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Question Paper Code: 21399

B.E./B.Tech. DEGREE EXAMINATION, NOVEMBER/DECEMBER 2015.

Seventh Semester

Computer Science and Engineering

CS 2403/CS 73 — DIGITAL SIGNAL PROCESSING

(Common to Fifth Semester – Information Technology)

(Regulations 2008)

(Also common to PTCS 2403 – Digital Signal Processing for B.E. (Part-Time) Sixth Semester – Computer Science and Engineering – Regulations 2009)

Time: Three hours

Maximum: 100 marks

Answer ALL questions.

 $PART A - (10 \times 2 = 20 \text{ marks})$

- 1. A discrete-time signal $x(n) = \{-2, -1, 0, 1, -1, 1\}$ is multiplied by u(-n-2). What is the resulting signal?
- 2. What is a shift-invariant system? Give an example.
- 3. Find the circular convolution of two sequences $x_1(n) = \{1, 2, 2, 1\}$ and $x_2(n) = \{1, 2, 3, 1\}$.
- 4. Calculate the number of multiplications needed in the calculation of DFT using FFT algorithm with 32-point sequence.
- 5. Compare digital and analog filters.
- 6. What is meant by impulse invariant method of designing IIR filter?
- 7. What conditions on the FIR sequence h(n) are to be imposed in order that this filter can be called a linear phase filter?
- 8. What is the effect of quantization on pole locations?
- 9. What is the need for anti-imaging filter after up sampling a signal?
- 10. What is a channel vocoder?

PART B — $(5 \times 16 = 80 \text{ marks})$

- 11. (a) (i) Determine whether each of the following systems below is
 - (1) Causal (2) Linear (3) Dynamic (4) Time invariant (5) Stable.
 - $(A) y(n) = e^{-x(n)}$

(B)
$$y(n) = x(n) \sum_{k=-\infty}^{\infty} \delta(n-2k).$$
 (8)

(ii) Explain sampling theorem and reconstruction of the analog signal from its samples. (8)

Or

- (b) (i) Explain the properties of cross correlation and autocorrelation sequences. (8)
 - (ii) Find the discrete convolution of the following sequences. $u(n)*u(n-3). \tag{8}$
- 12. (a) (i) Explain any four properties of DFT. (8)
 - (ii) Compute the eight-point DFT of the sequence $x(n) = \{0.5, 0.5, 0.5, 0.5, 1, 2, -1, 0\}$ using the inplace-radix-2 DIT algorithm. (8)

Or

- (b) (i) Draw the flowgraph of a two-point radix-2 DIF-FFT algorithm.

 What is the basic operation of DIF algorithm? (8)
 - (ii) Find the IDFT of the sequence $X(k) = \{4, 1-j2.414, 0, 1-j0.414, 0, 1+j0.414, 0, 1+j2.414\} \text{ using DIF algorithm.} \tag{8}$
- 13. (a) (i) Design an analog Butterworth filter that has a -2dB passband attenuation at a frequency of 20 rad/sec and atleast -10dB stopband attenuation at 30 rad/sec. (10)
 - (ii) Explain the steps of design of digital filters from analog filters. (6)

Or

- (b) (i) Using the bilinear transform, design a high pass filter, monotonic in passband with cutoff frequency of 1000 Hz and down 10 dB at 350 Hz. The sampling frequency is 5000 Hz. (10)
 - (ii) Explain the methods of realization of digital filters. (6)

L 4.	(a)	(i)	Using a rectangular window technique design a low pass filt passband gain of unity, cut off frequency of 1000 Hz and wor a sampling frequency of 5 KHz. The length of the impulse reshould be 7.	rking at
		(ii)	Compare FIR and IIR filters.	(6)
			\mathbf{Or}	
	(b)	(i)	Obtain the cascade realization of system function	
			$H(z) = (1 + 2z^{-1} - z^{-2}) (1 + z^{-1} - z^{-2}).$	(10)
	•	(ii)	Explain the quantization errors due to finite word length r in digital filters.	egisters (6)
5 .	(a)	(i)	Explain aliasing effect in down sampling.	(8)
		(ii)	Explain subband coding technique used in speech coding.	(8)
			\mathbf{Or}	
	(b)	(i)	Explain digital processing of audio signals.	(8)
		(ii)	Explain digital signal processing in image enhancement.	(8)

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