BLG 354E
Signals & Systems for Computer Engineering
Spring 2020
Homework - 3

Due 27.05.2020 23:59

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Policy:

- Cheating is highly discouraged. It will be punished by a negative grade. Also disciplinary actions will be taken. Please do your homework on your own. Team work is not allowed. Pattern of your solutions must belong to only you.
- Upload your solutions through Ninova. Homeworks sent via e-mail and late submissions will not be accepted.
- You should write all your codes in Python language using Jupyter notebook. You can install Jupyter Notebook by following these steps on this documentation. If you are not familiar with Jupyter Notebook, you can check this tutorial.
- Prepare a report including all your solutions, codes and their results.
- You do not have to use Latex for the report but if you use Latex, you will get 10% more points. You can use this Latex template for the report.
- If you do not use Latex, the handwritten parts of the solutions must be presented on a paper legibly and scanned clearly. 10% penalty will be applied for illegible reports.

For your questions: Abdullah Ekrem Okur (okurabd@itu.edu.tr)

1. **[20 points]**

A linear time-invariant filter is described by the difference equation

$$y[n] = x[n] + x[n-1] + x[n-2] + x[n-3] + x[n-4]$$

(a) Obtain the frequency response of the system as a mathematical formula in polar form.

(b) Determine the period of frequency response, if it is periodic function.

(c) Sketch the magnitude and phase of the frequency response versus frequency for $-\pi \le \hat{\omega} \le \pi$. Label all important points: peaks, valleys, zeros, etc. Give numerical values where it is easy to estimate, e.g., at $\hat{\omega} = 0, \pi, \frac{\pi}{2}, \frac{\pi}{4}$, etc.

(d) Determine and plot the impulse response of the system.

(e) What is the output of the system, if input is

$$x[n] = 4 + 2cos[0.5\pi(n-1)] - 3cos[0.3\pi n]$$

(f) If applied to the rows or columns of an image, would this filter blur the image, or sharpen it? Briefly explain your answer.

2. **[20 points]**

In this question, you are expected to design a LTI system using two filters(a low pass filter and a high pass filter) to prove that "convolution in the time domain is equivalent to multiplication in the frequency domain."

i) Design a LTI system and select its parameters.

ii) Obtain overall impulse response and frequency response of your system. Then, show that the results are same in order to prove that property above is correct.

iii) Comment on the features of the system you designed and give an example of areas where the designed system can be used.

3. **[20 points]**

Check whether DTFT exists for the following signals. If it is, obtain the DTFT of these signals using properties of DTFT. Briefly explain your suggestions and specify which properties of DTFT you have used.

i)

$$x_1[n] = n3^{-n}u[k] + e^{j(0.3\pi n + \frac{\pi}{4})}$$

ii)

$$x_2[n] = \begin{cases} 3 - |n|, & |n| < 3 \\ 0, & |n| = 3 \end{cases} \quad \text{and} \quad x[n+7] = x[n]$$

iii)

$$x_[n] = \frac{\sin 0.8\pi n}{\pi n} - \frac{\sin 0.5\pi n}{\pi n}$$

4. **[20 points]**

Using DTFT properties and lookup table below in figure 1, calculate the inverse DTFT for the following signals:

(a)

$$X(e^{j\hat{\omega}}) = \frac{2 + \frac{1}{4}e^{-j\hat{\omega}}}{-\frac{1}{8}e^{-j2\hat{\omega}} + \frac{1}{4}e^{-j\hat{\omega}} + 1}$$

(b)

$$X(e^{j\hat{\omega}}) = \begin{cases} 1, & 0.25\pi \leq |\hat{\omega}| < 0.75\pi \\ 0, & |\hat{\omega}| \leq 0.25\pi \quad \text{and} \quad 0.75\pi \leq |\hat{\omega}| < \pi \end{cases}$$

5. **[20 points]**

With the HW file, you can find two folders containing the audio files of two prestigious historians of Turkish academic community: İlber Ortaylı and Emrah Safa Gürkan. Each audio file has the same length of nearly 5 seconds (22501 data samples).

a) Write a script that calculates fft for these audio files and plots the fft magnitude. You can use only one of the sound channels. To make *np.fft.fft* function work properly, normalize the signal by dividing it by *max(abs(signal))*

b) Implement a basic algorithm (it may contain some machine learning methods or even just some if-elses) to classify an audio file as belonging to İlber Ortaylı or Emrah Safa Gürkan. Test your approach on the files given in the test folder. Explain your approach. What makes this so easy?

Table of DTFT Pairs	
Time-Domain: $x[n]$	*Frequency-Domain:* $X(e^{j\hat{\omega}})$
$\delta[n]$	1
$\delta[n-n_d]$	$e^{-j\hat{\omega}n_d}$
$r_L[n] = u[n] - u[n-L]$	$\dfrac{\sin(\frac{1}{2}L\hat{\omega})}{\sin(\frac{1}{2}\hat{\omega})} e^{-j\hat{\omega}(L-1)/2}$
$r_L[n]\, e^{j\hat{\omega}_0 n}$	$\dfrac{\sin(\frac{1}{2}L(\hat{\omega}-\hat{\omega}_o))}{\sin(\frac{1}{2}(\hat{\omega}-\hat{\omega}_o))} e^{-j(\hat{\omega}-\hat{\omega}_o)(L-1)/2}$
$\dfrac{\sin(\hat{\omega}_b n)}{\pi n}$	$\begin{cases} 1 & \lvert\hat{\omega}\rvert \le \hat{\omega}_b \\ 0 & \hat{\omega}_b < \lvert\hat{\omega}\rvert \le \pi \end{cases}$
$a^n u[n] \quad (\lvert a\rvert < 1)$	$\dfrac{1}{1 - ae^{-j\hat{\omega}}}$
$-b^n u[-n-1] \quad (\lvert b\rvert > 1)$	$\dfrac{1}{1 - be^{-j\hat{\omega}}}$

Basic Discrete Time Fourier Transform Pairs[1]

References

[1] J. H. McClellan and R. Schafer, "Digital signal processing first," *IEEE Signal Processing Magazine*, vol. 16, no. 5, pp. 29–34, 1999.