JAX DYNAMICS SERIES

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The JAX Dynamic series is a set of really easy to use visual audio units for altering the levels of sound in time dependency. The signal flow and important action of the internal DSP is displayed in realtime, unleashing the secrets of operation, even easy to understand for a novice.

Preface

Dynamics processing is an often misunderstood and mismatched area in music production. It is confusing at the beginning, but it is so essential for music production.

Compressors, Expanders, Gates, Followers, Limiters, Maximizers... this are all time domain level processing devices and most of them are of a hybrid architecture, meaning doing many different things at ones. This is the reason, why there are more dynamics processors out there than any other effects.

We want to bring some light into these different level modification methods and isolate them a bit, making things more clear.

Compressor

A compressor will squash the peaks of a sound. This is its only desire. A compressor will make a sound more compact. If the peaks (loudest signals) are above a certain threshold, their level will be reduced, all other sound (quieter parts) should remain nearly untouched as far as even possible. So a compressor always needs a peak detector (follower) to continuously examine (analyze) the incoming sound levels.

Now, squashing the peaks will logically always result in a level reduced sound finally. So the first (noticeable) side effect of using compressors is a reduced sound loudness, rather than a sound, that is optimized in loudness. This very often confuses the novice user.

To compensate this, the level must be raised again, after the compressor has finished its work. Many compressors therefore will deliver a mechanism for auto-compensation of the internal loudness decrease in any way.

Expander

The expander is the complete opposite of a compressor and will make the quieter parts of a sound gradually quieter, keeping the loudest levels nearly intact.

An expander often is used to give a sound more dynamics, if the levels are too dense, too prominent already. The expander can reduce overly penetrant sounds and transients to better fit into the mix.

Like all dynamics processors, the core of the expander is the peak follower for level detection.

Expanders can also, like the compressor form the transients of a sound. While the compressor usually squashes the transients, the expander will not so much alter the loudness of the transients but better separate them.

The expander has some similarity with the gate, as we will see later.

Limiter

The limiter is very much a descendant of a compressor, because it also will squash the peaks above a certain threshold.

The main difference to the compressor is, that the limiter will not reduce the final level in the same manner as a compressor does.

The limiter will just ensure, the loudness peaks do not exceed a given maximum. The compressor does not look for the output volume, it only looks for the peaks and reduces these to the amount of adjustment.

With other words, with a limiter you adjust the maximum output level and it squashes all above that level and with the compressor you specify, how much a particular sound should be squashed, without taking directly into account what the final volume will be.

A compressor can be constructed from a limiter with ease, by just raising the input volume, doing the limiting and then lower the output gain to the inverse amount the input gain after this. Viola, we just compressed the sound.

In fact, many so-called compressors and limiters are a hybrid between compressors and limiters and possibly other dynamics processors too and are not clearly defined in their basic function.

Transparent dynamics processors will not add saturation to the sound. So-called tube compressors are in fact dynamic saturators, rather than dynamics processors and will heavily alter the sound colour.

Gate

A gate is a special kind of an (extreme) expander, because it will reduce the level of quieter parts against a certain threshold. The gate can even silence (mute, fade out) the sound under the threshold.

A gate usually will pass everything untouched, that is above the threshold, not limiting anything, even if it is above the 0db mark. So its usually needs a combination with other processors in a production setup.

Gates are often used for noise canceling. This uses the psycho-acoustic law, that a noise is heard only in the quiet parts of a sound and usually (successfully) masked in the louder parts. The noise will be audible never the less there, but sometimes this can give a kind of “excitement” to dull sounds, which can profit then from the high frequency components of the noise. In fact, this is often audible in sampled drum sounds (with audible hiss component).

Gates are often also be used to create controlled rhythmic effects, by using extreme values, fitting to the main beat. This can in combination with mixing also emphasize a certain punch and alter all the following dynamics processors.

AGC

An auto gain control is a specialized method for equalizing the volumes if these are too loud or quiet over a longer amount of time. It is thought for improving the perceived loudness and to equalize really extreme level differences for an optimal audition.

Extreme level differences can happen with many different pieces from different sources (i.e. with broadcasting) and with music, that has extreme dynamics, like classical music for instance. Or with movies, which often require an artificial dynamics compensation for even being consistently audible.

The AGC often has an inbuilt limiter, which ensures, that the maximum levels will not be exceeded. So there are mostly 2 important parameters for the AGC, which are the minimum and the maximum thresholds.

