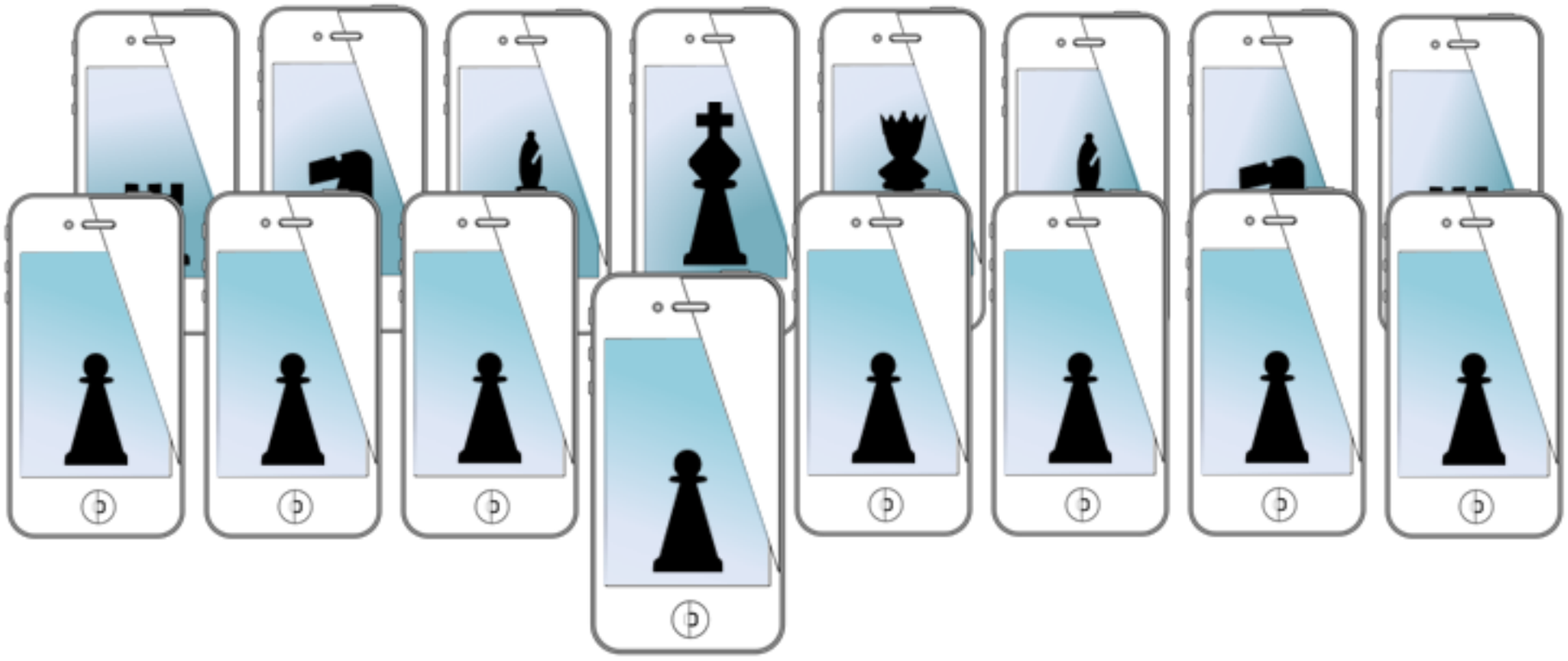
MOBILE SENSING LEARNING



CSE5323 & 7323

Mobile Sensing and Learning

week 4, lecture one: audio graphing, sampled data, & accelerate

Eric C. Larson, Lyle School of Engineering,
Computer Science and Engineering, Southern Methodist University

course logistics

- lab two due at end of next week
- in-class assignment on audio next time!

course logistics

- Look at A2

Module A
Create an iOS application using the NovocaineExample template that:

- Reads from the microphone
- Takes an FFT of the incoming audio stream
- Displays the frequency of the two loudest tones within 6Hz accuracy
- Is able to distinguish tones as least 25Hz apart, lasting for 100ms or more

The sound source must be external to the phone (i.e., laptop, instrument, another phone, etc.).

Module B
Create an iOS application using the NovocaineExample template that:

- Reads from the microphone
- Plays a settable (via a slider or setter control) **inaudible** tone to the speakers (15-20kHz)
- Displays the magnitude of the FFT of the microphone data in decibels
- Is able to distinguish when the user is {not gesturing, gestures toward, or gesturing away} from the microphone using Doppler shifts in the frequency

agenda

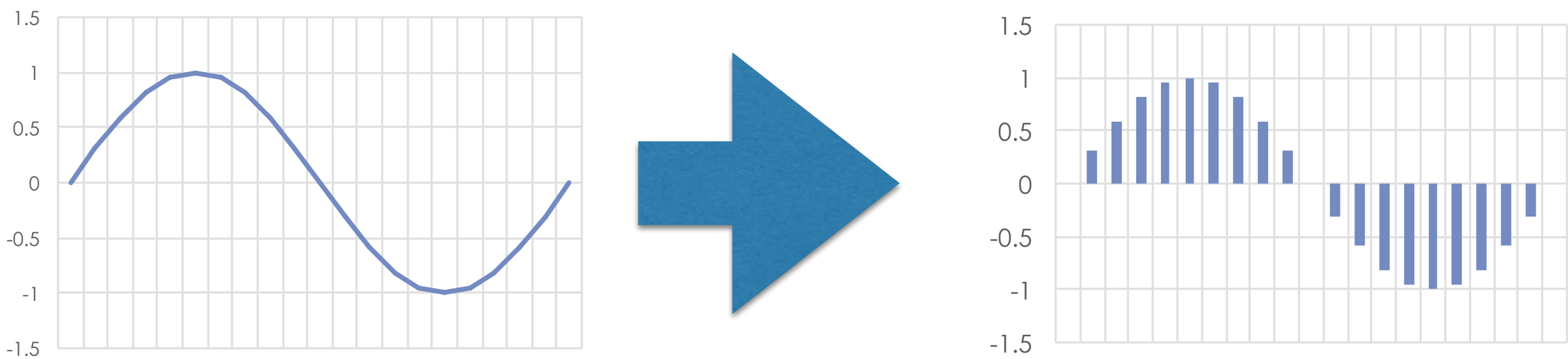
- dealing with sampled data
- the accelerate framework
  - massive digital signal processing library
- graphing audio fast (well, graphing anything)
  - must use lowest level graphing, OpenGL

intro to sampled data

- why is understanding sampled data important?
  - because we'll be dealing with it all semester
  - it's important to understand basic mistakes that can be made
- there are entire courses dedicated to sampled time series
  - actually entire courses on analyzing frequency content
- we'll touch on a few guidelines to help you design your projects better

intro to sampled data

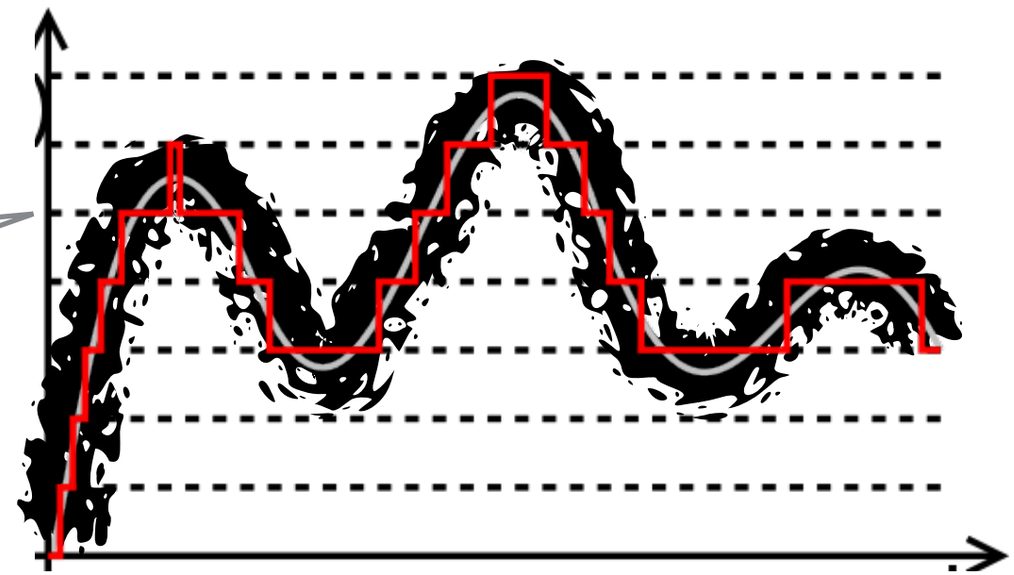
- physical processes are continuous
  - to process with computers, we must digitize it
  - digitization can change how we understand the signal
- digitization occurs in time and amplitude
  - time: sampling
  - amplitude: quantization



sampled data

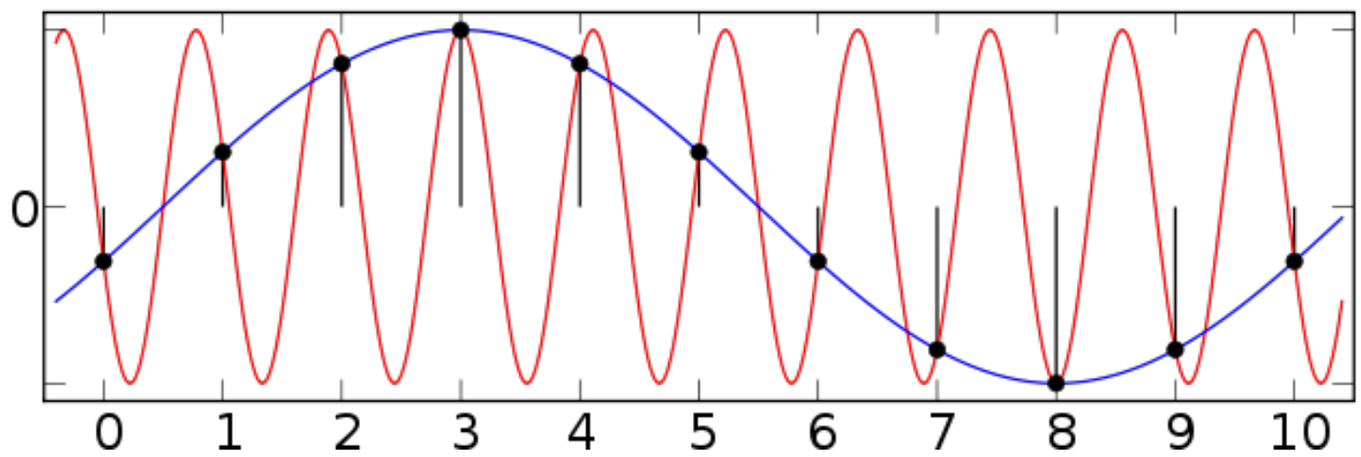
- quantization (amplitude)
  - introduces error in estimating amplitude of a signal
  - error can be reduced by adding more “bits per sample”
- most ADCs are 16 bits, considered “good enough”
- sufficient for most uses
  - not for others!

iPhone uses LPCM 32 bits, Q8.24



sampling errors

- sampling in time
  - introduces errors through ‘aliasing’
  - limits the range of frequencies able to be accurately captured
  - root of most common mistakes with sampled data



so how do I sample?

- heuristics
  - don't try to sample extremely small increments or values!
  - if capturing an "X"Hz signal, need to sample at least 2"X" Hz
  - changing sample rates is complicated, don't just drop every other sample
- for example, speech
  - majority of necessary energy in speech is located < 8000Hz
  - phones (for speech) typically capture at 16KHz or lower
  - good enough for speech, not music!

sanity check

- I need to detect an 80Hz signal
  - what sampling rate should we use?

- I want to detect a feather dropping next to the microphone
  - can the sound be detected?

making a sine wave

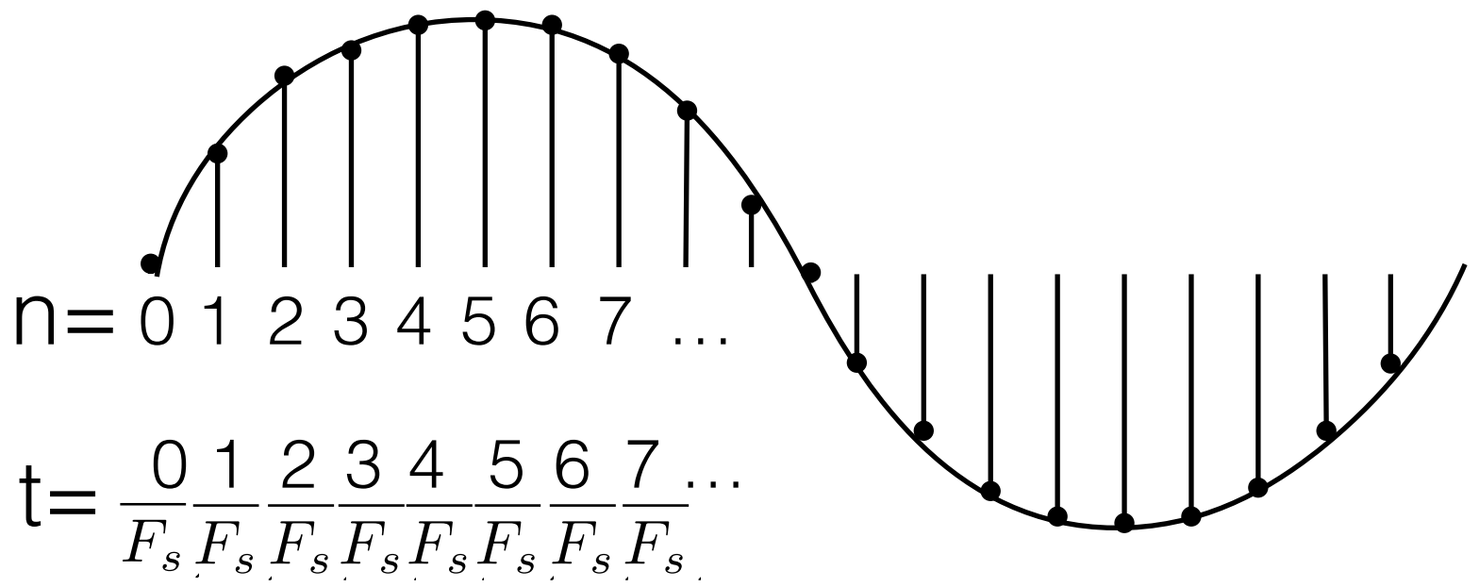
- we want to create a sine wave and play it to the speakers

$$g(t) = \sin(2\pi f t)$$ equation for sine wave

frequency in Hz

time in “seconds”

but we are working digitally, so we have an “index” in an array, not time!



$$g[n] = \sin\left(2\pi f\left(\frac{n}{F_s}\right)\right)$$

making a sine wave

$$g[n] = \sin\left(2\pi f\left(\frac{n}{F_s}\right)\right)$$ how to program this?

```
for (int n=0; n < numFrames; ++n)
 {
    data[n] = sin(2*M_PI*frequency*n/samplingRate);
 }
```

is this efficient?

```
float phase = 0.0;
double phaseIncrement = 2*M_PI*frequency/samplingRate;
 for (int n=0; n < numFrames; ++n)
 {
    data[n] = sin(phase);
    phase += phaseIncrement;
 }
```

