note

CSClub March 22, 6pm. MC4059

CS Club Talk

"Black box testing of sound gear"

Richard Mann, Computer Science (mannr@uwaterloo.ca)

Date: Thu Mar 22, 2018, 6pm.
Location: MC4059

Recruiting

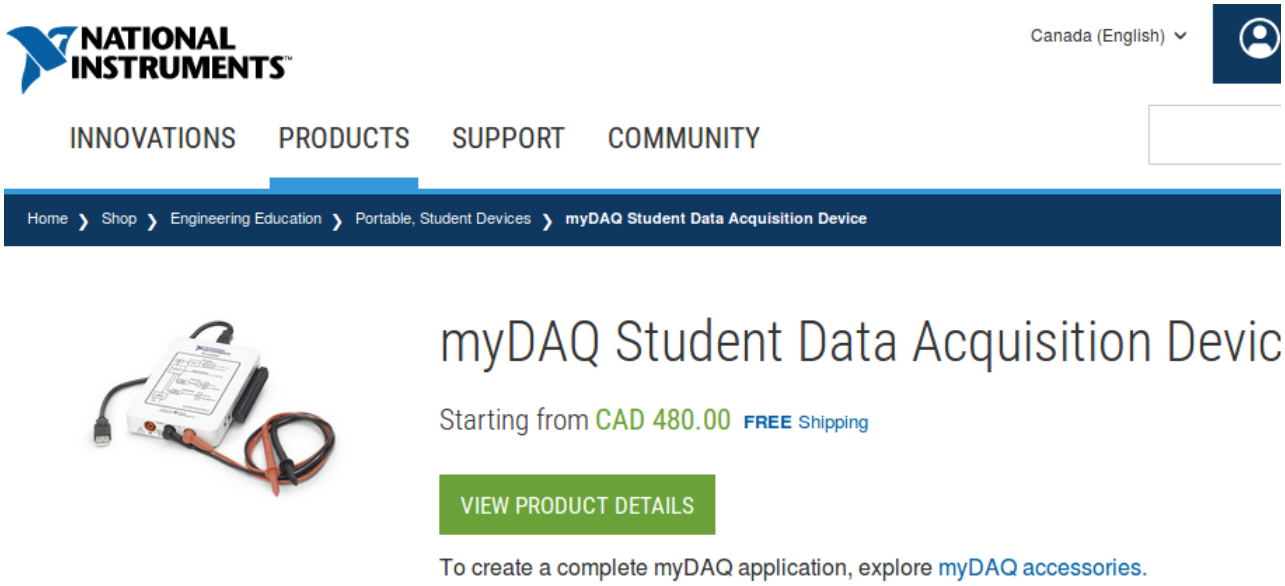
- Undergrad students CS498/698 W19. CS Advanced topics, Computer Audio. Project course, no exam.

- URA (part time): F18
  Interests: Raspberry Pi audio programming,

PART I: SOUND

SOUND DEMO

National Instrument myDAQ C$480.
Proprietary (Windows only) software.
Function generator, oscilloscope, spectrum analyzer



Frequency

Sound level (decibels)

Timbre (Pure tone vs square wave vs other forms)

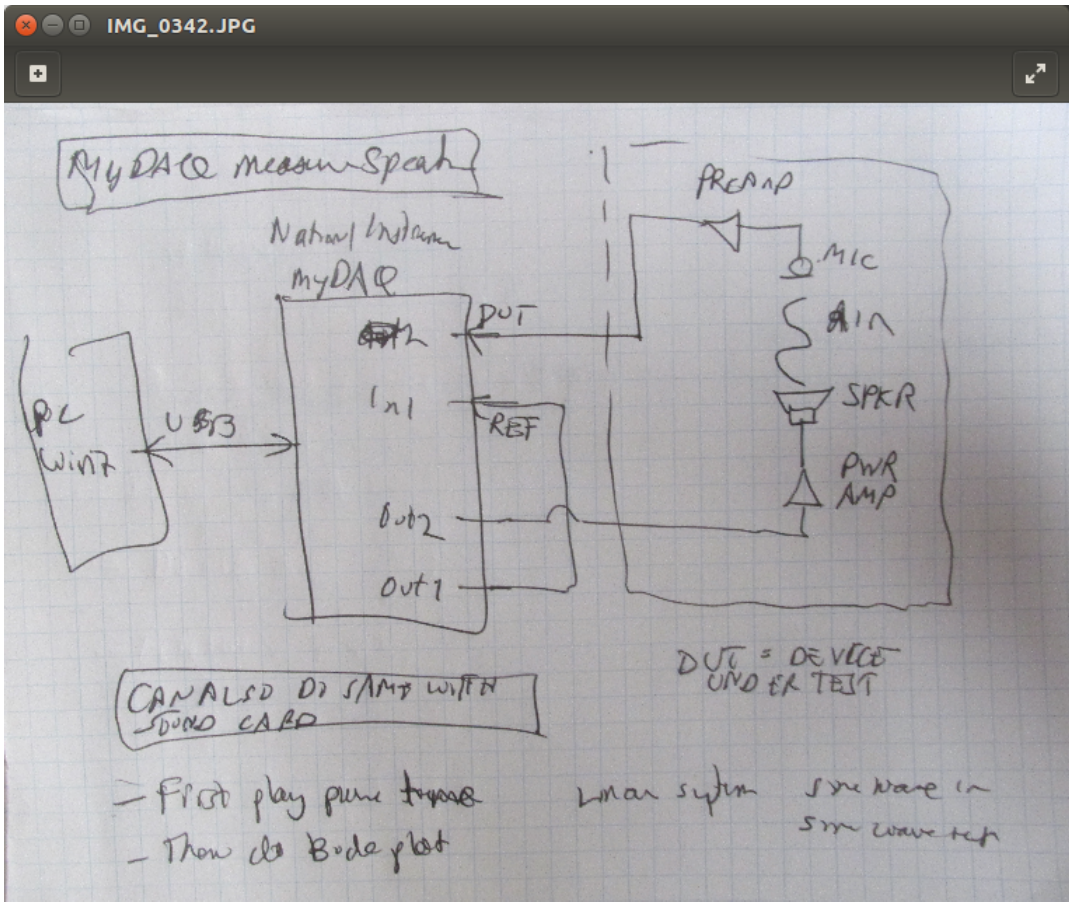
Low frequencies,
Perception reduced,

speaker output reduce, need subwoofers

High frequencies, ear more sensitive to 3kHz (smoke detector alarm), vs 1kHz (button beep of microwave).

Testing the sound system.
A linear system should produce a pure tone out when a pure tone is input.
Each tone will have a certain gain and time/phase shift.
This may be measured by a sequence of tones, a sweep or a random noise source.

DEMO: Measure speaker



Introduction to Sound

Sound pressure level, Pascal (Pa).

Air static pressure is 101.3kPa = 101,300 Pa, Local barometric pressure.

Air pressure depends on:

- local weather barometric pressure (xx to xx Pa)

- elevation changes (approx 12 Pa/ m)

- temperature

For now let's just take Air pressure as 100,000 Pa unless otherwise noted.

