Mini project: **Z transform and sampling process** Due Date: 11/08(sun)/2020

You can work with up to 2 people in a group.

1. Name: ID:
2. Name: ID: by JK

The main focus of this mini project is to implement understanding of Z transform and sampling process. A file, which contains a song with some disturbing tones, is attached with this file. Your mission is to get rid of the tones so that you can enjoy the clean music. To accomplish it, you need to do the following.

a) Listen to the attached sounds and explain what is going on in the song. (What do your ears say?)

b) Define the song as $x[n]$ and plot it in the time domain, $[n \text{ vs. } x[n]]$

   i. What is the sampling rate, fs, and the length of the song in seconds?

c) Make a DTFT (discrete time Fourier transform) function and process the $x[n]$ and plot $[\Omega \text{ vs. } X(\Omega)]$ & $[f \text{ vs. } X(\Omega)]$ (you need to plot both magnitude and phase information for both of them)

d) After observing the previous spectrum (c), what is disturbing frequencies? (to see clearly you need to adjust the resolution of the frequencies-remember in DTFT, the frequency response is continuous but you want to represent in the computer (your computer can do only discrete signals))

   i. How many of them (the disturbing frequencies in the music) are there and what are those frequencies?

e) You are going to design a IIR filter, especially a notch filter that attenuates the disturbing frequencies (I showed you in the class using programs)

f) Define and write the $H(z)$ (Let's define $H(z) = \dfrac{N(z)}{D(z)}$)

g) Show coefficients of $N(z)$ & $D(z)$ (**for an example**, if you defined $\left[H(z) = \dfrac{(z-0.5)(z-0.7)}{(z-0.8)(z-0.9)} = \dfrac{z^2 - 1.2z + 3.5}{z^2 - 1.7z + 7.2}\right]$ then the filter coefficients will be $N = [1 \ \ -1.2 \ \ 3.5]$ & $D = [1 \ \ -1.7 \ \ 7.2]$)

h) Plot the frequency responds of the $H(z)$, which is $[\Omega \text{ vs. } H(\Omega)]$ in normal and dB scales (Make sure that your filter has to be normalized properly-it means that the max is 0dB or 1)

i) Plot the pole zero locations using "zplane" or similar functions in the Matlab.

j) Do the filtering process to get rid of the tone noises using the function in the Matlab called "**filter**" $[y = \text{filter}(N, D, x)]$. It means that the input signal $x[n]$ has been filtered using IIR filter (infinite impulse response filter) (You need to read the filter function in the Matlab commend line by typing help filter)

k) Listen to the sounds of filtered signal, $y[n]$

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l) Plot the frequency response of $y[n]$

m) Write observations after listening to the filtered sound. (Obviously you are supposed to hear better sound. If your filtered sound contains tone noise, you need to redesign the filter again to make it work better).

n) If the filtered output is not as clean as what you were expecting, write your opinion that why it is not working as what you thought it was. (this is important part of the solution that if you cannot find the 100 % perfect solution then you need to find out why it is)

o) Write total observation after going through this mini project