Audio Measurements Workshop

Fons Adriaensen
Casa della Musica, Parma

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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

2 Philosofical 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.

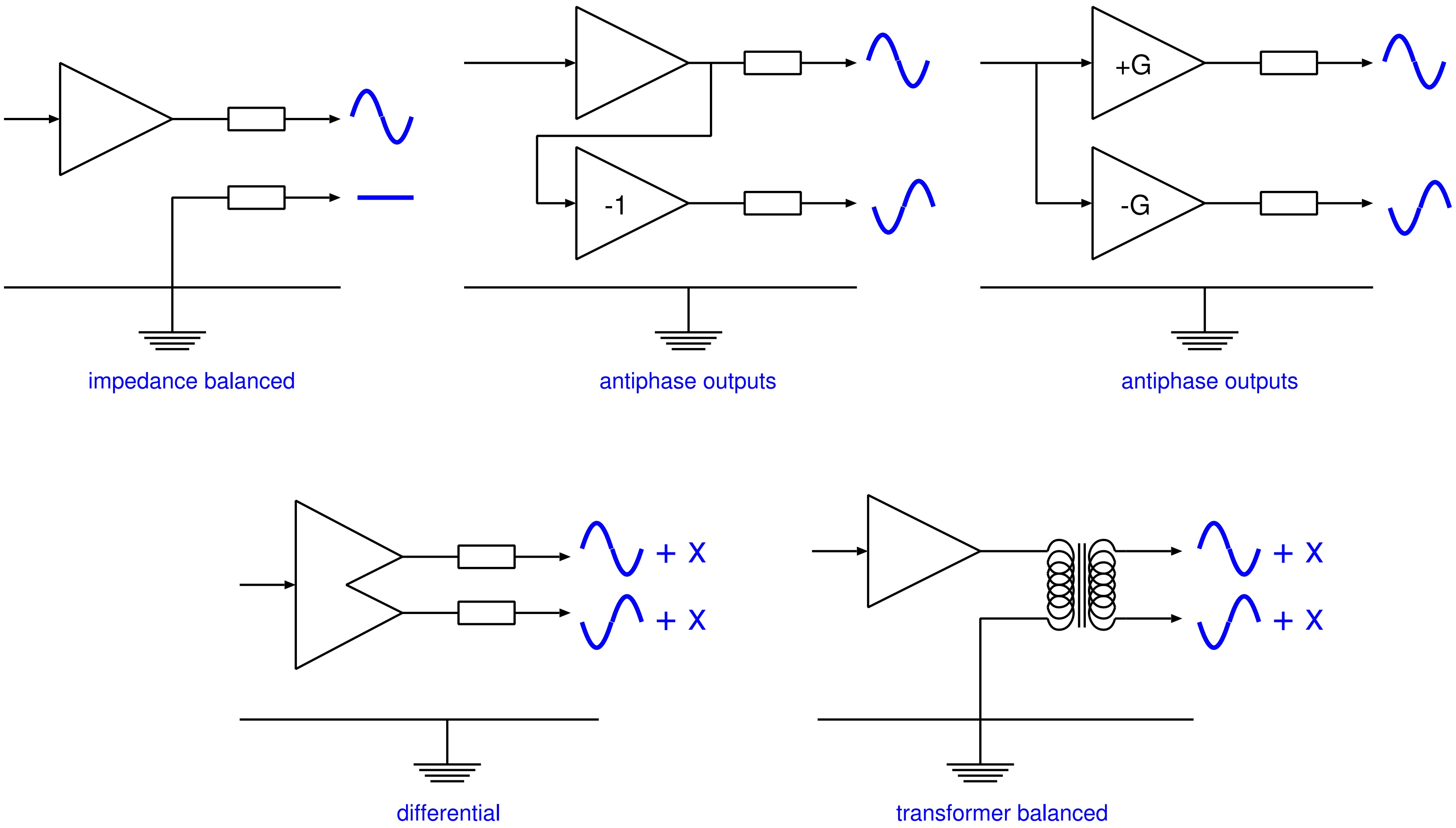
4 Analog signals

- Measured in Volts (V).
  * Rough levels: mics: 1 mV, consumer: 100 mV, pro: 1 V.
  * Current: $i = u/Z$ (A).
  * Power: $P = i * u = u^2/Z = i^2 * Z$ (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.

