ST. ANNE'S COLLEGE OF ENGINEERING AND TECHNOLOGY

(Approved by AICTE, New Delhi. Affiliated to Anna University, Chennai)

ANGUCHETTYPALAYAM, PANRUTI – 607 106.

DEPARTMENT OF ELECTRONICS AND COMMUNICATION ENGINEERING

QUESTION BANK

Regulation	**:**	**2021**
Class	**:**	**II Year, IV Semester ECE**
Subject Name	**:**	**Digital Signal Processing**

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UNIT I – DISCRETE FOURIER TRANSFORM

Review of signals and systems, concept of frequency in discrete-time signals, summary of analysis & synthesis equations for FT & DTFT, frequency domain sampling, Discrete Fourier transform (DFT) - deriving DFT from DTFT, properties of DFT - periodicity, symmetry, circular convolution. Linear filtering using DFT. Filtering long data sequences - overlap save and overlap add method. Fast computation of DFT - Radix2 Decimation-in-time (DIT) Fast Fourier transform (FFT), Decimation-in-frequency (DIF) Fast Fourier transform (FFT). Linear filtering using FFT.

PART – A

Q. No	Questions	BT Level	Competence
1	List the classifications of signals?	K1	Remembering
2	What is an LTI system?	K4	Analyzing
3	Compare energy and power signal of Discrete time signal.	K2	Understanding
4	Check the causality of y (n) = x(n^2).	K4 Ar	alyzing
5	Check and explain whether the system y(n) = e $^{x(n)}$ is linear or not?	K3	Applying
6	Define sampling theorem.	K1	Remembering
7	Write the condition for system stability.	K1	Remembering
8	Distinguish between linear convolution and circular convolution?	K2	Understanding
9	Obtain the circular convolution of $x(n)$ = {1,2,3,1} ; $h(n)$ = {4,3,2,1}	K3	Applying
10	What is zero padding? What are its uses?	K3	Applying
11	State about overlap save method.	K1	Remembering
12	State and prove periodicity property of DFT.	K4	Analyzing
13	Write the time shifting property of DFT?	K2	
Unde	rstanding 14 Express the Parseval's relation of DTFT.	K4	Analyzing

15	Describe about relation between Discrete Fourier Transform and Discrete time Fourier Transform	K2	Understanding
16	Find the 4-point DFT of the sequence x(n) = {1,1,-1,-1}.	K3	Applying
17	Find the DFT of the sequence x(n) = {1,2,3,0} using DIF algorithm.	K1	Remembering
18	How many multiplications and additions are required to compute N point DFT using radix-2 FFT algorithm?	K3	Applying
19	What is meant by in-place computation?	K3	Applying
20	Outline the concept of bit reversal in FFT?	K2	Understanding
21	Draw the basic butterfly diagram of radix-2 DIT FFT.	K2	Understanding
22	List the differences and similarities between DIT and DIF.	K4	Analyzing
23	Define twiddle factor and write the properties of twiddle factor.	K1	Remembering
24	What are the advantages of FFT algorithm over direct computation of DFT?	K4	Analyzing
	PART – B		
1	Determine whether the following signals are energy signals or power signals a. $x(n) = \left(\frac{1}{3}\right)^n u[n]$ (6) b. $x[n] = \begin{cases} n & for\ 0 \le n \le 5 \\ 10 - n & for\ 5 \le n \le 10 \\ 0 & otherwise \end{cases}$ (7)	K2	Understanding
2	(i) Consider an analog signal x(t) = 5cos200πt. a. Examine the minimum sampling rate to avoid sampling. (4) b. If sampling rate Fs =400Hz.What is the Discrete time signal after sampling? (5) (ii) write short notes on aliasing effects. (4)	K3	Applying
3	How will you determine the circular convolution of the following sequence x(n) = {1,1,2,1}, h(n) = {1,2,3,4} using DFT and IDFT method? (13)	K1	Remembering
4	Illustrate the 8-point DFT of a sequence $x(n) = \{\frac{1}{2}, \frac{1}{2}, \frac{1}{2}, \frac{1}{2}, 0,0,0,0\}$ (13)	K2	Understanding
5	Summarize the following properties of DFT: a. Periodicity (3) b. Time Reversal (3) c. Circular frequency shifting (3) d. Multiplication. (4)	K2	Understanding
6	Determine the output y(n) of a filter whose impulse response h(n) = {1,2} and input signal x(n) = {1, 2, -1, 2, 3, -2, -3, -1, 1, 1, 2, -1} using overlap save method and overlap add method. (13)	K2	Understanding

