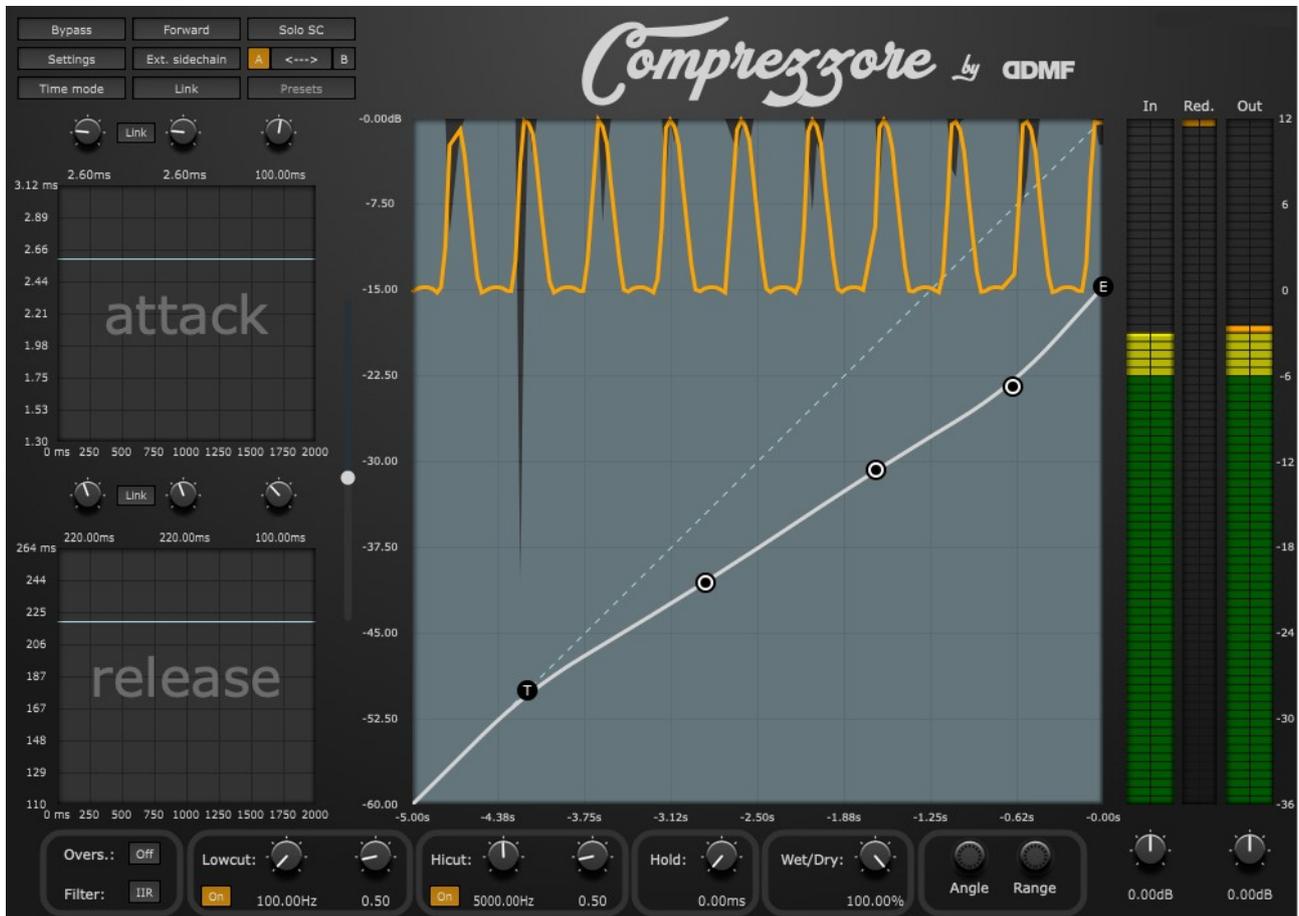


Manual for DDMF's Comprezzore Audio Plugin



Thank you for using Comprezzore! Since this is not your standard run-of-the-mill compressor plugin, it is highly recommended to read through this manual at least once. It will help you to get the most out of the numerous possibilities you have with Comprezzore to fine-tune the dynamics of your audio material.

