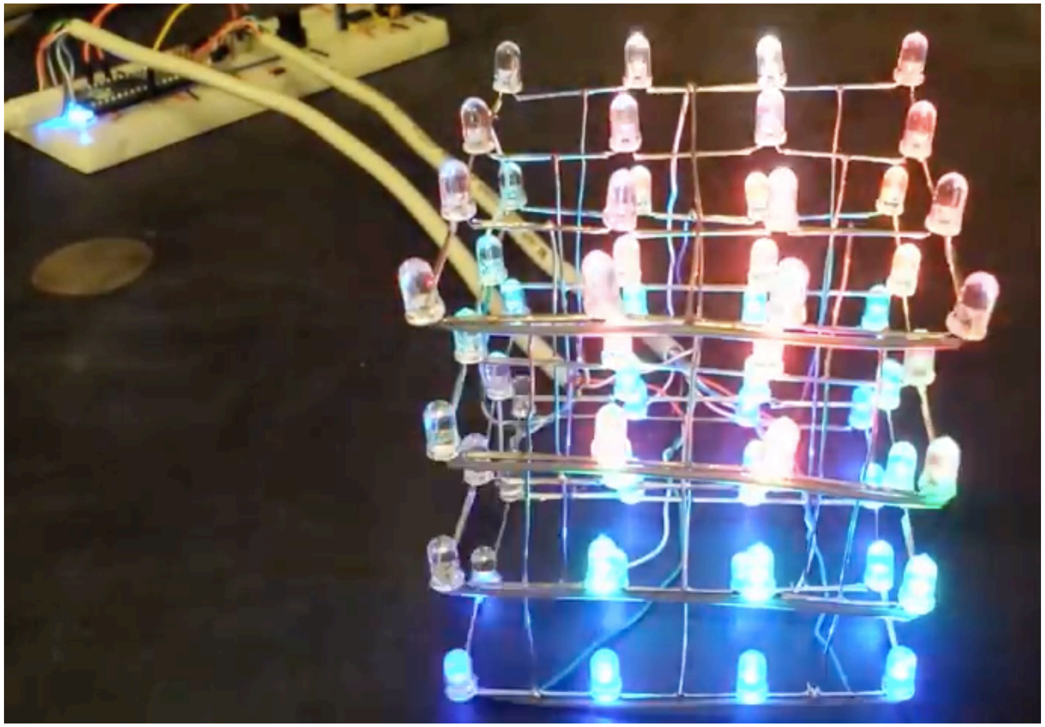
E40M
Sound and Music

LED Cube – Project #3

- In the next several lectures, we'll study

- Concepts
  - Coding
  - Light
  - Sound
  - Transforms/equalizers

- Devices
  - LEDs
  - Analog to digital converters

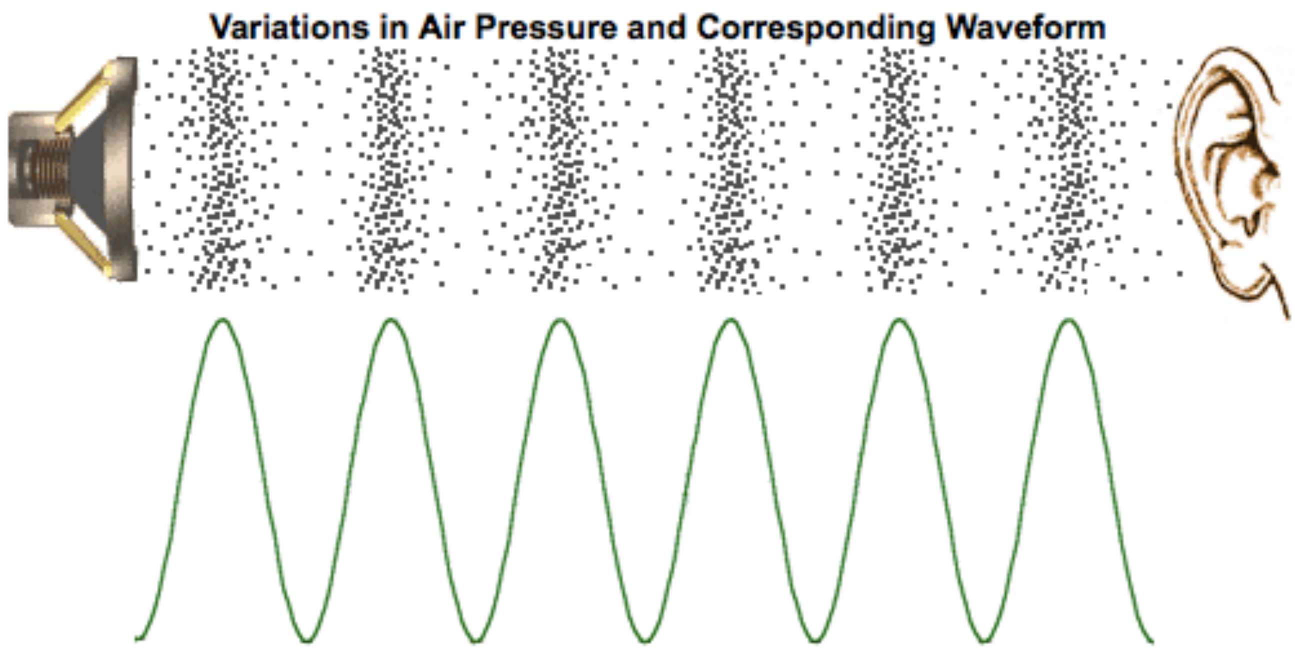


Music responsive LED Cube

https://www.youtube.com/watch?v=FRXDTiOHFIl&feature=youtu.be

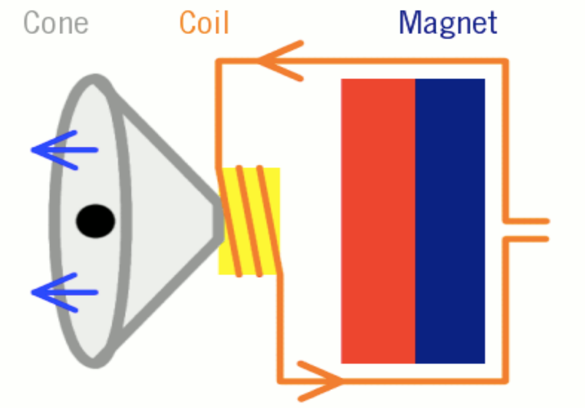
What is Sound Anyway?

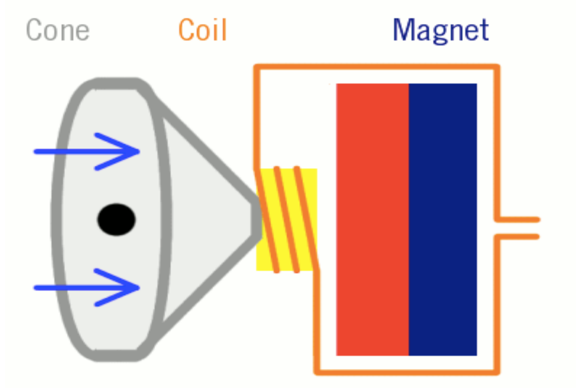
- It is a pressure wave that moves in air
  - Created by voice, instruments, speakers



http://www.mediacollege.com/audio/01/sound-waves.html

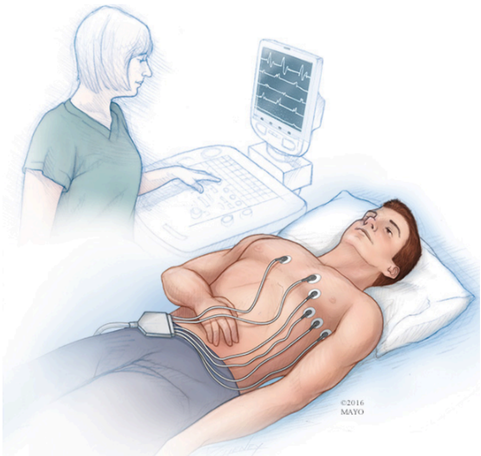
How Does a Speaker Create Sound?





