

Audio Levels and Linux

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Audio Levels and Linux

- What are audio levels?
- Uses of level metering
- Techniques for measuring audio levels
- Software Metering
- Metering on Linux
- PostQC

Audio Levels

- Sound pressure level is measured in decibels
- It is a logarithmic relation between two measures of power
- Not so practical since in audio we generally measure voltage or pressure

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

Audio Levels

- Power is related to voltage and pressure by squares
- The formula for decibels can be expressed for voltage or pressure like this:

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

Audio Levels- digital samples

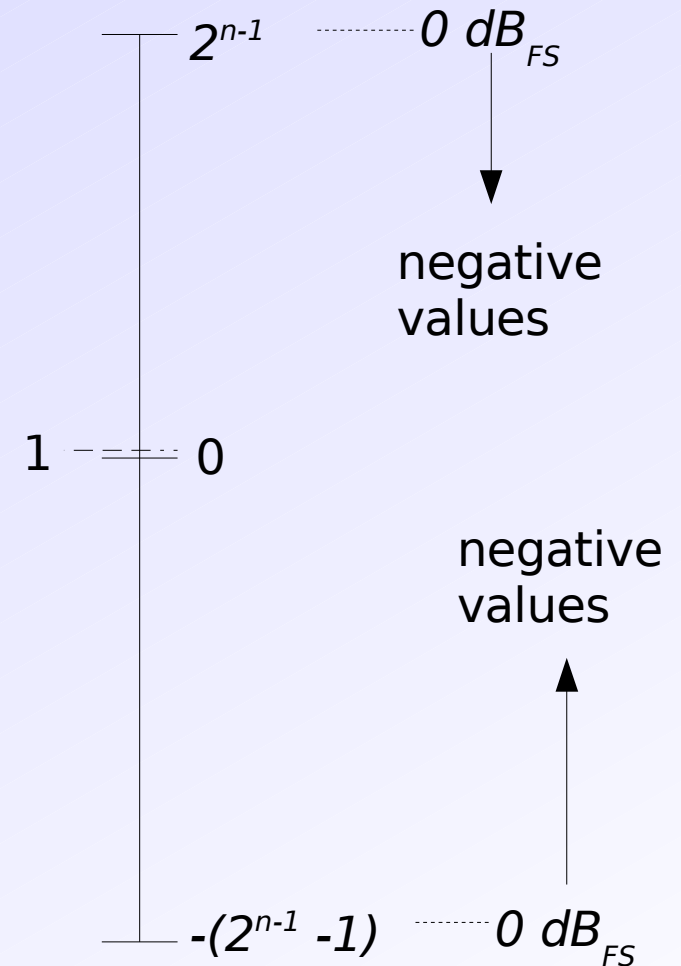
- For single signed integer samples, their individual decibel value (in dB_{FS}) can be calculated as:

$$\begin{aligned} L_{FS} &= 20 \frac{\ln |A_i|}{\ln 10} - 20 \frac{\ln A_{max}}{\ln 10} \\ &= 20 \frac{\ln(|A_i| / A_{max})}{\ln 10} \quad |A_i| > 0 \\ &= -\infty \quad A_i = 0 \end{aligned}$$

$$A_{max} = 2^{n-1}$$

Audio Levels-digital samples

- A_{max} is the maximum amplitude that can be represented in the digital system.
- In dB_{FS} scales, the maximum amplitude takes a value of 0, and all values are negative numbers below it.



Audio Levels- digital samples

- For signed float samples (normalized to 1), their individual decibel value (in dB_{FS}) can be calculated as:

$$L_{FS} = 20 \frac{\ln(|A_f|)}{\ln 10} \quad |A_f| > 0$$
$$= -\infty \quad A_f = 0$$

Root Mean Square

- A way of calculating “energy” for an oscillating signal is calculating its RMS, since individual samples don't reflect this.
- RMS is defined as:

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

Root Mean Square

- For discrete signals (like digital audio), RMS can be calculated as:

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

Loudness

- There is a fundamental difference between perceived loudness and nominal audio levels due to the imperfections of the auditory system.
- This has several causes:
 - Loudness is a function of frequency and this function in turn depends on pressure levels (Described in the well known Equal Loudness contours)
 - Short sounds are perceived softer than longer sounds particularly below 100 ms.

