CE1.1-R4: DIGITAL SIGNAL PROCESSING

NOTE:

1.	**Answer question 1 and any FOUR from questions 2 to 7.**
2.	**Parts of the same question should be answered together and in the same sequence.**

Time: 3 Hours **Total Marks: 100**

1.

a) Sketch the pole-zero plot of the following z-transforms and shade the Region Of Convergence (ROC)

$$X_1(z) = \frac{1 - \frac{1}{2}z^{-1}}{1 + 2z^{-1}}$$, $x_1[n]$ is non-causal sequence.

b) Find Impulse response h[n] of the stable linear time-invariant system, whose input and output satisfy the difference equation:

$y[n] - 0.5\, y[n-1] = x[n] - 0.25\, x[n-1]$.

c) Determine Fourier transform of the signal

$$x[n] = a^{|n|}, \qquad -1 < a < 1$$

d) What is the condition to avoid time domain aliasing to recover x[n] from its periodic extension in Discrete Fourier Transform (DFT)? What is the significance of zero padding in DFT?

e) Describe properties of Region of Convergence (ROC) of z-transform.

f) Draw direct-form structure of Finite Impulse Response (FIR) system represented as a non-recursive difference equation:

$$y[n] = \sum_{k=0}^{M-1} h(k)x(n-k)$$

g) Design a single-pole low pass digital filter with a 3-dB bandwidth of 0.2π, using Bilinear transformation applying to the analog filter:

$$H(s) = \frac{\Omega_c}{s + \Omega_c}$$

(7x4)

2.

a) Determine the z-transform and its ROC of the following sequence:

i) $x[n] = (1 + n)\, u[n]$

ii) $x[n] = (-1)^n\, 2^{-n}\, u[n]$

b) A causal Linear Time Invariant(LTI) system with impulse response h[n] and system function:

$$H(z) = \frac{(1 - 2z^{-1})(1 - 4z^{-1})}{z(1 - 0.5z^{-1})}$$

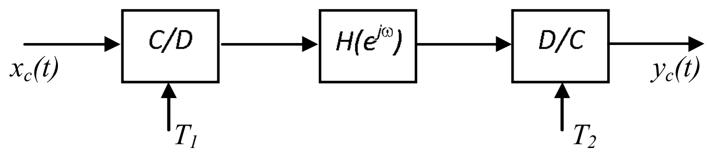
i) Draw a direct form II flow graph.

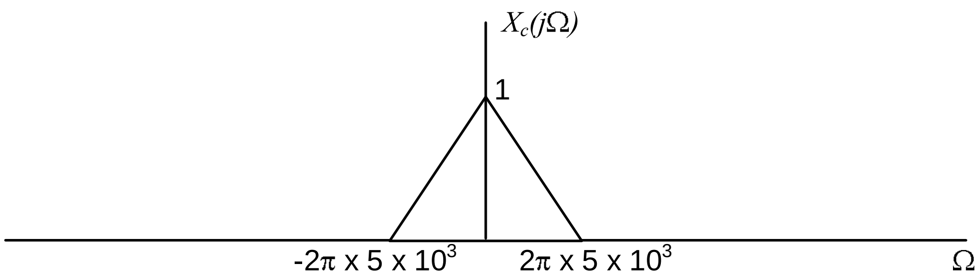
ii) Draw the transposed form of the flow graph in **Part i)**.

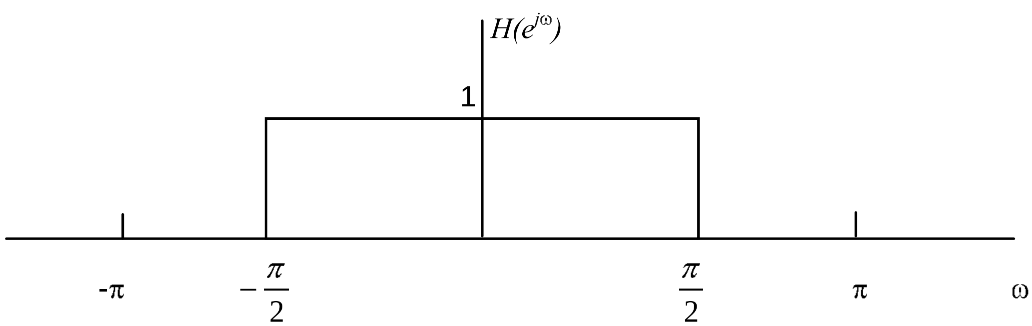
c) Consider the following system, sketch and label the Fourier transform of $y_c(t)$ for the following two cases.

i) $1/T_1 = 2 \times 10^4$, $1/T_2 = 10^4$.

ii) $1/T_1 = 10^4$, $1/T_2 = 2 \times 10^4$.