6 Balanced outputs



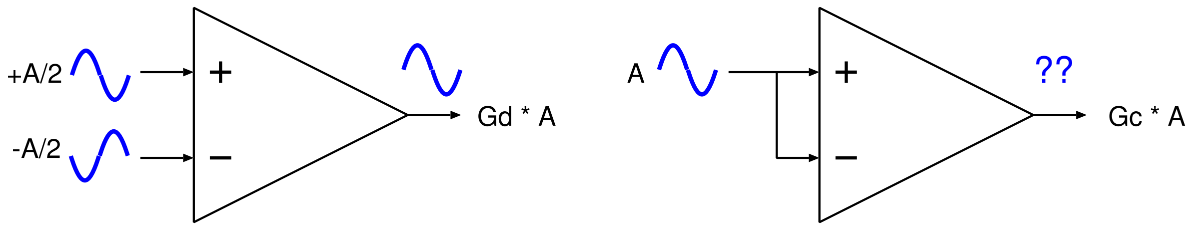
7 Balanced inputs : CMRR

- A 'perfect' differential input ignores the common signal.
- Real-life balanced inputs are not perfect.

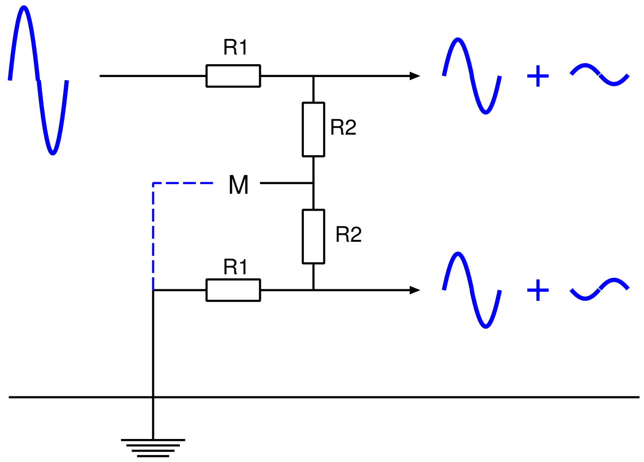
- The Common Mode Rejection Ratio indicates
  by how much the common signal is attenuated.
- CMRR is usually a function of frequency.
- Typical figures:
  * Cheap: 20...25 dB
  * Reasonable: 40 dB
  * Transformer balanced: 80...90 dB

- 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 = Gd / abs (Gc)



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

9 Audio level meters

- Defined by filter, detector, ballistics.

- Filter: flat, lowpass, A, C, ITU 468,. . .
- 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 meters

- 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, . . .)

11 Decibels

- 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.

Decibels – reference levels

- 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 * 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 $P = u * i = u^2/R$.
- For acoustic signals we have $v = p/Z$ and $I = p * v = p^2/Z$.
  * 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.

Acoustic levels – example 1

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 W / 113.1 m^2, or 8.842e-3 W / m^2.
- Now $I = p^2/Z$, or $p = \sqrt{I * Z}$.
- Hence the pressure is sqrt (8.842e-3 * 413.3) = 1.91 Pa.
- 1.91 Pa = 20 * log10 (1.91) + 94 = 99.6 dB SPL.

Acoustic levels – example 2

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 ?

- $I = p^2/Z = 1$ / 413.3 W / m^2.
- $P = I * S$ ($S =$ surface area) so
- Power = 5e-4 / 413.3 = 1.21e-6 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.