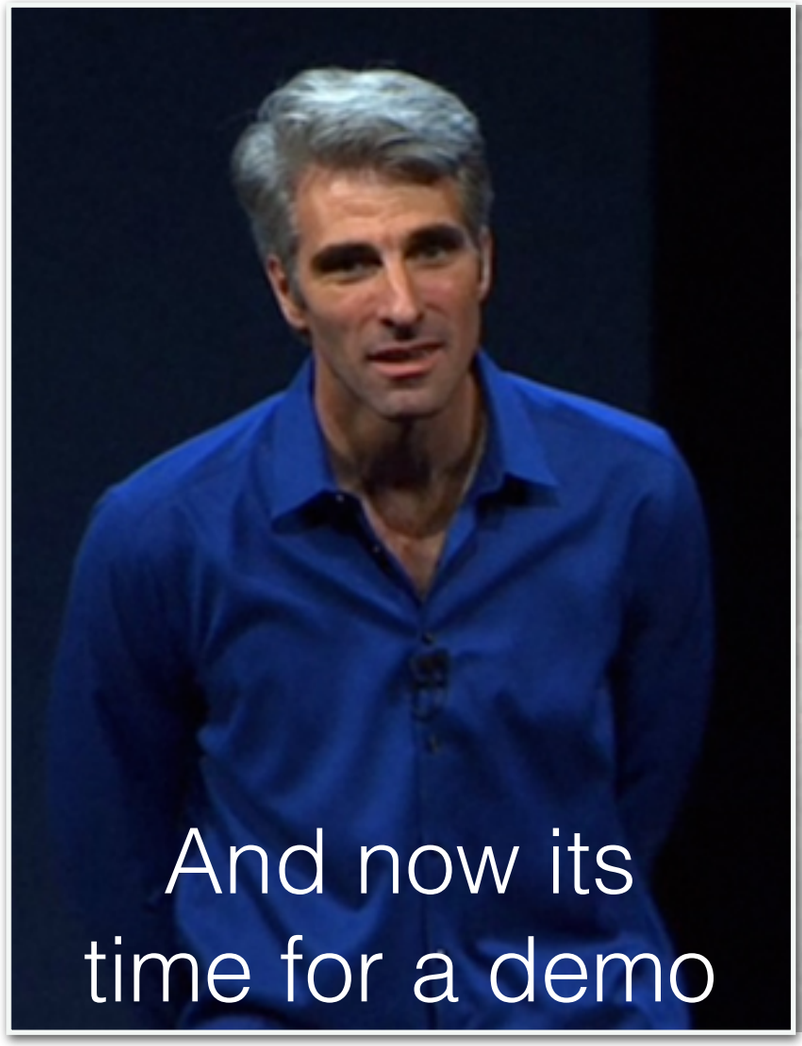
making a sine wave

- bringing it all together $g[n] = \sin\left(2\pi f\left(\frac{n}{F_s}\right)\right)$

```
frequency = 18000.0; //starting frequency
__block float phase = 0.0;
__block float samplingRate = audioManager.samplingRate;

[audioManager setOutputBlock:^(float *data, UInt32 numFrames, UInt32 numChannels)
 {
     double phaseIncrement = 2*M_PI*frequency/samplingRate;
     double sineWaveRepeatMax = 2*M_PI;
     for (int i=0; i < numFrames; ++i)
     {
         data[i] = sin(phase);

         phase += phaseIncrement;

         if (phase >= sineWaveRepeatMax) phase -= sineWaveRepeatMax;
     }

 }];
```

sample from the mic

- demo, play sine wave



and rolling stones, if time

the accelerate framework

- very powerful digital signal processing (DSP) library
  - look at vDSP Programming Guide on developer.apple.com for the complete API
- provides mathematics for performing fast DSP

input data | stride | scalar | output | array length

```
vDSP_vsmul(data, 1, &mult, data, 1, numFrames*numChannels);
```

```
void vDSP_vsmul (
    const float __vDSP_input1[],
    vDSP_Stride __vDSP_stride1,
    const float *__vDSP_input2,
    float __vDSP_result[],
    vDSP_Stride __vDSP_strideResult,
    vDSP_Length __vDSP_size
);
```

examples

```
[audioManager setInputBlock:^(float *data, UInt32 numFrames, UInt32 numChannels) {
        float volume = userSetVolumeFromSlider;
        vDSP_vsmul(data, 1, &volume, data, 1, numFrames*numChannels);
        [ringBuffer AddNewInterleavedFloatData:data withNumFrames:numFrames];
    }];
```

```
[audioManager setInputBlock:^(float *data, UInt32 numFrames, UInt32 numChannels) {

        // get the max
        float maxVal = 0.0;
        vDSP_maxv(data, 1, &maxVal, numFrames*numChannels);

        printf("Max Audio Value: %f\n", maxVal);

    }];
```

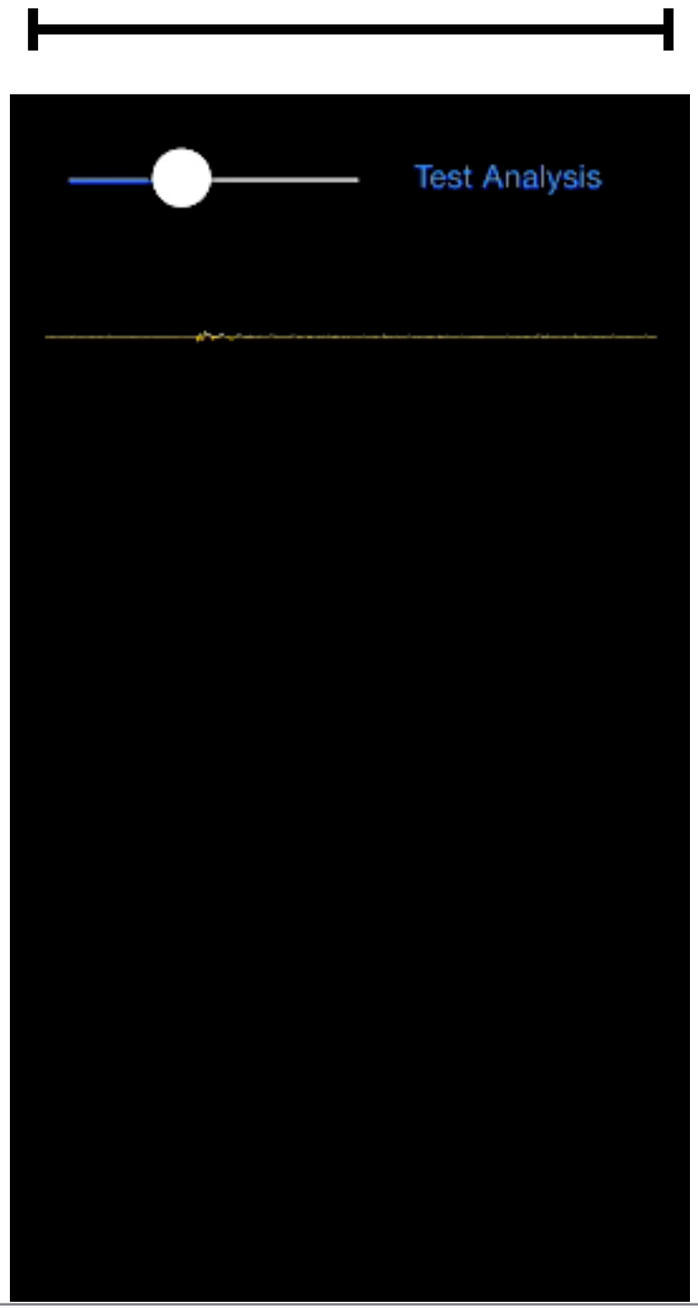
```
[audioManager setInputBlock:^(float *data, UInt32 numFrames, UInt32 numChannels)
 {
     vDSP_vsq(data, 1, data, 1, numFrames*numChannels);
     float meanVal = 0.0;
     vDSP_meanv(data, 1, &meanVal, numFrames*numChannels);
 }];
```

audio graphing

- we want to see the incoming samples
  - good for debugging
  - equalizers
  - oscilloscope type applications

how much data to show?

- sampling at 44.1kHz == 44100 samples per second



0.5 seconds is 22050 samples

display is 640 pixels wide

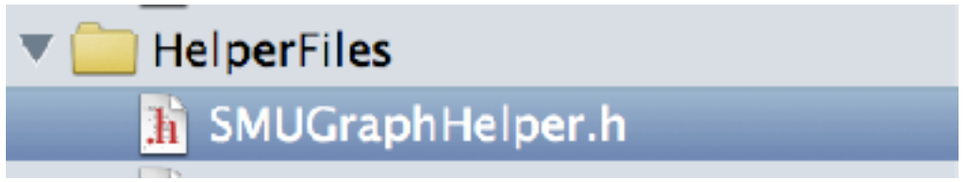
what if we want lots of graphs?

solution

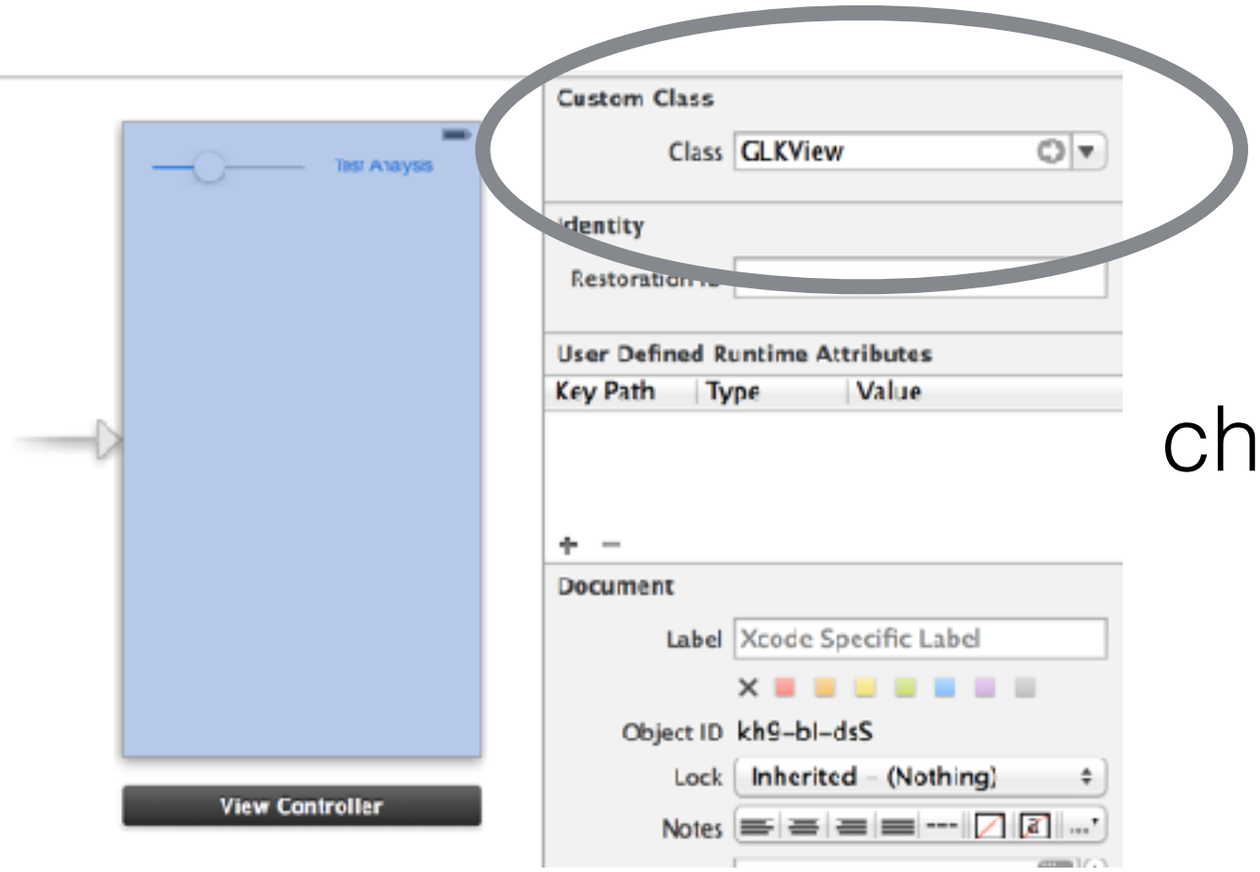
- use the GPU
- set vectors of data on a 2D plane
- let the renderer perform scaling, anti-aliasing, and bit blitting to screen
- …this is not a graphics course

easy solution

- use graph helper, which uses GLKView and GLKViewController



add .h and .m to project



change view to GLK View

the graph helper

```
#import <GLKit/GLKit.h>
@interface YourCustomViewController : GLKViewController

@property (strong, nonatomic) SMUGraphHelper *graphHelper;
```

inherit from open GL

declare property

```
When setting up:
   // start animating the graph
     _graphHelper = [[SMUGraphHelper alloc]initWithController:self
                                      preferredFramesPerSecond:15
                                                     numGraphs:1
                                                     plotStyle:PlotStyleSeparated
                                             maxPointsPerGraph:BUFFER_SIZE];
```

setup GLKViewController

```
    enum PlotStyle {
        PlotStyleOverlaid,
        PlotStyleSeparated
    };
```

```
In view did load:
[self.graphHelper setScreenBoundsBottomHalf];
[self.graphHelper setScreenBoundsTopHalf];
[self.graphHelper setFullScreenBounds];
[self.graphHelper setBoundsWithTop:(float) bottom:(float) left:(float) right:(float)];
```

bounds for screen different options

setting data

called for each draw to screen

```
//  override the GLKView draw function, from OpenGLES
- (void)glkView:(GLKView *)view drawInRect:(CGRect)rect {
    [self.graphHelper draw]; // draw the graph
}
```

prototypes for setting scatter data

```
-(void) setGraphData:(float*)
     withDataLength:(int)
      forGraphIndex:(int)
  withNormalization:(float)
       withZeroValue:(float)
```

```
-(void) setGraphData:(float*)
     withDataLength:(int)
      forGraphIndex:(int)
```

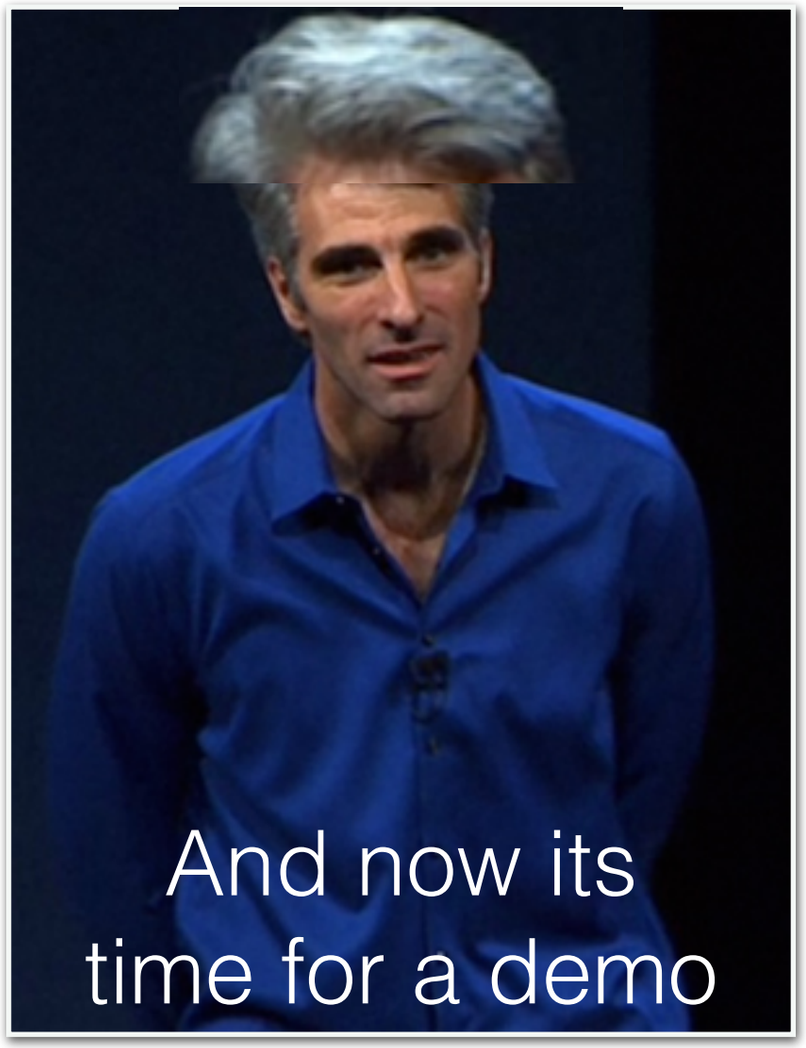
```
//  override the GLKViewController update function, from OpenGLES
- (void)update{
    // just plot the audio stream

    // get audio stream data
    float* arrayData = malloc(sizeof(float)*BUFFER_SIZE);
    [self.buffer fetchFreshData:arrayData withNumSamples:BUFFER_SIZE];

    //send off for graphing
    [self.graphHelper setGraphData:arrayData
                    withDataLength:BUFFER_SIZE
                     forGraphIndex:0];

    [self.graphHelper update]; // update the graph
    free(arrayData);
}
```

get data (shown here: from buffer)

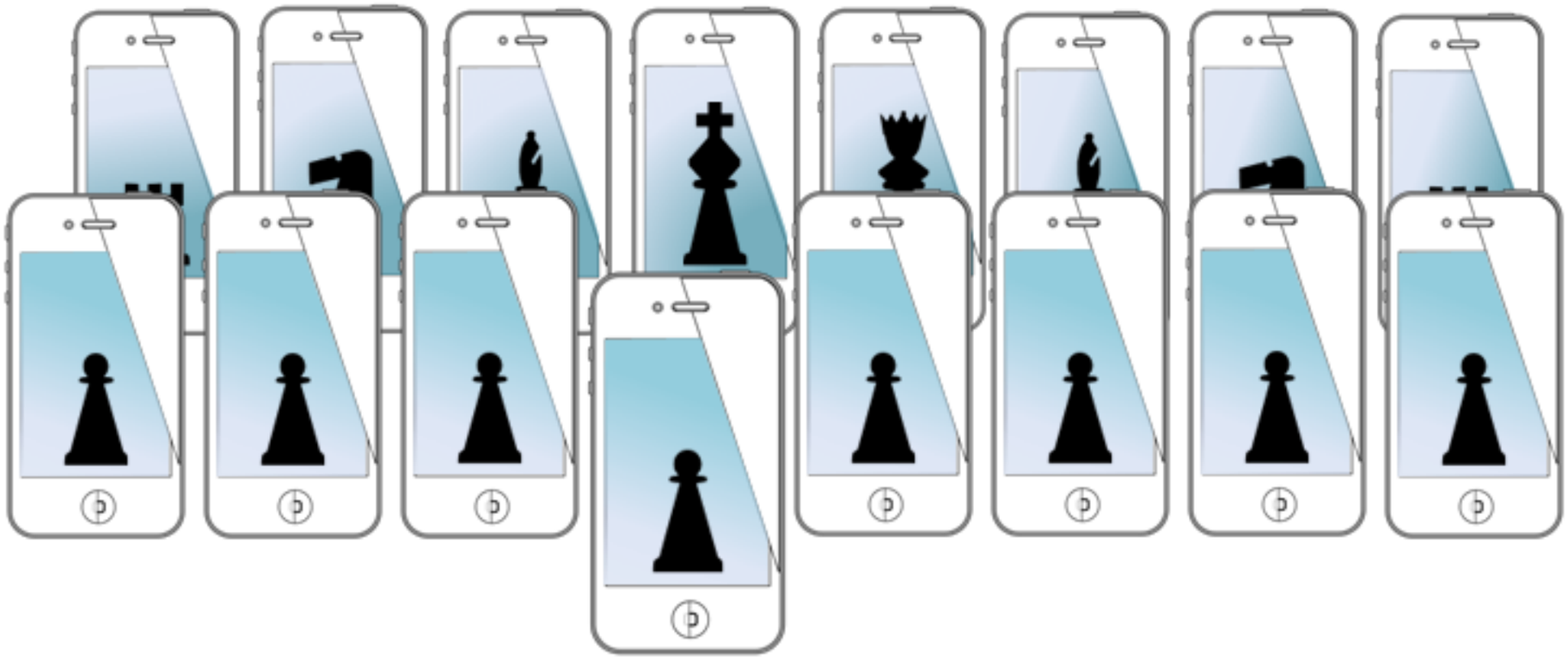
set data for 0th graph
no normalization

update render state and free memory

audio graphing demo!



