



Visual guide

LUFScale for macOS - version 2.2.3

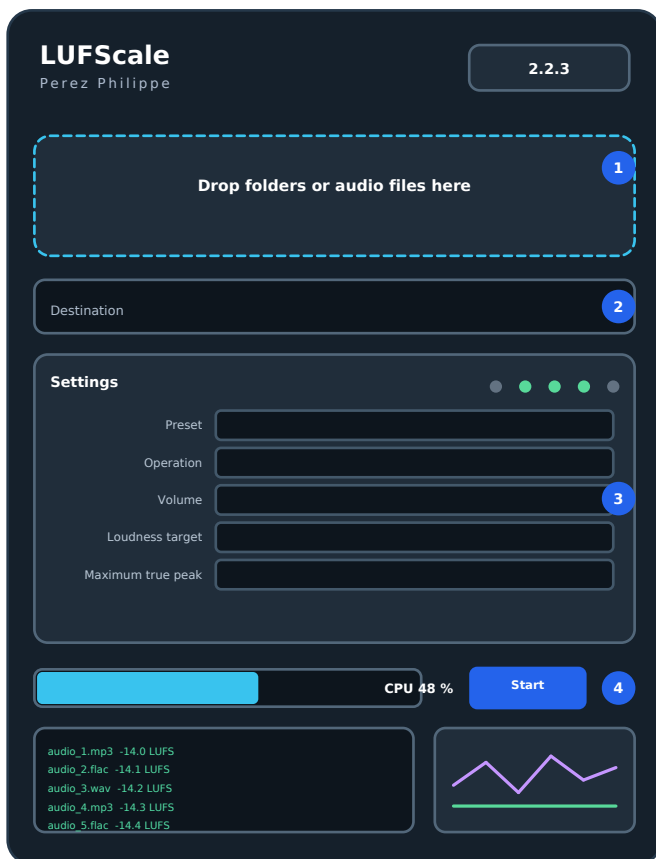
Quick start

What does LUFScale do?

LUFScale measures file loudness and, with Normalize, physically adjusts perceived volume toward a LUFS target while controlling true peak. Each source is analysed over its full duration, then the output is remeasured and verified. The result does not depend on tags or a compatible player: levels become more consistent across files, deviations are flagged, and originals remain untouched.

Free software and redistribution

LUFScale is free software distributed under GNU GPL-3.0-or-later. This licence permits use, study, modification and redistribution under its terms. Source code, notices and third-party licences accompany the distribution. The software is provided without warranty.



1

Add audio files

Drop one or more folders or audio files. Every subfolder is scanned.

2

Destination

Select the folder that will receive processed copies with the same tree.

3

Settings

Choose a preset and the required operation. Every file is processed separately.

4

Start

Click Normalize. Progress, elapsed time and CPU use are displayed live.

Main features

- True normalization, ReplayGain or analysis for MP3, FLAC, WAV, AIFF, M4A, OGG and Opus.
 - Every file is measured and processed separately toward the selected target.
 - Folder structure, metadata and artwork are preserved when FFmpeg can copy them.
 - Originals are never moved or modified.
 - Auto parallelism, analysis cache and resume after interruption.
 - Quality control, CSV report, progress, CPU and Before/After LUFS histories.
 - Interface and PDF guides in 12 languages.
- Two macOS distributions are available: Apple Silicon (arm64) and Intel (x86_64). Use the one matching your Mac processor.
- The macOS application already contains Python, PySide6/Qt, FFmpeg, codecs, guides and licences: end users install no additional tool.

Settings - Audio / Options

Main features

Format scope

LUFSscale supports MP3, FLAC, WAV, AIFF, M4A, OGG and Opus. Loudness measurement, the LUFS target, the True Peak ceiling in Normalize mode, processing settings and quality control apply to supported formats, not only MP3. The codec and some corrections depend on the output format. MP3 examples illustrate a particular case; they do not restrict general features to MP3.

Preset

Applies a consistent loudness target, maximum true peak and audio quality at once. Any manual change selects Custom.

File-by-file processing

Every file receives its own measurement and gain to approach the selected LUFS target under the True Peak limit.

ITU-R BS.1770

LUFSscale automatically uses the full historical measurement, the only method validated on the reference corpus.

Operation

Normalize physically changes audio toward target. ReplayGain copies the stream and adds tags. Analyze-only measures each source without creating audio files.

Volume

This setting is a shortcut to the loudness target; it does not change the Mac playback volume.

