Technical Council

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Recommendation for Loudness of Audio Streaming and Network File Playback

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Study Group on Streaming Loudness

Recommendation for Loudness of Audio Streaming and Network File Playback

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1. Introduction

Streaming is rapidly becoming a major vehicle for media delivery. As a result, the ways that audio is recorded, mixed, post-produced and delivered have been radically affected. Audio quality has begun to suffer as a result of loudness differences among and within streams, as well as some very high loudness targets, resulting in distortion. Thus, streaming requires a leveling solution based on loudness, with an appropriate loudness target.

Loudness is the listener's perception of "audio volume". An **audio stream** is a continuous transmission to listeners over a network (typically the Internet) that consists of one or more programs presented sequentially. It's analogous to a "radio station" in over-the-air broadcasting. A **streamer** is a content provider offering a streaming service to customers. **Normalization** is a method of regulating the loudness to be more consistent for the listener. **Network file playback** is on-demand download of complete programs from the network, such as podcasts. In this document, the terms **stream** and **streaming** take into account network file playback.

These recommendations primarily are intended for "radio-like" mono and stereo streams as opposed to very dynamic stereo and surround sound streams with content such as movies or video specials. See the Appendix for notes on such highly dynamic streams.

2. Primary Goals

The intent of this document is to provide recommendations for loudness normalization of streaming and network file playback content. There are many good reasons to set some basic loudness requirements:

- Improve the audience experience.
- Provide reasonable consistency across different online streams from different sources.
- Provide reasonable consistency within a specific online stream for its different programs.
- Provide a consistent real-time production target for stream loudness.
- Obtain a loudness that is well-suited for mobile listening.
- Avoid loudness jumps when external material (such as advertisements) is inserted into stream content.
- Prevent excessive peak limiting or other processing from degrading perceived audio quality.
- Avoid a loudness war among streamers.

3. Recommendations

- It is recommended that the Target Loudness of the stream not exceed -16 LUFS: to avoid excessive peak limiting, and allow a higher dynamic range in a program stream.[1]
- It is recommended that the Target Loudness of a stream not be lower than -20 LUFS: to improve the audibility of streams on mobile devices.
- It is recommended that short-form programming (60 seconds or less) be adjusted by constraining the Maximum Short-term Loudness to be no more than 5 LU higher than the Target Loudness: This ensures that commercials and similar short-form content are consistent with the stream loudness.
- It is recommended that the maximum peak level not exceed −1.0 dB TP: to prevent clipping when using lossy encoders.

1 **Target Loudness** is the intended or desired loudness of a stream, in **LUFS,** "loudness units relative to full scale". See Section 4 below. In ATSC A/85, the unit of measure "LKFS" is used instead of "LUFS" for absolute loudness. LKFS and LUFS are identical. See definitions in the Appendix.

4. Choosing and Measuring Loudness

A. Technical Definitions

Loudness is measured relative to digital full scale, not SPL. Absolute loudness is measured in **LUFS**, “loudness units relative to full scale”. Relative loudness is measured in **LU**, “loudness units”. ITU-R BS.1770-3 defines **Integrated Loudness**, a measurement of the total amount of audio energy between two points in time divided by the duration of the measurement. The measurement is frequency-weighted to approximate the sensitivity of the ear to different frequencies, and is level-weighted to emphasize the parts of the program contributing most to the sensation of loudness. **Program Loudness** is a measurement of one program from top to tail. EBU - TECH 3341 and ITU-R BS.1771 distinguish the measurement windows of **Integrated Loudness** and **Short-term Loudness**. EBU R 128 defines **Maximum Short-term Loudness**. See links and definitions in the Appendix.

B. Target Loudness

Target Loudness is the intended Integrated Loudness for a stream. It is impossible to measure and verify the actual Integrated Loudness of an infinite length stream, so loudness must be verified by using an integration time long enough (typically 24 hours) to characterize the stream as a whole.