MOBILE SENSING LEARNING



CSE5323 & 7323

Mobile Sensing and Learning

week 3, lecture b: audio graphing, sampled data, & accelerate

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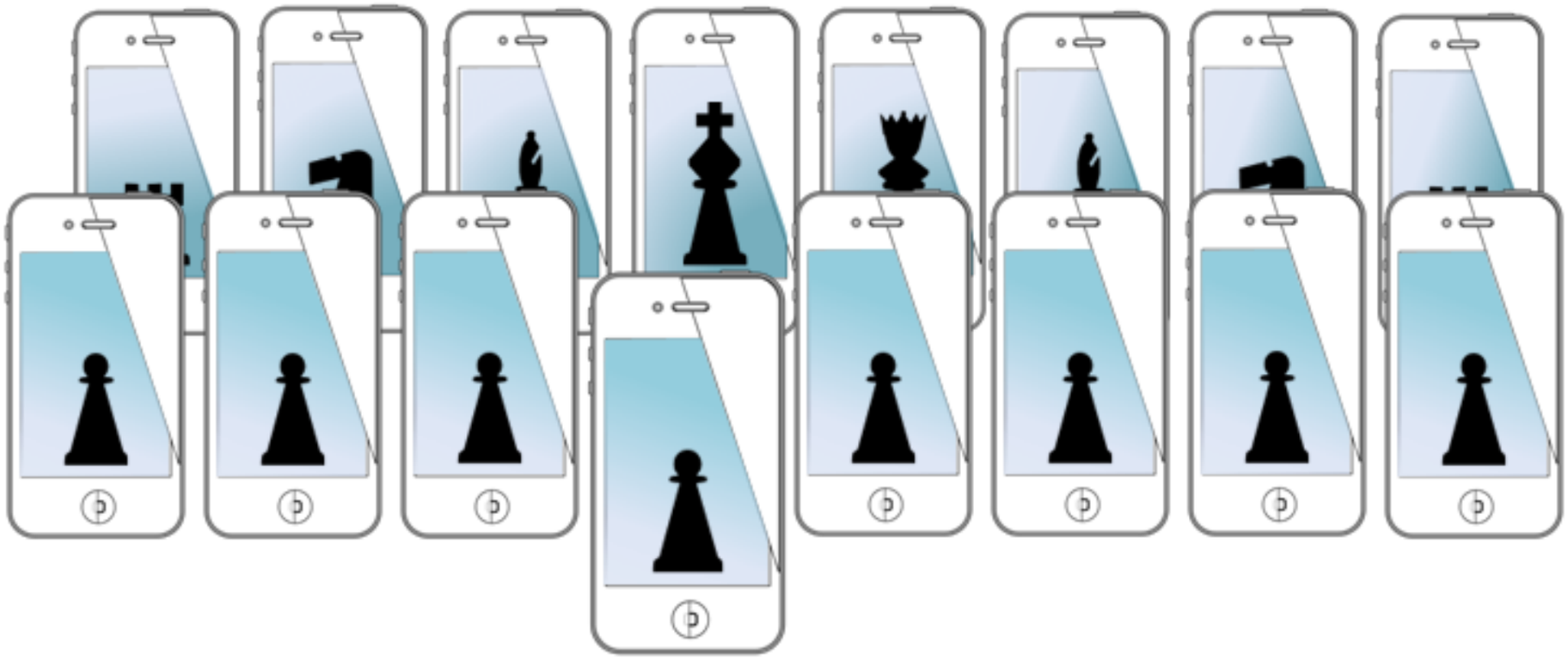
MOBILE SENSING LEARNING



CSE5323 & 7323

Mobile Sensing and Learning

week 4, video lecture: accelerate & FFT

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agenda (Video)

- high level: what is audio processing
- audio analysis: the FFT
- how to you use the FFT
- how to program the FFT
- an example: graphing whistles

processing audio

- lots of space to explore, processing signals is big!
  - **great reference**: “DSP First” by McClellan, Schafer, and Yoder
    - http://www.rose-hulman.edu/DSPFirst/visible3/contents/index.htm
  - **a reference for CSE’s**: “Signal Computing” by Stiber, Stiber, and Larson
    - http://faculty.washington.edu/stiber/pubs/Signal-Computing/
  - filtering
    - only let certain frequencies through
  - analysis
    - analyze characteristics of signal (like pitch or amplitude)
  - synthesis
    - play around with different ideas, and see what sounds good!
    - not just pure synthesis, but also manipulation (like guitar effects)

processing audio

- the accelerate framework is great for analyzing audio
- for instance, what is the max value of an audio chunk?

```
[audioManager setInputBlock:^(float *data, UInt32 numFrames, UInt32 numChannels) {

        // get the max
        float maxVal = 0.0;
        vDSP_maxv(data, 1, &maxVal, numFrames*numChannels);

        printf("M  x Audio Valu  : %f\n", maxVa );

}
```

input data from microphone

output: max

array length

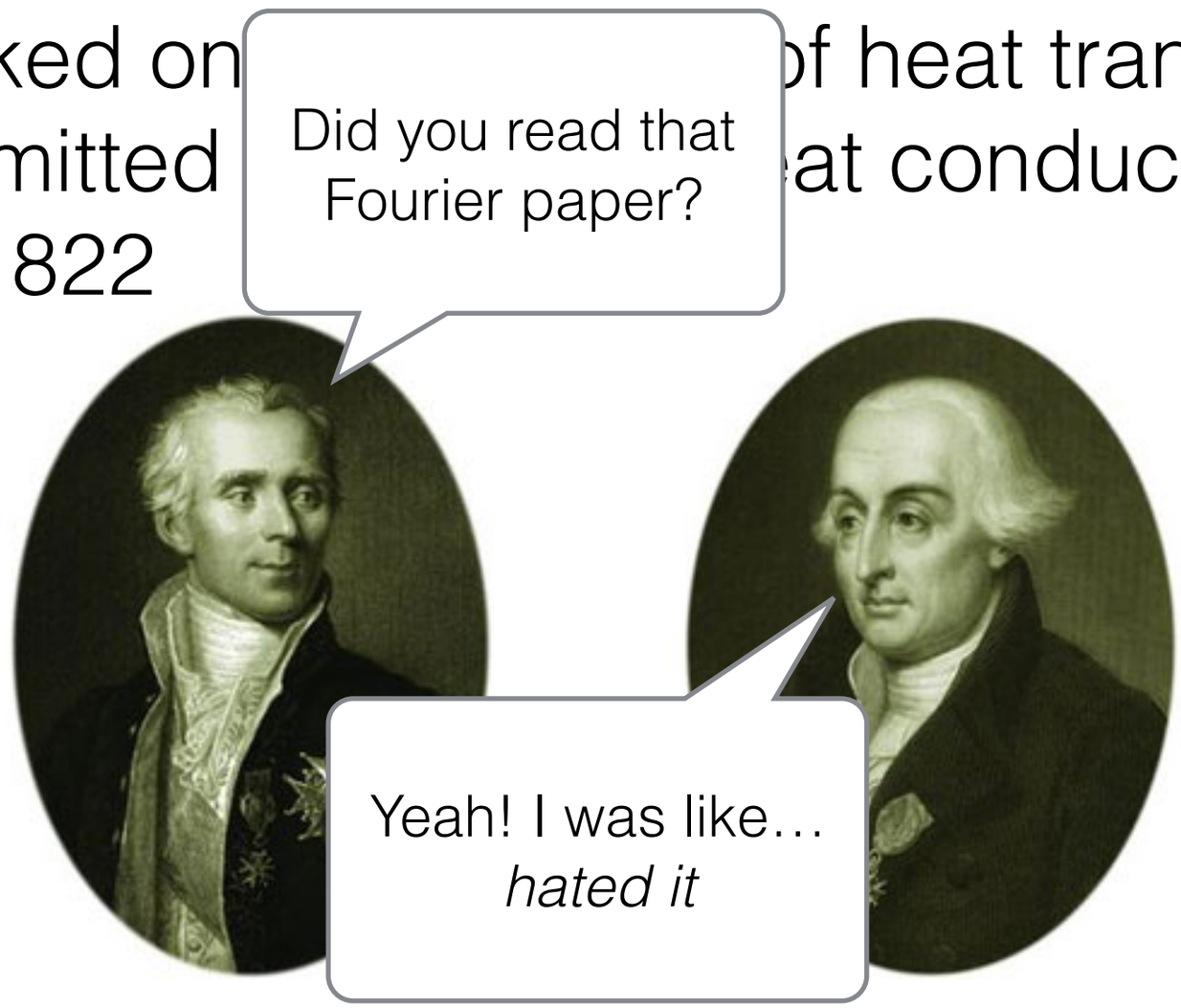
max of audio might be used to detect:

taps next to **phone**

- for now, we're going to stick with analysis
  - specifically, the **Fourier Transform**

Fourier

A mathematician
Worked on f heat transfer
Submitted at conduction
in 1822



Laplace Lagrange

Fourier

A mat
Worke
Subm
in 18

of heat transfer
eat conduction

He said all signals could be defined as a sum of simpler functions

Simpler functions!? Like sine waves? Come on!

Laplace

Lagrange

Fourier

He developed a method that allowed one to transform a signal into a **sum of sine waves**.

The transform was eventually called The Fourier Transform

~150 years later Cooley and Tukey would discover a way to calculate it really fast on a computer

Cooley

I created the FFT to commit espionage against the Russians!

I coined the term “bit”

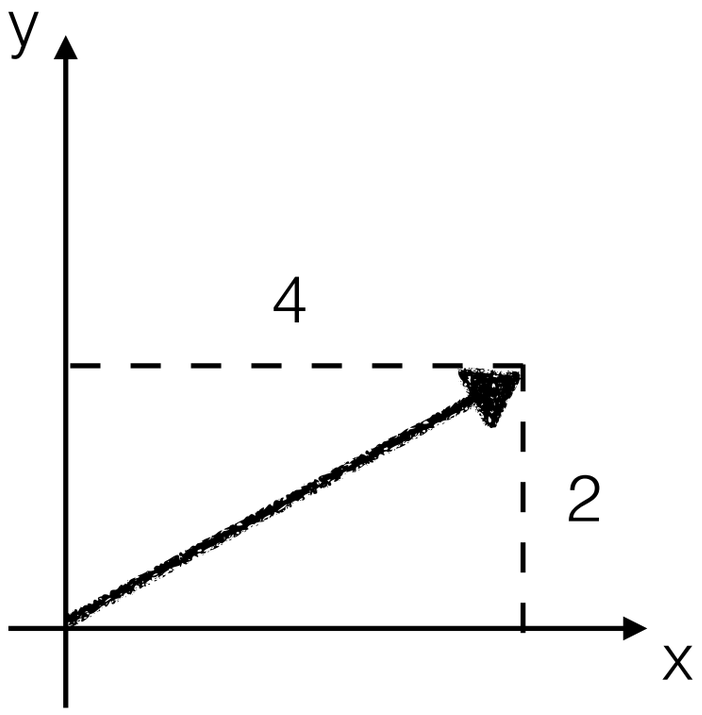
Tukey

the Fourier transform

- extremely useful, not just for signal junkies but also:
  - computer scientists, engineers, physicists, mathematicians, astronomers, oceanographers, health care professionals, etc.
- the Fourier Transform (FT) converts a time series into a frequency spectrum
  - the spectrum is an array of complex numbers which we will represent in polar form (i.e., with magnitude and phase)
  - each complex number represents a sinusoidal wave at a specific frequency
- we will use the FFT in the accelerate framework
  - complexity is $O(N \log_2(N))$ (for radix 2 FFT)

FT by vector intuition

think of it as a vector projection (intuitively)



(4,2)

where did these numbers come from?

$\vec{x}$ $(1,0) \bullet (4,2) = 4$

$\vec{y}$ $(0,1) \bullet (4,2) = 2$

point by point multiplication and add it up!

tells us "how much" of the vector is the result of these vectors

$\vec{x}$

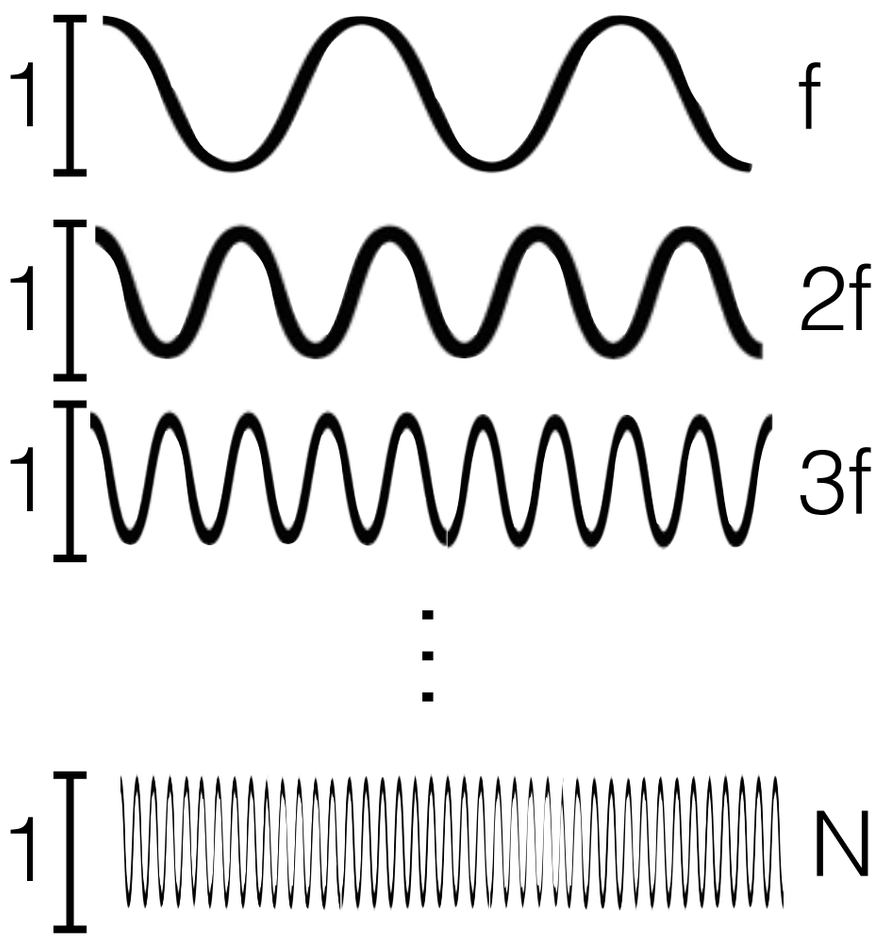
$\vec{y}$

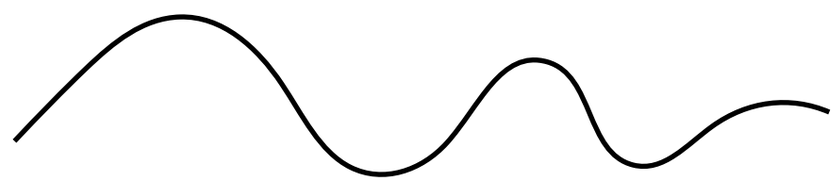
$\vec{x} \bullet \vec{y} = 0$

$|\mathbf{x}| = |\mathbf{y}| = 1$

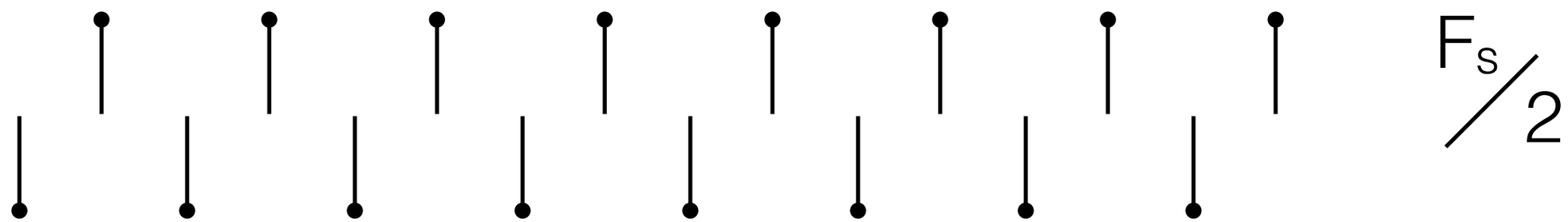
FT by sine waves

what if the orthogonal vectors were functions?





