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Audio Mastering and Metering:
Peak Normalization vs. Loudness Normalization

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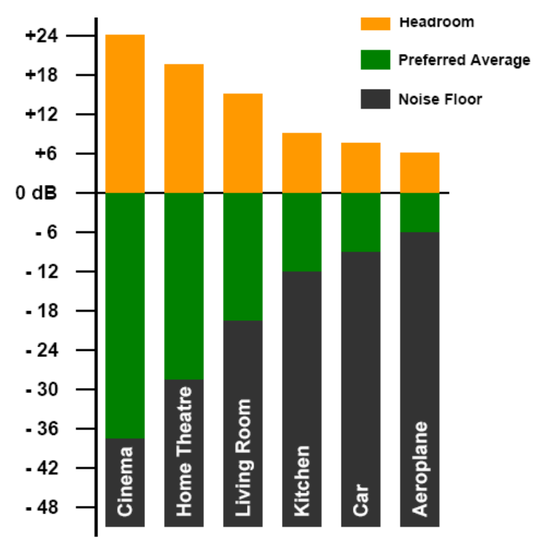
ABSTRACT

After years of fighting between content producers in order to make their track sounding louder than any other one (i.e. commercials, music, etc.), a new approach is gaining respect by sound professionals and companies: providing loudness consistency and an higher audio definition is a must and is now possible.
Moving from Peak Normalizing to Loudness Normalizing increases audio quality, generates more pleasant sound design, allows greater headroom and produces audio programs that will be perceived at a constant equivalent volume, regardless the content and the electric values of the soundwave.
Manufacturing industry is very active in developing new tools that allow the sound designers to measure and monitor the audio mix during sound design, the mixing and the final mastering, in order to achieve the best results in terms of loudness consistency and audio quality.

1. WHAT IS MASTERING?

Mastering is the final process that aims to optimize the mix in order to bring it to the listener at the maximum enjoyable level. That means that it is a crucial point to know what the final delivery media will be, in order to address the sound design and the mix in the right way. Each media has specific requirements that will influence the Sound Engineer since the early audio process, up until the final master creation. That is mainly due to the recording/playback system that is going to carry the final audio mix, and to the so called Dynamic Range Tolerance associated to the environment where that media is supposed to be used. I.e. cinema theatres, domestic rooms, cars, etc. are characterized by different listening conditions and require appropriate dynamic processing in order to be perceived at a pleasant level.



2. AUDIO METERING

Besides knowing how our mix will be carried and into which environment it will be listened, to print a trustable audio master we need a linear monitoring system, powering mastering tools, and a good pair of ears, of course. We also have to assist the sound design process with the support of reliable audio meters.

2.1 ELECTRICAL LEVEL METERING

Several scales measuring the electrical level of the audio signal have been developed during time being, and adopted by different regions in the world.

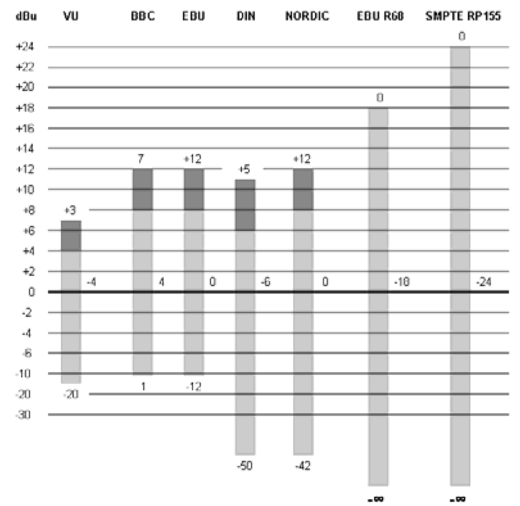
2.1.1 VU Meter

- VU (Volume Unit), standard audio meter since 1939. Originally intended for telephone lines, due to its integration time (300ms) and typical slow reaction to peaks, it soon became the main sound metering tool in audio applications and equipment, until the advent of digital audio. After that, its reading characteristics made it inaccurate and not usable in modern operations.

