Sardar Patel Institute of Technology
Bhavan's Campus, Munshi Nagar, Andheri (West), Mumbai-400058, India
(Autonomous College Affiliated to University of Mumbai)

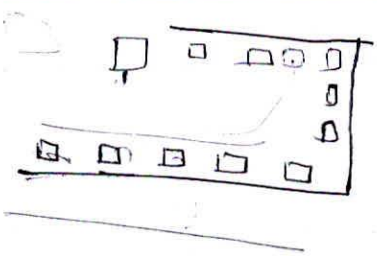
End Semester Examination
DECEMBER 2023

Max. Marks: 100
Class: TE ELECTRONICS & TELECOMMUNICATION
Course Code: EC303
Name of the Course: Digital Signal Processing

Duration: 180 Min
Semester: V
Branch: EXTC

Instruction:
(1) All questions are compulsory.
(2) Use of scientific calculator is allowed.
(3) Draw neat diagram.
(4) Assume suitable data if necessary with justification.

		Max. Marks	CO
Q.1	Attempt the following Short Answer Questions.		
(1)	For the DFT of each real sequence compute the missing values (1.1) P[k] = {0, ---, -12-10j, -10, ---, 1+j } (1.2) Q[k] = {10, 2+2j, ---, 1+j, 0, 1-j, -2.4+4.8j, ---}	2	CO2
(2)	Let h[n] be the unit impulse response of a Low Pass Filter with a cutoff frequency w_c, what type of filter has a unit sample response $g[n] = (-1)^n h[n]$?	2	CO4
(3)	Let $x[n]=\left(-0.4\right)^{n-1}u[n-1]+\left(-0.5\right)^{n}u[n]$ Find X(z)	2	CO3
(4)	Show that if $Z = 0.25$ is ZERO of the filter then $Z= 4$ is also a ZERO of the Linear Phase filter.	2	CO4
(5)	Derive the relationship between Analog Filter Frequency and Digital Filter Frequency when BLT Method is used for filter design?	2	CO5
(6)	Let x[n] be a four-point sequence with X[k] = {1, 2, 3, 4}. Find the DFT of p[n] using X[k] where p[n] = x[n-2].	2	CO2
(7)	An Audio signal x(t) band-limited to 4 KHz is sampled with a sampling frequency of 8 KHz for a duration of 10 seconds of time. The DFT of samples of x[n] is then computed. To what Analog frequency does the index k=0 correspond?	2	CO1
(8)	Given $H(z) = 0.1 + 0.2\, z^{-1} + 0.3\, z^{-3}$ Find the response of the filter to the input $x[n] = (0.5)^n \mathrm{Sin}(0.25\pi n)\, u[n]$.	2	CO4



(9)	A continuous Time Signal $x(t) = 1.8 \sin(120\pi t) + 20 \sin(80\pi t)$ is sampled with Fs = 1000 times per seconds for a period of 60 seconds. How many samples will be obtained in 60 seconds of time. Justify your Answer.	2	CO1
(10)	What is the effect of zero padding of signal in frequency domain?	2	CO2
Q.2 (A)	Assume that a Real Multiplication takes 1 microsecond & a Real Addition takes 1 microsecond. How much time it will take to find circular convolution of M=21 point h[n] and L = 480 point x[n] using FFT-IFFT.	10	CO2
Q.2 (B)	Given $H(z) = \frac{z}{(z-0.3)\ (z-2)\ (z-0.6)}$ Find h(n) for the following ROC conditions: a) $\lvert z\rvert > 2$ b) $\lvert z\rvert < 0.3$ c) $0.6 > \lvert z\rvert > 0.3$ d) $2 > \lvert z\rvert > 0.6$ e) State in which of the above system is stable and justify.	10	CO3
Q.3 (A)	Design 4^{th} order Digital Butterworth HPF with pass band cutoff frequency of 0.5π radians. [6 Marks] Draw Realization Diagram [4 Marks] **OR** A Digital Butterworth Low Pass Filter is required to meet the following specifications: Pass band ripple $\leq$ 1.5 dB Stop band attenuation $\geq$ 42 dB Pass band edge = 4.2 KHz Stop band edge = 5.8 KHz Sampling rate = 24 KHz Find the filter order and cutoff frequency of the filter if (a) Impulse Invariant Method is used (b) BLT technique is used.	10	CO5
Q.3 (B)	Design a second order Linear Phase Low Pass FIR filter with one of the ZERO at z = - 0.5. [4 Marks] Find the response of the filter to the input x[n] = {1, 2, 3, 4, 5, 6, 7 } using Overlap Add Method. [6 Marks]	10	CO2
Q.4 (A)	A filter is required to be designed with the following specifications, $H_d(e^{jw}) = \begin{cases} 2\, e^{-j3w} & -0.45\pi \leq w \leq 0.45\pi \\ 0 & Otherwise \end{cases}$ Using the Hamming window. [8 Marks] Hamming window function is given by,	10	CO5

$$w[n] = \begin{cases} 0.54 - 0.46 \cos\left(\dfrac{2\pi n}{N-1}\right) & 0 \le n \le N-1 \\ 0 & \textit{otherwise} \end{cases}$$

