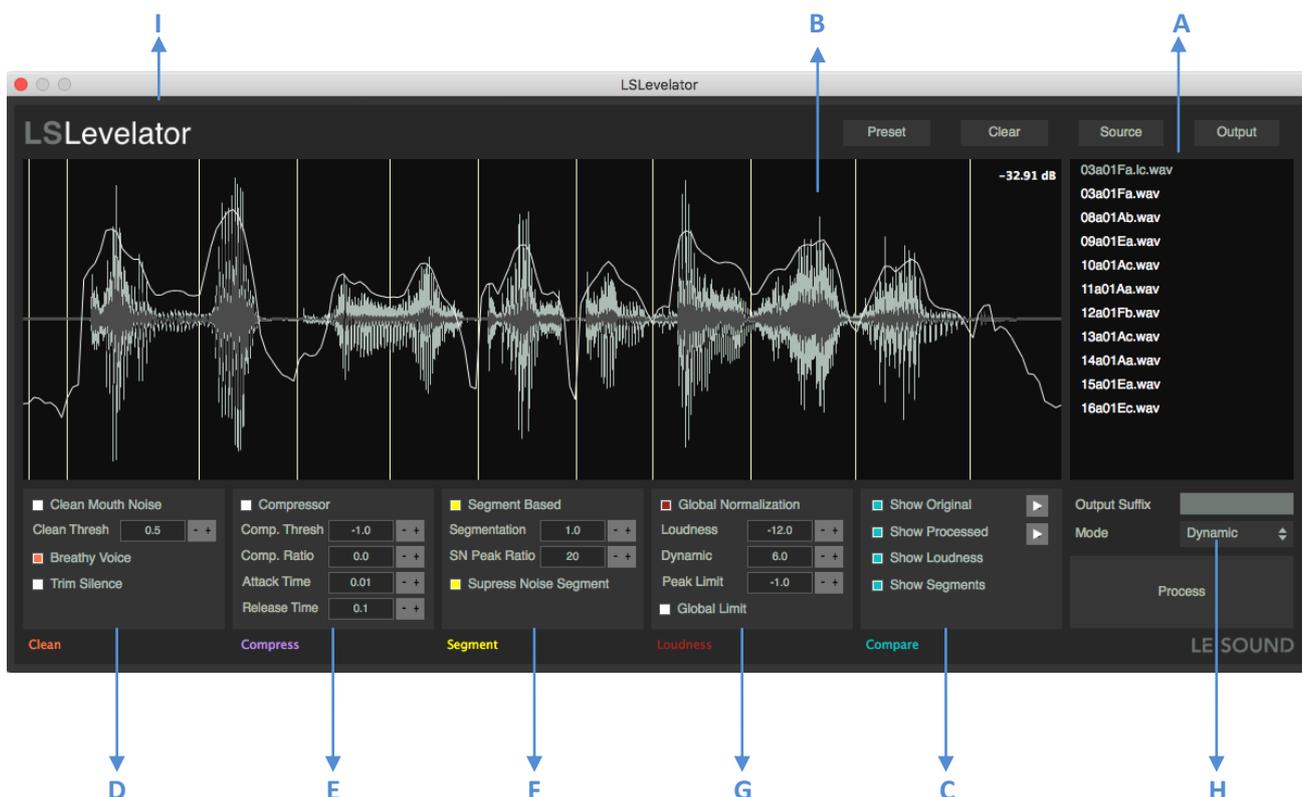


## LS Levelator 2 Quick Start Guide

LS Levelator is an energy/loudness normalization tool for batch processing of large amount of files. This tool not only makes use of local energy variation but also takes into account the loudness model to enable a perceptually relevant normalization. The batch process is expected to help localization and broadcast professionals to render the dialogue dynamics more efficiently and effectively.

LS Levelator 2 has added some more preprocessing features including mouth noise cleaning and dynamic range processor. It also supports custom presets.



