Name:__________________

Section:_____2111_____

EE432 Fall 2011 Exam 3

Problem	Possible Points	Score
1	45	
2	55	
Total	100	

- Turn this in Fri, 12/09/2011 at the beginning of class.
- This is an open book, open notes and open computer exam. USE ONLY YOUR COURSE TEXTBOOK ("Fundamentals of Digital Signal Processing"), your notes from this course, and MATLAB HELP.
- You ***must show your work*** to get full credit for problems. Use additional sheets as necessary.
- Label your plots carefully, and turn in all of your code when using MATLAB. Put all the code into one document to turn it in and save paper.
- Organize and clearly label the additional materials you turn in, so it is obvious to me what problem or part of a problem the material refers to.

- If you are stuck on a problem, you may ask for guidance…but it might cost you in points. You ask your question, and I will let you know how much it will cost. Then you can agree to obtain the guidance for the specified number of points off of your final score, if you wish.

1. (45 pts) DFT/FFT.

    a. Write a MATLAB function called *plotfftmag* that will take in a signal and its sample frequency and produce a plot of the magnitude of the FFT of that input signal. The x-axis should be labeled : "Frequency (Hz)", the y-axis is magnitude of the FFT and should be labeled : "|X[k]|" The plot's title should be : "FFT Magnitude-xxx", where xxx is replaced by your last name. The plot must display the correct number of samples of the FFT from 0 Hz to $\leq$ the Nyquist frequency. The function does NOT return any values, but if the input is all zeros, or the sample frequency is a negative value, the function will output a warning message on the command line stating that either the input is all zeros or that the input sample frequency is negative.

    Usage : >> plotfftmag(x, fs)

    Turn in your well-commented code for this function.

    b. Generate 128 samples of a 737 Hz cosine with amplitude 1.0 added to a 1205 Hz cosine with amplitude 0.3, using a sample frequency of 8000 Hz. If for some reason you cannot figure out how to do this, download the file called "exam03-problem1b.wav" and use that instead for partial credit. Note that this file is DIFFERENT than the one I asked for. Indicate if you used this signal below. If *x* is the name of your signal, record the 89^{th} value; i.e, what value appears when you type "x(89)" on the MATLAB command line.

    x(89) = ___________   Used "exam03-problem1b.wav"? Yes / No

    Turn in your code you used to create this signal.

    c. Use your *plotfftmag* function to generate a plot of the magnitude of the signal's FFT. Print out and turn in your plot.

    Turn in a hardcopy of this plot.

    d. What is the frequency resolution of the FFT of this signal? How many samples of the sinusoid would need to be collected to have a 0.2 Hz frequency resolution in its FFT?

Continued on the next page.

e. Based on the Fourier transform of a cosine, the frequency content of this signal should be two impulses (i.e., delta functions). Explain why does *plotfftmag* not show two delta functions?

f. Create the same signal as in part b, but with a different sample rate and number of samples: Generate 1000 samples of a 737 Hz cosine with amplitude 1.0 added to a 1205 Hz cosine with amplitude 0.3, using a sample frequency of 2000 Hz. Produce a plot of the magnitude of the FFT, and describe why this plot looks different than the one in part b. If for some reason you cannot create this signal, for partial credit, describe what you'd expect to be the effects on the plot.

Turn in a hardcopy of this plot and annotate the differences between this plot and the plot from part b.

2. (55 pts) Download the file called exam03-problem2.wav from the course website. This is a music clip from James Brown's "Living in America" that has been corrupted by two types of noise. Listen to the clip and you will hear the corruption.

   a. Use your *plotfftmag* function to plot the frequency spectrum of this signal and determine the corrupting noise. Annotate on your plot the corrupting frequency content, e.g. specific frequency or frequency range(s), white or colored noise or tonal, etc.

      Turn in a hardcopy of this plot.

   b. Design two filters to reduce/eliminate this noise while maintaining as much of the quality of the music as possible. One of your filters must be a low pass filter that is a windowed-sinc approximation to an ideal LPF, and you must use the procedure from Tables 9.3 and 9.4 in the textbook to create that filter.

      For the LPF, the design constraints are:
      FIR filter
      No more than 500 terms
      Stop band attenuation: > 50 dB

      Carefully describe the steps you used to create the LPF.

      Also, use *fvtool* to create a plot of the frequency response of this LPF along with the frequency content of the noisy signal in the SAME *fvtool* window. Turn in a hardcopy of this plot.

      The other filter can be created using *fdatool*. For this filter, the design constraints are:
      IIR filter
      Minimum order
      Filter must be stable

      Turn in a hard copy of the *fdatool* window after your filter is designed, to show me your specifications and frequency response. You can Print Screen to copy and paste the *fdatool* window into a word document, and print the word document.

   c. Filter the noisy signal with your filters and write out your result to a file called "EE432Exam03.wav" and email it to me. Be sure to listen to your file before emailing it, and ensure that the corrupting noise is sufficiently reduced.