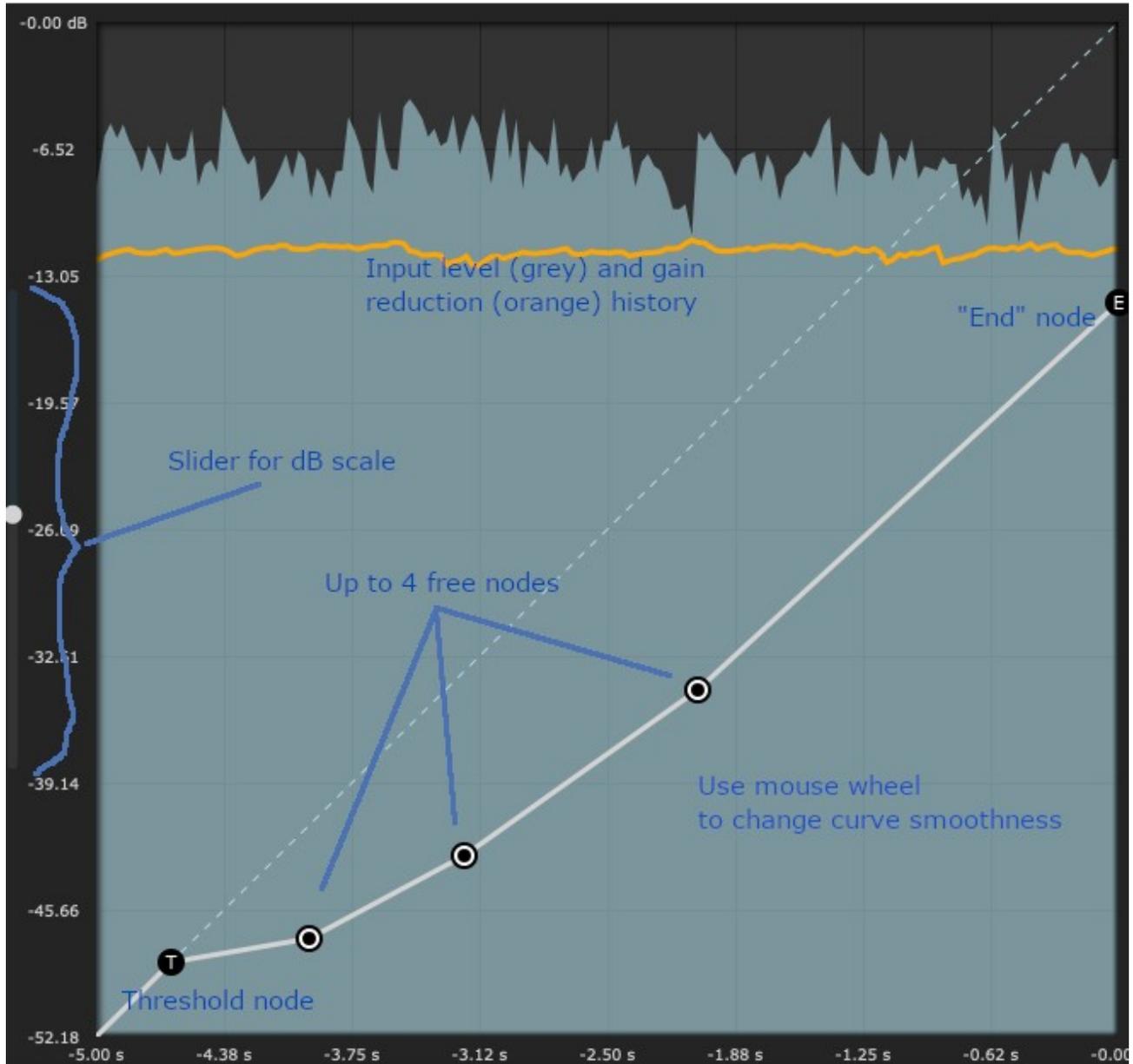
Comprezzore is an audio plugin that needs a host program into which it can be loaded. The plugin is compatible with all VST, VST3, Audio Unit (Mac only) and AAX hosts.

