Audio Measurements Workshop

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1 Overview

- Techniques and tools to measure
  * Soundcards
  * Analog hardware
  * DSP software

- Theory
  * Levels, decibels, noise, calibration, . . .

- Tools
  * jaaa, jnoisemeter, jsignal, . . .

- Practice
Philosophical issues

• Why measure things?
  * Verify your design and programming.
  * Have you been ripped off?
  * To know limits and create a level of confidence.
  * Curiosity.

• Always expect the unexpected. It happens. If your measurements are exactly as you imagined they would be, then
  * Congratulations!
  * It’s time to do some checks and ask some questions.

• Audio measurements often involve a mix of electrical and acoustic units as well as purely numerical values, and conversions between them. This can be very confusing unless you have a solid grip on the basics.
3 Theory

- Signals
  - Levels, power, impedance, balancing, . . .
  - Measurement methods.
- Decibels.
  - Reference levels, . . .
- Acoustic units and levels.
- Noise.
  - Distribution, spectrum, density.
  - Measurement methods and standards.
  - Thermal noise.
  - Equivalent input noise.
- Understanding audio specs.
  - Microphones.
- Calibration.
• Measured in Volts (V).
  * Rough levels: mics: 1 mV, consumer: 100 mV, pro: 1 V.
  * Current: \( i = \frac{u}{Z} \) (A).
  * Power: \( P = i \times u = \frac{u^2}{Z} = i^2 \times Z \) (W).

• Analog audio connections are almost always voltage driven.
  * Input impedance is much higher than output impedance.
  * Allows splitting the signal without level changes.
  * Exception: long analog lines (rarely used today).

• For numerical signals (no physical units), 'power' means the square of amplitude.

• The 'gain' of AD and DA converters requires some care to define without ambiguity as one side uses physical units and the other not.
5 Balancing

- Balanced connections:
  * Inputs take difference of two signals, cancels interference.
  * Different kinds of outputs and inputs are not always compatible.

- Outputs
  * Impedance balanced – very common.
  * Antiphase outputs – different variations.
  * Differential – rare.
  * Floating – requires transformer.

- Inputs:
  * Differential with unbalanced impedance – very common.
  * Differential with balanced impedance.
  * Floating — requires transformer.
  * Common mode rejection of input defines performance.
Balanced outputs

- Impedance balanced
- Antiphase outputs
- Transformer balanced

Differential

Transformer balanced
Balanced inputs: CMRR

- A ‘perfect’ differential input ignores the common signal.
- Real-life balanced inputs are not perfect.
- The **Common Mode Rejection Ratio** indicates by how much the common signal is attenuated.
- CMRR is usually a function of frequency.
- Typical figures:
  - Cheap: 20...25 dB
  - Reasonable: 40 dB
  - Transformer balanced: 80...90 dB
- Common mode signals can be the source of large errors.
- Common mode input or output impedances are usually not the same as the differential impedance.
8 Balanced inputs: CMRR

CMRR = Common Mode Rejection Ratio = $\frac{G_d}{\text{abs}(G_c)}$

Unbalanced signal with balanced attenuator.
If M is not connected to ground the common mode signal is not attenuated.
Audio level meters

- Defined by filter, detector, ballistics.


- Detector: what is measured.
  * Peak or pseudo peak value.
  * Average of absolute value.
  * RMS.

- Ballistics: response to level variations.
  * Rise and fallback times.
  * Burst response, can be different from rise time.
• RMS = Root Mean Square =
  * The square root of the average value of the square of the signal.
  * The average power expressed as an amplitude.

• The RMS value is independent of the relative phases of the different frequencies in a signal.

• 'Mean' or 'average' means some form of lowpass filter:
  * Rectangular window.
  * First or second order IIR, which is an exponential window.

• AC voltages are always shown as an RMS value, even if the meter is not an RMS one. In that case the measured value is correct only when the signal is a sine wave (e.g. almost all multimeters, VU, . . . )
• Logarithmic unit used to indicate ratios.

• 1 dB = 0.1 Bell. One Bell is a ratio of 10 to 1.

• Always a ratio of powers or something proportional to power.

  10 * log10 (ratio of powers)
  20 * log10 (ratio of amplitudes)

• Absolute measurements require a reference value.

