DIGITAL SIGNAL PROCESSING LABORATORY

Course Name: Digital Signal Processing **Course Code**: ECE-205

Using the computational programming platform MATLAB, students are required to write the following computational programs and in turn simulate for their deep understanding and learn the classroom fundamentals via MATLAB.

1. (a) Represent the discrete-time (DT) signal:

$$x(n) = \begin{cases} 2n, & -3 \le n \le 3 \\ 0, & otherwise \end{cases}$$

and plot it using stem function.

(b) Represent the DT signal: $x(n) = e^{j\left(\frac{\pi}{8}\right)n}$ for 0≤n≤32 and plot the real, imaginary, absolute and angle (phase angle in radians) parts. Now convert and plot the phase angle in degrees and make sure that your plot should have a title, as well as labels for both axes.
(c) Determine (using MATLAB) and plot even and odd parts of the following DT sequence: $x(n) = 0.8^{n}$

2. Write the program to:
(a) Generate unite impulse, unit step and signum function.
(b) Generate square wave (*both continuous plot and discrete plot*) of frequency 50 Hz and sampling frequency 1000 Hz: having duty cycle of 25%, 50% and 75%.
(c) Generate sinusoidal signal with user input multiple frequencies, amplitudes and its phases

3. Write a MATLAB program to
(a) Determine linear convolution (without direct MATLAB command) of any two user defined finite length sequences.
(b) Determine linear convolution of any two user defined finite length sequences using direct MATLAB command. Verify the result of part (a).
.
4. Write a MATLAB program to (a). Determine N-point DFT of the user input sequence (without using direct command) and plot its magnitude and phase spectrum.
(b) Compute IDFT of the result obtained in (a). Verify the results using direct MATLAB commands.

5. For the sequences $x(n) = [1, 1,1,2,1,1] and\ h(n) = [1,1,2,1]$
(a) Compute circular convolution (without using direct command).
(b) Compute linear convolution using circular convolution (as computed in (a))
(c) Compute circular convolution using linear convolution and verify with result obtained in (a).

(d) Compute circular convolution using DFT-IDFT method. In all cases verify the results using direct MATLAB commands.

6. Write a MATLAB program to compute 4-point DFT of the user input sequence using

(a) Divide and Conquer Method

(b) Radix-2 DIT algorithm.

7. Write a MATLAB program to determine the time domain and frequency domain response of fixed windows by taking N=25 and N=1024.

8. Write a MATLAB program to design FIR low pass filter using all windows with the following specifications: pass band edge frequency= 1.5 KHz, transition width 0.5 KHz, stop band attenuation > 20dB, sampling frequency=8 KHz.

9. To design filter using Kaiser with the following specifications: pass band 150-250 Hz, transition width 50 Hz, pass band ripple 0.1 dB, stop band attenuation 60 dB, sampling frequency 1 KHz.

10. (a) Design a Butterworth analog high pass filter with the specifications as: pass band ripple=0.2, stop band ripple=40, pass band frequency=2 KHz, stop band frequency=3.5 KHz, sampling frequency=8 KHz.

(b) Convert designed analog filter into digital filter using impulse invariant transformation.

11. (a) Design a Chebyshev Type-1 analog low pass filter with the specifications as: pass band ripple=0.23, stop band ripple=47, pass band frequency=1.3 KHz, stop band frequency=1.55 KHz, sampling frequency=7.8 KHz.

(b) Convert designed analog filter into digital filter using bilinear transformation.