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

Date: March 13, 2015 Course: EE 445S Evans

Name: Corneil & Bernie
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		Audio Effects System
4	24		Potpourri
Total	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:

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

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.*

IIR filter. There is a non-zero pole, and from part (b), the input-output difference equation requires the previous output value.

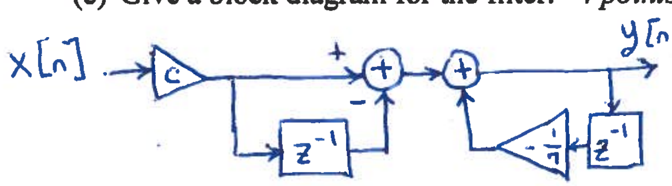
(b) From the transfer function, give the difference equation governing the filter with input $x[n]$ and output $y[n]$. *4 points.*

$$H(z) = \frac{Y(z)}{X(z)} = \frac{C - C z^{-1}}{1 + \frac{1}{7} z^{-1}} \Rightarrow \left(1 + \frac{1}{7}z^{-1}\right) Y(z) = \left(C - C z^{-1}\right) X(z)$$

$$Y(z) = -\frac{1}{7} z^{-1} Y(z) + C X(z) - C z^{-1} X(z)$$

$$y[n] = -\frac{1}{7} y[n-1] + C x[n] - C x[n-1]$$

(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.*

Let $n=0$: $y[0] = -\frac{1}{7} y[-1] + C x[0] - C x[-1]$

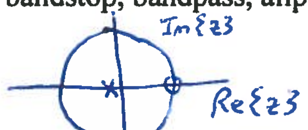
Initial conditions are $x[-1]$ and $y[-1]$. They should be set to zero to satisfy LTI properties.

(e) Find the equation for the frequency response of the filter. *4 points.*

Because the LTI system is stable, or equivalently because the region of convergence $|z| > \frac{1}{7}$ includes the unit circle,

$$H_{freq}(\omega) = H(z)\Big|_{z=e^{j\omega}} = C\,\frac{1 - e^{-j\omega}}{1 + \frac{1}{7} e^{-j\omega}}$$

(f) What is the best description of the frequency selectivity of the filter: lowpass, highpass, bandstop, bandpass, allpass or notch? Why? *4 points*



Zero at $z = 1$. Stopband centered at $\omega = 0$.

Pole at $z = -\frac{1}{7}$. Little effect on magnitude response because it isn't close to unit circle.

Highpass

(g) Find a numeric value for C that normalizes the magnitude response. *4 points.*

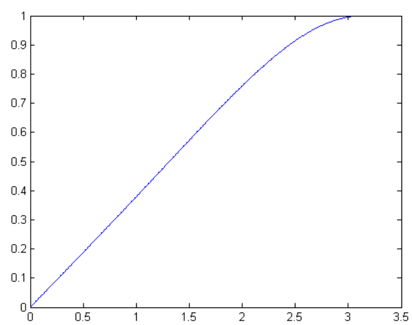
Since the filter is highpass, normalize the frequency response at $\omega = \pi$, which is $z = e^{j\omega} = -1$.

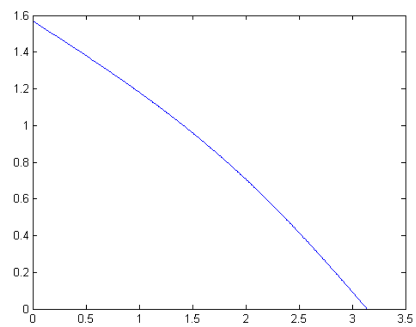
$$1 = H_{freq}(\pi) = C\,\frac{1-(-1)}{1+\frac{1}{7}(-1)} = C\,\frac{2}{\frac{6}{7}} = 1 \Rightarrow C = \frac{1}{2}\cdot\frac{6}{7} = \frac{3}{7}$$

Problem 1.1 Al-Alaoui Differentiator (more information)

