Andhra Loyola Institute of Engineering and Technology
Department of Electronics and Communication Engineering

Year/Sem: III/II **Subject: Digital Signal Processing** **Academic Year: 2021-2022** **Faculty: Dr. T Lakshmi Narayana**

Question Bank

UNIT-1: Introduction to Digital Signal Processing: Discrete time signals & sequences, Classification of Discrete time systems, stability of LTI systems, Invertability, Response of LTI systems to arbitrary inputs. Solution of Linear constant coefficient difference equations. Frequency domain representation of discrete time signals and systems. Review of Z-transforms, solution of difference equations using Z-transforms, System function.

S. NO	Question	Marks	BT Level
1.	(a)Find the solution to the following linear constant coefficient difference equation with initial conditions y(-1)=1, y(-2)=0. $y(n) - 0.25y(n-2) = x(n)$	7	L2
	(b)Derive the relationship between impulse response and frequency response of a discrete time system.	3	L2
2	(a)Find the impulse response $h[n]$ of the system described by the difference Equation $8y(n) + 6y(n-1) = x[n]$	5	L2
	(b)What are the conditions for stability and causality of an LTI system? Explain.	5	L2
3	(a)State and prove the properties of convolution.	6	L2
	(b)State and explain the transfer function of an LTI system	4	L1
4	(a)Using the z-transform, find the total solution to the following difference equation with initial conditions, for discrete time $n \geq 0$. $5y(n+2) - 3y(n+1) + (y)n = (0.8)^n u(n)$, $y(0) = -1$, $y(1) = 10$	7	L2
	(b)Define the terms : linearity, time invariance and causality for a discrete time system	3	L1
5	(a)Determine whether each of the following systems defined below is Casual, linear, dynamic, time invariant (i) $y(n) = x(-n-2)$ (ii) $y(n) = \cosh[nx(n) + x(n-1)]$ (iii) $y(n) = \log_{10}[\{x(n)\}]$,	5	L2
	(b) Find the convolution of the signals $x(n)=a^n u(n)$, $h(n)=b^n u(n)$	4	L2
6	(a)Determine and sketch the magnitude and phase response of the following systems	7	L2
	(b) State and prove final-value theorem of z-transform.	3	L2
7	(a)Explain in detail the classification of discrete-time systems.	4	L2
	(b)Explain and calculate the frequency response of discrete time system.	6	L2
8	(a)Discuss the frequency domain representation of linear time- invariant systems	5	L2
	(b) Determine the Inverse Z-Transform of: $X(Z)=1/(1-Z^{-1})(1-Z^{-1})^2$.	5	L2
9	(a) Determine the stability for the following systems: i) $h(n) = 2^n u(n)$ ii) $h(n) = 5^n u(3-n)$ iii) $h(n) = e^{-6\|n\|}$.	7	L2
	(b) Discuss the stability of the systems described by the impulse response below: i. $h(n) = 2^{-n}(n)$. ii. $h(n) = 0.5^n u n - 0.5^n (4-n)$.	3	L2

10	(a) Find the step response of the system described by the difference Equation $8y(n) + 6y(n-1) = x[n]$	6	L2
	(b)What are the advantages of DSP over ASP? Explain	4	L2

UNIT II: DISCRETE FOURIER SERIES & FOURIER TRANSFORMS: Properties of discrete Fourier series, DFS representation of periodic sequences, Discrete Fourier transforms: Properties of DFT, linear filtering methods based on DFT, Fast Fourier transforms (FFT) - Radix-2 decimation in time and decimation in frequency FFT Algorithms, Inverse FFT