Sound waves: local variations that "ride on top" of local pressure.
<INSERT FIGURE>

The figure above shows a signal going from +/- 1 Pa at the rate of twenty cycles per second.

1 Pa corresponds to 94 dB (Decibels), a high sound level.

The over pressure p is very small, 1/100,000 of the atmospheric pressure!

Converting Pascals to Decibels

Measuring Sound levels (decibels)

Decibels (dB) is a logarithmic scale.ie., adding in decibels corresponds to multiplying in pressure.

Sound Pressure Level (SPL) in DB = 20 * log10 (P/Pref)

P: pressure
Pref: reference value. 20 x 10^-6. 20 micro pascals.

20 dB increase/decrease --> x 10 increase/decrease in pressure
40 dB --> x 100
60 dB --> x 1000
...

Also, convenient rule,
+/- 6 dB increase/decrease. --> 2 x increase/decrease in pressure

Some typical Sound Pressure Levels in air

Sources at 1 m	Sound Pressure	SPL re 20 µPa
Rifle	200 Pa	140 dB
Threshold of pain	20 Pa	120 dB
Pneumatic hammer	2 Pa	100 dB
	1 Pa	**94 dB**
Street traffic	0.2 Pa	80dB
Talking	0.02 Pa	60 dB
Library	0.002 Pa	40 dB
TV Studio	0.0002 Pa	20 dB
Threshold of hearing	0.00002 Pa	0 dB

Question: what happens higher pressures:
20,000 Pa = 180dB
200,000 Pa = 200dB --> Sonic boom.

Waves and the speed of sound

$\lambda = cf$

λ: wavelength, meters (m),

f: frequency Hertz (Hz, aka cycles per second cps)

c: speed of sound, approx 343 m/s @ 20C, (1,125 ft/s; 1,235 km/h; 767 mph)

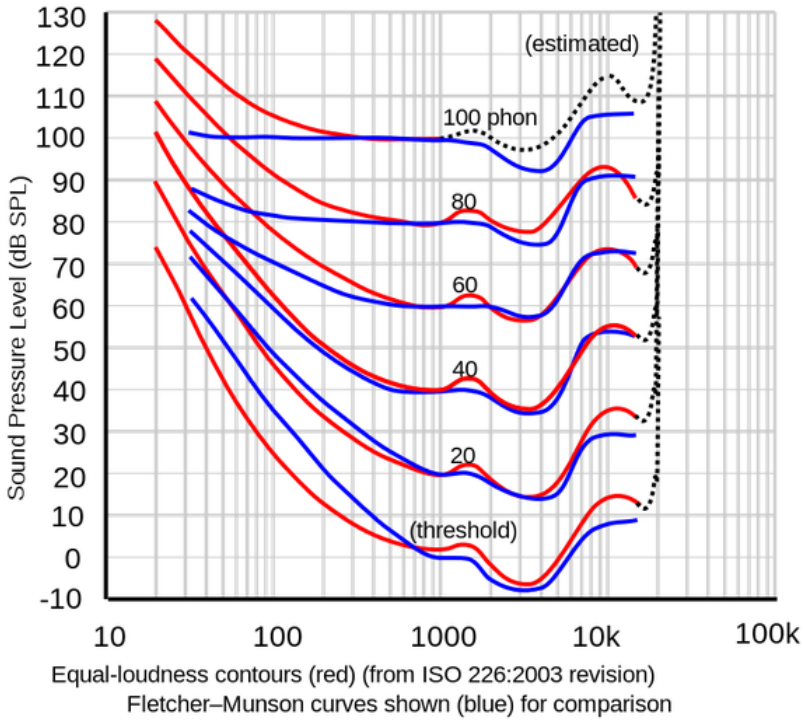
A 20 Hz wave has a wavelength of 343/20 = 17.5m

A 1000 Hz wave has a wavelength of 343/1000 = 0.343m (1.125 ft).

Human Hearing

https://en.wikipedia.org/wiki/Fletcher%E2%80%93Munson_curves

The **Fletcher–Munson curves** are one of many sets of equal-loudness contours for the human ear, determined experimentally by Harvey Fletcher and Wilden A. Munson, and reported in a 1933 paper entitled "Loudness, its definition, measurement and calculation" in the *Journal of the Acoustic Society of America*.[1]

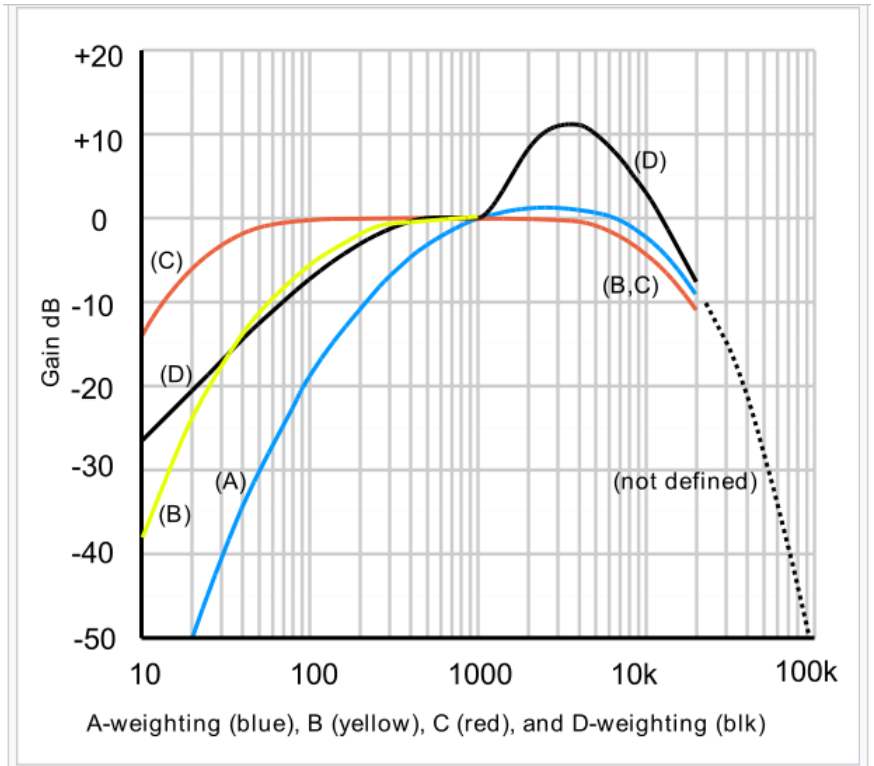


Contents [hide]

1 Background

Draw regions of Voice, Music in this space.

https://en.wikipedia.org/wiki/A-weighting



A graph of the A-, B-, C- and D-weightings across the frequency range 10 Hz – 20 kHz

These curves are approximated by so-called "A-weighting" of frequencies. This significantly down weights sound at low frequencies.
The designation dB(A) stands for decibels, A weighted. Measurements specifying dB alone is usually assumed to be A weighted.
There are other weightings, such as dB(C) which still reduce low frequencies, but less so than A weighting.

*** NOTE *** Infra sound refers to frequencies below the lower range of human hearing, usually taken to be 20Hz. The exact lower cut off depends, but can be as lo
0.1 Hz. In this case special equipment is needed to measure this sound, and an unweighted measure should be taken.