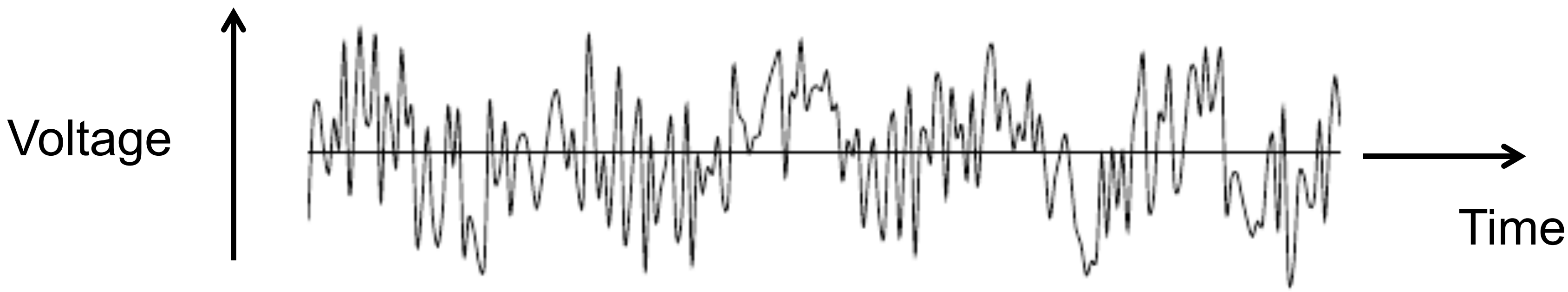
- Electrical signals from a sound system pass through the electromagnet attached to the speaker. The electromagnet is attracted or repelled by the permanent magnet, causing the speaker to vibrate, creating sound waves
- Power
  - 100W sound system, speakers are 8 Ω
    - Vi =100; i=V/R $\therefore$ $V^2 = 800$, so V swing > +/- 30V
  - Earbuds use much lower voltages (R is 10-100 Ω).
- http://www.explainthatstuff.com/loudspeakers.html

Sensors are Everywhere and Produce Electrical Signals

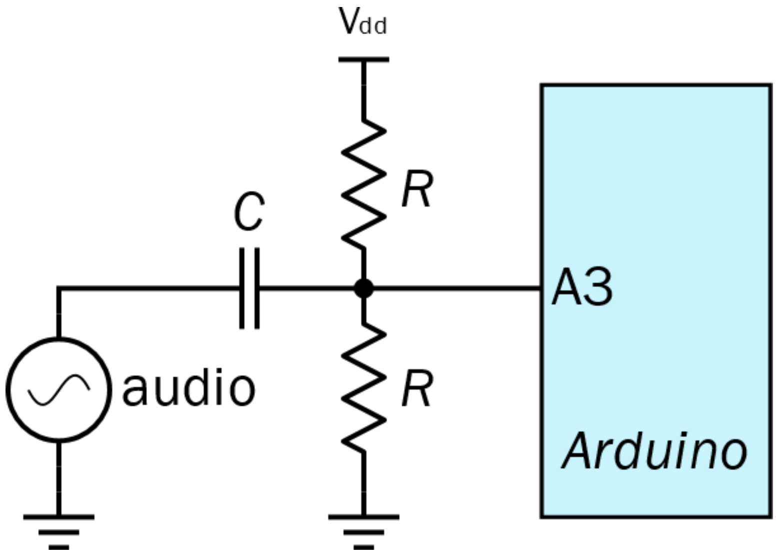


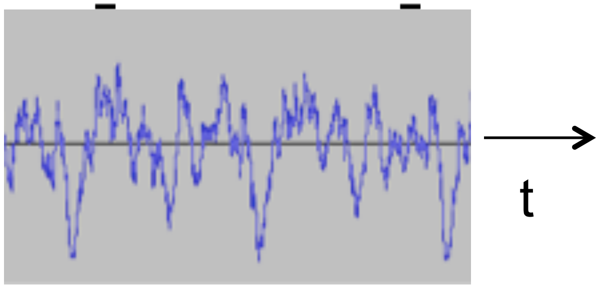
Sound pressure converted to voltage vs. time

Electrical signals plotted as voltage vs. time

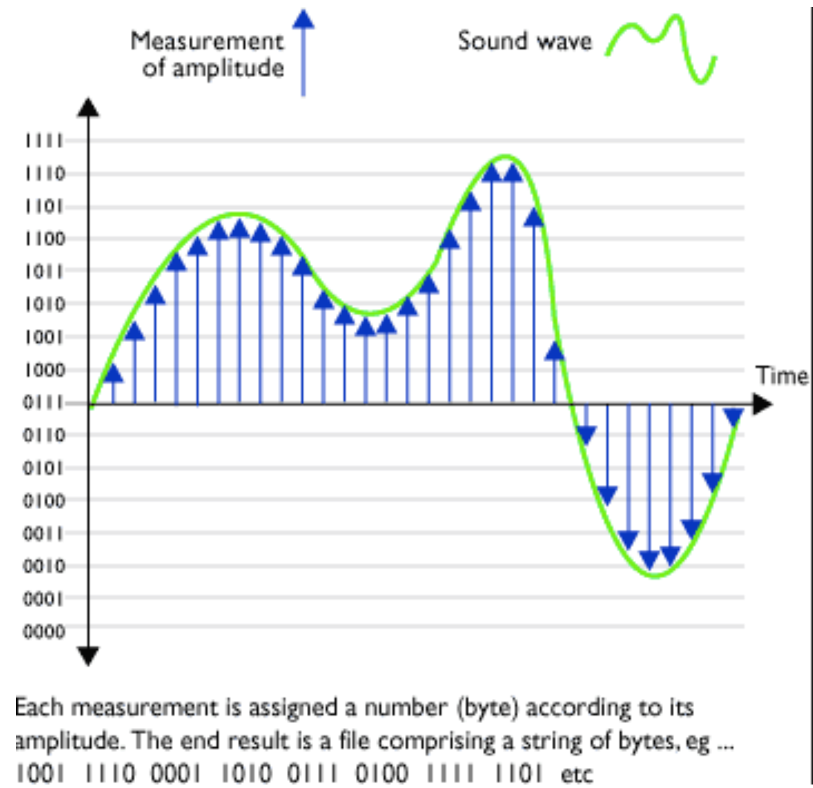


Converting Analog to Digital Signals





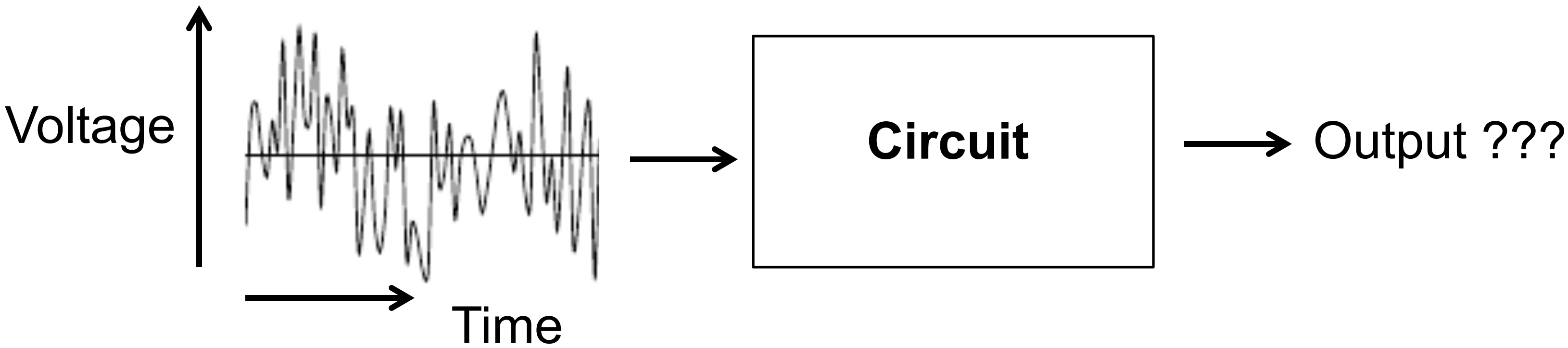
Audio Signal



http://www.planetoftunes.com/digital-audio/how-do-analogue-to-digital-converters-work.html#.WuOS0y-ZPOQ

- Analog signals can be converted to a stream of digital numbers corresponding to magnitude vs. time in an A to D converter.

