The American University in Cairo
Computer Science and Engineering Department
Fall 2022

CSCE 363/3611 – Digital Signal Processing

Assignment #2

(Due on: November 6, 2022 at mid-night)
(The assignment is individual – Submit on Blackboard as one .zip file)

Problem 1

Determine the z-transforms of the following signals including the Region of Convergence (ROC) and sketch the corresponding pole-zero plot. In the solution, you can use the z-transform pairs and properties tables:

a) $x(n) = \{-1, -2, 0, 1, \underset{\uparrow}{1}, 1, 0, 0\}$

b) $x(n) = \{10, \underset{\uparrow}{3}, -1, 5\}$

c) $x(n) = \left(\frac{1}{5}\right)^n u(n-5)$

d) $x(n) = \begin{cases} \left(\frac{1}{2}\right)^n, & 3 \le n \le 10 \\ 0, & Otherwise \end{cases}$

e) $x(n) = n^2 u(n-5)$

Problem 2

Determine the causal signal that corresponds to the following z-transforms:

a) $X(z) = \dfrac{2}{1 + 1.5z^{-1} - 0.5z^{-2}}$

b) $X(z) = \dfrac{1 - z^{-1} + z^{-7} - 0.5z^{-8}}{1 - 1.5z^{-1} + 0.5z^{-2}}$

c) $X(z) = \dfrac{z^{-4} + z^{-5}}{1 - 0.5z^{-1}}$

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Problem 3

For each of the systems defined by their impulse response function below, find the output of the system in response to the given input:

a) $h(n) = \{\underset{\uparrow}{1}, -1\}, \qquad x(n) = \{1, \underset{\uparrow}{0}, 1, 0\}$

b) $h(n) = \left(\frac{1}{2}\right)^n u(n), \qquad x(n) = \{1, \underset{\uparrow}{0}, 1, 0\}$

c) $h(n) = \left(\frac{1}{5}\right) u(n), \qquad x(n) = \left(\frac{1}{2}\right)^n u(n)$

Problem 4

Implement the following function using MATLAB or Python:

ExponentialMovingAverageFilter: A function that takes as input the file name of an input audio file, the moving average window size M, the parameter a, and the file name of the output filtered audio file. This function should convolve the exponential moving average filter with the input signal. The impulse response of the exponential moving average filter takes the form

$$h(n) = \frac{(1-a)^{|n|}}{\sum_{i=-L}^{L} (1-a)^{|i|}}, \qquad -L \le n \le L$$

where $M = 2L + 1$.

To adjust for the attenuation that occurs after filtering, scale the output obtained after filtering to have the same maximum value as the input audio.

Deliverables:

i- Your code (either MATLAB .m files or Python .py or Jupyter notebook files).

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ii- Apply the implemented function to the audio file provided "NoisyTone.wav" using $a = 0$, $a = 0.25$, and $a = 0.9$. In all cases, use $M = 201$. Name the output audio files "Filtered_a_0.wav", "Filtered_a_0_25.wav" and "Filtered_a_0_9.wav", respectively. The sampling rate of the original audio is 10 KHz.

iii- The file "CleanTone.wav" includes the same audio without noise. Plot the first 1000 samples of the clean tone, the noisy tone and each of the 3 filtered signals obtained in (ii). Comment on the quality of the output obtained in (ii) relative to the clean tone based on the figure.

iv- Apply the implemented function to the audio file provided "NoisyTone.wav" using $M = 3$, $M = 30$ and $M = 50$. In all cases, use $a = 0.25$. Name the output audio files "Filtered_M_3.wav", "Filtered_M_30.wav" and "Filtered_M_50.wav", respectively.

v- Comment on the impact of increasing M on the quality of the filtered output obtained in (iv).

Important Notes:

- All deliverables should be included in one .zip file.
- For Problems 1, 2 and 3, scanned version of handwritten solutions is acceptable.
- For Problem 4, you can use the convolution function available in MATLAB or Python to apply the filter to the input audio. Don't used ready made functions that you might find available in MATLAB or Python that directly implement the Exponential Moving Average Filter.
- This is an individual assignment.