PART II: SOUND MEASUREMENT

"Black Box Testing" : General Systems

Stimulus -----> System
Hardware, Software, -----> Response

Specification
(Mathematical model) -----> Predicted response

"Black Box Testing": Audio systems

Unlike general-purpose computer systems, there is often a fixed pallet of (open source/ proprietary) audio effects.
Effects may be analog, but most are now emulated in DSP (Digital signal processing).
System may be open source or proprietary.

Typical audio effects:
- sample rate conversion (eg., for aligning audio/ video)
- frequency equalization
- compression
- echo/ reverb/ room simulation
- headphone vs speaker vs surround sound listening
- fuzz/ overload/ distortion
- etc

Audio Testing techniques:

Test with pure tones, stepped tones, sweeps
Also noise sources
- random noise
- pseudo random noise

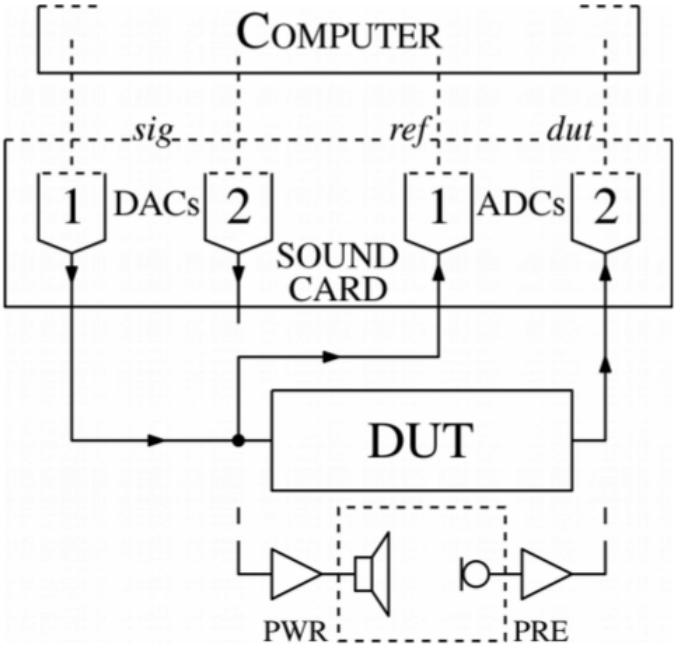
We will use standard USB Audio Hardware (not specialized DAQ equipment)
Open Source software.
Octave programming language (Open source Matlab(TM) work alike)

Open Source Software for Electrical and Acoustic measurement.

Joint work with John Vanderkooy (Physics), Audio Research Group.

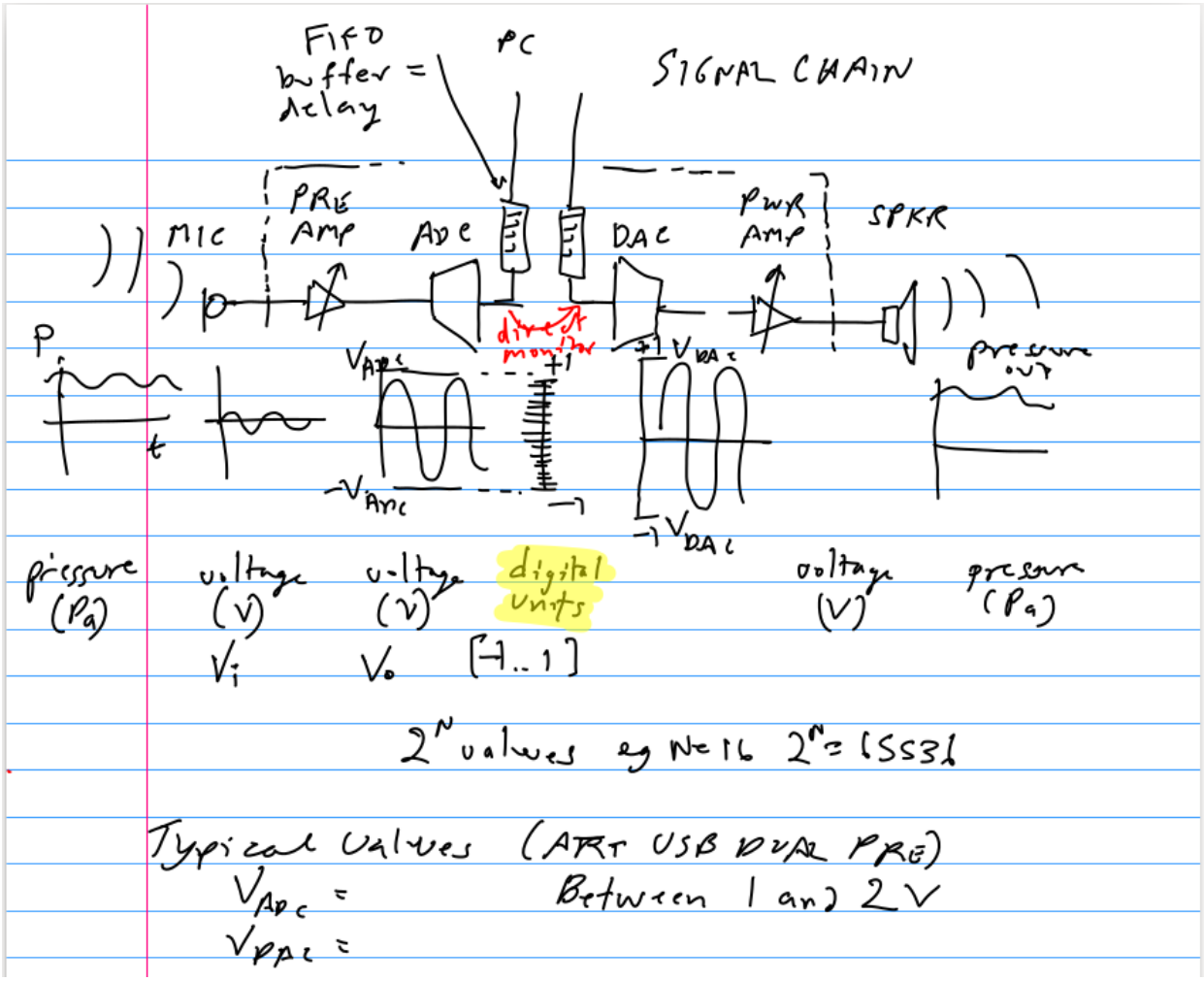
- John Vanderkooy and Richard Mann, An Open-Source Electroacoustic Measurement System Theory, Practicalities & Acoustic Examples. Linear Audio, Volume 13, 1 April 2017. LinearAu Paper (PDF) M-files & readme for part1
- Richard Mann and John Vanderkooy, An Open-Source Electroacoustic Measurement System Sound Card Setup, System Characterization and a few more Examples. Linear Audio, Volume 2017. LinearAudio,net. Paper (PDF) M-files & readme for part 2.

CONNECTION



By convention, REF is Left channel, DUT is Right channel.