7	Construct the circular convolution of two finite duration sequences $x_1(n) = \{1, -1, -2, 3, -1\}$; $x_2(n) = \{1,2,3\}$. (13)	K4	Analyzing
8	(i) Prove that FFT algorithm helps in reducing the number of computations involved in DFT computation. (7) (ii) Discuss about overlap add method for convolution. (6)	K1	Remembering
9	Find the 8-point DFT of a given sequence $(n) = \{1,2,2,1,1,2,2,1\}$ using DIF-FFT algorithm. (13)	K1	Remembering
10	(i) Write the steps for radix-2 DIT FFT algorithm. (7) (ii) Solve the 8-point of a given sequence $x(n) = n + 1$ using DITFFT algorithm. (6)	K3	Applying
11	Calculate IDFT of the sequence X(K) = {7, −0.707− j0.707, −j ,0.707− 0.707, 1, 0.707+ j0.707, j, −0.707+ j0.707 using DIT algorithm. (13)	K4	Analyzing
12	Apply DIT algorithm to compute DFT of the given sequence $x(n) = \{1, 1, 1, 1, 0, 0, 0, 0\}$. (13)	K3	Applying
13	Compute the DFT of the sequence $x(n) = cos\frac{n\pi}{2}$ where $N = 4$ using DIF FFT algorithm. (13)	K1	Remembering
14	(i) Analyze the N – point DFT of the sequence $x(n) = \delta(n)$ (3) (ii) Compute 8 – point DFT of the sequence x(n) = {0, 1, 2, 3, 4,5, 6, 7} using radix – 2 DIT algorithm. (10)	BTL4	Analyzing
15	Examine the 8-point DFT of the sequence $x(n) = \{2,2,2,2,1,1,1,1\}$ using decimation in time FFT algorithm. (13)	K4	Analyzing
16	Find the DFT for the sequence {1,2,3,4,4,3,2,1} using Radix-2 decimation in frequency algorithm. (13)	K3	Applying
17	(i) State and analyse convolution property of DFT? (7) (ii) Find the 4-point inverse DFT of X(k)={10,-2+2j,-2,-2-2j} using DIT-FFT algorithm. (6)	K3	Applying
	PART – C		
1	Using linear convolution construct $y(n) = x(n) * h(n)$ for the sequence $h(n) = \{1,1,1\}$ and input signal $x(n) = \{3,-1,0,1,3,2,0,1,2,1\}$ using overlap save method and overlap add method. (15)	K3	Applying
2	Examine the properties linearity, causality, time invariance and dynamicity of the given systems. a. $y(n) = x(2n)$ (5) b. $y(n) = x(n)x(n-1)$ (5) c. $y(n) = \log\lvert x(n)\rvert$ (5)	K4	Analyzing
3	Determine the DFT of the sequence $x[n] = \{\underset{\uparrow}{1},2,3,2,1,0\}$ (15)	K3	Applying
4	Formulate the 8-point DFT using FFT	K1	Remembering

	$x[n] = \begin{cases} 1 & for \quad 0 \le n \le 6 \\ 0 & otherwish \end{cases}$ (15)		
5	Calculate the 8 point for the given sequence using DIT FFT algorithm $x[n] = \begin{cases} 1 & for \quad -3 \le n \le 3 \\ 0 & otherwise \end{cases}$ (15)	K2	Understanding

UNIT II – INFINITE IMPULSE RESPONSE FILTERS			
Characteristics of practical frequency selective filters. Characteristics of commonly used analog filters - Butterworth filters, Chebyshev filters. Design of IIR filters from analog filters (LPF, HPF, BPF, BRF) - Approximation of derivatives, Impulse invariance method, Bilinear transformation. Frequency transformation in the analog domain. Structure of IIR filter - Direct form I, Direct form II, Cascade, Parallel realizations.			
PART – A			
Q.No	**Questions**	**BT Level**	**Competence**
1	What are the different types of structures for realization of IIR systems?	K 2	Understanding
2	Distinguish between recursive realization and non-recursive realization?	K2	Understanding
	Convert the given analog transfer function $H(s) = \frac{1}{s+a}$ into digital by impulse invariant method?	K 3	Applying
4	List the different types of filters based on frequency response.	K 1	Remembering
5	Write the properties of Butterworth filter.	K 1	Remembering
6	Justify why impulse invariant method is not preferred in the design of IIR filter other than LPF?	TL 4	Analyzing
7	Identify the expression for location of poles of normalized Butterworth filter.	K3	Applying
8	What is the relation between digital and analog frequency in bilinear transformation?	K 4	Analyzing
9	Why do we go for analog approximation to design a digital filter?	K 4	Analyzing
10	Outline the steps in design of a digital filter from analog filters.	K 2	Understanding
11	Mention the requirements for the digital filter to be stable and causal.	K1	Remembering
12	Write the need for prewarping.	K1	Remembering
13	Give the properties of bilinear transformation.	K 1	Remembering
14	Use the backward difference for the derivative to convert analog LPF with system function $\mathrm{H}(s) = \frac{1}{s+3}$	BTL 3	Applying
15	Compare Butterworth with Chebyshev filters.	K 2	Understanding
16	Justify why the Butterworth response is called a maximally flat response.	K 4	Analyzing
17	List the parameters that can be obtained from Chebyshev filter specification?	BTL 3	Applying
18	What is the advantage of direct form II realization when compared to direct form I realization?	K 1	Remembering

