<u>Homework#9</u> Due Friday 6 December 2019 (last one!) Counts as 2 homework assignments!

1) Compact disk technology currently records/reads data from CD at a rate of 44,100 samples per second (44.1kHz).

a) In order to avoid aliasing, what is the highest significant audio frequency we should allow into the digital sampling system at the recording studio?

b) Suppose we insist on a minimum of 40dB anti-aliasing protection, and suppose we want a relatively flat audio response for our CD recording system out to approximately 15kHz.

i) Calculate the required order for a Chebyshev anti-aliasing low pass filter, given that we will allow 2dB of pass band ripple.

ii) If we choose to opt for a Butterworth filter instead (with Amax=2dB), calculate the required order.

2) You are given the following difference equation that describes a <u>causal</u> linear time-invariant discrete-time system:

$$y[n] + 0.25y[n-1] = 0.5x[n]$$

a) Find the transfer function, H(z), for this system (including the region of convergence) and plot the pole/zero diagram.

b) Calculate the impulse response, h(n), of the system.

c) Does this system have a finite impulse response (FIR) or an infinite impulse response (IIR)? Explain.

d) Does this system represent a recursive or non-recursive implementation? Explain.

e) Is this system stable? Explain.

f) Calculate the step response, s(n), of the system.

3) Design a Digital LOW pass Chebyshev filter to satisfy the specifications below. By design, I mean obtain the transfer function H(z) (i.e. the numerator and denominator coefficients, **and the K value** to get the desired DC gain). Use MATLAB to plot the magnitude response of your filter from 0 to 0.5Fs. (We will discuss how to do this in class.)

   DC Gain: 40dB; Amax=1dB; Amin=25dB
   Pass Band Limit: 1.5kHz; Stop Band Limit: 3kHz; Sampling Freq.: 10kHz

4) Design a Digital HIGH pass Chebyshev filter to satisfy the specifications below. By design, I mean obtain the transfer function (H(z)) (i.e., the numerator and denominator coefficients **and the K value** to get the desired high frequency gain). Use MATLAB to plot the magnitude response of your filter from 0 to 0.5Fs. (We will discuss how to do this in class.)

   High Freq ($0.5F_s$) Gain=40 dB; A_{max} = 1 dB; Amin=25 dB
   Pass Band Limit: 3k Hz; Stop Band Limit: 1.5kHz; Sampling Freq: 10kHz

5) Design a Digital NOTCH Butterworth filter to satisfy the specifications below. By design, I mean obtain the transfer function (H(z)) (i.e., the numerator and denominator coefficients and the K value to get the desired DC gain).. (We will discuss how to do this in class.)

   DC Gain: 20 dB; A_{max} =1 dB; A_{min}=25 dB
   Pass band Limits: 2kHz and 3kHz; Stop band Limits: 2.4kHz and 2.6Hz
   Sampling Freq.: 10kHz

6) Suppose you were testing your digital filter design code, and you wanted to try designing a Butterworth Digital Band-Pass filter with a center frequency of approximately Fs/8. After entering in your specification, suppose your program produced an answer for your design of:

$$H(z) = k\left(\frac{z^2+1}{z^2+1.4319z+0.8217}\right)\left(\frac{z^2+1}{z^2+1.1175z+0.7805}\right).$$

Unfortunately, you do not have any "correct answers" with you for comparison purposes, so you are on your own to figure out if your code is working correctly for this example. Comment on whether or not you think your code for the Butterworth Digital Band-Pass filter is producing proper results. If you think your code is fine, <u>please state that</u>, and provide some comment regarding why you think this is a valid result. If you think your code has problems, explain <u>everything</u> that concerns you about this result.