SEAS Winter 2020
Semester-6
Digital Signal Processing

LAB 4

Objectives:

Understand different concepts of Fourier transform along with its applications.

Prerequisites:

- DTFT, DFT and circular convolution.

Important notes:

- Initially each student needs to execute DFT using FFT MATLAB command. Then create your own DFT function and again execute all following problems using your own created function. At the end all need to compare the output, obtain using FFT MATLAB command and your own function. Verify answer using IFFT MATLAB command wherever required.

Problems

1. Consider analog sinusoidal wave $x(t) = 4\cos(100\pi t)$ is sampled at twice the Nyquist rate for full one period. Find out its DFT. If x(t) is sampled at Nyquist rate for one period what will be its DFT?. Instead of one period if two full periods are sampled then how the result is affected in both cases?

   Write a program to compute the DFT of given sinusoidal wave by specifying amplitude, frequency, sampling frequency and number of periods to obtain DFT from user. Plot magnitude spectrum with horizontal axis indicating the analog frequency. Assume the phase of the sinusoidal wave is zero.

2. Use MATLAB to compute DFT of following sequences and verify the answer by finding IFFT command. Display output sequences on command window and also obtain magnitude and phase spectrum.

   a) x(n) = {1,2,3,4}
   b) x(n) = $u(n) - u(n-3)$ for N=4 and N= 8

3. Convolution Application using DFT-IDFT

   a. Write a MATLAB program to find circular convolution of two sequences using DFT-IDFT based approach.

      - $x_1(n) = \{1,-1,-2,3,-1\}$ and $x_2(n) = \{1,2,3\}$
      - $x_1(n) = \{1, 2, 1, 2\}$ and $x_2(n)= \{3,2,1,4\}$

   b. Write a program to perform linear convolution using above DFT-IDFT approach based circular convolution.
   c. Plot and Verify above sequences for (a) and (b)

4. Application of Fourier transform in Audio signal processing Application (Transform and Noise removal)

   a. Explore following commands

      - **Audioread**
      - **length**
      - **audiowrite**
      - **fftshift**
      - **filter**
      - **butter (Butterworth filter) [i.e [b,a] = butter(3, [0.3 0.7],'bandpass')]**

   b. Take an audio file "sample_sound.wav" from shared folder. Apply different commends which are mentioned above on given audio file to find sampling frequency, frequency range of audio input file and normalized frequency. Plot the normalized audio signal. Take Fourier transform to plot and analyze magnitude spectrum of the same.

   c. Now generate one noise signal inform of sinusoidal wave and add it to audio file which you have read. Generate noisy audio file and save it. By applying Butterworth equation find different filter coefficients. Once filter coefficients has been received, apply filter on noisy audio file using generated filter coefficients. Generate the filtered output audio file and save it. Plot Filtered Signal. Now listen original file, noisy audio file and filtered output audio file one by one.