The University of Texas at Austin
Dept. of Electrical and Computer Engineering
Midterm #1

Date: March 8, 2013 Course: EE 445S Evans

Name: ____
Last, First

- The exam is scheduled to last 50 minutes.
- Open books and open notes. You may refer to your homework assignments and the homework solution sets.
- Calculators are allowed.
- You may use any standalone computer system, i.e. one that is not connected to a network. ***Please disable all wireless connections on your computer system(s).***
- Please turn off all cell phones.
- No headphones allowed.
- All work should be performed on the quiz itself. If more space is needed, then use the backs of the pages.
- **Fully justify your answers.** If you decide to quote text from a source, please give the quote, page number and source citation**.**

Problem	*Point Value*	*Your score*	*Topic*
1	28		Filter Analysis
2	24		Filter Implementation
3	24		Filter Design
4	24		Potpourri
Total	100		

Problem 1.1 *Discrete-Time Filter Analysis*. 28 points.

A causal stable discrete-time linear time-invariant filter with input $x[n]$ and output $y[n]$ is governed by the following transfer function:

$$H(z) = 1 - z^{-3}$$

for $z \neq 0$.

(a) From the transfer function, derive the difference equation relating input $x[n]$ and output $y[n]$. *6 points*.

(b) Give the block diagram for the filter. *3 points*.

(c) What are the initial conditions? What values should they be assigned and why? *4 points*.

(d) Find the equation for the frequency response of the filter. Justify your approach. *6 points*.

(e) What is the group delay through the filter? *3 points*.

(f) Draw the pole-zero diagram. Is the filter lowpass, highpass, bandpass, bandstop, allpass or notch? *6 points*.

Problem 1.2 *Discrete-Time Filter Implementation*. 24 points.

Consider a causal fourth-order discrete-time infinite impulse response (IIR) filter with transfer function $H(z)$. A filter is a bounded-input bounded-output stable linear time-invariant system.

Input $x[n]$ and output $y[n]$ are real-valued.

Cascade of biquads. We factor H(z) into a product of two second-order sections (biquads)

$$H(z) = H_1(z)\, H_2(z)$$

Parallel combination of biquads. We perform partial fraction decomposition on $H(z)$ to write it as a sum of two second-order sections (biquads)

$$H(z) = G_1(z) + G_2(z)$$

(a) Draw the block diagrams for the *cascade of biquads* and the *parallel combination of biquads*. Each block in the block diagram would correspond to a biquad. *6 points*.

(b) Consider the implementation of the two filter structures on the TI TMS320C6700 DSP.

i. Compare the memory usage for the two structures. *3 points*.

ii. Compare the execution cycles for the two structures. *6 points*.

(c) Consider the implementation of the two filter structures on a processor with two TI TMS 320C6700 DSP cores (CPUs). The cores share the same on-chip memory.

i. Compare the memory usage for the two structures. *3 points*.

ii. Compare the execution cycles for the two structures. *6 points*.

Problem 1.3 *Filter Design*. 24 points.

In North America, there is a narrowband WWVB timing signal being broadcast at 60 kHz.

The G.hnem powerline communication standard uses a sampling rate 800 kHz and operates in the 34.4 kHz to 478.1 kHz band.

G.hnem receivers experience in-band interference from the WWVB signal.

(a) Design a discrete-time second-order infinite impulse response (IIR) filter for a G.hnem transceiver to remove the 60 kHz WWVB interferer. Give poles, zeros and gain. *12 points*.

(b) Design a discrete-time second-order infinite impulse response (IIR) filter for a G.hnem transceiver to extract the 60 kHz WWVB signal for use in generating timestamps for power load profiles at the consumer's site. Give poles, zeros and gain. *12 points*.

Problem 1.4. Potpourri. *24 points*.

(a) You want to design a linear phase finite impulse response (FIR) filter with 10,000 coefficients that meets a magnitude specification. Which FIR filter design method would you advocate using? *6 points*.

(b) Consider a causal first-order IIR filter with non-zero feedback coefficient a_1 and input signal $x[n]$. Output signal is $y[n] = a_1 \, y[n-1] + x[n]$. Input data, output data and feedback coefficient are unsigned 16-bit integers. As n increases, does the number of bits needed to keep calculations from losing precision always increase without bound? If yes, show that it is true for all non-zero values of a_1. If no, give a counter-example. *6 points.*

(c) Give three reasons why 32-bit floating-point data and arithmetic is better suited for audio processing than 16-bit integer data and arithmetic? *6 points*.

(d) What three instruction set architecture features would accelerate finite impulse response (FIR) filtering? *6 points*.