Visvesvaraya Technological University, Belagavi
CBCS Scheme: 2015-16

MODEL QUESTION PAPER
Fifth Semester Electronics & Instrumentation Engineering

15EI/BM52

Fundamentals of Signals and DSP

Time: 3 Hrs Max. Marks: 80

Note: Answer FIVE FULL Questions, selecting ONE FULL Question from each Module

Question Number	Question	Marks Allotted
	Module -1	
1a	Give the Classification of the Discrete Time Signals with suitable examples.	5
1b	Determine the response y[n] of the system, given $x[n] = \begin{cases} \lvert n\rvert, -3 \le n \le 3 \\ 0, \quad \text{otherwise} \end{cases}$ & y[n] = x[n] + x[n-1]	5
1c	The impulse response of a linear time-invariant system is $h[n] = \begin{bmatrix} 2,3,1,-4 \\ \uparrow \end{bmatrix}$. Compute the response of the system to an input signal $x[n] = [1,3,5,2]$	6
OR		
2a	With a suitable example, describe the different representations of a discrete time signal.	4
2b	State the Sampling theorem and also compute the Nyquist rate for the analog signal $x[n] = 3\cos 50\pi t + 10\sin 400\pi t - \cos 150\pi t$.	6
2c	Obtain the cross correlation of the sequences $\text{x}[n] = \begin{bmatrix} 1, & 2, & 3,1 \\ & \uparrow & \end{bmatrix}$ and $y[n] = \begin{bmatrix} 3, & 1,2,2 \\ & \uparrow \end{bmatrix}$	6
	Module -2	
3a	Evaluate the Z- transform (with ROC) of the sequences $x[n] = \begin{bmatrix} 1,2,3,-4 \\ \uparrow \end{bmatrix}$ & $y[n] = 3^n u[n]$	6
3b	Using partial fraction expansion, obtain x[n] for $X(z) = \dfrac{z}{3z^2 - 4z + 1}$ $\lvert z\rvert > 1$	6
3c	Check the causality of the LTI system given by	4

	$H(z)=\frac{z^3+2z^2}{z-\frac{1}{2}}; ROC: \left\|z>\frac{1}{2}\right\|$	
	OR	
4a	Determine the system function and unit sample response of the system described by the difference equation y[n] = 0.5 y[n-1] + 2 x[n]	6
4b	List the properties of the Z-transform.	4
4c	Draw the direct form I & II structures of the filter given by $H(z)=\frac{1-3z^{-1}}{1+2z^{-2}-4z^{-3}}$	6
	Module -3	
5a	State and explain the following DFT properties: Time Reversal and Periodicity	6
5b	Draw the 8-point DITFFT Signal Flow Graph.	5
5c	Explain how linear filtering is carried out by using DFT.	5
	OR	
6a	Using DFT and IDFT, compute the circular convolution of the sequences: x[n]= {2, 1, 3} and h[n]= {1, 2, 5, 2}	12
6b	Given X(k) = [0 , 1+j , 1 , 1-j], using the properties of the DFT, compute the DFT W(k) of the sequence $w(n) = x((n-1))_4$	4
	Module -4	
7a	Labeling the important specifications, draw the figure of the magnitude characteristics of a physically realizable low pass filter.	4
7b	Compute the order and cut-off frequency of an analog Butterworth filter for the following specifications : attenuation of -2dB at 20 rad/sec, stopband attenuation of more than 10dB beyond 30 rad/sec	8
7c	Draw the structure of a single stage lattice filter.	4
	OR	
8a	For the analog transfer function $H(s)=\frac{1}{s+1}-\frac{1}{s+3}$, determine H(z) using impulse invariant method if T=1s.	8
8b	Design an FIR linear-phase, digital filter approximating the ideal frequency response.	8

	$$H_d(e^{j\omega}) = \begin{cases} 1, \text{ for } \lvert\omega\rvert < \frac{\pi}{4} \\ 0, \text{ for } \frac{\pi}{4} < \lvert\omega\rvert < \pi \end{cases}$$. Using a rectangular window of length, N = 9	
	Module -5	
9a	For the signal $x[n] = \begin{bmatrix} 1,2,3,-4,5,6,8,9 \\ \uparrow \end{bmatrix}$, compute the decimated signal w[n] = x[3n] decimated by a factor of 3 and also the interpolated signal y[n] = x[n/2] interpolated by a factor of 2.	6
9b	Describe the realization of the Analysis and Synthesis Filter bank.	6
9c	What is the principle of operation of an adaptive filter?	4
	OR	
10a	Illustrate a few applications of the Adaptive filter.	6
10b	With a neat block diagram, discuss the architecture of the TMS320c54xx Processor.	10