S. No.	Question	Marks	BT Level
1	(a) Compute the DFT of the three point sequence x(n) = { 2, 1, 2 }. Using the same sequence, compute the 6 point DFT and compare the two DFTs.	5	L2
	(b) Give the steps involved in implementing Radix –2, DIT FFT algorithm.	5	L2
2	(a)Prove that the convolution in time-domain leads to multiplication in frequency domain for discrete time signals.	4	L1
	(b)Compute the 4-point DFT using Radix–2 DIT FFT algorithm.	6	L2
3	(a)Compute the circular convolution of the sequences $x_1(n) = \{ 1, 2, 0, 1 \}$ and $x_2(n) = \{ 2, 2, 1, 1 \}$	4	L2
	(b)Compute the DFT of the three point sequence using the same sequence, compute the 6 point DFT and compare the two DFTs.	6	L2
4	(a)Explain Properties of Discrete time fourier series.	5	L2
	(b) Compute the DFT of the given sequence $x(n)$ using DIT FFT algorithm. $x[n] = \{1, -1, 1, -1, 1, -1, 1, -1\}$ Show the intermediate result on the flow graph.	5	L3
5	(a) Compute the DFT of a sequence $x(n)$ ={1/2,1/2,1/2,1/2,1,1,1,1} using DIF-FFT.	5	L2
	(b) Find the inverse DFT of X(k) = {1,2,3,4}.	5	L2
6	(a) Establish the relation between DFT and Z-transform.	5	L2
	(b)Find the DFT of the sequence x[n]={1,2,1,2,1,2,1,2} using decimation in time algorithm	5	L2
7	(a)Find the DFT of x[n]={0.5,0.5,0.5,0.5,-1,-1,-1,-1} using decimation in time algorithm	5	L2
	(b)State all properties of DFT	5	L2
8	(a)Compute the DFT for the sequence (0.5,0.5,0.5,0.5,1,1,1,1) using DIF-FFT	6	L2
	(b)Derive the equation to implement a butterfly structure In DIT-FFT algorithm	4	L2
9	(a)Explain the use of FFT algorithms in linear filtering and correlation. How is the FFT algorithm applied to determine inverse discrete Fourier transform	5	L2
	(b)Determine IDFT of the following (i) X(k)={1,1-j2,-1,1+j2} (ii)X(k)={1,0,1,0}	5	L2
10	(a)What is FFT? How many multiplications and additions are required to compute N point DFT using redix-2 FFT?	5	L1
	(b)Explain how linear convolution can be derived from Circular Convolution.	5	L2

Unit-III: DESIGN OF IIR DIGITAL FILTERS& REALIZATIONS: Analog filter approximations – Butter worth and Chebyshev, Design of IIR Digital filters from analog filters, Design Examples, Analog and Digital frequency transformations. Basic structures of IIR systems, Transposed forms			
1	(a)With an example explain the design procedure for Butterworth filter	5	L2
	(b)Explain about different types of analog to digital filter conversion methods.	5	L2
2	Using BLT, design a butterworth filter which satisfies the following conditions: 0.8 < |H(jω)| < 1 for 0 < ω < 0.2Π, |H(jω)| < 0.2 for 0.6Π < ω < Π,	10	L3
3	(a) Design a single pole LPF with a 3dB bandwidth of 0.2Π by the use os BLT applied to the analog filter $H(s) = \frac{\Omega_c}{\Omega_c + S}$, where Ωc is 3dB frequency of analog filter.	6	L2
	(b)Compare bilinear transformation and impulse invariant mapping	4	L2
4	(a)By impulse invariant method obtain the digital filter transfer function and the differential equation of the analog filter whose transfer function is given below. H(s)=1/(S+1)	6	L2
	(b)Convert the analog filter to a digital filter whose system function using bilinear transformation $H(s) = \frac{1}{(s+2)^2 + (s+1)}$	4	L2
5	Using BLT, design a butterworth filter which satisfies the following conditions: 0.7 < |H(jω)| < 1 for 0 < ω < 0.3Π, |H(jω)| < 0.3 for 0.7Π < ω < Π,	10	L3
6	Determine the system function $H(z)$ of the lowest order Butterworth digital filter with the following specification i)*3db ripple in pass band* $0 \le \omega \le 0.2\pi$ *ii)25db attenuation in stop band* $0.45\pi \le \omega \le \pi$	10	L3
7	(a)With an example explain the design procedure for Butterworth filter	3	L2
	(b)Design a low pass digital filter that will operate on sampled analog data such that the analog cut off frequency is 200 Hz (1 dB acceptable ripple) and at 400 Hz the attenuation is at least 20 dB with monotonic shape past 400 Hz. The sample rate is 2000 samples/sec. Use impulse invariant transformation	7	L2
8.	(a)Describe various Structures of IIR filters.	5	L2
	b) Design a Chebyshev filter with a maximum passband attenuation of 2 dB; at Ωp=20rad/sec and the stopband attenuation of 35 dB at Ωs=50 rad/sec.	5	L3

9	(a)Develop the cascade and parallel forms of the following causal IIR transfer functions. $$H(z) = \frac{(3+5z^{-1})(0.6+3z^{-1})}{(1-2z^{-1}+2z^{-2})(1-z^{-1})}$$	5	L2
	(b)Why IIR filters do not have linear phase	5	L2
10	Determine direct form I and cascade realization of the following system: $$H[z] = \frac{2\left(1-z^{-1}\right)\left(1+\sqrt{2}z^{-1}+z^{-2}\right)}{\left(1+0.5z^{-1}\right)\left(1-0.9z^{-1}+0.81z^{-2}\right)}$$	10	L2

