

[Instr Note] CS Club Talk. Tues Jul 17, 2018.

Subject: [Instr Note] CS Club Talk. Tues Jul 17, 2018.
From: "CS 489, SECTION 001 on Piazza" <no-reply@piazza.com>
Date: 07/17/2018 11:57 PM
To: mann@uwaterloo.ca

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Instructor Richard Mann posted a new private Note.

CS Club Talk. Tues Jul 17, 2018.

CS Club Talk

"Computer Sound: From Microphones to Loudspeakers and Everything In Between"

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

Date: Tues July 17, 2018. 6pm.

Location: MC 4041

Recruiting

- Undergrad students [CS498/698 W19](#). CS Advanced topics, Computer Audio. Project course, no exam.

The course will be 50% assignments, 50% project.

- URA (Part time only) for F18/W19. Linux audio programming, Arduino and Raspberry Pi.

MORE ON THE COURSE

Audience

- 3B or later (CS370, 371, and/or Matlab/Octave experience)

- sound/ music/ math/ electronics interests

- students who want to do independent project and get course credit

- open to Math/ Physics/ Engineering with similar interests

- all necessary material will be developed in class, lectures + demo code

GRADING:

- Lectures with term work, 50%
- Term work: Experiments, homework exercises,
in the language of your choice Matlab/Octave, Python, C, etc..
- Student Project 50% Alone or in groups Option to present your work. NO FINAL :)

SCHEDULE.

Project topic determined/ negotiated during term,

P0 (one page proposal, week 6),

P1 (three pages, algorithm/ data ready, week 7)

Pfinal (ten to twenty pages, including figures, EOT)

SOME PAST PROJECTS:

Music synthesis (SW modeling of old school analog synths)
Automatic pitch detection (eg., guitar tuner, note transposer)
Analysis of bird calls ("bio acoustics")
Speech processing/ speech recognition
Music indexing (Shazam algorithm)
Recording and frequency analysis of (student's own) singing voice.
Building, testing and evaluating a new plug in for SW package X.
Time-frequency shifting of audio ("vocoder/ auto tune effects")
Realtime Audio effect programming on Arduino
Design/ measure your own loudspeakers

TRY DOWNLOADING THIS NOW:

"Elevator Pitch", Student project, a video game App for voice / singing training. Check it out! App Store: <https://itunes.apple.com/app/id1370102089>, Play Store: <https://play.google.com/store/apps/details?id=com.waznop.ElevatorPitch>

This is supposed to work with voice, but should also work with any instrument. Try it out!

SOUND: INTRODUCTION

AUDIO TONE DEMO

Audacity -> USB DUALPRE --> AUDIO AMPLIFIER (Digital Audio) --> SPEAKER

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).

SOUND WAVES

$$\lambda = cf$$

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 $343/20 = 17.5\text{m}$

A 1000 Hz wave has a wavelength of $343/1000 = 0.343\text{m}$ (1.125 ft).

SOUND: MEASUREMENT

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 = $20 * \log_{10} (P/P_{ref})$

P: pressure

Pref: reference value. 20×10^{-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 μ Pa
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

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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 ???