2.5	10°
0.1	45°
0.5	-12°
⋮	⋮
.005	0°



FT by the numbers

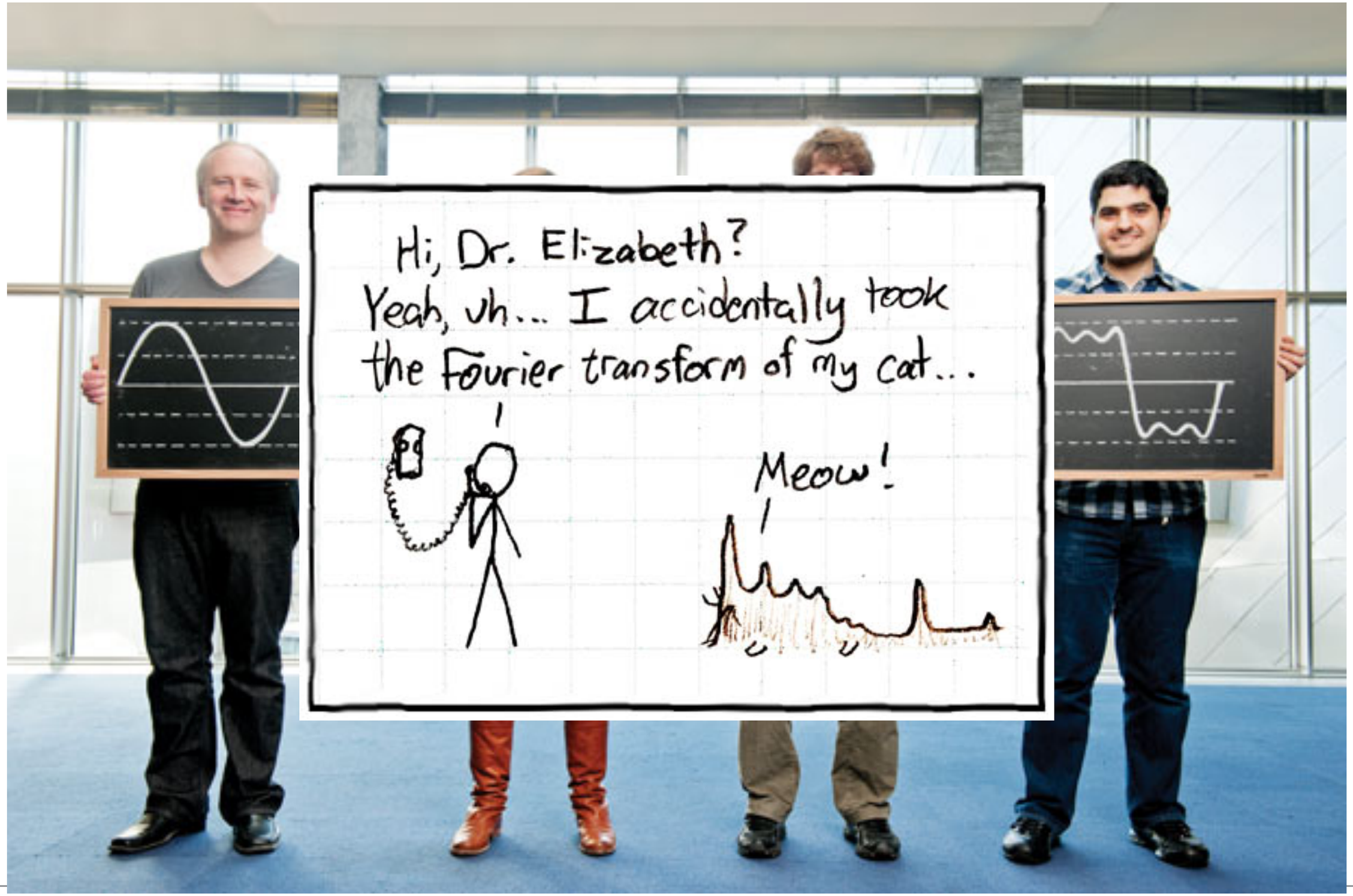
content	frequency									
	0f	0.7	0°						=	0.7
	1f	2.5	10°	+	2.5	-10°	-1f		~	2.5 cos(2*pi* (f) t+10°)
	2f	0.1	45°	+	0.1	-45°	-2f		~	0.1 cos(2*pi* (2f) t+45°)
	3f	0.5	-12°	+	0.5	12°	-3f		~	0.5 cos(2*pi* (3f) t-12°)
	⋮	⋮	⋮		⋮	⋮				
	N f	.005	0°	+	.005	0°	-N f		~	0.005 cos(2*pi* (Nf) t)

FT by video

http://en.wikipedia.org/wiki/User:LucasVB/Gallery

FT by hilarity

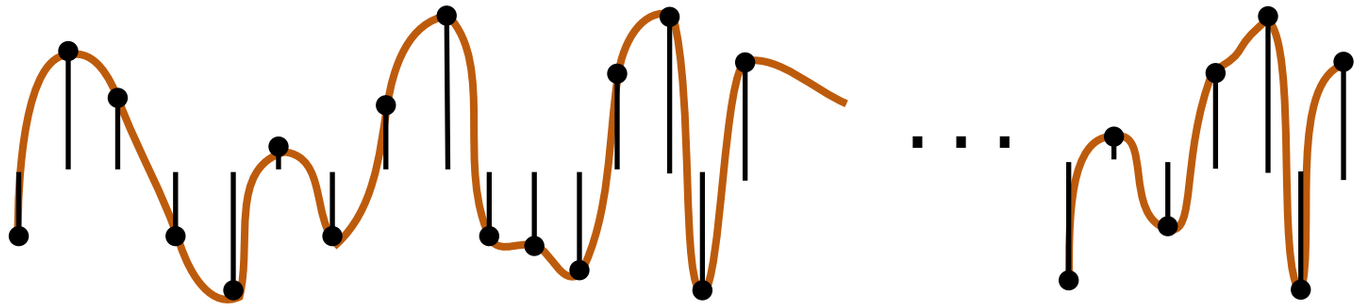
http://www.preposterousuniverse.com/blog/2014/11/27/thanksgiving-9/



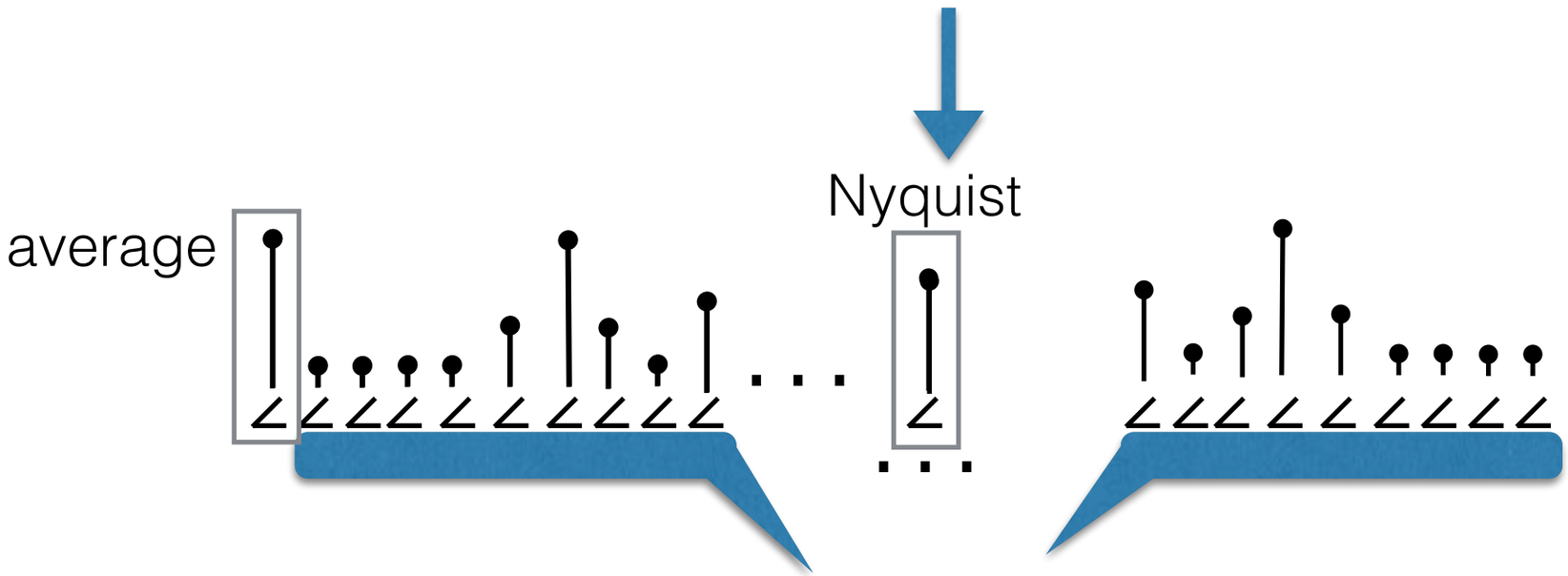
the FFT

- we will use the FFT in the accelerate framework
  - complexity is O(N $\log_2$(N))
- the FFT takes an array of numbers and gives you back an array of complex numbers
  - both input and output arrays are the same length
  - the indices of the input array denote increasing time
  - the indices of the output array denote increasing frequency
- don't quite get the math? so what? understand the output array…

