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

Question Paper Code : 70437

B.E./B.Tech. DEGREE EXAMINATIONS, NOVEMBER/DECEMBER 2021.

Fifth/Sixth Semester

Medical Electronics Engineering

EC 6502 — PRINCIPLES OF DIGITAL SIGNAL PROCESSING

(Common to Electronics and Communication Engineering and Biomedical Engineering)

(Regulations 2013)

(Also common to : PTEC 6502 – Principles of Digital Signal Processing for B.E. (Part-Time) – Electronics and Communication Engineering – Fourth Semester (Regulations – 2014))

Time : Three hours Maximum : 100 marks

Answer ALL questions.

PART A — (10 × 2 = 20 marks)

1. Compare Radix 2 DIT, DIF FFT algorithm.

2. Test the causality and stability of $y(n) = \sin x(n)$.

3. What are the requirements for the digital filter to be stable and casual?

4. Discuss the need for prewarping.

5. What are the two kinds of limit cycle behaviour in DSP?

6. List out the advantages of FIR filters.

7. What are the two kinds of limit cycle behavior in DSP?

8. Why rounding is preferred to truncation in realizing digital filter?

9. List the areas in which multirate processing is used.

10. State sampling theorem for a band limited signal.

PART B — (5 × 13 = 65 marks)

11. (a) (i) State and prove if $x_3(K) = x_1(K)\, x_2(K)$, then $x_3(n) = \sum_{m=0}^{N-1} x_1(m)\, x_2((n-m))_N$. (5)

(ii) Using the equation given in 11 (a)(i), for the 8 point DFT of the sequence

$x(n) = 1,\ 0 \le n \le 3$

$0, 4 \le n \le 7$, compute the

DFT of $x_1(n) = 1,\ n = 0$

$0,\ 1 \le n \le 4$

$1, 5 \le n \le 7$ (8)

Or

(b) (i) Compute the 8 point circular convolution

$x_1(n) = \{1, 1, 1, 1, 0, 0, 0, 0\}$

$x_2(n) = \sin \dfrac{3\pi n}{8},\ 0 \le n \le 7$

using matrix method. (10)

(ii) State the differences between (a) overlap-save (b) overlap-add. (3)

12. (a) Design a third order Butterworth digital filter using impulse invariant technique. Assume sampling period T = 1 sec. (13)

Or

(b) Convert the single pole low pass filter with system function $H(z) = \dfrac{0.5(1+z^{-1})}{1-0.302\,z^{-2}}$ into band pass filter with upper and lower cut off frequencies w_u and w_L respectively. The LPF has 3dB BW of $w_p = \dfrac{\pi}{6}$ and $w_u = \dfrac{3\pi}{4}$, $w_l = \dfrac{\pi}{4}$. (13)

13. (a) (i) List the steps involved by the general process of designing a digital filter. (7)

(ii) List the advantages of FIR filters. (6)

Or

(b) (i) The transfer function $H(z) = \sum_{N=0}^{M-1} h(n)\, z^{-n}$ characterizes a FIR filter $(M = 11)$. Find the magnitude response. (7)

(ii) Use Fourier series method to design a low pass digital filter to approximate the ideal specifications given by

$$H(e^{jw}) = \begin{cases} 1, & |f| \le f_p \\ 0, & f_p < |f| \le F/2 \end{cases}$$

Where f_p pass band frequency

F = sampling frequency. (6)

14. (a) The input to the system

$y(n) = 0.999 \quad y(n-1) + x(n)$ is applied to an ADC. What is the power produced by the quantization noise at the output of the filter if the input is quantized to (i) 8 bits (ii) 16 bits.

Or

(b) For the given transfer function $H(z) = H_1(z)\, H_2(z)$, where $H_1(z) = \dfrac{1}{1 - 0.5z^{-1}}$ and $H_2(z) = \dfrac{1}{1 - 0.4\, z^{-1}}$, find the output round off noise power. Calculate the value if $b = 3$ (excluding sign bit).

15. (a) Explain the concept of deciation by a factor D and interpolation by factor I. With help of equation explain sampling rate conversion by a rational factor I/D. (13)

Or

(b) Explain the operation of adaptive filter with suitable diagrams and equations. (13)

PART B — (1 × 15 = 15 marks)

16. (a) Determine the coefficients of a linear phase FIR filter of length M = 15 which has a symmetric unit sample response and a frequency response that satisfies the conditions. (15)

$$H_r\left(\frac{2\pi k}{15}\right) = \{1 \text{ for } k = 0, 1, 2, 3$$

$$H_r\left(\frac{2\pi k}{15}\right) = \{0.4 \text{ for } k = 4$$

$$H_r\left(\frac{2\pi k}{15}\right) = \{0\, for\ k = 5, 6, 7$$

Or

(b) Determine the system function $H(z)$ of the lowest order chebyshev and butterworth digital filter with the following specifications. (15)

(i) 3dB ripple in the passband $0 \le w \le 0.2\pi$

(ii) 25dB attenuation in the stop band $0.45\pi \le w \le \pi$. Use the bilinear transformation.