Each stream has a single target loudness, with the possible exception of multi-formatted streams, e.g. talk shows in the morning and music programs at night. In such cases, it is recommended that the maximum difference between any targets be as small as possible, and not less than −20 nor greater than −16 LUFS. It is recommended that the Integrated Loudness of each program match its target as closely as practical. For live streams, a wider tolerance may be necessary, while prerecorded files may be matched within ±0.5 LU, for example.

Users may choose a Target Loudness that is lower than the −16 LUFS maximum, e.g., −18 LUFS, to better suit the dynamic characteristics of the program. A lower Target Loudness helps improve sound quality by permitting the programs to have a higher peak-to-loudness ratio without excessive peak limiting.

The −20 LUFS lower limit has been chosen as the lowest current practical value for streaming, as some current mobile devices have insufficient gain to allow the common production targets of −23 or −24 LUFS to be heard at a satisfying loudness even if the volume control is turned all the way up.

C. Short-Form Content (e.g. Commercials)

It is recommended that the Program Loudness (PL) of program breaks lasting 60 seconds or less not exceed the Target Loudness of the stream. Additionally, it is recommended that the Maximum Short-term Loudness of these short program breaks not exceed 5 LU above the Target Loudness of the stream.

For example, if a stream is targeted to −20 LUFS Integrated Loudness, the Maximum Short-term Loudness of a commercial (or other short length) segment would not exceed −15 LUFS. Moreover, the PL of the commercial would not exceed −20 LUFS. If the PL of the commercial is −20 LUFS but its Maximum Short-term Loudness is, for example, −13 LUFS, the commercial would have to be attenuated by 2 LU, which would reduce its PL to −22 LUFS. In this case, the commercial provider may wish to remix the commercial to reduce the difference between its Maximum Short-term Loudness and its PL to 5 LU or less so that this attenuation is not required.

D. Choosing the Optimum Target

Every streamer has to choose a target. When choosing a Target Loudness, streamers must take several audio characteristics into account, such as providing adequate acoustic output from player headphones. Higher target values (approaching −16 LUFS) require greater dynamic control, which can result in reduced sound quality. The minimum Target Loudness of −20 LUFS is believed to support sufficient acoustic output with most consumer playback equipment, though not always with portable media players obeying European regulation EN 50332 (see Appendix). Another factor is the variability of Integrated Loudness over time in a live stream, which may require lowering Target Loudness to avoid inadvertently exceeding −16 LUFS.

Some will stream spoken word material that sounds more natural with a less-processed dynamic range. Others will stream “fine arts” material that sounds more natural when streamed at a lower target with its original dynamic range. Others may target mobile devices or high-fidelity playback systems. Regardless, if all streamers follow this document’s recommendations, there will be no more than a 4 LU spread among all participants.

5. Peak Control

Peaks generally do not affect a loudness measurement, though they do affect perceived signal quality. A recording with high peak to loudness ratio (PLR) is often perceived as clearer and less fatiguing than one that has been excessively peak-limited. In this discussion, “dB TP” refers to peak levels measured using a “true peak” meter according to ITU-R BS.1770-3, Annex 2.

If the streamer chooses a lower target loudness than the recommended –16 LUFS (e.g. −18 LUFS), peak overloads are rarely an issue. Peak limiting is not normally required unless the incoming audio level must be increased to meet target loudness. Audio that has been attenuated to achieve the target loudness will have its peak level decreased by the same amount. Highly processed audio where original peak levels exceed 0 dB TP will not normally overload as the loudness would have to be greatly reduced to meet the target loudness. Incoming material that has been gently processed or is unprocessed will rarely exceed 0 dB TP.

However, peak level can increase after lossy coding, so we recommend using a safety limiter with a threshold of −1.0 dB TP prior to encoding. See Appendix for technical suggestions on dealing with peak limiting and codecs.

