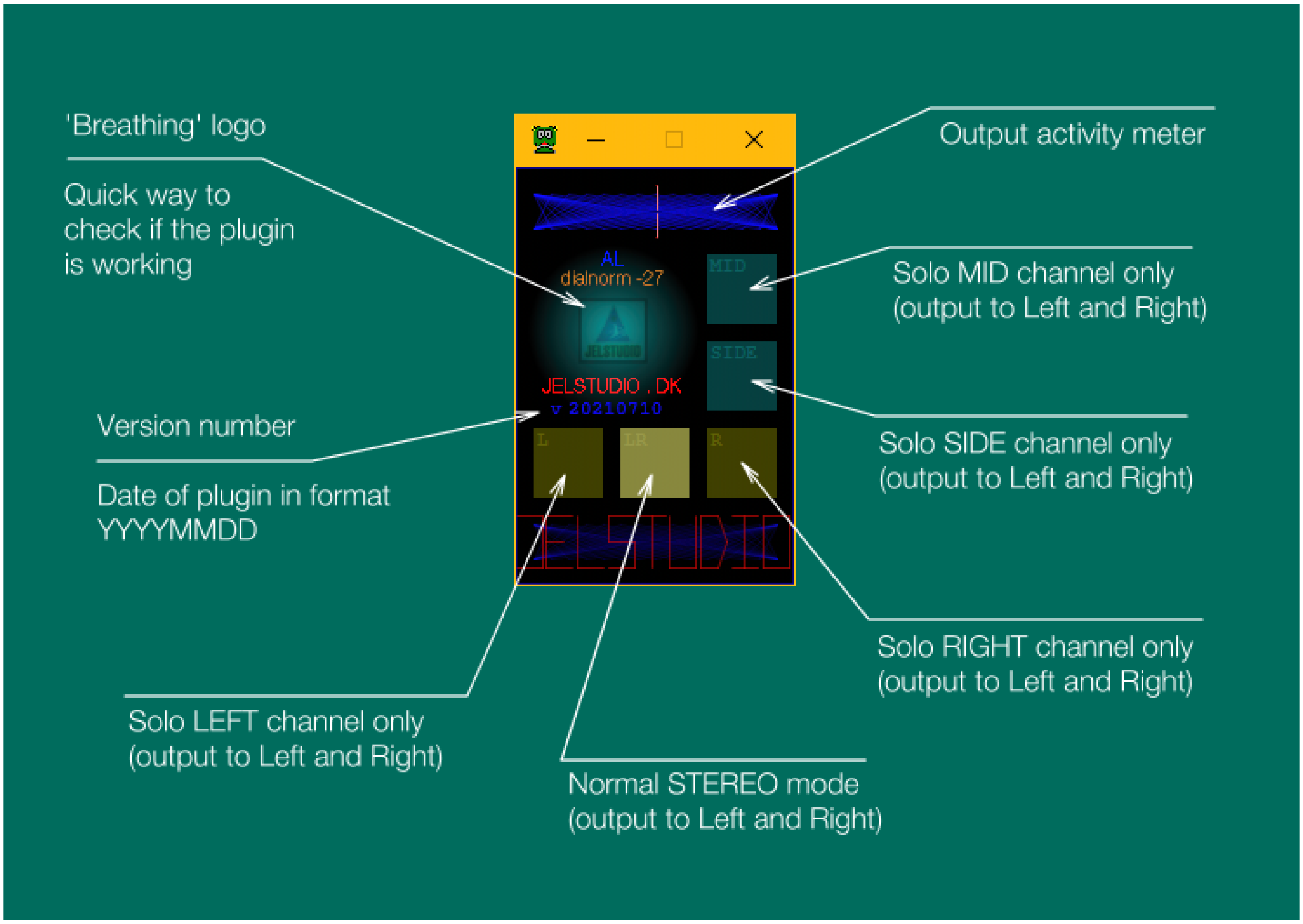

JELSTUDIO's "AL dialnorm -27" (Auto Loudness for dialogue)

version 20210710

AL - Automatic Loudness (for movie/television or radio dialogue)

AL will automatically turn the volume up and down, so the audio you play through it (Or render/export to a file) will constantly have the same overall loudness.