Installation: for Mac, simply choose which formats you want to install, and the plugin will be copied to the various default folders for audio plugins. Under Windows, again, select the plugin formats you want to install. The only format for which you can specify your own folder is the VST format; for the rest, everything happens automatically. After installation (and a possible re-scan of your plugins by your host program), you should find Comprezzore available in

the list of plugins.

Usage: Comprezzore is a dynamics processor: it will, quite generally, map certain input levels to certain output levels. It can both compress and expand a signal, i.e. map a wider range of levels to a smaller range or vice versa.

At the heart of mapping input to output levels is the central screen shown here:



Here, you can change your desired dynamic response by simply using your mouse to drag around various nodes. By default, a threshold and an endpoint node are present. You can move the threshold node diagonally, and the endpoint node vertically. In a traditional compressor, those would be the two variables that control threshold and ratio. In Comprezzore, you can add up to four more nodes in between the threshold and the end point node. You do this with a single mouse click in the desired node position. You delete one of those previously added nodes by double-clicking on them. Those additional nodes can be moved around rather freely, the only requirement being that their

horizontal ordering cannot be changed.

To guide your decisions on where to best place your nodes, the input level of your audio material is shown in a moving image with a history of 5 seconds (the light grey area). On top of that, the five-second history of the applied level reduction is also shown. From this you can directly see the influence of any parameter change, not only of the node positions but also of the attack and release timings to which we come in a second.

It is advised to position your nodes "where the action is happening": you will have the biggest impact on your levels when you adjust your curve such that there will be a noticeable change of output levels over the range of input levels that you see. If you create a curve such that you will have, say, -20 dB reduction from an input level of -50 dB all the way up to 0 dB, there won't be much dynamical change except for the whole signal being lowered by 20 dB. Play around with the node positions to get a feeling for where to best place them for a variety of source materials.

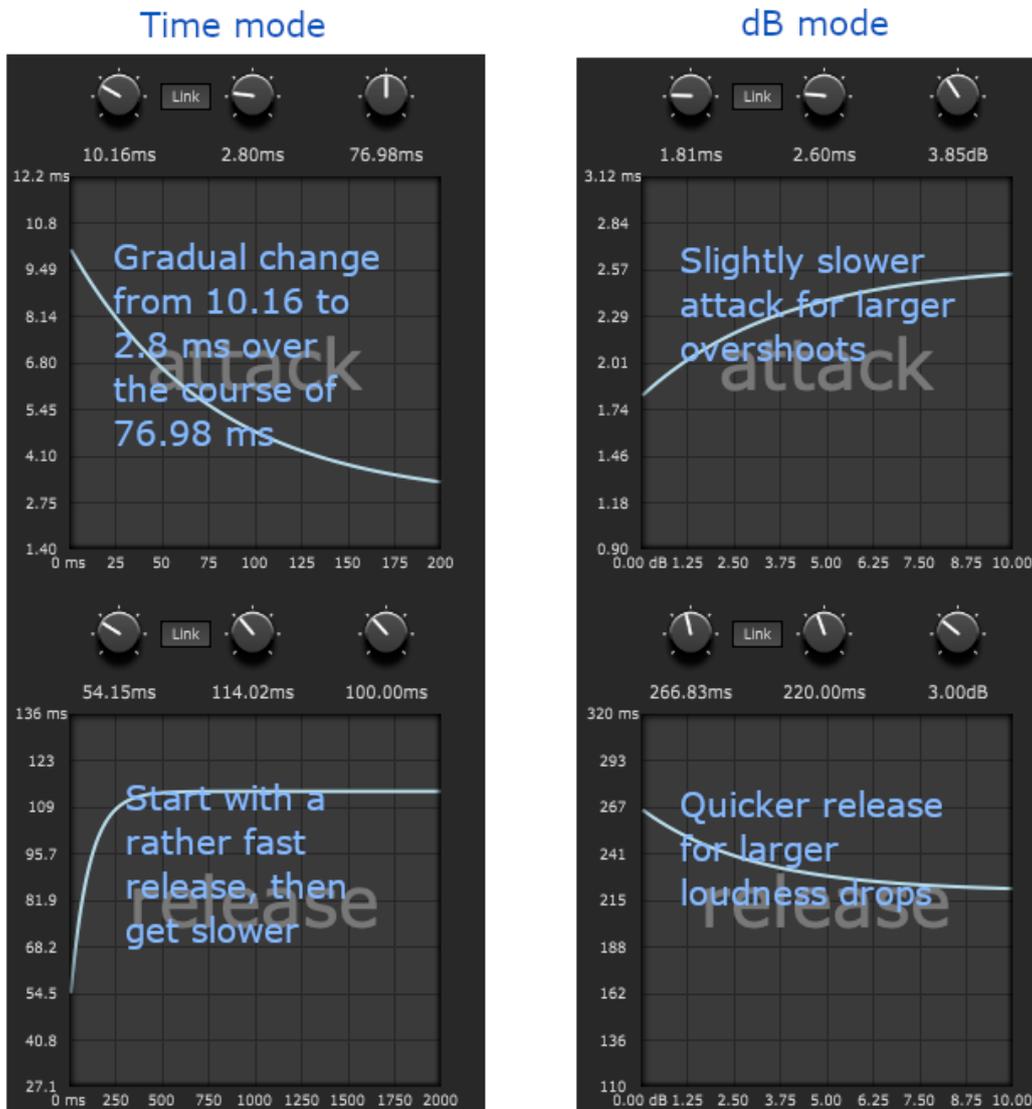
You can change the scale of the dB axis using the zoom slider on the left of the display. Finally, you can change the smoothness of the resulting curve using your mouse wheel, from a very smooth interpolation between the nodes to a curve with sharp edges. Again, there is no general rule on which smoothness level sounds the best, it always has to be decided per case.