- Soft: -18 LUFS - calmer level, more dynamic headroom and less chance of engaging the limiter.
- Normal: -16 LUFS - balanced compromise and a useful starting point for a personal library.
- Loud: -14 LUFS - more forward playback, close to Spotify's Normal playback target, but more likely to require limiting.
- Custom - lets you enter another LUFS target directly.

These are practical choices, not a universal standard.

Loudness target

The loudness target is the intended integrated loudness across the whole track, expressed in LUFS. A less negative value produces a louder file: -14 LUFS is louder than -16 LUFS. A 2 LU difference is approximately a 2 dB level difference before any peak limiting.

Guidance: -18 LUFS for a calmer and more dynamic result; -16 LUFS for a general balance; -14 LUFS for a louder streaming-style result. Platforms may then apply their own playback normalization.

This target does not by itself flatten the dynamics inside the track. If the maximum true peak prevents the target from being reached without clipping, the result may remain slightly lower.

Maximum true peak

Maximum true peak is a ceiling, not a level to reach. It limits the highest reconstructed waveform peaks in dBTP, including peaks between samples, to reduce clipping after encoding or transcoding.

- -1.0 dBTP - common delivery ceiling with the highest output peak level.
- -1.5 dBTP - the default and a cautious compromise for MP3 files.
- -2.0 dBTP - extra headroom, useful when files may be encoded again or when using a high loudness target.
- 0 dBTP - no headroom and not recommended for MP3.

A more negative value is safer, but may prevent highly dynamic tracks from reaching the LUFS target exactly.

The True Peak ceiling applies to all formats in Normalize mode. MP3 examples also help explain precautions for other lossy formats such as M4A/AAC, OGG/Vorbis and Opus. The additional internal 1 dB margin described in the appendix is specific to dynamic processing of MP3 output.

Settings - Audio / Options

Main features

Audio quality

Controls the quality/size trade-off for compressed formats. Lower numbers use higher quality and bitrate. It changes neither the LUFS target nor maximum true peak.

- 0 - maximum quality, recommended to preserve detail.
- 1 to 2 - very high quality.
- 3 to 4 - a good quality/size compromise.
- 5 to 9 - smaller files with progressively more loss.

FLAC remains lossless at every value. WAV and AIFF ignore this setting and preserve source-compatible rate and PCM depth. For MP3, M4A, OGG and Opus, a low value may request a higher bitrate than the original, making the output larger. A higher value generally reduces size, without guaranteeing the same byte count because these encoders often use variable bitrate. Re-encoding a lossy source cannot restore discarded information.

Parallel processes

Determines how many files can be processed at the same time.

- Auto starts with at most 4 jobs. When CPU measurement is available, it checks once per second: it adds one job below 70% usage and removes one above 92%.
- Auto never exceeds the detected logical CPU count, with an absolute ceiling of 16 jobs.
- If CPU measurement is unavailable, Auto uses that detected ceiling directly without dynamic adjustment.
- A numeric value fixes the maximum number of simultaneous jobs; it is not a CPU-usage target.

More jobs can speed up a large batch, but increase CPU load, heat and disk activity. Press - until Auto is shown.

Overwrite existing files

Allows an audio file already present in the destination to be replaced. Source files are never overwritten.

Resume after an interruption

Files already completed with the same settings are recognized and not processed again.

Automatic quality control

Quality control covers all supported formats. M4A, OGG and Opus are also checked, but do not use the corrective loops specific to dynamic MP3 or FLAC/WAV/AIFF. In ReplayGain mode, checks concern signal preservation and tags; they do not guarantee physically normalized audio.

With quality control, a dynamic MP3 can be recalculated up to three times. FLAC/WAV/AIFF may receive up to two corrections. Every attempt is remeasured and the best output is retained.

Without QC, no compliance is declared and no corrective retry runs. A normalized output is still remeasured for the log and CSV. Analyze-only creates no output and therefore performs no output QC.

Create a CSV report

Creates a detailed CSV report in the destination with measurements, durations and warnings. No diagnostic JSON file is added.

Start automatically after a drop or paste

Automatically starts processing after sources are added by drag-and-drop or paste, when a destination has already been selected.

Do not re-encode files that already comply

Enabled by default. After analysis, a file within ± 0.10 LU of target and below the True Peak limit is copied unchanged without re-encoding.

What happens during processing

LUFScale automatically uses the full historical measurement, the only method validated on the reference corpus.



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Total Mac CPU usage, updated every second during processing.

Analyze

Analyze-only fully remeasures every source with FFmpeg on each run. The Before graph and progress advance file by file. No audio file or output quality control is created.