AL dialnorm is based on AL12, but is designed specifically to make speech sound more coherent and meet loudness delivery specifications of -27 LUFS/LKFS with a maximum of -2 dB 'True' Peaks.

It has a latency of about a quarter of a second.

In order to achieve better overall sound-quality AL is NOT fully compliant with the loudness-specs defined by ITU (Including ATSC and EBU), but is generally close to those defined targets.

Quick-use:

User interaction:

Buttons are controlled by left-clicking on them with the mouse and are lit brighter when ON.

Blue button column:

MID = Solo the mid-image of the input-audio
In this mode only the mid-channel can be heard. It will play in both the Left and Right channel.

SIDE = Solo the side-image of the input-audio
In this mode only the side-channel can be heard. It will play in both the Left and Right channel. Mono audio will obviously be silent in this mode, since it has no side information.

Yellow button row:

L = Solo the LEFT channel of the input-audio
In this mode only audio present in the Left-side can be heard. It will play in both the Left and Right channel.

LR = Play both channels of the stereo input-audio
In this mode a normal stereo-image can be heard. This is the default start-up mode.

R = Solo the RIGHT channel of the input-audio
In this mode only audio present in the Right-side can be heard. It will play in both the Left and Right channel.

Information display:

The 'needle-meter' at the top shows whether AL is amplifying or reducing audio-gain.

When the needle is in the:
Center: output-gain is similar to input-gain.
Below center (left side): output-gain is below input-gain (AL is reducing the volume)
Above center (right side): output-gain is above input-gain (AL is amplifying the volume)

The meter technically shows* from -20 LUFS/LKFS (Fully left) to +20 LUFS/LKFS (Fully right), but is only meant to give a general indication of whether the incoming audio is amplified or attenuated so you can use it as a 'gain-stage' indicator if you want (most often this is not necessary, but the 'optimal' input-level is when the meter mostly moves somewhere near the center area)

The bottom ‘spot-light display’ has 2 functions:

Two yellow lights from the bottom, lighting upwards, mean the input-audio is below the gate-level (When audio is below the gate-level, the gain will slowly reset to center)
If you get frequent flashing yellow lights during play, your input-audio must have its volume raised before entering AL.

Two green lights from the top, lighting downwards, mean the soft-limiter has been triggered (The lights will come ON for a few seconds and then turn OFF again)
This is normally not a cause for concern, but just to make you aware that the output-peaks may be close to the ‘legal’ limits (In this case louder than about -4 dB FS)

The JELSTUDIO logo is 'breathing' (oscillating in brightness) to give visual indication the plugin is still running correctly and has not 'crashed'/malfunctioned.

Some of AL's key-points:

- Audio-chain sample-peak overload protection (Max output is 0 dB FS unweighted)
- Quarter of a second latency (Plugin has delay compensation, so can be used in multi-track environments)
- Inspired by "Recommendation ITU-R BS.1864-0" (See technical description for more info)
- Inspired by "Recommendation ITU-R BS.1770-4" (See technical description for more info)
- Has 1 loudness-algorithm (JELSTUDIO's own mode, based on the "Advanced Television Systems Committee standard A/85" but modified for automation)
- Can handle incoming audio-signals that has 'overs' without clipping (Internally the plugin can operate with sample-peak values up to about +10 dB FS)
- Works on mono and stereo signals with loudness-compliant panning.
- Employs soft-rolloff band-pass filter (full pass-through of audio above 100 Hz and below 10,000 Hz, with soft roll-off outside those frequencies)
- GUI layout is optimized for 'peripheral vision overview' (when looking at the center of the GUI; all important displays can be read at a glance with peripheral vision) based on aviation-safety studies.

