
# JELSTUDIO's "AL" (Auto Loudness)

version 20170110

AL is an automatic perceived-loudness normalizer for LIVE use.

AL will automatically and continuously adjust audio-gain to maintain a similar overall perceived-loudness of the audio going through it.

AL has 18 different methods of adjusting loudness.

Technically it is a 32/64 bit VST2 audio-effect plugin mainly for use in a LIVE VST-host (for example with LIVE stage-performance audio, or with music/TV home-stereo audio), but it can also be used in DAWs and sound/video-editors on Windows.

A few ideas of what AL can be used for: to balance dialog and sound-effects when watching movies, to avoid loud commercials when watching TV-shows, to balance loudness between songs when listening to music. It can also be used on LIVE internet-broadcasts.

## Some of AL's key-points:

- Runaway volume peaks are clipped at -0.1 dB FS
- Zero latency (for LIVE use or for tracking)
- Inspired by "Recommendation ITU-R BS.1864-0" (See preamble for more info)
- Inspired by "Recommendation ITU-R BS.1770-4" (See preamble for more info)
- Can handle incoming audio-signals that has 'overs' (internally handles signals hotter than 0 dB FS, either real peaks or ISPs)
- Works on mono and stereo signals.
- 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)

**Preamble:**

When initiated; AL may need up to 60 seconds to stabilize loudness (generally less)

If attempting to use AL in a mixing/mastering context; it is recommended to feed it some 'calibration-audio', if possible, prior to the audio you want adjusted (for example, a 30-second part of the track's coming audio)

The pause between 'calibration-audio' and target-audio should be as little as possible (if a reverb-effect is to be added to a track, consider running the dry track through AL first and apply the effect afterwards)

AL is designed to be rigidly time-stabile (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)

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 small, 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 tends to output a loudness of around -26 to -23 LUFS/LKFS)

## Main controls:

“FAST” / How quick to respond to loudness-changes.

“SLOW” / How quick to respond to loudness-changes.

"PASS" (AKA Infinity-mode) / How to respond to loudness-level (low levels vs high levels)

"TIME" (AKA Time-lock or Dynamic-lock) / How fast to correct for long-term loudness-changes.

"LOCK" (AKA Static-lock) / How much amplification to allow.

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

## Information display:

The 'needle-meter' 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 is not calibrated to any specific level-units, but simply a general indication of whether amplification or reduction is taking place.

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

## Operation:

It is recommended to spend some time experimenting with different settings on various audio-material, to get familiarized with the impact it has on the sound.

Since AL allows 18 different ways of handling loudness, which each react differently to different types of audio, it is difficult to give a specific description of all possibilities.

In general terms:

PASS affects how low-level signals are considered.

This can have significant impact, depending on how loud the input-signal is.

When ON; noise can get increased and sound can get more 'fatiguing'. It can better normalize dynamic-range, but can also lead to 'loudness-sausage' (flat-line). Fade-ins and fade-outs can get exaggerated. Dialogue can get 'roomy' (a room's reverberation sound may get exaggerated between spoken words in a piece of dialogue).

When OFF; original dynamic-range may be better preserved, which may lead to unwanted loudness-variations. Signals that are input at a low-level might not get corrected (for example; recordings that are intentionally recorded at a low level to allow a lot of headroom, such as movies or high-bit nature-recordings)

TIME affects long-term dynamic-range.

This is more useful on longer listening-sessions, such as full concerts, full music-albums, etc.

When ON; short-term dynamic-range is better preserved, while long-term dynamic-range is still attempted normalized. It can lead to momentary periods with lower loudness.

When OFF; original long-term dynamic-range is ignored.

LOCK limits the maximum gain that can be applied.

This is often useful in combination with PASS, since it allows low-level signals to be considered without allowing gain to get excessively high.

When ON; avoid low-level noise/sounds from getting too much 'in your face'. Will make musical/instrument diminuendo, and general fade-ins and fade-outs, sound more natural.

A few examples of suggested settings:

If listening to a commercial recording of classical music (full orchestra or perhaps a single pianist):

SLOW, TIME

If watching a movie with a theatrical soundtrack:

FAST, PASS, LOCK (if watching at lower volume, for example at night. Optionally LOCK can be turned OFF to maximize quiet parts of dialogue-loudness the most)

SLOW, PASS, LOCK (if watching at higher volume, but sound-effects should still be contained)

If listening to a commercial recording of popular upbeat music (such as dance and rock):

FAST, LOCK

If highest degree of 'loudness-sausage' is desired:

FAST, PASS

If most original dynamic-range should be preserved:

SLOW, TIME, LOCK

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