Calculating Circuit behavior



- We could construct the output signal by considering the input at each time t and construct the output point by point.
- This could get pretty tedious!
- Maybe there's another way to think about this?

BREAKING DOWN SIGNALS INTO FREQUENCY COMPONENTS

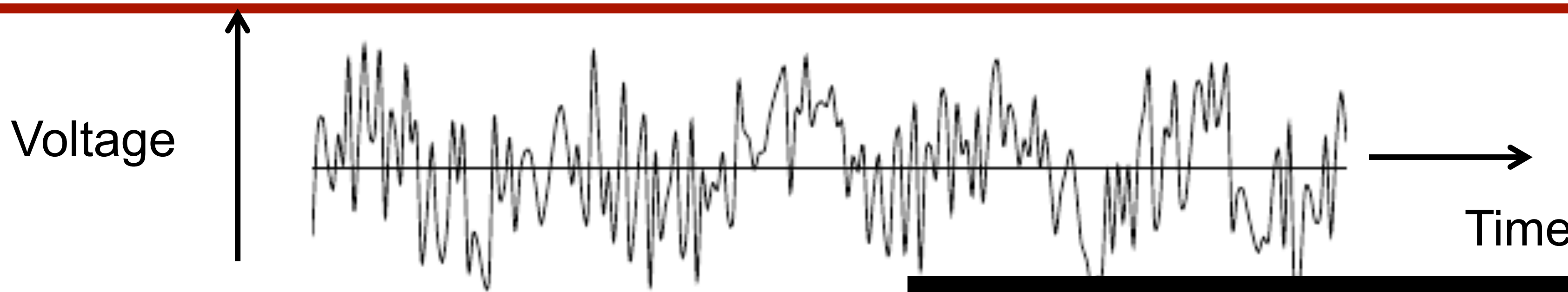
Representing Signals In Different Ways

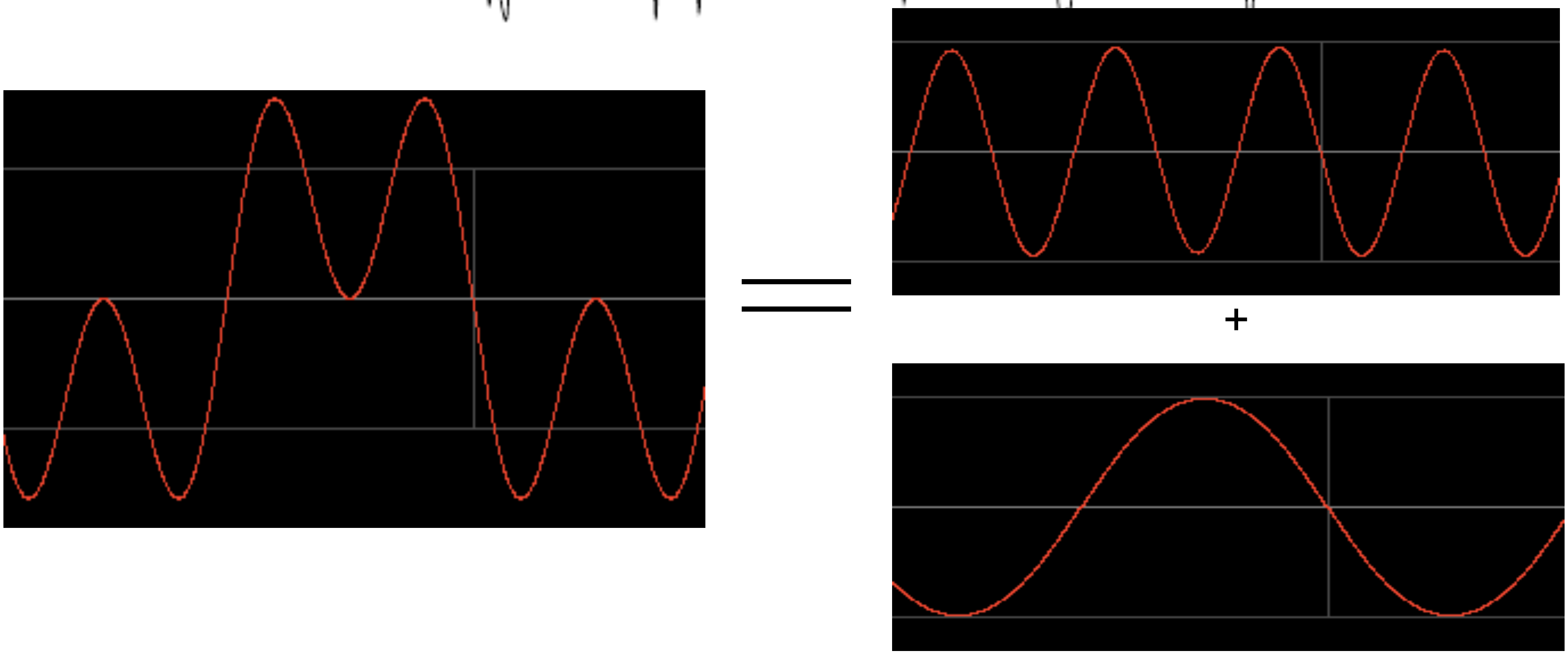
- We could represent sound or other signals as a string of numbers (using an A to D converter as on page 6)
  – Which represent voltage at different times
  – Our brain doesn't process sound that way

- We think and talk about sound/music as combinations of tones
  – Summation of different sinewaves
  – And you can represent sound this way too

- All signals can be represented in two ways
  – Voltages in time
  – Sum of tones of different amplitudes and frequencies