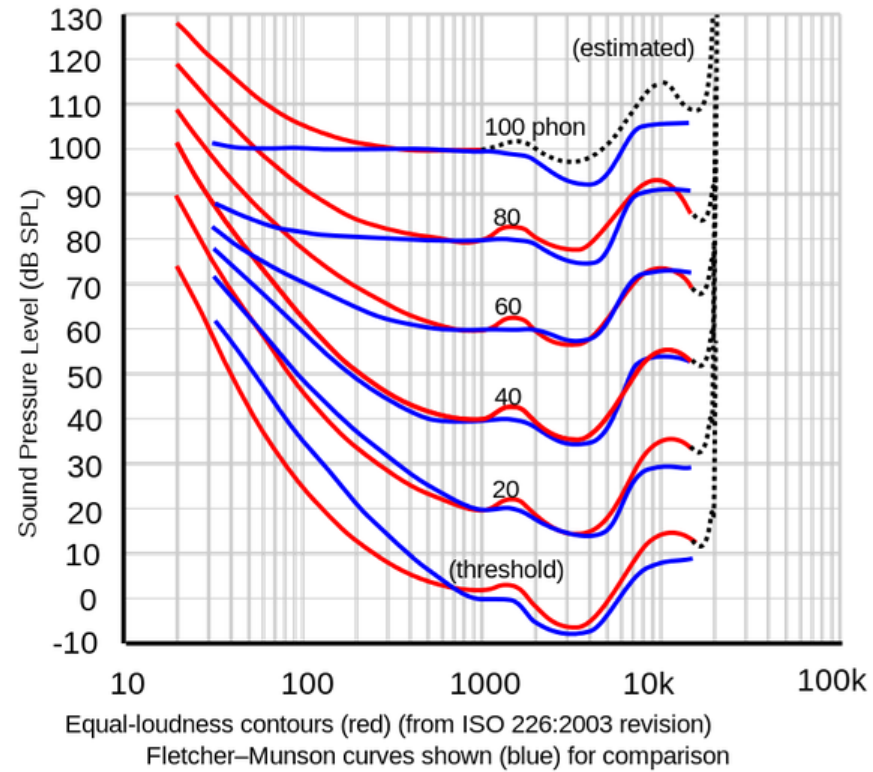
Human Hearing

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]

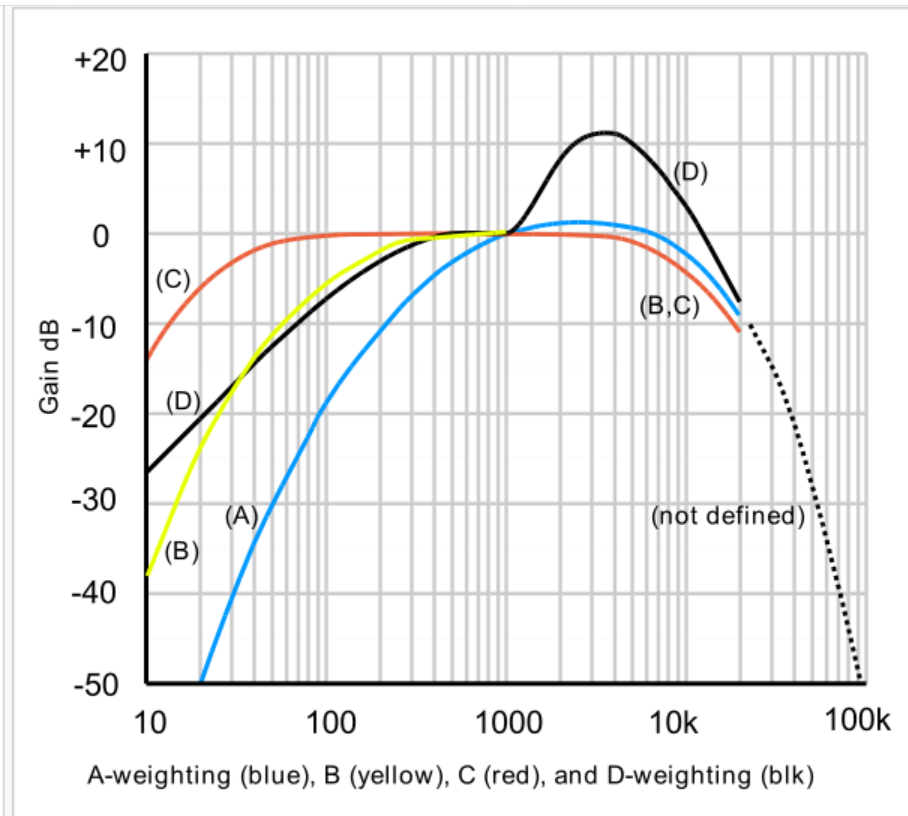
Contents [hide]

1 [Background](#)



[Draw regions of Voice, Music in this space.](#)

