EE 341
Lab 3: The FFT and Digital Sound Transformations

Note: *Each group provides only one report for all group members. All MATLAB files (.m files) must be uploaded.*

When using a digital computer, frequency analysis means using a Fast Fourier Transform (FFT). This necessitates we spend some time becoming familiar with using the FFT to study the frequency content of a discrete-time signal. Most parts of this lab are given by Mari Ostendorf class website for EE 341, Spring 2014.

1. **MATLAB function FFT**
In this problem you will learn how to use the MATLAB command fft. First, use the help feature in MATLAB to learn the syntax of the *fft* function. The FFT function computes the Discrete Fourier Transform (DFT) of a sequence. In general the FFT of a sequence will be a complex function so you will need to look at the magnitude and phase separately. The MATLAB commands abs and angle are useful for obtaining the magnitude and phase of a complex valued sequence. Also, since the FFT only has values at discrete frequencies, it may be useful to do the plots with stem to reinforce that idea, but continuous frequency plots (i.e. using plot) are often used since they are closer to the DTFT that you are ultimately interested in.

The FFT outputs a sequence that corresponds to the range $0 \leq \omega \leq 2\pi$. You are probably more familiar with seeing the spectrum plotted over the range $-\pi \leq \omega < \pi$ (or $-0.5 \leq f \leq 0.5$). The *fftshift* function can be used for this purpose.
Plot the magnitude of the FFT of the following signal before and after the *fftshift*

$$x[n] = 1 + \cos(2\pi fn); 0 \leq n \leq 127$$

for the cases where $f = 0.25$ and $f = 0.5$.

WRITTEN REPORT: Generate a 3-part plot for each signal: unshifted DFT, shifted DFT, and the shifted DFT with a Hz frequency scale assuming a sampling rate of 10kHz. Discuss why the frequency peak locations make sense.

2. **Frequency Shifting**
Recall that multiplying by a complex exponential in time (or a cosine, which is comprised of complex exponentials) results in a frequency shift. For each of the following sequences, let $f_1 = 0.15$ and $0 \leq n \leq 255$. Use the built-in *sinc* function in MATLAB. Plot the magnitude and phase plots (using plot), where the magnitude and phase plots are over the range $-0.5 \leq f \leq 0.5$ (normalized frequency), i.e. use *fftshift*.

a) $x_1[n] = \text{sinc}(f(n-32))$.
b) $x_2[n] = \text{sinc}(f(n-32))(-1)^n$.
c) $x_3[n] = \text{sinc}(f(n-32))\cos(2\pi f n)$ where $f_2 = 0.2$.
d) $x_4[n] = \text{sinc}(f(n-32))\cos(2\pi f n)$ where $f_3 = 0.4$.

WRITTEN REPORT: Display the plots for (a)-(c). State what type of signals each corresponds to (low pass, high pass, etc.). For (d), explain why (d) does not have a flat frequency response in the passband.

3. Starting from Continuous Time Signals

Get two sounds from the class web site, picking one that you think will have more high frequency content and one that will have more low frequency content. Load each sound (using *wavread*) and play it out (using *soundsc*). Record the sampling rate, length of the data (sec), samples.

Plot the time signal and FFT of each sound. For the frequency plot, use your understanding of the relation between discrete and continuous time and knowledge of the sample rate to scale the frequency axis to match the continuous time range in Hertz. Comment on whether the frequency content matches your expectation.

a) Modify your signals by multiplying the time signal by either $(-1)^n$, as in 2(b), or a cosine as in 2(c). Play the sounds and plot the frequency content of the new sounds. What is the effect of the frequency shift on how this sounds relative to the original?

b) Modify your signals by time scaling ($y[n]=x[2n]$), which results in frequency scaling. Play the sounds and plot the frequency content. How does frequency scaling compare to frequency shifting?

c) One way to implement an ideal filter is to zero out frequency terms. Implement a high-pass filter with a cut-off of 0.25 by zeroing out the low frequency terms in the FFT of the sounds you chose. Then take the inverse FFT of the result to get the modified signal. Play the sounds and plot the frequency content of the new sounds. (You may need to scale the filtered sound if it is hard to hear, since you've eliminated a lot of the energy in it.) Discuss the impact of this operation on the sounds.

WRITTEN REPORT: Turn in the frequency plots for the original and modified versions of each of your sounds. Provide details of the modifications that you did in each part and explain the impact of these operations on what you hear.