The reaction times of an AGC are clearly larger than the attack/release times in other dynamics processors.

Maximizer

The maximizer is mostly a quite heavy combination of several multi-band dynamic processors (compressors and limiters, AGCs ...), that are combined with one “glory” goal: making the sound louder than usually possible, without noticeable distortion.

However, these methods often also will incorporate saturation and other psycho-acoustical methods and tricks to intensify the result. Some Loudness maximizers will even heavily colorize the sound and alter the stereo image of the sound, which is not everyones taste.

An overcompressed sound always tends to become dull and has usually very low stereo excitement.

Conclusion

All these basic dynamic processors have one important thing in common. It is their inbuilt peak follower, which is able to analyze the realtime stream of audio in the time domain for any level changes.

In the digital domain sometimes a “trick” is used, the so-called “look ahead” mechanism. This always will introduce a processing latency an this theorem will not work in the analog domain, the real world. Because nobody actually can tweak the time that way (looking into the future). It is working in the digital domain only because there is ALWAYS a latency (even if very small) of audio sample buffers anywhere, which can be analyzed prior the sound actually will be audible on the output.

NONE of our DYNAMICS SERIES units will use a look ahead mechanism. So these are well suited for realtime usage, because there is virtually no latency.

The JAX DYNAMICS SERIES

We have build a set of core modules with well defined functions. Only most useful controls are available with the intent for simplifying the usage as much as even possible. There are way too many complicated dynamics tools out there, which are overly complex and mostly also intransparent of what they even do, have confusing undescriptive adjustment terms and are difficult to master.

We think, a dynamics processor should be as transparent as possible and not introduce saturation or distortion by default and even not without any user option and control.

People tend to use merely the presets on complex dynamics tools, which are not really improving (rather damaging) their mixes, because these presets are not optimized for their individual sound, they are in fact not optimized for anything.

We highly suggest to optimize any dynamics processing to the essential things that a mix currently is needing. This requires careful listening and good visual supporting tools. It also requires some amount of experience and distance from the creation process.

Real stereo processing

All our dynamic effects will operate in linked stereo mode by default. If true stereo or mid/side operation is chosen, both channels will be processed independently with their own parameter sets and displays.

Stereo linked processing ensures, both channels will not run out of the stereo phase due to the automatically applied level adjustments by the independent stereo dynamics processors. However, there may be special cases (i.e. binaural audio or surround or such), where explicitly independent stereo channels are required.

Mid/side processing is useful, if the stereo image or the mid (mono) part must be handled separately or if the audio material has a strong emphasis on the stereo image. There is no general rule for that.

No Latency

None of the effects in our series will introduce any audio latency in normal operation (there is intentionally no look ahead), which make them useful for delay-free realtime recording and effect processing. Most audio units we tested, introduced generally a clear latency (look ahead), which was not even able to be compensated by the host applications while down mixing and using send busses. We also could not find, that the final results were significantly better with such tools. In fact, there were allot of problems with the stereo phase of a mix caused by such audio processors.

Please do not try to directly compare our JAX Dynamics Series effects with usual dynamic effects or other available tools, because there are some things operating very differently. We do not aim to emulate any available analog equipment, nor did we “copy & paste” the commonly used paradigms with our new dynamics series. And thats actually intentionally.

The PRO Mode.

~~Selected dynamic processors in the JAX Dynamics Series (where it makes sense) have a special feature for specifying the effective frequency range for the dynamics processing. Only the selected range will be effected by the DSP, all other frequencies will be passed thru untouched. This of course will not make any sense with a brickwall limiter.~~

~~The range also can be inverted for basically providing a edge sharp bandpass/bandstop filter for the dynamics processor. This approach is somewhat different to usual multi-band dynamics processors, which is often used but quite complicated and difficult to use, along with allot of other mixing problems due to frequency band overlapping.~~

~~Our filter delivers ultra sharp surgical results in the frequency domain. It consequently uses the equally tempered MIDI note frequency scale for providing musically useful results by default.~~

~~Please note that this special mode introduces a clear latency because we use FFT based bandpass filters for the effective frequency range. This approach is merely thought for special mastering purposes instead of realtime recording or realtime effect processing. The latency of the FFT filter is fixed to 4096 samples.~~

(Remark: We changed the PRO Mode with version 1.5.)

The PRO mode is available for selected units (where it makes sense).

