Full Name: ____

EEL 3135 (Spring 2025) – Lab #9

Question #1: *(Designing a Filter)*

There is (somewhat) no Question #1 this week (you do not have to submit anything)! Instead, we have included a demonstration on how you can design and apply your own filter. The demo isolates a single note from a clip of xylophone audio. There is also a new function in the comment code and skeleton code, `pz2ba`, which you can use to convert pole and zero locations into filter coefficients `b` and `a`. This should help you in the lab.

Question #2: *(Thinking in Three Domains)*

The next two problems represent the culmination of the past three z-transform weeks. We want you to design and implement filters with little additional guidance from us (excluding what is in `eel3135_lab10_comment`). Use your knowledge about filters in the time-domain, frequency-domain, and pole-zero space to design appropriate filter systems.

Included with the lab is an audio file "music.wav." Listen to the audio and you should hear two instruments: a string bass and a guitar. Use MATLAB to design a filter that retains the **string bass** and removes the guitar. Follow these filter design requirements: (1) can be FIR or IIR, (2) must be a causal, (3) must have no more than 10 non-zero coefficients (i.e., no more than 20 non-zero poles / zeros) in the transfer function (i.e., the filter should not be expensive to implement), (4) the output should maintain the amplitude of the desired signal.

Hint: The bass has a low frequency and the guitar has a higher frequency (Note: you may not be able to entirely retain / remove instruments).

(a) **Answer in your comments:** What type of filter did you design: low-pass, high-pass, band-pass, band-stop, all-pass? Why?

(b) **Answer in your comments:** Did you create an FIR or IIR filter? Why?

(c) Plot the frequency-domain magnitudes and the time-domains for the input and output signals ($|X(\omega)|$ and $|Y(\omega)|$) (use Question #1 as guidance).

(d) **Answer in your comments:** What do these plots tell should you about your filter?

(e) Plot the filter impulse response $h[n]$.

(f) **Answer in your comments:** What does the impulse response tell you about your filter?

(g) Plot the filter magnitude response $|H(w)|$.

(h) **Answer in your comments:** What does the frequency response tell you about your filter?

(i) Plot the pole-zero plot of your filter.

(j) **Answer in your comments:** What does the pole-zero plot tell you about your filter?

(k) Submit a `q2.wav` file with your filter audio. Also, submit your filter coefficients in a q2.mat file (using MATLAB's `save` function). The filter coefficients should be in variables called `b1` (numerator) and `a1` (denominator).

Question #3: *(Thinking in Three Domains)*

Included with the lab is an audio file "music.wav." Listen to the audio and you should hear two instruments: a string bass and a guitar. Use MATLAB to design a filter that retains the **guitar** and removes the string bass. Follow these filter design requirements: (1) must be FIR, (2) must be a causal, (3) must have no more than 20 non-zero coefficients (i.e., no more than 20 non-zero poles / zeros) in the transfer function, (4) the output should maintain the amplitude of the desired signal.

Note: You may not be able to remove all of the string bass.

(a) **Answer in your comments:** What type of filter did you design: low-pass, high-pass, band-pass, band-stop, all-pass? Why?

(b) **Answer in your comments:** Did you create an FIR or IIR filter? Why?

(c) Plot the frequency-domain magnitudes and the time-domains for the input and output signals ($|X(\omega)|$ and $|Y(\omega)|$) (use Question #1 as guidance).

(d) **Answer in your comments:** What do these plots tell you about your filter?

(e) Plot the filter impulse response $h[n]$.

(f) **Answer in your comments:** What does the impulse response tell you about your filter?

(g) Plot the filter magnitude response $|H(w)|$.

(h) **Answer in your comments:** What does the frequency response tell you about your filter?

(i) Plot the pole-zero plot of your filter.

(j) **Answer in your comments:** What does the pole-zero plot tell you about your filter?

(k) Submit a `q3.wav` file with your filter audio. Also, submit your filter coefficients in a q3.mat file (using MATLAB's `save` function). The filter coefficients should be in variables called `b2` (numerator) and `a2` (denominator).