Using the following Matlab code,

```
C = 3/7;
feedforwardCoeffs = C*[1 -1];
feedbackCoeffs= [1 (1/7)];
[H, w] = freqz(feedforwardCoeffs, feedbackCoeffs);
figure(1); plot(w, abs(H));
figure(2); plot(w, angle(H));
```





Magnitude Response (Linear Scale)

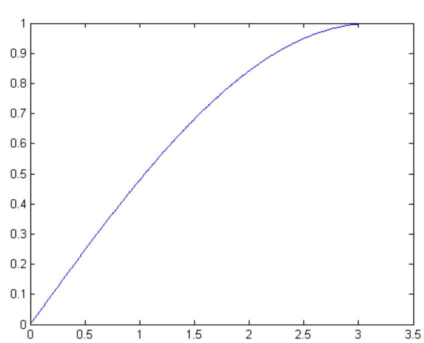
Phase Response (Radians)

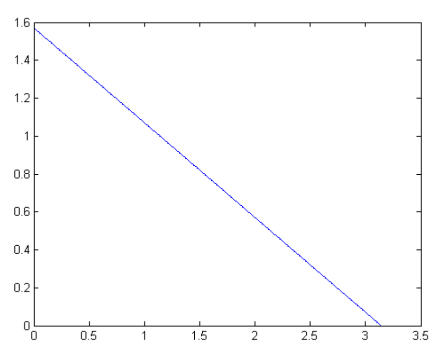
The Al-Alaoui Differentiator was first reported in the following article:

M. A. Al-Alaoui, "Novel Digital Integrator and Differentiator", *IEE Electronic Letters*, vol. 29, no. 4, Feb. 18, 1993.

Here are the magnitude and phase plots for the traditional differentiator:

```
C = 1/2;
feedforwardCoeffs = C*[1 -1];
 [H, w] = freqz(feedforwardCoeffs);
figure(1); plot(w, abs(H));
figure(2); plot(w, angle(H));
```





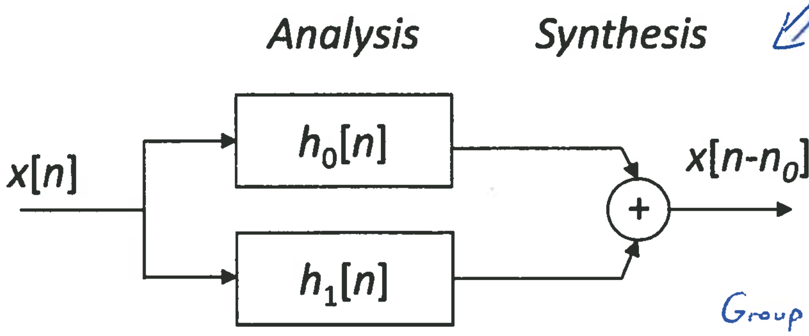
Magnitude Response (Linear Scale)

Phase Response (Radians)

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

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

$h[n] = h_0[n] + h_1[n]$



Group delay = n_0

For a constant group delay, we need to use linear phase FIR filters.

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.*

$\omega_{passband} = \frac{\pi}{2} - \frac{\pi}{40}$

$\omega_{stopband} = \frac{\pi}{2} + \frac{\pi}{40}$

order = $2n_0$

design method = Parks-McClellan.

- Cutoff frequency is near midpoint of $\omega_{passband}$ and $\omega_{stopband}$.
- We also want 10% rolloff from passband to stopband

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

$\omega_{stopband} = \frac{\pi}{2} - \frac{\pi}{40}$

$\omega_{passband} = \frac{\pi}{2} + \frac{\pi}{40}$

order = $2n_0$

design method = Parks-McClellan

- Note: If the filter order gets too high and the Parks-McClellan algorithm fails to converge, we can use the Kaiser window 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.*

Synthesis output: $y[n] = h_0[n] * x[n] + h_1[n] * x[n]$

$x[n-n_0] = y[n] = h[n] * x[n]$ where $h[n] = h_0[n] + h_1[n]$

Normalize the FIR filter coefficients to give $\sum_n h[n] = 1$

Problem 1.3 *Audio Effects System.* 24 points.