16 Noise

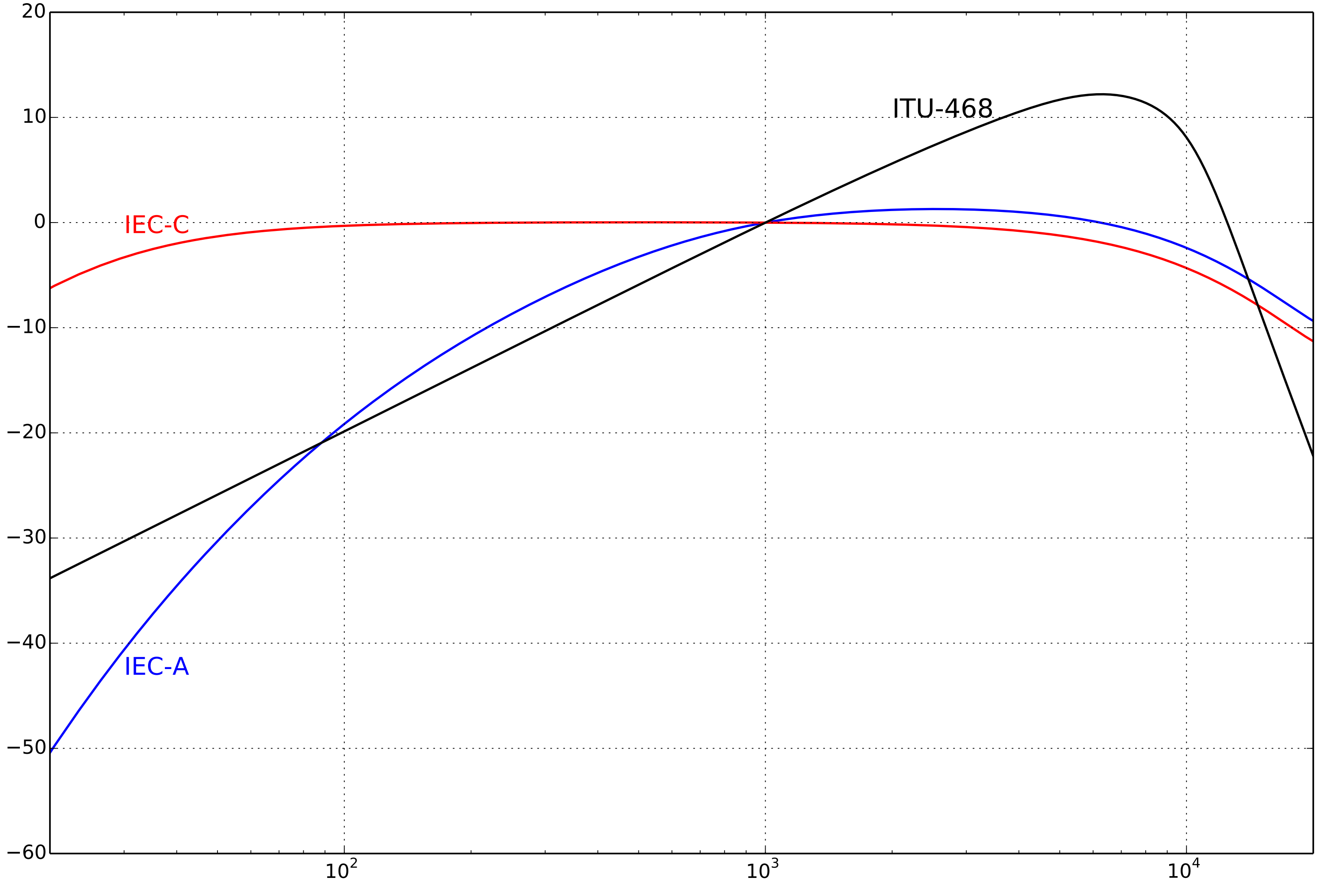
- A random or pseudo random signal.
- Probabilty distribution
  - Rectangular, Gaussian,...
  - We normally don't hear differences in distribution.
  - A sum of many independent random values will have a Gaussian distribution.
  - Filtered noise tends to have a Gaussian distribution.
- Density spectrum
  - Noise density N_0 = power per Hz.
  - At any particular frequency there is zero power.
  - White noise: constant density. $P = B * N_0$
  - 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.