19	How to represent the frequency warping in IIR filter?	K 4	Analyzing
20	Write the expression for location of poles of normalized Butterworth filter.	BTL 4	Analyzing
21	Sketch the frequency response of an odd and even order Chebyshev low pass filters.	K 3	Applying
22	Compare digital and analog filters.	K2	Understanding
23	Determine the order of the analog butterworth filter that has a -2db pass band attenuation at a frequency of 20rad/sec and at least -10db stop band attenuation at 30 rad/sec.	BTL 3	Applying
24	Write the various frequency translations in analog domain.	K 2	Understanding
	PART – B		
1	Enumerate the steps for IIR filter design by impulse invariance with example. (13)	K1	Remembering
2	Obtain the direct form I ,direct form II and cascade form realization of the following system functions $y[n] = 0.1\ y[n-1] + 0.2\ y[n-2] = 3\ x[n] + 3.6\ x[n-1] + 0.6\ x[n-2]$ (13)	K1	Remembering
3	Explain the bilinear transform method of IIR filter design. What is wrapping effect? Explain the poles and zeros mapping procedure clearly. (13)	K 4	Analyzing
4	Summarize the steps in the design of IIR filter using bilinear transformation for any one type of filter? (13)	K 2	Understanding
5	Given the specification $a_p = 3dB$; $a_s = 16dB$; $f_p = 1KHz$; $f_s = 2KHz$. Solve for H(s) using Chebyshev approximation. (13)	K3	Applying
6	For the given specifications, design an analog Butterworth filter $$0.9 \le \lvert H(j\Omega)\rvert \le 1 \quad for\ \ 0 \le \Omega \le 0.2\pi$$ $$\lvert H(j\Omega)\rvert \le 0.2 \quad for\ \ 0.4\pi \le \Omega \le \pi$$ (13)	K4	Analyzing
7	Convert the analog filter into a digital filter whose system function is $H(s) = \frac{s+0.2}{(s+0.2)^2+9}$ using bilinear transformation technique. The digital filter should have a resonant frequency of $\omega_r = \pi/4$. (13)	K3	Applying
8	Using the bilinear transformation design a high pass filter 3 dB monotonic in pass band with cut off frequency of 1000Hz and down 10 dB at 350Hz. The sampling frequency is 5000Hz. (13)	K4	Analyzing
9	Analyze a digital Chebyshev filter to satisfy the constraints $$0.707 \le \lvert H(e^{j\omega})\rvert \le 1 \qquad 0 \le \omega \le 0.2\pi$$ $$\lvert H(e^{j\omega})\rvert \le 0.1 \quad 0.5\pi \le \omega \le \pi$$ using Bilinear transformation and assuming $T = 1sec$. (13)	K4	Analyzing
10	Design a butterworth digital filter using bilinear transformation to satisfy the constraints $$0.89 \le \lvert H(e^{j\omega})\rvert \le 1 \qquad 0 \le \omega \le 0.2\pi$$	K3	Applying

