: the “A/B” button: the LP10 offers two independent configurations. Simply switch and compare between two alternative settings by switching between “A” and “B”.

IV: the “Copy” button: copies the current program (A or B) to the program not in use (B or A). This way you can test the effect of small parameter changes, starting from identical setups.

Tilt functionality:

The whole spectrum can be tilted up- or downwards with respect to a variable center frequency. This is very helpful if you want to adjust the overall balance of low- and high frequency content in your mix. Here's how to do this: hold the “alt” button while left-clicking somewhere in an empty area of the frequency display window, and move the mouse up- or downwards to change the amount of tilt (max. range is -2 db → 2 dB/octave). Move the cursor left or right to change the tilt center frequency. To switch off the frequency tilt again, hold the alt-key and double-click somewhere in the frequency window.

Spectrum analyzer:

The built-inspectrum analyzer allows you to view the frequency content of your audio material in real time. The analyzer is activated/deactivated by clicking on the “Analyzer” button. The spectrum of the signal at the input of the equalizer is shown in dark green, the spectrum at the output in light green. This way you can immediately judge the changes you have made.

Zooming/shifting the analyzer curve: when holding the “shift” key pressed on your keyboard while dragging the mouse vertically with the left mouse button pressed will shift the analyzer curve up and down. When you keep the “Ctrl” button pressed while dragging the mouse, you can zoom in and out of the analyzer curve (increase/decrease the dB resolution).

True Stereo operation:

LP10 offers the possibility to equalizer either the left and the right or the mid and the side channel of the stereo signal separately. When clicking on the “2-channel” button to the right of the main display, a second LP10 instance will show up. By default the upper user interface acts on the left and the lower interface on the right channel, as indicated by “L” and “R” on the left side of the GUIs. You can now make independent settings on each channel or lock both channels with the “Locked” button that appears in two channel mode. You can also change the mode from “LR” to “MS”, so that the upper GUI will refer to the mid signal (L+R) and the lower to the side signal (L-R).

Latency: the LP10 operates with a latency of 12288 samples. This means that the incoming signal is delayed by this amount. Many hosts have automatic delay compensation built in; in other hosts you have to manually adjust the timing of your tracks to account for the latency. Note that the LP10 is intended to be a mixdown/mastering EQ which should be used on the sum bus, where latency is no problem anyway. Pressing the overall bypass button bypasses the complete signal chain, so that the latency is also zero in bypass mode. You will therefore notice a time shift in the signal.

Audio buffer settings: while not creating an unusually high average CPU load, the LP10 performs computationally expensive calculations a few times per second. If the audio buffer in your host program is set to a too small value, this can lead to stuttering/glitches. In that case, please readjust the size of the audio buffer.

Demo restrictions: Noise bursts will be added every now and then. If you haven’t done so yet, you should get the full version at http://www.ddmf.eu !

Questions/Feedback: support@ddmf.eu