BLG 354E Homework - 2

Due 07.04.2018 23:59

Policy:

- Cheating is highly discouraged. It could mean a zero or negative grade. Please do your homework on your own. Team work is not allowed. Pattern of your solutions must be belong to only you.
- Prepare reports using LaTeX. Otherwise, you will get 0 point.
- After the deadline, your point will decrease with slope -10 according to the number of days past.
- You will get 50 points from completeness of your report and 50 points from selected 3 questions.
- In Problems 2, 6, and 8, you will write code in Python 3.5+.
- Upload your solutions with code files through Ninova (Do not forget to upload code files separately.).
- There will be no postponement on the due date.
- If you see any mistake in the homework, inform me as soon as possible.

For your questions: albay@itu.edu.tr

1. Give derivation of DFT starting from CTFT and passing through DTFT ("Give derivation" means "show how to obtain" not "derivative ("türev" in Turkish)"). Explain your derivation process in very detail. Why discrete signals have different frequency in a limited interval?

2. In this question you will implement very simple voice recognition method using Fourier Transform. You will use Python library functions.

   Record your own voice 20 times for each word by saying "One" and "Two" (There will be 40 audio files in total.). Load your voice files into your program (Be careful! Your voice records contain two channels in general. You can use only one of them.). Take Fourier Transform of your records using fft function of Numpy library (not other than Numpy). Get a sufficient number of coefficients of Fourier Transform. You will use these coefficients for classification.

There is a class called Pipeline in scikit learn library. It helps you to apply different methods sequentially. Using Pipeline class of scikit learn library, first of all reduce dimension of your Fourier Transform coefficients using PCA algorithm of scikit learn lbirary. After dimensionality reduction, using LogisticRegression class of the same library, classify your voice as one or two. Use 30 of your voice (15 ones and 15 twos) to train your classifier and remaining 10 of them to validate your classifier.

I suggest you that start with first 10 fourier coefficients of the transform and increase one by one. During PCA start with 1 components and increase one by one. You must try to find best classification using minimum dimension (I mean you must try to reduce minimum dimension using PCA. For example, you can select first 100 fourier coefficients and using PCA you can reduce it 2.). Explain your code in detail using comments.

3. Give definition of the following terms: *unit impulse*, *unit impulse response*.

$$DONOTFORGET$$

Side note: The impulse response provides a complete characterization of the filter, because the convolution sum gives a formula for computing the output from the input when the unit impulse response is known.

4. Derive the convolution sum formula step by step using LTI system properties. Explain each step.

   (a) What is the intuition behind the convolution? Explain. (*Hint:* Use linearity and time-invariance of systems)

5. Determine whether each of the following LTI systems are Casual and Stable.

   (a) $y(t) = x(t-4) + x(t+2) + 5\frac{d(x)}{dt}$

   (b) $y(t) = \int_{-\infty}^{t} x(\tau)d\tau$

(c) $h(t) = e^{-(t-5)}u(t-5)$

(d) $h(t) = u(t) - e^{-3t}u(t)$

6. Implement *MyConv* function without using built-in function of Python such as *convolve*. Test your function using signal $x[n] = \{2, 4, 6, 4, 2\}$ and impulse response $h[n] = \{3, -1, 2, 1\}$.

7. $x[n] = \{2, 4, 6, 4, 2\}$ and $h[n] = \{3, -1, 2, 1\}$. Calculate $y[n] = x[n] * h[n]$ using matrix-vector multiplication.

8. Explain briefly *convolve2d* library function of Scipy. Convolve noisyCameraman.png image with following 3x3 smoothing box filter using *convolve2d*. Discuss the resulting image briefly.

$$K = 1/9 \begin{bmatrix} 1 & 1 & 1 \\ 1 & 1 & 1 \\ 1 & 1 & 1 \end{bmatrix}$$