	$\lvert H(e^{j\omega})\rvert \leq 0.18 \quad 0.3\pi \leq \omega \leq \pi$ using Bilinear transformation and assuming $T = 1sec$. (13)		
11	Explain the conversion of analog BPF into digital IIR filter using backward difference for the derivative $H_a(s) = \frac{1}{(s+0.2)^2+8}$ (13)	BTL 2	Understanding
12	(i) for the given specifications Ap = 3dB, As=15 dB, Ωp =500rad/sec and Ωs=1000rad/sec. Design a high pass filter. (6) (ii) Convert the following analog transfer function into digital using impulse invariant technique with sampling period T=1sec. H(s)=[s+1]/[(s+3)(s+5)] (7)	K3	Applying
13	Apply Bilinear transformation to determine (z) for Butterworth filter satisfying the following specifications. $0.8 \leq \lvert H(e^{j\omega})\rvert \leq 1 \quad 0 \leq \omega \leq \pi/4$ $\lvert H(e^{j\omega})\rvert \leq 0.2 \quad \pi/2 \leq \omega \leq \pi$ (13)	K3	Applying
14	Find the system function H(z) of the Chebyshevs low pass digital filter with the specifications ap= $1dB$ *ripple in the pass band* $0 \leq \omega \leq 0.2\pi$; as= $15dB$ *ripple in the stop band* $0.3\pi \leq \omega \leq \pi$; using bilinear transformation assume $T = 1sec$. (13)	K1	Remembering
15	An Analog filter has a transfer function $$H(s) = \frac{10}{s^2 + 7s + 10}$$ Design a digital filter equivalent to this using impulse invariant method for $T = 0.2sec$. (13)	BTK 2	Understanding
16	Summarize the design steps followed by discrete time IIR filter from analog filter. (13)	BTK 2	Understanding
17	(i) Convert the analog filter with system function $H_a(s)=[s+0.1]/[(s+0.1)^2+9]$ into a digital IIR filter by means of the impulsive invariance method. (7) (ii) Draw the direct form I structures for the given difference equation y(n)=2y(n-1)+3y(n-2)+x(n)+2x(n-1)+4x(n+2). (6)	BTK4	Analyzing
Part C			
1	Realize the direct form I, direct form II, cascade and parallel form realization of LTI system governed by the equation: $y(n) = -\frac{3}{8} + y(n-1) + \frac{3}{32}y(n-2) + \frac{1}{64}y(n-3) + x(n) +$ $3\,x(n-1) + 2\,x(n-2)$ (15) (15)	BTK4	Analyzing
2	Obtain a digital Butterworth filter with the following specifications : $0.707 \leq \lvert H(e^{j\omega})\rvert \leq 1 \quad 0 \leq \omega \leq 0.5\pi$ $\lvert H(e^{j\omega})\rvert \leq 0.2 \quad 0.75\pi \leq \omega \leq \pi$ using bilinear transformation determine system function $H(Z)$ assuming	BTK3	Applying

	$T = 1sec.$ (15)		
3	For the given specifications, design a low pass digital filter using impulse invariance method satisfying the constraints. Assume T=1sec $0.9 \le \ |H(e^{j\omega})| \ \le 1 \qquad 0 \le \omega \le 0.25\pi$ $|H(e^{j\omega})| \le 0.24 \qquad 0.5\pi \le \omega \le \pi$ (15)	BTK3	Applying
4	For the given specifications, design an digital Butterworth filter using impulse invariance method satisfying the constraints. Assume T=1sec $0.8 \le \ |H(e^{j\omega})| \ \le 1 \qquad 0 \le \omega \le 0.2\pi$ $|H(e^{j\omega})| \le 0.2 \qquad 0.6\pi \le \omega \le \pi$ (15)	BTK1	Remembering
5	A digital low pass IIR filter is to be designed with butterworth approximation using bilinear transformation technique. Find the order of filter to meet the following specifications. (i) Passband magnitude is constant within 1dB for frequencies below 0.2 π (ii) Stopband attenuation is greater than 15 dB for frequencies between 0.3 π to π. (15)	BTK 2	Understanding

UNIT III - FINITE IMPULSE RESPONSE FILTERS

Design of FIR filters - symmetric and Anti-symmetric FIR filters - design of linear phase FIR filters using Fourier series method - FIR filter design using windows (Rectangular, Hamming and Hanning window), Frequency sampling method. FIR filter structures - linear phase structure, direct form realizations

PART – A

Q.No	Questions	BT Level	Competence
1	Name the different types of filters based on frequency response.	BTK 1	Remembering
2	Summarize the advantages of FIR filters.	BTL 2	Understanding
3	Mention the necessary and sufficient condition for the linear phase characteristic of an FIR filter.	BTK 4	Analyzing
4	Illustrate the condition for the impulse response of FIR filter to satisfy for constant phase delay and for only constant group delay.	BTK 3	Applying
5	What is Window? Why it is necessary?	BTK 1	Remembering
6	Classify the properties of FIR filter.	BTL 3	Applying
7	What is the impulse response condition for a FIR filter to have linear phase characteristics?	BTK 2	Understanding
8	Infer the advantages and disadvantages of window.	BTK 4	Analyzing
9	Write about phase delay and group delay	BTL 2	Understanding

