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

Date: October 20, 2017 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		Mixers
3	24		Filter Design
4	24		Potpourri
Total	100		

Problem 1.1 *Filter Analysis*. 28 points.

Consider the following causal finite impulse response (FIR) linear time-invariant (LTI) filter with input $x[n]$ and output $y[n]$ described by

$$y[n] = x[n] - a\, x[n-1]$$

(a) Give a formula for the impulse response $h[n]$. Plot $h[n]$. *3 points*.

(b) What are the initial conditions? What are their values? *3 points*.

(c) Draw the block diagram of the FIR filter relating input $x[n]$ and output $y[n]$. *6 points*.

(d) Give a formula for the discrete-time frequency response of the FIR filter. *4 points*.

(e) Does the FIR filter have linear phase? If yes, then give the conditions on the coefficient a for the filter to have linear phase. If no, then show that the coefficients cannot meet the conditions for linear phase. *6 points*

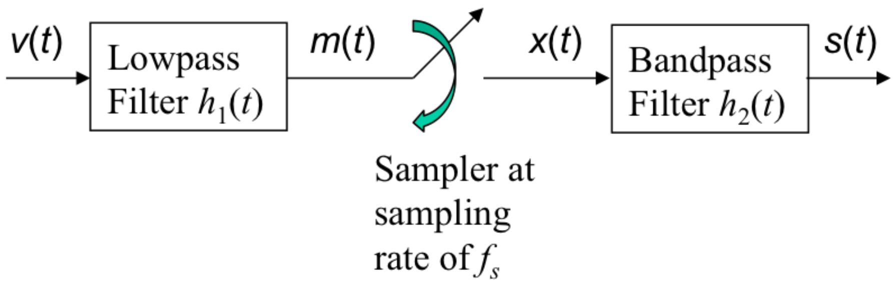
(f) If parameter a were real-valued, what are all of the possible frequency selectivities that the FIR filter could provide: lowpass, highpass, bandpass, bandstop, allpass, notch? *6 points*

Problem 1.2 *Mixers.* 24 points.

Mixing provides an efficient implementation in analog continuous-time circuits for sinusoidal amplitude modulation of the form

$$s(t) = m(t) \cos(2 \pi f_c t)$$

where $m(t)$ is the baseband message signal with bandwidth W, and
f_c is the carrier frequency such that $f_c > W$



(a) Give the passband and stopband frequencies for the lowpass filter. *3 points.*

(b) Give the passband and stopband frequencies for the bandpass filter. *3 points*

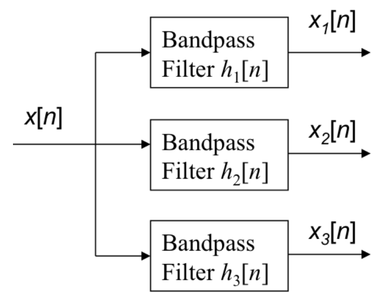
(c) Draw the spectrum for $m(t)$, $x(t)$, and $s(t)$. You do not need to draw the spectrum for $v(t)$. *9 points*.

(d) In order to simulate the mixer in discrete-time, e.g. in MATLAB, we use discrete-time filters for the lowpass and highpass filters and replace the sampling block with an upsampling block.

   i. Give the constraints on the sampling rate to convert the mixer to discrete time. *6 points.*

   ii. Determine the upsampling factor. *3 points*.

Problem 1.3 *Filter Design.* 24 points.



Some audio systems split an audio signal in three frequency bands for playback over sub-wolfer, wolfer, and tweeter speaker elements.

The block diagram on the right performs the split in discrete time:

> $x_1[n]$ contains sub-wolfer frequencies 20-200 Hz.
>
> $x_2[n]$ contains wolfer frequencies 200-2,000 Hz.
>
> $x_3[n]$ contains tweeter frequencies 2,000-20,000 Hz.

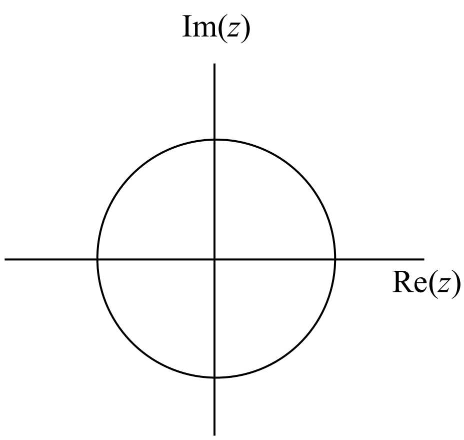
Assume that the sampling rate is 48000 Hz.

Each bandpass filter should have group delay of less than 10 ms.

(a) Give passband ripple and stopband attenuation values for these filters. *6 points.*

(b) Give passband and stopband discrete-time frequencies to design the bandpass filter $h_2[n]$. *6 points*.

(c) Draw the pole-zero diagram for a fourth-order infinite impulse response (IIR) bandpass filter $h_2[n]$. *6 points*.



(d) For a linear phase finite impulse response (FIR) filter, indicate the maximum length that would still meet the group delay constraint. The maximum length would apply to all three filters. *6 points*.

Problem 1.4. *Potpourri.* 24 points.

(a) Consider a linear time-invariant (LTI) system that has bounded-input bounded-output stability. To measure its frequency response, one could input a discrete-time unit impulse $\delta[n]$ for $-\infty < n < \infty$, find the output signal $h[n]$, and take the discrete-time Fourier transform of $h[n]$. In practice, we cannot go back to $n = -\infty$ or wait until $n = \infty$. Give a practical method using a finite-length discrete-time input signal to estimate the frequency response of the LTI system. *12 points.*

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(b) Consider the following method to compute a cosine value by using a Taylor series at $\theta = 0$:

$$\cos(\theta) = \sum_{n=0}^{\infty} \frac{(-1)^n}{(2n)!}\theta^{2n} = 1 - \frac{1}{2}\theta^2 + \frac{1}{24}\theta^4 - \ldots$$

Suppose that 10 non-zero terms were kept in the series expansion (i.e. $n = 0, 1, 2, \ldots 9$).

i. How would you minimize the number of multiplications? *6 points*.

ii. Please complete the last row of entries for the new method. *6 points*.

Method	**Multiplication-Add Operations**	**ROM (words)**	**RAM (words)**	**Quality in floating point**
C math library call	30	22	1	Second best
Difference equation	2	2	3	Worst
Lookup table	0	L	0	Best
Taylor series				

L is the smallest discrete-time period for the cosine signal.