B.E.(with Credits)-Regular-Semester 2012 - Instrumentation Engineering Sem VI

IN605 - Digital Signal Processing

P. Pages : 2 **GUG/W/16/5389**

Time : Three Hours Max. Marks : 80

Notes :
1. Same answer book must be used for each section.
2. All questions carry marks as indicated.
3. Due credit will be given to neatness and adequate dimensions.
4. Assume suitable data wherever necessary.

1. a) Write atleast six advantages and mention application of DSP. **6**

b) Explain any five properties of z transform. **10**

OR

2. a) State and prove the circular convolution property of DFT. **4**

b) Given x(n)={0,1,2,3,4,5,6,7} find x(k) using DIT FFT algorithm. **8**

c) Compute the DFTs of the sequence $x(n) = \cos\frac{n\pi}{2}$ where, N=4, using DIF FFT algorithm. **4**

3. a) Sketch the block diagram for the direct form realization and the frequency sampling realization of the M=32, $\alpha = 0$; linear phase (symmetric) FIR filter which has frequency samples. **6**

$$H\left(\frac{2\pi k}{3_2}\right) = \begin{cases} 1 & k = 0,1,2 \\ 1/2 & k = 3 \\ 0 & k = 4,5,\ldots.15. \end{cases}$$

b) Give three stage lattice filter with coefficients $k_1 = 1/4, k_2 = 1/2, k_3 = 1/3.$ Determine the FIR filter coefficients for the direct form structure. **5**

c) Determine the lattice coefficient corresponding to the FIR filter. With system function. **5**

$$H(z) = A_3(z) = 1 + \frac{13}{24} Z^{-1} + \frac{5}{8} Z^{-2} + \frac{1}{3} Z^{-3}$$

OR

4. a) Draw a signal flow graph to implement the following system **8**

i) Parallel form ii) Cascade form

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

b) Obtain the i) direct form - I & ii) direct form - II realization for the following system. **8**

$$y(n) = -0.1y(n-1) + 0.2y(n-2) + 3x(n) + 3.6x(n-1) + 0.6x(n-2)$$

5. a) A low pass filter has the desired response as given below. **8**

$$H_d\left(e^{j\omega}\right)=\begin{cases} e^{-j3w} & 0 \le \omega < \pi/2 \\ 0 & \pi/2 \le \omega \le \pi \end{cases}$$

Determine the filter coefficients h(n) for M=7 using Type - I frequency sampling technique.

b) Design an FIR digital filter to approximate an ideal low pass filter with passband gain of unity, cut- off frequency of 850 Hz and working at a sampling frequency of f_S = 5000 Hz. The length of the impulse response should be 5. Use rectangular window. **8**

OR

6. a) Design a high pass filter using Hamming window with a cut-off frequency of 1.2 rad/sec and N=9. **10**

b) Compare design methods for linear phase FIR filters. **6**

7. a) Determine H(z) using the impulse invariant technique for the analog system function. **10**

$$H(s)=\frac{1}{(S+0.5)\left(S^2+0.5S+2\right)}$$

b) Obtain H(z) using bilinear transformation if $H(s)=\frac{1}{(S+1)^2}$ and T=0.15. **6**

OR

8. a) A digital low pass filter is required to meet the following specification. **10**

Pass-band ripple $\le$ 1 dB
Pass band edge = 4 kHz
Stop- band attenuation $\ge$ 30 dB
Stop- band edge = 6 kHz
Sampling rate = 24 kHz
Design the filter by performing a bilinear transformation on an analog system function. Use Butterworth approximation.

b) Explain advantages of FIR filters over IIR filter. **6**

9. a) Discuss Up- sampler and Down sampler in brief. **8**

b) Explain the Lagrange's interpolation algorithm. **8**

OR

10. a) Discuss at least two applications is multirate digital signal processing. **8**

b) Describe how DSP is useful for speech application. **8**

****