Detail of USB DUAL PRE Interface



SOFTWARE SYSTEM OPERATION

Create test stimulus
Loop
  turn on recorder,
   play stimulus
  end recorder

  compare DUT signal (through device) to REF signal (loopback)
  compute Transfer function (or other properties)

  update plots
Forever

PROBLEM: USB devices have uncertain delay, anywhere from 0.05 to 0.25 seconds!

SOLUTION: Output periodic test signal of Length N. Then we don't need to worry about time shift!

Points N (N-1)	f_s=44100	f_s=48000
8192 (8191)	0.19 s	0.17 s
16384 (16383)	0.37 s	0.34 s
32768 (32767)	0.74 s	0.68 s
65536 (65535)	1.49 s	1.37 s

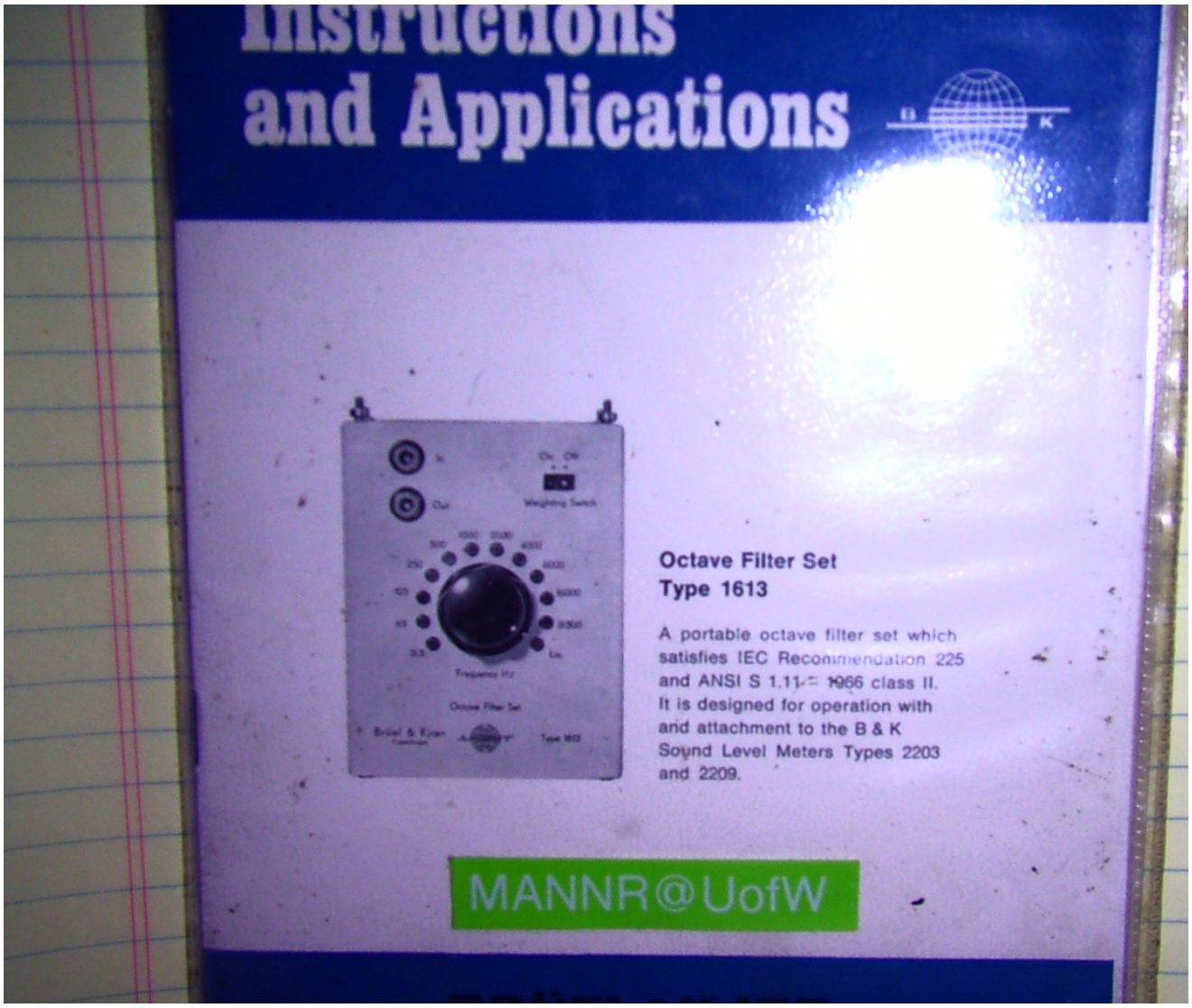
Case 1: Sound card itself

"Loop back mode".
Left output --> Left input
Right output --> Right input

Verify passage of signal (without distortion)
Verify frequency response of system.
Verify no extra signals (artifacts) interfere with measurement

Case 2: Analog Octave filter bank

- Bruel Kjaer Model 1613 Octave filter bank


Instructions
and Applications
Octave Filter Set
Type 1613
A portable octave filter set which satisfies IEC Recommendation 225 and ANSI S 1.11-1966 class II. It is designed for operation with and attachment to the B & K Sound Level Meters Types 2203 and 2209.
MANNR@UofW

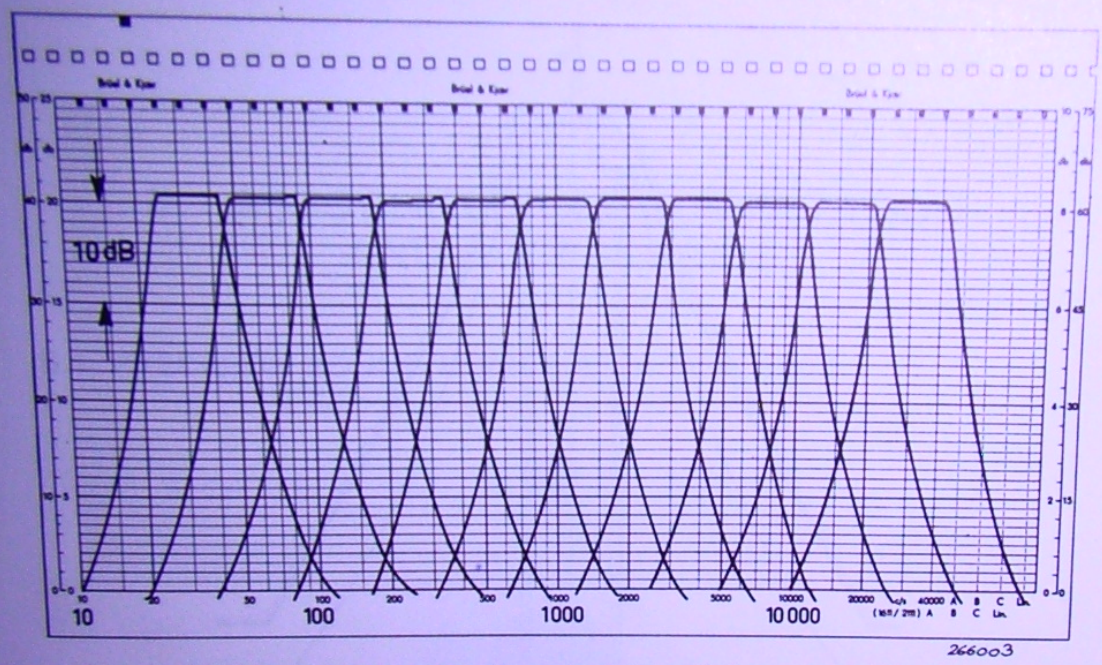
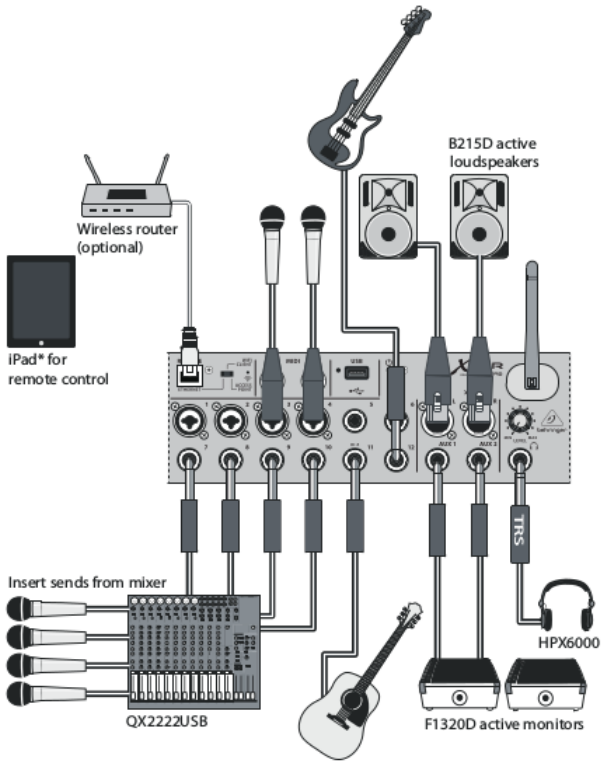