### Source and output directory setup

Click **Source** to select the input directory containing files of WAV or AIFF format. Click **Output** to select the output directory. Type output file suffix if necessary.

Before processing, the list of sound files in the input directory will be listed in **A**. While selecting a file in the list, its waveform preview is displayed in the panel **B**.

After processing, panel **A** will list the output files.

### Preview/Compare

The preview panel **B** displays the original waveform (dark) and the processed waveform (light). One can switch to check each waveform using the Compare panel **C**. By clicking the transport buttons, one can listen to the original or the processed file for comparison.

There are options for displaying the loudness curve and the segment markers. **Show Loudness** displays the loudness curve or the original file which gives an indication of how one would like to normalize. **Show Segments** displays the segment markers, which provides interactive visual feedbacks while adjusting the Segmentation parameter.

## Clean D

The first preprocessing module is **Clean Mouth Noise**. For dialogue recording, one may produce noticeable clicks due to watering mouth or unstable pronunciation. By default, LS Levelator considers the recording contains **Breathy Voice** and uses a soft attenuation algorithm. If the voice contains little breathy sounds and the recording is of very low noise, one can switch off this option and a more aggressive cleaning algorithm will be applied.

Increase the **Clean Thresh** allows the algorithm to clean more potential clicks, vice versa.

**Trim Silence** removes the heading/trailing silence (continuous 0 in amplitude). Notice that the duration changes if any silence is removed.

## Compressor E

The second preprocessing module is dynamic range processing based on the usual compressor. This allows to compressing strong peaks before applying the loudness normalization.

Set the **Comp. Ratio** to 0 to use it as a limiter.

## Segment F

Treat the automatically detected segments respectively. LS Levelator automatically detected (speech) signal segments and the processing will then be based on the segment's loudness for normalization.

Reduce the **Segmentation** parameter to allow more segments, vice versa. Preview the segments using **Show Segments** in the Compare panel.

When Segment Based option is on, LS Levelator classifies them into signal and noise segments. By activating **Suppress Noise Segment**, the noise segments will not be normalized.

Decrease the **SN Peak Ratio** when the recording is noisy, vice versa.

## Loudness G

This panel contains parameters for loudness normalization. The units are in dB.

**Global Normalization** : Derive loudness distribution from all the input files. When combined with segment-based processing, it allows adapting normalization for all segments of the input files.

Adjust **Loudness** for the target loudness and **Dynamic** for the loudness range (only for Dynamic mode).

Set **Peak Limit** as the maximum allowed peak amplitude. When the **Global Limit** option is enabled, it simply takes the maximal peak in all files as the limit regardless of the custom Peak Limit. Notice that the peak limit applies a constraint to normalization and therefore one could use Compressor to attenuate large peaks before normalization.

## Normalization mode **H**

**Peak:** This is a traditional normalization w.r.t. the maximum peak level. The maximum peak level can be set manually using **Peak Limit** or automatically using **Global Limit**.

**Static:** This mode normalizes a signal from the file's average loudness to the target loudness, with constraints defined by the peak level. When **Segment Based** processing is enabled, it allows to applying respective normalization to the automatically detected signal segments.

**Dynamic:** Similar to Static mode, Dynamic mode further allows adjusting the loudness dynamics among segments or files. When **Global Normalization** is enabled, all segments detected in all the input files are taken into account for deriving the best loudness distribution.

**Bypass:** This bypasses loudness normalization and carries out the preprocessing only (Clean and Compressor).

## Output Suffix

Type a suffix name for better distinguishing different batches.

## Preset

Save and Load custom presets.

## Clear

Delete all files in the selected Output directory. It is handy to test different parameter settings but please use it carefully.

## License Activation

Click on the logo **I** will switch to the license activation panel. Copy and paste the product key to activate. The product key is available in the purchase confirmation mail or in your lesound.io account.

Notice that you will need internet connection while activating LS Levelator.