• Memo trick: the 1/3 octave band frequencies: 1, 1.25, 1.6, 2, 2.5, 3.16, 4, 5, 6.3, 8, 10. Each step is 1 dB for powers or 2 dB for amplitudes.
• 0 dBm = a power of 1 mW. A standard impedance is assumed.
  * Audio: usually 600 ohm
  * RF: 50 or 75 ohm.

• 0 dBu = 0.7746 Volt, the voltage corresponding to 1 mW in 600 ohm. Quite often 'dBm' is used when 'dBu' is meant.

• 0 dBV = 1 Volt. Simple and easy. 0 dBV = +2.22 dBu.

• 0 dB FS = the amplitude of a maximum level sine wave in a digital system.

  If this is RMS, and the range is ±1, then the actual RMS amplitude of a sine wave at 0 dB FS is not 1 but sqrt (0.5) = 0.7071.
13 Acoustic levels

- **Sound Pressure Level** (SPL) is measured in Pascal (Pa).
- 0 dB SPL = an RMS sound pressure of $2 \times 10^{-5}$ Pascal.
  - * This is the threshold of human hearing at 1 kHz.
  - * A sound pressure of 1 Pa is +94 dB SPL.
- For electrical signals we have $i = u/R$ and $P = u \times i = u^2/R$.
- For acoustic signals we have $v = p/Z$ and $I = p \times v = p^2/Z$.
  - * $p =$ sound pressure (Pa)
  - * $v =$ particle velocity (m / s)
  - * $Z =$ acoustic impedance (N s / m$^3$)
  - * $I =$ acoustic intensity (W / m$^2$)
- The acoustic impedance of air at 20 degrees Celsius is 413.3 N s / m$^3$.

- Note: electrical current and power are scalars, but particale velocity and intensity are vectors – they have a direction.
We have a sound source with an acoustic power of 1 Watt.
What is the SPL at a distance of 3 meters?

- The power is spread over the surface of a sphere with radius 3 m.
- This surface is $4 \pi R^2$ or 113.1 m$^2$.
- So the intensity $I$ is $1 \text{ W} / 113.1 \text{ m}^2$, or $8.842 \times 10^{-3} \text{ W} / \text{ m}^2$.
- Now $I = p^2/Z$, or $p = \sqrt{I \times Z}$.
- Hence the pressure is $\sqrt{8.842 \times 10^{-3} \times 413.3}$ = 1.91 Pa.
- $1.91 \text{ Pa} = 20 \times \log_{10} (1.91) + 94 = 99.6 \text{ dB SPL}$.
We have an SPL of 1 Pa (+94 dB) and a microphone with a membrane of 5 cm\(^2\) (1 inch diameter). How much acoustic power does the mic receive?

- \(I = \frac{p^2}{Z} = \frac{1}{413.3} \text{ W} / \text{m}^2\).
- \(P = I \times S\) (\(S = \text{surface area}\)) so
- Power = \(5\times10^{-4} / 413.3 = 1.21\times10^{-6} \text{ Watt}\)

- This should be compared to thermal noise power (later).

- If all this acoustic power would be converted to electrical power, then a passive mic with an impedance of 200 ohm would produce 15.6 mV output.
• A random or pseudo random signal.

• Probability distribution
  – Rectangular, Gaussian, . . .
  – We normally don’t hear differences in distribution.
  – A sum of many independent random values will have a Gaussian distribution.
  – Filtered noise tends to have a Gaussian distribution.

• Density spectrum
  – Noise density $N_0 = \text{power per Hz}$.
  – At any particular frequency there is zero power.
  – White noise: constant density. $P = B \times N_0$
  – Pink noise: density proportional to $1/F$.
  – White and pink noise must be limited in bandwidth, or they would have either infinite power or zero density.
Measuring noise

- Requires either a true RMS meter, or one that is very tightly specified.
- Meter ballistics must be slow to have a stable value.
- Usually weighted using a standard frequency response.
- If no other filter is used the bandwidth must be defined.
- A number without specified measurement method is meaningless.
- Standard methods
  - 20 kHz equivalent bandwidth + RMS dB(20kHz)
  - IEC-A filter + RMS dB(A)
  - IEC-C filter + RMS dB(C).
  - ITU468 filter and pseudo-peak meter dB(ITU468)
- dB(A) is typically 1.9 dB lower than 20 kHz equivalent BW.
Standard noise weighting filters

- ITU-468
- IEC-A
- IEC-C

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• The 'best' method for measuring mics and mic preamps.