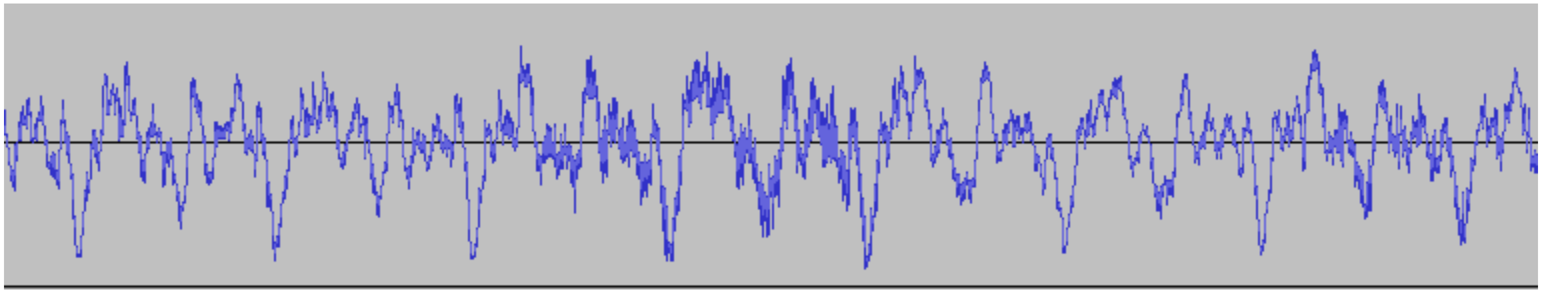
Representing Signals





Sound as Tones

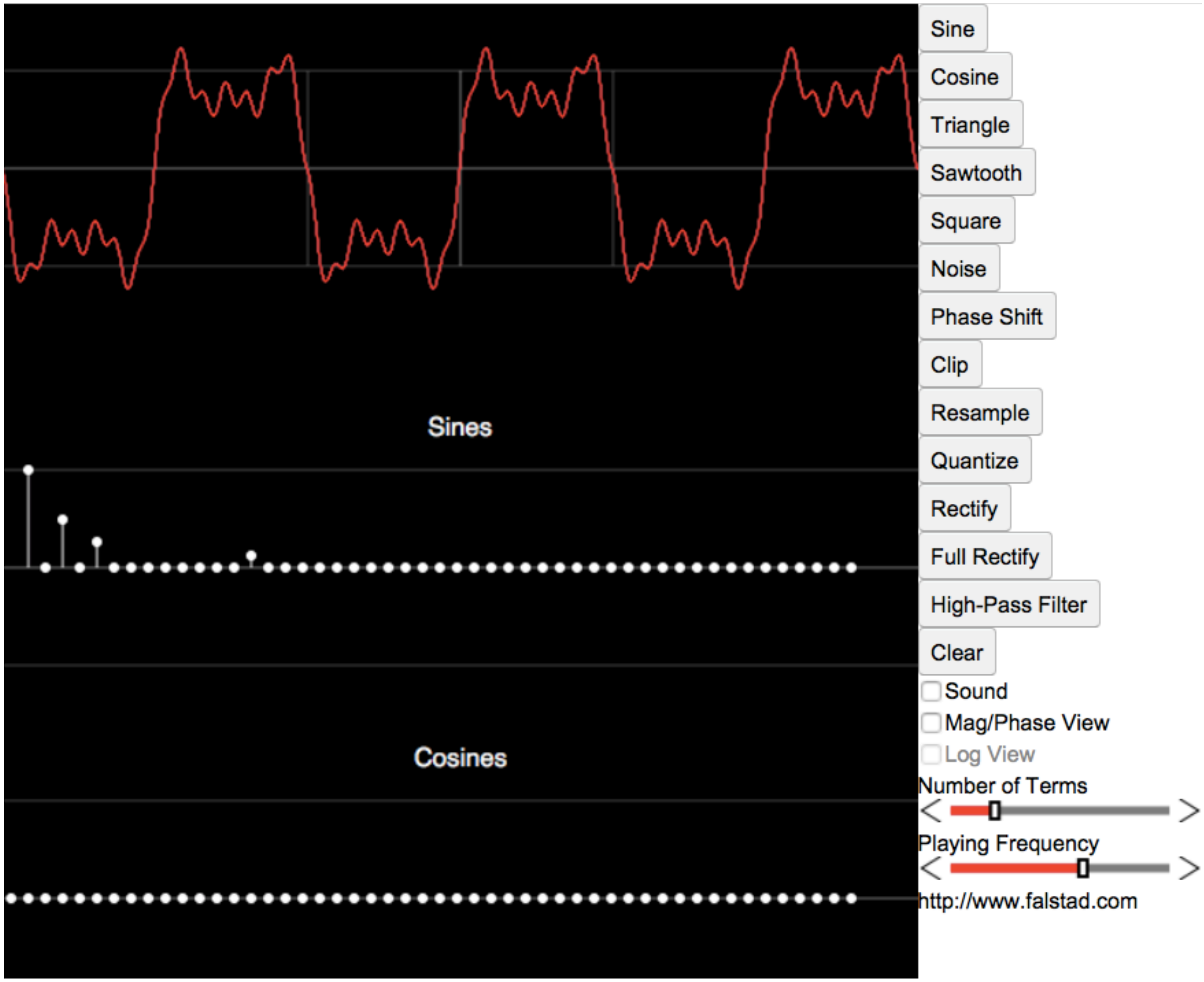
- We perceive sound as a composition of tones
  - Each tone is a sine wave of pressure
    - Which is a sinewave in voltage
- The funny waveforms that we see in time



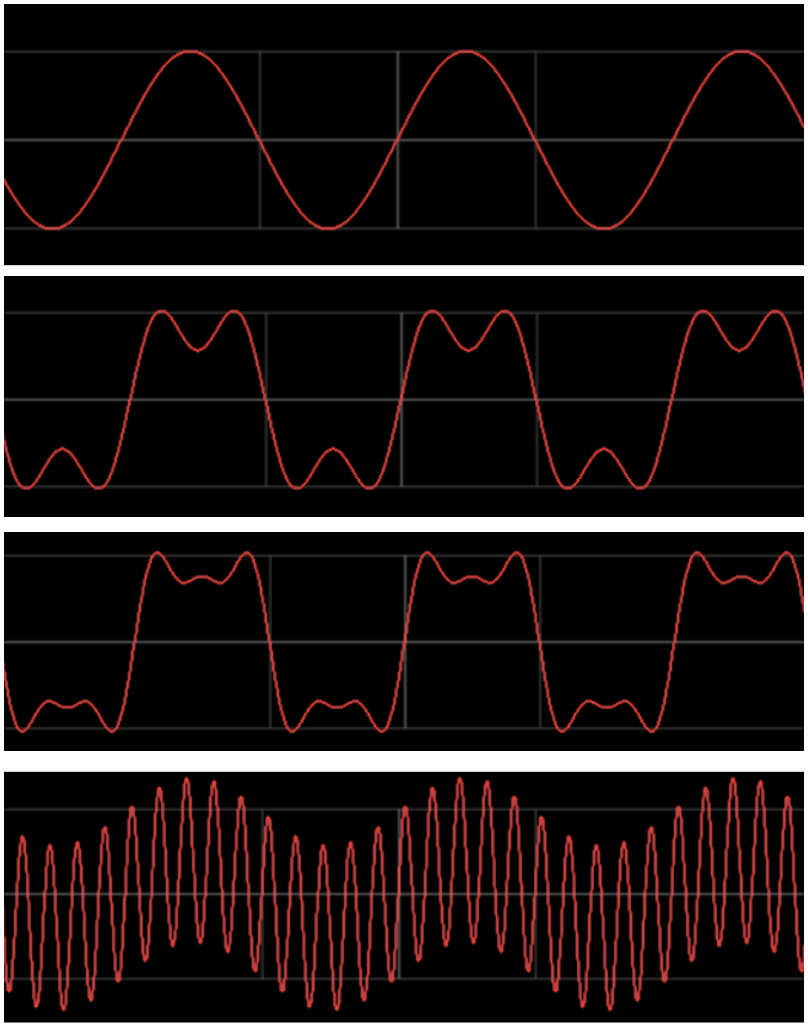
  - Can be created by adding many tones (sinewaves) together
- Java applet from: http://www.falstad.com/fourier/
  - You may have to override security features in your computer to run it after you download it (see class website). Or you can run it from the above website.

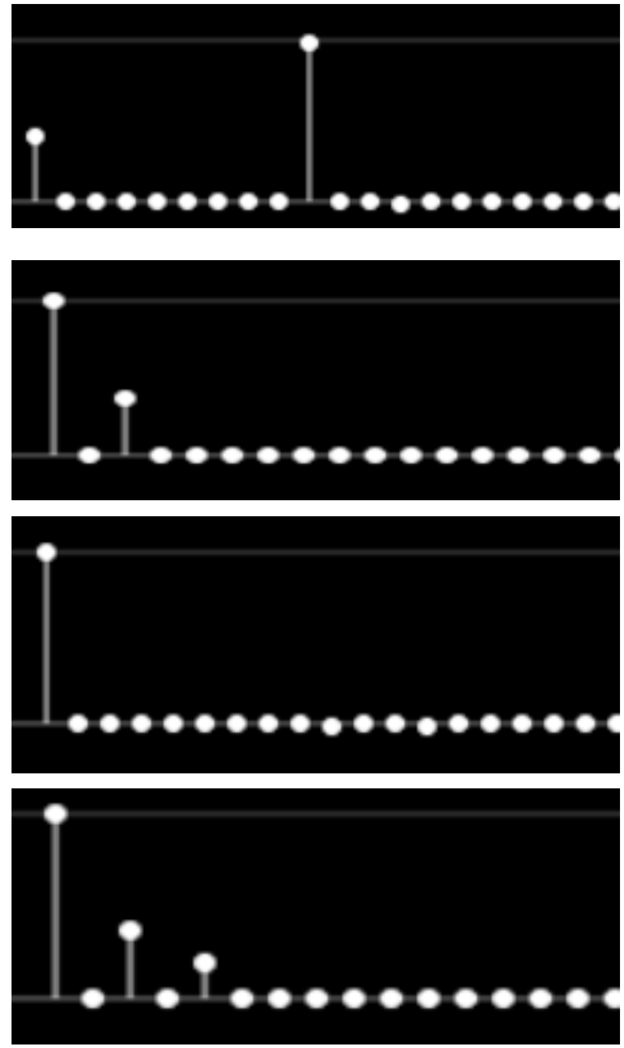
Relating Voltage to Sinewaves – Demo
http://www.falstad.com/fourier/