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



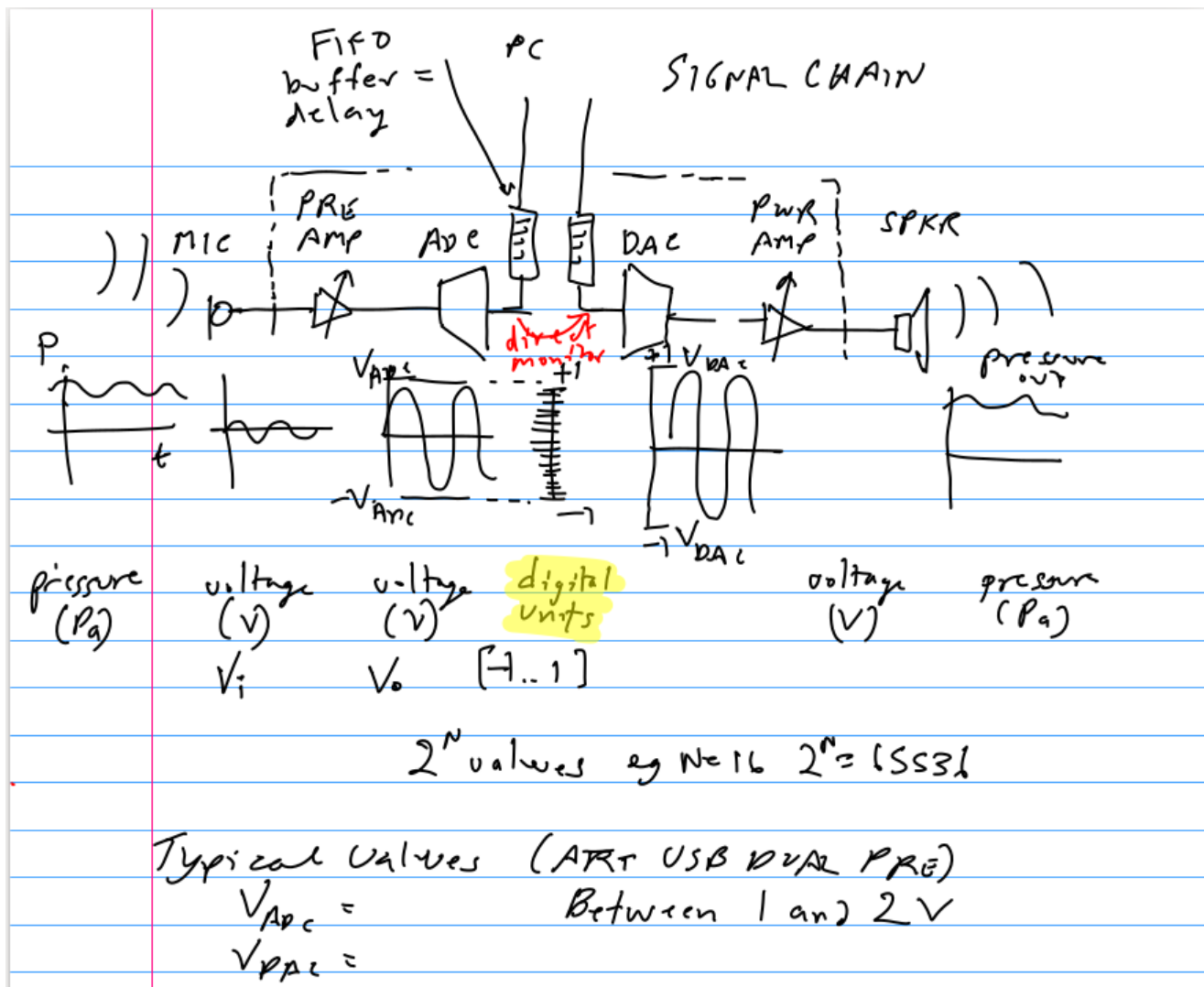
A graph of the A-, B-, C- and D-weightings across the frequency range 10 Hz - 20 kHz

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 low as 1 Hz or even 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