10	Define Gibbs phenomenon.	TK 1	Remembering
11	Write the need for employing window technique for FIR filter design.	BTLK2	Understanding
12	Points out the desirable characteristics of FIR filter using windows.	BTK 4	Analyzing
13	Write the general expression of hanning, hamming and rectangular window.	BTK 3	Applying
14	Compare Hamming and Hanning window.	BTLK4	Analyzing
15	List the characteristics features of Rectangular window.	BTK 1	Remembering
16	What are the desirable characteristics of window?	BTK 4	Analyzing
17	Justify that frequency-sampling method is suitable for narrow band filters.	BTL 4	Analyzing
18	Draw the direct form realization of FIR filter.	BTK 3	Applying
19	Why FIR filters are always stable?	BTL 1	Remembering
20	Express why cascade realization is preferred in FIR filters.	BTK 2	Understanding
21	Write the definition for linear phase response of a filter.	BTK 3	Applying
22	Illustrate the various methods of designing FIR filters.	BTL 3	Applying
23	Differentiate symmetric FIR filters and antisymmetric FIR filters.	BTK 2	Understanding
24	What are the Antisymmetric FIR filters? What are its applications?	BTL 1	Remembering
	PART – B		
1	Prove that an FIR filter has linear phase if the unit sample response satisfies the condition $h(n) = h(N-1-n)$. (13)	BTK 1	Remembering
2	Determine the frequency response of linear phase FIR filter when impulse response is symmetrical and N is even. (13)	BTK3	Applying
3	Obtain the frequency response of linear phase FIR filter when impulse response is antisymmetrical and N is odd. (13)	BTK 2	Understanding
4	Find the FIR LPF with cut-off frequency of 1KHz and sampling frequency of 4KHz with 11 samples using Fourier series method. (13)	BTK 4	Analyzing
5	Solve and design a FIR filter with the following desired specifications using hanning window with N=5. $Hd(e^{j\omega}) = \begin{cases} e^{-j2\omega} & for \quad -\frac{\pi}{4} \le \lvert\omega\rvert \le \pi \\ 0 & for \quad -\frac{\pi}{4} \le \lvert\omega\rvert \le \frac{\pi}{4} \end{cases}$ (13)	BTK3	Applying
6	Obtain an ideal low pass filter with a frequency response $Hd(e^{j\omega}) = \begin{cases} 1 & for \quad -\frac{\pi}{2} \le \omega \le \frac{\pi}{2} \\ 0 & for \quad \frac{\pi}{2} \le \lvert\omega\rvert \le \pi \end{cases}$ Find the values of $h(n)$ for $N = 11$. Find $H(z)$. (13)	BTK 4	Analyzing

7	Using a rectangular window technique, Illustrate a low pass filter with pass band gain of unity, cut-off frequency of 1000 Hz and working at a sampling frequency of 5 KHz. The length of the impulse response should be 7. (13)	BTK 2	Understanding
8	By Choosing N = 7, Examine a filter with $Hd(\omega) = \begin{cases} e^{-j3\omega} & ; for|\omega| \leq \frac{\pi}{4} \\ 0 & ; \frac{\pi}{4} \leq |\omega| \leq \pi \end{cases}$ Using Hamming window. (13)	BTK 1	Remembering
9	Design a length 5 FIR Band reject filter with a lower cutoff frequency of 2KHz and upper cutoff frequency 2.4KHz and the sampling rate of 8000Hz using Hamming window. (13)	BTK 4	Analyzing
10	How to design a FIR band stop filter to reject frequencies in the range 1.2 to 1.8 rad/sec using hamming window, with length $N = 6$. (13)	BTK 1	Remembering
11	Determine the filter coefficients for an FIR filter approximating the ideal frequency response having N=7 using Hamming window. $Hd(\omega) = \begin{cases} e^{-j\alpha\omega} & ; for|\omega| \leq \frac{\pi}{6} \\ 0 & ; \frac{\pi}{6} \leq |\omega| \leq \pi \end{cases}$ (13)	BTK 2	Understanding
12	Describe the procedure of designing FIR filters by windows. (13)	TK 1	Remembering
13	Determine the Lowpass FIR filter using frequency sampling technique having cutoff frequency of π/2 rad/sample. The filter should have linear phase and length of 7. $H_d(e^{j\omega}) = e^{-j3\omega} \quad for\ 0 \leq \omega \leq \frac{\pi}{2}$ (13)	BTK3	Applying
14	Obtain the direct form and cascade form realizations of the following system equation y(n) = 0.1 y(n-1) + 0.2 y(n-2) + 3x(n) +3.6x(n-1) + 0.6 x(n-2) (13)	BTK 2	Understanding
15	Briefly explain the procedure for design of linear phase FIR filter using frequency sampling technique or discuss the design procedure of FIR filter using frequency sampling method. (13)	BTK 4	Analyzing
16	Illustrate the direct form I & II structure of the system function $H(z) = 1 + 2z^{-1} - 3z^{-2} + 4z^{-3} + 5z^{-4}$ (13)	BTK3	Applying
17	What is the need for realization of FIR filters? Describe the various different types of linear phase FIR structures. (13)	BTK 1	Remembering
	PART – C		