- Calculates Fourier Series representation (later) of time varying wave.
- Or calculates time varying wave from Fourier components (tones).
- Let's play with it a little bit



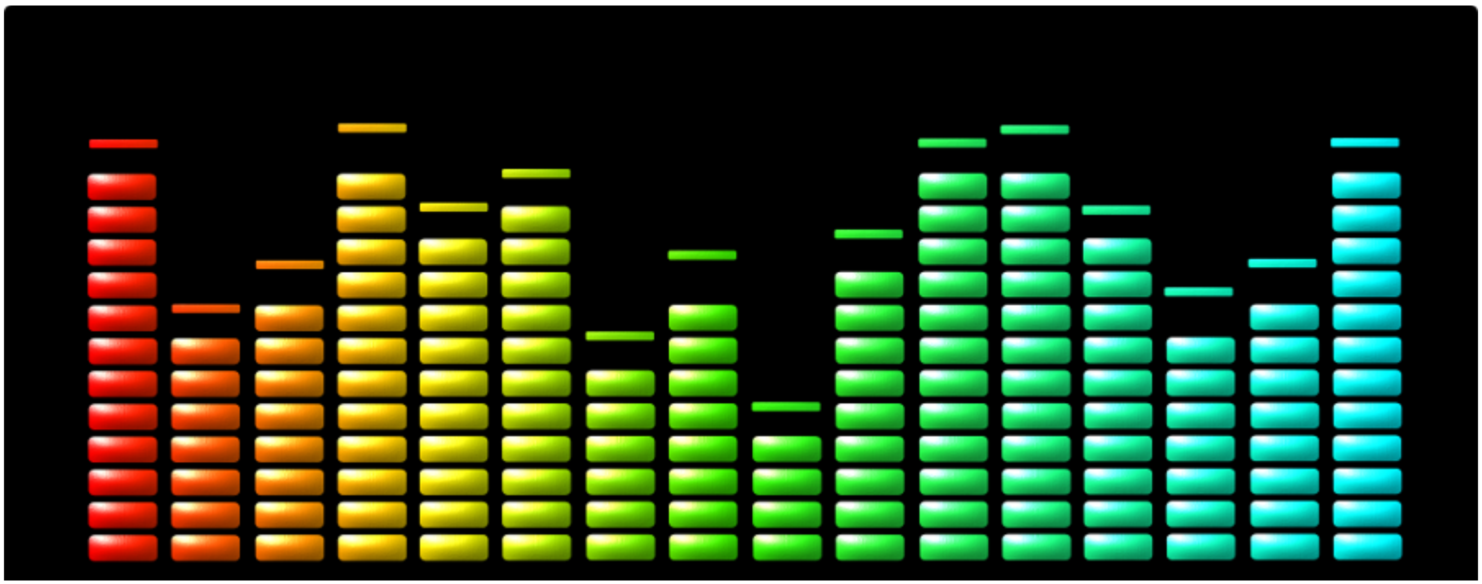
Which Goes With Which?





Equalizers

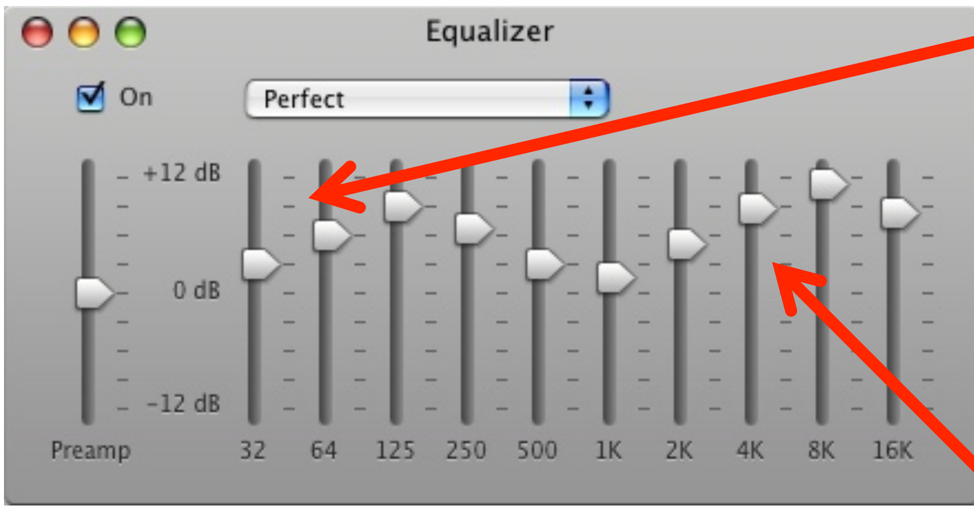
- We have all seen this type of display

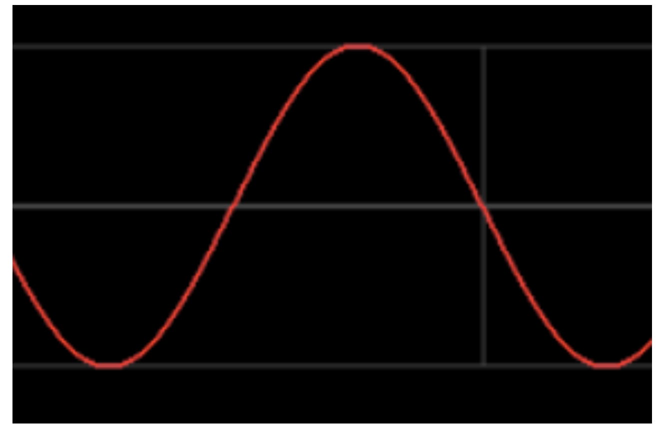


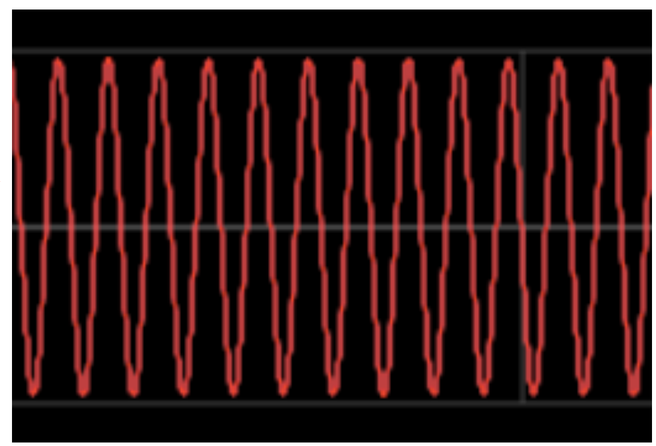
- What information does it represent?