Details of USB Sound Card

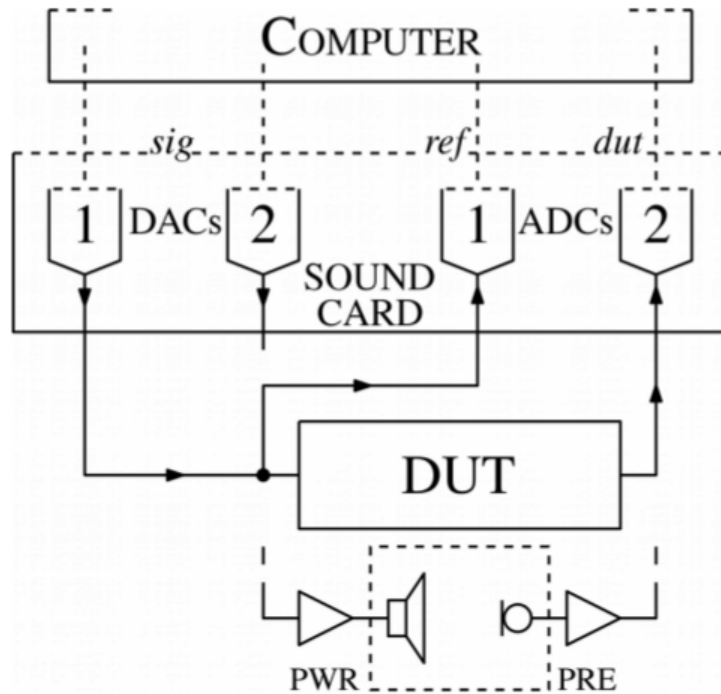


Open Source Software for Electrical and Acoustic measurement.

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

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

CONNECTION



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

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

Experiment 0: Test 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

Experiment 1: Measure speaker response vs Distance.

Experiment 2: Measure performance of Behringer XR12 Wireless Mixer

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Audio mixer technology (for live sound)

Wireless control via GUI,

Tactile control using MIDI

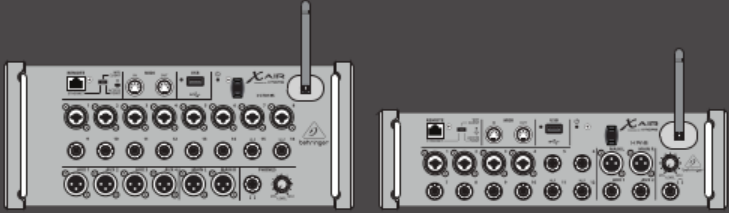
All processing happens in real time on device, DSP on ARM processor.

The delay time, from input to output is constant, and only 0.8ms!

(approx 38 samples, at $f=48000\text{Hz}$).


NOTE: Remote interface does not monitor audio, only signal levels.

However, device can record directly on to two (or more) tracks via USB interface.



