Signal Processing and Communications Hands-On Using scikit-dsp-comm Part 0

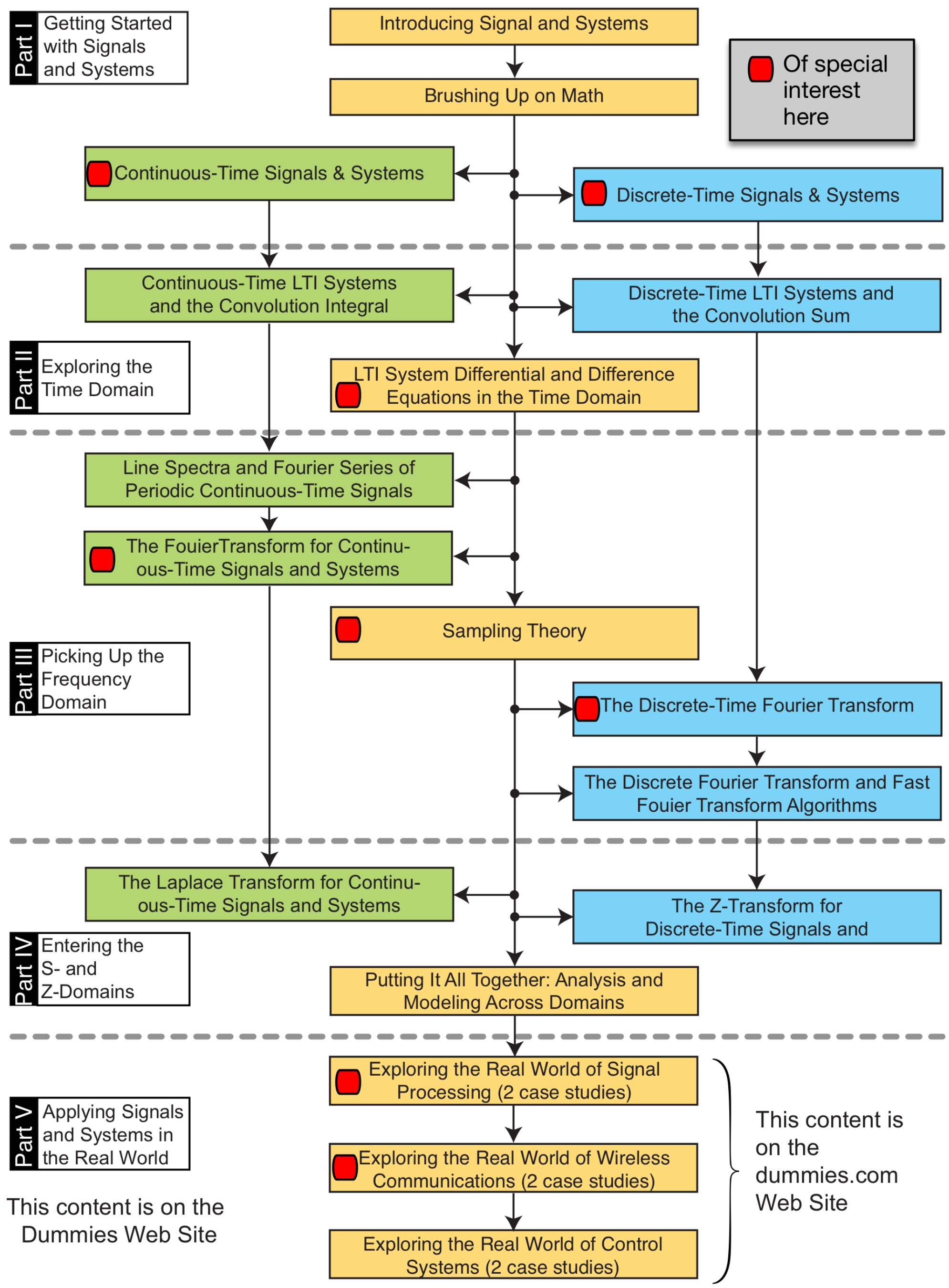
> Reading the PDF version of this `markdown` document is best, unless you have say [Typora](#) installed. The equations will be properly rendered.