Uses of level metering

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

Techniques for measuring audio

- VU (Voltage Units)
 - Developed in 1939
 - Power measurements with time averaging
 - 0 VU = 1000 Hz tone @ +4 dB_U (4 dB above 0.775 Volts(RMS))
 - 300 ms to reach 99% of the target value (Rise time)
 - Slow, doesn't show short transients
 - OK for approximation of speech loudness and avoiding distortion on consoles with little headroom.

Techniques

- PPM (Peak programme meter)
 - Designed to address short comings of VU's.
 - Faster response, similar to an ear's response to transients and distortion.
 - Different standards
 - Nordic: Attack time 5 ms for 77.77% target level, Decay time 1.5 seconds for 20dB. Scale from -36 to 9.
 - BBC/EBU: Attack time 10 ± 2 ms for 77.77% target level, Decay time 2.8 ± 0.3 seconds in fast mode, 3.8 ± 0.5 seconds in slow mode for 24 dB. BBC scale from 1 to 7 (4 = 0 dB). EBU scale from -12 to 12
 - DIN defines another type of fast PPM metering

Techniques for measuring audio

- Sample peak meter
 - Shows the maximum sample value
 - Implements some “fall off” mechanism to improve readability. Usually the last peak is multiplied by a factor, and compared to the new sample. The greatest of the two is the current peak value
 - Can miss intersample peaks that may appear on the DAC. (corrected with interpolated upsampling)

Techniques for measuring audio

- Equivalent continuous sound Level
 - Identified by some studies as an accurate measure of perceived loudness.
 - Implemented mostly on sound pressure level meters, not recording equipment
 - Adopted by Dolby on their flagship LM-100 level meter.
 - Basically an A-weighted RMS measurement.

Equivalent Continuous Sound Level

- Equivalent continuous sound level is defined as:

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

- Expressed for discrete systems as:

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

Other loudness calculation techniques

- Zwicker Loudness model
- LARM and HEIMDAL algorithms from TCElectronics
- Claim to be better models, but neither is broadly accepted or implemented.

Weighting

- An important part of loudness measurement involves simulating the ear's relation between frequency and loudness.
- Weighing networks are used (basically filter network frequency response specifications)

Weighting

- A,B and C weighting
 - For low, medium and high pressure levels respectively
 - Low pass and high pass filters with a flat middle range
- M weighting
 - “Movie” weighting used to measure film trailer loudness

Weighting

- RLB and RL2B
 - Revised low frequency B-weightings
 - Some studies suggest this weighting as the most accurate for perceived loudness measurements
 - Standardized by ITU for an $L_{eq}(RLB)$ measurement.
- Still no concensus on the best weighting. Several contradictory results.

Software Metering

- Spectrafoo
- Signal Tools
- Pinguin Audio Meter Pro
- Many others.....

Metering on Linux

- Audio recording and editing software usually implement Sample Peak Meters, which are easy and efficient to implement
 - Sometimes as graphics (Ardour, Rosegarden, etc.)
 - Sometimes as text (Csound, PD, etc.)
- Meterbridge by Steve Harris
 - VU, Peak and PPM meters
 - Visually appealing

What's missing

- Modern standard compliant Loudness metering (LAeqT, Leq(RLB), etc.) especially for Digital TV post production.
- Meter logging
 - Text log
 - Timeline
 - Histogram
- Accurate perceived loudness measurements for use in automatic loudness matching.

