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

Date: October 12, 2022 Course: EE 445S Evans

Name: ____
Last, First

- **Exam duration**. The exam is scheduled to last 75 minutes.
- **Materials allowed**. You may use books, notes, your laptop/tablet, and a calculator.
- **Disable all networks**. Please disable all network connections on all computer systems. You may not access the Internet or other networks during the exam.
- **Electronics.** Power down phones. No headphones. Mute your computer systems.
- **Fully justify your answers**. When justifying your answers, reference your source and page number as well as quote the particular content in the source for your justification. You could reference homework solutions, test solutions, etc.
- **Matlab**. No question on the test requires you to write or interpret Matlab code. If you base an answer on Matlab code, then please provide the code as part of the justification.
- **Put all work on the test**. All work should be performed on the quiz itself. If more space is needed, then use the backs of the pages.
- **Academic integrity.** By submitting this exam, you affirm that you have not received help directly or indirectly on this test from another human except your instructor, Prof. Evans, and that you did not provide help, directly or indirectly, to another student taking this exam.

Problem	*Point Value*	*Your score*	*Topic*
1	28		FIR Filter Analysis
2	25		IIR Filter Analysis
3	27		Filter Design
4	24		Potpourri
Total	104		

Instructor caught the error of the points adding to 104 while grading the test. The original intent was to have problem 1 count as 24 points.

Problem 1.1 *FIR Filter Analysis*. 28 points.

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

$$y[n] = a\, x[n] + b\, x[n-1] - b\, x[n-3] - a\, x[n-4]$$

for $n \geq 0$, where a and b are real-valued positive coefficients.

Please note that the coefficient in front of the $x[n-2]$ term is zero.

(a) What are the initial conditions and their values? Why? *6 points*.

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

(c) Derive a formula for the transfer function in the z-domain and the region of convergence. *4 points*.

(d) Give a formula for the discrete-time frequency response of the filter. *3 points*.

(e) Give a formula for the phase response vs. discrete-time frequency. *6 points*.

(f) Give a formula for the group delay vs. discrete-time frequency. *3 points*.

Problem 1.2 *IIR Filter Analysis*. 25 points.

For a second-order infinite impulse response (IIR) filter with poles p_0 and p_1 and zeros z_0 and z_1, the transfer function in the z-domain is

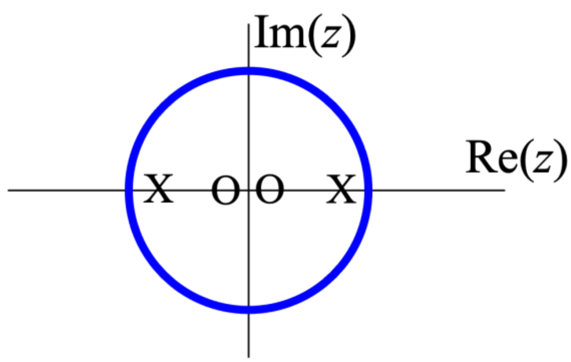
$$H(z) = \frac{(z - z_0)(z - z_1)}{(z - p_0)(z - p_1)}$$

The poles will remain at $p_0 = -0.9$ and $p_1 = 0.9$ in this problem. Region of convergence is $|z| > 0.9$.

Each question below will ask you to determine the frequency selectivity (lowpass, highpass, bandpass, bandstop, allpass and notch) of the second-order IIR filter with different choices of zero locations.

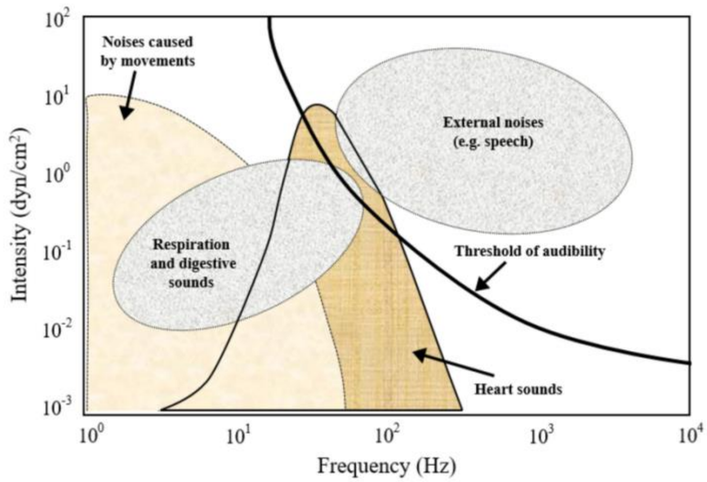
(a) From the transfer function $H(z)$, derive an expression for the magnitude response of the filter. *3 points.*

(b) Let the zeros be $z_0 = -0.1$ and $z_1 = 0.1$ as shown on the right. What is the frequency selectivity? *4 points.*



(c) Let the zeros continue being real-valued and negatives of each other, i.e. $z_0 = -z_1$. For every frequency selectivity (lowpass, highpass, bandpass, bandstop, allpass and notch) that is possible for the second-order IIR filter to achieve, give the values of zeros z_0 and z_1. You may reuse your answer from part (b). *18 points.*

Problem 1.3 *Filter design.* 27 points.