time and frequency



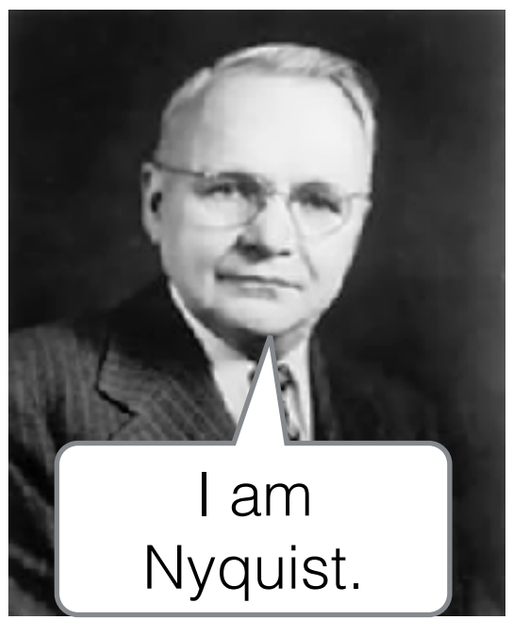
time array, N=512



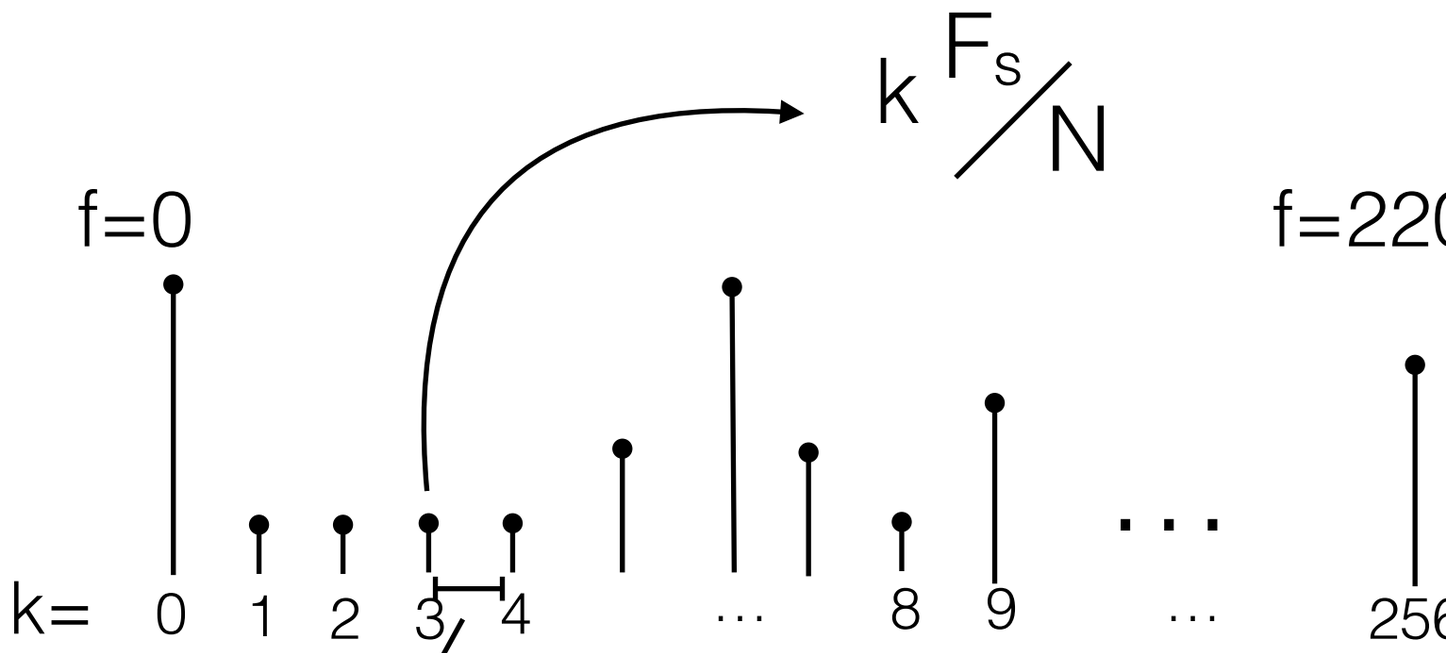
FFT array, N = 512

magnitude, even symmetry

magnitude spectrum



time and frequency



FFT array, N = 512

Spectrum, N = 257

Fs = 44100 Hz

$$df = \frac{F_s/2}{N/2} = F_s/N = 44100/512 = \sim 86\ \text{Hz}$$

using the FFT

raw audio

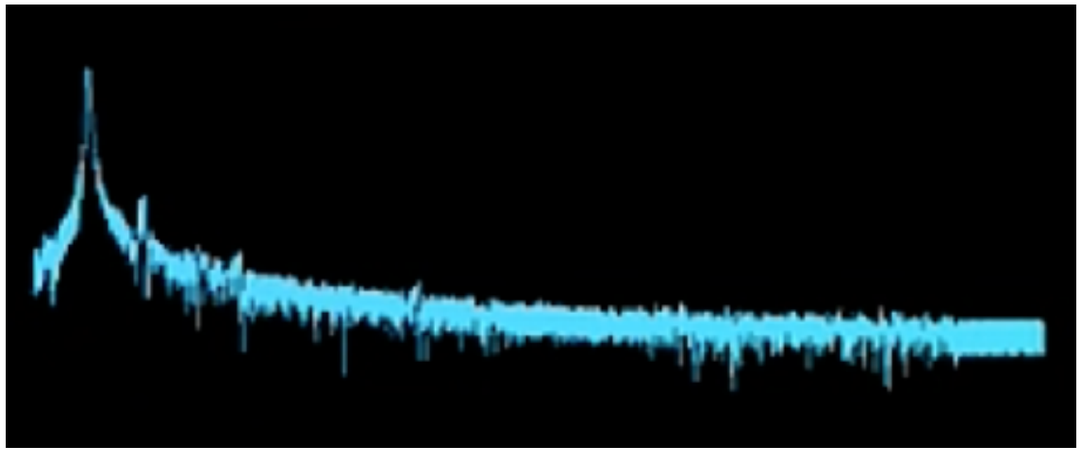
magnitude FFT

100

0

magnitude FFT
in dB

$20 \log_{10}(|FFT|)$

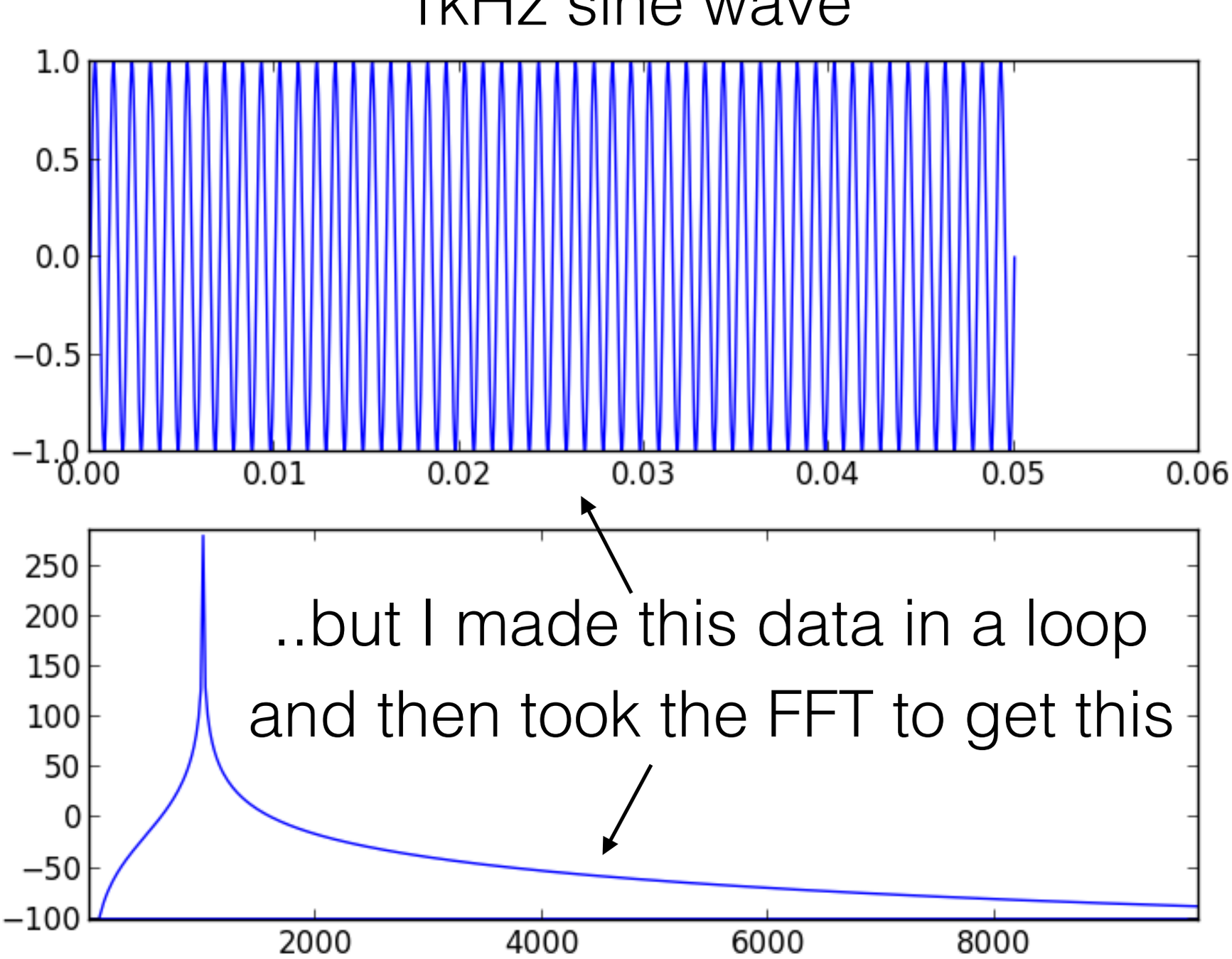


40

-30

some fft examples

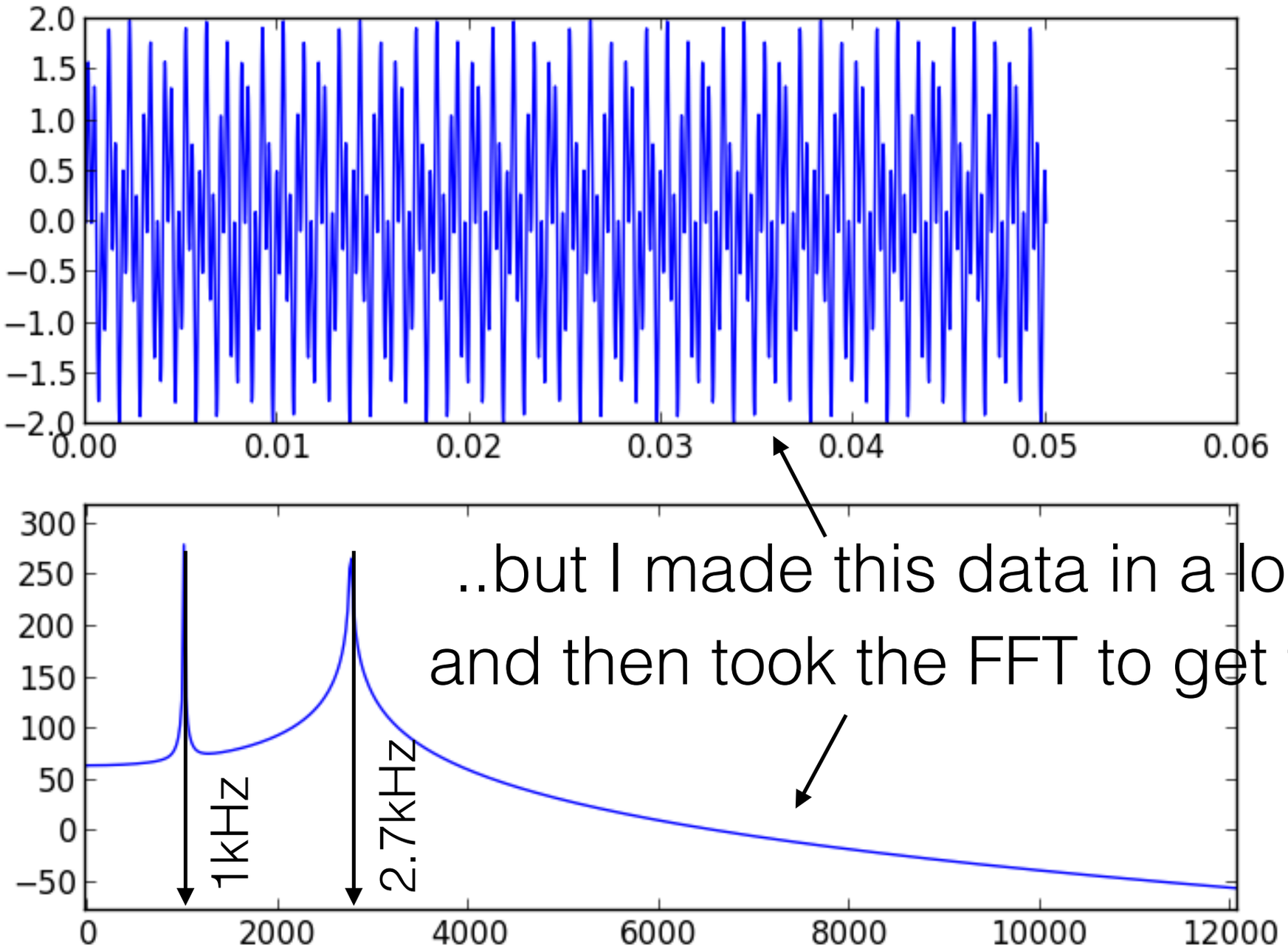
this is just a plot of an array of values sampled **over equal time intervals**



this is just a plot of an array of FFT values **its indices are equally sampled over frequency**

