Homework #8: Chapters 9 and 10 (due Fri. Nov. 30, 2012)

Preliminary

- All problems are worth +10 points unless otherwise noted.
- Download the CompanionFiles8.zip file

  `http://www.ece.nmsu.edu/~pdeleon/Teaching/EE395/Homework/CompanionFiles8.zip`
- Code the following tool from *DSP Software Toolkit*: Chapter 5 **freqshft.m** and **modulate2.m** and Chapter 7 **fir_wind.m** (option 0 only),
- Please attach at the *end* of your assignment, printouts of **main1.m** (code to solve software problem 1), **main2.m** (code to solve software problem 2), etc... as well as any new tools developed in this assignment.

Software Problems

Use your software tools to solve the following problems.

1. (+40 points) Dual-tone multi-frequency signaling (DTMF) is used for telecommunication signaling over analog telephone lines in the voice-frequency band between telephone handsets and other communications devices and the switching center. For more information see

   `http://en.wikipedia.org/wiki/Dual-tone_multi-frequency_signaling`

   In DTMF signaling, a combination of a high-frequency tone and a low-frequency tone represent a specific digit or the characters * and # (there are also A, B, C, and D but these digits are typically left off the keypad). The eight frequencies are arranged in Table 1. Determine the phone number being dialed from spectral analysis of the DTMF signal in phone_n.wav (in CompanionFiles10.zip) where n is the last digit of your student number.

   Table 1: DTMF frequency definitions

	Col 1 (1209 Hz)	Col 2 (1336 Hz)	Col 3 (1477 Hz)	Col 4 (1633 Hz)
Row 1 (697 Hz)	1	2	3	A
Row 2 (770 Hz)	4	5	6	B
Row 3 (852 Hz)	7	8	9	C
Row 4 (941 Hz)	*	0	#	D

   Hints/warnings to get started:

   - Plot (but don't print) the signal to determine the digit boundaries. Parse the signal according to the digit boundaries and within each digit, extract a segment for spectral analysis. Analyze each segment separately with the DFT. You may apply either a rectangular or Hamming window to your signal segments prior to computing the DFT. Your segments should be long enough for desired physical resolution (see Probs. 9.9 and 9.10 below).
   - Estimate the frequencies in the segment with the usual conversion

   $$\hat{f} = k f_s / N \tag{1}$$

where k is the DFT index number, N is the length of the segment (or length of DFT), and f_s is the sampling frequency. You are not to playback the WAV file through the telephone handset (dialing) and ask the answering party what phone number you dialed.

- **(a)** Plot the magnitude spectra of the first 2 segments (Digit #1 and Digit #2). Your plots should be of $|X(k)|$ vs. k where $0 \leq k \leq N/2-1$. Be sure and title your spectrum with the segment number and indicate the value of N.
- **(b)** What are the two DFT indices corresponding to the two spectral peaks (be sure to indicate the value of N)? Which two DTMF tones do these peaks correspond with?
- **(c)** What is the phone number being dialed?

2. Window method for FIR filter design.
   - **(a)** Use the fir_wind.m tool to reproduce Figs. 10.1.6(a) and 10.1.7(a) for the filter which uses the rectangular window. Also plot the magnitude response (in dB) and phase response for the filter.
   - **(b)** Use the fir_wind.m tool to reproduce Figs. 10.1.6(b) and 10.1.7(b) for the filter which uses the Hamming window. Also plot the magnitude response (in dB) and phase response for the filter.
   - **(c)** Comment on the relations between the window's mainlobe bandwidth/peak sidelobe amplitude and the filter's transition bandwidth/stopband attenuation using the development in Sections 10.1.2 and 10.1.3 in your text.
3. Frequency-shift a Hamming-windowed low-pass filter (LPF) to form a high-pass filter (HPF) with half-amplitude frequency, $\omega_c = 0.3\pi$ and plot the filter coefficients and magnitude response in dB.
4. Modulate a Hamming-windowed LPF to form a band-pass filter with half-amplitude frequencies, $\omega_a = 0.4\pi$ and $\omega_b = 0.6\pi$ (see Figure 10.1.1 for edge definitions) and plot the filter coefficients and magnitude response in dB. Be sure to scale your coefficients by $2\times$ so the passband gain is 0 dB.

Textbook Problems

9.2, 9.5, 9.9, 9.10, 9.23, 9.32