Attack and release section: this is where Comprizzore is really offering something new, which is why I encourage you to familiarize yourself with the concept. The basic idea is that, instead of constant attack and release rates, you can choose how the time constants should change, either as a function of time or as a function of dB-over or -undershoot.

For this end, the left side of the UI contains two smaller graphs, with three knobs on top of each graph. The upper section controls the attack rate, the lower section the release rate. To recapitulate what attack and release in a compressor are doing: whenever the currently applied gain reduction is lower than what you'd expect based on the compression curve and the current incoming level, the gain reduction is increased over a time span that is dictated by the attack time. Conversely, when the applied gain reduction is too high (after a louder section where more compression took place), the value of gain reduction is reduced, and the timing of that reduction is dictated by the release time constant (where, typically, attack times are shorter than release times).

Now in Comprizzore, this concept is extended: attack and release times can be adjusted in greater detail. There are two principal modes: "time mode" and "dB mode". When in time mode, the leftmost knob above either the attack or the release section controls the start value of the time constant (which is the value that is applied right after an attack or release phase has begun); the second knob is the "target" time constant, which is where the time constant will eventually level out, and the third knob controls the time scale with which the transition between the initial and the final value is happening. Sounds confusing? Let's look at an example: setting your first attack time constant to 1 ms, the second one to 20 ms and the third one to 100 ms, this means that your attack time will increase from 1 ms to 20 ms over the course of (roughly)

100 ms. The display below the attack sliders will show a curve depicting the exact evolution of your attack time. The procedure is, of course, identical for the release section. If you press the "Link" button between the first and the second knob, both values will be identical, which corresponds to a traditional compressor with constant attack and/or release times. The third knob has no function in that case.



Buttons in the upper left corner:

Now that we have the difficult stuff out of the way, let's go through the rest of the UI elements. We'll start with the set of buttons in the upper left corner.

- Bypass: this bypasses the whole plugin
- Forward/Backward: this changes the compression topology. Forward compression measures the audio level at the beginning of the chain, backward compression at the end. So forward compression is always a

bit quicker to react, since it immediately sees a transient and can apply the necessary gain reduction, whereas backward compression (which is usually what has been used in all those legendary hardware compressors) is always one step behind, so to say (but is, at least by some, said to sound slightly more musical).

