DSP – Practice Problem Set #3

1. For the following continuous-time signals, (a) compute and plot the discrete-time signal with a suitable sampling rate and (b) compute and plot its discrete Fourier transform as magnitude and unwrapped phase.
    a. $x_1(t) = 3\,Sinc(100\,x)$
    b. $x_2(t) = \begin{cases} 0 & t < 1 \\ 1 & 1 \le t \le 10 \\ 0 & t > 10 \end{cases}$
    c. $x_3(t) = \begin{cases} 0 & t < 0 \\ \sin(5\pi t) & 0 \le t \le 8 \\ 0 & t > 8 \end{cases}$
2. For the discrete signals given as

$$x_1(n) = 3\,Sinc(n), \quad -16 \le n < 16$$

$$x_2(n) = \begin{cases} 0 & n < 4 \\ 1 & 4 \le t \le 15 \\ 0 & n \ge 16 \end{cases}$$

Compute and plot: (a) the circular convolution $x_1 \circledast x_2$, (b) the linear convolution $x_1 * x_2$.

3. Design discrete-time FIR and IIR filters for the following specifications:
    a. Low-pass filter with cutoff frequency of 40 Hz and passband/stopband attenuations of 1dB and 40dB respectively and given that the stopband frequency is 50 Hz and that the sampling rate is 1kHz.
    b. Low-pass filter with normalized discrete cutoff frequency of 0.3 and stopband frequency of 0.45 and passband/stopband attenuations of 1 and 1000 respectively.
    c. Band-pass filter with passband between the discrete frequencies of 0.5 and 1.0 and stopband attenuation of 60 dB at discrete frequencies of 0.2 and 1.4 respectively.

    Plot the response of these filters including the magnitude and phase response as well as the group delay associated with it. Compare the FIR and IIR filters for each case and comment on the results.

4. Compute the DFT of the following window functions and compare their magnitude response to produce plots similar to those in the lecture notes and textbook:
    a. Rectangular window
    b. Hamming window
    c. Kaiser window

5. Design an IIR filter based on a Butterworth transfer function with order 10 and cutoff frequency of 5 kHz given that the sampling frequency is 20 kHz. In particular, use Matlab to compute and plot the following:
    a. Filter coefficients
    b. Magnitude and phase responses
    c. Group delay
    d. Filter poles and zeros
    e. Filter realization using basic gain, delay and sum elements