Introduction and Session Overview

- Who am I?
- What is `scikit-dsp-comm` and what is its history
- Key signal processing and communications topics I will cover
- Working the first *Lab* involving speech capture using PyAudio and then simple speech processing

What is `scikit-dsp-comm`

- The package started while I was writing the book *Signals and Systems for Dummies*, Wiley 2013
- I needed to create a lot of technical charts and diagrams for the book and started use `matplotlib`, but then needed a code base, and then well, others like my students, might want to use it too
- The module `sigsys.py` was developed to support the topics of the book as shown in the flowchart below:



- I started using Python and the SciPy-Stack in my teaching Fall 2014, after returning from a sabbatical, I started adding more capability and see more to go
  - Python is now the mainstay modeling and simulation tool in all of my classes and my consulting work
  - C and C++ are also import in my work; especially for embedded systems and calculation intensive simulations

Modules

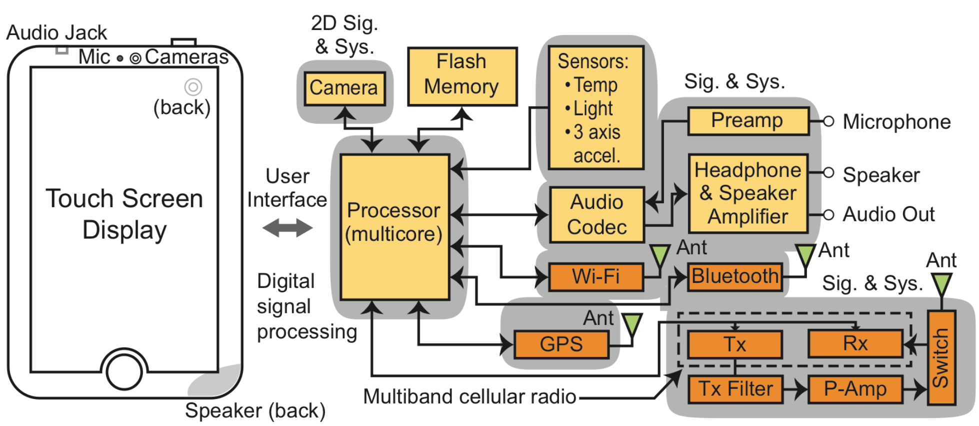
- The current modules:

coeff2header.py
digitalcom.py
fec_conv.py
fir_design_helper.py
iir_design_helper.py
multirate_helper.py
optfir.py (portions of a GNU Radio module)
pyaudio_helper.py
rtlsdr_helper.py
sigsys.py
synchronization.py

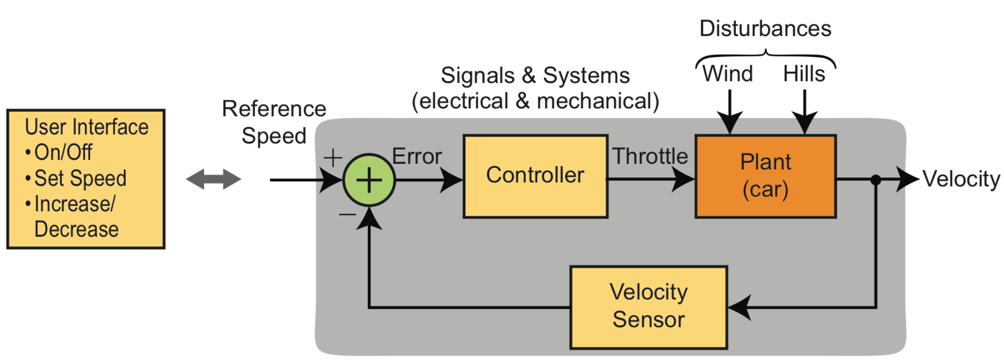
- There are more functions and classes developed, just need to find the time (and the students) to move them into the code base
- The newest module is `pyaudio_helper`; something I have wanted to start on for the past few years, but was waiting for the right time

What is Signal Processing and Where is it Used?

Cellphone Block Diagram - A Signal Processing/Communications Slant



Automobile Cruise Control - A Signal Processing/Control Systems Slant



DSP-Comm Topics Covered

- Introduction/review of continuous and discrete signals and systems modeling

    - How `scikit-dsp-comm` plays a part
    - An emphasis here is on filter design from amplitude response requirements; both finite impulse response (FIR) and infinite impulse response (IIR)

- Audio processing applications
    - Post processing and playback using the `Audio()` widget in Jupyter notebook
    - Real-time processing and playback using `Pyaudio` via `pyaudio_helper`

- Communications theory and signal processing implementation
    - Analog modulation theory and simulation
    - Digital modulation theory and simulation
    - Tools for modeling and simulation of digital comm systems
    - Receiver signal processing using `pyrtlsdr` to capture radio signals from a repurposed digital TV tuner

