THAPAR INSTITUTE OF ENGINEERING & TECHNOLOGY, PATIALA

Electronics and Communication Engineering Department

(188)

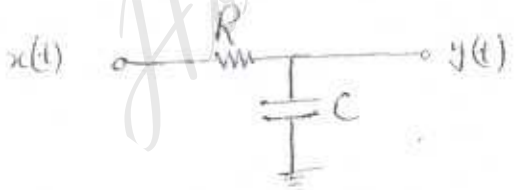
END Semester Examination BE (ECE and EIC) Final Year

EC- 011 (Digital Signal Processing)

Max Marks: 72 Instructor: Dr. Kulbir Singh Time allowed: 3Hrs

NOTE : 1. Attempt any **FIVE** questions.
2. Assume missing data, if any, appropriately.
3. All parts of a question must be done at one place.
4. OVER ATTEMPTED QUESTIONS WILL NOT BE EVALUATED.

1	a)	Determine the response of the LTI system $y(n)=5\,y(n-1) - y(n-2) + 6\,x(n)$ to the input signal $x(n)= \delta(n) - 0.2\,\delta(n-1)$.	6
	b)	Comment upon the linearity, causality, stability and time invariance propert of the discrete time system and jutify your answer **i)** $y(n) = x(2n)$ **ii)** $y(n) = x(-n+2)$ **iii)** $y(n) = x(n)\,u(n)$	6
	c)	Explain the concept of frequency in continuous and discrete time domain.	2.4
2	a)	Find the Z-transform of the following: **i)** $x(n) = n^2u(n)$ **ii)** $x(n) = na^nu(n)$ $-1 < a < 1$ **iii)** $x(n) = (1/2)^n\,[u(n)-u(n-10)]$	6
	b)	Find the inverse Z-transform of the following: **i)** $X(z) = \log(1 - z^{-1})$ $\quad |z| > \frac{1}{2}$ **ii)** $X(z) = \frac{z^2}{(z-0.5)(z-1)^2}$ **iii)** $X(z) = \frac{z^2 + 2z + 1}{z^2 - z + 0.3561}$	6
	c)	Explain the requirement of Region of Convergence (ROC) in Z-transform.	2.4
3	a)	Calculate the linear convolution, circular convolution, auto correlation and cross correlation of the sequences $x(n) = [1,2,3,4]$ and $h(n) = [1,2,1,1]$.	6
	b)	Compute the 8-point DFT of Hanning window using radix-2 DIT algorithm with the help of neat sketch.	6
	c)	List the properties of Discrete Fourier transform.	2.4
4	a)	A filter function $h(n)= [2\ ,1\ ,0,1]$ is given and it is desired to filter a long data sequence $x(n)= [1, 2, 4, 6, 5, 3, 4, 2, 1, 3, 5, 7, 5, 3, 2, 1, 3, 4, 5, 6, 5, 4\ldots\ldots]$. Calculate the output of the filter for the given filter function using a method of filtering of long data sequences.	6
	b)	Using Divide and conquer approach calculate the discrete Fourier transform of signal $x(n) = [1, 2, 1, 2, 1, 1, 1, 1, 0, 1, 0, 1, 1, 2, 3, 4]$	6
	c)	What is alias frequency? Explain with suitable example.	2.4

5	a)	Obtain the coefficients of a linear phase FIR filter to meet the specifications given below using the window method. Stopband attenuation 41 dB Passband ripple 0.01 dB Transition width 5 kHz Sampling frequency 100 kHz Ideal cutoff frequency 12 kHz	6
	b)	Using pole-zero placement method, obtain the transfer function, realization and the difference equation of a digital notch filter that meets the following specifications: Notch frequency 50 Hz 3 dB width of notch ± 5 Hz Sampling Frequency 500 Hz	6
	c)	Differentiate FIR and IIR filters.	2.4
6	a)	A requirement exists to simulate in a digital computer an analog system with the following normalized characteristics: $H(s) = \frac{1}{s^2 + s + 1}$ Obtain a suitable transfer function using the impulse invariant method. Assume a sampling frequency of 10 kHz and a 3 dB cutoff frequency of 1.5 kHz.	
	b)	Determine, using BZT method the transfer function and difference equation for the digital equivalent of resistance- capacitance (RC) filter shown in figure below. Assume a sampling frequency of 150 Hz and cut off of 30 Hz.  Figure	6
	c)	Discuss the design of an FIR filter using Kaiser window. Also give the required equations.	2.4
7	a)	Explain in detail the process of conversion of an analog signal to digital signal.	6
	b)	Discuss Butterworth, Chebyshev and Elliptical Filter approximations in detail.	6
	c)	List various applications of DSP and discuss any one of them in detail with neat sketch.	2.4