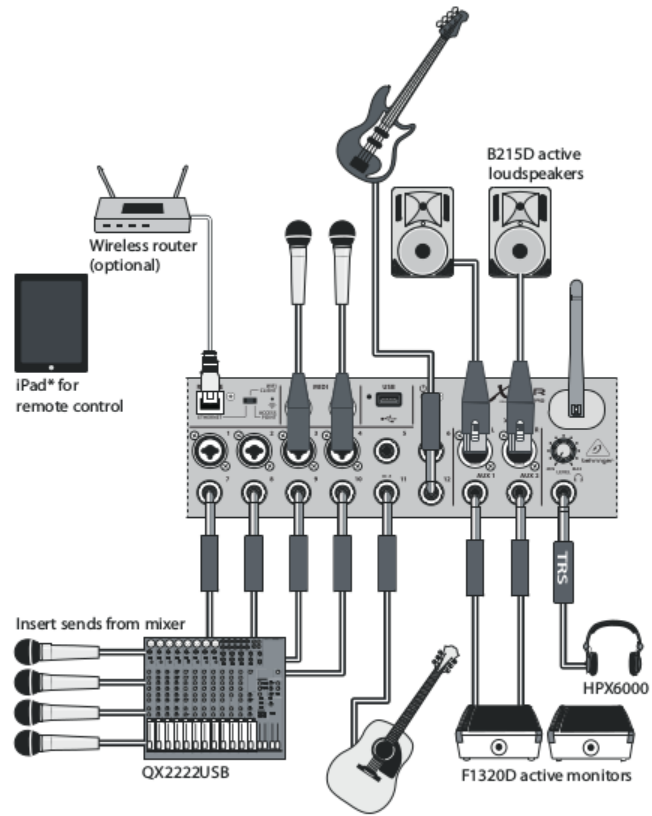
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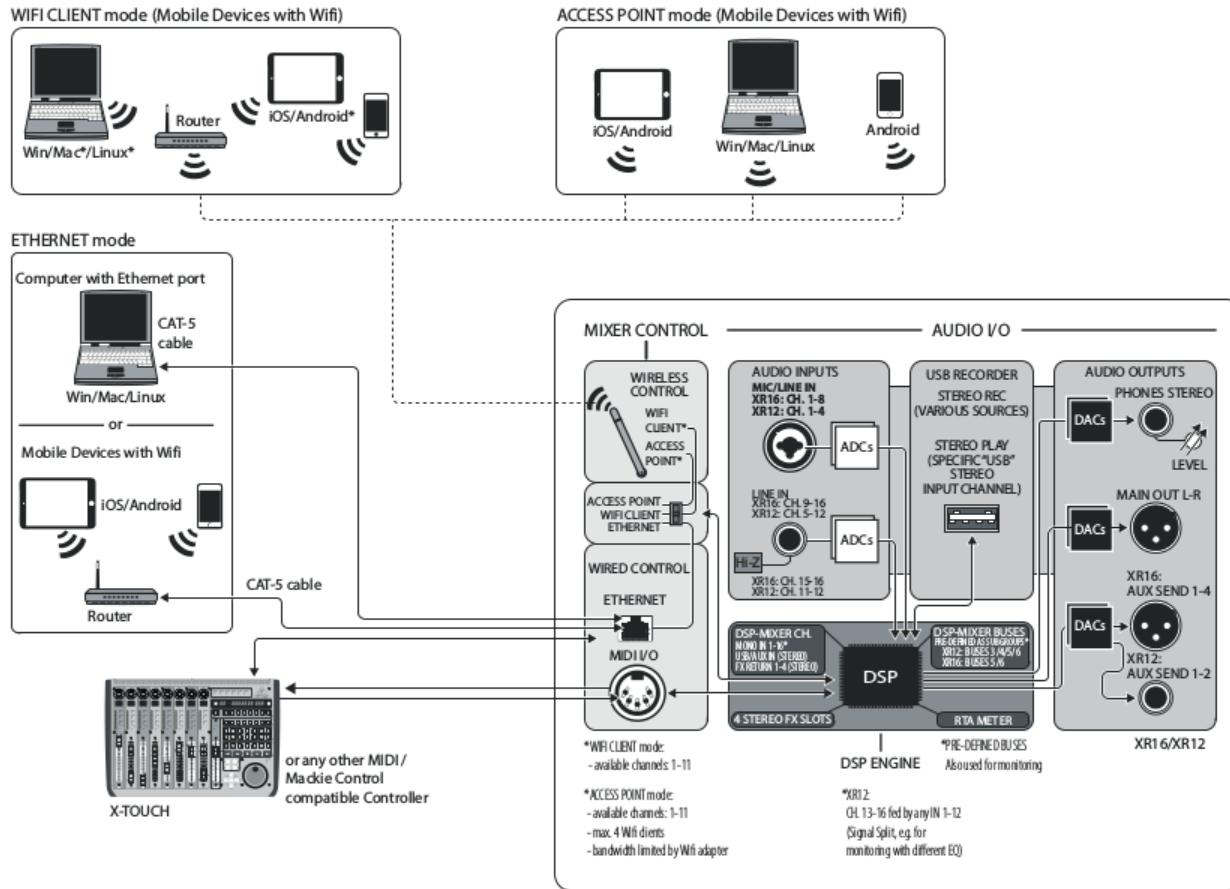
Quick Start Guide