Highlights of *Hands-on* Experiences

- Audio recording playback slow down/speed up without changing the pitch
- Waveform/signal synthesis, e.g., continous and discrete-time respectively, $x(t)$ and $x[n]$, from primitives
    - The frequency spectrum of a signal, e.g., continous and discrete-time respectively, $X(f) = \mathcal{F}\{x(t)\}$ and $X(e^{j2\pi f/f_s}) = \mathcal{F}\{x[n]\}$

- Systems (linear in most cases) that operate on signals, e.g., the *system function* representation continous and discrete-time respectively, $H(s)$ and $H(z)$, as ratios of polynomials in complex variables s and z
    - Linear constant coefficient difference equations (LCCDE), e.g., for *causal* systems

$$y[n] = -\sum_{k=1}^{N} a_k y[n-k] + \sum_{k=0}^{M} b_k x[n-k] \tag{1}$$

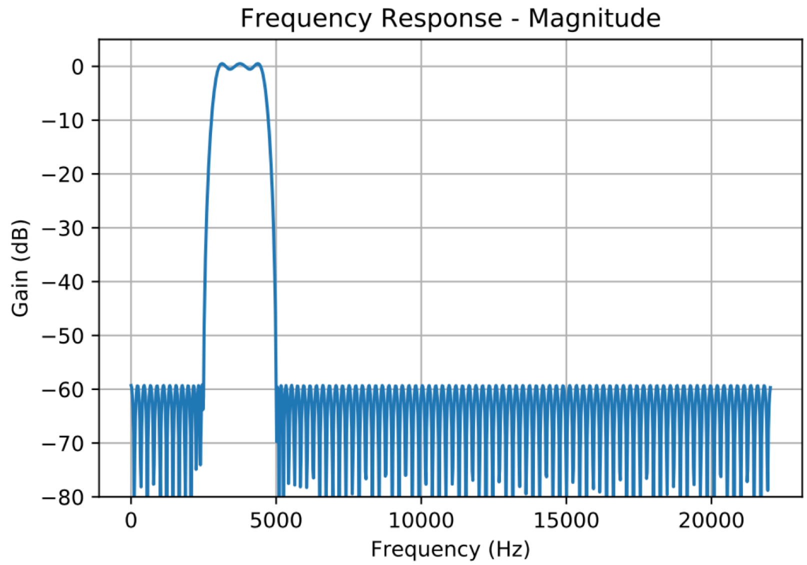
    - Frequency response, e.g., evaluate the system function, continuous and discrete respectively, be setting $s = j2\pi f$ and $z = e^{j2\pi f/f_s}$, where $j = \sqrt{-1}$ and $f_s = 1/T$ is the sampling frequency and period respectively
    - Simulation, e.g., turn the crank on the LCCDE using `y=signal.lfilter(b,a,x)`, where `b` and `a` are coefficient arrays corresponding to a_k and b_k in (1)

- Digital filter design, characterization, and test
    - Simple systems, e.g., M-tap moving average, first-order IIR, IIR notch
    - Design from amplitude response requirements, e.g., `fir_design_helper.py` and `iir_design_helper.py`

```
import sk_dsp_comm.fir_design_helper as fir_d
```

```
b = fir_d.fir_remez_bpf(2500,3000,4500,5000,.5,60,44100,18)
fir_d.freqz_resp_list([b],[1],'dB',44100)
ylim([-80,5])
grid();
```

```
Remez filter taps = 193.
```

