CS Club Talk

"Black box testing of sound gear"

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

Recruiting

- Undergrad students CS 498/698 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 $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,
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.

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, i.e., adding in decibels corresponds to multiplying in pressure.

Sound Pressure Level (SPL) in DB = $20 \times \log_{10} \left( \frac{P}{P_{ref}} \right)$

- $P$: pressure
- $P_{ref}$: reference value. $20 \times 10^{-6} = 20$ micro pascals.

$20$ dB increase/decrease $\rightarrow x 10$ increase/decrease in pressure

$40$ dB $\rightarrow x 100$

$60$ dB $\rightarrow x 1000$

... 

Also, convenient rule,

$+/- 6$ dB increase/decrease $\rightarrow 2 x$ increase/decrease in pressure

#### Some typical Sound Pressure Levels in air

<table>
<thead>
<tr>
<th>Sources at 1 m</th>
<th>Sound Pressure SPL re 20 $\mu$Pa</th>
</tr>
</thead>
<tbody>
<tr>
<td>Rifle</td>
<td>200 Pa</td>
</tr>
<tr>
<td>Threshold of pain</td>
<td>20 Pa</td>
</tr>
<tr>
<td>Pneumatic hammer</td>
<td>2 Pa</td>
</tr>
<tr>
<td>1 Pa</td>
<td></td>
</tr>
<tr>
<td>Street traffic</td>
<td>0.2 Pa</td>
</tr>
<tr>
<td>Talking</td>
<td>0.02 Pa</td>
</tr>
<tr>
<td>Library</td>
<td>0.002 Pa</td>
</tr>
<tr>
<td>TV Studio</td>
<td>0.0002 Pa</td>
</tr>
<tr>
<td>Threshold of hearing</td>
<td>0.00002 Pa</td>
</tr>
</tbody>
</table>

Question: what happens higher pressures:

- $20,000$ Pa = $180$dB
- $200,000$ Pa = $200$dB $\rightarrow$ Sonic boom.

### Waves and the speed of sound

$\lambda = cf$

- $\lambda$: wavelength, meters (m),
- $f$: frequency Hertz (Hz, aka cycles per second cps)
- $c$: speed of sound, approx $343$ m/s @ $20^\circ C$, ($1,125$ ft/s; $1,235$ km/h; $767$ mph)
A 20 Hz wave has a wavelength of $\frac{343}{20} = 17.5\text{m}$.

A 1000 Hz wave has a wavelength of $\frac{343}{1000} = 0.343\text{m}$ (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*. \(^1\)

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**Contents** [hide]

1 **Background**

Equal-loudness contours (red) (from ISO 226:2003 revision) Fletcher–Munson curves shown (blue) for comparison

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


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!
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

<table>
<thead>
<tr>
<th>Points N (N-1)</th>
<th>f_s=44100</th>
<th>f_s=48000</th>
</tr>
</thead>
<tbody>
<tr>
<td>8192 (8191)</td>
<td>0.19 s</td>
<td>0.17 s</td>
</tr>
<tr>
<td>16384 (16383)</td>
<td>0.37 s</td>
<td>0.34 s</td>
</tr>
<tr>
<td>32768 (32767)</td>
<td>0.74 s</td>
<td>0.68 s</td>
</tr>
<tr>
<td>65536 (65535)</td>
<td>1.49 s</td>
<td>1.37 s</td>
</tr>
</tbody>
</table>
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.
Quick Start Guide

X AIR XR16/XR12
16/12-Input Digital Mixer for iPad/Android Tablets with
8/4 Programmable MIDAS Preamps, 8 Line Inputs,
Integrated WiFi Module and USB Stereo Recorder
X Air Channel

<table>
<thead>
<tr>
<th>CH</th>
<th>TRIM</th>
<th>GATE</th>
<th>EQ</th>
<th>COMPRESSOR</th>
<th>FX</th>
</tr>
</thead>
<tbody>
<tr>
<td>Ch1</td>
<td></td>
<td>FLAT</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Ch2</td>
<td></td>
<td></td>
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<td></td>
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</table>

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.

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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.
COMPRESSION CIRCUIT OF A MULTIBAND ANALOG SYSTEM HEARING AID


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Figure 1: Nonlinear loudness compensation function.

Figure 2: Linear loudness compensation function.
Compressor. "hard vs soft"

Example:
T: -20 dB
R: 2

Input, \( I = -10 \) dB

Output = \( T + (I - T)/R \)
= \(-20 + (-10 -1)\)
= \(-20 + 10/2\)
= \(-15\) dB

In other words, for every decibel input signal, we get gain 1/2 decibels.

When \( I = 0 \) dB (maxi
Output = \( T - T/R = -2 \)
= \(-1\) dB
followup discussions for lingering questions and comments