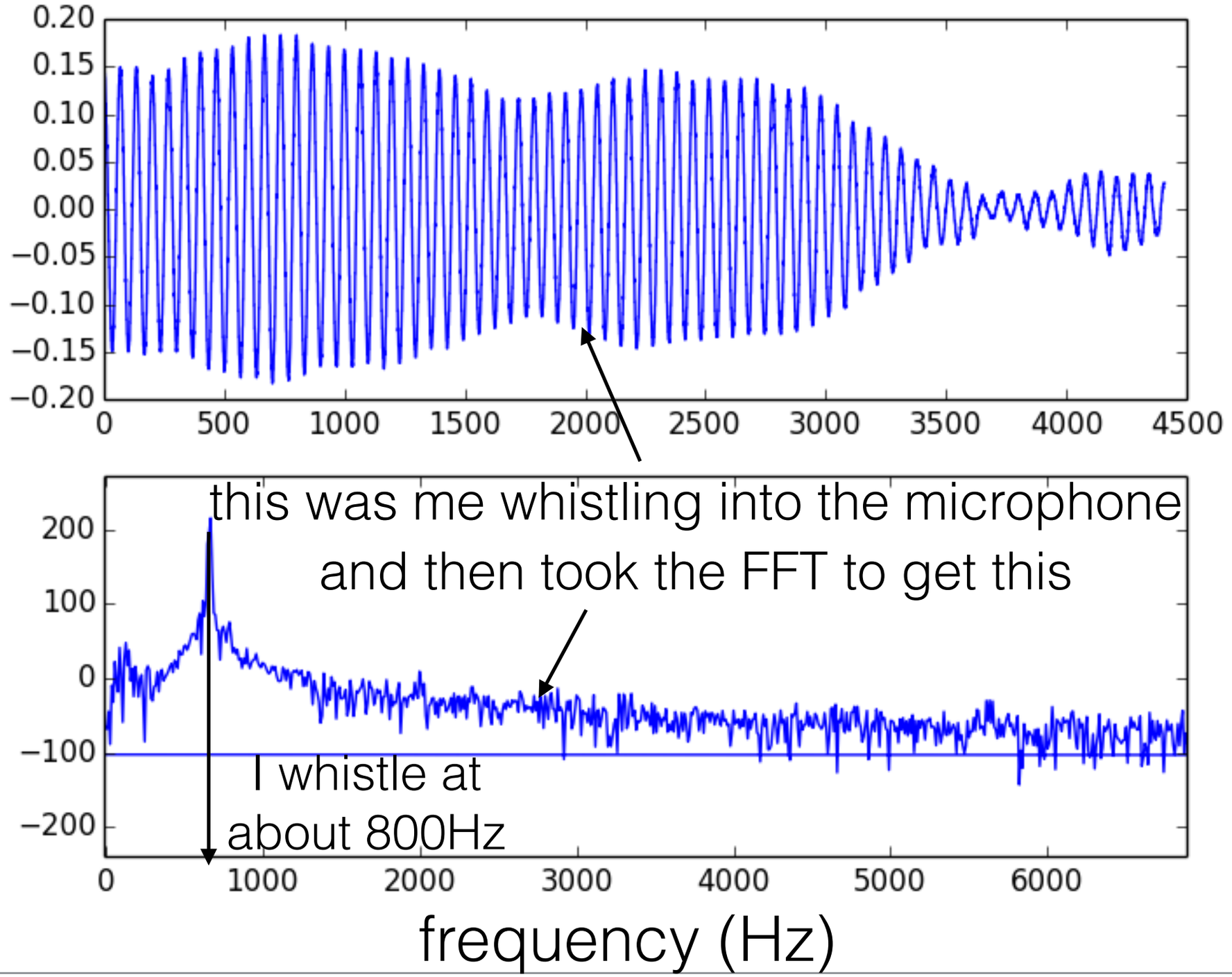
some fft examples

1kHz sine wave +2.7kHz sine wave



some fft examples

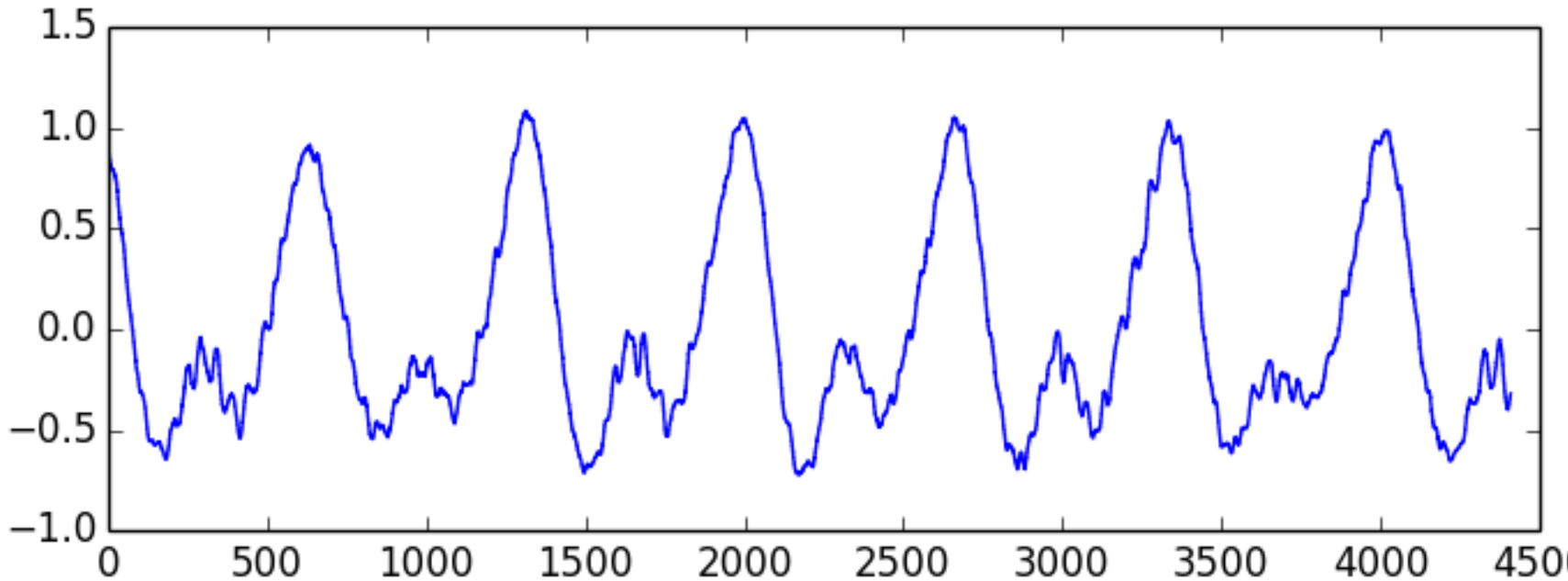
a person whistling

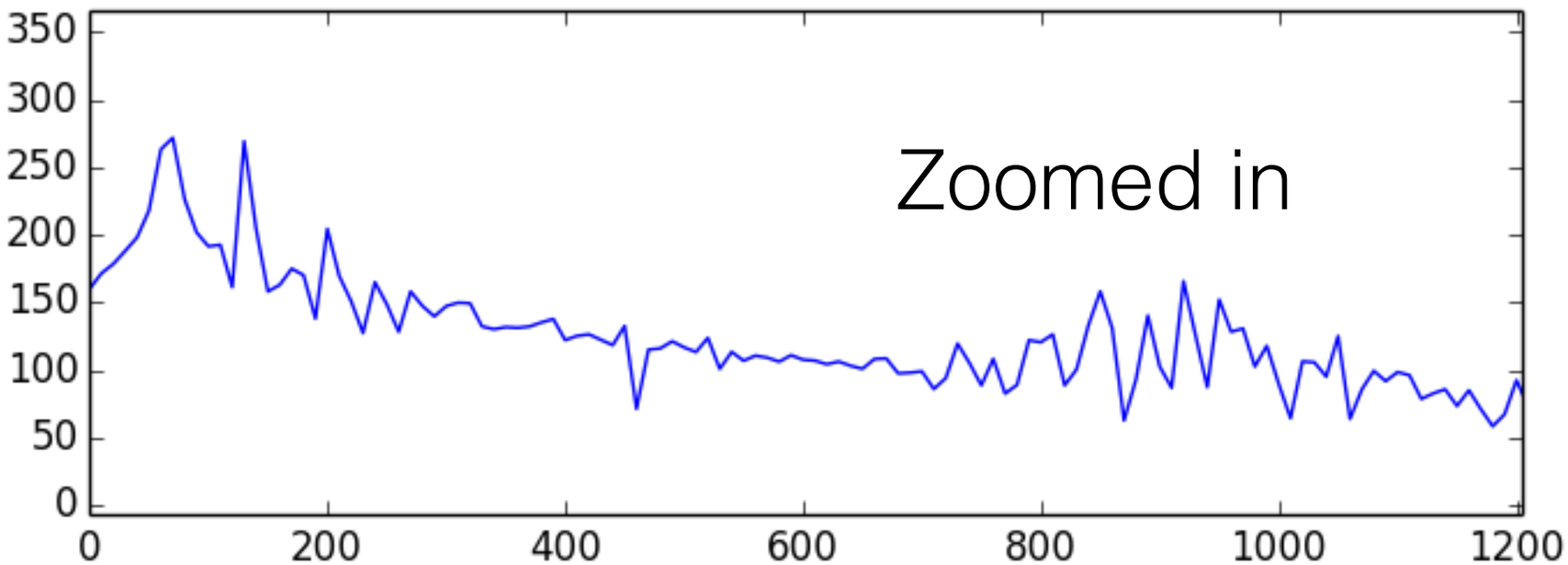


some fft examples

humming

oooooooooooo

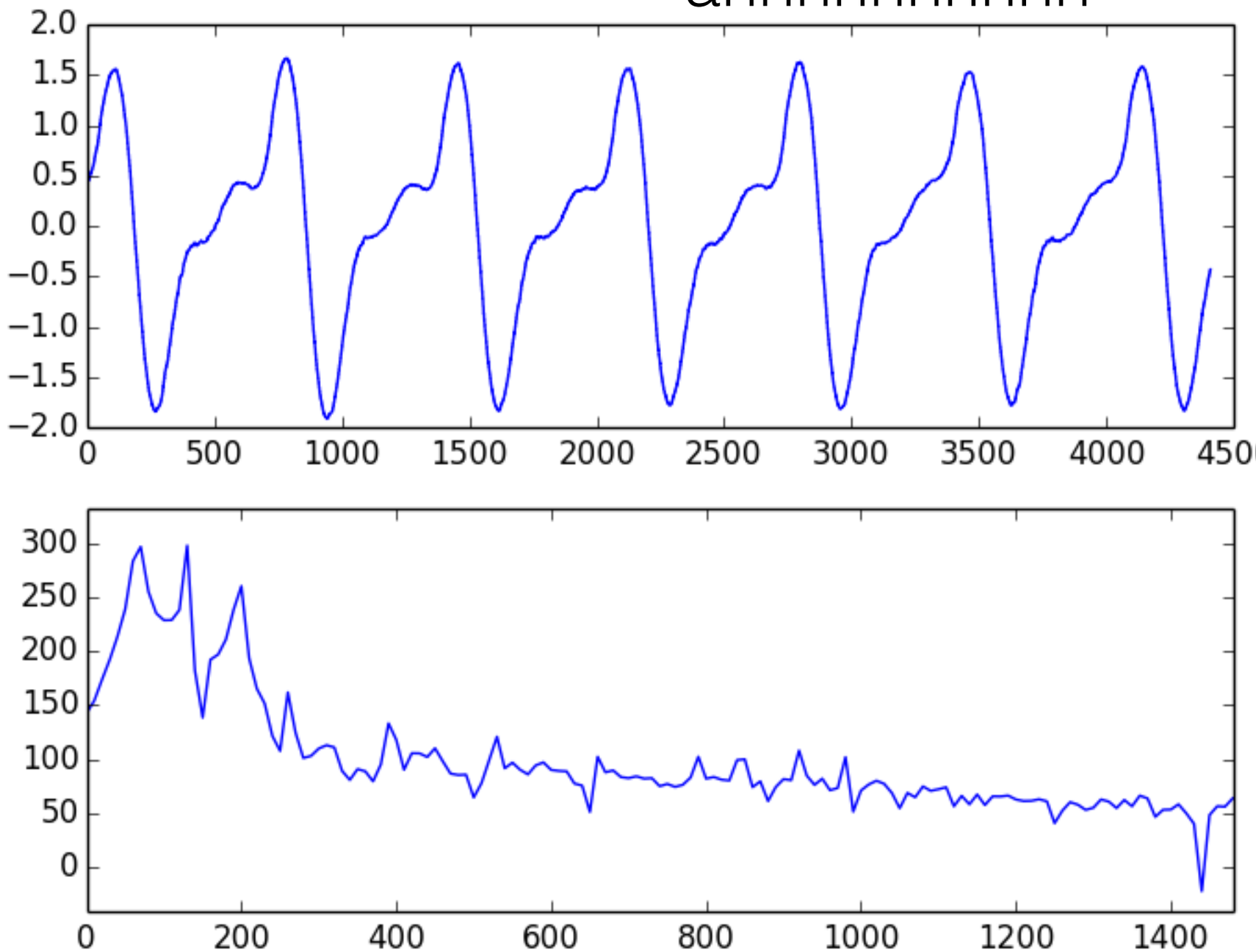




some fft examples

humming again

ahhhhhhhhhh

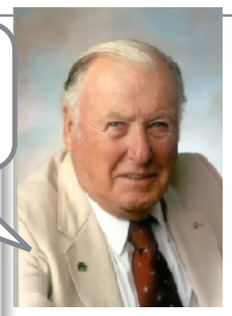


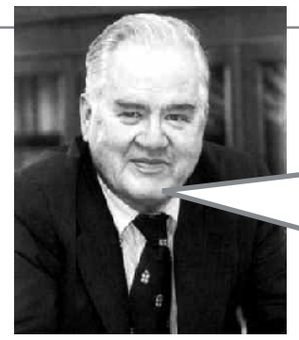
programming the FFT

```
#import "SMUFFTHelper.h"

@property (strong, nonatomic) FFTHelper *fftHelper;
```

Yeah!





This is how to use our stuff

```
//setup the fft
_fftHelper = [[FFTHelper alloc]initWithFFTSize:BUFFER_SIZE];
```

fft size == window size

```
float* fftMagnitude = malloc(sizeof(float)*BUFFER_SIZE/2);

// take forward FFT
[self.fftHelper performForwardFFTWithData:arrayData
                andCopydBMagnitudeToBuffer:fftMagnitude];

free(fftMagnitude);
```

input array
N=window size

magnitude out
N=half window size

highly optimized!! even using tricks we have not discussed

programming the FFT

```
#import "SMUFFTHelper.h"
@property (strong, nonatomic) FFTHelper *fftHelper;
```

```
//setup the fft
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```

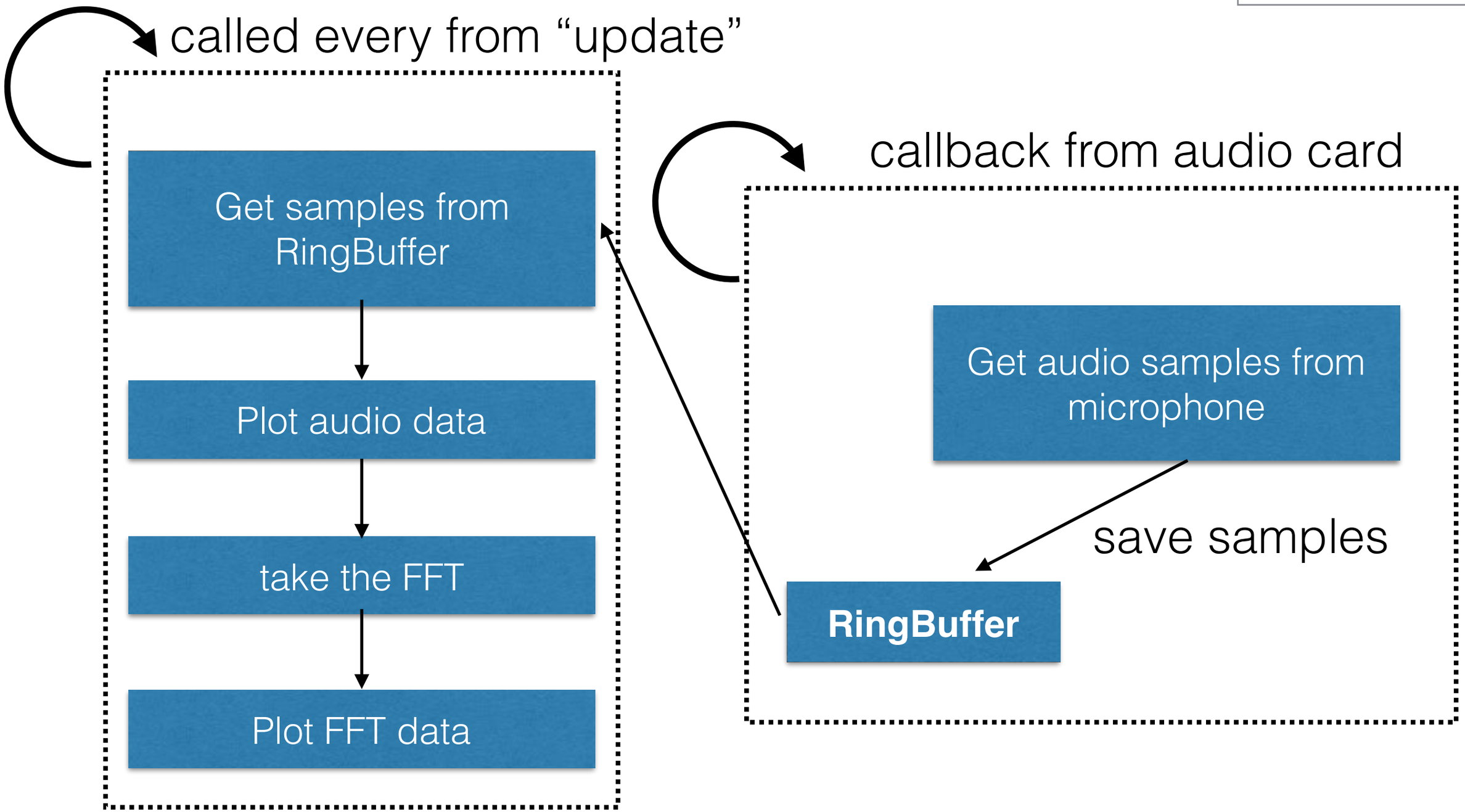
input array
N=window size

magnitude out
N=half window size

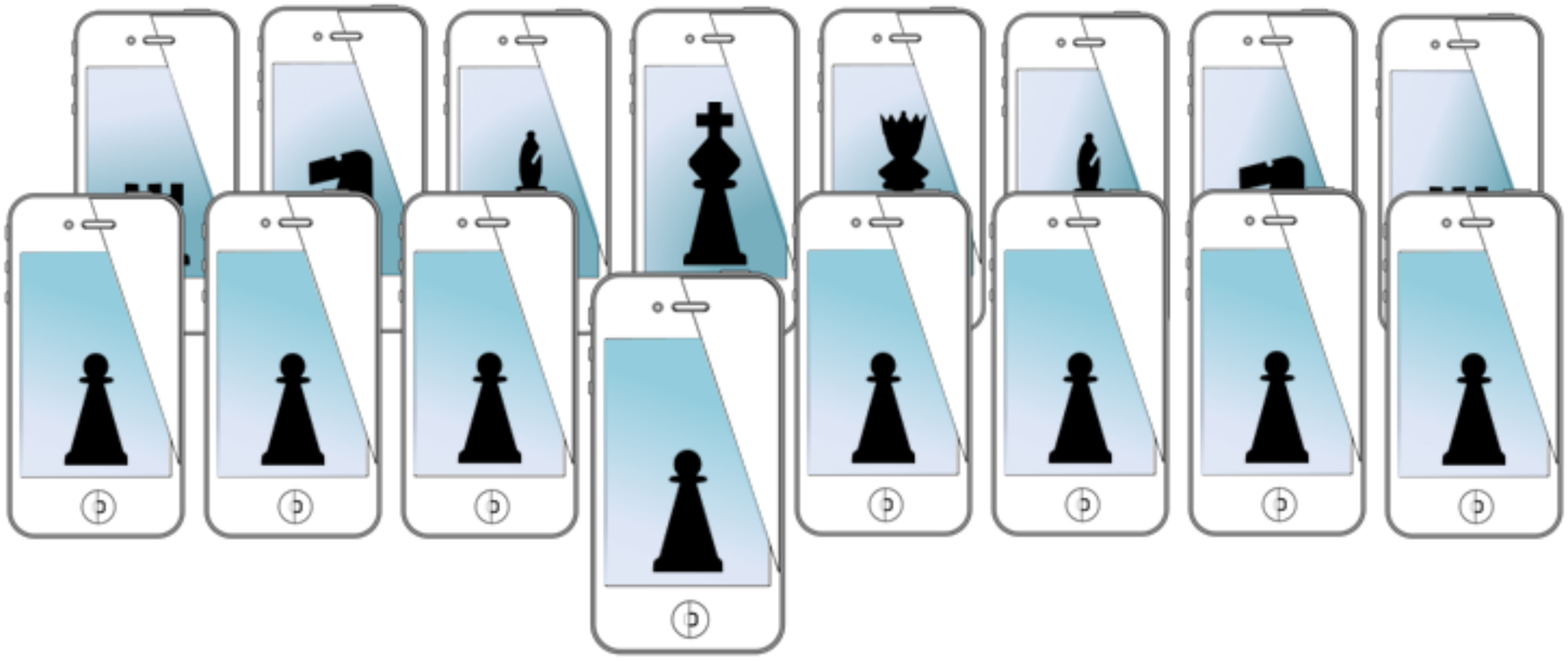
FFT tradeoffs

- sampling rate
  - dictates the time between each sample, (1 / sampling rate)
  - max frequency we can measure is half of sampling rate
- resolution in frequency
  - tradeoff between length of FFT and sampling rate
- each frequency “bin” is an index in the FFT array
  - each bin represents (Fs / N) Hz
  - what does that mean for 6 Hz accuracy?

The program


called every from “update”
Get samples from RingBuffer
Plot audio data
take the FFT
Plot FFT data
callback from audio card
Get audio samples from microphone
save samples
RingBuffer

MOBILE SENSING LEARNING



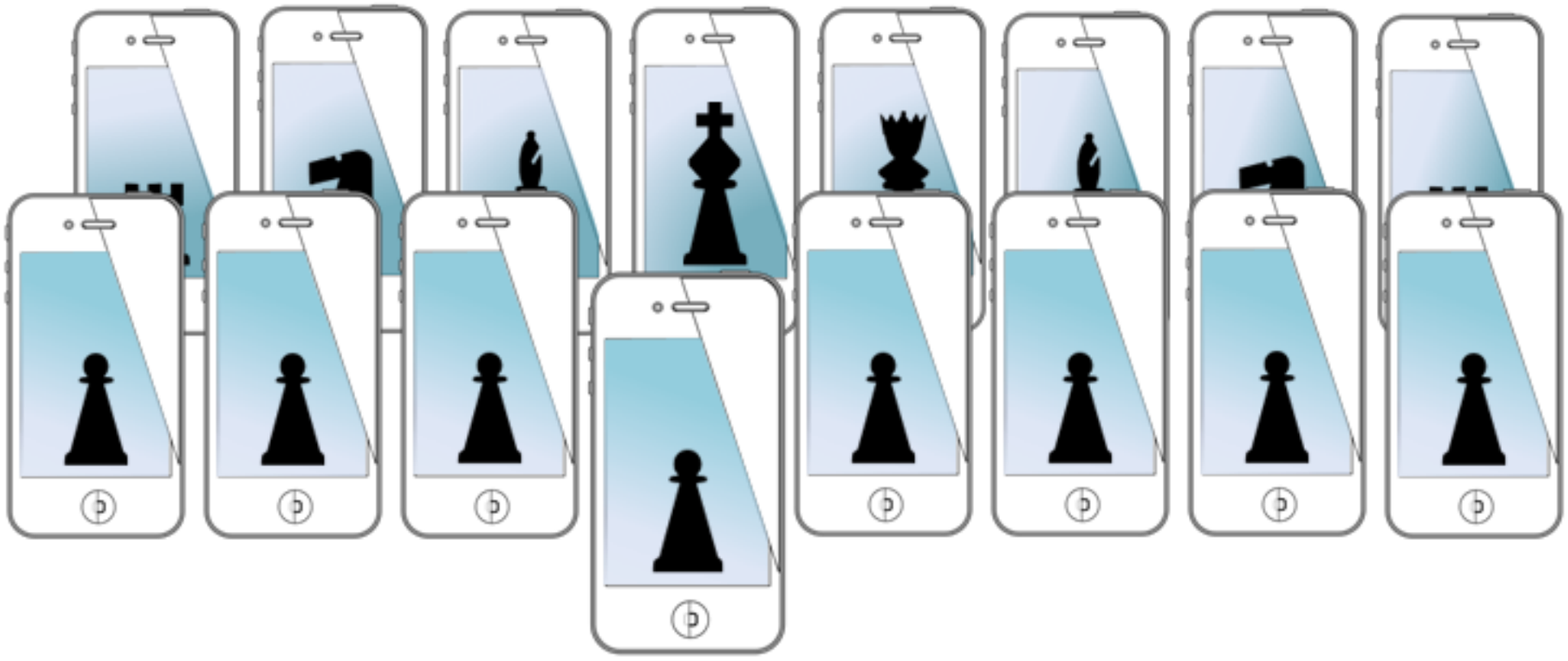
CSE5323 & 7323

Mobile Sensing and Learning

week 4, video lecture: accelerate & FFT

Eric C. Larson, Lyle School of Engineering,
Computer Science and Engineering, Southern Methodist University

MOBILE SENSING LEARNING



CSE5323 & 7323