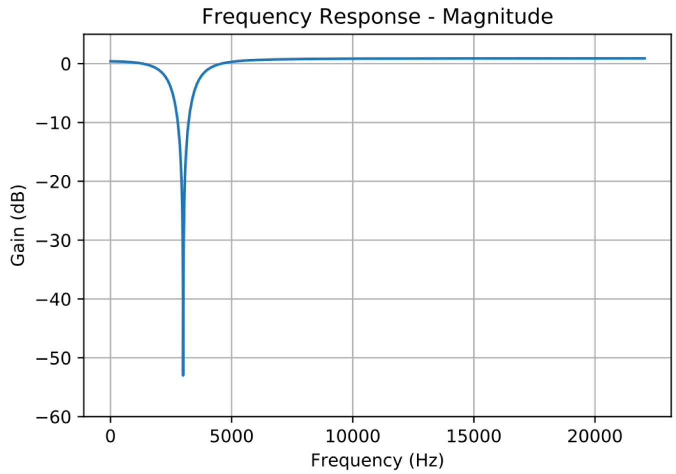


- Simulation
  - Real-time implementation using Pyaudio via `pyaudio_helper`

```
# define callback (#2)
def callback2(in_data, frame_count, time_info, status):
    global b, a, zi
    DSP_IO.DSP_callback_tic()
    # convert byte data to ndarray
    in_data_nda = np.fromstring(in_data, dtype=np.int16)
    #***********************************************
    # DSP operations here
    # Here we apply a linear filter to the input
    x = in_data_nda.astype(float32)
    #y = x
    # The filter state/(memory), zi, must be maintained from frame-to-frame
    y, zi = signal.lfilter(b,a,x,zi=zi) # for FIR or simple IIR
    #y, zi = signal.sosfilt(sos,x,zi=zi) # for IIR use second-order sections
    #***********************************************
    # Save data for later analysis
    # accumulate a new frame of samples
    DSP_IO.DSP_capture_add_samples(y)
    #***********************************************
    # Convert from float back to int16
    y = y.astype(int16)
    DSP_IO.DSP_callback_toc()
    return y.tobytes(), pah.pyaudio.paContinue
```

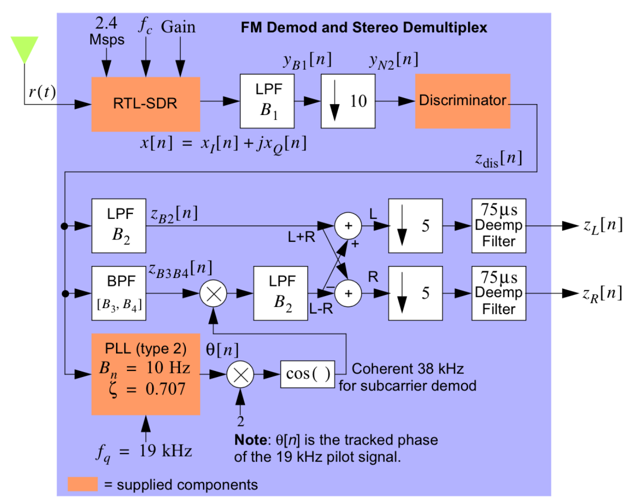
- Overview of real-time implementation in C using an embedded system; mock design for ARM Cortex-M4 processor with measured data provided in a file
- Removing narrowband interference from audio signals
  - Cascade of notch filters, e.g., excise signals not of interest and leave the rest of the signal intact

```python
# Design an IIR Notch
b, a = ss.fir_iir_notch(3000,44100,r= 0.9)
fir_d.freqz_resp_list([b],[a],'dB',44100,4096)
ylim([-60,5])
grid();
```

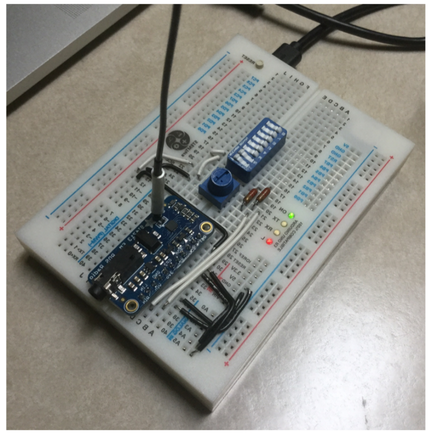


- Adaptive FIR filter, e.g., adapt to statiscal characteristics of signals present removing say only narrowband interferers

* Audio special effects, e.g., echo, reverb, and *flanging*
  - Flanging in particular the function `digitalcom.time_delay`, which implements a time varying time delay (hence can introduce Dopper)
* Modulation and complex frequency translation to move the center frequency of a software defined radio (SDR) signal to a new center center frequency, e.g., $x[n]e^{j2\pi n f_0/f_s} \Leftrightarrow X(e^{j2\pi(f-f_0)/f_s})$
* Amplitude and frequency modulation
  - Transmitter
  - Receiver
* Processing software defined radio (SDR) signals using multi-rate DSP



* Receiving frequency-Shift Key (FSK) digital modulation from Arduino-based FM radio transmitter

