Reg. No. : E N G G T R E E . C O M

Question Paper Code : 90988

B.E./B.Tech. DEGREE EXAMINATIONS, APRIL/MAY 2025

Fourth Semester

Electronics and Communication Engineering

EC 3492 — DIGITAL SIGNAL PROCESSING

(Common to Electronics and Telecommunication Engineering/Medical Electronics)

(Regulations 2021)

Time : Three hours | Maximum : 100 marks

Answer ALL questions.

PART A — ($10 \times 2 = 20$ marks)

1. If $x(n) = \{1, 2, 3, 1\}$ and $h(n) = \{1, 1, 1\}$, find output response $y(n)$ using Circular Convolution with zero padding method.

2. Consider a sequence $x(n) = \{2, 4, 6, 8, 0, 1, 3, 5, 7, 9\}$. Down sample the sequence by 3 and up sample by 2. Then what will be resulting sequence?

3. What is the relationship between analog and digital frequency in impulse invariant transformation?

4. Determine the order of the analog Butterworth filter that has a –2 dB pass band attenuation at a frequency of 20 rad/sec and atleast –10 dB stop band attenuation at 30 rad/sec.

5. What is the need for scaling in the digital filter realization?

6. List the desirable and undesirable features of FIR Filters.

7. What is meant by limit cycle oscillation in digital filter?

8. Define dead band of the filter. How to calculate the dead band of an IIR system?

9. State the applications of multirate signal processing.

10. Mention the role of DSP in speech processing.

PART B — ($5 \times 13 = 65$ marks)

11. (a) (i) A continuous time signal $x(t)$ is obtained at the output of an ideal low pass filter with cut off frequency $\omega_c = 10^3\pi$ rad/sec. If impulse train sampling is performed on signal $x(t)$ which of the following sampling periods will guarantee that $x(t)$ may be recovered from its sampled version using an appropriate low pass filter? (6)

(1) $T_s = 5\times10^{-4}$ sec

(2) $T_s = 2\times10^{-3}$ sec

(3) $T_s = 1\times10^{-4}$ sec

(ii) Compute the DFT of sequences $x_1(n) = \{3,2,1,2\}$, $x_2(n) = \{j,0,j,1\}$. (7)

Or

(b) (i) Determine the Nyquist sampling rate and Nyquist sampling interval for the signals

$x_1(t) = \sin 200\pi t + \sin 350\pi t$, $x_2(t) = \sin 200\pi t \cdot \sin 350\pi t$, and $x_3(t) = \sin^2 200\pi t$. (6)

(ii) Compute the DTFT of the following sequences $x_1(n) = 0.8^n u(n-2)$, $x_2(n) = 0.8^n u(n)\cos(0.1\pi n)$, and $x_3(n) = 0.8^{-n}u(-n)$. (7)

12. (a) Determine the Direct form II realization of the second order IIR transfer function $y(n) = 2b\cos\omega_0 y(n-1) - b^2 y(n-2) + x(n) - b\cos\omega_0 x(n-1)$. (13)

Or

(b) Convert the analog filter with system function (13)

$$H(s) = \frac{s+0.1}{(s+0.1)^2+9}.$$

into a digital IIR filter using bilinear transformation. The digital filter should have a resonant frequency of $\omega_r = \pi/4$.

13. (a) (i) Realize the following non-causal linear phase FIR system (7)

$$H(z) = \frac{2}{3}z + 1 + \frac{2}{3}z^{-1}.$$

(ii) Determine the Impulse Response $h(n)$ of a Band-stop filter to reject frequencies in the range 1 to 2 rad/sec using Rectangular Window with $N = 7$. (6)

Or

(b) (i) Realize the following causal linear phase FIR system (7)

$$H(z)=\frac{2}{3}+z^{-1}+\frac{2}{3}z^{-2}$$

(ii) Consider the FIR filter $\frac{1}{2}[x(n)+x(n-8)]$. Compute and sketch its magnitude and phase response. Also compute its response to the input $x(n)=\cos\frac{\pi}{2}n+\cos\frac{\pi}{4}n+\cos\frac{\pi}{8}n$. (6)

14. (a) Consider the transfer function $H(z)=H_1(z)H_2(z)$, where $H_1(z)=\frac{1}{1-a_1z^{-1}}$ and $H_2(z)=\frac{1}{1-a_2z^{-1}}$. Find the output round off noise power. Assume $\alpha_1=0.5$, $\alpha_2=0.6$ and find output round off noise power.

Or

(b) Find the effect of coefficient quantization on pole locations of the given second order IIR system, when it is realized in direct form I and in cascade form. Assume a word length of 4 bits through truncation.

$$H(z)=\frac{1}{1-0.9z^{-1}+0.2z^{-2}}.$$

15. (a) Draw the architecture of TMS320C6XX DSP Processor and explain its functionalities in detail.

Or

(b) Explain the pipelining concept of instruction execution and how it is advantageous compared to the serial execution.

PART C — (1 × 15 = 15 marks)

16. (a) Determine the system function *H(z)* of Chebyshev digital filter having least possible circuit complexity to meet the following specifications:

(i) 1 dB ripple in the passband $0 \le \Omega \le 2\,kHz$.

(ii) At least 30-dB attenuation in the stop band $4 < \Omega \le 5\,kHz$.

(iii) Convert this filter into an equivalent high-pass filter. Use Bilinear Transform for both the filters design.

Or

(b) Compute the 8-point DFT of $x(n)=[2,2,2,2,1,1,1,1]$ using (i) Radix 2 DIT FFT (ii) Radix 2 DIF FFT.

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