17 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.

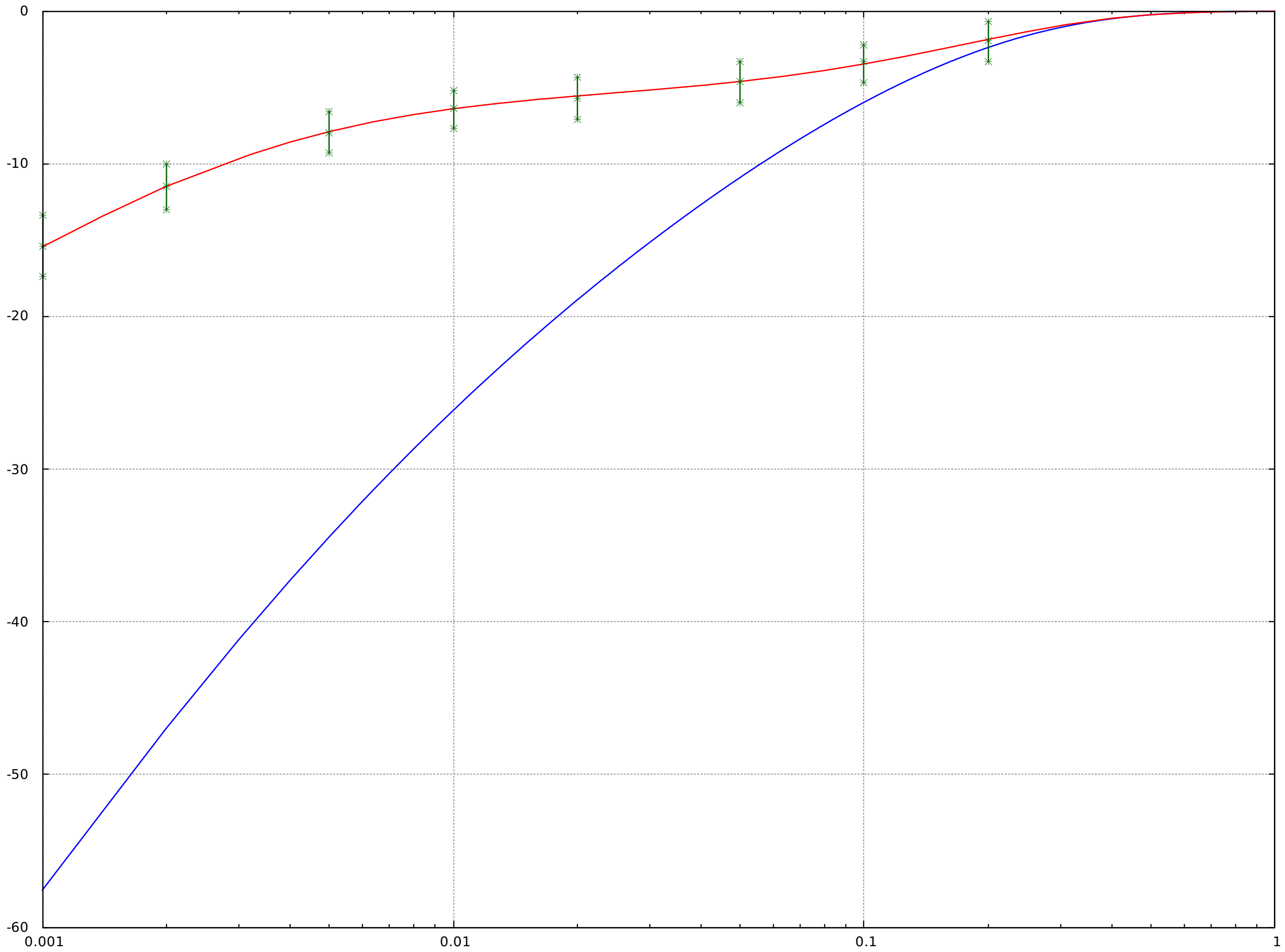
Standard noise weighting filters



19 ITU468

- 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.