1	(i) Draw the linear phase structure of FIR filter with the following impulse response $h(n) = \delta(n) + \frac{1}{2}\delta(n-1) - \frac{1}{4}\delta(n-2) + \delta(n-4) + \frac{1}{2}\delta(n-3)$. (8) (ii) Explain the steps involved by the general process of designing a digital filter. (7)	BTK1	Remembering
2	Prove that an FIR filter has linear phase if the unit sample response satisfies the condition $h(n) = h(N-1-n)$. Also discuss symmetric and antisymmetric case of FIR filter when N is odd. (15)	BTK2	Understanding
3	Design an ideal high pass filter using hanning window with a frequency response $H_d(e^{j\omega}) = \begin{cases} 1 & for\ \frac{\pi}{4} \le \lvert\omega\rvert \le \pi \\ 0 & for\ \lvert\omega\rvert \le \frac{\pi}{4} \end{cases}$ Assume $N = 11$. (15)	BTL4	Analyzing
4	Determine the coefficients of a linear phase FIR filter of length $M = 15$ which has a symmetric unit sample response and a frequency response that satisfies the conditions. $H_r\left(\frac{2\pi k}{15}\right) = \left\{\begin{matrix} 1 & k = 0,1,2,3 \\ 0 & k = 4,5,6,7 \end{matrix}\right\}$ (15)	BTL3	Applying
5	Examine the coefficients of a linear phase FIR filter of length $M = 15$ which has a symmetric unit sample response and a frequency response that satisfies the conditions. $H_r\left(\frac{2\pi k}{15}\right) = \left\{\begin{matrix} 1 & k = 0,1,2,3 \\ 0.4 & k = 4 \\ 0 & k = 5,6,7 \end{matrix}\right\}$ (15)	BTK4	Analyzing

UNIT - IV FINITE WORD LENGTH EFFECTS			
Fixed point and floating point number representation - ADC - quantization - truncation and rounding - quantization noise - input / output quantization - coefficient quantization error - product quantization error - overflow error - limit cycle oscillations due to product quantization and summation - scaling to prevent overflow.			
PART – A			
Q.No	**Questions**	**BT Level**	**Competence**
1	List the different types of number representations in digital systems.	BTK1	Remembering
2	Define Finite word length effect.	BTL1	Remembering

3	Point out the some of the finite word length effects in digital filters.	BTK4	Analyzing
4	Mention the different formats of fixed point representation.	BTL3	Applying
5	State the advantages of floating-point representation.	BTL1	Remembering
6	Express the fraction 7/8 and -7/8 in sign magnitude, 1's complement and 2's complement.	BTK2	Understanding
7	Compare the fixed and floating point number representation.	BTL4	Analyzing
8	Illustrate what are the errors occurred due to finite word length registers in digital filter.	BTK3	Applying
9	List the two types of quantization employed in digital system.	BTL4	Analyzing
10	Why rounding is preferred over truncation in relating digital filter?	BTK3	Applying
11	What is quantization?	BTL1	Remembering
12	What is quantization error?	BTK1	Remembering
13	What is the effect of quantization on pole location?	BTL4	Analyzing
14	How would you relate the steady state noise power due to quantization to the b bits representing the binary sequence?	BTK2	Understanding
15	What do you understand by input quantization error?	BTL3	Applying
16	Interpret the meaning of coefficient quantization error.	BTK2	Understanding
17	Define product quantization error.	BTL2	Understanding
18	Write about the product round-off noise.	BTK4	Analyzing
19	Summarize about limit cycles.	BTL3	Applying
20	Classify the two kinds of limit cycle behavior in DSP?	BTK3	Applying
21	Infer the dead band of the filter.	BTK4	Analyzing
22	How overflow limit cycles can be eliminated?	BTK2	Understanding
23	Summarize zero input limit cycle oscillations.	BTLK2	Understanding
24	Which realization is less sensitive to the process of quantization?	BTK1	Remembering
PART – B			
1	Explain in detail about finite word length effects in digital filters. (13)	BTK1	Remembering
2	Realize the first order transfer $\mathrm{H(z)} = \frac{1}{1-\mathrm{a}z^{-1}}$ and draw its quantization noise model. Find the steady state noise power due to product round off. (13)	BTK4	Analyzing
3	For the given transfer function $H(z) = H_1(z).H_2(z)$, Where $H_1(z) = \frac{1}{1-0.9z^{-1}}$ *and* $H_2(z) = \frac{1}{1-0.8z^{-1}}$. Solve the output round off noise power. Identify the value if B=3bits. (13)	BTK3	Applying