The PRO Mode uses now a conventional multi band scheme, whereby 3 independent stereo processors work together with their own selective frequency range. The range can be adjusted freely. Each of the dynamics processors has its own set for the adjustable parameters.

If the ALLPASS mode is used (default mode), only a single dynamics processor will process on the full frequency range, the other bands are suspended. As soon one of the selective bands is chosen, the processor will operate in 3-band mode. The user can select and adjust the parameters for each single range. The frequency range selector is scaled to a musical useful scale, which closely corresponds to the logarithmic MIDI tone scale.

So users can switch between these 2 different processing modi in real time for a comparison with the original. With the multi band mode, the frequency splits can be adjusted with our horizontal range sliders. The upper range will separate the highest frequencies, the lower range will separate the bass frequencies and the medium range will adjust the remaining mid frequencies. All 3 bands will get their own independent dynamics processor instance with its own set of parameters.

The frequency split is made with analog modelled filters for extra warmth and latency free natural fading sound. The analog filters do not introduce any latency as the FFT based approach. They also do not alter the phases of the signal. These are adjusted to be as transparent as even possible. The price for this is, that these bands do more or less overlap, because strict frequency separation is not well possible with analog filter models. And strict isolation is also not wanted with our dynamic processors.

If you switch to multi band processing, the single frequency ranges can be muted temporarily for audition and special filtering effects. It usually allows the fine adjustment of the dynamics effects by listening to separated bands while the dynamics processors are working.

Special Volume Controls

Our DYNAMICS SERIES units all have independent input and output volume controls for each single band. The input volume adjusts the volume prior the dynamics processor, the output volume adjusts the volume of the output from the dynamics processor for each band separately.

This is very special, compared to other multi-band processors, as it allows to actively and effectively alter the frequency adjustment of the final sound to quite extreme values.

This way no extra equalizer is necessary. With the bandpass adjusters, the frequency range for each band is adjusted separately and the input/output volumes will work like simplified auto-adjusting mastering equalizers with no emphasis (bass, mid, treble)

There is also a final limiter device inbuilt, that prevents bogus boosting of loudness values, that would cause digital clipping. If the mix starts pumping or extreme frequency damping is audible, which was not intended, so please always check the volume values for each band and possibly reduce the shown bogus volume values. The graphs will also hint to such values and should be checked in regularity.

This flexibility of adjusting the levels freely can modify the final sound quite and the user must respect this fact for successful results across all of our DYNAMICS SERIES units.

Our units are NOT build to simulate any available or ancient hardware devices and we also did NOT apply commonly copied digital paradigms with our series (you will for instance not find any controls like “ratio” or “knee” and such on our units), but we rather created a powerful suite of new generation dynamics processors, that are easy to use and will deliver excellent results, based on a clear logic and our understanding of dynamics processing.

The Visual Aids.

The main screen of our plugins will show the new horizontal “flow” display. We specially developed this for our JAX Dynamics Series. The flow display tracks several things from the core DSP kernel with different color overlays in realtime.

At first, there is the level peak detection, which draws continuously grey shades for the peak levels of the output signal. ~~By default, this is the mid signal, extracted from the stereo source. You can switch the display to draw db scaled lines and also switch the modi for operation, which may change the display (i.e. for independent stereo processing or mid/side processing).~~
We changed this with version 1.5 to displaying always a bi-directional waveform, either left/right or mid/side (regarding the selected processing mode) above and below the center of the display.

Default mode for the level scaling is b.t.w. always drawing linear levels, which scale from 0 to 1.0 and correspond to the absolute sample values or to “percent”, rather than decibel. We find this mode much more comfortable to read and also more logical than the somewhat “artificial” decibel approach.

In the middle of the screen, a red horizontal activity line will indicate the compression/expansion/limiting action (it is bi-directional too) and there is also plotted the current amount of multiplying action reduction/boost of the signal in other emphasized colors.

You can see that reduced levels will always increase to the bottom and boosted levels will increase to the top from the horizontal center of the flow display. This way you easily can see, what happens with the modified signal in realtime. The input level is not drawn extra, because we find this rather confusing and makes everything quite difficult to read and understand.

In multi band mode, the displayed levels will automatically switch with the selected frequency range. This means, each band will have its own realtime display.

There may be several special controls and options for adjusting the flow display for each kind of dynamics processor. The limiters and the gate for instance usually will merely reduce levels and never boost anything. Other units, like the AGC and the Compander will operate in both directions and possibly with different methods dynamically.