Resume after an interruption

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Loudness change

Each file adds a point on the right and moves history left. Before always shows the measured source. With Normalize, After shows the actually remeasured output. With ReplayGain, the dashed second graph shows a playback estimate: source loudness plus the gain stored in the Track tag. The $\approx$ sign and Compatible player note make clear that this is not a physical measurement of the delivered file. An incompatible player keeps the original level; a compatible player may also alter the result through preamp or clipping prevention. Both graphs keep the same fixed ± 6 LU scale around target. Analyze-only has no After output.

Analysis only

In Analyze-only mode, the Before graph, log, and progress bar advance file by file as each measurement finishes; After stays still.

What happens during processing

LUFScale automatically uses the full historical measurement, the only method validated on the reference corpus.

Processing log

Each line concerns a file or a general processing step.

- A successful line starts directly with the filename: SUCCESS is no longer repeated.
- COMPLIANT, RESUMED, SKIPPED, CANCELLED and ERROR remain when they add useful information.
- Levels show input -> remeasured output, followed by any quality-control result.
- Warnings and Errors open separate lists containing filename, path and details. They are available while paused or after processing, and each list can be saved.

Colours: green = success; orange = warning; red = unfinished file; blue-purple = resumed; grey = information, skipped item or cancellation.

QC WARNING - loudness means the remeasured output differs from its expected value by more than ± 0.60 LU. A more negative value is quieter; a less negative value is louder. The deviation is the absolute difference: -14.69 instead of -14.00 is 0.69 LU. The file is still created; this is not a conversion failure. No action is required if it sounds acceptable. For a strict target, inspect the details and CSV, then check the target and True Peak ceiling before retrying. The message alone cannot prove whether the ceiling, encoder or a correction limit caused the deviation.

QC WARNING - peak means the remeasured true peak exceeds the selected limit by more than 0.25 dB. The file is still created. For a persistent warning, choose a lower LUFs target or a safer peak ceiling, for example -2.0 dBTP, then process the file again.

Cumulative times add the work of all parallel jobs. Total time is the actual elapsed duration.

In ReplayGain, the log shows unchanged physical loudness, the gain written to metadata and the configured target. With quality control enabled, 'audio unchanged and tags verified' means loudness and peak were compared with the source and tags were read back; it does not mean that the file physically measures at the target.

Estimated total time: calculating...

Estimated total time: 12 min - finishing around 14:30. "12 min" is the estimated total duration and "14:30" the expected finish time. If completion passes midnight, the day count is automatically added before the time, for example "2 d. 14:30".

Processing-log colour key

SUCCESS	Processing completed with no detected anomaly.
COMPLIANT	Unchanged audio copy: the source already met the target and peak limit.
WARNING	The output exists, but one measurement is outside the expected tolerance.
CANCELLED	Processing was intentionally stopped; this is not an error.
ERROR	The affected file could not be completed.

Recommended settings and reference points

Preset	Loudness target	Maximum true peak	Audio quality
Music library - recommended	-16 LUFs	-1.5 dBTP	0
Louder streaming	-14 LUFs	-1 dBTP	0
Dynamic music	-18 LUFs	-2 dBTP	0

These presets are practical application choices, not universal standards.

Three useful checks

No processing

Check that at least one compatible audio source and a different destination are selected.

FFmpeg not found

Run the application builder again or install FFmpeg, then restart.

Peak warnings

Choose a more negative true-peak ceiling or a lower loudness target, then process the file again.

Official technical references

FFmpeg filters - loudnorm / ebur128:
<https://ffmpeg.org/ffmpeg-filters.html>

FFmpeg codecs - libmp3lame and audio encoders:
<https://ffmpeg.org/ffmpeg-codecs.html>

ITU-R BS.1770-5 (11/2023), programme loudness and true peak:
<https://www.itu.int/rec/R-REC-BS.1770-5-202311-I/en>

EBU R 128 v5.0 (21/11/2023):
<https://tech.ebu.ch/publications/r128>

Original ReplayGain specification:
https://wiki.hydrogenaudio.org/index.php?title=Original_ReplayGain_specification

Technical appendix

Source files are never moved or modified.

Operation

Normalize processes every file separately, targets LUFS under the True Peak limit, and remeasures the output. ReplayGain does not change samples. Analyze-only produces measurements and an optional CSV report, but no audio file.

With quality control enabled, ReplayGain remeasures the delivered file to confirm unchanged physical loudness and peak, then checks the Track tags. Success confirms preserved audio and present tags, not physical target loudness.

Without QC, no compliance is declared and no corrective retry runs. A normalized output is still remeasured for the log and CSV. Analyze-only creates no output and therefore performs no output QC.

ITU-R BS.1770

FFmpeg loudnorm measures integrated loudness I, LRA, threshold and true peak TP according to ITU-R BS.1770.

