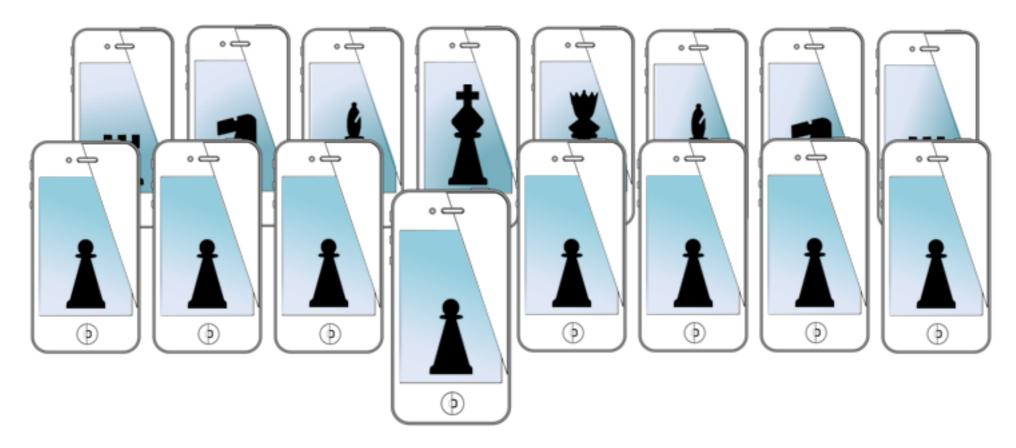
MOBILE SENSING LEARNING



CS5323 & 7323

Mobile Sensing and Learning

core audio

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agenda

- core audio intro
- next lecture: more audio and graphing audio
- next week: second flipped module, the FFT

Core Audio

- many audio packages exist, but we want low level signals
- Audio Sessions (high level, setup audio hardware)
 - access the shared instance (for all applications)
 - access/set category (play, record, both)
 - choose options: like mixing with ambient sources
 - access/set audio route (mostly we will ignore this)
 - set specific hardware within audio route
 - Audio Units (once hardware is ready handle the input/output)
 - set stream format, buffer sizes, sampling rate,
 - initialize memory for audio buffers
 - set callback rendering procedure

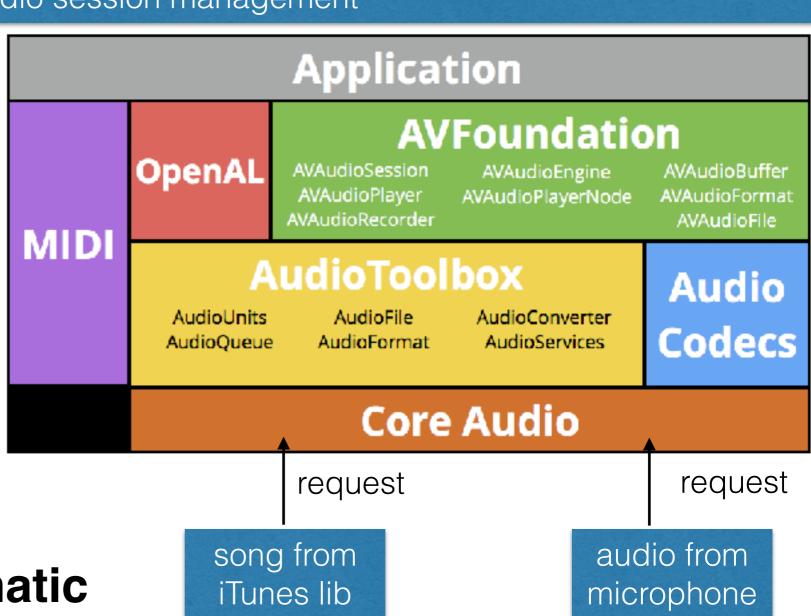
audio sessions



background audio

any request can alter management of other audio requests

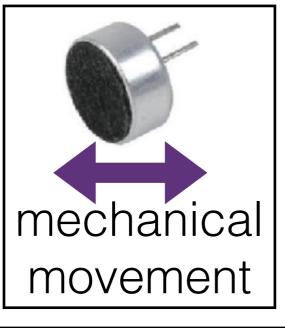
request

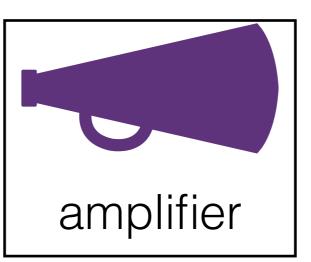


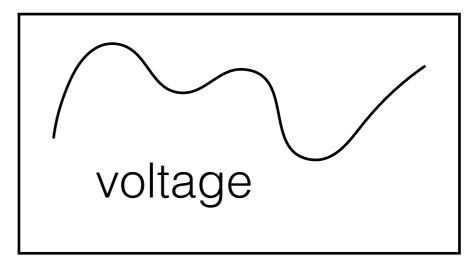
mixing can be automatic — impressive!