Just tweak the parameters and see, what happens. Visual control of this sort is great for understanding the signal flow of basic dynamics processors, even if the results are mostly clearly audible, sometimes strange side effects can be easily identified with the visual aids.

Some units may also feature an optional frequency flow display, which is merely for informative purposes and for the adjustment of the PRO features. In this mode the selected frequency bands can be analysed visually. Each dot or line will correspond to a MIDI tone. So melodies are visible like in usual MIDI sequencers or in our SPECTRO and CHROMATIC Series.

The Auditioning Aids.

with the „listen“ button, the currently selected band can be made audible (isolated). This allows much better frequency band adjustment.

Some of our dynamics units have a special function, that is able to invert the signal, meaning that the processed audio is subtracted from the original. So if loud peaks are passed thru a gate, for instance, now the cut parts will be audible instead. With other words, it will output the part of the signal that would be removed while normal operation (in the case of a gate). However there are complex side effects by inverting the signal with the other kinds of dynamics processors.

But by inverting the signal, a better understanding of the internal working via a sonic effect can be achieved or certain parameters even can be much better adjusted by inverting the signal temporarily. We find this is a really useful aid and you will not find such with many other dynamics processors.

Inverting works on each band separately.

The JAX DYNAMICS | Visual Compander

A compander is a combination of a compressor (increasing loudness density) and an expander (doing the exact opposite, stretching the loudness relations). This combination may sound contradictory at first but it is not.

Both modi are mutual exclusive per band. The adjustment can be done with only one “multiplier” (exp./comp. knob) parameter in realtime, which can be negative. A negative value will expand levels, a positive value will compress. This can be automated.

However, expanding is rarely used nowadays but it can help to recover lost or too much squashed dynamics. In multi band mode the compressor can be used alongside with an expander (on different bands) this way. So with the JAX DYNAMICS | Compander you can do expansion and compression on different frequency bands at same time. This is extremely powerful and creative.

Compressors and expanders are often used to alter the rhythmic and the loudness presence of mixes by using relatively short values fitting to the beat, creating more or less a ‘pumping’ auto gain effect. It also can be used to change the transients and the entire presence and loudness and the pressure of tracks and single sounds.

With our units even the frequency content can be adjusted with this in one step.

Saturation is Optional

Our compander will produce very transparent sound, not adding any saturation or desaturation by default. Such modification of the signal is implemented as an extra (optional saturation) parameter.

The special saturation parameter is also bi-directional. Values below zero will de-saturate and values above will saturate. De/Saturation adds new harmonics (or will remove these) to/from the fundamental frequencies. This is comparable with creating even and odd harmonics from the available fundamentals.

Please note, that the de/saturator is implemented for rare special purposes. It effectively can boost loudness, if positive and add grid if negative. But usually the effect should be applied sparingly to selective bands for all kinds of excitation. This effect is not oversampled and may introduce aliasing artefacts with extreme, unthought usage.

Using companders, generally needs some good listening, rhythmic feel and also some amount of musical experience. Wrong or unthought usage of companders can damage the sound and musical feel quite easily.

Please note, this is in difference to usual compressors/expanders an auto-levelling compander, which means, it usually will not reduce the gain of compressed audio. Instead it will always tend to boost the level against 0 db, similar to a classic “maximizer’ effect. However, the output level can be adjusted manually with post processing (output gain) to fit other needs. I.e. The classic compressor can be emulated by increasing the input level and reducing the output level at same time with the inverted value.

The usage with the great visual control of our compander is really simple and does not need special expertise. The most important control is the exp./comp. knob, which can be adjusted on each band separately. You will notice, that multiband compression immediately can fresh up a sound, even if it is rather complex, like complete mixes, that are already mastered.

While the allpass mode quickly tends to pumping and becoming muddy, the multiband mode will deliver much smoother results and give control over the emphasized frequency content at same time. You even often can omit an extra EQ and exciter, if you carefully adjust the 3 frequency bands with only compression/expansion and saturation parameters.

Users have absolute control over the internal dynamics processors, because their action is displayed in realtime at selectable speed with meaningful graphs. Users will learn in very short time, how to reach the desired results by adjusting some controls. The visual aids also enable to adjust mixes purely visually to useful results, without even listening to the sound.

