Experiment 1

DFT and Its Properties

1. Purpose
   Main purpose of the experiment is to study the generation of discrete-time signals, reviewing some properties of the Discrete-Fourier transform (DFT) and using it as a tool for signal analysis. Another aim of this experiment is to study thhe effects of changing the sampling rate in discrete-time systems. Refer to the references at the end of the sheet for the required background on this topic.

2. Preliminary Work
   a. Write a MATLAB function $x = gen(A, f, T, F_S)$, that generates a discrete-time sinusoidal signal $x[n]$ with amplitude A. Assume that x[n] is the output of a C/D converter at sampling rate F_S, with a sinusoidal input of duration T seconds, at $f\ Hz$.
   b. Run your function gen with the following parameters:
   $$A = 1;\ f = 25\,Hz;\ T = 2\,sec;\ F_S = 100\,Hz$$
   And obtain $x_1[n]$. Using MATLAB's fft function obtain the DFT of $x_1[n]$ and comment on the resulting spectra.
   Note on the MATLAB function fft: The fft function computes the DFT of a vector $\boldsymbol{x}$. If the length of the vector $\boldsymbol{x}$ is an integer power of two, then it is possible to compute the DFT using faster algorithms, which are called as Fast-Fourier Trasnform (FFT) algorithms. MATLAB has a built in function named fft, which depending on the data length, computes the DFT either directly or using the FFT algorithm.

   c. Use the function gen one again to generate $x_2[n]$, with parameters:
   $$A = 3;\ f = 10\,Hz;\ T = 2\,sec;\ F_S = 100\,Hz$$
   Take DFT of $x_2[n]$ and call it $X_2[k]$. PLot the magnitude and phase spectra of $x_2[n]$ and comment on the resulting spectra.

   d. Now generate the signal $x[n] = x_1[n] + x_2[n]$. Check numerically tht DFT is a linear transform. PLot the magnitute spectrum of $x[n]$ and comment on the resulting spectrum.
   e. Generate the signal $x[n] = x_1[n]x_2[n]$. Plot the magnitute spectrum of $x[n]$ and comment on thhe resulting spectrum.
   f. Generate the signal $x[n] = x_1[n-1]$ which is the one sample sihfted version of $x_1[n]$. Assuming $x_1[n]$, $0 \leq n \leq N-1$, generate $x[n]$ such that $x[0] = x_1[N-1]$. Observe numarically the effect of this operation on the fft of both signals.
   g. Load the file Television.wav (You would use MATLAB command 'audioread' to load this file. Use MATLAB help to learn the usage of 'audioread'). This file contains a portion of speech waveform. Plot the waveform and its magnitude spectrum.

Up- and Down-Sampling

In saome cases, signals have to be processed at a sampling rate other than they have been obtained. This is the subject of Multirate-Signal Processing. Below we consider two examples oon this topic in which the effect of reducing the sampling the sampling rate is studied.

   h. Generate signals:
   $$x_1[n] = sinc(0.2 * (n - 128)), n = 0, \ldots, 255$$
   $$x_2[n] = sinc(0.8 * (n - 128)), n = 0, \ldots, 255$$

i. Downsample $x_1[n]$ by two such that $xdown1[0] = x_1[0]$. Plot $x_1[n]$, $xdown1[n]$ and their magnitude spectra. Comment on the plots. Using MATLAB's interp function interpolate $xdown1[n]$ by two to obtain $xinterp1$. Compare the signals $x_1[n]$ and x$interp1$ both in time and frequency domains.

j. Downsample $x_2[n]$ by two such that $xdown2[0] = x_2[0]$. Plot $x_2[n]$, $xdown2[n]$ and their magnitude spectra. Comment on the plots. Using MATLAB's interp function interpolate $xdown2[n]$ by two to obtain $xinterp2[n]$. Compare the signals $x_2[n]$ and $xinterp2[n]$ both in time and frequency domains.