EE395: Introduction to Digital Signal Processing
Final
December 8, 2008

You are allowed to use a dumb calculator on this test and two 8.5x11" notesheet (both sides may be written on). You are not allowed to use the textbook, homework solutions, or any other references. Your answers must be written in the space provided on the exam sheets, but you may attach additional sheets containing your work if necessary. Do not talk during the test: if you have questions, ask the exam proctor. Show your work (including intermediate steps) unless otherwise notes in the problem. You may use properties but you **must** state which property you are using when you use it.

Name:____

Problem Number	***Max Points***	***Points***
1	20	
2	30	
3	20	
4	20	
5	10	
Total	100	

1. (20 pts) Discrete-time Fourier Transform (DTFT). Clearly state which properties (if any) you have used and show your work.

a) (8 pts) Determine the DTFT of the following sequence: $x[n]=\alpha^{n+1}\mu[-n+1]+\beta^{n-1}\delta[n-2]$.
Define the range of α and β for which this DTFT will exist.

b) (8 pts) Determine the inverse DTFT *h[n]* of $H(e^{j\omega})=\frac{3e^{j4\omega}}{1-0.5e^{j\omega}}$. Use properties.

c) (4 pts) Let the *h[n]* found in part b) correspond to the impulse response of a system with a corresponding transfer function *H(z)*. What is the ROC of *H(z)*? Explain your reasoning briefly.

2. (30 pts) Let the transfer function *H(z)* of a system have the following poles and zeros:

poles: 0.75, 1.1, -0.75

zeros: 1.1j, -1.1j, 0.5

a) (5 pts) Roughly sketch the magnitude of the frequency response. What kind of filter is it (e.g., lowpass, highpass, etc.)? What is the implied region of convergence here?

b) (5 pts) Determine the transfer function *H(z)* in polynomial form (i.e., the most common way we write them). Assume that the gain multiplier is 1.

c) (1 pt) Is *H(z)* minimum phase, maximum phase, mixed phase, or linear phase?

d) (10 pts) Determine the impulse response $h[n]$ of this system assuming now that the system is causal. Is this system stable? Justify your answer.

e) (9 pts) Write down the transfer function for a system $A(z)$ which, when cascaded with $H(z)$, will create a new filter $G(z)$ having the same magnitude response but being minimum phase: i.e, $G(z) = A(z)H(z)$ where $|G(e^{j\omega})| = |H(e^{j\omega})|$. Is $A(z)$ stable and causal? Is the impulse response of $A(z)$ real valued?

3. (20 pts) You have been tasked with replacing an obsolete DSP-based filtering system in your company's widget polishing facility! If you succeed with this one, the company has promised to send you on an all-expense-paid vacation to Cancun for Spring break!!! If not, you will be fired and your job will be given to a UNM grad...

What you know about the existing digital filtering system:

- The A/D and D/A are both ideal with ideal antialiasing and reconstruction filters
- Input frequencies in the range from DC to 55 kHz are preserved by the existing system
- Sampling is performed ***exactly*** at the Nyquist rate
- The discrete-time filter in the existing system is a real bandstop filter which eliminates the following two frequency components: $\omega_1 = \pm 0.3\pi$ radians/sample and $\omega_2 = \pm 0.8\pi$ radians/sample.

a) (10 pts) Assuming ideal sampling at the Nyquist rate with ideal antialiasing and reconstruction filters in the A/D and D/A, plot the magnitude of the frequency response for the effective analog filter. The x-axis of your plot should be labeled in units of radians/second (i.e., Ω). Show your work and clearly label the frequencies of all of the transitions points in the filter.

c) (5 pts) Off-the-shelf A/Ds and D/As are available with the following sampling rates: 40 kHz, 65 kHz, 80 kHz, 100 kHz, 150 kHz, 200 kHz, 300 kHz, and 500 kHz. Using higher sampling rates will make your system more expensive. Which sampling rate should you choose to make the system as cheap as possible yet still operate just as well as the original system? Justify your answer.

d) (5 pts) Assume now that the sampling rate of 300 kHz is selected. Plot the frequency response of the digital filter that will result in the newly redesigned system having exactly the same effect on the analog signal as the original system. The x-axis of your plot should be labeled in units of radians/sample (i.e., ω). Show your work and clearly label the frequencies of all of the transitions points in the filter.

4. (20 pts) Given the transfer function *H(z)* for a system as follows:

$$H(z) = 2 + \frac{3 + z^{-1}}{1 + 1.2z^{-1} + 0.7z^{-2}}$$

a) (10 pts) Draw the block diagram for the direct form II implementation of *H(z)*. Show your work and clearly give the values of all of the multipliers in the system. *Is the resulting system cannonic?* Briefly justify your answer.

b) (10 pts) Create a cascade implementation for H(z) having exactly two stages. Implement each stage using DF-II and show your work. *Is this system stable?* Briefly justify

5. (10 pts) *True or False*

a) The computational complexity of an FIR satisfying some set of specifications is generally higher than that of an IIR filter satisfying the same specifications:_______

b) Spectral transformations cannot be applied to FIR filters:_______

c) The output of the radix-2 decimation in time FFT algorithm only approximates the output of the actual DFT output:_____

d) The type of window selected in designing FIR filters directly affects the length of the filter required to achieve the required design specifications:________

e) The bilinear transformation is a linear mapping from the Laplace domain to the z-transform domain:________

f) If your original analog signal was sampled at 8Khz, then T in the bilinear transformation must be selected to be $1.25e^{-4}$:_________

g) Linear phase FIR filters can be realized with reduced complexity by sharing multipliers:________

h) Linear phase filtering cannot be achieved using causal IIR filters:_________

i) A length N Type 3 linear phase filter having real coefficients must have at least one zero valued coefficient within the range $0 \leq n \leq N$:________

j) The z-transform of a sequence will not exist unless that sequence is absolutely summable:______