6. Continuous Programs Presented in Segments

Some streamers will play multi-part programs, e.g. symphonies, which are presented in movements. It is desirable to normalize the whole symphony to the target, not the individual movements**,** or the quiet movements will be played too loudly. The best algorithm is to find the loudest movement, determine its normalization, and apply that gain to each of the other movements of the symphony. If a single pre-recorded program is presented as a whole with the symphony integrated in the program, normalizing the program is easy to accomplish, but if the symphony is presented as a playlist in segments an automatic normalizer may mistakenly override the need for whole program normalization. Exercise caution: normalize programs ahead of time keeping these issues in mind.

7. Genre and Format Issues

Numerous independent tests of the ITU-R BS.1770 algorithm against human listeners show that it's among the best metrics for normalizing a wide variety of broadcast programs and music tracks. However, it is possible to find examples where average listeners prefer normalization by an experienced engineer with fresh ears. Furthermore, normalizing each generic element (such as speech and music) in a given program to the same loudness can yield inartistic balances that are inconsistent with balances made by experienced human operators. Hence, the balances between elements in a program should be chosen by program creators based on their artistic goals.[2]

Within a given program, the largest perceived difference to be noted is speech versus music. Speech normalized to the same Integrated Loudness as a music stream inevitably sounds too loud. It is recommended to normalize speech (dialog) segments within other segments 2 to 4 LU (or more) below the loudness of the other segments.

Ideally, listeners should not have to adjust their volume controls when switching between streams with similar formats. Although two streams of different formats having identical measured Integrated Loudness may not sound equally loud, the difference is not jolting to the listener, so it is thus usually tolerated and accepted.

Multi-formatted streams: This is a special case where a streamer may produce fine arts content with a high PLR at a target of, for example, −20 LUFS in the evening and in the morning, talk shows with a target of −16 LUFS. This is not a problem because listeners will only have to adjust their volume controls once when the format change occurs. The Integrated Loudness of the whole stream will still lie within the accepted window of –16 LUFS through –20 LUFS.

8. Live Streams

Live streams obviously cannot be normalized in advance. The most transparent way to deal with live streams is to set gains in advance on a moderately-loud segment so the Short-term Loudness is approximately −20 LUFS and adjust as the stream progresses. Include a protection limiter at −1 dB TP prior to the encoder to prevent accidental overloads.

9. Appendix

A. Technical Notes

1. Portable Media Players (PMPs) and Hearing Loss

If PMPs are played too loudly for too long, they will cause hearing loss. The Scientific Committee on Emerging and Newly Identified Health Risks estimates that between 2.5 and 10 million people in the EU are at risk of developing early hearing loss as a result of listening to PMPs (see Bibliography: "Prevention of Hearing Loss…"). Europe is the first region to implement regulations to protect the hearing of PMP users. This has successfully brought down the maximum SPL from PMPs sold across Europe, but with the adverse effect that music and programs not produced like modern pop cannot be played loudly enough to be heard under demanding listening conditions.

The current regulation specifies a test signal rather than a method of determining the range and duration of loudness of actual program material. In terms of its ability to predict hearing loss, this test signal is inapplicable to program material having a different character, loudness and duration. The regulation also does not reflect the actual loudness and dosage heard by the listener. Furthermore, the regulation specifies an SPL of the test signal, resulting in a gain limit for players that leads producers to over compress real world material in order to be heard. Inadvertently, the very regulations intended to prevent hearing loss have set off a loudness race to produce overly-compressed and fatiguing-sounding music. This development cannot be good for our hearing or our music heritage.

CENELEC TC108X/WG3 is working to improve the standard, EN 50332, by adding sound dose estimates that take actual audio into account. However, until the revised regulation comes into effect, European PMPs may have insufficient gain to allow satisfactory playing of material with high PLR and LRA (Loudness Range). Furthermore, until the regulation is revised, loudness targets below −20 LUFS and possibly even below −16 LUFS may not be compatible with current-generation European PMPs.