UNIT-4:DESIGN OF FIR DIGITAL FILTERS& REALIZATIONS: Characteristics of FIR Digital Filters, frequency response. Design of FIR Digital Filters using Window Techniques and Frequency Sampling technique, Comparison of IIR & FIR filters. Basic structures of FIR systems, Lattice structures, Lattice-ladder structures

Q.NO	QUESTIONS	Marks	Level
1	(a)What are the characteristics of linear phase FIR digital filters?	5	L1
	(b)Design a Finite Impulse Response low pass filter with a cut-off frequency of 1 kHz and sampling rate of 4 kHz with eleven samples using Fourier series method.	5	L3
2	(a)Distinguish between IIR and FIR filters.	5	L1
	(b)Outline the steps involved in the design of FIR filter using windows	5	L2
3	Design a linear phase FIR filter with the magnitude response $$\left\|H(e^{j\Omega})\right\| = 1 \; for \; \|\Omega\| \le \frac{\pi}{8}$$ $$= 0 \quad for \; \pi/8 \le \|\Omega\| \le \pi$$ Use Hamming window. The length of the impulse response is limited to 9. Draw the Direct form structure of the filter.	10	L3
4	Design a linear phase FIR filter with the magnitude response $$\left\|H(e^{j\Omega})\right\| = 1 \; for \; \|\Omega\| \le \frac{\pi}{4}$$ $$= 0 \quad for \; \frac{\pi}{4} \le \|\Omega\| \le \pi$$ Use Rectangular window. The length of the impulse response is limited to 7. Find the magnitude response of designed filter.	10	L3
5	Design an ideal high pass filter with a frequency response $Hd(jw) = 1$ for $\pi/4 \le \|w\| \le \pi$ $= 0$ for $\|w\| \le \pi/4$ Find the values of h(n) for N = 11 using Hamming window. Find H (z) and determine the magnitude response.	10	L2
6	(a)Write a short note on Kaiser window	5	L2
	(b) What are the characteristics of linear phase FIR digital filters?	5	L2
7	(a)Write a short note on Frequency sampling method of realization	5	L2
	(b) Design an FIR digital low pass filter with cutoff frequency 1.2 radian and length N = 7. Use frequency sampling method.	5	L2

8	. (a)Explain the frequency-sampling method of FIR filter design with an example.	5	L2
	(b) List out the characteristics of FIR digital filters.	5	L2
9	(a) Distinguish between IIR and FIR filters.	4	L2
	(b) Design an FIR digital low pass filter with cutoff frequency 1.2 radian and length $N = 7$. Use frequency sampling method.	6	L3
10	(a)Explain Polyphase realization of FIR Filter	5	L2
	(b)Explain the frequency response of N Point rectangular window	5	L2

UNIT-5 : Introduction to programmable DSPs: Multiplier and Multiplier Accumulator, Modified bus structures and memory access schemes in P-DSPs, Multiple Access Memory, Multiported memory, VLIW architecture, Pipelining, Special addressing modes, On-Chip Peripherals. Architecture of ARM processors: Technical details of ARM Processors, Introduction to Cortex-M3 and cortex M4 processors - Processor type, processor architecture, instruction set, block diagram, memory systems.

Q.NO	QUESTIONS	Marks	Level
1	(a)Explain the difference between Von Neumann and Harvard architectures. Which architecture is preferred for DSP applications and why?	5	L1
	(b)What are the advantages of DSP processors over conventional microprocessors?	5	L2
2	(a)Discuss the on chip peripherals available on the ARM processor and explain their function.	5	L2
	(b)Explain with help of block diagram the architecture of ARM processor	5	L2
3	(a)Discuss various interrupt types supported by Cortex-M3	5	L2
	(b)Explain the concept of pipelining in DSP processors	5	L2
4	(a)Explain the Implementation of convolver with single multiplier/adder	5	L2
	(b)compare and contrast CISC and RISC Processors?	5	L2
5	(a)What are the special addressing modes of DSP? Explain.	5	L2
	(b)What are the architectural features of Cortex M4 DSP?	5	L2
6	Draw the block diagram of ARM digital signal processor and explain the functionality of ALU.	10	L2
7	(a)Write notes on VLIW architecture	5	L2
	(b)Write notes on Multiported memory	5	L2
8	(a)What is meant by bit reversed addressing mode? What is the application for which this addressing mode is preferred?	5	L2
	(b)Explain Memory Access schemes in DSPs.	5	L2
9	(a)Draw the pipelined MAC configuration to perform convolution operation and explain with neat timing diagrams.	5	L2
	b) What are the special addressing modes of DSP? Explain.	5	L2
10	Write notes on the following: a) VLIW architecture b) Multiported memory	10	L2