2.1.2 PPM Meter

In the latest decades, PPM meters (Peak Programme Meter) have gradually replaced VU meters.
Their integration time is usually 5 or 10 ms, while the fallback time is usually 20dB/1,5sec.
Several PPM scales have been developed by different countries. Among other, the most popular ones are:

- IEC I, Nordic Scale: used by broadcasters in Northern Europe, ranges from -42 to + 12 dB;
- IEC IIa, BBC: developed by BBC in UK, also used in other countries, ranges from 1 to 7 (each step represents 4 dB);
- DIN, mostly used in Central Europe, ranges from -50 to +5 dB;
- IEC IIb, EBU: adopted in transmission by EBU, ranges from -12 to +12 dB.



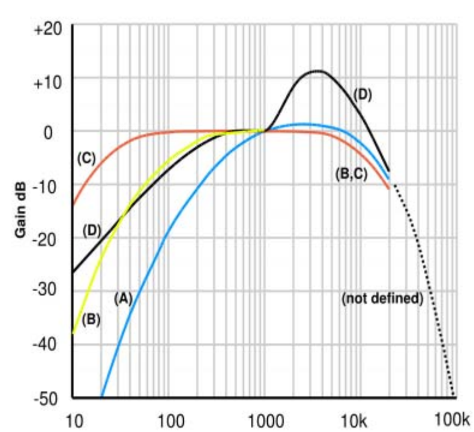
Due to their reading characteristics, they are typically used in broadcast operations, but are not fast enough to detect digital overload, since these meters would not read transients/peaks that could reach 3 dB higher than the displayed level.

2.1.3 DMU

The DMU is a full scale digital meter, with 0ms integration time. Its scale ranges from -∞ to 0 dBFS, which represents the maximum possible level. It is particular indicated for digital operations.

2.2 LOUDNESS METERING

We could define Loudness as the perception of the sound volume perceived by the human hearing.
Among others, it is dependent on many factors such as: frequency, RMS level, duration, and masking. On the opposite, it is not related to the fast audio transients and peaks.
For these reasons, all the above mentioned metering tools require specific interpreting skills in order to determine the correspondent perceived loudness of the audio program. In particular, broadcast transmissions have been suffering of very annoying inconsistent loudness, due to objective difficulties in interpreting the loudness of TV programs, and a lack of standardization in this matter.
To solve this issue, several loudness algorithms and methods have been developed during the years, since 1960. The most popular ones are: Zwicker's (ISO 532-1975), K-system, TC-Electronic LARM, Leq(m), Leq(A), Leq(C), Leq(RLB), and ITU-R.BS1770. They differ the way the audio signal is weighted, integration time and methodology.



A-weighting (blue), B (yellow), C (red), and D-weighting (blk)

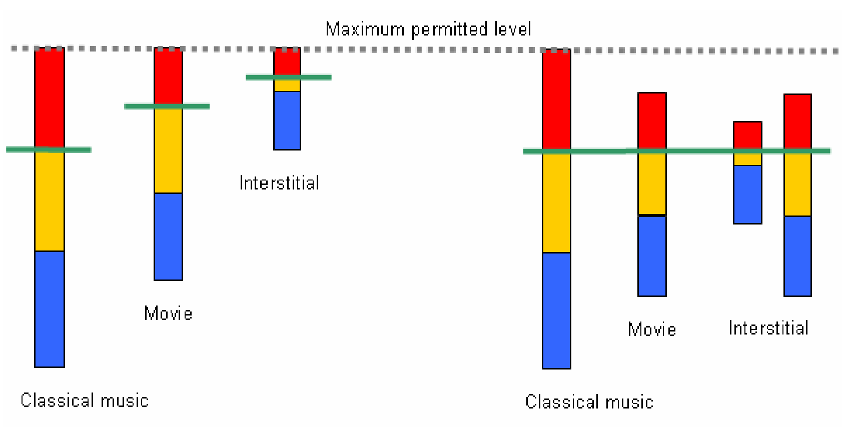
Consequently, audio manufacturers have designed reliable tools able to give an easy reading of the real equivalent volume perceived by the listener. Several studies have been carried out in order to determine which algorithm better applies to various sound contents and production fields.

