Reg. No. : | | | | | | | | | | | | |

Question Paper Code : 40986

B.E./B.Tech. DEGREE EXAMINATIONS, NOVEMBER/DECEMBER 2024.

Fourth/Fifth Semester

Electronics and Communication Engineering

EC 3492 — DIGITAL SIGNAL PROCESSING

(Common to: Computer and Communication Engineering/Electronics and Telecommunication Engineering/Medical Electronics)

(Regulations 2021)

Time : Three hours Maximum : 100 marks

Answer ALL questions.

PART A — (10 × 2 = 20 marks)

1. State the condition for the existence of DTFT for an aperiodic sequence.

2. Find the four point DFT of a sequence $\begin{array}{ll} x(n) = 1 & \text{for } 0 \le n \le 2 \\ x(n) = 0 & \text{otherwise} \end{array}$

3. Define warping effect in Bilinear transformation.

4. Realize $y(n) + y(n-1) + \frac{1}{4}y(n-2) = x(n)$ in cascade form.

5. Distinguish between recursive and non recursive realization.

6. For what kind of applications symmetric and anti symmetric impulse response can be used?

7. Write an account on floating point arithmetic with an example.

8. What is overflow oscillations? Discuss the methods to prevented overflow.

9. If the spectrum of a sequence $x(n)$ is $X(e^{j\omega})$ then what is the spectrum of the down sampled by the factor 2?

10. Give the use of echo cancellation in DSP.

PART B — (5 × 13 = 65 marks)

11. (a) (i) State and prove the Time Reversal and Complex Conjugate properties of DFT. (4)

(ii) Using linear convolution find $y(n) = x(n) * h(n)$ for the sequences $x(n) = \{1, 2, -1, 2, 3, -2, -3, -1, 1, 1, 2, -1\}$ and $h(n) = \{1, 2\}$. Compare the result by overlap save method. (9)

Or

(b) An 8-point sequence is given by $x(n) = \{2, 2, 2, 2, 1, 1, 1, 1\}$. Compute 8-point DFT of $x(n)$ by radix DIT-FFT method. Sketch the magnitude and phase.

12. (a) Design a digital filter equivalent of a 2nd order Butterworth low-pass filter with a cut-off frequency $f_c = 100\,\text{Hz}$ and a sampling frequency $f_s = 1000$ samples/sec. Derive the finite difference equation and draw the realization structure of the filter. Given that the analogue prototype of the frequency-domain transfer function $H(s)$ for a Butterworth filter is $H(s) = \dfrac{1}{s^2 + \sqrt{2}s + 1}$ using Bilinear transformation.

Or

(b) Obtain an analog Chebyshev filter transfer function that satisfies for conditions $\dfrac{1}{\sqrt{2}} \le |H(j\Omega)| \le 1; 0 \le \Omega \le 2$.

13. (a) Design an ideal high pass filter with a frequency response

$$H_d(e^{j\omega}) = 1 \text{ for } \frac{\pi}{4} \le |\omega| \le \pi$$

$$= 0 \text{ for } |\omega| \le \frac{\pi}{4}$$

Find the value of $h(n)$ for $N = 11$ using Hamming window.

Or

(b) Using frequency sampling method, design a band pass filter with specifications Sampling frequency = 8000 Hz cut off frequencies $f_{c_1} = 1000\,\text{Hz}$ and $f_{c_2} = 3000\,\text{Hz}$ determine the filter coefficients for $N = 7$.

14\. (a) Realize the first order transfer function $H(z)=\frac{1}{1-az^{-1}}$ and draw the quantization noise model. Find the steady state noise power due to product round off.

Or

(b) Explain the characteristics of a limit cycle oscillations with respect to the system described by the difference equation $y(n)=0.95\ y(n-1)+x(n)$ with $x(n)=0.875$ for $n=0$

$x(n) = 0$ otherwise

Determine the dead band of the filter.

15\. (a) Explain the effective implementation of polyphase structures for decimation and interpolation filters.

Or

(b) Explain the architecture of the fixed point DSP processor. Discuss the applications of fixed point processor.

PART C — (1 × 15 = 15 marks)

16\. (a) Design an adaptive channel equalization algorithm in a typical digital communication system using a adaptive channel equalization filter.

Or

(b) Explain sampling rate increase by an integer factor I and derive the input-output relationship in both time and frequency domains.

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