..E.(Instrumentation Engineering)(SEM-VI)(Choice Base) / DEC 2019 / 12.12.2019

[3 Hours] **[Total Marks : 80]**

Instructions:
1. Question.No.1 is compulsory.
2. Attempt any three questions from remaining questions.
3. Assume suitable data wherever necessary.

1 Attempt the following: 20

a. State Sampling Theorem. Determine the minimum sampling rate required to convert the following analog signal into discrete-time signal? 5

$$x(t) = 5\sin(250\pi t) + 7\cos(800\pi t)$$

b. Compare FIR and IIR filters. 5

c. Sketch the block diagram of Digital Signal Processing (DSP) system. State the advantages of DSP over analog signal processing. 5

d. Convert the analog filter with following transfer function into a digital filter using approximation of derivative method: 5

$$H_a(s) = \frac{2}{(s+2)^2 + 16}$$

2 a. Compute circular convolution of following sequences using DFT-IDFT method: 10

$x_1(n) = \{5, 4, 3, 2\}$, $x_2(n) = \{2, 2, 1, 1\}$

b. Realize the discrete-time system having following transfer function using direct-form I structure: 10

$$H(z) = \frac{-10(0.5 + z^{-1} + 1.25\, z^{-2} - 1.5\, z^{-3})}{15 + 25\, z^{-1} - 6.5\, z^{-2} + 8.5\, z^{-3}}$$

3 a. Design a FIR high-pass filter with following desired frequency response: 10

$$H_d(\omega) = \begin{cases} e^{j5\omega}, & 0.6\pi \le \omega \le \pi \\ 0, & otherwise \end{cases}$$

Use length of filter, $M = 11$ and Bartlet and Hamming window functions.

b. Determine 8-point DFT of following sequence using decimation-in-time (DIT) FFT algorithm and sketch the signal flow graph: 10

$x(n) = \{1, 2, 3, 4, 5, 6, 7, 8\}$

4 a. Design a digital Butterworth low-pass filter with following specifications: 10
Passband attenuation, $\delta_p = 0.89$
Stopband attenuation, $\delta_s = 0.25$
Passband frequency, $\omega_p = 0.3\pi\ rad/sample$
Stopband frequency, $\omega_s = 0.6\pi\ rad/sample$
Use Bilinear transformation method with sampling time, $T = 1\ sec.$

b. Explain the architecture of TMS 320C54XX DSP processor with the help of neat diagram. 10

5 a. What are the applications of adaptive filters? Describe the Least Mean Square (LMS) adaptive filter algorithm. 10

b. Determine 4-point DFT of following sequence using decimation-in-frequency (DIF) FFT algorithm and draw the signal flow graph: 6

$x(n) = \{1, 0, 2, 4\}$

c. State any two properties of DFT. 4

6 a. Design a digital type-I Chebyshev low-pass filter with following specifications: 10
Passband attenuation, $\delta_p = 0.9$
Stopband attenuation, $\delta_s = 0.2$
Passband frequency, $\omega_p = 0.25\pi\ rad/sam$
Stopband frequency, $\omega_s = 0.6\pi\ rad/sample$
Use Impulse Invariance method. Assume sampling time, $T = 0.5\ sec.$

b. Design a FIR low-pass filter with cut-off frequency $0.6\pi\ rad/sample$ and length of filter, M = 11 Use Blackman and Hanning window functions. 10