- Solo SC: the "sidechain" signal is what actually drives the level detection circuit. If no external sidechain source has been connected, and no sidechain filters have been applied (we'll get to those when introducing the UI elements in the lower part of the UI), this would simply be a copy of the original incoming signal. With filters applied and/or with an external sidechain source, you'd hear that audio signal. Very helpful to judge what is actually going making the compressor behave like it does.
- Settings: this will open a separate rectangle where you can select your own colour palette for a number of Comprizzatore's UI elements. Plus: you can freely customize your VU meters! Select the min and max value of the meters, as well as the dB values where you want the change between the four available colours to happen (traditionally those are green, yellow, orange and red, but with the colour selectors you can choose anything you like). Perfect if you want to work with a custom metering scale.
- Ext. sidechain: Comprizzatore has 4 input channels, where the first two will always carry your signal to be compressed, and channel 3 and 4 can be filled with an external signal which would then duck the signal on the first two channels. This button switches between using the signal at incoming channels 1+2 and between a possibly connected source signal to the incoming channels 3+4 for level reduction. If no source is connected to channels 3+4, Comprizzatore will not do anything to your audio material when that button is enabled.
- A ↔ B: the plugin has two internal states, called A and B. A is the default state. You can switch between A and B by clicking on either letter (the currently active one will be shown in orange) and you can copy the active state to the inactive state by clicking on the double arrow between the letters. Very handy to compare two sets of compression parameters.
- Time mode/dB mode: this button switches between the two different modes for the attack and release time curves, as discussed in the previous section.
- Presets: to give you a small introduction to the possibilities of Comprizzatore, a set of presets has been added to the plugin. Simply use this drop-down menu to change between them

Controls in the lower row (left to right):

- Overs.: lets you choose between no, 2x and 4x oversampling. Comprizzatore uses a very clean algorithm, but if you want to decrease the level of distortion even more, you can apply oversampling here (at the expense of a somewhat higher CPU consumption).

- Link: when you have a stereo signal you can, in principle, either opt to compress each side separately using a mono algorithm, or you can decide to calculate an overall gain reduction factor and apply that same factor to both channels (hence keeping the stereo image intact which might get destroyed with the former approach when you have a lot of loudness difference between left and right). Comprezzore lets you smoothly vary between the unlinked approach or the fully linked approach to find the best compromise between keeping the stereo image and not applying too much compression on a channel which might not need it.
- Filter (IIR/FIR): the filter type that is applied when using oversampling to downsample your signal back to the original sample rate. IIR uses an Infinite Impulse Response filter, which introduces almost no latency, but a considerable phase shift that may or may not be a problem in your situation. If you decide it is a problem, switch to the FIR (Finite Impulse Response) filter which will introduce a bit more latency, but without phase distortion.
- Lowcut section: this is an optional filter in the sidechain path, applied before level detection. It's a 12dB/octave lowcut filter, you can select its operating frequency and width here. Everything that you filter out here will not lead to a reaction of the compressor. So if you set this to 100 or 200 Hz, you will notice more of the low bass of your audio coming through uncompressed. Can be helpful for that nice "oomph".
- Hicut section: same as before, but a hicut filter. If you have a lot of signal in the barely audible frequency range of 10 kHz and higher, here is the point where you can filter that part out before letting it influence the compression engine.
- Hold: normally, when there is a change from attack to release mode, that switch would happen instantaneously. Here you can tell the plugin to wait some time before releasing the compression (from 0 to 500 ms). You should immediately see the effect in the orange gain reduction signal. Think of it as an infinite release time for the number of microseconds set here.
- Wet/dry: sometimes you want the best of both worlds: a mix of your compressed and your uncompressed signal. Can be great to achieve a certain fullness while keeping your transients under control.
- Ratio/Threshold: two knobs that correspond to traditional ratio and threshold parameters. The Ratio knob will move your whole compression curve vertically, while the Threshold knob will move the threshold node (and any other nodes that may or may not be present) closer to the endpoint node.
- In/Out: probably self-explaining, isn't it?

That's all! Thank you for reading this far, I hope you can now make the most use of Comprezzore. For any questions or remarks, please contact support@ddmf.eu. Thanks again for using this fine piece of audio software!