

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

Date: March 13, 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.

| <i>Problem</i> | <i>Point Value</i> | <i>Your score</i> | <i>Topic</i>                  |
|----------------|--------------------|-------------------|-------------------------------|
| 1              | 28                 |                   | Discrete-Time Filter Analysis |
| 2              | 24                 |                   | Discrete-Time Filter Design   |
| 3              | 24                 |                   | Audio Effects System          |
| 4              | 24                 |                   | Potpourri                     |
| <i>Total</i>   | 100                |                   |                               |

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

The Al-Alaoui Differentiator is a causal stable discrete-time linear time-invariant filter with a transfer function in the following form:

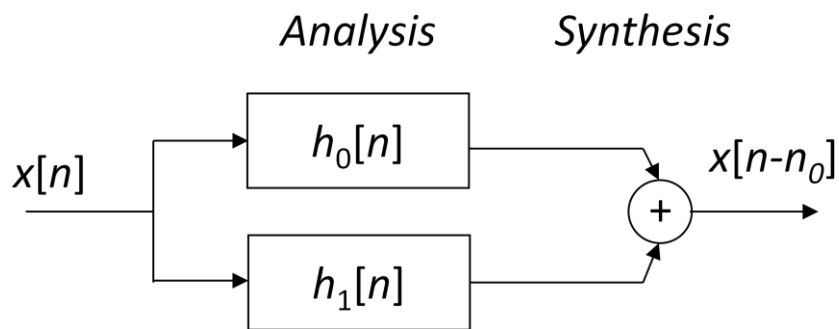
$$H(z) = C \frac{z-1}{z + \frac{1}{7}}$$

Constant  $C$  is real-valued and is not equal to zero.

- (a) Is this a finite impulse response filter or an infinite impulse response filter? Why? *4 points.*
  
- (b) From the transfer function, give the difference equation governing the filter with input  $x[n]$  and output  $y[n]$ . *4 points.*
  
- (c) Give a block diagram for the filter. *4 points.*
  
- (d) What are the initial condition(s)? What value(s) should they be assigned and why? *4 points.*
  
- (e) Find the equation for the frequency response of the filter. *4 points.*
  
- (f) What is the best description of the frequency selectivity of the filter: lowpass, highpass, bandstop, bandpass, allpass or notch? Why? *4 points*
  
- (g) Find a numeric value for  $C$  that normalizes the magnitude response. *4 points.*

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

Consider the design of the two discrete-time filters shown below:



Here,  $n_0$  is a given constant integer delay in samples where  $n_0 > 0$ .

FIR filter #0 is lowpass with cutoff frequency of  $\pi/2$  rad/sample.

FIR filter #1 is highpass with cuton frequency of  $\pi/2$  rad/sample.

At the cutoff/cuton frequency, which is in the transition band, the magnitude response is -6 dB.

The filters should have the highest stopband attenuation possible for their filter lengths.

(a) Give the filter specification and design method for FIR Filter #0 below. 9 points.

$\omega_{\text{passband}} =$

$\omega_{\text{stopband}} =$

order =

design method =

(b) Give the filter specification and design method for FIR Filter #1 below. 9 points.

$\omega_{\text{stopband}} =$

$\omega_{\text{passband}} =$

order =

design method =

(c) How would you adjust the FIR filter coefficients to make sure that the output of the synthesis section is  $x[n - n_0]$ ? 6 points.

**Problem 1.3** *Audio Effects System.* 24 points.

This problem asks you to design a discrete-time audio effects system that will

- Accept a sinusoidal signal representing a musical note as the input
- Output the input sinusoidal signal plus the same note from the next two highest octaves

For example, if a 440 Hz note ('A' in the Western scale) is the input, then the output will be the 440 Hz note plus notes at 880 Hz and 1760 Hz.

Assume that the sampling rate  $f_s$  is 44.1 kHz.

- (a) Draw a block diagram for your system. If your system generates a DC or zero-frequency component, please add a DC notch filter to your system. Sketch the Fourier transform of the output signal when the input is a sinusoidal signal. *18 points.*

- (b) How many multiplications per second does your audio effects system require? *6 points.*