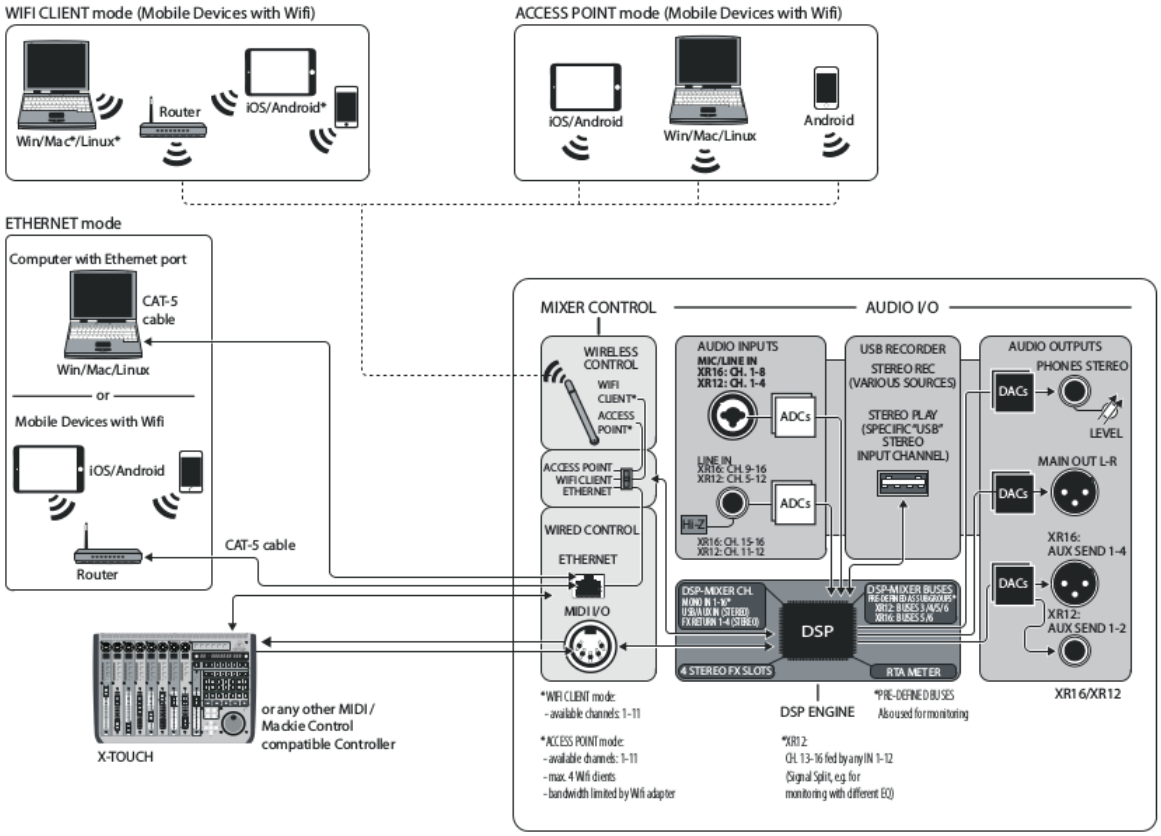
Fig. 4.2. Frequency characteristics of the octave filters

Each filter conforms to the requirements of ANSI S 1.11 — 1966 for octave band filters class II and IEC Recommendation 225. Fig. 4.3 shows the typical response of an octave filter of the 1613 compared with the limits set by ANSI and IEC. Uniformity in the pass band is within ± 0.5 dB. Attenuation at one, two and three octaves from centre frequency is approximately 25, 50 and 68 dB respectively.

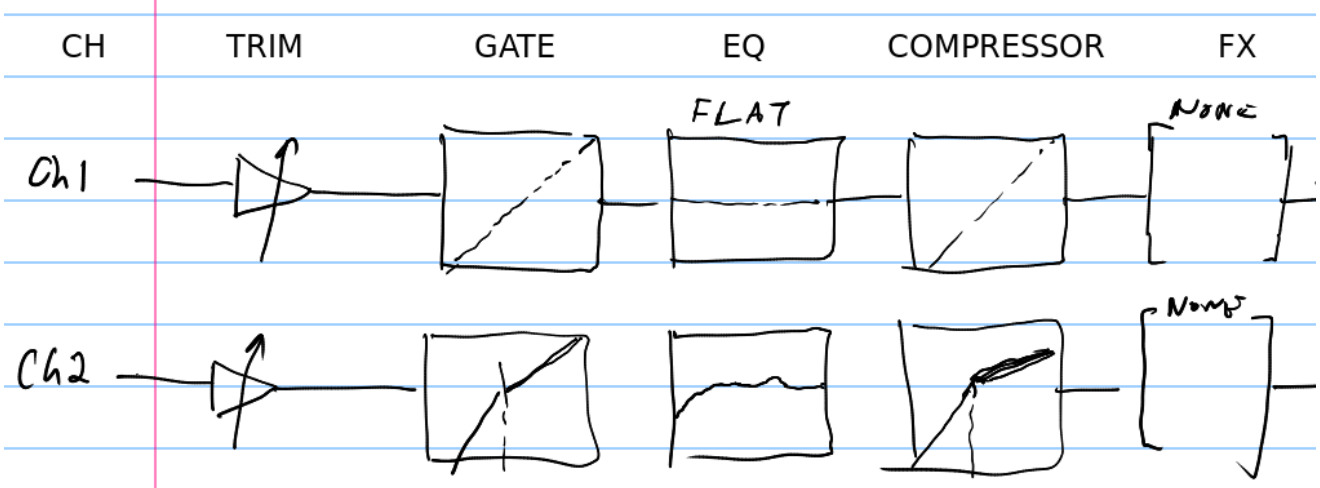
Case 3: Behringer X AIR Mixer

Analog electronics


B215D active loudspeakers
Wireless router (optional)
iPad* for remote control
Insert sends from mixer
QX2222USB
TRS
HPX6000
F1320D active monitors