Note: We do not recommend to use our dynamics effects in the “loudness war”. Our units are not intended for everything squashing to death. So please use other tools, if your intent is to make your sounds indistinguishable from the screaming, egocentric commercial blast.

Our compander also has an integrated limiter.

The rule for the integrated limiter is: as long the peaks are above 0 db, the processor at first will reduce it (normalize), before specifically compressing or expanding anything. Otherwise, the audio unit will operate in the supposed active compression/expansion mode, regarding the current parameter adjustment. Out internal compander algorithms can work only correctly, if the sound levels are consequently adjusted below 0 db and normalized. If the multiplier control is adjusted to zero, the device will effectively work as a (soft) limiter, reducing exceeding values above 0db.

There are controls for attack time, hold and release time, which alter the integrated followers action speed. These are very important for the final result. If these values are adjusted wrongly, the sound may pump or transients may be lost and damaged.

The amount of the multiplier will adjust the intensity of the compression/expansion up to the chosen value. But the effect is limited to useful amounts and usually cannot badly destroy the sound.

The compander explicitly supports the PRO mode with multi band processing. With multi band processing, best optimized results can be achieved. But there are intentionally no “wonder presets” that make up the ultimative sound. Each sound must be individually optimized. This is the only “good” usage of professional dynamics processors like ours.

Advanced Stereo Processing

Stereo processing has many different possibilities, especially for dynamics processors. It very much depends on the kind of audio material, which method to chose for best results.

Generally the linked modi are recommended if stability of the stereo image is required. Sometimes the independent modi are preferable, if stereo enhancement is welcome. The four available modi may sound completely different with the same set of parameters. Users may switch between the different modi anytime and select the best result.

Unlinked mid/side processing often will increase the psychoacoustic loudness, but the effect should be carefully watched for mono compatibility, which is still extremely important today. We recomment the JAX TRIPLE SERIES 3Stereo for this task.

The following modi for advanced stereo processing are available:

- Stereo Linked: This mode will temporarily extract the mid signal for the follower and process both stereo channels with this signal synchronized. The display shows the mid signal below the center and the side signal above. So it will most likely appear asymmetric, which allows the visual judgement of the resulting stereo information.

Note: The display will show M/S (mid/side signal), because the dynamics processing will be driven by the mid signal peak follower. So the waveform plot may look more or less asymmetric, which is not a bug but an intended feature. If the side signal is very strong, you should watch the stereo phases with special Tools, like out JAX TRIPLE SERIES 3Stereo.

- Stereo Independent: The left and the right channels are processed independently. Both channels have its own peak follower. The display will show the left (below center) and the right stereo channel (above center). The waveform most likely will look nearly symmetric, regarding of the amount of stereo information.

- Mid/Side Independent: The mid and the side channels are isolated and processed independently. Both channels have their own peak follower. The display shows the mid signal below the center and the side signal above. Strong stereo phases will have a large wave plot on top, weak stereo will have a small wave above the center. This mode is excellent for boosting the stereo image. However, it cannot generate stereo content from pseudo stereo (mono) recordings.

Note: If only the lower waveform part is displayed, this is not a bug, but the audio is obviously pseudo stereo, which means, that both channels will be exactly the same. So no side channel can be extracted.

- Mid/side Linked: In this mode the mid and side signals are isolated and processed with data from one peak follower (the mid signal) synchronized. The display shows the mid signal below the center and the side signal above.

Users should experiment with these different modi, because our experience shows, that different audio material has particularly drastic impact on the sonic result. Sometimes astonishing aspects of a mix come out with the different stereo processing modi.

Our JAX DYNAMICS SERIES processors are highly creative and flexible sound modification devices, that even can create new sounds from existing. They also can be subtile enougth to be well suited for using as high end mastering processors.

The JAX DYNAMICS | Visual Follower

The follower is a special audio unit in our JAX Dynamics Series. It does no directly modify the audio stream but does peak analysis and (optional) continuously sends MIDI data of the internal peak detection result.

The user can specify the MIDI controller to be sent and is also able to invert the signal direction and adjust other parameters. The interval of sending can be specified to some extend, whereby the effect always ensures, that data is sent only, if peaks actually changed.

The sent MIDI signal can be used externally to adjust parameters of other plugins in realtime, that are peak/level dependent, even if these do not support level dependent audio processing internally.

Some hosts may support mapping MIDI data to audio unit parameters, so plugins, that do not support direct MIDI data for input can be automated via a user defined host mapping this way.