Mobile Sensing and Learning

Supplemental Slides: filtering and windowing

Eric C. Larson, Lyle School of Engineering,
Computer Science and Engineering, Southern Methodist University

Supplemental Slides

- these slides were removed from the course because of their complexity
- you need a good background in signal representation and time series analysis to really understand these slides

Optional Concepts not Covered in Course Anymore

filters!

- we will cover what we can…

signals and systems

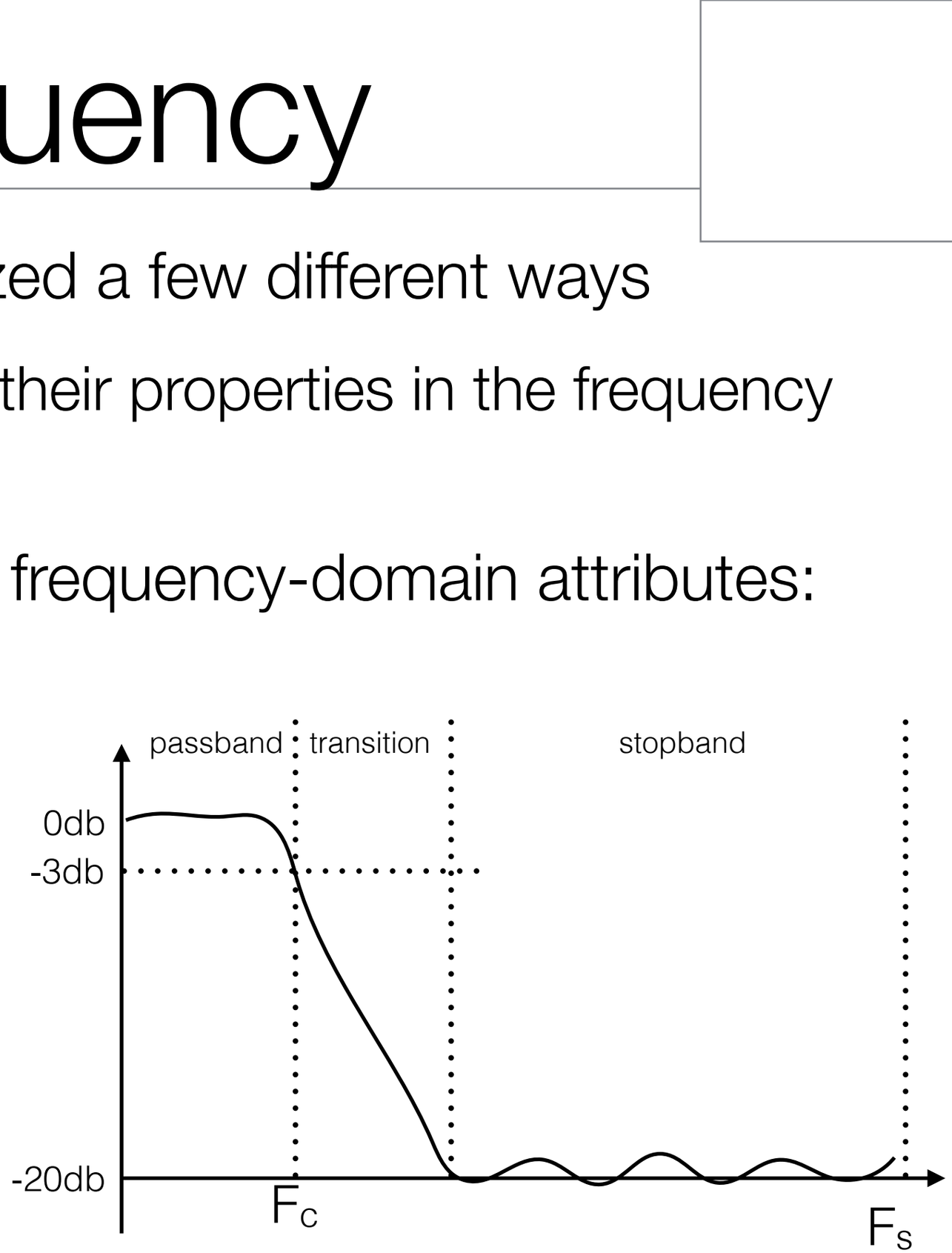
- signals are collections of sampled data (arrays)
  - such as audio, accelerometer, etc.
  - can also be 2D, like images
- systems are objects which manipulate signals
  - characterized by their “input/output” relationships
  - we say “x[n] is passed through H, resulting in y[n]”

filters

- filters are systems which manipulate frequencies
  - certain frequencies to pass through, but not others
  - “lowpass” filter allows low frequencies to pass through
  - “highpass” filter likewise allows high frequencies through
- keep in mind: no filter is perfect!!
  - no filter will pass everything you want while stopping everything you don’t
  - everything is a balance between different parameters you can control
- we won’t study how to design filters
  - we will study properties of filters and how to use them
  - so we need to know what filters can and cannot do

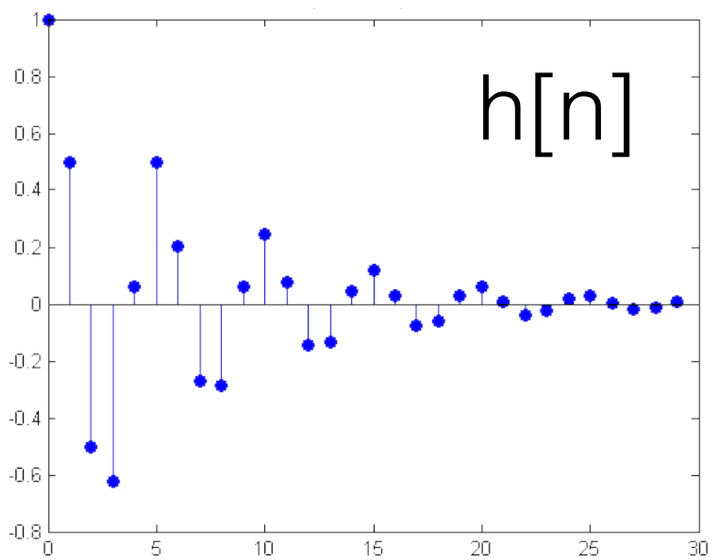
filters in frequency

- filters can be characterized a few different ways
  - let's start by looking at their properties in the frequency domain
- filters have the following frequency-domain attributes:
  - passband gain
  - passband bandwidth
  - stopband attenuation
  - transition bandwidth



filters in time

- filter are also signals (time series)
  - the series is called the “impulse response” of the filter
  - the frequency-domain plots are just Fourier transforms of the impulse response (magnitude)
- the time-domain property we care about is length
- everything else is best left to a filter design course



so how to design a filter?

- scipy.signal in python (try to use remez)
- decent tutorial:
  - http://mpastell.com/2010/01/18/fir-with-scipy/
- matlab
  - fdatool