(6+6+6)

3.

a) Determine and sketch for the linear convolution y[n] of the signals:

$$x[n] = \begin{cases} \frac{1}{3}n, & 0 \le n \le 6 \\ 0, & elsewhere \end{cases}$$

$$h[n] = \begin{cases} 1, & -2 \le n \le 2 \\ 0, & elsewhere \end{cases}$$

b) Describe mathematically the conversion of lattice coefficients to direct-form filter coefficients in Finite Impulse Response Lattice structure.

c) A real finite-length sequence,

$$x[n] = \{\underset{\uparrow}{1}, \tfrac{3}{4}, \tfrac{1}{2}, \tfrac{1}{4}\}$$

The 4-point DFT of x[n] is denoted as X[k]. Plot the sequence y[n] whose DFT is $Y[k] = W_4^{3k} X[k]$.

(6+6+6)

4.

a) Determine the response of the relaxed system characterized by the impulse response:

$$h[n] = \left(\frac{1}{2}\right)^n u(n)$$

and to the input signal

$$x[n] = \begin{cases} 1, & 0 \le n < 10 \\ 0, & Otherwise \end{cases}$$

b) Explain Machine Vision ***Or*** Video Segmentation.

c) A signal x[n] is discrete time sequence

$$x[n] = \{-1, 2, -\underset{\uparrow}{3}, 2, -1\}$$

With its Fourier Transform X(ω). Determine the following quantities, without explicitly computing X(ω).

i) X(π)

ii) Angle of X(ω)

iii) $\int_{-\pi}^{\pi} X(\omega)\, d\omega$

(8+6+4)

5.

a) Explain the Application of DSP in Global Positioning System (GPS).

b) The complex sequence:

$$x[n] = \begin{cases} e^{jw_0 n}, & 0 \le n \le N-1 \\ 0, & otherwise \end{cases}$$

i) Find the Fourier transform X(ω) of x[n].

ii) Find the N-point DFT X[k] of the finite length sequence x[n].

c) Determine the lattice coefficients corresponding to the Finite Impulse Response filter with given system function

$$H(z) = A_3(z) = 1 + \frac{13}{24} z^{-1} + \frac{5}{8} z^{-2} + \frac{1}{3} z^{-3}$$

(8+6+4)

6.

a) Explain the basic features and Advantages of TMS320C40 DSP Co-processor.

b) Develop a radix-3 decimation-in-time FFT algorithm for $N = 3^v$ and draw the corresponding flow graph for N = 9. What is the number of required complex multiplications?

c) Using the radix-2 decimation-in-frequency algorithm, compute the 8-point DFT of the sequence, $x(n) = \{0.5, 0.5, 0.5, 0.5, 0, 0, 0, 0\}$.

(6+6+6)

7.

a) An IIR digital low-pass filter is required to meet the following specifications:

Passband ripple: ≤ 0.5 dB, Passband edge: 1.2 kHz, Stopband attenuation: ≥ 40 dB

Stopband edge: 2 kHz, Sample rate: 8 kHz

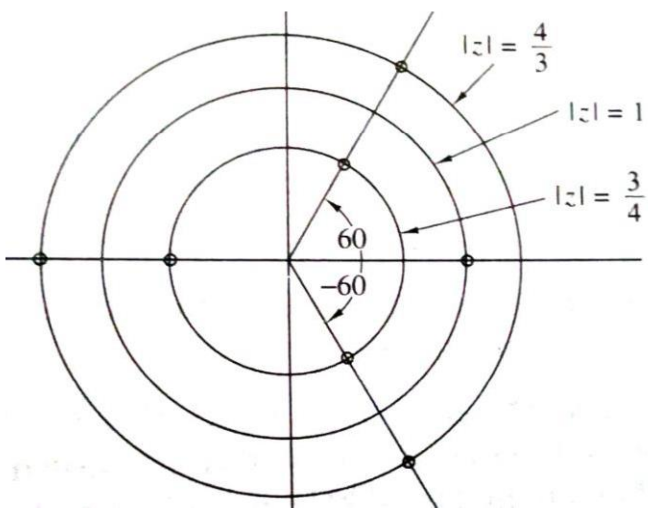
The filter is to be designed by performing a bilinear transformation on an analog system function. To meet the specifications in the digital implementation what should be the order of Buttorworth and Chebychev analog designs.

b) Determine the inverse z-transform of

$$X(z) = \frac{1}{1 - 1.5z^{-1} + 0.5z^{-2}}$$

Using long division method, if (i) ROC: $|z| > 1$ and (ii) $|z| < 0.5$.

c) The pole-zero plot shown in given figure:



i) Does it represent an FIR filter?

ii) Is it a linear-phase system?

(6+6+6)