21 Thermal noise

- 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 * 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 * 10^{-16} Watt
  - * RMS voltage v_n = 18 nV * $\sqrt{R}$
  - * 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

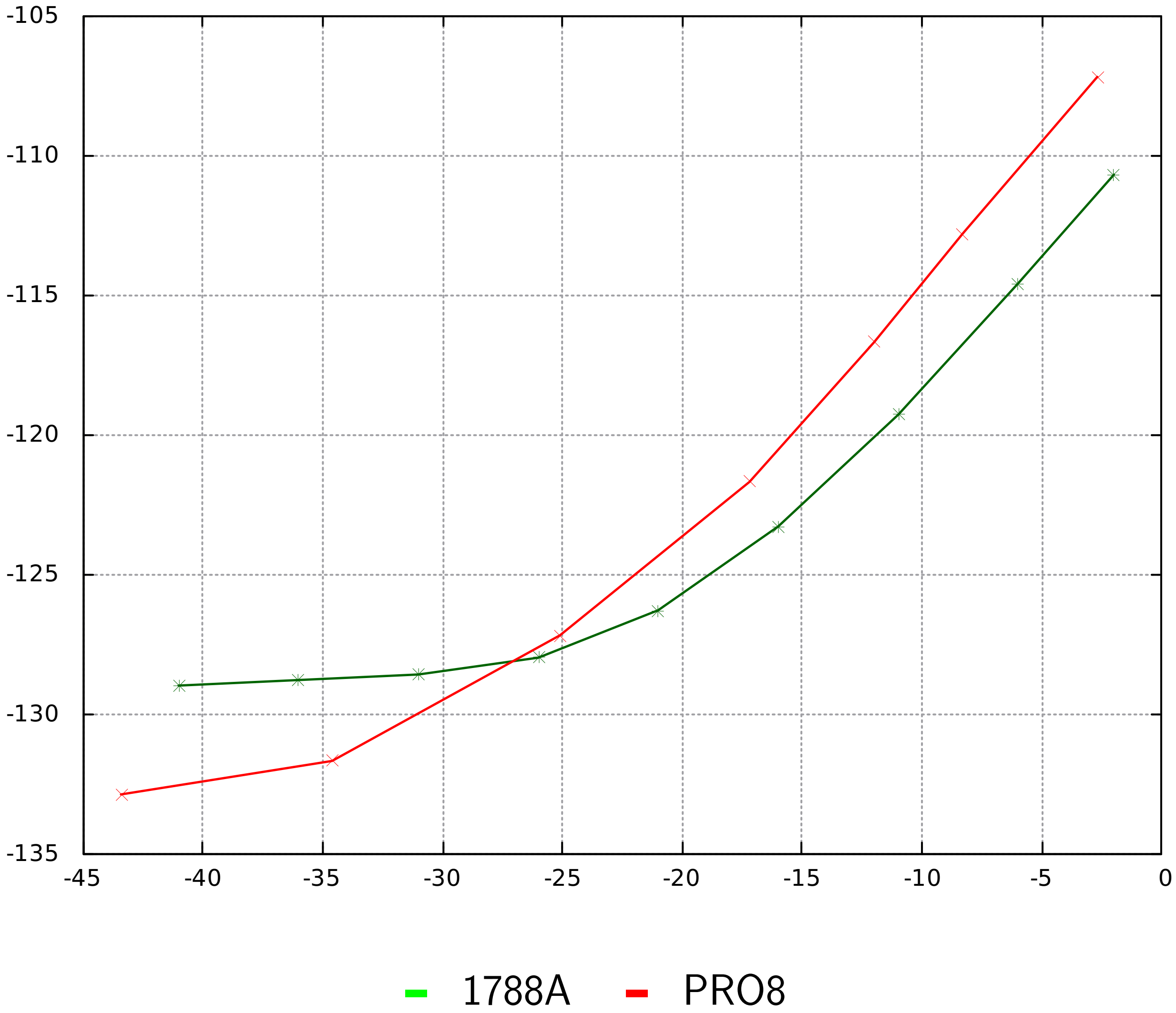
- All components of an electronic circuit generate thermal noise.
- It is always possible to model an amplifier or an AD converter as a 'perfect' noiseless one with a single noise source at the input.
- EIN = noise measured at the output / gain.
- The noise generated near the input will contribute most, 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).