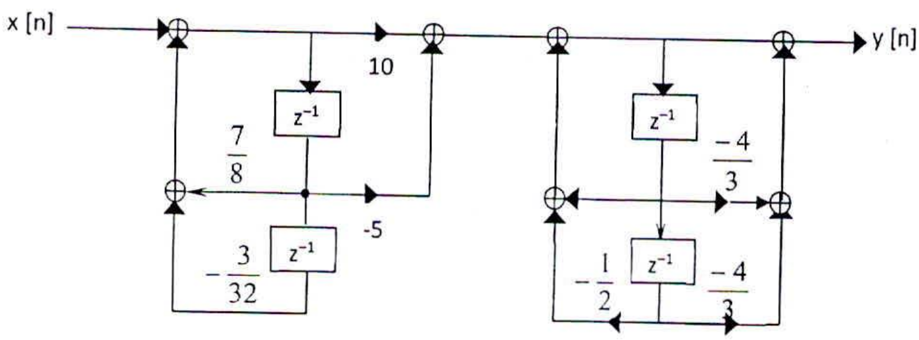
Draw Linear Phase Realization Diagram. [2 Marks]

OR

Design sixth order Linear Phase Band Pass FIR filter with pass band frequencies $w_1 = 0.45\,\pi$ and $w_2 = 0.65\,\pi$ using Frequency Sampling Method [8 Marks]

Draw Linear Phase realization diagram. [2 Marks]

Q4 (B) Given | 10 | CO4



(1) Find H(z). [6 Marks]

(2) Calculate Magnitude Response at w = 0 and w = π [2 Marks]

(3) Comment on Stability and Causality of the system. [2 Marks]

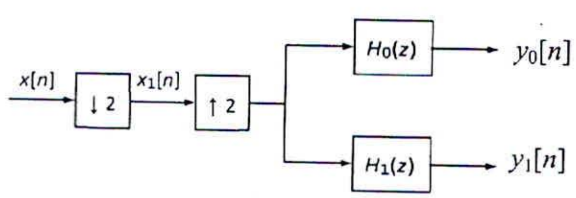
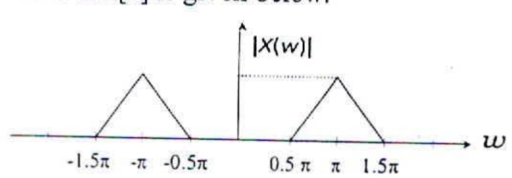
OR

Given $H(z) = \dfrac{2z^2}{z^2 + 0\cdot 1z - 0\cdot 06}$

(a) Find the impulse response of the filter. Identify the filter type (IIR Filter Or FIR Filter)? [4 Marks]

(b) Draw POLE and ZERO Diagram. Is the filter Stable? Justify your answer. [3 Marks]

(c) Find Difference Equation. State whether the System is Recursive or Non recursive. Justify. [3 Marks]

Q.5 **(A)**	What will be the reconstructed analog signal? Consider the signal, **$x(t) = 120 \cos(50\pi t)\, u(t) + 130 \sin(300\pi t)\, u(t) - 140 \cos(100\pi t)\, u(t)$** 1. If the signal is sampled with Fs = 200 Hz, What will be the DT signal obtained after sampling? Plot first five samples of DT Signal thus obtained. [4 marks] 2. Classify the sampled signal: Even/Odd Signal, Periodic/Non Periodic, Causal/Anti-casual/Both Sided Signal, [4 Marks] 3. Each sample is scaled in the range [0 to 255] & digitized using ADC with eight-bit resolution. If the binary samples thus obtained are transmitted serially, what will be the data bit rate? [2 Marks]	10	CO1
Q.5 **(B)**	It is required to design an audio signal processing system that will decompose input digital audio signal into four frequency bands without overlapping. The output of each filter is compressed with 50% compression factor and the compressed files are stored separately. Specify the type of filters to be used with cutoff frequencies of each filter. Draw the Framework/block diagram of the DSP system. Justify the need of each process. Assume that Sample Audio Signal of user is already captured for 3 seconds of time. **OR** For the system below, sketch $X_1(w)$, $X_2(w)$, $Y_0(w)$ and $Y_1(w)$  Magnitude Spectrum of x[n] is given below.  $H_0(z)$ is LPF and $H_1(z)$ is HPF.	10	CO6