## DDMF LinComp (Linear Phase Multiband Compressor) Manual Version 1.0.5

Thank you for using this fine DDMF audio plugin! To get the most out of it, we recommend to read this manual at least once before operation.

## Setup:

LinComp comes with an installer, in which you can select which plugin format(s) you want to install (VST, VST3, AU (Mac only) and AAX are the available options). On Windows, if you select the VST option, you will need to specify the location of your VST folder(s) (two of them, if you opt to install both the 32 and the 64 bit version). On Mac, the plugins automatically contain both the 32 and the 64 bit version). On Mac, the plugins automatically contain both the 32 and the 64 bit version, and the folders where they are installed are also fixed to their system-wide values, so you don't have to worry about anything here. After installation, DirectionalEQ should be available in your DAW's list of plugins. Occasionally, a rescan of your plugin folder might be necessary.

## Usage:

LinComp splits the incoming audio signal into four separate frequency bands, using linear phase crossover filters. Each of the bands is then compressed using individual settings, and afterwards the four compressed signals are summed back together again. Because of the linear phase filters, no artefacts at all are created in this process, i.e. if the compressors were set to bypass, the signal would be 100 % identical at input and output of LinComp. Multiband compression is superior to ordinary compression because the attack and release times for each band can be adjusted separately. It's easy to see why this is important: let's say you want to compress the range below 100 Hz. One oscillation period of all the audio waves in that region is then at least 10 ms long, so that attack and release times should be longer than 10 ms. If you want to compress the high-frequency region above 5000 Hz, on the other hand, one oscillation would be 0.2 ms or shorter, so that the attack and release times can be set to much shorter values.

LinComp induces a latency of 1536 samples due to the linear phase crossover filters, and while we have done our best to optimize the code as much as possible, due to the very high sonic demands the CPU load is definitely not ultra-low. Since LinComp should primarily be used in your mastering chain, this is usually not an issue.

The user interface of LinComp is shown below. All sliders are numerical sliders, which means if you place your mouse above on of them and start dragging vertically, the values will change accordingly. Also, clicking in one of the sliders, numerical values can be entered directly if you need very precise control.

In the top row, the three frequencies that separate the four bands can be adjusted. Below the graphical display, four columns that represent the controls of the four compressors are visible. The threshold determines at which level the compressor kicks in, the ratio (from 1:1 to 10:1) adjusts the strength of the compression, the attack time is the time the compressor waits after the threshold level has been reached to actually start compressing, and the release time, conversely, is the time the compressor still keeps on compressing after the signal has become smaller than the threshold again. With the gain control you can increase or decrease the level of each band after compression has taken place. The "level" display shows you what the maximum levels for each band is; this can be reset by clicking on the number. In practice, the threshold levels should roughly follow the leves that are displayed here.

All control values for the four compressors can be adjusted simultaneously using the controls with the up/down arrows in the M (=master)-column. On the very right hand side, we first have a master output control at the top, showing also the levels of the two stereo channels (or the mono channel, then twice). Below that, there are five buttons: "Bypass" bypasses the whole plugin (leaving the overall latency intact though). "Soft" sets the knee curve of the compressors to a softer shape. With "Clip" all samples above 0 dB are clipped; with this setting on, you can basically use LinComp as the last plugin in your mastering chain and be sure you'll never end up with signals above 0 dB. The "State" and "Copy" button control the two states the plugin has, A and B. Switch between the two states to compare two different settings, and copy the settings of the current state (be it A or B) to the other state, to start from equal values.

| Crossover 10     | 9 Hz 628         | Hz 2           | 275 Hz DD                 | MF LinComp                   |
|------------------|------------------|----------------|---------------------------|------------------------------|
|                  |                  |                | 1.1.1.1.1.1.1.1           | -0 dB 0.3 dB                 |
|                  |                  |                |                           | 10                           |
|                  |                  |                |                           | 20                           |
|                  |                  |                |                           | 30                           |
|                  |                  |                |                           | 40 -3.6 -3.5                 |
| Thrsh -10.9 dB S | Thrsh -26.6 dB S | Thrsh -20.0 dB | S Thrsh -20.0 dB          | S Bypass                     |
| Gain 0.0 dB      | Gain 0.0 dB      | Gain 0.0 dB    | Gain 0.0 dB               | Soft                         |
| Ratio 1.8:1 Byp  | Ratio 1.8:1 Byp  | Ratio 1.8:1 By | <sup>/p</sup> Ratio 1.8:1 | Byp Clip                     |
| Atk 2.0 ms       | Atk 2.0 ms       | Atk 2.0 ms     | Atk 2.0 ms                | State: A                     |
| Rel 63.9 ms      | Rel 63.9 ms      | Rel 63.9 ms    | Rel 63.9 ms               | Сору                         |
| Level -5.2 dB    | Level -3.5 dB    | Level -8.8 dB  | Level -9.8 dB             | M LINEAR PHASE<br>COMPRESSOR |

Finally, the graphic display: we have three vertical yellow lines which represent the crossover frequency values that separate the frequency bands. Hover with your mouse over them and you will see that you can move the horizontally, which is just another way of setting these frequencies. Then, there is a grey and a red block for each frequency band. The red band displays the signal strength of each band after compression, on a scale of -40 dB to 0 dB. The grey block indicates the threshold and the ratio of the compression. The way it works is as follows: the lower border of a grey block is the threshold for that band, and the upper border is where a hypothetical 0-dB-signal would end up after compression. So, with a threshold of -30 dB and a ratio of 3:1, there would be a grey block ranging from -30 dB to -20 dB (because the 30 dB dynamic range before compression become only 10 dB after compression). You can move the grey blocks up and down, another way to set the threshold (using the visual feedback from the red blocks).

Alright, that's about it. As you can imagine, the possibilities are endless, so just go ahead and play! It's pretty hard to come up with something that sounds really bad, so don't be afraid to make mistakes. A good way to start is to let your song run one time, with roughly evenly distributed frequency crossovers, and to record the maximum levels in each band. Then, copy these values into the threshold controls and drag down the master threshold fader until it starts to sound too compressed. Now you've got the thresholds set right and can take things from there...

Demo restrictions: audio dropouts about once per minute, and no saving of settings in most hosts.

Feedback/requests: support@ddmf.eu