• Originally developed to measure noise on long analog audio lines.

• Filter emphasizes the most critical frequency region.

• **Pseudo-peak** meter having
  * a very slow response, as required for noise,
  * but very sensitive to short bursts and impulsive noise
    e.g. from a switching power supply or digital electronics.

• Typical measured values are around 9 dB higher than A-weighted.

• Used mostly in Europe, Americans use dB(A) because it looks better.

• The 'Dolby variant' uses lower gain and an average meter, but it is **not**
  and official standard.
ITU468 step and burst response

Step response and burst response of the ITU468 meter.

Burst response specification.
• Aka Johnson or Nyquist noise. Generated by thermal motion of electric charge carriers in all conductors.

• Essentially white (up to very high frequencies).

• Power density is proportional to absolute temperature.

• Power $P = 4kBT$
  - $k =$ Boltzmann’s constant, $1.381 \times 10^{-23}$ Joule / Kelvin
  - $B =$ Bandwidth in Hz
  - $T =$ Temperature in Kelvin.

• At room temperature and a BW of 20 kHz this means
  * Power $P = 3.24 \times 10^{-16}$ Watt
  * RMS voltage $v_n = 18$ nV $\times \sqrt{R}$
  * For $R = 150$ ohm $v_n$ is $220$ nV $= -133.1$ dBV $= -130.9$ dBu

• Other noise sources are present in real-life electronics, but mostly at low frequencies (below 100 Hz).
EIN: Equivalent input noise

- All components of an electronic circuit generate thermal noise.
- It is always possible to model an amplifier or an AD converter as a 'perfect' noiseless one with a single noise source at the input.
- \( EIN = \frac{\text{noise measured at the output}}{\text{gain}}. \)
- The noise generated near the input will contribute most, as it is amplified.
- For a well-designed amplifier, \( EIN \) will be independent of gain as long as the gain is high.
- Other low level signals may be present (e.g. 50 or 60 Hz and harmonics).
The EIN of mic preamps is usually measured at maximum gain with a source impedance of 150 ohm.

Some preamps can be a dB or so less noisy when measured with a lower impedance source or short-circuit.

The EIN can be compared to the self-noise of the mic to find out which generates most noise.

Some example values:
note: 150 ohm generates -135 dBV(A)

<table>
<thead>
<tr>
<th>Preamp</th>
<th>EIN dBV(A)</th>
</tr>
</thead>
<tbody>
<tr>
<td>RME Micstasy</td>
<td>-131.2</td>
</tr>
<tr>
<td>Aphex 1788A</td>
<td>-129.0</td>
</tr>
<tr>
<td>Sony SPR-V110</td>
<td>-133.5</td>
</tr>
<tr>
<td>Behringer PRO8</td>
<td>-132.9</td>
</tr>
<tr>
<td>Edirol UA5</td>
<td>-122.2</td>
</tr>
</tbody>
</table>
EIN vs full scale input level
EIN vs gain: why

\[ \text{Gain} = 1 + \frac{R_1}{R_2} \]

\[ E_n = \text{thermal noise due to } R_1 \parallel R_2 \]
\[ \text{+ amplifier input noise voltage} \]
\[ \text{+ amplifier input noise current } \times R_1 \parallel R_2 \]

- Except at low gains, \( R_2 \ll R_1 \) and \( R_1 \parallel R_2 \simeq R_2 \).
- \textbf{A}: For lower gain \( R_2 \) increases and \( E_n \) increases.
- \textbf{B}: \( R_2 \) is fixed and \( E_n \) is almost constant.
• Very few manufacturers of ’prosumer’ HW provide specs that have any real meaning, or that can be compared to others. This is of course entirely intentional.

• Example: M-Audio ’octane technology’ mic inputs. All you are supposed to know about those is: signal/noise ratio = 97 dB.

• Specs that don’t define measurement conditions are completely useless.