2. Portable Player Performance

We encourage providers of player devices to remove the loudness, gain and headroom limitations in future versions of software and hardware. This would eventually permit the EBU R 128 standard of −23 LUFS to be suitable for streaming, even in moderately noisy environments. This would be a big advantage because virtually all the program content can be played without sonic alteration, without

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2 For further information, see BS.1770-3 standard and other references cited in the appendix.

any processing other than loudness normalization. Lowering all targets to the broadcast standard of −23 LUFS would also provide consistency between broadcast material and streaming material without a need for level translation, and would cause fewer headaches for broadcasters who also provide internet streams.

As portable players get rid of the current gain and headroom limitations, all targets could be lowered together so that the difference between lowest and loudest streams would continue to be no greater than 4 LU.

3. Players with Limiters

Many mobile devices have built-in peak limiters of uncertain subjective quality, and these act to protect the power amplifiers in battery-operated mobile devices from clipping. Hence, a stream with a high peak-to-average ratio (such as a −23 LUFS classical music stream) may activate the peak limiters in player devices, with unpredictable subjective results.

Similarly, some common operating systems (like Microsoft Windows, Vista or higher) have built-in peak limiters that produce gain reduction on codec overshoots that would otherwise cause clipping in a float-to-fixed-point conversion following the decoder. Consequently, failure to provide sufficient headroom for codec overshoots on the transmission side can produce as much as 3 dB of gain reduction in the decode-side peak limiter.

4. Peak Level Control and Lossy Codecs

Safety limiting should take into account the peak signal level appearing at the decoder's output, which may be higher than the peak level at the encoder's input. High rate (e.g., 256 kbps) coders may work satisfactorily with as little as −0.5 dB TP for the limiting threshold. However, lower bit rate coders tend to overshoot peaks even more, so the limiting threshold may need to be reduced below −1.0 dB TP. A maximum of 1 dB of limiting prior to final encoding is recommended as a starting point. Do this only if frequently-occurring peaks cause audible distortion or artifacts at the decoder. Use the ear as the final judge, since limiting more than about 1 dB may produce more audible artifacts than simply letting the program clip on occasional transients. Even better, don't raise the gain enough to cause potential clipping. Given this document's recommended allowable range of loudness (−20 to −16 LUFS), most types of content can be accommodated by simply using a lower target level instead of applying egregious peak limiting. This could easily produce a cleaner "station" sound and be very desirable. In other words, instead of deciding on 2 dB of peak limiting, a combination of −1 dB TP peak limiter threshold with an overall attenuation of 1 dB from the previously-chosen target may produce a more desirable result.

True peak measurement calls for the audio signal to be oversampled by at least 4x so that it approximates the true peak level following D/A conversion or sample rate conversion. True peak meters typically have an error less than 0.6 dB, assuming an ideal D/A converter with a linear-phase reconstruction filter.

Lowpass filtering can add overshoots, so lowpass filtering (if used) should occur before the peak limiter. If the signal path after the peak limiter has a high pass characteristic (as do most analog paths), the −3 dB frequency must be below 0.15 Hz to prevent the path from introducing more than 0.1 dB of overshoot.

5. Metadata, Interoperability with R 128 Content

Until an agreed loudness metadata scheme is firmly established and widespread, it is challenging to assume that the listener's receiver will perform the necessary gain changes for different programs and different streams. To help speed up the transition to a metadata-based system, it is recommended to insert loudness metadata at the stream encoder indicating the loudness of the content for player-side loudness control. Without using metadata, files need to be stored pre-normalized, or a real-time normalization scheme at the content provider's side can be established using a playlist method that reads the metadata in each file to be streamed. Nevertheless, such playlists may not respect the integrity of movements of a symphony, or the relative levels of songs within an album, so proceed with caution.