WIFI CLIENT mode (Mobile Devices with Wifi)
Win/Mac*/Linux*
Router
iOS/Android*
ACCESS POINT mode (Mobile Devices with Wifi)
iOS/Android
Win/Mac/Linux
Android
ETHERNET mode
Computer with Ethernet port
CAT-5 cable
Win/Mac/Linux
or
Mobile Devices with Wifi
iOS/Android
Router
CAT-5 cable
MIXER CONTROL
AUDIO I/O
WIRELESS CONTROL
WIFI CLIENT*
ACCESS POINT*
ACCESS POINT
WIFI CLIENT
ETHERNET
WIRED CONTROL
ETHERNET
MIDI I/O
AUDIO INPUTS
MIC/LINE IN
XR16: CH. 1-8
XR12: CH. 1-4
ADCs
LINE IN
XR16: CH. 9-16
XR12: CH. 5-12
ADCs
HI-Z
XR16: CH. 15-16
XR12: CH. 11-12
USB RECORDER
STEREO REC (VARIOUS SOURCES)
STEREO PLAY (SPECIFIC "USB" STEREO INPUT CHANNEL)
AUDIO OUTPUTS
DACs
PHONES STEREO
LEVEL
MAIN OUT L-R
XR16: AUX SEND 1-4
XR12: AUX SEND 1-2
DSP-MIXER CH.
MONO IN 1-X*
USB/AUX IN (STEREO)
FX RETURN 1-4 (STEREO)
DSP
DSP-MIXER BUSES
PRE-DEFINED AS SUBGROUPS*
XR12: BUSES 3/4/5/6
XR16: BUSES 5/6
4 STEREO FX SLOTS
RTA METER
DSP ENGINE
*PRE-DEFINED BUSES
Also used for monitoring
XR16/XR12
*WIFI CLIENT mode:
- available channels: 1-11
*ACCESS POINT mode:
- available channels: 1-11
- max. 4 Wifi clients
- bandwidth limited by Wifi adapter
*XR12:
CH. 13-16 fed by any IN 1-12
(Signal Split, e.g. for monitoring with different EQ)
X-TOUCH
or any other MIDI / Mackie Control compatible Controller

X Air Channel

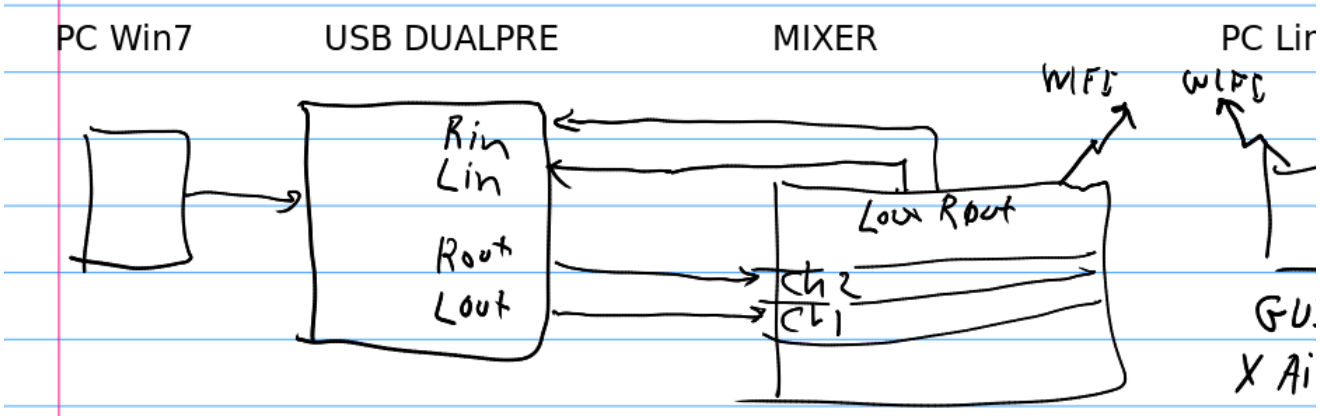


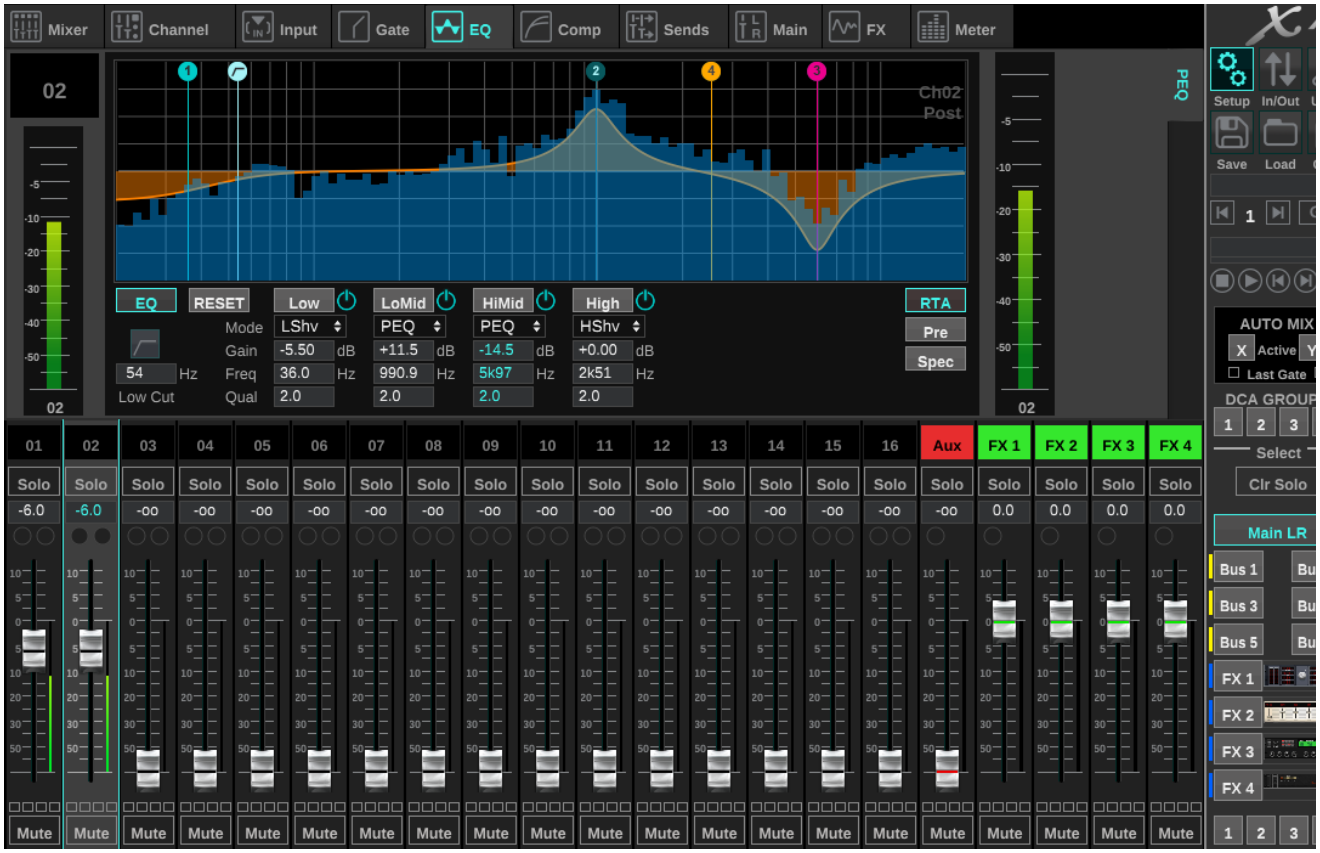
TRIM: Microphone gain, Level adjustment for line inputs
GATE (aka EXPANDER): increases input level if needed
EQ: Frequency equalization
COMPRESSOR (aka LIMITER): limits output level if needed.

