SREENIVASA INSTITUTE OF TECHNOLOGY AND MANAGEMENT STUDIES

(Autonomous)

DEPARTMENT OF ELECTRONICS AND COMMUNICATION ENGINEERING

QUESTION BANK

18ECE 323– DIGITAL SIGNAL PROCESSING

Question No.	Questions	PO Attainment
	UNIT – 1: DISCRETE FOURIER TRANSFORM	
	PART A (2 Marks)	
1	Define digital signal processing.	PO1
2	Define circular convolution?	PO1
3	What is zero padding? What are its uses?	PO1
4	Define DFT of a discrete time sequence?	PO1
5	State the properties of DFT?	PO1
6	. Define discrete time signals and classify them?	PO1
7	What is meant by radix-2 FFT?	PO1
8	Why FFT is needed?	PO1
9	State periodicity property with respect to DFT?	PO1
10	State time reversal property with respect to DFT?	PO1
	UNIT – 2: IIR FILTER DESIGN	
11	What are the different types of filters based on frequency response?	PO2
12	State the structure of IIR filter?	PO2
13	.What is bilinear transformation?	PO2
14	What is Warping Effect?	PO2
15	Which types of structures are used to realize IIR systems??	PO2
16	.Write the expression for order of Butterworth filter?	PO2
17	. Why feed back is required in IIR systems?	PO2
18	What are the advantages & disadvantages of bilinear transformation?	PO2
19	.What is meant by impulse invariant method of designing IIR filter?	PO2
20	How one can design digital filters from analog filters?	PO2
	UNIT3 – : FIR FILTER DESIGN	
21	.What are FIR filters?	PO3
22	Write the steps involved in FIR filter design?	PO3
23	What are the advantages of FIR filters?	PO3
24	What are the disadvantages of FIR filters?	PO3
25	Distinguish between FIR and IIR filters?	PO3
26	What is the necessary and sufficient condition for linear phase characteristic in FIR filter?	PO3
27	List the steps involved in the design of FIR filters using windows.	PO3
28	Give the equation specifying Hanning and Blackman windows	PO3
29	Draw the direct form realization of a linear Phase FIR system for N even	PO3
30	Draw the direct form realization of FIR system?	PO3
	UNIT – 4: FINITE WORD LENGTH EFFECTS	
31	What is meant by floating point representation	PO4

32	What is meant by fixed point number?	PO4
33	What are the advantages of floating point representation?	PO4
34	What is input quantization error?	PO4
35	What is product quantization error?	PO4
36	What are the different quantization methods?	PO4
37	What is truncation?	PO4
38	What is rounding?	PO4
39	What are the two types of limit cycle behavior of DSP?	PO4
40	What are the methods to prevent overflow?	PO4
UNIT –5: MULTIRATE DIGITAL SIGNAL PROCESSING		
41	Define sampling rate conversion.	PO5
42	State some applications of DSP?	PO5
43	State the methods to convert the sampling rate.	PO5
44	What is multirate signal processing?	PO5
45	State the applications of multirate signal processing.	PO5
46	What is decimation?	PO5
47	What is interpolation?	PO5
48	Mention the types of sample/hold?	PO5
49	What is anti - aliasing filter?	PO5
50	What are the factors that influence selection of DSP's?	PO5
PART-B (10 Marks)		

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18ECE 323– DIGITAL SIGNAL PROCESSING

UNIT – 1: DISCRETE FOURIER TRANSFORM		
1	Explain the operations on signals. i. Shifting ii. Time reversal iii. Time scaling iv. Scalar multiplication **v.** Signal multiplier	PO1, PO2
2	Check whether the following signal is linear or nonlinear, time variant or time invariant i. $y(n) = 2x(n) + \frac{1}{x(n-1)}$ ii. $y(n) = nx^2(n)$	PO1, PO2
3	Check whether the following signal is causal, static or not i. $y(n) = ax(n)$ ii. $y(n) = x(n^2)$ PO1, PO2 iii. $y(n) = x(n) + x(n+1)$ **iv.** $y(n) = x^2(n)$	PO1, PO2
4	Determine the convolution of two sequences and justify with DFT and IDFT with the same result $x(n) = \{3,2,1,2\}; h(n) = \{1,\underset{\uparrow}{2},1,2\}$	PO1, PO2.
5	Compute the 8-point DFT of the sequence $x(n) = \{0.5,0.5,0.5,0.5,0,0,0,0\}$ using the radix-2 DIT FFTalgorithm	PO1, PO2.
UNIT – 2: IIR FILTER DESIGN		
6	Design a digital butterworth filter that satisfies the following constraint using bilinear transformation. Assume T = 1 sec. $$0.9 \le \left\|H(e^{j\omega})\right\| \le 1 \;\; for\; 0 \le \omega \le \frac{\pi}{2}$$ $$\left\|H(e^{j\omega})\right\| \le 2 \;\; for\; \frac{3\pi}{4} \le \omega \le \pi$$	PO1, PO2, PO3, PO5
7	Convert the analog filter to a digital filter whose system function is $$H(s) = \frac{1}{(s+2)^2 + (s+1)}$$ Use bilinear transformation.	PO1, PO2, PO3, PO5
8	1. A chebye low pass filter has to meet the following specifications i) Pass band gain of -1dB at$\Omega_p = 4\, rad/sec$ ii) Stop band alternations greater than or equal to 20 dB at $\Omega_s = 8\, rad/sec$ Determine the transfer function H(s) of the chebye filter to meet the above specifications	PO1, PO2, PO3, PO5
9	A third – order Butterworth low pass filter has the transfer function	PO1, PO2, PO3, PO5