PostQC

- In early alpha stage (horrible code and design...)
- Some features already working, though

The screenshot displays the PostQC software interface. The main window, titled "PostQC Report", shows the following information:

PostQC Report
PostQC
Tue Mar 20 11:41:46 2007
File : /home/andres/csound/marmstk44100.wav
Timecode Format : 29.97 DF
Sample Rate: 44100

| Channel | Amp Peak | Amp Ave | LAeq Peak | LAeq Ave | RMS(S) Peak |
|---------|------------|-------------|-------------|-------------|-------------|
| 1-Mono | -5.60 dBFS | -23.34 dBFS | -17.43 dBFS | -17.43 dBFS | -17.43 dBFS |

To the right of the report is a waveform display showing a signal with a peak level of approximately -17.43 dBFS. The y-axis ranges from 0.0 to -60.0 dBFS.

Below the main report is a "PostQC Overshoot Report" window, which shows the following information:

PostQC Overshoot Report
PostQC
Tue Mar 20 11:41:49 2007
File : /home/andres/csound/marmstk44100.wav

Peak Overshoot

| Peak # | Channel | Location | Duration | Amount |
|--------|---------|----------|----------|---------|
| 1 | 1-Mono | 9 | 1 samps | 4.71 dB |
| 2 | 1-Mono | 12 | 1 samps | 6.17 dB |
| 3 | 1-Mono | 25 | 2 samps | 3.88 dB |
| 4 | 1-Mono | 30 | 1 samps | 3.30 dB |
| 5 | 1-Mono | 38 | 4 samps | 6.73 dB |

PostQC Goals

- Realtime and offline analysis
- Many types of standard metering
- Real-time graphical meters
- Upsampled peak metering
- Written report of results
- Lashified
- Emulation of LM100 dialog detection
- Overshoot report
- Histograms
- Careful adherence to standards

PostQC

Document Control Report Help

PostQC

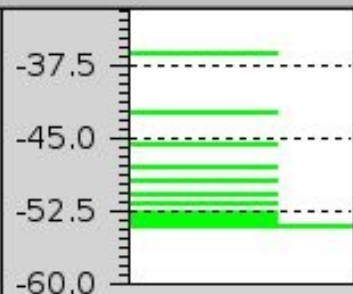
Tue Mar 20 11:50:21 2007

File :/home/andres/tonostereo.flac

Timecode Format : 29.97 DF

Sample Rate: 48000

| Channel | Amp Peak | Amp Ave | LAeq Peak | LAeq Ave | RMS(S) Peak | RMS(S) Ave | DC Offset |
|------------------|------------|-------------|------------|------------|-------------|------------|-----------|
| 1-Left Front | -6.00 dBFS | -15.96 dBFS | -8.90 dBFS | -9.06 dBFS | -8.90 dBFS | -9.06 dBFS | 0.00 % |
| 2-Right Front | -6.00 dBFS | -15.96 dBFS | -8.90 dBFS | -9.06 dBFS | -8.90 dBFS | -9.06 dBFS | 0.00 % |
| 3-Left Surround | -6.00 dBFS | -15.96 dBFS | -8.90 dBFS | -9.06 dBFS | -8.90 dBFS | -9.06 dBFS | 0.00 % |
| 4-Right Surround | -6.00 dBFS | -15.96 dBFS | -8.90 dBFS | -9.06 dBFS | -8.90 dBFS | -9.06 dBFS | 0.00 % |



Tue Mar 20 11:50:30 2007

File :/home/andres/tonostereo.flac

Peak Overshoot

| Peak # | Channel | Location | Duration | Amount |
|--------|--------------|----------|----------|----------|
| 1 | 1-Left Front | 2 | 20 samps | 14.00 dB |
| 2 | 1-Left Front | 26 | 20 samps | 14.00 dB |
| 3 | 1-Left Front | 50 | 20 samps | 14.00 dB |
| 4 | 1-Left Front | 74 | 20 samps | 14.00 dB |

PostQC

- <http://sourceforge.net/projects/postqc>
- GPL
- All and any help most welcome!

Questions...