EE422G Homework #8 (12 points)

Due March 8,2007

1. (4 points) Given the following two transfer functions

   $$H_1(s) = \frac{s^2 + 3s + 4}{s^3 + 6s^2 + 11s + 6} \text{ and } H_2(s) = \frac{s^3 + 2s^2 + 3s + 2}{s^4 - 10s^3 + 35s^2 + 50s + 24}$$

   Use MATLAB

   a. To plot the step response of each system.
   b. To determine what type of stability each system exhibits.
   c. To plot the Bode plot of each system. Does the Bode plot make sense?
   d. To plot the output response of each system for input $x(t) = \ln(2t)\cos(5t)$ between t=0.01s to t=10s. Also show the input signal in the same plot.

2. (4 points) To realize her "American Idol" dream, a friend of yours has recently bought a microphone for her own recording studio. Unfortunately, the microphone is inexpensive and the quality is poor. She has recorded a sample sound file which you can find in `distortedSound.wav` from the homework webpage. Armed with your recent knowledge of MATLAB, you try to help her out to post-process the recording. After some research on the specifications, you find out that the transfer function of the microphone is as follows:

   $$H(s) = 0.04\frac{s^2 + 2\times10^2 s + 1.0001\times10^8}{s^2 + 1.52\times10^3 s + 2.92\times10^6}$$

   a. Could you design a linear system to compensate the distortion caused by the microphone? Please submit the MATLAB code for both the design of your "compensating filter" and the experiments.

   b. Due to your success in part a., your friend wants to further cut the cost by buying an even cheaper microphone. The sample sound file is stored in `distortedSound2.wav` and the transfer function is almost the same as before except for a sign change in the numerator and a small change in gain:

      $$H(s) = 0.03\frac{s^2 - 2\times10^2 s + 1.0001\times10^8}{s^2 + 1.52\times10^3 s + 2.92\times10^6}$$

      Can you use the same approach as in part a) to compensate for the distortion in this case? I highly recommend plotting the output first before attempting to play it with your computer speaker. Also try the compensating filter you obtained from part a). Please comment your results.

   Hint: in MATLAB, you can use the following command to read from the wav file

```
>> x = wavread('distortedSound.wav');
```

and the following command to play the sound, which is sampled at 44.1 kHz.

```
>> sound(x,44100)
```

3. (2 points) We have seen the aliasing effect visually during lecture. In this problem, you are asked to explore the aliasing effect in audio. Download the MATLAB script `aliasing_demo_audible.m` from the homework webpage and run it. You will hear six tones at different frequencies: 500Hz, 2kHz, 3kHz, 4.5kHz, 5.5kHz and 7kHz. All the signals are sampled at 5kHz. Explain what you hear using the Nyquist Theorem.

4. (2 points) The signal

$$x(t) = 4 + 8\cos(8\pi t) + 3\cos(16\pi t)$$

is sampled at a rate of 32 samples per second. Plot the amplitude spectrum of the sampled signal showing the weight and the frequency of each component for $|f| < 80$ Hz. Show how the signal can be reconstructed from the samples.