

Your meter reads samples; the listener hears the wave between them.

How dBTP is measured (ITU-R BS.1770-5, Annex 2)

1. Attenuate -12.04 dB (2-bit shift) - headroom for the math (skip in float).
2. Oversample 4x: 48 kHz -> 192 kHz, 48-tap FIR interpolation.
3. Low-pass filter - exclude spurious frequencies.
4. $|x|$, then $20 \cdot \log_{10}$ and add back 12.04 dB -> report in dBTP.

Delivery ceilings (max true peak)

DESTINATION	CEILING	NOTE
EBU R128 broadcast	-1 dBTP	production, +/-0.3 dB tol.
AES streaming (codec input)	-1 dBTP	lower at low bitrate
Spotify	-1 dBTP	-2 if above -14 LUFS
Apple Music	-1 dBTP	Apple Digital Masters
YouTube	-1 dBTP	common practice
Amazon Music	-2 dBTP	stricter ceiling
Netflix deliverable	-2 dBTP	-27 LKFS dialogue-gated
Low-bitrate / HE-AAC	-2 dBTP+	overshoot grows (AES TD1008)

The math

Sample peak (dBFS) reads stored samples; true peak (dBTP) reads the rebuilt wave between them.

Bright material hides 0.5-1.5 dB of true peak above sample peak.

Worked: master +0.6 dBTP + AAC 128k overshoot +0.4 dB = +1.0 dBTP at decoder -> CLIPS.

Fix: limit to -1.0 dBTP first; -1.0 + 0.4 = -0.6 dBTP at decoder -> SAFE.

Test before delivery (FFmpeg)

Measure true peak (input_tp) before delivery:

```
ffmpeg -i master.wav -af loudnorm=I=-14:TP=-1:print_format=json -f null -
```

Or measure with the dedicated meter:

```
ffmpeg -i master.wav -af eburl28=peak=true -f null -
```

Remember

- Trust dBTP, not dBFS - a sample-peak meter under-reads by up to ~1 dB.
- Use a TRUE-peak limiter (oversamples internally), not a sample-peak limiter.
- Limit true peak AFTER any downsampling - SRC adds its own overshoot.
- Deliver at -1 dBTP; use -2 dBTP for low bitrate or strict platforms.
- Lower is always safe: normalization sets loudness, not the peak ceiling.