Audio Metering and Linux

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Abstract

This documents presents an overview of current audio level and loudness measuring techniques and concepts, in the context of musical and post-production environments. It proposes a metering application, which is currently in early development, which should address metering needs currently unsatisfied in Linux.

Keywords

Audio level metering, Peak Meter, RMS Meter, PPM Meter,Equal loudness

1 Introduction

Sound is energy propagating as compression and rarefaction through a medium, but when captured, or artificially generated it is represented as a variation in voltage (or a representation of this variation as digital samples, grooves on a record or variations in tape magnetization). The ear is a sound pressure sensor which together with the brain produce the experience of hearing. There is a technical and practical need to quantify sound pressure levels and perceived loudness, which is done with sound level and loudness meters. A particular setting for the usage of audio metering presents itself in the production of music and sound for audiovisual productions.

1.1 Decibels

The ear senses sound pressure non-linearly, so the unit developed to quantify audio level, called decibel (dB), is logarithmic, and has been defined as:

$$L_1 = 10log_{10}\frac{W_1}{W_2} \tag{1}$$

Where the value in decibels of power W_1 is L_1. Decibels are always expressed in relation to a reference power W_2. This definition can be used to calculate decibels when the measurement is in acoustic power or intensity, or electric power. This is not practical, since most often we measure air pressure, voltages or current. From their relation to power, we can express equation (1) as:

$$\begin{aligned} L_p &= 10log_{10}\frac{p_1^2}{p_2^2} \\ &= 20log_{10}\frac{p_1}{p_2} \end{aligned} \tag{2}$$

The standard reference pressure for sound is 20 μPa. If this reference level is used to measure sound pressure level (SPL) in the air, the result is expressed as dB_{SPL}. Decibels in reference to voltage level are also indicated with subscripts, like dB_u for a reference level of 0.775 $Volts$.
Equation (2) can be used to calculate decibels in full scale (dB_{FS}) for PCM digital systems, where the absolute maximum sample value corresponds to $0dB_{FS}$, and amplitudes are expressed as negative numbers below it. When signed integer samples (8-bit *short*, 16-bit *int* or 24-bit *long* types) are used, the following formula can be used to calculate full-scale decibels:

$$\begin{aligned} L_{FS} &= 20\frac{ln \mid A_i \mid}{ln10} - 20\frac{lnA_{max}}{ln10} \\ &= 20\frac{ln(\mid A_i \mid / A_{max})}{ln10} \quad \mid A_i \mid > 0 \\ &= -\infty \quad A_i = 0 \end{aligned} \tag{3}$$

Where the value in full-scale decibels L_{FS} for absolute amplitude A_i is calculated as the difference with the decibel value for the greatest possible sample value A_{max} in a given bit precision. The reference value is the minimum sample value available (i.e. 1) and natural logarithm is used since it is often more practical in computer systems to use them to take advantage of the *exp()* function from the standard *math.h* library, when calculating amplitude from dB_{FS} values.
The value of A_{max} is:

$$A_{max} = 2^{n-1} \tag{4}$$

Where n is the integer sample bit depth, and 1 is subtracted from the exponent because samples values oscillate around a center, so the total number of values must be divided by two to represent positive and negative displacement (when using signed data types).
When using floating point samples, there is no set reference level, but when using the usual range from 1 to -1, the following can be used to calculate dB_{FS}:

$$\begin{aligned} L_{FS} &= 20\frac{ln(\mid A_f \mid)}{ln10} \quad \mid A_f \mid > 0 \\ &= -\infty \quad A_f = 0 \end{aligned} \tag{5}$$

Note that the absolute value of the amplitude must be used in both cases.

1.2 Root Mean Square

The previous method of calculating decibels yields "instantaneous" values reflecting a temporary state that doesn't represent actual energy in an oscillation. For this reason, sound volume (and voltage average) is sometimes calculated using Root Mean Square (RMS), which presents an average that better describes an oscillating signal's level. The RMS value is always calculated for a certain period of time between T_1 and T_2 and is defined as:

$$f_{rms} = \sqrt{\frac{1}{T_2 - T_1} \int_{T_1}^{T_2} [f(t)]^2 \, dt} \tag{6}$$

For discrete values, like digital PCM samples, a signal's RMS value for a group of N samples is:

$$x_{rms} = \sqrt{\frac{1}{N} \sum_{k=1}^{n} x_k^2} \tag{7}$$

1.3 Loudness

There is a fundamental difference between perceived loudness and nominal audio levels. Audio with a high level might sound softer than one with lower levels. There are several reasons for this.
It is well known that the human ear perceives loudness as a function of amplitude and frequency. The recent ISO 226:2003 standard [1] presents a revision of the well known Robinson-Dadson (in turn based on the Fletcher-Munson) curves. It defines a set of equal loudness contour lines on a frequency vs. sound-pressure level graph, relating sound pressure level and frequency to subjective human perception of loudness. The curves show that the human ear is most sensitive to frequencies between 3000 and 5000 Hz, and that at higher sound pressures, the difference in sensitivity is reduced, making the curves flatter.