Setting An Equalizer

- You might have even played with setting levels

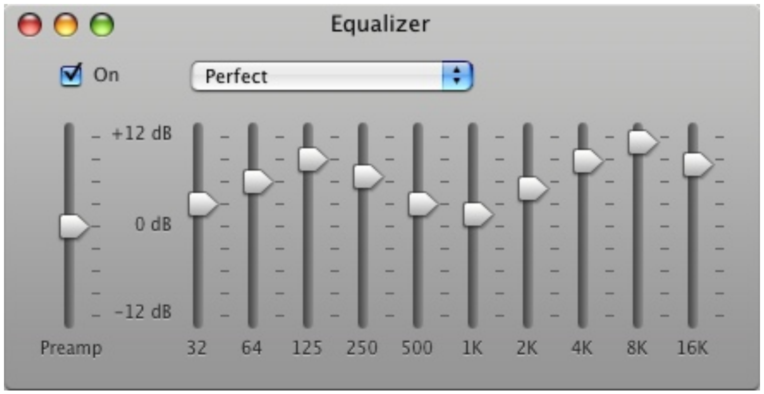


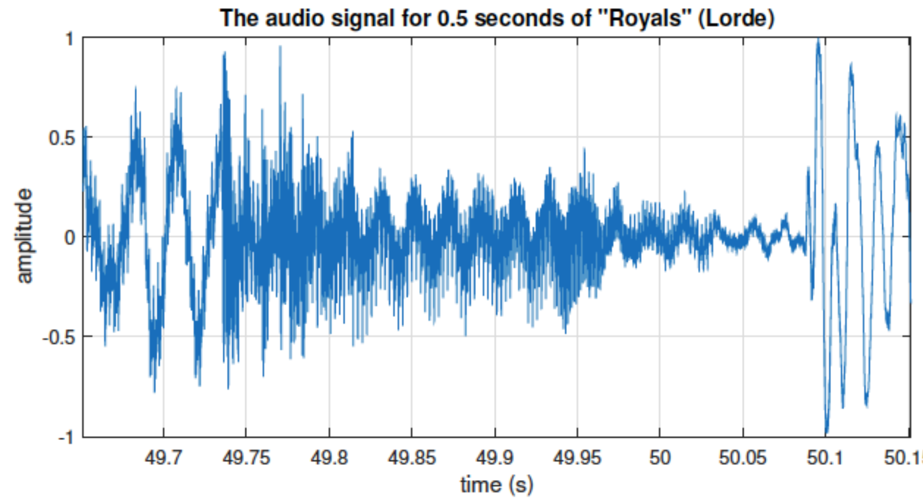


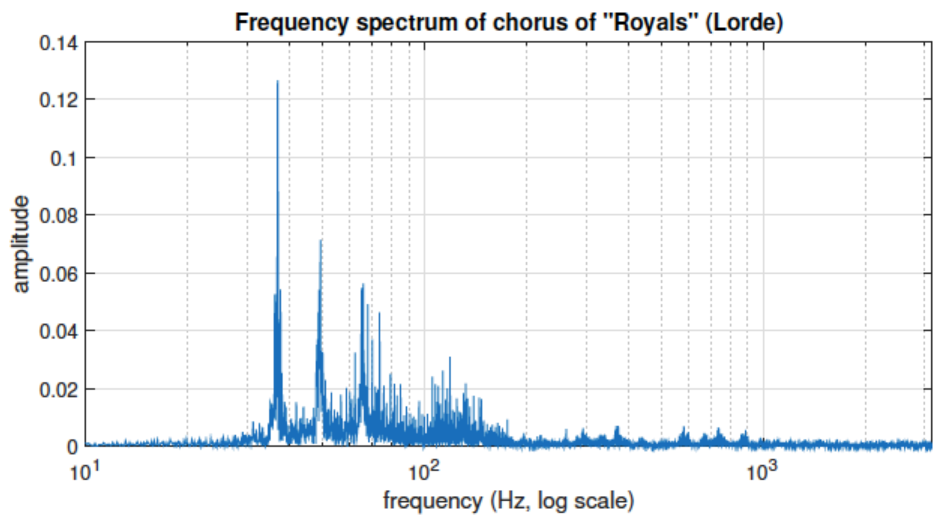


- Ever think about what you are really doing here?
  - The music is a set of voltages vs. time.

What You Are Doing



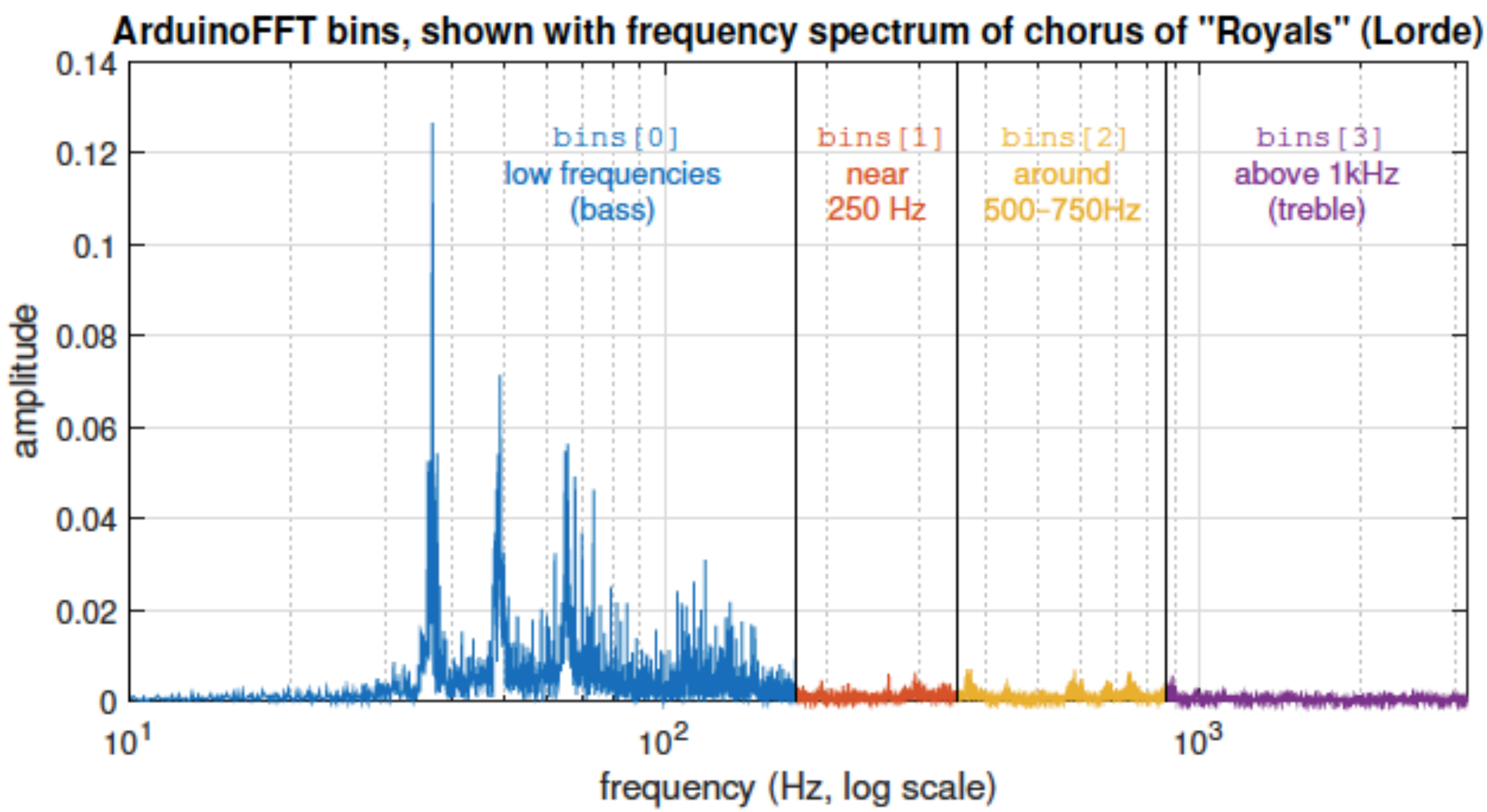




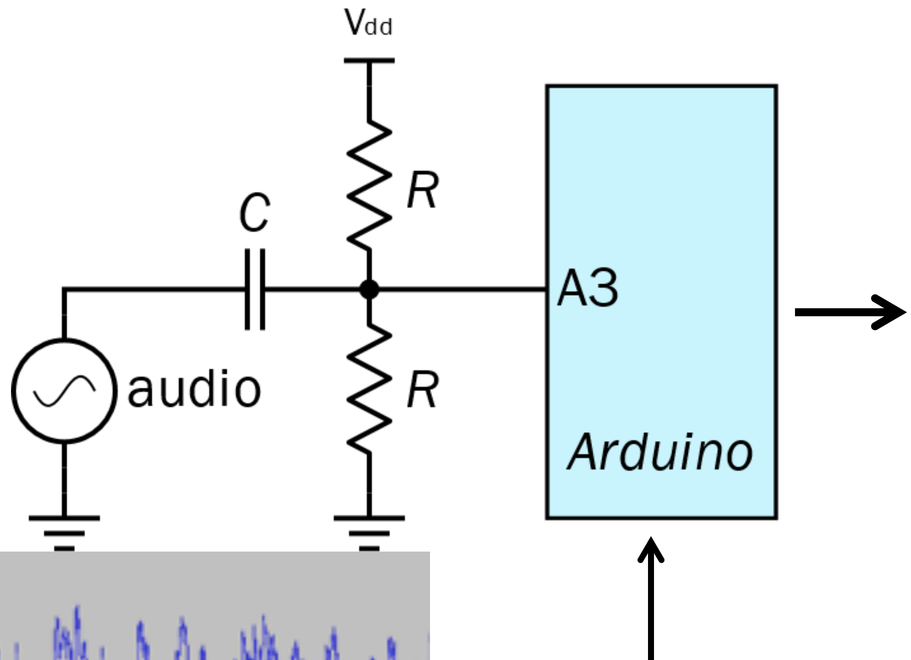
- Changing the amplitude of sinewaves in different frequency bands
- Scale is weird – dB - logarithmic gain, more on that later