EIN of microphone preamps

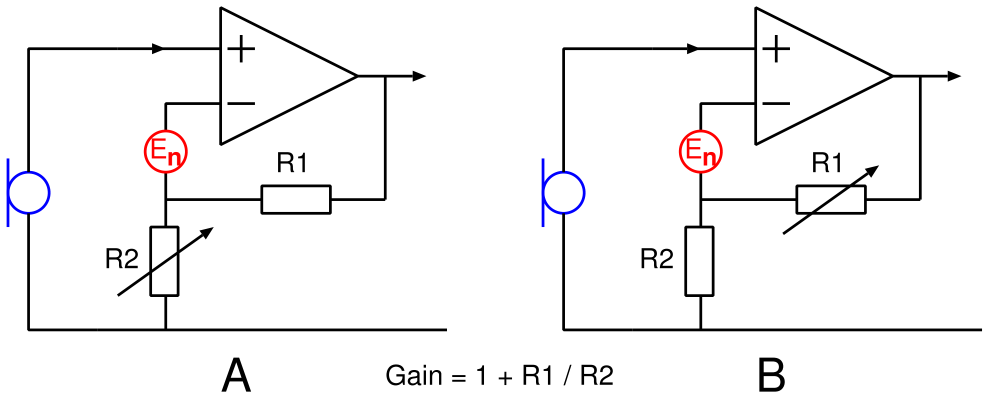
- The EIN of mic preamps is usually measured at maximum gain with a source impedance of 150 ohm.
- Some preamps can be a dB or so less noisy when measured with a lower impedance source or short-circuit.
- The EIN can be compared to the self-noise of the mic to find out which generates most noise.
- Some example values:
  note: 150 ohm generates -135 dBV(A)

Preamp	EIN dBV(A)
RME Micstasy	-131.2
Aphex 1788A	-129.0
Sony SPR-V110	-133.5
Behringer PRO8	-132.9
Edirol UA5	-122.2

EIN vs full scale input level



EIN vs gain: why



$$
\begin{aligned}
E_n &= \text{thermal noise due to R1} \parallel \text{R2} \\
&+ \text{amplifier input noise voltage} \\
&+ \text{amplifier input noise current } \times \text{R1} \parallel \text{R2}
\end{aligned}
$$

- Except at low gains, R2 $\ll$ R1 and R1$\|$R2 $\simeq$ R2.
- A: For lower gain R2 increases and E_n increases.
- B: R2 is fixed and E_n is almost constant.

Audio specs and unspecs

- 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.

Microphone specs

- 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 condensor mics may require some calculations.

Dynamic mics

- Sensitivity and impedance are specified. Self noise is thermal noise corresponding to impedance. Given sensitiviy, this can be converted to SPL.

- Example: Beyer M160: 1.0 mV/Pa, 200 ohm.

  1.0 mV = -60 dBV.

  Thermal noise is 18 nV * $\sqrt{200}$ = 254 nV = -131.9 dBV.

  Self noise is -131.9 + 60 - 94 dB = 22 dB = 20.1 dB(A).

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

  1.85 mV = -54.7 dBV

  Thermal noise is 18 nV * $\sqrt{300}$ = 312 nV = -130.1 dBV.

  Self noise is -130.1 + 54.7 - 94 dB = 18.5 dB = 16.6 dB(A).

- To use the full dynamic range a preamp with an EIN better than -130 dBV is required.

Condensor mics

- 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

- 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.

Other tools

- 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.

Q & A

Questions and answers, hands-on practice.

PyJacktools - LAC2014 Prerelease

- Some Python extensions and objects using Jack.

  AudioFile: read/write audio files into/from numoy 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.