It has also been shown that the ear perceives short transients softer than a longer equivalent sound [2]. Short transients might report a high level on a peak meter, but not be perceived as loud by a listener. This effect is significant for sounds shorter than 100ms, and becomes more evident for shorter sounds.

Another important concept is long-term loudness . This is an average loudness over a longer period of time (usually a complete program) which quantifies perceived loudness, to be able to compare and match with other material [3].

2 Uses of Level Metering

There are three uses for level metering:

- To keep levels within equipment limits
- To assist subjective judgement of levels
- To comply with delivery or industry standards

Controlling signal levels is important in all aspects of audio production and delivery, from recording to mastering. Audio equipment, whether analogue or digital, has a threshold above which, audio isn't accurately represented. If this threshold is exceeded, distortion occurs. This is sometimes used as an effect, but is undesirable in most cases. If a signal level is too low it might be degraded by a device's own noise floor. Metering guarantees a clean signal path, helping the musician or engineer stay within equipment limits. For this application, the best suited meters are ones that can show potential problems clearly. Meters with a fast response, that can clearly show peaks, are the most suited.

Meters can be helpful as an objective means of measuring audio, when deciding relative mix levels, or to help achieve consistent loudness between different sources. Even though the ear should be the final judge, meters can help a tired ear, or an operator in an unusual or inadequate environment to better judge audio levels.

When delivering content for broadcast or mastering, it is ideal and sometimes compulsory to comply with certain standards, which vary greatly depending on the situation and destination. Adequate metering will help audio material pass quality control standards and delivery requirements. Adequate metering together with proper speaker calibration will assist in producing adequately loud masters, while respecting standard program loudness. Film sound has had a "standard" calibration setting of $85dB_{SPL}(C)$ for $0VU$[4]. This, though not really a standard, is a respected practice -mostly- throughout the industry. This technique is explored for other settings in Bob Katz's K-system [5], which is a combination of metering/calibration guidelines.

3 Techniques for measuring audio

Throughout the history of audio technology, different techniques have been used to measure audio levels. Most of these techniques are deeply related to the medium they measure, and vast differences can be seen for instance in meters targeted at analogue and digital systems. Several techniques have been developed to imitate subjective perception of loudness.

3.1 VU

Originally developed in 1939, the VU (Volume unit) meter is the oldest type of metering still in usage, and consists of power measurements with time averaging (sometimes referred to as the meter's ballistics). This type of meter is calibrated so that 0 Volume Units represent 1 milliwatt of sine-wave power at 1000 cycles per second (a 1000 Hz tone at $+4dB_u$) [6]. A VU meter has rise and fall times of 300 ms, sometimes called the time integration constant. This means that the VU should take 300ms to reach 99% of the target voltage value. The VU meter is not particularly well suited to detect problematic peaks, or to measure music loudness. Still, it can be useful, as many analogue consoles (particularly low and mid-range studio consoles) provide little headroom above 0VU and produce audible distortion even for slight peaks above 0VU. Although today there are better systems to measure speech loudness (See section 3.4) and detect problematic peaks, VU meters are still ubiquitous, specially in analogue equipment, and their measurements are still used for delivery standards in certain cases.

3.2 PPM

To address some of the short-comings of VU meters, Peak Programme Meters (PPM) were developed and standardized. These meters, sometimes called quasi-peak meters, are also time averaged RMS meters, but have a much faster attack than VU meters, therefore showing potential peak problems more accurately. They don't have instantaneous response, but they were designed according to the ear's ability to detect short distortion. There are several types of PPM meters[1] [7] :

Nordic These meters have an attack time of $5ms$ (for 77.77% target level) and a decay time of 1.5 seconds for $20dB$. The scale shows values from -36 to +9.

BBC/EBU These meters have an attack time of $10 \pm 2ms$ (for 77.77% target level) and a decay time of 2.8 ± 0.3 seconds in fast mode and 3.8 ± 0.5 for 24 dB. The difference between the BBC and EBU standard resides only in the scale used. The BBC uses a scale from 1 to 7 where 4 equals 0dB_u, 6 being considered the maximum acceptable peak, and EBU uses a scale from -12 to +12.

