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

Date: October 16, 2015 Course: EE 445S Evans

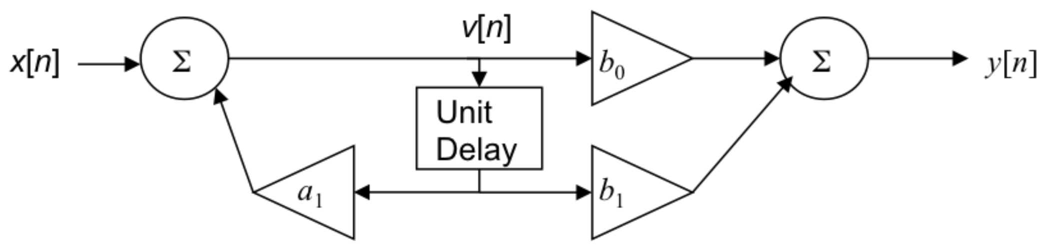
Name: ______________________________________________
Last, First

- 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 | 27 | | Discrete-Time Filter Design |
| 3 | 27 | | Modulation and Demodulation |
| 4 | 18 | | Potpourri |
| *Total* | 100 | | |

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

A causal stable discrete-time linear time-invariant filter with input signal $x[n]$ and output signal $y[n]$ is described by the following block diagram:




(a) Is this a finite impulse response filter or an infinite impulse response filter? Why? *4 points*.

(b) From the block diagram, give the difference equation relating input signal $x[n]$ and the intermediate signal $v[n]$. *4 points*.

(c) From the block diagram, give the difference equation relating the intermediate signal $v[n]$ and the output signal $y[n]$. *4 points*.

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

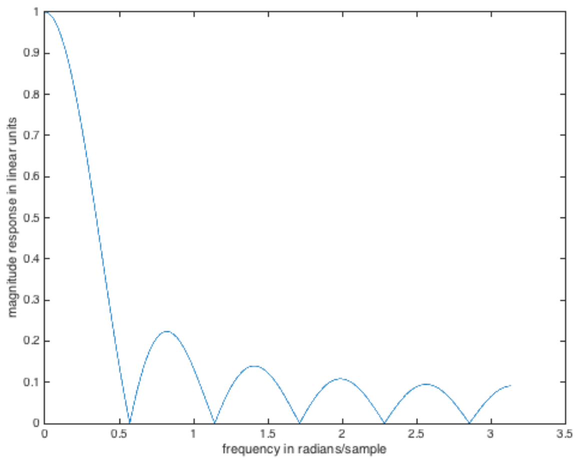
(e) Based on your answer in parts (b), (c) and (d), derive the transfer function in the $z$-domain of the filter given in part (a). *4 points*.

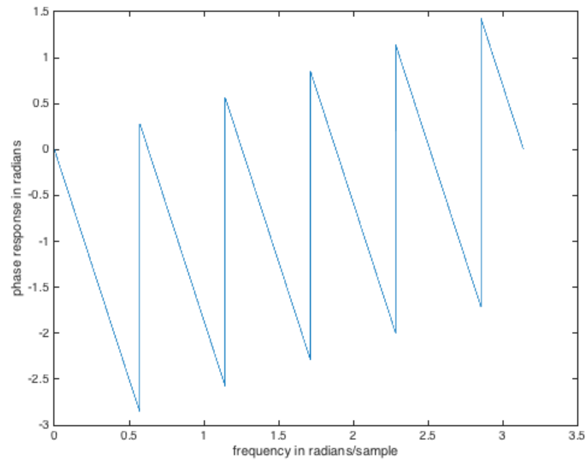
(f) Based on your answer in (e), give the frequency response of the filter. *4 points*.

(g) For $a_1 = 0.8$, $b_0 = 1$, and $b_1 = -1.05$, what is the best description of the frequency selectivity of the filter: lowpass, highpass, bandstop, bandpass, allpass or notch? Why? *4 points*

**Problem 1.2** *Discrete-Time Filter Design.* 27 points.

While searching online, you find the following magnitude and phase responses vs. rad/sample for a finite impulse response (FIR) filter. Magnitude response is in linear units (not deciBels).




(a) What is the order of the FIR filter? Why? *6 points*.

(b) Is the phase response linear? Why or why not? *6 points*.

(c) From the plots, infer the magnitude response specification for the filter. *12 points*.

i. Passband frequency (in rad/sample)

ii. Stopband frequency (in rad/sample)

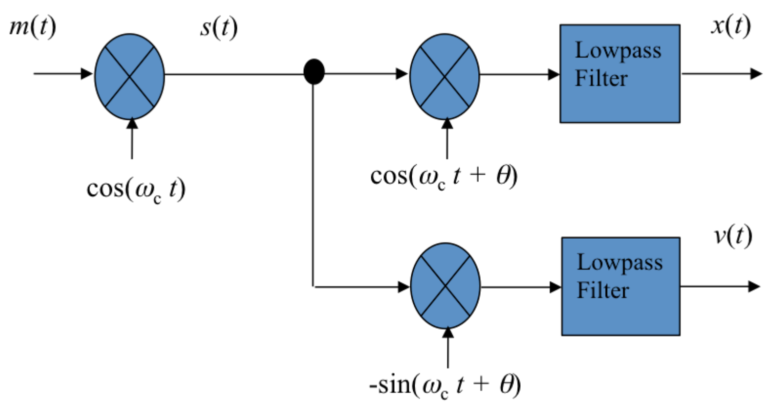
iii. Passband ripple (in dB)

iv. Stopband attenuation (in dB)

(d) Consider designing an infinite impulse response (IIR) filter to meet the magnitude response specification in part (c). Give an IIR filter design method that would give the fewest coefficients and still meet the magnitude response specification. *3 points*.

**Problem 1.3** *Modulation and Demodulation.* 27 points.

Consider sinusoidal amplitude modulation using the cosine, and sinusoidal amplitude demodulation using both the cosine and sine, as shown below. The phase offset between the receiver and transmitter is given by $\theta$.




Assume the lowpass filters are ideal and have a gain of 2. Please ignore the delay through the filters.

(a) When $\theta = 0$, derive formulas for $x(t)$ and $v(t)$ in terms of the message signal $m(t)$. *9 points*.

(b) When $\theta = \pi / 2$, derive formulas for $x(t)$ and $v(t)$ in terms of the message signal $m(t)$. *9 points*.

(c) Describe an algorithm to adjust $\theta$. *9 points*.

**Problem 1.4**. *Potpourri.* 18 points.

(a) Describe design tradeoffs in signal quality vs. implementation complexity for the following two linear phase finite impulse response (FIR) filter designs. *9 points*.

| | **Linear Phase FIR Filter Design** | | |
|---|---|---|---|
| | **100 coefficients** | **1000 coefficients** | **Increase by Factor of** |
| Multiplications/sample | | | |
| Memory size in words | | | |
| Group delay in samples | | | |

(b) Two real-time digital bandpass filter designs are being considered for an audio speaker.

- $9405^{th}$ -order linear phase finite impulse response (FIR) filter
- $12^{th}$-order infinite impulse response (IIR) filter

The group delay for the IIR filter design over passband frequencies is plotted below.

Which filter design would you advocate using? Why? *9 points*.