Reference analysis

LUFSscale automatically uses the full historical measurement, the only method validated on the reference corpus.

File-by-file processing

Every file receives its own measurement and gain to approach the selected LUFS target under the True Peak limit.

MP3 only: dynamic processing

An MP3 not eligible for the linear path receives an internal true-peak target 1.00 dB lower, never below -9 dBTP. The requested target remains the user and QC reference.

Audio quality

MP3 uses LAME, M4A FFmpeg AAC, OGG Vorbis and Opus libopus. FLAC stays lossless; WAV/AIFF remain source-compatible PCM.

Do not re-encode files that already comply

Enabled by default. After analysis, a file within ± 0.10 LU of target and below the True Peak limit is copied unchanged without re-encoding.

Automatic quality control

Quality control covers all supported formats. M4A, OGG and Opus are also checked, but do not use the corrective loops specific to dynamic MP3 or FLAC/WAV/AIFF. In ReplayGain mode, checks concern signal preservation and tags; they do not guarantee physically normalized audio.

With quality control, a dynamic MP3 can be recalculated up to three times. FLAC/WAV/AIFF may receive up to two corrections. Every attempt is remeasured and the best output is retained.

Parallel processes

FFmpeg tasks run concurrently. Cumulative times add task work, not elapsed time. Cache and resume verify source, settings and output; Analyze-only deliberately ignores the cache.

Build for Apple Silicon Mac and Intel Mac

Paths are relative to the extracted source-package root:

Apple Silicon Mac (arm64) - On macOS 12 or later, run:

```
./Create_Community_Distribution_macOS.command
```

Created files:

```
dist/LUFSscale.app
dist/LUFSscale-2.2.3-macOS-arm64-community.zip
dist/LUFSscale-2.2.3-macOS-arm64-community.zip.sha256
```

Intel Mac (x86_64) - On an Intel Mac, run:

```
./Create_Community_Distribution_macOS.command
```

If macOS installs the Xcode tools, keep Terminal open: the builder resumes automatically.

Created files:

```
dist/LUFSscale.app
dist/LUFSscale-2.2.3-macOS-x86_64-community.zip
dist/LUFSscale-2.2.3-macOS-x86_64-community.zip.sha256
```

.app (self-contained application for local testing or regular use)

.zip (archive to share or publish, with the application, sources, licences and notices)

.sha256 (SHA-256 fingerprint of the .zip file)

Technical appendix - calculations and formulas

L = LUFS, TP = dBTP, G = dB

ITU-R BS.1770

$$z_i = (1/T_g) \int x_i^2(t) dt$$
$$L_K = -0.691 + 10 \log_{10}(\sum G_i z_i)$$

K-weighted channel energies are combined to calculate integrated loudness.

ITU-R BS.1770 - Gating

$$T_g = 400 \text{ ms}; \text{overlap} = 75 \%$$
$$\Gamma_a = -70 \text{ LKFS}; \Gamma_r = \Gamma_{\text{loud}} - 10 \text{ LU}$$

Absolute and relative gates select the blocks retained by the integrated measurement.

File-by-file processing

$$G_{\text{file}} = L_{\text{target}} - L_{\text{input}}$$

Every file receives its own measurement and gain to approach the selected LUFS target under the True Peak limit.

Estimate without output

$$G_{\text{est}} = L_{\text{target}} - L_{\text{input}}$$
$$TP_{\text{est}} = \min(TP_{\text{input}} + G_{\text{est}}, TP_{\text{limit}})$$

Analyze-only may calculate an estimate from the source measurement, but creates no audio file and no output quality control.

Do not re-encode files that already comply

$$|L_{\text{in}} - T| \leq 0.10 \text{ LU}$$
$$TP_{\text{in}} \leq TP_{\text{max}}$$

Enabled by default. After analysis, a file within ± 0.10 LU of target and below the True Peak limit is copied unchanged without re-encoding.

Loudness change

$$\Delta = L - T$$
$$R_{\text{display}} = [-6, +6] \text{ LU}; \tau_{\text{QC}} = \pm 0.60 \text{ LU}$$

Before shows physical loudness. In ReplayGain, the second graph estimates compatible-player playback from the stored gain; audio remains unchanged.

Automatic quality control

$$|L_{\text{out}} - L_{\text{expected}}| > 0.60 \text{ LU}$$
$$TP_{\text{out}} > TP_{\text{limit}} + 0.25 \text{ dB}$$

Quality control covers all supported formats. M4A, OGG and Opus are also checked, but do not use the corrective loops specific to dynamic MP3 or FLAC/WAV/AIFF. In ReplayGain mode, checks concern signal preservation and tags; they do not guarantee physically normalized audio.