PPM meters usually report around 4dB higher than VU meters. These meters are useful to make sure equipment doesn't generate any audible distortion, but still have a slow decay to approximate perceived loudness. They are widely used (specially in Europe) for broadcast delivery standards. PPM meters are technical tools and don't seek to measure loudness. VU meters are typically better loudness meters.

3.3 Sample Peak Meters

Digital audio equipment and software typically implement sample peak meters which show the maximum sample received. This type of meter is simple to implement as you only need to check if the new sample is greater than the current meter position and if not, divide the meter value by some factor to generate a smooth movement. This "fall-off" improves the readability of the meter (otherwise very short peaks might be too short for the eye), while making sure all the sample peaks are reported.

These meters can miss inter-sample peaks that may appear on the digital-to-analog conversion. Interpolated upsampling can increase the precision of this meters [8].

3.4 Equivalent Continuous Sound Level Measurement

Some studies [9] have identified Equivalent Continuous Sound Level measurements (implemented mostly on sound level meters rather

[1] DIN 45406 specifies another type of fast PPM metering.

than audio equipment) as the most accurate method to determine perceived loudness for long term measurements[2]. Equivalent Continuous Sound Level has been defined as [10]:

$$L_{A_{eq}T} = 20log_{10}\left[\frac{\sqrt{(1/T)\int_{t-T}^{t} p_A^2(\xi)d\xi}}{p_0}\right] \quad (8)$$

Equal Loudness is the value in decibels of the RMS value for a time period T of an A-weighted (see section 4) audio signal with pressure p, referenced to pressure p_0 [3]. This can be expressed for N number of samples in a digital system as:

$$L_{A_{eq}T} = 20log_{10}\left[\frac{\sqrt{(1/N)\sum_{k=1}^{n} x_k^2}}{p_0}\right] \quad (9)$$

$L_{A_{eq}T}$ measurement with a few additions has been adopted by Dolby for their flagship broadcast meter, the LM100 [11]. This device is slowly gaining ground (at least in the USA) and becoming a standard for audio delivery requirements [4].

3.5 Other loudness calculation techniques

Different manufacturers and standards institutes have developed other loudness measurement techniques. An interesting example is the Zwicker Loudness model (DIN 45631/ISO 532B), which performs separate measurements for different spectral bands. Also noteworthy are the LARM and HEIMDAL algorithms from TC electronics [12], which performed exceptionally well in two separate studies. Neither of these has achieved widespread usage in broadcast or music production.

4 Weighting

To compensate for the variation in detected loudness with respect to frequency and pressure (See section 1.3), several "weighting" networks have been standardized. Weighting networks are filter networks that approximate ear response characteristics in a simple and efficient way. There are several filter networks in use and study today in audio loudness metering:

A, B and C Weighting These weightings, designed to be used for low, medium and high pressure levels respectively, implement low pass and high pass filters, leaving a flat middle section [7]. They are frequently found on SPL meters and A-weighting is the basis for Equal Loudness level calculations ($L_{A_{eq}T}$).

M Weighting This weighing, detailed in ITU-R 468 (previously CCIR 468), has been dubbed "M" for Movie, since it has been used to compare loudness between different sections of film, or with movie trailers.

RLB and R2LB Weighting The Revised Low-frequency B-weightings [13] have the closest approximation to subjective loudness among the weightings according to some studies [12]. This weighting has been proposed in ITU-R BS.1770 as the basis of an equal loudness measurement ($L_{eq}(RLB)$).

There's still no concensus on the most appropriate method or weighting to evaluate loudness. Some techniques appear better than other for certain material or listening conditions, and experimental data is sometimes conflicting.

5 Software Metering

Metering, though traditionally implemented in hardware devices, has seen PC software counterparts. Apart from meters available within audio applications, there are many software packages dedicated to level and loudness metering, however few of them are open source, even though some are freeware. Worthy of notice are:

SpectraFoo This is one of the most complete metering tools available. It is a standalone application or TDM/MAS plugin for ProTools TDM and Digital Performer. Apart from a complete array of metering tools, with logging, it includes spectral, phase and other types of analysis. It can work in real time or from a file. Only for Mac OS [14].

[2] The study has been contested by both TC Electronics and Dolby Labs according to the article "Real-time loudness control for broadcast" by Thomas Lund, found at: http://www.broadcastpapers.com/whitepapers/Realtime-Loudness-Control-For-Broadcast.cfm?objid=32&pid=35&fromCategory=26.