Experimental Setup

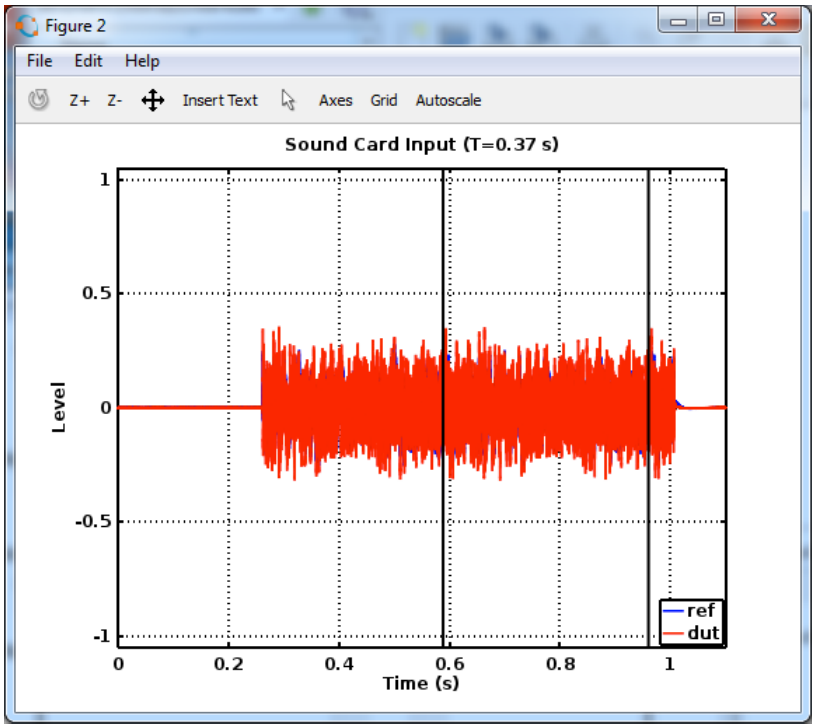
Compare Ch1 (no effects)
Ch2 (effects)

Various stimuli (noise, tones, music notes) to test channel response.

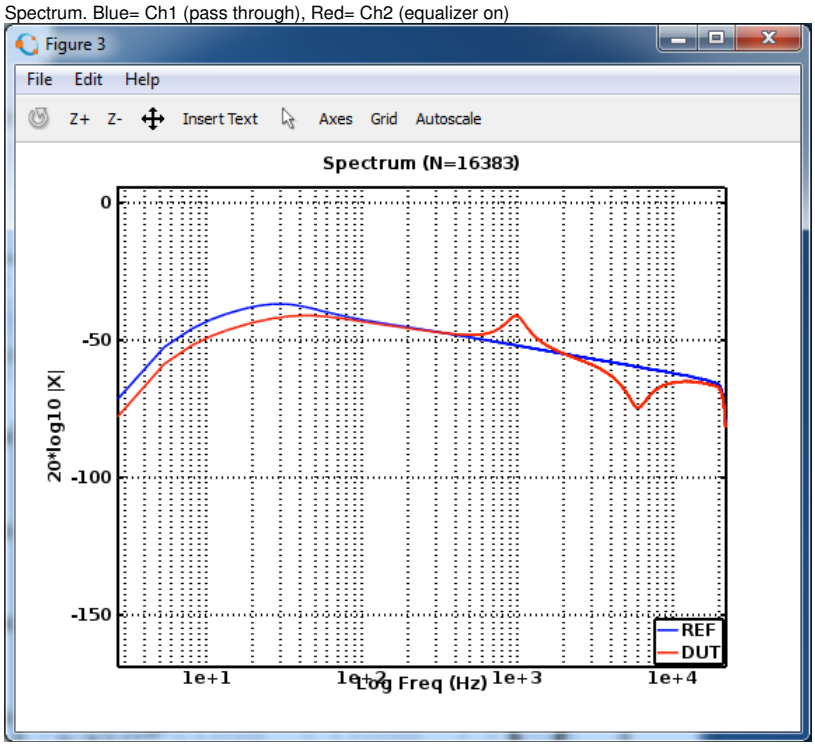




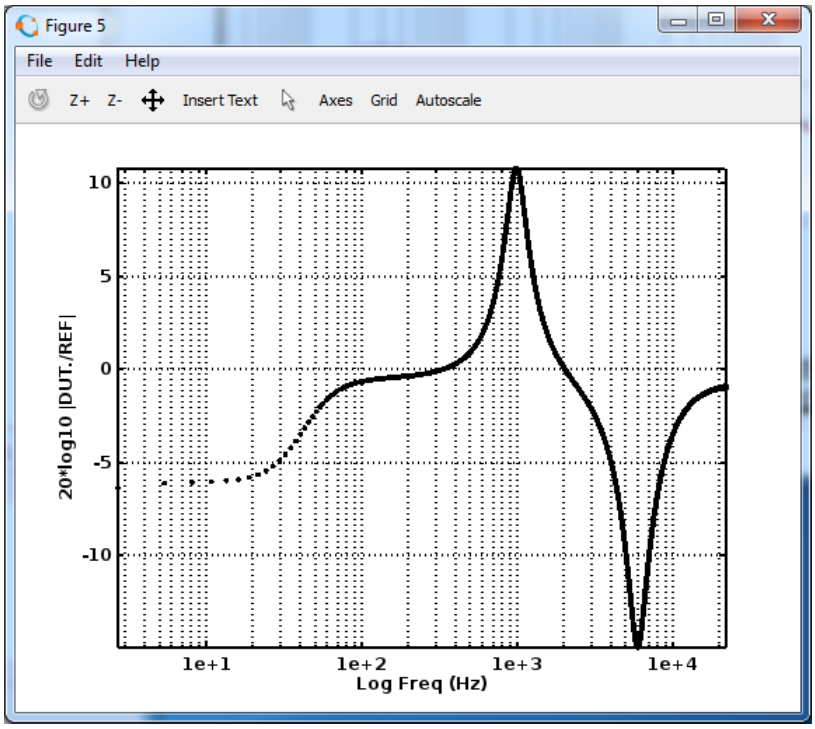
Stimulus signal. Pink MLS noise, 44.1kHz, 16383 samples.