3. PEAK NORMALIZATION

As stated before, the perception of loudness doesn't depend on fast audio peaks but on average level, also in accordance to its spectrum image and structure. As a consequence, the more compressed the audio mix is, the loudest it would sound, if normalized at the maximum permitted level allowed by its media.
In music production, where the master level is normalized by peaks at 0dBFS, we have entered the so called "loudness war", where the main goal is to make the final mix as loud as possible. Mastering engineers have developed skills and techniques aimed at that, but in some cases this has reduced the overall quality of the final mix in order of dynamic and linearity. The risk of introducing audio artifacts such as saturation and digital distortion is also very high.
In broadcast, as well, the current international recommendations (EBU R68, SMPTE RP155) only indicate two values: the alignment level (used for setting playback and transmission equipment) and the MPL (Maximum Permitted Level) over which transmission would not be allowed. No indication of loudness is given. As a consequence, the loudness varies according with the content put to air, and more specifically with the compression degree of its audio signal. Some short program such as commercials are heavily compressed in order to get an increase in loudness, while others that maintain their original wide dynamic range and volume modulation (i.e. movies) will be perceived at much lower level. This results in loudness inconsistency when switching from program to interstitial, and from channel to channel. Commonly, broadcasters approach this problem inserting audio processors (automatic gain control, compressor, limiter, compander) in the play-out/transmission chain. In some cases this generates an unpredictable modification of the original audio mix, which some times results in over-compression and decrease in quality.

4. LOUDNESS NORMALIZATION

Especially in broadcast (but not exclusively), it appears obvious how a different approach to audio metering is needed in order to offer audio programs at higher quality and at constant loudness. The audio industry has made enormous efforts in studying this phenomenon and finding tools and procedures capable of dealing with it.



In particular, some broadcasters (Discovery, SKY Italia, FOX Italy) have adopted technical specifications that include loudness metering and processing for every audio operation (production, post-production, ingestion and transmission). This has generated several benefits, including: standardization of production procedures, loudness consistency through programs and channels, reduction of using of audio processors in transmission, wider dynamic range available for interstitial contents.

4.1 ITU-R.BS1770 and BS1771

In 2006 the ITU published two new recommendations aimed to offer a standard practice to broadcasters in regard to loudness metering. The BS1770 describes an algorithm labeled R2LB (derived from the RLB weighting curve). It is recommended for metering the equivalent perceived loudness of audio signals, including mono, stereo and multichannel. Besides that, another recommendation (BS1771) indicates some requirements for designing loudness meters complying with the BS1770. The unit is called LU and suggests to distinguish between short term values (F = Fast) end long term ones (I = Integration). Some manufacturers have already implemented this algorithm into their tools, and it is highly auspicial that others would follow soon.

4.2 DIALNORM and PROGRAM OPTIMIZERS

Another way of normalizing the loudness of audio transmission is handling metadata. In particular, Dolby Dialnorm is utilized in authoring Dolby Digital streams. This technology allows the audio to be recorded (and transmitted) at its maximum level. According with the loudness value indicated in the Metadata, the master audio playback level is shifted in order to match with a fixed target level.
Another solution recently introduced by some manufacturers, consists in analyzing the audio program of the media files ingested in the audio/video server, and processing them prior to transmission, in order to make all them match with a predetermined loudness value.

5. MIXING and MASTERING MEMO

The best results in terms of quality, definition, linearity and loudness consistency of audio mixes are given if adequate care is given right from the early production process: bearing in mind what the final media is and its Loudness Range Tolerance, checking the alignment of the meters, inserting the master dynamic processor at the beginning of mixing, using the dialogue as the main audio element as loudness reference, applying dynamic/loudness control on each discrete track rather that on the master, applying adequate compensation in leveling and filtering the Centre/LFE/Surround channels when down-mixing into stereo.

AUTHOR’S BIO

Alessandro Travaglini is born in 1970. After studying Electronic and Telecommunication technology, he began his career as an Audio Engineer, including Live music and theatre productions, TV shows and music recording. In the middle 90s he focused on Digital Audio, with particular regard to post-production for moving pictures, spending the last 13 years in working as Sound Designer and Audio Engineer for the broadcast industry.
Since 2005 he applied his background in studying the loudness issue and consulting for SKY Italia, for which he has developed new audio specifications aimed to balance the loudness of the platform transmission. He is currently covering the position of Senior Sound Designer at FOX Italia, a News Corp. company that produces and transmits more than a dozen of TV channels (including 5.1 HD) in several European countries.
Alessandro is a music composer, a multichannel audio mixing/authoring engineer and an approved Dolby trainer.
He is member of:
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