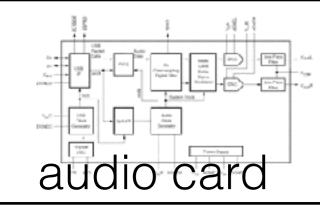
audio hardware



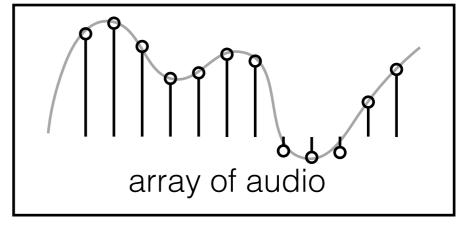




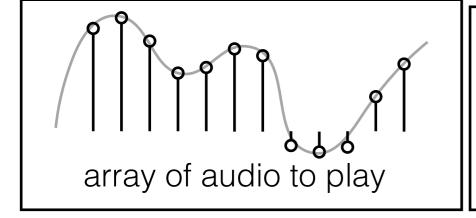


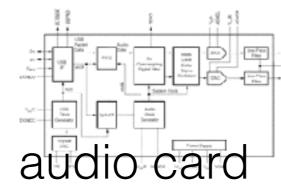


buffer and **ADC**



notify software a buffer is ready





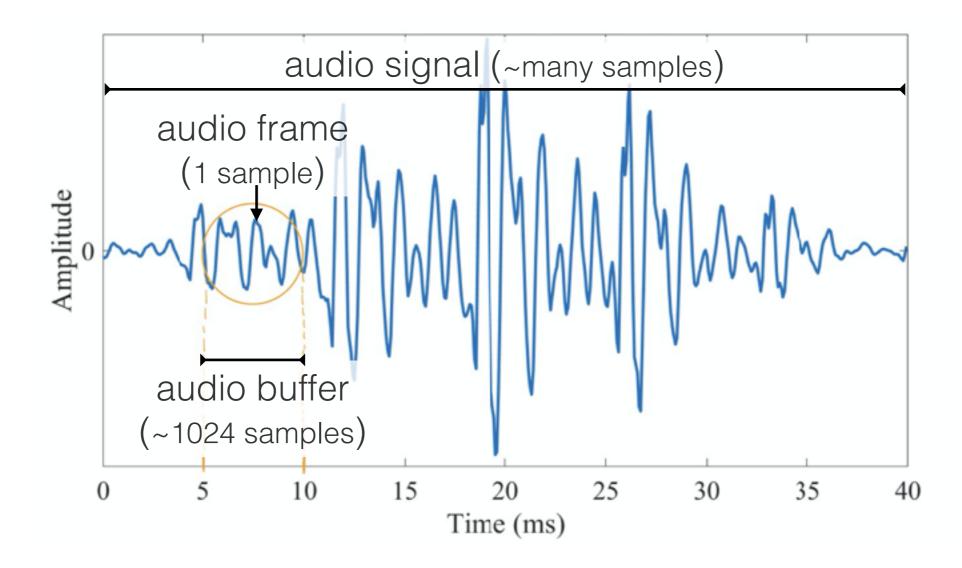
buffer and DAC





audio buffering

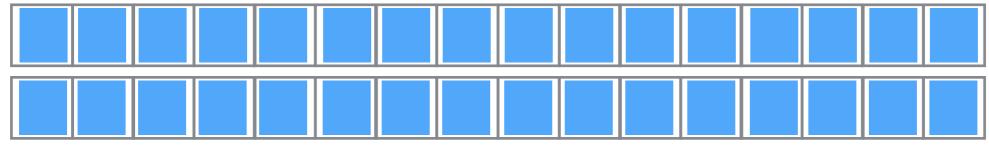
 audio card buffers up audio samples before sending to the main processor



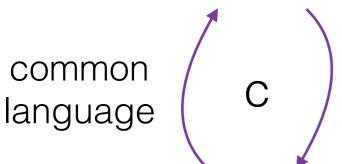
https://medium.com/better-programming/audio-visualization-in-swift-using-metal-accelerate-part-1-390965c095d7

audio units

audio input buffer procedure, double buffer shown



Audio Card (memory allocated on card)



sent to audio session callback

CPU

copy over samples, convert

(memory in RAM) exit from call as soon as possible!

do not allocate memory, take locks, or waste time!!



audio unit formats





right speaker

left speaker



32 bits

callback preallocates buffers developer fills the output buffer OS handles playing the buffer if you don't fill fast enough, audio is choppy

audio nodes solution...

