[05 – 4110]

2016

IV/IV B.E. DEGREE EXAMINATION.

First Semester

Electronics and Communication Engineering

DIGITAL SIGNAL PROCESSING

(Common with EEE)

(Effective from the admitted batch of 2006–2007)

Time : Three hours   Maximum : 70 marks

Answer any FIVE questions, question No. 1 is compulsory.

All questions carry equal marks.

1. (a) What are the classification of discrete-time systems? $(7 \times 2 = 14)$

(b) What is LTI system?

(c) What is aliasing effect?

(d) What are the properties of region of convergence?

(e) Define discrete Fourier Series.

(f) What is main advantages of FFT?

(g) What are the properties of Chebyshev filter?

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2. (a) What is sampling and aliasing? Discuss advantages and applications. $(2 \times 7 = 14)$

(b) Determine the following system are time-variant and time invariant?

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

3. (a) Discuss the properties of Region of convergence? $(2 \times 7 = 14)$

(b) Find the stability of the system whose impulse response $h(n) = 2^n u(n)$.

4. (a) Write a detailed note on properties of D.F. series? $(2 \times 7 = 14)$

(b) Determine the 8-point DFT of the sequence of $x(n) = \{1, 1, 1, 1, 1, 1, 0, 0\}$.

5. (a) Discuss the require steps for Radix –2 DIF-FFT algorithm? $(2 \times 7 = 14)$

(b) Compute 4-point DFT of a sequence $x(n) = \{0, 1, 2, 3\}$ using DIT-DIF algorithm?

6. (a) Discuss the require steps to design of IIR filter using bilinear-Transformation? $(2 \times 7 = 14)$

(b) Apply bilinear transformation to $H(s) = 2/(s+1)(s+2)$ with $T = 1$ sec and find $H(z)$?

7. (a) Write a note on realization of FIR filter? $(2 \times 7 = 14)$

(b) Briefly explain the procedure for locate the zeros in FIR-Filter.

8. (a) Write a detailed note on criteria for selecting the digital signal processor? $(2 \times 7 = 14)$

(b) Discuss the techniques of speech coding?

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8. (a) What are the applications of digital signal processing in respect to speech analysis?

(b) Explain the role of DSP in speech processing.

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(Common with Electrical and Electronics Engineering

Time : Three hours Maximum : 70 marks

Questions No. 1 is compulsory.

Answer any FOUR from the remaining.

All questions carry equal marks.

1. (a) What is the stability condition for a discrete signal and find whether the following signal is stable or causal?

$$h(n) = -(0.5)^n\, u(-n-1)$$

(b) State and prove initial value theorem of Z-transforms.

(c) Explain how FFT reduces the computational complexity of DFT.

(d) Check the system $y(n) = \cos x(n)$ for stability.

(e) Differentiate between circular convolution and linear convolution.

(f) How many computations in 8-point system are reduced in FFT compared to DFT?

2. (a) State whether the following discrete time systems are linear, time invariant, causal, stable. Justify your answers with appropriate proof.

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

(ii) $y(n) = 2x(n-1) + nx(n+1)$

(iii) $y(n) = ne^{|x(n)|}$.

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(b) A discrete system is given by the following difference equation $y(n) - 4y(n-1) = x(n) + 5x(n-1)$ where $x(n)$ is the input and y(n)is the output. Determine its magnitude and phase response as a function of frequency for $w \le \pi$.

3. (a) State and prove time shifting property of Z-transform.

(b) Determine Z-transform, ROC and pole-zero locations of $x(n) = \alpha^n u(n) + \beta^n u(-n-1)$.

4. (a) Determine the 4-point circular convolution of the sequences:

$X_1(n) = \{1, 2, 3, 1\}$

$X_2(n) = \{4, 3, 2, 1\}$

Evaluate linear convolution sum of the above sequences using circular.

(b) State and prove the convolution property and shift of a sequence for DFT.

5. (a) Obtain the eight point DFT of the sequence $x(n) = \{2, 1, -1, 0, 0, 2, 0, -2\}$ using Radix-2, DIT FFT algorithm.

(b) Explain radix-2 Decimation-In-frequency algorithm with the diagram.

6. Design a digital High pass filter with 3 dB cut-off frequency at 60 rad/sec and attenuation of at least 60 dB for frequencies larger than 200 rad/sec. The sampling frequency is 1000 rad/sec. Obtain $H(Z)$ of the desired filter using Chebyshev approximation.

7. (a) What are the differences between FIR and IIR filters?

(b) Explain the Hamming window technique to design FIR filter.