Final Lab
May 2, 2017

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INSTRUCTIONS:
All lab submissions include a written report and source code in the form of an m-file. The report contains all plots, images, and figures specified within the lab. All figures should be labeled appropriately. Answers to questions given in the lab document should be answered in the written report. ***The written report must be in PDF format***. Submissions are done electronically through **my.ECE**. NO LATE SUBMISSIONS ACCEPTED.

1 Sakanaya

(25 pts) Load *signal.mat*. This signal is sampled at $f_s = 100$ hz. Plot the magnitude and phase response of this signal. How many tones are there? What is the frequency of each tone (in Hz)?

2 Chipotle

Time domain must be in seconds. Frequency domain must be in hertz.

(5 pts) Load *samplerate.mat*. Assume that the analog sampling frequency f_s is 40 hz. Plot the magnitude spectrum and time domain plots of the original signal x using *plot* and *stem*, respectively.

(5 pts) Upsample the original signal by 3 by inserting zeros between samples. Plot the resulting magnitude spectrum and time domain plots for this new signal using *plot* and *stem*, respectively. What do you notice about the magnitude response compared with the magnitude response of the original signal? Why does this effect occur (*HINT:* think about your fourier transform properties)? If you cannot think of the fourier transform property, then you may present a math proof.

(10 pts) In the frequency domain, apply an ideal LPF to get rid of the extra spectral content introduced by zero insertion. Using *subplot*, plot both the magnitude spectrum after applying the LPF and the corresponding time-domain waveform.

(5 pts) Downsample the signal from section 2.3 by 2. Using *subplot*, plot both the time domain waveform and its associated magnitude response. What is the final sample rate of the waveform? What is the maximum value of D you can choose without aliasing?

3 Legends

(5 pts) Load *q1_signal.mat*. Plot the magnitude and phase spectrum of 'x'.

(10 pts) The sampling rate for 'x' and 'sig' are the same. Plot the magnitude and phase spectrum of 'sig' such that we can make a meaningful comparison with the spectrum of 'x'.

(10 pts) The several copies of the signal 'sig' is present in 'x'. Count the number of times 'sig' occured in 'x'. (Hint: you will have to implement some form of filtering. Your code does not have to generate the count, you can generate a plot that allows you to count easily).

4 Blackdog

Load *q2_signal.mat*. It should contain 2 variables, a signal 'x' and sampling rate 'fs'.

(5 pts) Listen to 'x' using soundsc. Describe what you hear, plot the spectrogram of signal 'x', using a hamming window, window size of 256, step size of 128. Make sure to label your axes correctly. You may use the information derived from the spectrogram to help you describe signal 'x'.

(10 pts) Your friend doesn't understand DSP and aliasing. He downsampled x by removing every even element of 'x'. Why is this wrong? What do you expect to hear when you listen to the his downsampled version of 'x'? Verify what you described by downsampling in the 'wrong' way and plotting the spectrogram of this new signal (Use the same window size and step size as before).

(10 pts) Implement the right way of downsampling 'x'. What do you expect to hear now? Verify this by plotting the spectrogram of the correctly downsampled 'x' (Use the same window size and step size as before).