Spectrum. Blue= Ch1 (pass through), Red= Ch2 (equalizer on)



Transfer function,
Ratio of spectrum of Ch2/Ch1.
Note similarity to EQ curve on mixer.



DEMO 2. Compression.

Compressor

HearingAidAnalogCompressor.pdf

COMPRESSION CIRCUIT OF A MULTIBAND ANALOG SYSTEM HEARING AID

J. Fernández Ramos, J.C. Tejero, J.A. Hidalgo, M. J. Martin, A. Gago

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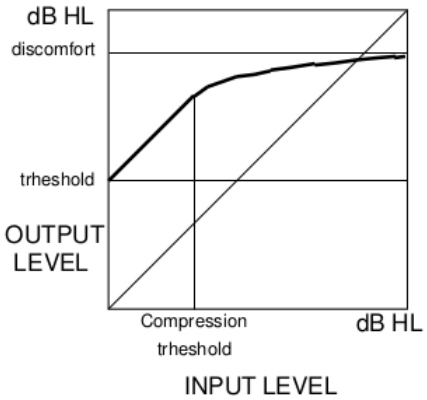


Figure 1: Nonlinear loudness compensation function.

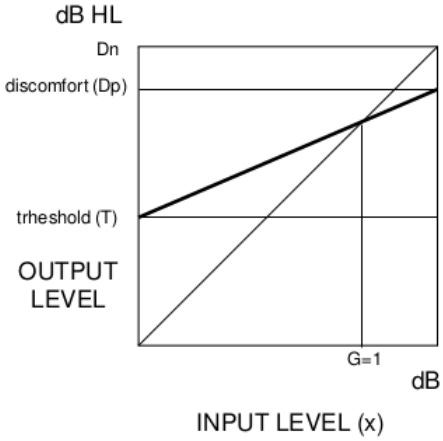
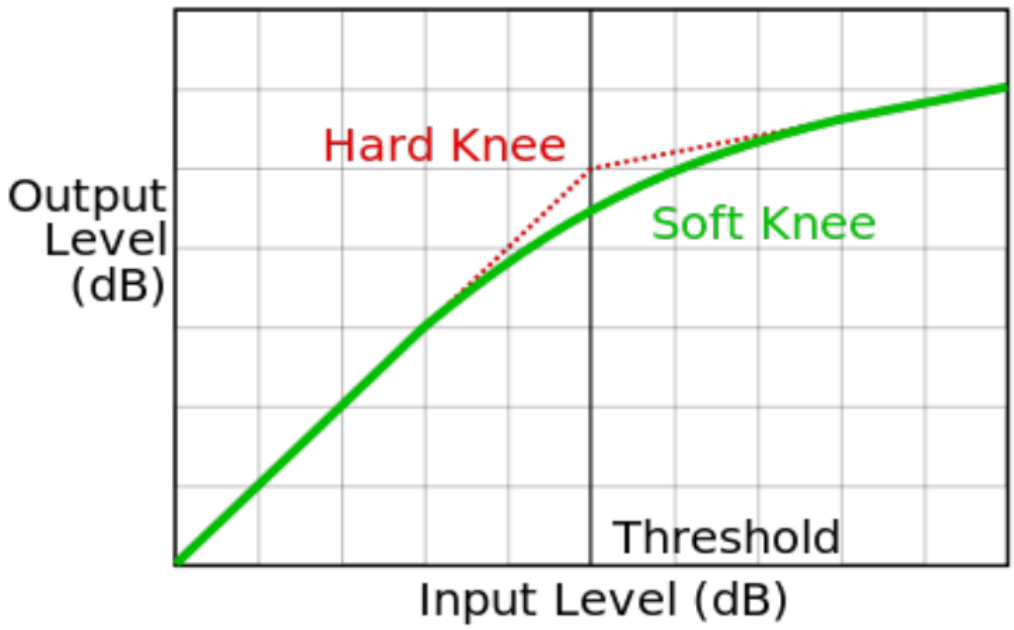


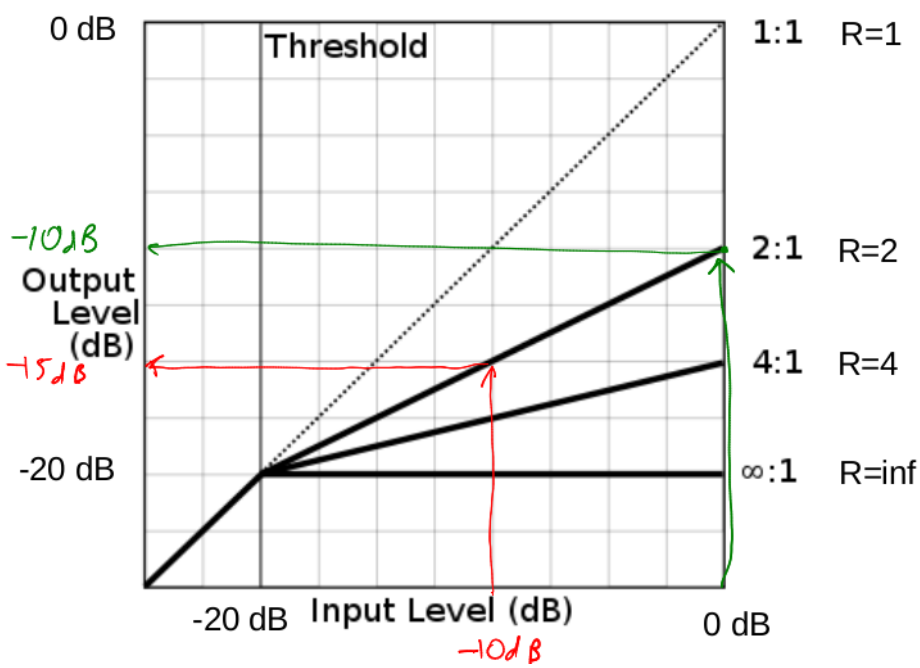
Figure 2: Linear loudness compensation f

Compressor. "hard vs soft"
(https://en.wikipedia.org/wiki/Dynamic_range_compression)



Compressor Notation

Compressor Ratio (R:1)
(https://en.wikipedia.org/wiki/Dynamic_range_compression)



Example:
T: -20 dB
R: 2

Input. I = -10 dB

Output = T+ (I - T)/R
= -20 + (-10 -
= -20 + 10/2
= -15 dB

In other words, for ev
decibel input signal is
we get gain 1/2 decib

When I = 0 dB (maxi
Output = T -T/R = -2
= -1

other

Updated Ju

followup discussions *for lingering questions and comments*