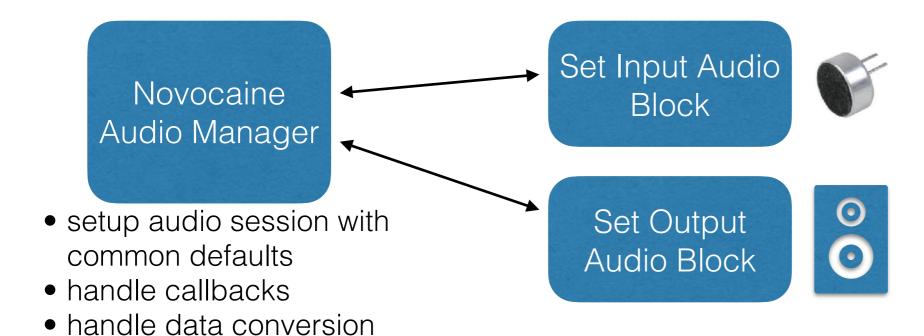
```
// The audio engine used to record input from the microphone.
private let audioEngine = AVAudioEngine()
                                                                               Setup Audio
// setup audio
let audioSession = AVAudioSession.sharedInstance()
do{
    try audioSession.setCategory(AVAudioSession.Category.record)
    try audioSession.setMode(AVAudioSession.Mode.measurement)
    try audioSession.setActive(true, options: .notifyOthersOnDeactivation)
catch { fatalError("Audio engine could not be setup") }
let inputNode = audioEngine.inputNode
let recordingFormat = inputNode.outputFormat(forBus: 0)
inputNode.installTap(onBus: 0, bufferSize: 1024, format: recordingFormat)
```

Get Samples

but, audio unit taps are slower than using core audio...

wouldn't it be **great** if there was a module that **handled** all the specifics of **audio units for us**?

Novocaine: takes the pain out of audio processing Originally developed by **Alex Wiltschko**Heavily manipulated by **eclarson**