[3] ξ is a dummy variable of time integration

[4] $L_{A_{eq}T}$ measurements are suggested as the way to set the *dialnorm* parameter in ATSC Digital Television Standard A/53 Revision E. The DVB Project has not set any audio standards yet, but also uses AC-3 for audio encoding.

Signal Tools A ProTools plugin capable of long-term $L_{A_{eq}T}$ measurements on the TDM version [15].

Pinguin Audio Meter Pro Only for Windows, provides K-system and PPM metering [16].

6 Metering on Linux

Audio software typically provides some form of sample peak metering. Most Linux programs follow this practice. Multitrack environments like Ardour and Rosegarden, and audio editors like Rezound and Audacity provide the usual peak meter strips for each channel. Other software like Pure Data, Csound or Ecasound can provide text information about sample peaks. Currently the most advanced option for audio level metering on Linux is Steve Harris' Meterbridge package [17]. This package contains sample peak, PPM and VU metering (apart from other stereo metering tools). It is a simple but effective and visually appealing package. However, it is designed to be used as a real-time visual aid only.

6.1 What's missing

The available metering options are probably adequate for most music production projects, but for serious post production work, particularly with the advent of digital television standards, and for audio quality control and analysis it becomes important to have other standard compliant metering options like $L_{A_{eq}T}$ and $L_{eq}(RLB)$, and also options for logging and for calculating long-term loudness.

Logging in this context can take two main forms:

- Graphical or text log of the measured levels for regular time periods
- Histogram of density of occurrence of levels

Implementing loudness measuring algorithms as a library might prove useful for other programs like media players, to help achieve good quality automatic loudness matching. Other operating systems like Windows Vista have implemented similar schemes [18], but no technical details are available, though it is likely, since there is no mention of patented technology, that some known algorithm is used.

7 PostQC

A graphical metering tool called PostQC is currently under development by the author, which will support many of the standard metering options mentioned in this article, and will implement needed logging and long-term measurements. It is still in early development, although some features are already working. Figure 1 shows a screenshot of the current state of PostQC. It can be seen that sample peak and $L_{A_{eq}T}$ metering has been implemented and logging can be shown in a somewhat primitive histogram form. Level threshold overshoot is logged showing the channel in which the overshoot occurred, the duration and the amount of overshoot. Most of the work has been done and tested for file input, but the jack real-time portion is almost ready, as all the jack engine is done, and all that needs to be done is the jack callback function, which is a slight variation to the file block process function. PostQC has been programmed in C++ using QT 3.3 and currently depends on libsndfile 1.0.17 and jack. It is to be publicly available soon under the GPL.

Some of the goals of the project include:

- Real-time (Jack) and Offline Audio File metering
- Support for many types of standard level and loudness metering
- Real-time graphical meters
- Upsampled sample peak metering
- Written report of measured data
- Easy usage and useful in many contexts
- Lashified
- Emulation of LM100 dialogue detection by using a clean dialogue/narration signal as gate trigger to turn on and off long-term averaging of loudness.
- Level threshold overshoot report with occurrence time
- Level histograms
- Graphical and text level and loudness logs
- Make sure all meters adhere to standards

7.1 Other relevant audio measurements

Other aspects that might be useful to determine audio quality apart from metering, that could be included, include real bit depth metering, phase correlation and distortion (clipping) detection.

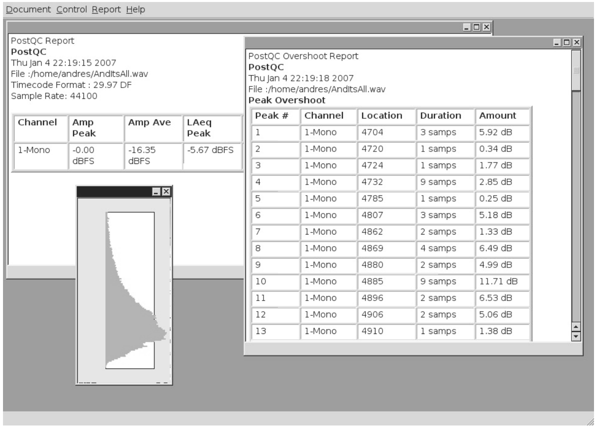

Figure 1: Screenshot of current development version of PostQC

8 Conclusions

Level metering is an important part of the technical and subjective production of audio material. Linux is not far behind from other platforms in this respect, but an additional tool like the one proposed will certainly make using Linux for metering very appealing.

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