The creative use is quite endless with our follower audio unit.

Please note, that the receiving plugin must support internal parameter automation smoothing to not to produce audio artefacts with possibly fast jumping follower data.

The follower also includes the frequency flow display for informative purposes. For pitch tracking output per MIDI, please use one of the polyphonic frequency analyzers in our JAX CHROMATIC or JAX SPECTRO Series.

The JAX DYNAMICS | Visual Limiter

The limiter will (only) reduce the loudness of an audio stream down to a defined maximum level. There are parameters for adjusting the speed in (fractions of) seconds. This is not a hard limiter and the level will be always reduced in dependency of the chosen speed and amount. It will not force the level to the maximum value, so it can slightly exceed it for a short time. If a strict limiting is required, an additional, so-called brickwall limiter should be used.

A limiter can make sound more dense in loudness and equalize differing levels over time. If the pre-gain is boosted, it even can maximize general loudness. The correct adjustment for attack, hold and release must be chosen carefully to make it sound correct.

Our limiter will leave all signals below the adjusted maximum level untouched, if no peaks are exceeding. The core limiter is as transparent as possible, not adding any saturation by default.

The JAX DYNAMICS | Visual Gate

The gate will only pass thru audio, that has levels above a certain minimum value. It is very useful for noise canceling for instance and has a wide range of further creative use. The gate has controls for attack, decay and time and release time in (fractions) of seconds.

The Gate can be seen as the opposite of a limiter. So it will not affect any levels above the specified minimum, even if these clearly exceed the 0 db boundary. There is no adjusting of a maximum.

A gate can be used to emphasize the rhythmics of a composition or for creating interesting automatic fading effects.

Our gate supports multi band operation.

The JAX DYNAMICS | Automatic Gain Control (AGC)

A classic AGC is basically a specialized limiter with automatic gain control, if levels are too differing in time. A classic limiter will usually not alter levels below the threshold. An AGC therefore is often used to equalize the impact of clearly different loudness in pieces of sound (like for instance wildly mixed playlists) or extremely changing dynamic levels, like in some classical recordings for instance. Everything with the general goal of a better perception and optimized loudness balance.

AGCs should not be used for loudness maximizing with classic mastering or for usual limiting.

AGCs are often used in customer equipment and broadcasting therefore, automatically levelling extreme loudness differences, most likely present in surround sound movies, followed by speech and by commercials and so on.

There is a minimum and a maximum level control, which tells the processor how loud the level for operation must be at least, before the DSP will be effective at all and how much the sound will be boosted at maximum. Both values are to chose carefully, otherwise unwanted noise boosting can occur and possibly other unexpected results of sound raising.

Attack and the release times should be adjusted moderately (selecting clearly longer times above 1 second) to prevent audible pumping with extreme level changes. The maximum parameter should be adjusted moderately to the absolutely required boosting amount for keeping consistence, not just simply using any bogus values. Otherwise the audio unit cannot ensure quality of operation.

An extra brickwall limiter is strongly recommended to be placed behind an AGC, because due to larger reaction times this effect may exceed the 0 db mark for the active attack and release times.

If loudness maximization is a requirement additionally (i.e. a dance channel radio), an additional compressor can be chained after the AGC.

The JAX DYNAMICS | Brickwall

A brickwall limiter is a special limiter device, that ensures there are under no circumstances levels above 0 db at the output. It is mostly used for post production mastering and for protecting against signal bursts while mixing and editing or for signal processors, which occasionally “drive crazy”.

Our brickwall limiter will force the sound level to the 0 db boundary (normalizing) in realtime, but trying to apply as less distortion as even possible. Because of the reason, that our effect will not use any look ahead buffer (latency), it may may hard clip extreme loudness levels that could not brought under control in time, for the purpose of not exceeding the 0 db mark.

Some filters and effects will (for unknown reasons) not limit their levels at the output at all, which may cause NAN states and malfunctions on the used audio busses and other connected units. We have seen people blaming plugins because of refusing the supposed operation, but the real reason was another plugin connected anywhere in-front of the chain, generating constantly bogus output values which is generally a really bad practice.

A brickwall is also important for protecting sensitive external equipment, including speakers and other effect processors that do not like values above 0 db that much (expecting normalized values) while operation.

The brickwall has no special parameters, as this would not make any sense here. However the state of operation can be visualized with the user interface.