- L Twitter
- O Boston, MA
- alex.bw@gmall.com
- U Joined on Dec 4, 2009

novocaine

Novocaine needs callbacks

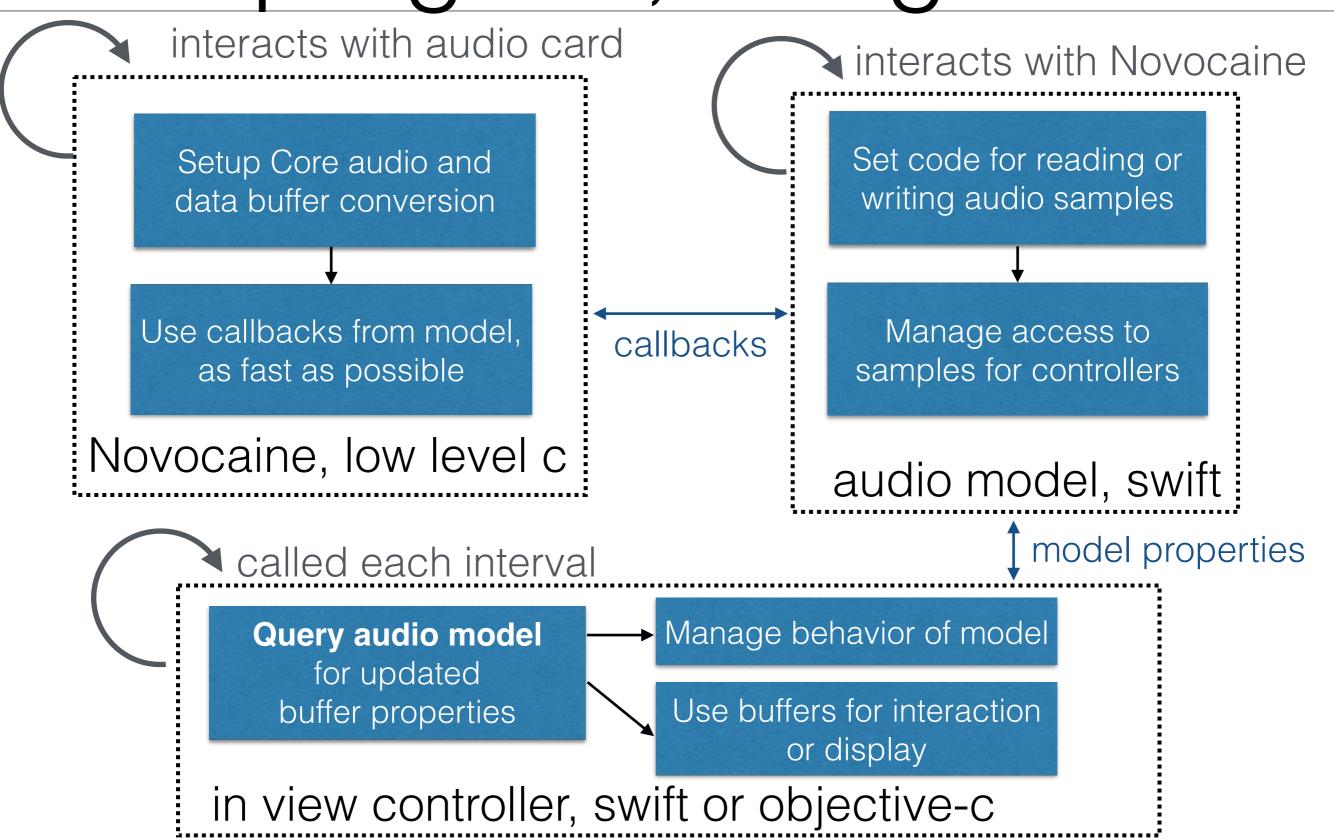
declare properties and setup manager

```
setup function to give
self.audioManager?.inputBlock = self.handleMicrophone
                                                                      novocaine for callback
private func handleMicrophone (data:Optional<UnsafeMutablePointer<Float>>,
                        numFrames: UInt32.
                                                             microphone samples as float array
                       numChannels: UInt32) {
       // copy samples from the microphone into circular buffe
       self.circBuffer?.addNewFloatData(data, withNumSamples: Int64(numFrames))
                                                                        data to write to speakers
self.audioManager?.outputBlock = self.handleSpeakerQueryWithAudioFile
private func handleSpeakerQueryWithAudioFile(data:Optional<UnsafeMutablePointer<Float>>,
                                     numFrames: UInt32,
                                   numChannels: UInt32){
       self.outputBuffer?.fetchInterleavedData(data, withNumSamples:Int64(numFrames))
                                                 microphone samples as float array
   [self.audioManager setInputBlock:^(float *data, UInt32 numFrames, UInt32 numChannels){
       [weakSelf.buffer addNewFloatData:data withNumSamples:numFrames];
   }];
                                                           data to write to speakers
    [self.audioManager setOutputBlock:^(float *data, UInt32 numFrames, UInt32 numChannels)
```

[weakSelf.buffer fetchInterleavedData:data withNumSamples:numFrames];

}];

The program, using MVC



novocaine setup demo

source code on GitHub

rolling stones, if time



Declare in info.plist

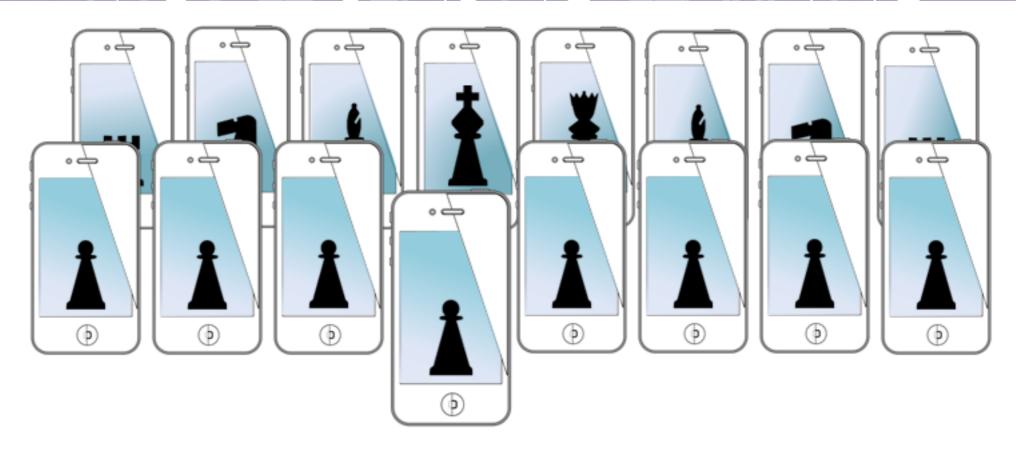
Southern Methodist University

Approacion regards it note environment	•	120
Privacy - Microphone Usage Description	String	This App requires microphone access.
Application Scano Manifest	▲ Distinger	10 itamal

for next time...

- more core audio
 - playing songs (if not covered today)
 - getting samples from microphone
 - showing samples with Metal
 - working with sampled data
 - the accelerate framework

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audio session

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