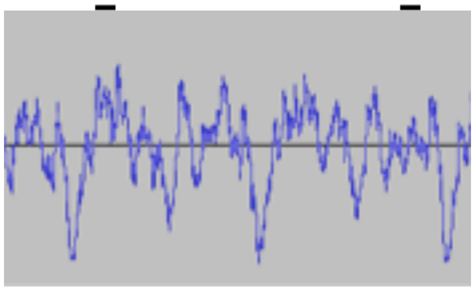
Sound Display on Your LED Cube

- In Lab 3c one of the options for your LED cube is to display the frequency components of music.



Lab 3C Audio Option

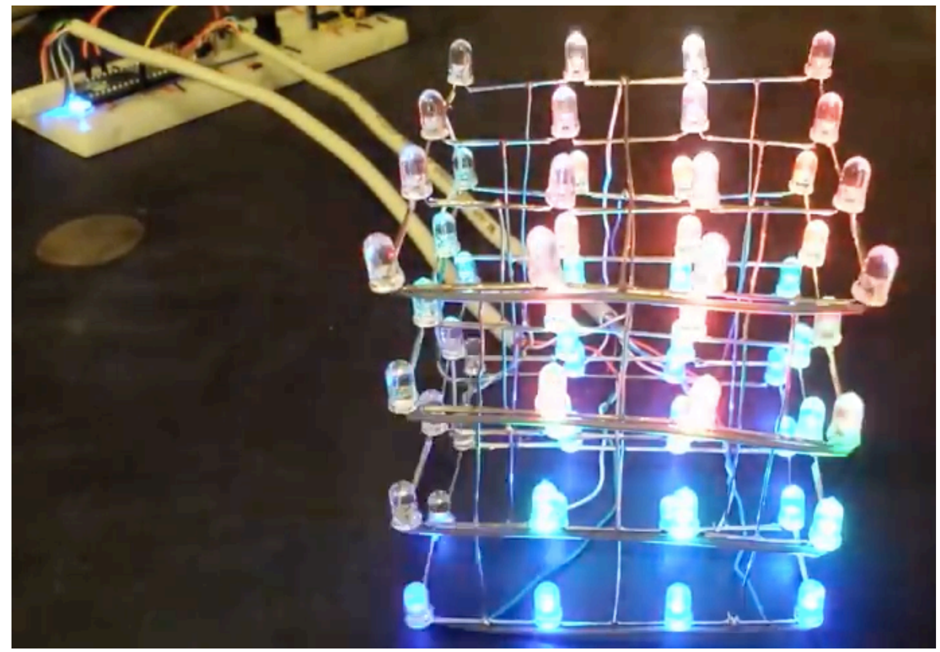




Audio Signal

A to D Converter

FFT Software



Music responsive LED Cube

FOURIER SERIES

Fourier Series

- The formal name for this alternative representation
- Officially it only works for repetitive signals
  - Since sine-waves repeat
- There is an extension for non repetitive signals
  - It is called the Fourier Transform
- Many people use Fourier series for a block of data
  - And just assume that the block of data repeats
  - That is what the demo did (http://www.falstad.com/fourier/)

Formal Definition

- Assuming a signal repeats every T seconds
  - Or we just have T seconds of data to look at . . .

$$\upsilon(t) = a_0 + \sum_{n=1}^{\infty}\left(a_n \cos\frac{2n\pi t}{T} + b_n \sin\frac{2n\pi t}{T}\right)$$

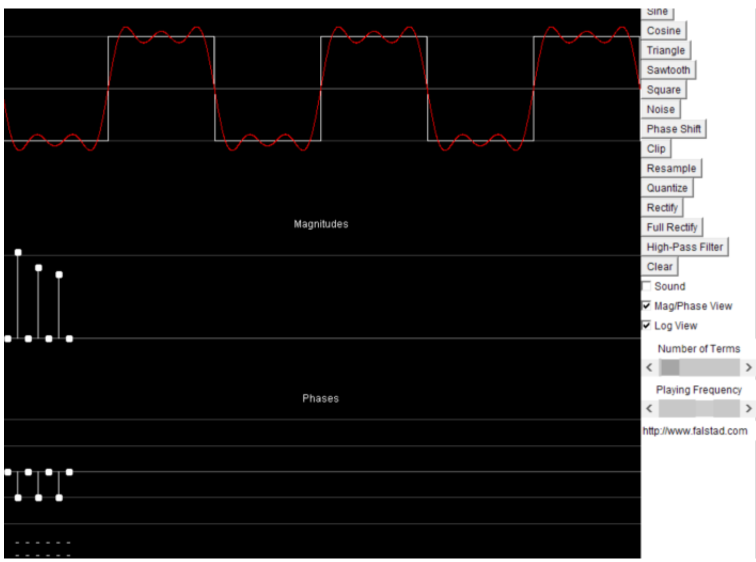
- The term with n=1 is called the fundamental term
  - It is the lowest frequency that exists in a period of T
  - The other terms are called harmonics
    - They are integer multiples of the fundamental frequency 2πT

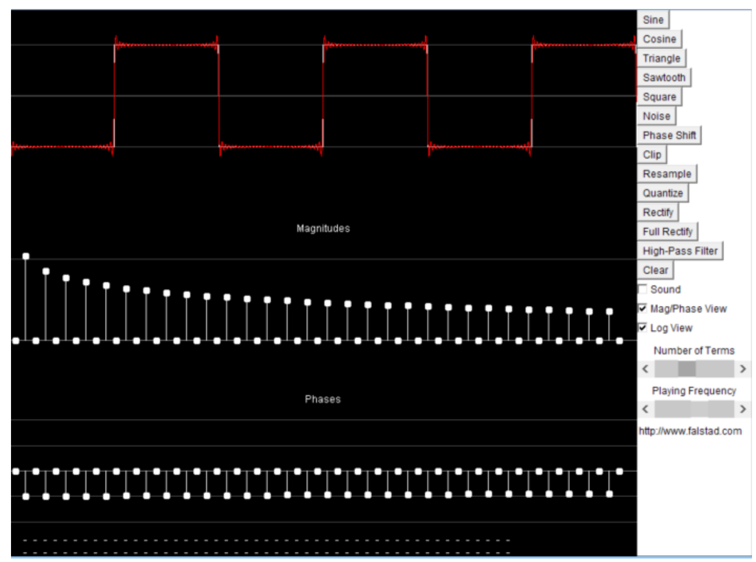
- A detailed discussion of Fourier Transforms is beyond the scope of this course. Take EE 102a if interested in more details.

