JAX DYNAMICS SERIES

BASICS

Manual (Version 1)

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BASICS

The JAX Dynamics Series - Basics include several classic dynamic (single band) processors for altering gain or loudness in time dependency with realtime streaming audio.

Currently there are the following audio processors in this collection:

- JAX DYNAMICS : Compressor
- JAX DYNAMICS : Limiter
- JAX DYNAMICS : Gate
- JAX DYNAMICS : Compander

All these dynamics processors operate in single band (all-pass) mode with optionally an input filter for the peak detectors (version 2 extension).

The JAX DYNAMICS Compressor

This processor is a classic single band (all-pass) compressor unit with its commonly used parameter set. The compressor usually looks for peaks in the audio stream and applies gain reduction based on a **threshold** parameter, an **attack** and a **release** time parameter for the integrated envelope follower. Additionally there is a **ratio** parameter, which adjusts the intensity of the gain reduction for the processing.

Peaks over the threshold value will be squashed down with the speed of the adjusted envelope follower. A time dependent gain reduction of the entire audio signal is the result. The adjustments of the attack and release times are important for achieving the desired results. Transients can be tamed or emphasized with the correct parameter adjustment of the internal follower. This very much is dependent of the musical tempo and rhythmic content of the audio stream.

With an additional gain parameter, the input signal can be adjusted in loudness prior entering the envelope detector and processing stage. This way also a loudness compensation can be adjusted. This is often called “makeup”, although we did not involve any facial MakeUp artists with this. ^^

Our compressor unit also has some special controls, the “audition” switch, which basically inverts the processing result for audition of what the processor is doing (by just subtracting the output from the input). The reduced (removed) signal can be made audible this way temporarily for better understanding and control or even for creative usage.

The “mix” parameter is effectively a **parallel compression** option, which allows tho adjust a mix between the original and the processed audio signal to any amount. This allows more complex static and dynamic sound adjustments as it may be manually faded in and out at realtime too.

A compressor conceptually will reduce the loudness / gain of a signal. With the **gain** parameter (which is effectively an input gain), this reduction can be effectively compensated. There is no automatic gain adjustment applied inside the signal chain by intention. So the final gain of the audio signal is dependent of proper adjustment of all parameters.

The reduction signal and the resulting final gain can be visually watched with the compressors display. Two emulated ‚needles‘ will display the compression actions around the adjusted threshold value and 2 output indicators will show the resulting output volume of the output, taking the current mix parameter into account.

The indicators will reflect the action of the selected operation modi. There are the modi stereo **dual**, stereo **linked** and mid/side **dual**, and **linked** implemented (with version 2 of the unit). In stereo, the left channel is displayed with the color red, with the mid side modi, the mid channel is displayed in red. The other in green. You may discover, that the needle indicators will react very differently with different selected processing modi and also the compression result may differ clearly. Which mode to select is a matter of the kind of input.

Optionally a final brickwall limiter (version 2) can be switched on, to prevent the audio stream exceeding the 0 db mark and allowing over-compression without the danger of audible distortion.

Our compressor is working quite transparently, meaning there are no saturation or tube flavors applied intentionally. However, very short attack and release times can introduce harmonic signal distortion on a fractional level, as it is possible to adjust very short envelope times with our compressor.

Later versions may add an optional tube saturation model to the signal flow, which allows to emulate some taste to the resulting sound, which is kind of ‚modern' recently.

Note: Envelope followers (used in dynamic processors) are driven by the audio frequency parts with the highest energy. Often this are the lowest frequencies in the audio material, which naturally have the highest energy, depending on the material. Sometimes this may be a problem, as the compression result will always be very much dependent of the highest energy frequency components.

Therefore we implemented a lowpass/highpass frequency range filter (version 2) for the input signal, which effectively alters the envelope detection for the followup compression. Please note, that this feature is currently not visible at the interface but exposed as audio unit parameters. Only the detection stage is affected by the input filter range, the compression itself will be applied to the complete input frequency spectrum of the processed audio signal.

For multi-band compression (where the signal is divided into several frequency bands and processed with multiple compressors independently), we recommend our specialized multi-band dynamics or master bus processors, i.e. the JAX MBC Pro.

The JAX DYNAMICS
Limiter

The limiter is a classic single band (all-pass) dynamics processor with its commonly used parameter set. A limiter is a specialized version of a compressor and used to limit the output to a certain threshold value. Usually the limiter does not implement the ratio parameter (always has a high ratio) and conceptually reduces the gain by suppressing loud signal levels. The additional knee parameter particularly replaces the ‚ratio‘ parameter of a compressor (somehow).

The gain compensation is effectively applied by the **gain** or “makeup” parameter, so usually the entire limiter will be driven by just adjusting the input gain and the threshold value is kept constantly to a certain critical value, the threshold (i.e. the 0db mark).

Our Limiter internally works slightly different than the compressor or the expander or gate and so the results may be slightly different too.

The JAX DYNAMICS
Gate

The JAX DYNAMICS Gate is a classic single band (all-pass) dynamics processor which gates (drastically lowers) the audio signal under a selected threshold value, dependent of adjustable attack and release times. The Gate is so a specialized (extreme) version of an expander and definitively the inverse version of a compressor/limiter.

While the compressor and limiter squashes down the signal peaks over a certain threshold level, the gate does the opposite and passes thru the the peaks over a threshold value. So it will effectively suppress (eliminate) the audio signals below the threshold value.

Gates have much potential for creative usage. Even with rhythmic audio material, gates can be used to shape transients. The mix parameter is very important for doing this.

Especially the **mix** parameter enables to simulate an expander with the correct adjustment.

The JAX DYNAMICS
Compander

With our compander, a fusion of a compressor and an expander, we went a somewhat different approach to conventional dynamics processing, although we tried hide its internal complexity and tried to publish its parameters in a more traditional way. So it also fits better into the series.

Like all dynamics processors, there is a stage of detecting the peaks of the realtime audio stream. A true stereo envelope follower will track the signal peaks with certain attack and release time and then apply bidirectional gain modification based on these results. The modi are selectable like on all our dynamics processors.

The JAX Compander can either boost or damp the volume via its bi-directional parameter **exp./comp** in time dependence of the follower signal. On left side (negative values), it will expand the audio, on the right side (positive values) it will compress. Both directions are mutually exclusive and directly related to the resulting gain.

This works internally entirely different than the traditional threshold/ratio control of similar processors. Also the classical **threshold** parameter is different here. We use a kind of “virtual threshold”. Meaning, the processor algorithm internally works with normalized audio values.

If you tweak the threshold parameter, you effectively raise or lower the gain of the audio prior the internal processing stage with the 0db limit, which is inverted again before the output will come out and be audible.

The result is our compander, which is a very flexible dynamics tool and for instance able to bring back some dynamics to an over compressed sound, a requirement!!! that you may find very often these days.

But it also is able to raise the loudness of the audio input to an extreme and deliver transparent classic compression with quite low distortion.

Our compander is very transparent, meaning it will not introduce additional harmonic saturation, like many tools intentionally do. It is also not modeled after any hardware but an invention for the digital domain.