Streams having an Integrated Loudness according to the recommendations of this document will be available on player devices that may also play program material having the R 128-recommended Integrated Loudness of −23 LUFS. This can cause loudness jumps of up to 7 LU, which is outside the comfort zone of most listeners. The best solution for this situation is to include Target Loudness metadata in the program stream. This allows a metadata-aware player to adjust its gain automatically to prevent loudness shifts between −16 LUFS and −23 LUFS streams. Such devices typically normalize all program material to an internal reference below -23 LUFS.

With metadata more and more widespread, it becomes viable not to use a fixed target loudness at all! Instead, the gain control ("volume slider") may adjust the target loudness ("moving target"). The lower the slider, the lower the target loudness and the higher the number of the programs that can be loudness normalized without any limiting. Care must be taken that the maximum position of the gain control is at or close to the upper target limit of −16 LUFS.

6. Normalization Practice

Normalization is the process of adjusting the loudness of a program to conform to the target via an algorithm. Three possible algorithms are:

a. Measure the Integrated Loudness and true peak level of the program. Determine the difference between the target loudness and the program loudness. There is no problem if its level has to be lowered to match the target. If its level has to be raised, raise it until it reaches target level or until the true peak reaches 0 dB TP, whichever occurs first. Thus, the sound quality of all material will be preserved, without introducing excessive peak limiting. However, some material with a high PLR will be streamed lower than the target loudness.

b. Perform paragraph a, but keep raising the level until the program level reaches target, and apply either peak limiting or allow some clipping to handle excessive peaks. The advantage is more consistent loudness in the stream, but this is a potential sonic compromise compared to paragraph a. The best way to retain sound quality and have more consistent loudness is algorithm a. with a lower target.

c. Short-form programs that stand on their own (60 seconds or less, typically commercial breaks): Measure the BS.1770-3 Program Loudness (PL) and the Short-term Loudness (which uses a 3 second measurement window — see EBU - Tech 3341 and ITU-R BS.1771-1). Raise or lower the level until the PL matches the target except if the maximum Short-term Loudness exceeds 5 LU above the target; in that case lower the level until the max Short-term Loudness is no higher than 5 dB above the target. Alternatively, a remix of the program may be delivered by the content provider.

7. Highly dynamic streams (e.g. movies)

Some streamers, usually video streamers, wish to stream highly dynamic content with a very high PLR, e.g. movies, music specials, drama, sports, etc., often with 5.1 surround sound. It is acknowledged that this type of content requires far more headroom and a target loudness of −23 LUFS or lower. It would be futile to stream these programs with their full dynamic range on personal media players or small stereo systems. However, it might be suitable to stream some radio-like content on the high PLR stream. In that case it is recommended to normalize the radio-like content downward to the −23 LUFS target of this stream. Having loudness metadata in each file would help to sort out the complexities.

At the time of this writing, there is no easy solution other than metadata to the coexistence of highly dynamic content and "radio-style streams" on the same stream. However, since highly dynamic content is designed to be played back in a quiet room on a good surround-capable playback system, then conversely, it would not normally be auditioned on a personal media player in a noisy room. So there is little conflict or overlap between these two types of streams, for now. In the future, metadata will help reconcile any conflicts.

B. Links to useful loudness standards

Cenelec EN 50332-2:2013, http://shop.bsigroup.com/ProductDetail/?pid=000000000030281663 Sound system equipment: Headphones and earphones associated with personal music players.

ITU-R BS.1770-3, https://www.itu.int/dms_pubrec/itu-r/rec/bs/R-REC-BS.1770-3-201208-I!!PDF-E.pdf Algorithms to measure audio program loudness and true-peak audio level. ITU-R BS.1770-3 defines a method of measuring the Integrated Loudness over entire program segments, using gating to emphasize the parts of the program that contribute most loudness perception.

EBU R 128, https://tech.ebu.ch/docs/r/r128-2014.pdf Loudness normalization and Permitted Maximum Level of Audio Signals.

