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

Date: October 18, 2013 Course: EE 445S Evans

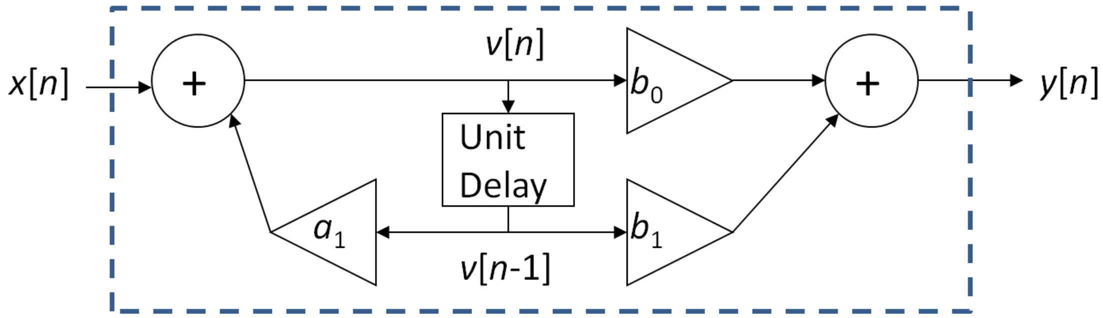
Name: ____
Last, First

- 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		Discrete-Time Filter Analysis
2	24		Discrete-Time Filter Design
3	24		System Identification
4	24		Modulation and Demodulation
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 block diagram:



Constants a_1, b_0 and b_1 are real-valued, and $| a_1 | < 1$.

(a) From the block diagram, derive the difference equation relating input $x[n]$ and output $y[n]$. Your final answer should not include $v[n]$. *6 points*.

(b) What are the initial condition(s)? What value(s) should they be assigned and why? *4 points*.

(c) What is the transfer function in the z-domain? What is the region of convergence? *5 points*.

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

(e) For $a_1 = -0.9$, $b_0 = 1$, and $b_1 = -1$, draw the pole-zero diagram. What is the best description of the frequency selectivity: lowpass, highpass, bandstop, bandpass, allpass or notch? *7 points*.

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

Consider a causal second-order discrete-time infinite impulse response (IIR) filter with transfer function $H(z)$.

The filter is a bounded-input bounded-output stable, linear, and time-invariant system.

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

The feedback and feedforward coefficients are real-valued.

You will be asked to design and implement a notch filter:

f_0 is the frequency in Hz to be eliminated, and

f_s is the sampling rate in Hz where $f_s > 2\, f_0$

Assume that the gain of the biquad is 1.

(a) Give a formula for the discrete-time frequency ω_0 in rad/sample to be eliminated. *3 points*.

(b) Give formulas for the two poles and the two zeros as functions of ω_0. *6 points*.

(c) Give formulas for the three feedforward and two feedback coefficients. Simplify the formulas to show that all of these coefficients are real-valued. *9 points*.

(d) How many multiplication-accumulation operations are needed to compute one output sample given one input sample? *3 points*.

(e) How many instruction cycles on the TI TMS3206748 digital signal processor used in lab will take to compute one output sample given one input sample? *3 points*.

Problem 1.3 *System Identification*. 24 points.

Consider a causal discrete-time finite impulse response (FIR) filter with impulse response $h[n]$.

The filter is a bounded-input bounded-output stable, linear, and time-invariant system.

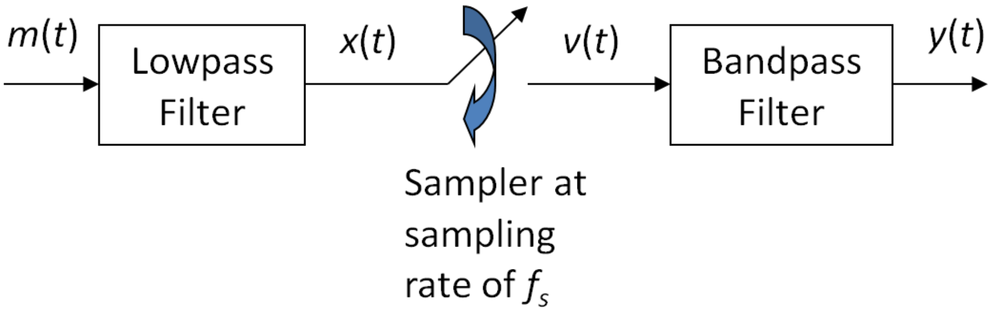
For input $x[n] = u[n]$, the output is $y[n] = \delta[n] + \delta[n-1]$.

(a) Determine the impulse response $h[n]$. *18 points*.

(b) Compute the group delay through the filter as a function of frequency. *6 points*.

Problem 1.4. Modulation and Demodulation. *24 points*.

A mixer can be used to realize sinusoidal amplitude modulation $y(t) = x(t)\cos(2\pi f_c t)$ for baseband signal $x(t)$:



Assume that $x(t)$ is a ideal baseband signal whose magnitude spectrum is zero for $|f| > f_{max}$.

Assume that $f_s > 2 f_{max}$ and $f_c = m f_s$ where m is a positive integer.

(a) Draw the magnitude spectrum of $x(t)$. *6 points*.

(b) Draw the magnitude spectrum of $v(t)$. *6 points*.

(c) Draw the magnitude spectrum of $y(t)$. *6 points*.

(d) Using only a lowpass filter, bandpass filter, and a sampler, give a block diagram for demodulation. *6 points*.