With quality control, a dynamic MP3 can be recalculated up to three times. FLAC/WAV/AIFF may receive up to two corrections. Every attempt is remeasured and the best output is retained.

Without QC, no compliance is declared and no corrective retry runs. A normalized output is still remeasured for the log and CSV. Analyze-only creates no output and therefore performs no output QC.

ReplayGain - no audio re-encoding

$$G_{\text{tag}} = L_{\text{target}} - L_{\text{input}}$$
$$P_{\text{peak}} = 10^{TP/20}$$

ReplayGain calculates a suggested gain and writes REPLAYGAIN_TRACK_GAIN/PEAK tags. The audio stream is copied without re-encoding (-c:a copy), so samples do not change. Only a compatible player applies the tags. The file therefore does not physically reach target LUFS and its actual True Peak is not limited.

ReplayGain is useful for reversible, no-re-encode playback leveling in a library used with a compatible player. To deliver a file that physically measures at the target in every player, use Normalize.

Technical appendix - Automatic quality control

L_t = target LUFS; TP_{req} = requested dBTP; TP_g = 0.05 dB guard

Format scope

Measurements and quality thresholds cover supported formats. Internal-target and feedback formulas marked “MP3 only” describe the dynamic MP3 loop. Correction limits separately identify FLAC/WAV/AIFF; they must not be generalized to M4A, OGG or Opus.

Linear loudnorm eligibility

$$\begin{aligned}G &= L_t - I \\TP_{pred} &= TP_{in} + G \\m_{TP} &= TP_{req} - TP_{pred} \geq 0.05 \\m_{LRA} &= LRA_t - LRA_{in} \geq 0.05\end{aligned}$$

Linear mode is used only when the predicted true-peak and LRA margins are each at least 0.05.

MP3 - Initial internal true peak

$$\begin{aligned}TP_{safe} &= TP_{req} - 1.00 \text{ dB} \\TP_{int,0} &= \max(-9 \text{ dBTP}, TP_{safe})\end{aligned}$$

For a dynamic MP3, the internal target starts 1.00 dB below the requested limit and is clamped at -9 dBTP.

Strict compliance

$$\begin{aligned}|L_{out} - L_t| &\leq 0.50 \text{ LU} \\TP_{out} &\leq TP_{req}\end{aligned}$$

Output is strict when loudness stays within ± 0.50 LU of target and true peak does not exceed the requested limit.

MP3 - Peak feedback

$$\begin{aligned}TP_g &= 0.05 \text{ dB} \\e_{TP} &= TP_{out} - (TP_{req} - TP_g) \\TP_{int,n+1} &= TP_{int,n} - e_{TP}\end{aligned}$$

If true peak exceeds the limit, the next internal target is lowered by the excess plus the 0.05 dB guard.

MP3 - Low-loudness feedback

$$\begin{aligned}r &= \min[(L_t - 0.45) - L_{out}, \\&\quad (TP_{req} - 0.05) - TP_{out}] \\TP_{int,n+1} &= TP_{int,n} + r\end{aligned}$$

If loudness is too low, the internal target rises toward $L_t - 0.45$ LU, capped by peak headroom; no step below 0.01 dB.

Best attempt

$$\begin{aligned}p &= \max(0, TP_{out} - TP_{req}) \\e_L &= \max(0, |L_{out} - L_t| - 0.50) \\d_L &= |L_{out} - L_t| \\score &= (p > 0, e_L > 0, p, e_L, d_L)\end{aligned}$$

Among available attempts, the score prioritizes the peak ceiling, then loudness tolerance, then absolute distance from the target. This selection applies to dynamic MP3 and FLAC/WAV/AIFF corrections.

Correction limits

$$N_{MP3} \leq 3; N_{lossless} \leq 2$$

A dynamic MP3 allows at most three retries; FLAC, WAV and AIFF allow two corrections. The loop may stop earlier.

QC warning vs retry target

$$\begin{aligned}\Delta L &= L_{out} - L_t \\QC_{alert} &\Leftrightarrow |\Delta L| > 0.60 \vee TP_{out} > TP_{req} + 0.25 \\strict_{OK} &\Leftrightarrow |\Delta L| \leq 0.50 \wedge TP_{out} \leq TP_{req} \\retry_{MP3} &\Leftrightarrow \neg strict_{OK} \wedge |TP_{int,n+1} - TP_{int,n}| \geq 0.01\end{aligned}$$

QC warns at ± 0.60 LU and $+0.25$ dB; a retry targets the strict limits with a step of at least 0.01 dB.