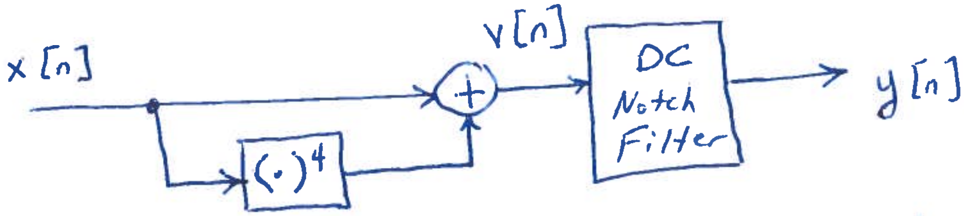
This problem asks you to design a discrete-time audio effects system that will

- 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

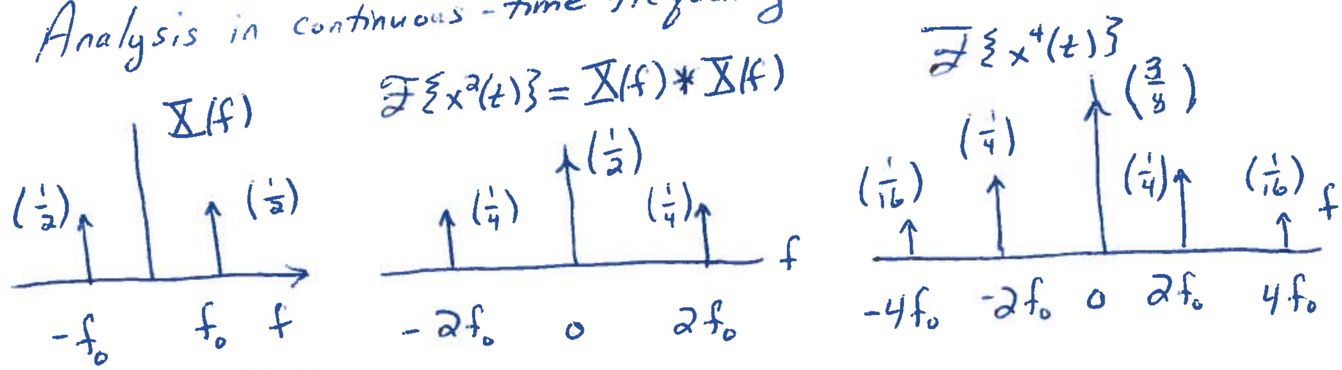
For example, if a 440 Hz note ('A' in the Western scale) is the input, then the output will be the 440 Hz note plus notes at 880 Hz and 1760 Hz.

Assume that the sampling rate f_s is 44.1 kHz.

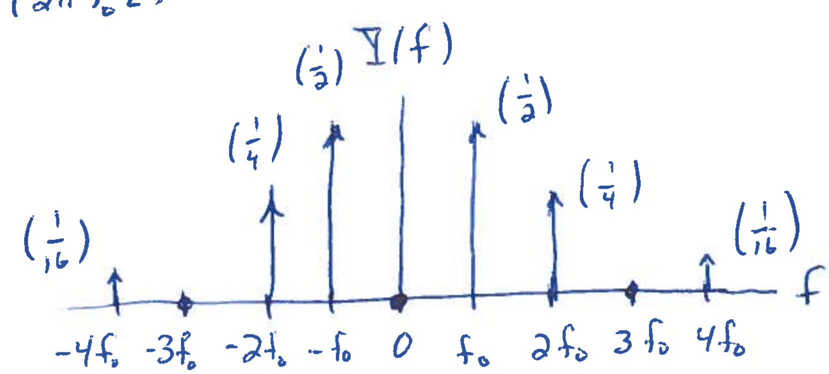
(a) Draw a block diagram for your system. If your system generates a DC or zero-frequency component, please add a DC notch filter to your system. Sketch the Fourier transform of the output signal when the input is a sinusoidal signal. *18 points.*



