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 calculators is allowed.
(3) Draw a neat diagram.
(4) Assume suitable data if necessary, with justification.

		Max. Marks	CO
Q.1	Attempt the following Short Answer Questions.		
(1)	Consider filter with transfer function $H(z)=\frac{z-4}{z-0.25}$ Identify the type of filter and justify it.	2	CO4
(2)	Given $H(e^{jw}) = e^{-j2w}$ [24 + 4.8 cos(2w) + 6.2 cos(w)]. Find h[n].	2	CO4
(3)	Given $h[n] = (0.4)^n$ for $0 \le n \le 2$ Find the response of the filter to the input $x[n] = (0.5)^n u[n]$	2	CO4
(4)	Justify that antisymmetric h(n) cannot be used for Linear phase Low pass FIR filter design?	2	CO5
(5)	Let $x[n]=(0.5)^{n-1}u[n-1]+(0.4)^{n-2}u[n-2]$ Find X(z)	2	CO3
(6)	Derive the relationship between Analog Filter Frequency & Digital Filter Frequency when Impulse Invariant Method is used for filter design?	2	CO5
(7)	Let x[n] be four-point sequence with X[k] = { 1, 2, 3, 4 }. Find the DFT of p[n] using X[k] where $p[n] = (-1)^n x[n]$	2	CO2
(8)	A signal x(t) that is band limited to 10KHz, is sampled with a sampling frequency of 20 KHz. The DFT of N = 1000 samples is then computed. To what analog frequency does the index k=150 correspond ?	2	CO2
(9)	Show that Frequency Mapping from Analog Filter to Digital filter is, One to One when BLT Method is used for filter design.	2	CO5

(10)	How many real multiplications and additions are required to find DFT of 256-point signal.?	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 the output of FIR filter with M=25-point h[n] and L = 1000 point x[n] using FFT-IFFT.	10	CO2
Q.2 (B)	Given x(n) = { 1, 1 , 1, 1, 0, 0, 0, 0}. (a) Find X[k] by using FFT. (b) Let p[n] = { 1, 1, 1, 1, 1, 1, 1, 1 } Find P[k] using X[k] (b) Let q[n] = { 1, 1, 1, 1, −1, −1, −1, −1} Find Q[k] using X[k] **OR** Derive DIT-FFT Flowgraph for N=6 using 3-point basic flowgraph.	10	CO2
Q.3 (A)	Given y[n] = 10x[n] + 0.5 y[n-1] Find the response of the given IIR filter to the input. (1) x[n] = 20 cos (0.25 π) u[n]. (2) x[n] = 10 Sin (0.5 π)	10	CO4
Q.3 (B)	A discrete time signal x(n) is defined as $x(n) = \begin{cases} 1+\frac{n}{3}, & -3 \le n \le -1 \\ 1, & 0 \le n \le 3 \\ 0, & elsewhere \end{cases}$ i) Determine its values. Can you express the signal x(n) in terms of δ(n) and u(n)? ii) Sketch the following signals and identify the type of operation involved: a) x (n – 2) b) x (3 – n) c) x (n – 1) · δ (n – 1) d) Even part of x(n) e) x(2n)	10	CO1
Q.4 (A)	Show that if Phase response of the FIR filter is Linear phase, then output of the filter is delayed version of the input signal during passband. [5 Marks] **AND** If the phase response of the FIR filter is NOT Linear Phase, then output of the filter is a distorted version of the input signal. [5 Marks]	10	CO5
Q.4 (B)	Let $x[n]=(0.25)^n \sin(0.5\pi n)u[n] + (0.5)^n \cos(0.3\pi n)\, u[n]$. Find X(z).	10	CO3

Q.5 (A)	A filter is required to be designed with the following specifications, $$\mathbf{H}(e^{jw}) = \begin{bmatrix} 2\,e^{-j3w} & \frac{-3\pi}{4} \le w \le \frac{3\pi}{4} \\ 0 & \textit{Otherwise} \end{bmatrix}$$ Calculate N=11-point h[n] using Frequency Sampling Method. **OR** Given Ap = 1.88 dB As = 13.89 dB $w_p = 0.75\,\pi$ $w_s = 0.35\,\pi$ (a) Design minimum order Digital Butterworth filter (b) Show Realization Diagram.	10	CO5
Q.5 (B)	A system based on sensor using DSP processor is required to be designed for Measurement & analysis of Sound pollution in the atmosphere. If the Sound Noise level is above the user defined threshold value, then an appropriate error message should get displayed on the screen. (1) Draw block diagrams of the complete DSP system. Justify the need of each block. (2) Write Algorithms/Flowchart to address the problem **OR** An Audio signal band limited to 3.4 KHz is sampled with a sampling frequency Fs = 44100 Hz and the sampled signal is stored on Pen Drive. It is required to obtain 48000 samples per second from 44100 samples obtained per second. 1. Devise a strategy to convert 44100 samples to 48000 samples per second. 2. Draw a block diagram and explain each block in detail to convert 44100 samples to 48000 samples per second.	10	CO6