Code: 20A04502T

(ii) Chebyshev filters.

B.Tech III Year II Semester (R20) Supplementary Examinations January 2024

DIGITAL SIGNAL PROCESSING

(Electrical & Electronics Engineering)

Time: 3 hours Max. Marks: 70

PART – A

(Compulsory Question)

1	(a) (b) (c) (d) (e) (f) (g) (h) (i)	Answer the following: (10 X 02 = 20 Marks) What are the advantages of digital signal processing over analog signal processing? What is an LTI system? What are twiddle factors of the DFT? Why FFT is called so? Distinguish between Butterworth and Chebyshev (Type-I) filter. What are the different types of structures for realization of IIR systems? What are the advantages of FIR filters? List the basic structures of FIR filters. What is the need for multirate signal processing? Define Quantization.	2M 2M 2M 2M 2M 2M 2M 2M 2M 2M
		PART – B	
		(Answer all the questions: 05 X 10 = 50 Marks)	
2	(a)	Define the following terms as referred to LTI discrete time system: (i) Stability, (ii) Causality.	4M
	(b)	Determine whether the following systems are Causal or not. (i) y(n) = x(n) + x(n-2), (ii) y(n) = x(2n).	6M
		OR	
3	(a)	Determine whether the following discrete time signals are Periodic or not. If Periodic, determine the fundamental period. (i) cos $2n$, (ii) $\sin 4\pi n$.	5M
	(b)	Explain about the classification of discrete-time signals.	5M
4	(a) (b)	Given $x(n) = \{1, 1, 1, 1, 1, 1, 1, 1\}$, find $X(k)$ using DIT FFT algorithm. Discuss the applications of FFT algorithms. OR	5M 5M
5		Find the DFT of the following discrete time sequence $x(n) = \{1, -1, -1, -1, -1, -1, -1, 1\}$ using Radix-2 Decimation-In-Time FFT algorithm.	10M
6		Explain the following: (i) Butterworth filters,	10M

Contd. in Page 2

OR

R20

Code: 20A04502T

7		Determine the order of the filter using Chebyshev approximation and H(s) for the given data $\alpha_p=3$ dB, $\alpha_s=16$ dB, $f_p=1$ kHz, $f_s=2$ kHz.	10M
8	(a)	Distinguish between FIR and IIR filters.	3M
	(b)	Design a FIR digital filter to approximate an ideal low-pass filter with pass-band gain of unity, cut-off frequency of 850 Hz and working at a sampling frequency of $f_s = 5000$ Hz. The length of the impulse response should be 5. Use a rectangular window.	7M
		OR	
9		Briefly explain about the different window functions used in FIR filter design.	10M
10		Explain about Quantization of filter coefficients in Digital Signal Processing.	10M
		OR	
11		With a neat sketch, explain the method for sampling rate conversion by a factor I/D.	10M

R20

Code: 20A04502T

B.Tech III Year I Semester (R20) Regular & Supplementary Examinations January 2024

DIGITAL SIGNAL PROCESSING

(Electronics & Communication Engineering)

Time: 3 hours Max. Marks: 70

PART – A

(Compulsory Question)

1		Answer the following: (10 X 02 = 20 Marks)						
	(a)	Find the system transfer function of given difference equation using Z-transform $y(n) - 0.5y(n-1) = x(n)$.	2M					
	(b)	Define LTI system.	2M					
	(c)	Compare DFT and FFT based on complexity.	2M					
	(d)	Mention the applications of zero padding.	2M					
	(e)	Define prewarping and mention its use.	2M					
	(f)	Write the frequency transformations from low pass to other filters.	2M					
	(g)	Mention the necessary and sufficient condition for the linear phase characteristic of FIR filter.	2M					
	(h)	Define Gibb's phenomenon.	2M					
	(i)	Define the term fixed point arithmetic with an example.	2M					
	(j)	Define decimation and interpolation.	2M					
		PART – B						
		(Answer all the questions: 05 X 10 = 50 Marks)						
2		Determine magnitude and phase response for the system described by the difference	10M					
		equation: $y[n] = \frac{1}{2}x[n] + x[n-1] + \frac{1}{2}x[n-2]$. Sketch the plots.						
		OR						
3	(a)	Find impulse response of the system described by the difference equation	5M					
		y[n] + y[n-1] - 2y[n-2] = x[n-1] + 2x[n-2].						
	(b)	For each of the following system, determine whether or not the system is static/dynamic, linear/non-linear, time variant/invariant, and causal/noncausal, stable/unstable. $y(n) = cos[x(n)]$	5M					
4		Compute 8-point DFT of the sequence $x(n) = \{1, 2, 1, 2, 1, 2, 1, 2, 1\}$ using radix-2 DIF-FFT algorithm.	10M					
		OR						
5	(a)	Find the 8-point DFT of the sequence $x(n) = \{1, 2, 1, 0, 2, 3, 0, 1\}$	6M					
	(b)	Determine the circular convolution for the two sequences $x_1(n) = \{1, 2, 3, 4\}, x_2(n) = \{1, 5, 1, 3\}.$	4M					
6	(a)	The analog transfer function $H(s) = 2/(s+1)(s+2)$. Determine $H(z)$ using impulse invariance method.	5M					
	(b)	Realize the following system with difference equation in cascade form.	5M					
		y(n) = y(n - 1) + 2y(n - 2) + x(n) - x(n-1).						
	OR							

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7 Design a digital Butterworth filter satisfying the following specifications 10M

 $0.7 \le |H(e^{jw})| \le 1, 0 \le \omega \le 0.2\pi$

 $|H(e^{jw})| \le 0.2, 0.6\pi \le \omega \le \pi \text{ with } T = 1 \text{ sec}$.

Determine system function H(z) for a Butter worth filter using bilinear transformation method.

8 Design an ideal HPF with desired frequency response 10M

$$H_d(e^{jw}) = 1, \pi/4 \le |w| \le \pi$$

$$= 0, |w| \le \pi/4$$

Find the values of h(n) for N = 11 and also find H(Z) using Hanning window technique.

(a) Realize FIR filter with system function in cascade form. 9

 $H(z) = 1 + 5/2z^{-1} + 2z^{-2} + 2z^{-3}$

5M 5M

- (b) Prove that an FIR filter has linear phase if the unit sample response satisfies the condition $h(n) = \pm h(M-1-n)$, $n = 0, 1, \dots$ M-1. Also, discuss symmetric and anti symmetric cases of FIR filter.
- 10 Explain the concept of decimation by a factor D and interpolation by factor I.

10M

11 Explain the characteristics of limit cycle oscillation with respect to the system described by the 10M difference equation:

$$y(n) = 0.95 y(n - 1) + x(n)$$

$$x(n) = 0$$
; and $y(-1) = 13$

Determine the dead band range of the system.
