Full Name: ____________________

EEL 3135 (Spring 2025) – Lab #05

Question #1: *(Frequency Response of Filters)*

Download `EEL3135_lab05_comment.m` from Canvas, replace each of the corresponding comments with the corresponding descriptions. This is designed to show you how to visualize the frequency response of FIR filters in MATLAB.

Note: You should run the code to help you understand how it works and help you write your comments. You will use elements of this MATLAB code for the rest of the lab assignment. **Submit only a single published PDF file** that contains both your code and the results.

Question #2: *(Frequency Filtering)*

This question will study a high-pass filter in the frequency domain and apply it to a sum of sinusoids signal. You will compare this result with the convolution in the time domain. Add all code into skeleton `eel3135_lab05_skeleton.m` from Canvas. Include all code (and functions) in this one file so that everything is published to a single PDF.

(a) Create a new function `H = FreqResponse(b,w)` that outputs the following frequency response of an FIR system:

$$H(e^{j\widehat{w}}) = \sum_{k=0}^{M} b_k e^{-j\widehat{w}k}.$$

where the input `b` is a vector of filter coefficients, input `w` is a vector of angular frequencies, and output `H` is the complex-valued frequency response. Include this function at the end of the skeleton file.

(b) Let the filter coefficients be $\{b_k\} = \{-1,-2,-1,1,2,1,-1,-2,-1\}$, generate the frequency response using the function `FreqResponse`. Plot the magnitude and phase responses. Use Question #1 as a guide.

(c) Use `FreqResponse` to determine the frequency response at $\widehat{w} = 0$, $\widehat{w} = \pi/3$, and $\widehat{w} = 2\pi/3$. Use the `disp` function to display each result.

(d) Given the following input signal:

$$x[n] = 1 + 2\cos((\pi/3)n) - \cos((2\pi/3)n + \pi/4)\ ,$$

compute the output signal $y[n]$. Use `subplot` and `stem` to plot $x[n]$ and $y[n]$ side-by-side for a range of n containing 1 fundamental period of $x[n]$. Label the horizontal axis “Samples” and vertical axis “x[n]” or “y[n]”.

(e) Using the `conv` function, compute the convolution between the input signal $x[n]$ (for a range of n containing 1 fundamental period) and coefficients $\{b_k\}$. Call this result $z[n]$. Use `subplot` and `stem` to plot $x[n]$ and $z[n]$ side-by-side for a range of n containing 1 fundamental period of $x[n]$. Label the horizontal axis “Samples” and vertical axis “x[n]” or “z[n]”.

(f) **Answer in your comments:** How does $z[n]$ compare to $y[n]$? Explain the cause for any differences.

Question #3: *(Audio Filtering)*

In this problem, we will apply filters to audio signals to observe how these filters change the audio's sound. The audio is loaded in the Lab's skeleton code. Note that the signals in this question correspond to $N = 50$ samples such that $-25 \leq n \leq 24$. For plotting frequency responses, choose a range of reasonable frequencies.

(a) For the filter coefficients `a` (or $a[n]$) in the skeleton code, compute the filter's frequency response with `FreqResponse` to get output `Ha`. Plot the magnitude and phase responses of `Ha`.
Answer in your comments: Is this a lowpass, highpass, bandpass, or bandstop filter?

(b) Use `soundsc` to play the input `x` in the skeleton code. Apply the filter `a` on `x` with `conv` and store the result in `xa`. Play `xa` with `soundsc`.
Answer in your comments: How does `x` (original sound) and `xa` (filtered sound) differ? Relate this difference to the filter magnitude response.

(c) For the filter coefficients `b` (or $b[n]$) in the skeleton code,compute the filter's frequency response with `FreqResponse` to get the output `Hb`. Plot the magnitude and phase responses of `Hb`.
Answer in your comments: Is this a lowpass, highpass, bandpass, or bandstop filter?

(d) Apply the filter `b` on `x` with `conv` and store the result in `xb`. Play `xb` with `soundsc`.
Answer in your comments: How does `x` (original sound) and `xb` (filtered sound) differ? Relate this difference to the filter magnitude response.

(e) Compute $c[n] = \delta[n] - b[n] - a[n]$ to get new filter coefficients `c`. Compute the resulting frequency response `Hc`. Plot the magnitude and phase responses of `Hc`.
Answer in your comments: Is this a lowpass, highpass, bandpass, or bandstop filter? Explain why we get this result.

(f) Apply the filter `c` on `x` with `conv` and store the result in `xc`. Play `xc` with `soundsc`.
Answer in your comments: How does `x` (original sound) and `xc` (filtered sound) differ? Relate this difference to the filter magnitude response.

(g) Plot the impulse response of the three filters above, using the `stem` function.
Answer in your comments: What is the distinguishing difference between these three impulse responses?