```
from pylab import *
import scipy.signal as signal
n = 61
a = signal.firwin(n, cutoff = 0.3,
                  window = "hamming")
```

- lots of other places!

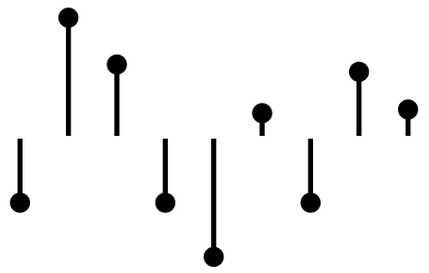
filtering by convolution

- we apply a filter using **convolution**
  - convolution allows us to combine frequency properties of two signals without taking an FFT
- basic principle:
  - convolution in time is multiplication in frequency
  - so the filter's frequency response will be multiplied by the frequency response of the signal

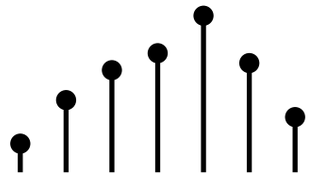
$$y[n] = \sum_{i=0}^{N-1} h[n-i]x[i]$$

convolution

x[i]

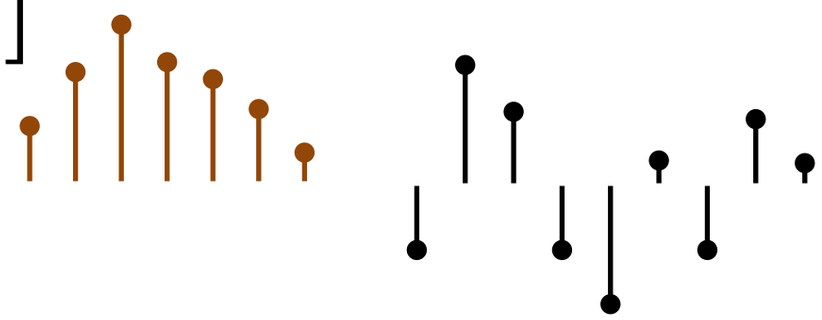


h[i]

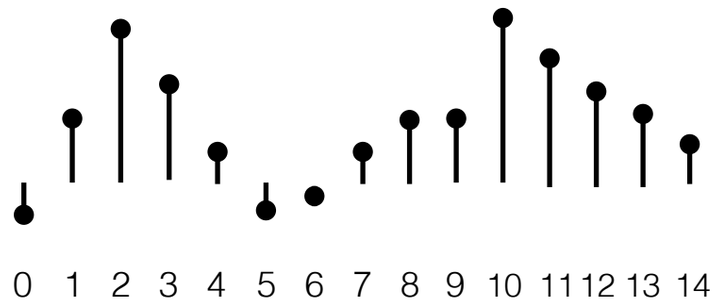


$$y[n] = \sum_{i=0}^{N-1} h[n-i]x[i]$$

h[-i]



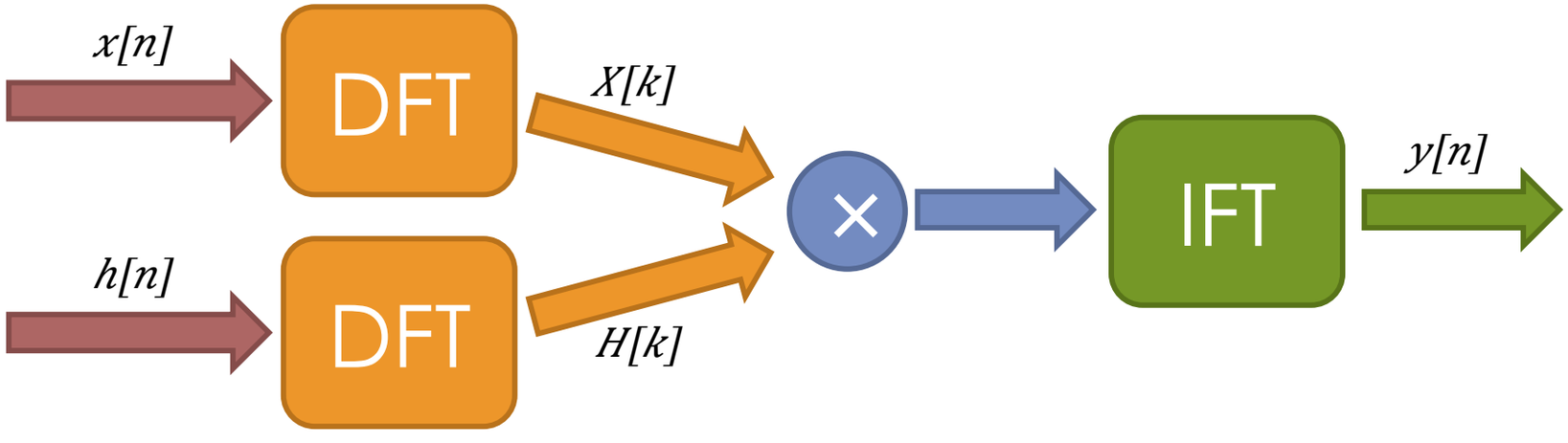
y[n]



length N M N+M-1

convolution efficiency

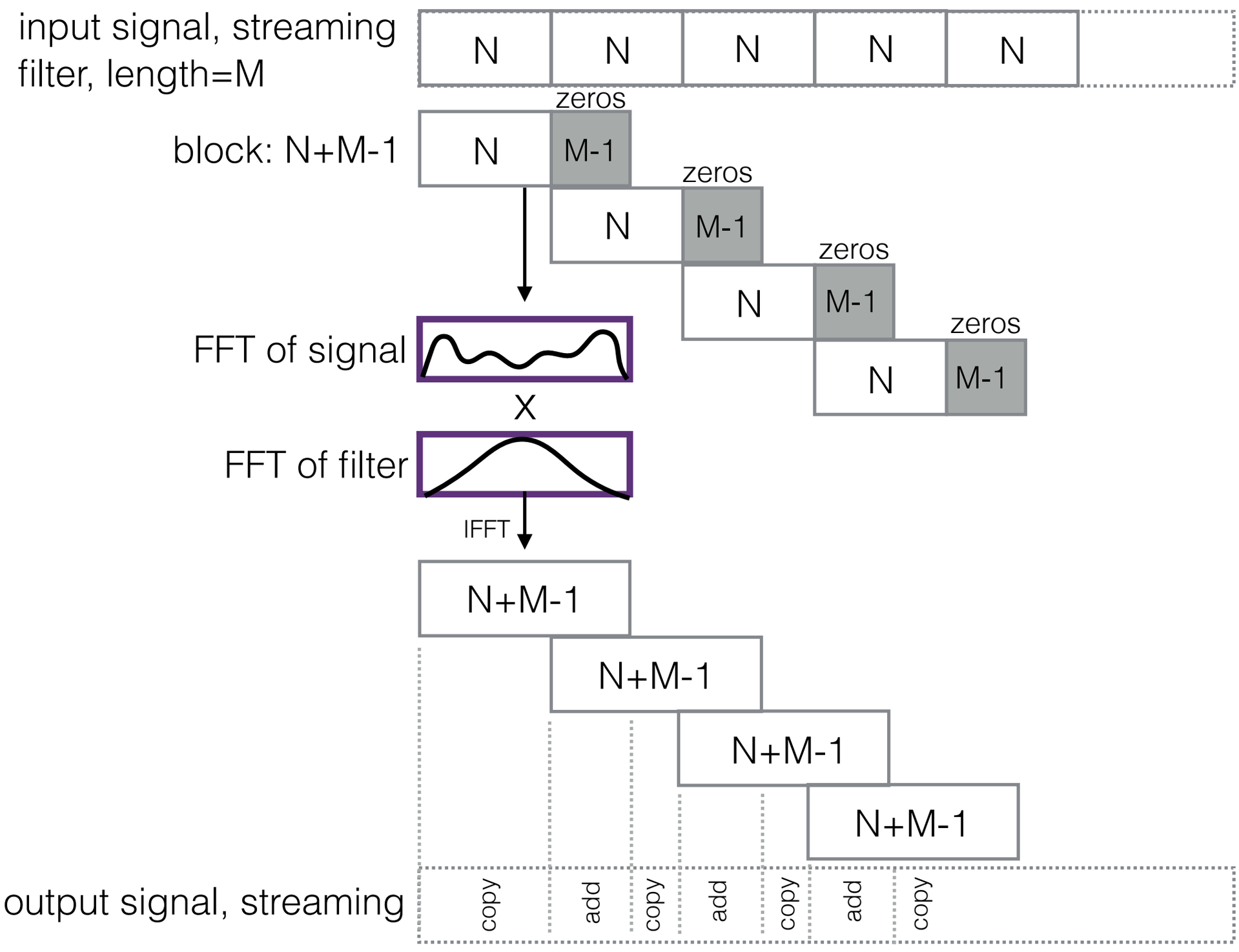
- algorithmic complexity
  - convolution is not particularly efficient, O(NxM)
- to convolve faster, use that Fourier property:
  - “convolution in time is multiplication in frequency”
- why not just multiply to begin with!



its circular

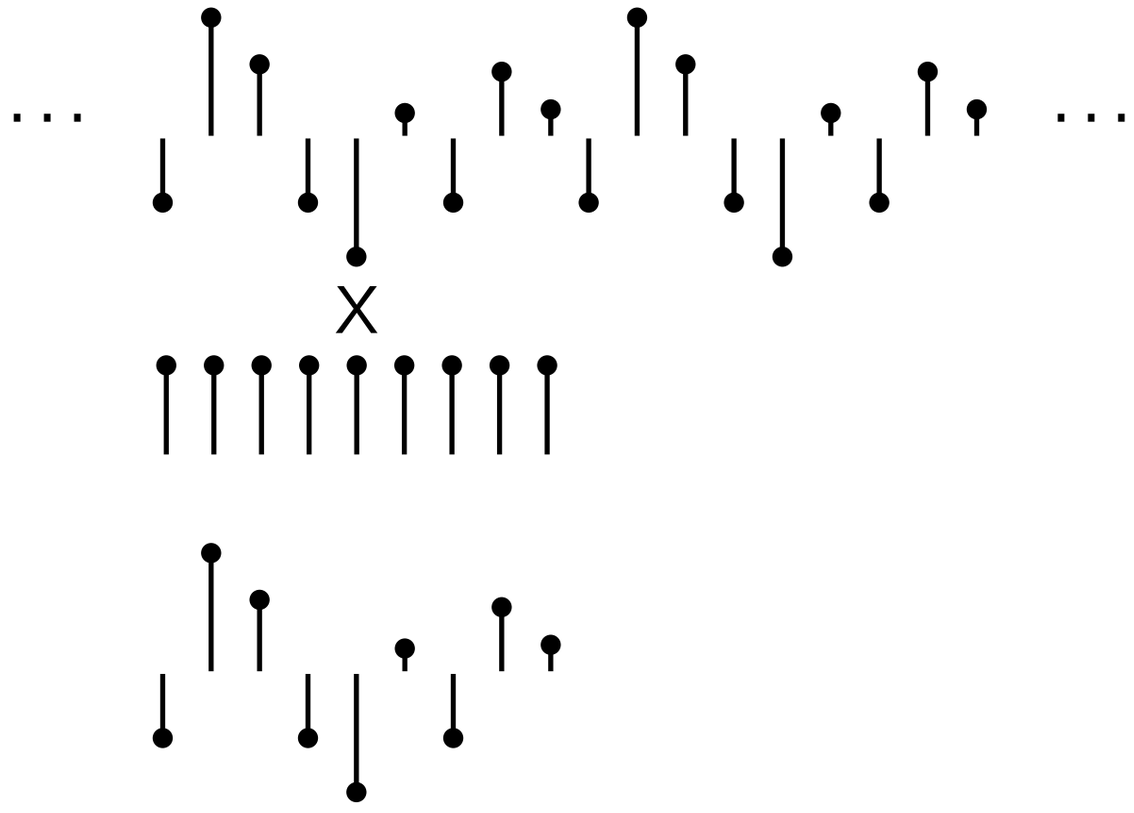
- just using N point FFT performs “Circular Convolution”
  - which is not linear convolution
  - causes the tail end of the convolution to “wrap around” to the beginning
    - FFT assumes the function is periodic (we did not talk about this)
- be aware of circularity when filtering your signal with the FFT
  - zero-padding can solve this for you!
  - zero-pad both signals to a length that will contain the entire convolution, N+M-1
  - for streaming, you must use overlap-and-add!
    - http://en.wikipedia.org/wiki/Overlap–add_method

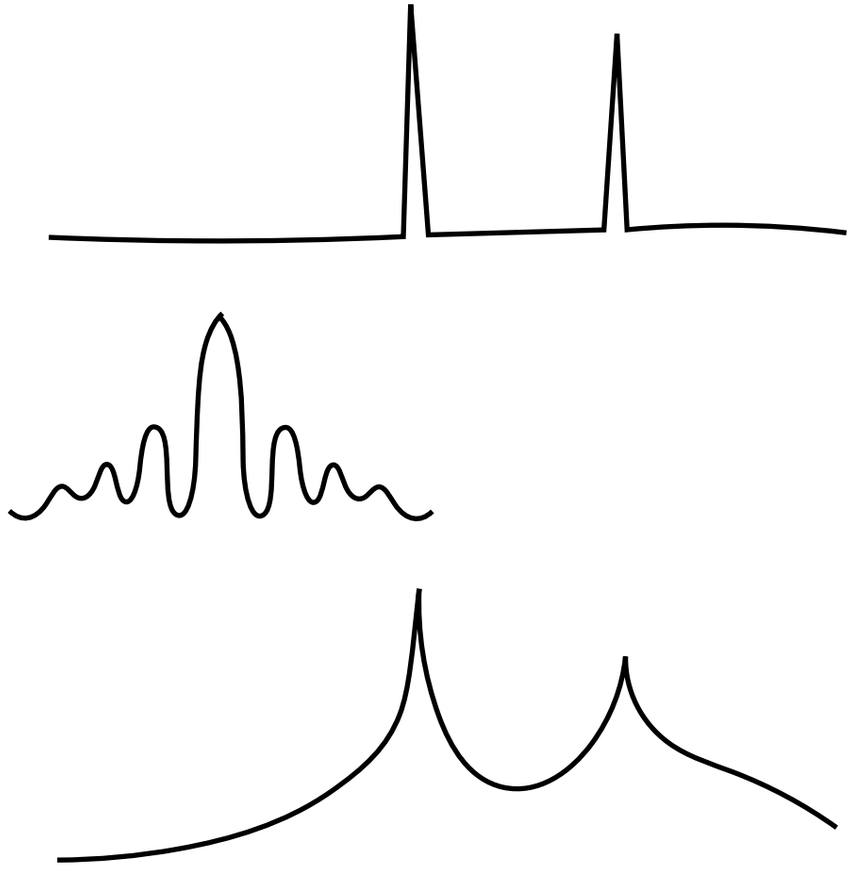
overlap and add



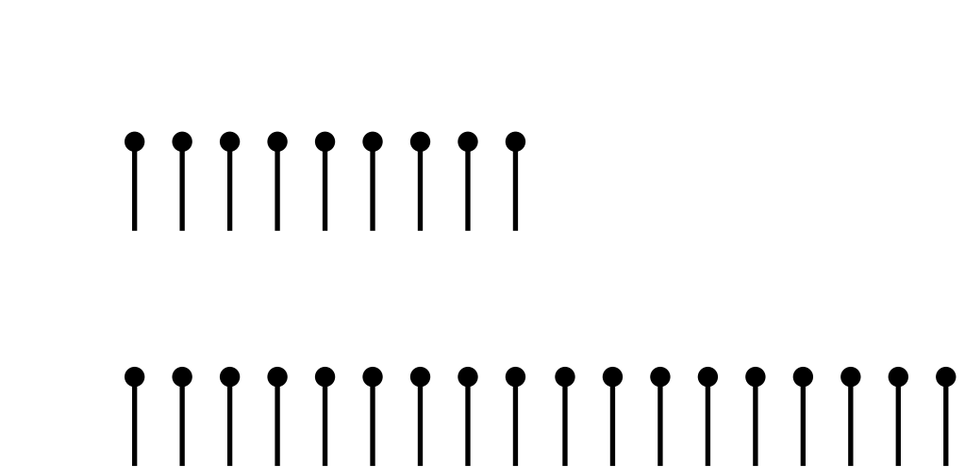
windowing: spectral BW widening

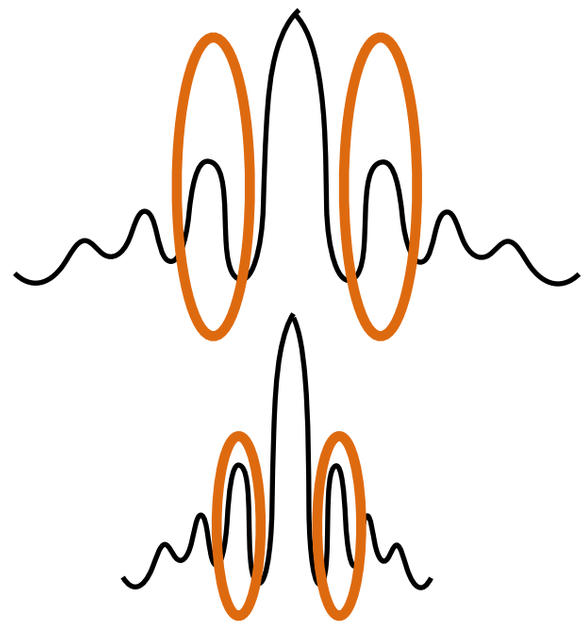
- multiplication in time is convolution in frequency
- a window is something we multiply in time with our signal
- windowing is unavoidable
  - why? we cannot take an infinite FFT…

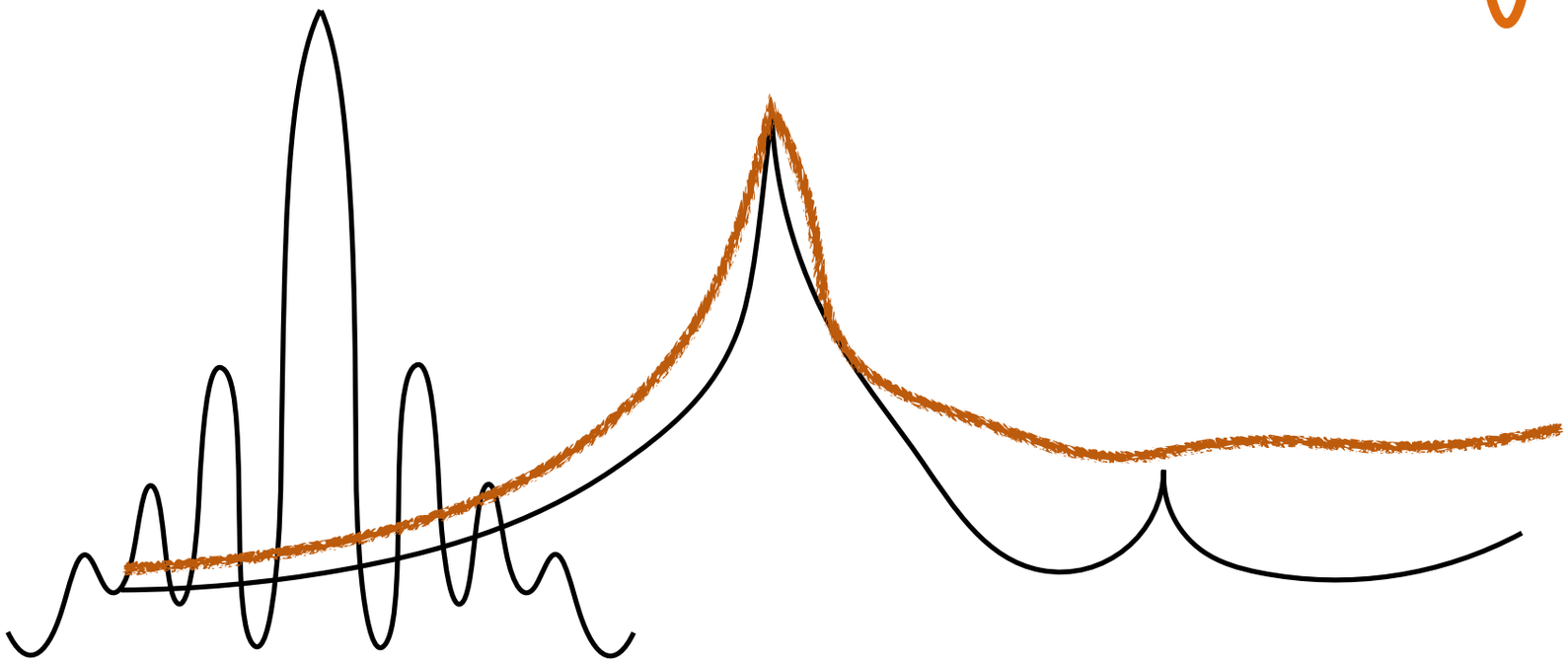




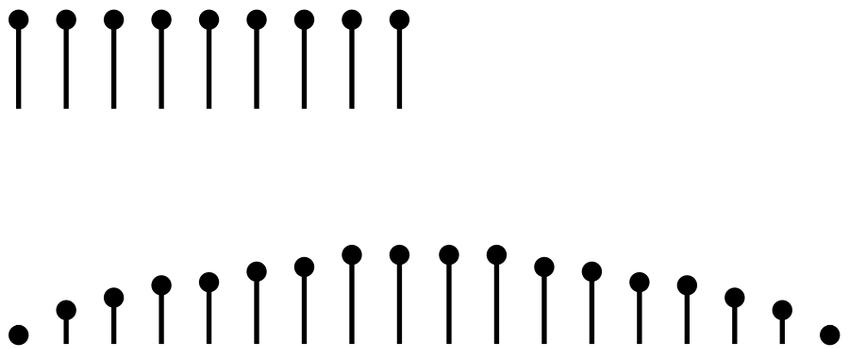
windowing: spectral leakage

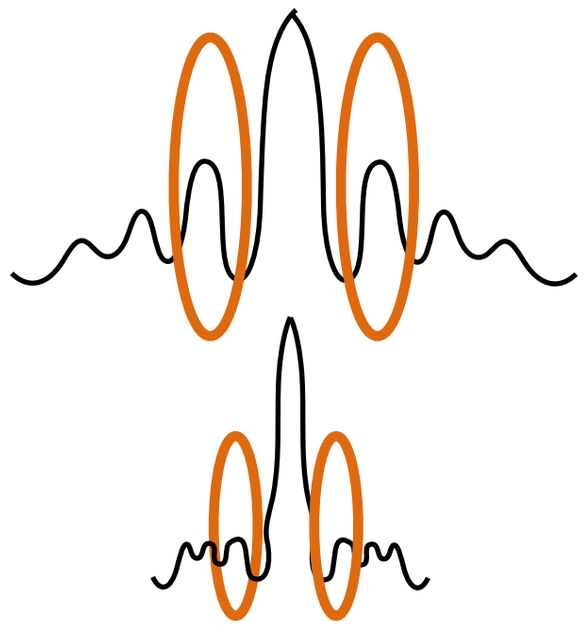


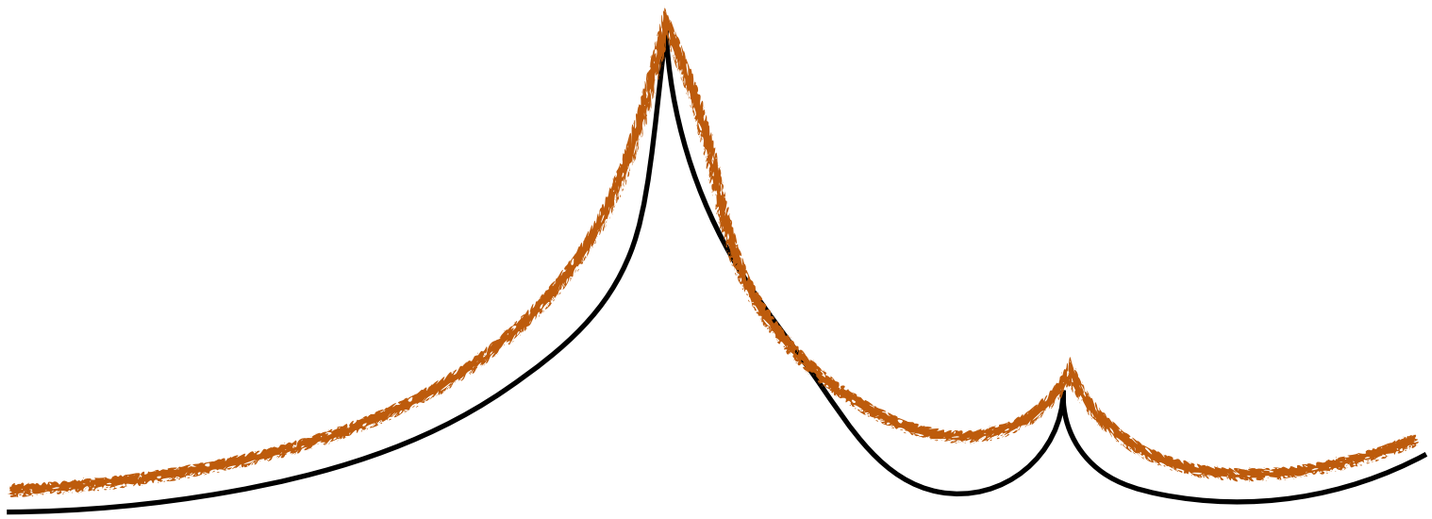




windowing: spectral leakage







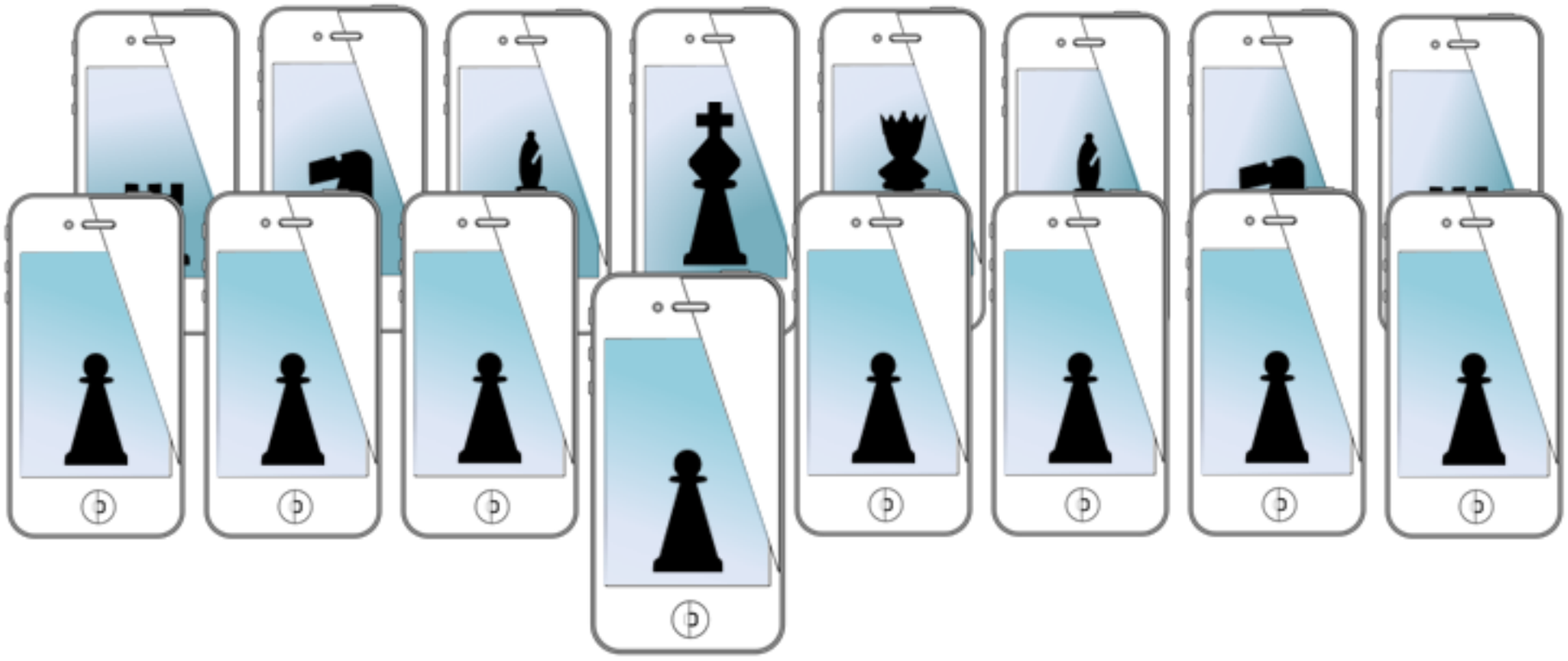
which window to use?

- depends
  - narrowest main lobe: rect
  - good tradeoff: hamming (or Von Hann)
  - optimal tradeoff for a given bandwidth:
    - discrete prolate spheroidal sequence (dpss, Slepian taper)

FFT review

- sampling rate
  - dictates the time between each sample, (1 / sampling rate)
  - max frequency we can measure is half of sampling rate
- resolution in frequency
  - tradeoff between length of FFT and sampling rate
    - each frequency “bin” is an index in the FFT array
      - each bin represents (Fs / N) Hz
      - what does that mean for 12 Hz accuracy?
- windowing is a result of “convolution” in frequency
  - some windows prevent “leakage” at the cost of frequency resolution

MOBILE SENSING LEARNING



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