ATSC A/85, http://atsc.org/wp-content/uploads/2015/03/Techniques-for-establishing-and-maintaining-audio-loudness.pdf ATSC Recommended Practice: Techniques for Establishing and Maintaining Audio Loudness for Digital Television (A/85:2013).

EBU R 128 s1, https://tech.ebu.ch/docs/r/r128s1.pdf Loudness Parameters for Short-Form Content (Adverts, Promos, etc.).

EBU - Tech 3341, https://tech.ebu.ch/docs/tech/tech3341.pdf Loudness Metering. 'EBU Mode' metering to supplement loudness normalization in accordance with EBU R 128. EBU − TECH 3341 defines three integration time constants derived from the BS.1770-3 algorithm.

EBU - Tech 3342, https://tech.ebu.ch/docs/tech/tech3342.pdf Loudness Range: A measure to supplement loudness normalization in accordance with EBU R 128.

EBU - Tech 3343, https://tech.ebu.ch/docs/tech/tech3343.pdf Practical guidelines for production and implementation in accordance with EBU R 128.

EBU - Tech 3344, https://tech.ebu.ch/docs/tech/tech3344.pdf Practical guidelines for distribution systems in accordance with EBU R 128.

ITU-R BS.1771-1, https://www.itu.int/dms_pubrec/itu-r/rec/bs/R-REC-BS.1771-1-201201-I!!PDF-E.pdf Requirements for loudness and true-peak indicating meters.

C. Bibliography

Difference between the EBU R-128 Meter Recommendation and Human Subjective Loudness Perception, http://www.aes.org/e-lib/browse.cfm?elib=16015 AES Convention Paper 8489, 131st AES Convention, by Begnert, Ekman, and Berg.

Loudness Descriptors to Characterize Programs and Music Tracks, http://www.aes.org/e-lib/browse.cfm?elib=14666 Lund and Skovenborg discuss subjects' loudness range tolerance. Summary: 50% of subjects react to +4/-6 LU systematic changes and 95% of subjects to do so for +6/−8 LU.

Prevention of Hearing Loss From The Use of Personal Music Players, http://www.aes.org/e-lib/browse.cfm?elib=17796 Lund demonstrates that the European PMP regulations have inadvertently set off a loudness war and suggests changes to the regulations that can protect our music heritage as well as our hearing.

Normalized Audio and 0 dBFS+ Exposure, http://www.indexcom.com/tech/0dBFS+/ by Greg Ogonowski. Why True Peak levels exceed 0 dBFS.

Streaming Codec Study Report, Kean and Sheffield discuss quality of codecs used for streaming http://nprlabs.org/sites/nprlabs/files/documents/codec/20130920%20Streaming%20Codec%20Study%20Report.pdf

Study of Audio Loudness Range for Consumers in Various Listening Modes and Ambient Noise Levels, https://drive.google.com/file/d/0B6yqUryPr1iaRFhiSDZzbk5IMGM/view Kean, Johnson and Sheffield, presented at NAB 2015.

D. Definitions

dB FS: decibels relative to full scale measured with a standard digital peak meter. Sometimes abbreviated without the space: **dBFS**

dB TP: decibels relative to full scale measured with a true peak meter. Often abbreviated without the space: **dBTP**

Format: A name, such as "contemporary hit radio" or "news," that describes the type or style of a given long-form program. A stream often has only one format, but some netcasters transmit different formats at different times ("mixed-format" streams).

Formats are often built from elements that are related to each other and presented sequentially, but that have different genres, such as music, studio-quality DJ announcements, and noisy, limited-bandwidth speech from a helicopter-based traffic reporter. This document treats commercial inserts as separate short-form program breaks having loudness constraints different from those of the main program.

Genre: A name, such as "full-bandwidth speech," "telephone-grade speech," "popular music," "classical music," etc., characterizing a program element that has a homogenous texture and style.