	$$H(s) = \frac{1}{(s+1)(s^2+s+1)}$$ Design H(z) using impulse invariance technique.	
10	Obtain a cascade realization for the system function given below: H(z)= $\frac{1}{(z+1)(z^2+z+1)}$	PO1, PO2, PO3, PO5
UNIT3 – : FIR FILTER DESIGN		
11	The desired frequency response of a low pass filter is $$H_d(e^{jw}) = \begin{cases} e^{-j3w} & \frac{-3\pi}{4} \le w \le \frac{3\pi}{4} \\ 0 & elsewhere \end{cases}$$ Determine $H_d(e^{jw})$ for M=7 using a rectangular window.	PO1, PO2, PO3, PO5
12	The desired frequency response of a low pass filter is $$H_d(e^{jw}) = \begin{cases} e^{-jw} & \frac{-\pi}{4} \le w \le \frac{\pi}{4} \\ 0 & elsewhere \end{cases}$$ Determine $H_d(e^{jw})$ for M=11 using a Hamming window	PO1, PO2, PO3, PO5
13	The desired frequency response of a low pass filter is $$H_d(e^{jw}) = \begin{cases} 1; & \frac{-\pi}{2} \le w \le \frac{\pi}{2} \\ 0; & elsewhere \end{cases}$$ Determine $H_d(e^{jw})$ for M=7 using a Hanning window.	PO1, PO2, PO3, PO5
14	Determine the filter coefficients h(n), using frequency sampling method of frequency response given by, $$H_d(e^{j\omega}) = \begin{cases} e^{-\frac{j(N-1)\omega}{2}}, 0 \le \omega \le \frac{\pi}{2} \\ 0, \frac{\pi}{2} \le \omega \le \pi \end{cases}$$ For N = 7.	PO1, PO2, PO3, PO5
15	Realize the system function $H(z) = \frac{1}{2} + \frac{1}{3}z^{-1} + z^{-2} + \frac{1}{4}z^{-3} + z^{-4} + \frac{1}{3}z^{-5} + \frac{1}{2}z^{-6}$ using linear phase realization.	PO1, PO2, PO3, PO5
UNIT – 4: FINITE WORD LENGTH EFFECTS		
16	Explain the characteristics of a limit cycle oscillation with respect to the system described by the difference equation, $$y(n) = 0.95y(n-1) + x(n)$$ Determine the dead band of the filter with b = 4.	PO1, PO2, PO4
17	Consider the transfer function H(z) = $H_1(z).H_2(z)$ where, $$H_1(z) = \frac{1}{1-a_1z^{-1}} \text{ and } H_2(z) = \frac{1}{1-a_2z^{-1}}$$	PO1, PO2, PO4

	Find the output round off noise power. Assume $a_1 = 0.5\ and a_2 = 0.6$	
18	Convert the following numbers into decimal. **i)** $(1110.01)_2$ ii) $(11011.1110)_2$	PO1, PO2, PO4
19	Convert the following decimal numbers into binary **i)** $(20.675)_{10}$ ii) $(120.75)_{10}$	PO1, PO2, PO4
20	The input to the system y(n) = 0.999y(n-1) + x(n) is applied to an ADC. What is the power produced by the quantization noise at the output of the filter if the input is quantized to (a) 8 bits (b) 16 bits.	PO1, PO2, PO4
	UNIT –5: MULTIRATE DIGITAL SIGNAL PROCESSING	
21	With a neat sketch, explain the method for sampling rate conversion by a factor I/D.	PO1, PO4
22	Explain decimation by a factor D.	PO1, PO4
23	Explain interpolation by a factor M.	PO1, PO4
24	Sketch multirate signal processing with neat examples.	PO1, PO4
25	Illustrate cascading of sample rate converters.	PO1, PO4

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