• Even levels are ambiguous. In pro audio +4 dBu is the ’work level’, peak level is at least 15 dB higher. Yet cards that can be switched between a peak output of -10 dBu and +4 dBu are presented as supporting pro signal levels.

• Most equipment reviews that can be found on the web don’t verify or even mention any technical specs. They are at best useless, if not completely bogus (the author got a free sample).

• If specs are not completely unambiguous, the only sane reaction is to mistrust them, and spend your money on something else.
When recording low-level signal (acoustic instruments, voice, natural sounds, . . . ) the combination of microphone and preamp will determine performance.

The important values are sensitivity and self-noise.

Sensitivity is usually specified as the output for 1 Pa (+94 dB) SPL.

Self noise is the acoustic level corresponding to the noise generated by the microphone, usually dB(A) or dB(ITU468).

Comparing specs for dynamic and condenser mics may require some calculations.
• Sensitivity and impedance are specified. Self noise is thermal noise corresponding to impedance. Given sensitivity, this can be converted to SPL.

• Example: Beyer M160: 1.0 mV/Pa, 200 ohm.
  
  1.0 mV = -60 dBV.
  
  Thermal noise is $18 \text{nV} \times \sqrt{200} = 254 \text{nV} = -131.9 \text{dBV}.$
  
  Self noise is $-131.9 + 60 - 94 \text{dB} = 22 \text{dB} = 20.1 \text{dB(A)}.$

• Example: Shure SM58: 1.85 mV/Pa, 300 ohm.

  1.85 mV = -54.7 dBV
  
  Thermal noise is $18 \text{nV} \times \sqrt{300} = 312 \text{nV} = -130.1 \text{dBV}.$
  
  Self noise is $-130.1 + 54.7 - 94 \text{dB} = 18.5 \text{dB} = 16.6 \text{dB(A)}.$

• To use the full dynamic range a preamp with an EIN better than -130 dBV is required.
• Sensitivity and self noise and/or S/N ratio are specified. S/N ratio is relative to 1 Pa, so self noise + S/N ratio = +94 dB. Given sensitivity, either can be converted to noise voltage.

• Example: Neumann KM184: 15 mV/Pa, self noise 13 dB(A).
  
  15 mV = -36.5 dBV.
  
  Noise voltage = 13 - 36.5 - 94 dBV(A) = -117 dBV(A).

• Example: Neumann TLM103: 23 mV/Pa, self noise 7 dB(A).
  
  23 mV = -32.8 dBV.
  
  Noise voltage = 7 - 32.8 - 94 dBV(A) = -119 dBV(A).

• A preamp with an EIN of around -122 dBV(A) or better will allow the full dynamic range of these mics to be used.
Calibration in this context means having a known relationship between analog signal levels (dBV) and digital levels (dBFS).

This can be expressed as a gain, in dB FS/V for inputs and dB V/FS for outputs.

The best soundcards to use for measurement are those having fixed or exactly repeatable analog gains. Real balanced outputs are a plus, but they can also be simulated using two unbalanced ones.

Beg steal or borrow a true RMS audio voltmeter (or attend this workshop). Check its frequency response by comparing 100 Hz, 1 kHz, and 10 kHz.

Calibrate at 1 kHz, check maximum output levels, distortion and noise.

Write down the results and store them with the audio interface.
Some calibrated passive attenuators come in handy, e.g. for
- line out to mic in,
- speaker signal to line in,

Use 1 percent metal film resistors, or measure them using a good digital resistance meter and calculate the attenuation.

When using passive attenuators, consider the effects of input and output impedances.

For measuring EIN: an XLR with a 150 ohm resistor between pins 2 and 3, and a 60...80 dB attenuator with the same or lower output impedance.

A junkbox with an assortment of connectors and short pieces of cable.

A notebook (paper). Write down your methods and results in detail, you will be happy you did later.
Questions and answers, hands-on practice.
• Some Python extensions and objects using Jack.
  AudioFile: read/write audio files into/from numpy arrays.
  JackControl: connections and transport.
  JackPlayer: multichannel audio file player.
  JackSignal: playback and capture from/into numpy arrays.

• Tested with Python 3.4, but should work with 2.7 as well.

• No manuals ATM, but full docstrings in the code and some examples.