Technical description:

Be aware that AL dialnorm exports a 2-channel audio-signal (LEFT and RIGHT channel, aka channel 1 and 2! If you plan on rendering the corrected audio from AL dialnorm to a true single-channel mono-track, rather than a 2-channel 'phantom' mono-track, you must make sure you understand how the 'pan law' of your DAW or editor is defined, since this can affect the resulting loudness level after downmixing (-27 LUFS/LKFS in a 2-channel phantom-mono audio-track is not always downmixed to -27 LUFS in a single-channel true-mono audio-track. It can be anywhere from -30 to -24, depending on the pan-rule of the system doing the downmixing. Test, and adjust if needed, your system to verify your exported renders meet delivery specs)

AL is designed to be rigidly time-stable (fully free of mathematical errors accumulating over time) and should be safe to use on long-term audio-streams running 24/7/360 (for example; automated broadcasts that run unsupervised)
(Technically 'set & forget')

AL is inspired by "Recommendation ITU-R BS.1864-0" and "Recommendation ITU-R BS.1770-4" but uses proprietary filtering and algorithms (The differences are noticeable, but AL gives more weight to the vocal frequency-spectrum. Compared to K-weight, this means AL allows more bass and treble. In a mixed dialogue/music program, such as television-comedy, this allows music to sit 'better' in the overall mix with dialogue).

Because of this difference, AL can not be said to be targeting a specific LUFS/LKFS value. However, generally speaking "AL dialnorm" tends to output a material-dependent loudness of around -29 to -25 LUFS/LKFS, with true-peaks soft-limited to about -3 LUFS/LKFS.

Most speech-tracks (in the format of either true 1-channel mono, 2-channel mono, or stereo) with clean dialogue (and even 'dirty' dialogue-tracks, as in tracks with added effects) should come out 'legal', in terms of popular (in the year 2021) delivery-specs specified for 'dialnorm -27 LUFS/LKFS' material.

Tracks where dialogue and music has been mixed can present a challenge in some special cases, but more often than not they should work fine as well.

The integrated loudness-target is -27 LUFS/LKFS in dialnorm-mode, but due to the nature of continuous level-adjustments it may land a bit below or above (generally within +- 2 dB FS, but in rare cases it may be more)

The Dolby DialNorm is NOT implemented to published specification, because constant level-adjustments are needed (Dolby DialNorm features a gate, which obviously is a problem for the automation since dialogue tracks with effects and/or music must still be controlled during the parts without speech)

It is recommended to spend some time getting acquainted with how AL 'sounds' with various audio-material, to get familiarized with the impact it has on the sound.

Despite the band-pass filter and the volume-automation, the dynamics of the dialogue should not sound 'pumping' or artificial.
If you need the speech to sound clearly compressed, you should use a dedicated compressor before AL, as AL

will attempt to make the speech sound un-compressed and neutral/natural.

As with all automation; it is not 100% perfect and there is no guarantee it will produce a ‘broadcast-legal’ stream even though it should with most ‘normal’ material (Most music and speech should give expected results, but test-tones will often not. This is considered normal)

END OF TEXT

Website: https://jelstudio.dk/VSTplugin_AutoLoudness/

Email*: jelstudio@hotmail.com

Twitter: https://twitter.com/JELSTUDIO

Facebook* (personal to Jacob Larsen): https://www.facebook.com/jacoblarsen.248

General DSP programming: Jacob Larsen

Loudness-mode algorithm-programming: Jacob Larsen

GUI programming: Jacob Larsen

VST compiler: Derek John Evans

BETA-tester and sound-quality inspector: Sébastien Wittebolle.

*If you receive 'weird' nonsense/spam email from this account, it is NOT sent by JELSTUDIO!

Feel free to use Twitter/Facebook instead.

(When using Facebook do NOT accept game-invitations or other ‘crap’ like that!)