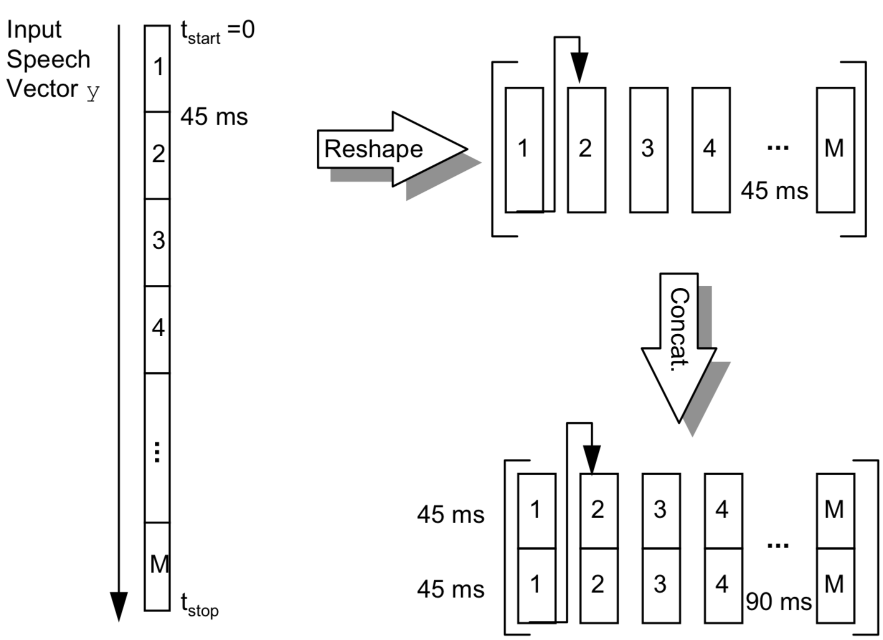


Lab 1: Use PyAudio to Record a Short Sentence (~5-10s) and then Process the Speech

- The use PyAudio to record your speech is optional, as test vectors are provided
- The primary objective is to use *butt splicing* via the `numpy` function `reshape()` to slow down and speed up the playback of a speech array without shifting the pitch

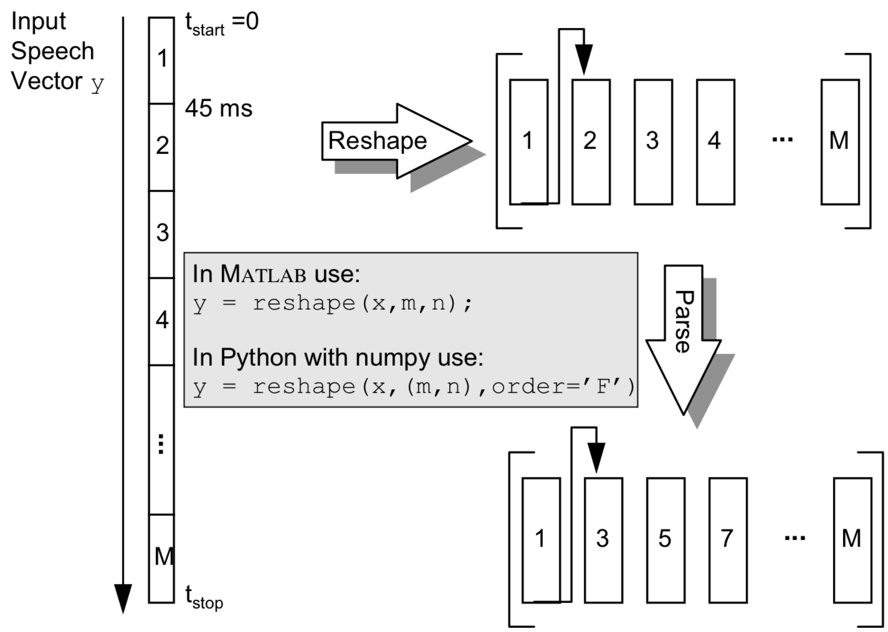
Slow Down Playback Approach

Repeat small slices of the speech ndarray as shown below:



Speedup Playback Approach

Remove small slices of the speech ndarray as shown below:



Move to the Jupyter Notebook

- Open the Jupyter notebook file `Speech Processing.ipynb`

wave wrapper functions in sigsys	
`to_wav( filename,rate,x)`	Write a wave file. A wrapper function for `scipy.io.wavfile.write` that also includes `int16` scaling and conversion. Assume input `x` is [-1,1] values.
`from_wav( filename)`	Read a wave file. A wrapper function for `scipy.io.wavfile.read` that also includes `int16` to float [-1,1] scaling.

Onward to Part 1

Following the *Speech Processing* lab we will move into an overview of signal and system modeling.