A stethoscope allows a physician to listen to sounds of the circulatory and respiratory system especially the heart and lungs.

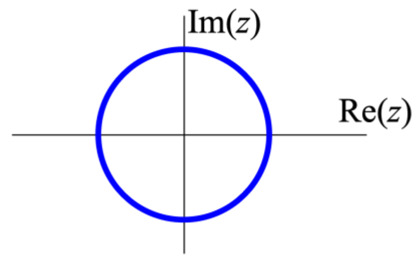
A conventional stethoscope uses a tube to directly transmit the vibration to the physician's ear.

A digital stethoscope uses a microphone and an analog-to-digital converter to record the signal.

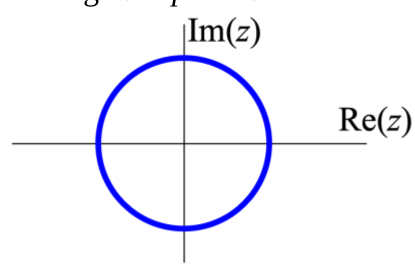
This problem will design three discrete-time filters placed in cascade for a digital stethoscope.

Assume a sampling rate of 8000 Hz.

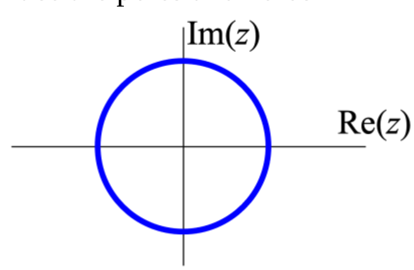
(a) Noise caused by physical motion of the patient or stethoscope occurs at extremely low frequencies (near 0 Hz). Design a first-order IIR filter to remove noise due to movement by giving thenumeric values of the one pole and one zero. Place the pole and zero on the pole-zero diagram. *9 points*.



(b) The sounds produced by the circulatory and respiratory systems occur at low frequencies (<500 Hz). Background noise in a hospital (especially from speech) occurs mostly at frequencies above 500 Hz. Design a second-order IIR filter by specifying the two poles and two zeros to remove background noise above 500 Hz. Place the poles and zeros on the pole-zero diagram on the right. *9 points.*



(c) The digital stethoscope has two modes: One for listening to respiratory and digestive sounds and one for listening to circulatory heart sounds. Design a second-order IIR filter by specifying the two poles and two zeros to extract sounds of the heart between 120 Hz and 500 Hz. Place the poles and zeros on the pole-zero diagram on the right. *9 points.*



Plot is from Nourelhuda Mohamed, Hyun-Seok Kim, Kyu-Min Kang, Manal Mohamed, Sung-Hoon Kim, and Jae Gwan Kim. "Heart and Lung Sound Measurement Using an Esophageal Stethoscope with Adaptive Noise Cancellation" *Sensors* 21, no. 20: 6757, 2021.

Problem 1.4. *Potpourri*. 24 points.

(a) Consider the signal $x[n] = (-1)^n$ observed for all time $-\infty < n < \infty$. *12 points*.

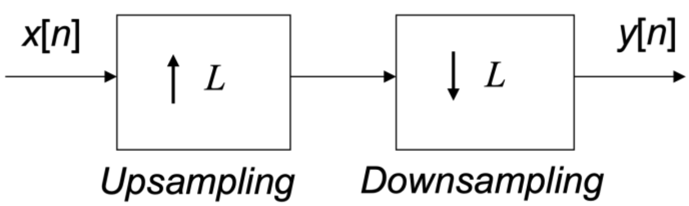
I. If we express $x[n] = \cos(\omega_0 n)$ observed for all time $-\infty < n < \infty$, give the value of ω_0. *4 points*.

II. Give a formula for $y[n]$ that is the output of downsampling by 2 applied to $x[n]$. *4 points*

III. In part II above, explain why the principal frequency of $x[n]$ became the principal frequency in $y[n]$. *4 points*

(b) Upsampling by L can be used to increase the sampling rate of the input signal by a factor of L and downsampling by M can be used to decrease the sampling rate of the input signal by a factor of M.

I. What is the sampling rate change from $x[n]$ to $y[n]$ for the system below? *6 points*.



II. What is the sampling rate change from $x[n]$ to $y[n]$ for the system below? *6 points*.