B.E. Electronics & Telecommunication / Communication Engg. / Electrical (Electronics & Power) Engineering / Instrumentation Engg. Sixth Semester

ET604 / EP602 / IN605 - Digital Signal Processing

P. Pages : 2 **GUG/W/18/1683**

Time : Three Hours Max. Marks : 80

Notes :
1. All questions carry marks as indicted.
2. Assume suitable data wherever necessary.
3. Illustrate your answers wherever necessary with the help of neat sketches.

1. a) Explain the following system properties with example. **8**

a) Linear & Non-linear system

b) Time variant & Invariant system

c) Causal & Non causal system

d) Static & dynamic system.

b) Compute circular convolution of the following two sequences. **8**

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

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

OR

2. a) Find out 8-point DFT of a sequence $x(n) = \{1, 2, 3, 4, 4, 3, 2, 1\}$. Using DIF-FFT algorithm. **8**

b) Determine the inverse 2 transform by using partial fraction expansion method **8**

$$H[2] = \frac{2^2 - 32 + 8}{(z-2)(z+2)(z+3)}.$$

3. Obtain the direct form I, direct form II, cascade & parallel structure for the following system. **16**

$$y(n) = y(n-1) - \frac{1}{2}y(n-2) + x(n) - x(n-1) + x(n-2).$$

OR

4. a) What are the different basic structures of IIR system? Explain Direct form-I structure for IIR system. **8**

b) Given a three stage lattice filter with coefficients $k_1 = 0.3, k_2 = 0.5 \;\&\; k_3 = 0.8$ determine the FIR filter coefficients for the direct form structure. **8**

5. a) What is Gibb's phenomenon ? Explain the need of window function in design of FIR filter. **8**

b) Design FIR filter using type-I frequency sampling technique which has the following specifications. **8**

$$Hd\left(e^{i\omega}\right) = \begin{cases} e^{-i\left(\frac{M-1}{2}\right)\omega} & ; \quad 0 \le |\omega| \le \frac{\pi}{2} \\ = 0 & ; \quad \frac{\pi}{2} \le |\omega| \le \pi \end{cases}$$

For M = 7

OR

6. a) Design a FIR filter with **8**

$$Hd\left(e^{-i\omega}\right) = \begin{cases} e^{-i3\omega} & \frac{-\pi}{4} \le \omega \le \frac{\pi}{4} \\ 0, & \frac{\pi}{4} \le |\omega| \le \pi \end{cases}$$

Using a Hamming window with M =7 M is the order of filter.

b) What is Linear phase filter? What conditions are to be satisfied by impulse Response of FIR system in order to have linear phase? **8**

7. a) Using Impulse Invariant technique with $T = 1\text{sec}$ determine $H(z)$ if **6**

$$H(s) = \frac{1}{(s+1)(s+2)}.$$

b) Design a Chebyshev filter with a maximum passband attenuation of 2.5dB at $\Omega_p = 20 \text{ rad/sec}$ & the stop band attenuation of 30 dB at $\Omega_5 = 50 \text{ rad/sec}$. **10**

OR

8. a) Convert the analog filter system function $H(s) = \frac{1}{(s+1)2}$ into digital IIR by means of BLT assume $T = 0.1 \text{ sec}$. **8**

b) Design an analog butter worth filter that has – 2dB passband attenuation at a frequency of 20 rad/sec & at least 10 dB stopband attenuation at 30 rad/sec. **8**

9. a) Explain the process of reducing the sampling rate by a factor D i.e. decimation by factor D. **8**

b) What is Multirate DSP? Explain the different technique of sampling rate conversion. **8**

OR

10. a) Explain Analysis filter bank and synthesis filter bank. **8**

b) Explain Quadrature-Mirror filter [QMF] Bank in detail. **8**

****