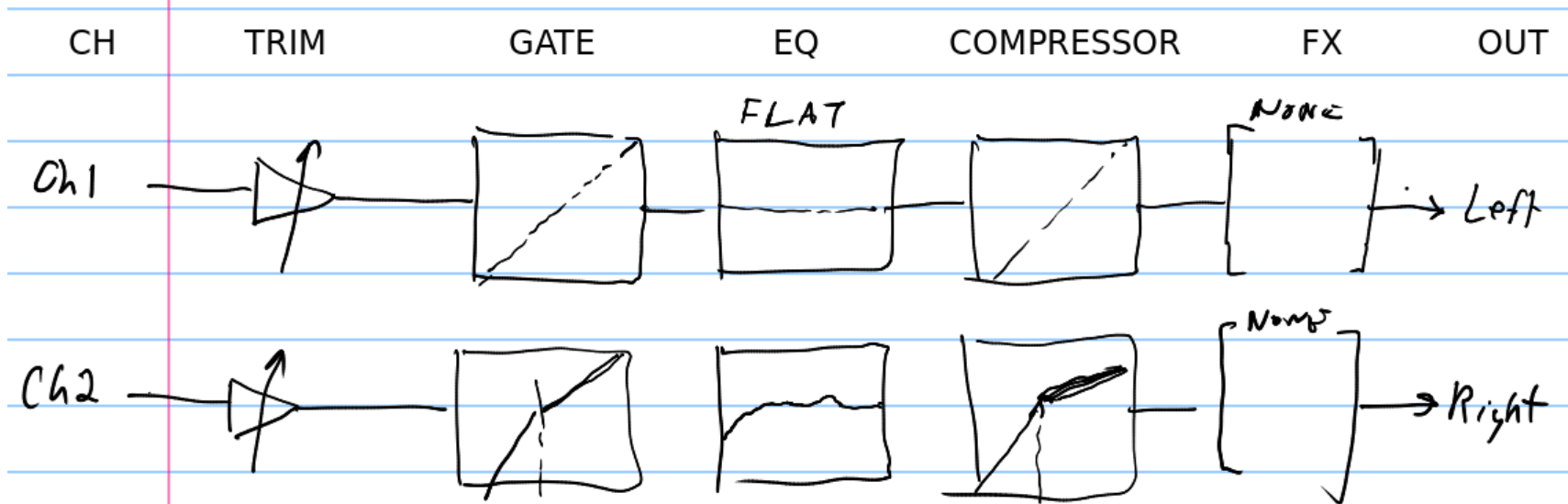
X AIR XR16/XR12
16/12-Input Digital Mixer for iPad/Android Tablets with
8/4 Programmable MIDAS Preamps, 8 Line Inputs,
Integrated Wifi Module and USB Stereo Recorder

Analog electronics





X Air Channel

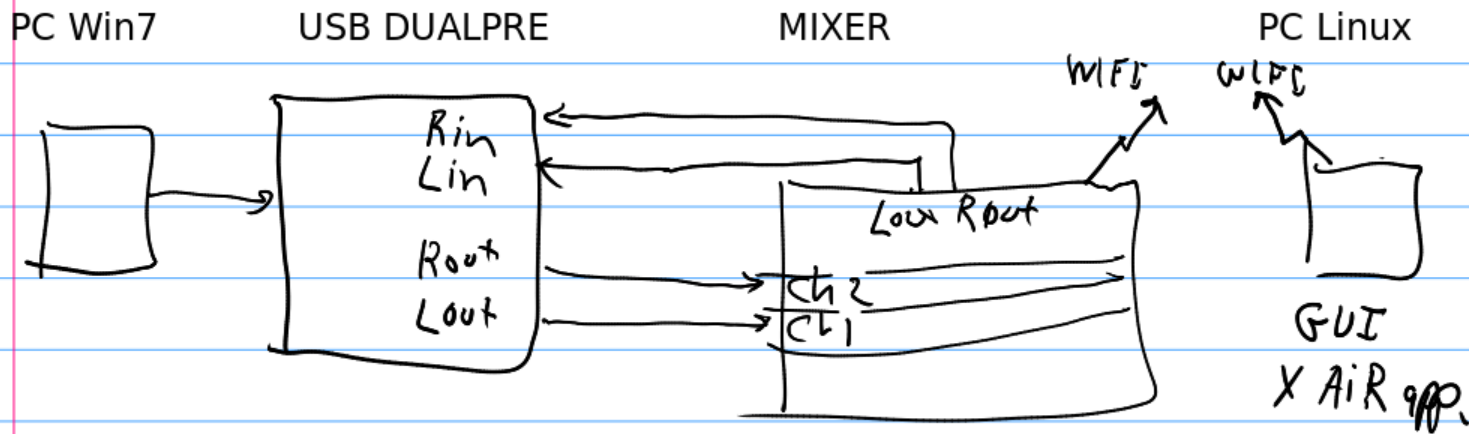


TRIM: Microphone gain, Level adjustment for line inputs
GATE (aka EXPANDER): increases input level if needed
EQ: Frequency equalization
COMPRESSOR (aka LIMITER): limits output level if needed.

Experimental Setup

Compare Ch1 (no effects)
Ch2 (effects)

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



RESULTS

Search or link to this question with @53.

Sign up for more classes at <http://piazza.com/uwaterloo.ca>.

Tell a colleague about Piazza. It's free, after all.

Thanks,
The Piazza Team

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Contact us at team@piazza.com

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