Equation For A Square Wave

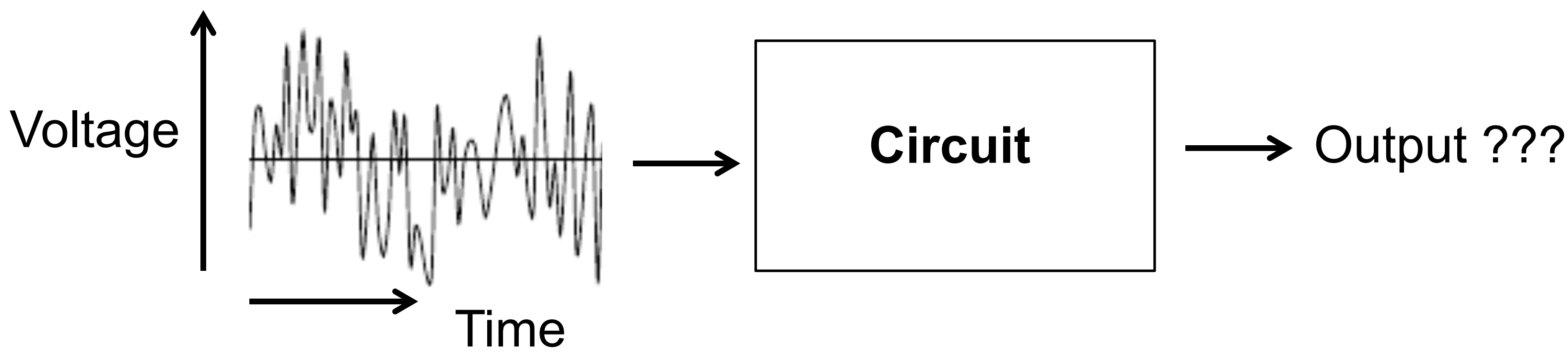
$$\sum_{n=0}^{\infty}\left(\frac{1}{2n+1}\sin\frac{2\pi(2n+1)t}{T}\right)$$

- It consists of all odd harmonics
  - Amplitude falls slowly (as 1/n)





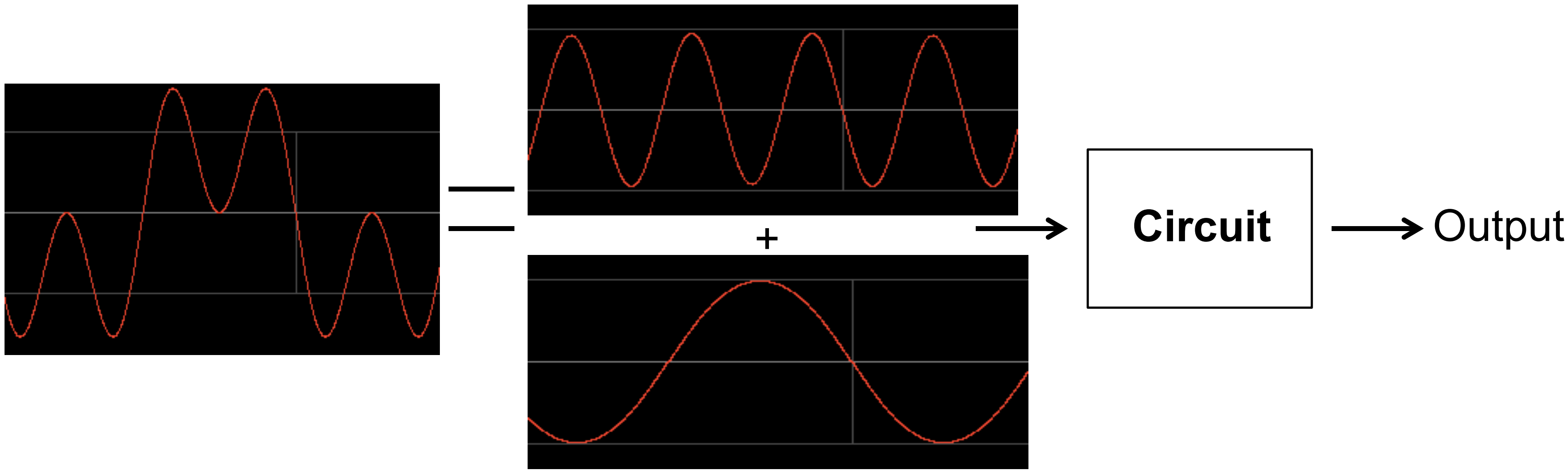
Frequency Domain Analysis



- If we have a circuit with an input voltage that varies with time, we can figure out what the output of that circuit will be by considering the individual frequency components of the input signal.

- Superposition will give us the resulting output.

Frequency Domain Analysis



- It's probably not obvious why this approach might make life simpler, but this will become clear starting next week when we talk about circuits that have capacitors and inductors in them.

Learning Objectives

- Understand what sound is
  - And how an electronic device stores and generates sound
    - It represents sound as a time varying voltage

- Understand that we can represent the sound in different ways
  - As a varying voltage vs. time
  - As the sum of different tones

- Understand how an equalizer works
  - You can amplify/attenuate tones in different bands

- You can convert from tones to voltages

$$\upsilon(t) = a_0 + \sum_{n=1}^{\infty}\left(a_n \cos\frac{2n\pi t}{T} + b_n \sin\frac{2n\pi t}{T}\right)$$

GENERATING FOURIER COEFFICIENTS

How To Go From Waveform to Sinewaves?

- Going from sinewaves to waveform is straightforward.
  - You just add all the sinewaves together.

$$\upsilon(t) = a_0 + \sum_{n=1}^{\infty} \left(a_n \cos \frac{2n\pi t}{T} + b_n \sin \frac{2n\pi t}{T} \right)$$

- But how does one figure out what the various a_n and b_n are if you have only v(t)?

- You use an interesting property of sinewaves.

Product of Sine Functions

$$\int_0^T dt\left(\cos\frac{2n\pi t}{T} * \cos\frac{2m\pi t}{T}\right)$$

- Is always zero unless m = n

- To see why this is true, remember that
  - cos(a+b) = cos(a) cos(b) – sin(a) sin(b)
- Which means
  - cos(a) cos(b) = ½ [cos(a+b) + cos(a-b)]

- So if m is not equal to n, the product will just be two sinewaves
  - One at the sum of the frequencies and one at the difference

- When n=m, cos(a-b) = cos(0), so the integral is T/2

This Means

- If v(t) is equal to

$$\upsilon(t) = a_0 + \sum_{n=1}^{\infty}\left(a_n \cos\frac{2n\pi t}{T} + b_n \sin\frac{2n\pi t}{T}\right)$$

- Then if I multiply v(t) by cos(2mπt/T) and integrate from 0,T
  - The only non-zero term will be the term where n = m
  - So the result will be T/2*a_m

- This gives us a way to extract a_n b_n from v(t)

$$\int_0^T dt\left(v(t) * \cos\frac{2m\pi t}{T}\right) = \frac{T}{2}a_m$$

Does n Really go to Infinity?

- No
  - All signals have limited bandwidth
    - Which means that they have a finite number of sinewaves
  - But the bandwidth of different signals are different
    - And this sets how large n can get

- For audio signals
  - 20kHz is the limit for human hearing

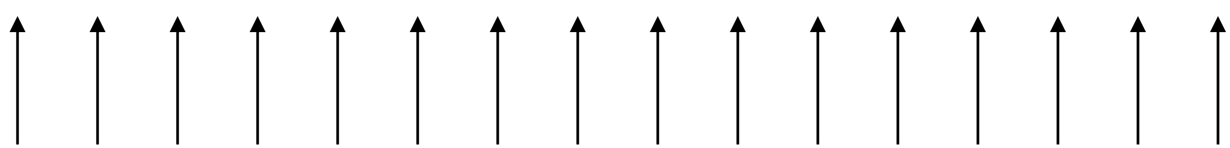
- Electronic signals are all over the map
  - Temperature, EKG, might be 100Hz
  - Wireless communication might be 5GHz

Sampling a Signal

- Computers don't like dealing with continuous variables
  - They like dealing with numbers
  - It is the only thing they can really handle

- So to deal with signals that change in time
  - Need to convert them to a series of numbers

- They do this by measuring the waveform at fixed interval in time



So How Fast Do You Need To Sample?

- Remember you need to capture the sinewaves of the signal
  - How many samples do you need per cycle of sine?

- Nyquist sampled
  - You only need two samples of the high-frequency sinewave