Analysis in continuous-time frequency domain



$x(t) = \cos(2\pi f_0 t)$



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

DC notch filter: $y[n] = 0.95\, y[n-1] + v[n] - v[n-1]$

Fourth-order power block: square the square of input

3 multiplications/sample

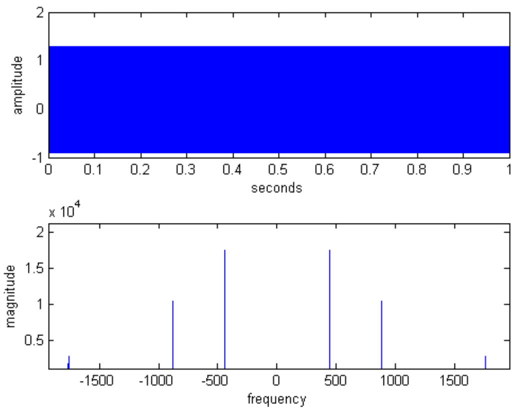
$3 f_s$

f_s samples/s

1.3 Audio Effects System

Here is the Matlab code to test the solution in part (a) for an input of a sinusoid at 440 Hz:

```
f0 = 440;
fs = 44100;
Ts = 1 / fs;
time = 1;
n = 1 : time*fs;
w0 = 2*pi*f0/fs;
x = cos(w0*n);
v = x + x.^4;
y = filter( [1 -1], [1 -0.95], v);
plotspec(y, Ts);
```



The frequencies at f_0, 2 f_0 and 4 f_0 are visible in the spectrum (bottom plot above).

One can playback the audio signal without and with DC removal. The Matlab command sound assumes that the vector of samples to be played is in the range of [-1, 1] inclusive.

```
sound(v, fs);          %%% without DC removal

sound(y, fs);          %%% with DC removal
```

To avoid clipping of amplitude values outside of the range [-1, 1], try

```
sound(v / max(abs(v)), fs);          %%% without DC removal

sound(y / max(abs(y)), fs);          %%% with DC removal
```

To hear the effect of DC offset, try

```
sound(v + 10, fs);
```

Problem 1.4. *Potpourri.* 24 points.

This problem will explore the design tradeoffs in working with a block of samples instead each sample individually.

(a) What is the advantage of moving blocks of samples from off-chip to on-chip instead of moving one sample at a time? *6 points.*

Increased throughput. Block transferred at one time – avoids overhead of reading each sample separately (bus arbitration, interrupt service routine).

(b) What is the disadvantage of moving blocks of samples from off-chip to on-chip instead of moving one sample at a time? *6 points.*

Increased latency for first sample to be processed. We have to read entire block of samples before first sample can be processed.

(c) Name and describe the subsystem on the TI TM320C6748 digital signal processor used in the laboratory component that moves blocks of samples from off-chip to on-chip? *6 points.*

Enhanced direct memory access (EDMA) controller. EDMA handles input/output off-chip/on-chip without using the CPU core. EDMA can support multiple channels at same time. EDMA can also reformat data.

(d) When implementing convolution between a finite-length signal stored on chip and a signal streaming into the processor from off chip, we encounter an issue when moving from one block of streaming samples to the next. Briefly describe what this issue is (why it exists) and how to correct for it. *6 points.*

h[n] finite length   x[n] infinite length & broken into blocks

Issue
- When block #2 is read, block #1 is overwritten (normally)
- However, convolving h[n] with block #2 requires the last samples of block #1 due to flip-and-slide.
- If h[n] has N samples, we need to keep N-1 samples of the previous block.

Solution
- We can implement this by using a circular buffer of length N-1 + BlockSize, and each new block overwrites the oldest samples.