4	Consider the transfer function $H(z) = H_1(z).H_2(z)$, Where $H_1(z) = \frac{1}{1-0.5z^{-1}}$ $and\ H_2(z) = \frac{1}{1-0.6z^{-1}}$. Estimate the output round off noise power. (13)	BTK2	Understanding
5	The output signal of an ADC is passed through a first order lowpass filter with transfer function given by $H(z) = \frac{(1-a)z}{(z-a)} for\ 0 < a < 1$. Calculate the steady state output noise power due to quantization at the output of the digital filter. (13)	BTK3	Applying
6	For the second order IIR filter, the system function is, $$H(Z) = \frac{1}{(1-0.5z^{-1})(1-0.45z^{-1})}$$ Examine the effect of shift in pole location with 3 bits coefficient representation in direct and cascade forms. (13)	BTK1	Remembering
7	(i) Write a note on Limit Cycle oscillation. (3) (ii) Explain the characteristics of limit cycle oscillations to the system described by the difference equation y(n)= 0.95y(n-10+x(n); x(n)=0 and y(n-1)=13. Determine the dead band of the system. (10)	BTK1	Remembering
8	Find the characteristics of a limit cycle oscillation with respect to the system described by the equation $y(n) = 0.95y(n-1) + x(n)$. Estimate the dead band of the filter. (13)	BTK2	Understanding
9	(i) Explain in detail the input quantization error and coefficient quantization error and its effect on digital filter design, with an example. (6) (ii) Illustrate quantization noise. Summarize the expression for quantization noise power at the output ADC. (7)	BTK2	Understanding
10	For a second order IIR filter $\mathrm{H(z)} = \frac{1}{(1-0.9\mathrm{z}^{-1})(1-0.8\mathrm{z}^{-1})}$ find the effect of shift in pole location with 3-bit coefficient presentation in direct from and cascade form. (13)	BTK2	Understanding
11	Explain the detail the 3 types of quantization error that occur due to the finite word length of register. (13)	BTK3	Applying
12	An IIR causal filter has the system function $H(z) = \frac{z}{z-0.97}$ Assume that the input signal is zero valued and the computed output signal values are rounded to one decimal place. Show that under those stated conditions, the filter output exhibits dead band effect. What is the dead band range? (13)	BTK3	Applying
13	Analyze the behavior of limit cycle oscillation with respect to the system described by the following equation $y(n) = 0.87\ y(n-1) + x(n)$.Determine the dead band of the system when $x(n) = 0$ and $y(-1) = 0.61$. Assume that the product is quantized to 4 bits by rounding. (13)	BTK4	Analyzing

14	In the IIR system given below the products are rounded to 4 bits (including sign bits). The system function is $$H(Z) = \frac{1}{(1 - 0.35z^{-1})(1 - 0.62z^{-1})}$$ Find the output round off noise power in (a) direct form realization and (b) cascade form realization. (13)	BTK1	Remembering
15	The input to the system $y(n) = 0.999y(n-1) + x(n)$ is applied to an ADC. Calculate the power produced by the quantization noise at the output of the filter if the input is quantized to 8 & 16 bits. (13)	BTK4	Analyzing
16	(i) Express decimal fraction 4.5, 6.5 and 1.5 in binary floating point format. (7) (ii) Compare fixed and floating point representation. (6)	BTK4	Analyzing
17	(i) What are the errors occurred during resulting from truncation and rounding? Explain. (10) (ii) Describe the various formats of the fixed point representation of binary numbers. (3)	BTK1	Remembering
PART –C			
1	The output of an A/D converter is applied to a digital filter with the system function; $H(z) = \frac{0.5z}{z-0.5}$. Determine the output noise power from the digital filter when the input signal is quantized to have 8 bits. (15)	BTK4	Analyzing
2	The input to the system $y(n) = 0.89y(n-1) + x(n)$ 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. (15)	BTK1	Remembering
3	Examine the effect of coefficient quantization on pole locations of the second order IIR system realized in Direct Form I and in Cascade. Assume word length of 4 bits through truncation. The transfer function of the realization is given as follows. $H(z) = \frac{1}{1-0.9z^{-1}+0.2z^{-2}}$ (15)	BTK3	Applying
4	A cascaded realization of the two first order digital filter is shown below. The system functions of the individual sections are $H_1(z) = \frac{1}{1-0.5z^{-1}}$ $and\ H_2(z) = \frac{1}{1-0.4z^{-1}}$. Draw the product quantization noise model of the system and determine the overall output noise power if b=3 bits (excluding sign bit). (15)	BTK4	Analyzing
5	A causal filter is defined by the difference equation $y(n) = x(n) - 0.9y(n-1)$ and $$x(n) = \begin{cases} 12, & n = 0 \\ 0, & n \neq 0 \end{cases}$$ The unit sample response h(n) is computed such that the computed values are rounded to one decimal place. Express that the filter exhibits dead band effect. Determine the dead band range. (15)	BTL2	Understanding