Headroom: The ratio of 0 dBFS to the integrated program level of a segment.

Integrated Loudness: The electrically measured average loudness between two points in time. Integrated Loudness according to the international standard ITU-R BS.1770-3 uses a gated algorithm. If Integrated Loudness is measured over the entire length of a program the result is called **Program Loudness** (**PL**). To determine the Integrated Loudness of a continuous stream it is necessary to choose an integration time window that is sufficiently long (typically 24 hours).

LKFS: Loudness, K-weighted, with reference to digital Full Scale. ATSC preferred term for the absolute loudness value. K-weighting is explained in ITU-R BS.1770-3. (See LUFS).

Loudness Normalization: A working practice in which a specific loudness level defines the reference point (also known as the "target"), and the loudness of individual signals is assessed so that they can then be adjusted to match the target loudness on replay. This approach allows a margin for peaks and thus encourages dynamic variations if a reasonably low target is chosen. (See also Normalization Practice, Peak Normalization).

LRA: Loudness range. Describes the variation of loudness levels within a program on a macroscopic scale. Not to be confused with "dynamic range", which is the distance between the noise floor and the highest possible peak of a signal path. It is based on statistics and uses the 3-second-short-term loudness levels. See EBU - Tech 3342, linked in the Bibliography.

LU: Loudness Units. A step of 1 LU is the same as the step of 1 dB. If one stream's target loudness is −16 LUFS and another stream is set to −18 LUFS, then they are 2 LU apart. Also a relative loudness value where the target level is 0 LU. If the target level is, for example, −23 LUFS = 0 LU, then an audio signal with a program loudness of −19 LUFS could be written as PL being +4 LU. See EBU Tech 3341 and ITU-R BS.1771-1.

LUFS: Loudness Units with reference to digital Full Scale. The EBU preferred term for the absolute loudness value, which includes a psychoacoustic filter shape known as K-weighting, explained in ITU-R BS.1770-3. (See LKFS).

Normalization: The process of adjusting the loudness of incoming material to conform to a target loudness. See Appendix above for a description of some algorithms.

Peak Normalization: The practice of adjusting the peak level of each program to (usually) full scale or 0 dBFS. This results in widely-varying loudness levels. This practice has been cited as the original cause of loudness wars in broadcast and music production.

PLR: Peak to loudness ratio. The ratio of maximum true peak level of a program segment to its integrated ITU-R BS.1770-3 loudness.

Program: A section of a stream having a single format such as news/talk, or popular music with DJ introductions and commentary. In this document, we treat short, inserted material like commercials as separate program breaks having special loudness constraints because different production teams typically produce the commercial and the program into which it is inserted, because the content of the commercial is often unrelated to the content of the surrounding program, and because obtrusively loud commercials annoy listeners.

Program Loudness: A measurement of Integrated Loudness over the entire length of a program. Abbreviated **PL**.

Short-Term Loudness: As defined in EBU Tech 3341, Short-Term loudness uses ITU-R BS.1770-3 algorithm, but with no gating and a sliding rectangular time window of length 3 s. The update rate for "live meters" must be at least 10 Hz.

Stream: A continuous transmission over a network (typically the Internet) that consists of one or more programs presented sequentially. Analogous to a "radio station" in over-the-air broadcasting.

Streamer: A content provider offering a streaming service to customers.

Target: The intended Integrated Loudness of the entire stream. In mixed-format streams it can also refer to the intended Integrated Loudness of programs having a given format within the stream.

Transcode: To convert material from one coded representation to another. In the worst case this is complete decoding of the material, and re-encoding. For example, to convert from one bitrate to another, or to insert processors in the signal path. Transcoding is frowned upon because it can multiply the artifacts of codecs.

True Peak Level: See ITU-R BS.1770-3, Annex 2.

To contact this committee with questions or suggestions, please email

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