UNIT V - INTRODUCTION TO DIGITAL SIGNAL PROCESSORS			
DSP functionalities - circular buffering – DSP architecture – Fixed and Floating point architecture principles – Programming – Application examples.			
PART – A			
Q.No	**Questions**	**BT Level**	**Competence**
1	List the applications of DSP.	BTK 1	Remembering
2	What is the role of the pipeline operation in a Digital Signal Processor?	BTK 1	Remembering
3	Mention the buses used in digital signal processors?	BTK 1	Remembering
4	Define circular buffering.	BTK1	Remembering
5	Brief the features of MAC unit.	BTK 1	Remembering
6	Point out the classification of instruction set in Digital Signal Processor?	BTK1	Remembering
7	Summarize the on-chip peripherals in ‘C5x’.	BTK 2	Understanding
8	Outline the different phases in pipelining process.	BTK 2	Understanding
9	Compare the difference between Von Neumann architecture & Harvard architecture.	BTK 2	Understanding
10	Enumerate the advantages and disadvantages of VLIW architecture.	BTK2	Understanding
11	Categorize the addressing modes of TMS320C54XX processor.	BTK 2	Understanding
12	Identify the important elements of program controller?	BTK2	Understanding
13	Choose the features to select digital signal processor.	BTK 3	Applying
14	Illustrate the need of accumulator.	BTK3	Applying
15	Identify any two logical instruction of DS processor.	BTK3	Applying
16	Specify the features of a Digital Signal Processor over Microcontroller?	BTK 3	Applying
17	List out the major functional units present in TMS32050.	BTK 3	Applying
18	Classify the types of special purpose DSP processors.	BTK 3	Applying
19	Write a program to add to numbers in DSP Processor.	BTK4	Analyzing
20	Distinguish between fixed and floating point arithmetic?	BTK 4	Analyzing
21	How the DS Processor pipeline differs from micro controller.	BTK 4	Analyzing
22	Analyze the various addressing modes of TMS32050.	BTK 4	Analyzing
23	Examine the arithmetic instructions of C5x processor.	BTK 4	Analyzing
24	Point out some example for floating point DSPs.	BTK 4	Analyzing
PART – B			
1	List and explain the various types of addressing modes of digital signal processor with suitable example. (13)	BTK 1	Remembering
2	(i) What are the factors used to select a Digital Signal processor? (5) (ii) Write in detail about few applications of programmable digital signal processor. (8)	BTK 1	Remembering

3	Summarize a detailed note about arithmetic instructions with necessary syntax. (13)	BTK 1	Remembering
4	(i) Name the different types of MAC functions in Digital Signal processor. (3) (ii) Describe about VLIW architecture and its advantages and disadvantages. (10)	BTK 1	Remembering
5	Explain the classification of instructions of TMS320C5X. (13)	BTK 1	Remembering
6	(i) Outline about different stages of pipelining and specify its importance. (6) (ii) Mention the features of Von Neumann and Harvard architectures. (7)	BTK 2	Understanding
7	With neat sketch explain the architecture of TMS320C54x processor. (13)	BTK 2	Understanding
8	(i) Specify the role of accumulator in TMS320C54x processor. (5) (ii) Explain the functionality of barrel shifter in TMS320C54x processor with neat sketch. (8)	BTK 2	Understanding
9	Draw and explain the basic architecture of fixed point processors TMS320C5X. (13)	BTK 2	Understanding
10	(i) Identify the need of MAC and its application in PDSP's. (8) (ii) List the instruction set of Digital Signal processor. (5)	BTL 3	Applying
11	(i) Examine the applications of PDSP's. (5) (ii) Write a simple program to generate square and saw tooth wav form. (8)	BTL 3	Applying
12	Illustrate in detail about Arithmetic Logic Unit with neat functional diagram of TMS320C54x. (13)	BTL 3	Applying
13	Discuss about the principle of operation of floating point architecture with necessary diagram. (13)	BTL 3	Applying
14	Draw and explain the bus structure and CPU of TMS320C50x. (13)	BTK4	Analyzing
15	Enumerate the various on chip peripherals in TMS320C54x processor. (13)	BTL 4	Analyzing
16	Examine about CSSU and Exponent encoder of TMS320C54x. (13)	BTK4	Analyzing
17	Write an assembly language program to generate a triangular and saw tooth waveform in TMS320C5X. (13)	BTL 4	Analyzing
PART C			
1	Explain in detail about the architecture of TMS 320C5416 Digital Signal Processor with neat sketches. (15)	BTL 1	Remembering
2	Discuss in detail with syntax for any six instructions used in TMS320C50X processors. (15)	BTL 1	Remembering
3	With neat functional diagram elaborate the following features of TMS320C54X : (i) Multiplier / Adder Unit (8)	BTK 2	Understanding

	(ii) Barrel Shifter (7)		
4	Write an assembly language program to perform linear and circular convolution through MAC operation in TMS320C5x. (15)	BTL 3	Applying
5	Obtain a suitable algorithm and illustrate the memory access used to calculate the value of the function $Y= A*X_1+B*X_2+C*X_3$. Write the necessary assembly code in TMS320C50 processor. (15)	BTL 4	Analyzing