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**Fifth Semester B.E. Degree Examination, Dec.2015/Jan.2016**  
**Digital Signal Processing**

Time: 3 hrs.

Max. Marks:100

**Note: 1. Answer FIVE full questions, selecting at least TWO questions from each part.**  
**2. Use of normalized filter tables not permitted.**

**PART – A**

- 1 a. Define N-point DFT and IDFT of a sequence. (03 Marks)
- b. Find the 8-point DFT of the sequence  $x(n) = \{1, 1, 1, 1, 1, 1, 0, 0\}$ . (08 Marks)
- c. Find the IDFT of  $X(K) = \{4, -2j, 0, 2j\}$ . (06 Marks)
- d. Obtain the relation between DFT and Z-transform. (03 Marks)
- 2 a. State and prove circular convolution property. (06 Marks)
- b. For  $x(n) = \{7, 0, 8, 0\}$ , find  $y(n)$ , if  $Y(K) = X((K - 2))_4$ . (06 Marks)
- c. Let  $x(n) = \{1, 2, 0, 3, -2, 4, 7, 5\}$ . Evaluate the following :
  - i)  $X(0)$       ii)  $X(4)$       iii)  $\sum_{K=0}^7 X(K)$       iv)  $\sum_{K=0}^7 |X(K)|^2$  (08 Marks)
- 3 a. In the direct computation of N-point DFT of  $x(n)$ , how many
  - i) Complex multiplications, ii) Complex additions      iii) Real multiplications
  - iv) Real additions and      v) Trigonometric function evaluations are required. (10 Marks)
- b. Find the output  $y(n)$  of a filter whose impulse response  $h(n) = \{1, 2\}$  and input signal  $x(n) = \{1, 2, -1, 2, 3, -2, -3, -1, 1, 1, 2, -1\}$  using overlap save method. (10 Marks)
- 4 a. Develop 8-point DIF-FFT radix-2 algorithm and draw the signal flow graph. (10 Marks)
- b. Find 8-point DFT of a sequence  $x(n) = \{1, 1, 1, 1, 0, 0, 0, 0\}$  using DIT-FFT radix-2 algorithm. Use butterfly diagram. (10 Marks)

**PART – B**

- 5 a. Given  $|H_a(j\Omega)|^2 = \frac{1}{(1 + 4\Omega^2)}$ , determine the analog filter system function  $H_a(s)$ . (08 Marks)
- b. Let  $H(s) = \frac{1}{(s^2 + \sqrt{2}s + 1)}$  represent transfer function of a low pass filter with a pass band of 1 rad/sec. Use frequency transformation to find the transfer functions of the analog filters,
  - i) A LPF with pass band of 10 rad/sec.
  - ii) A HPF with cut-off frequency of 5 rad/sec. (08 Marks)
- c. Compare Butterworth and Chebyshev filters. (04 Marks)
- 6 a. Realize the FIR filter  $H(z) = \frac{1}{2} + \frac{1}{3}z^{-1} + z^{-2} + \frac{1}{4}z^{-3} + z^{-4} + \frac{1}{3}z^{-5} + \frac{1}{2}z^{-6}$  in direct form. (04 Marks)
- b. Obtain direct form-I, direct form – II, cascade and parallel form realization for the following system:  $y(n) = 0.75y(n-1) - 0.125y(n-2) + 6x(n) + 7x(n-1) + x(n-2)$  (16 Marks)

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- 7 a. A LPF is to be designed with frequency response,

$$H_d(e^{j\omega}) = H_d(\omega) = \begin{cases} e^{-j2\omega}, & |\omega| < \frac{\pi}{4} \\ 0, & \frac{\pi}{4} < |\omega| < \pi \end{cases}$$

Determine  $h_d(n)$  and  $h(n)$  if  $\omega(n)$  is a rectangular window,

$$\omega_R(n) = \begin{cases} 1, & 0 \leq n \leq 4 \\ 0, & \text{Otherwise} \end{cases}$$

Also, find the frequency response,  $H(\omega)$  of the resulting FIR filter. **(10 Marks)**

- b. Explain the design of linear phase FIR filter using frequency sampling technique. **(10 Marks)**
- 8 a. Explain the design of IIR filter by using Impulse Invariance Method (IIM) technique also explain mapping of analog to digital filter by IIM. **(10 Marks)**
- b. Convert the analog filter with system function,  $H_a(s) = \frac{s+0.1}{(s+0.1)^2+16}$  into a digital IIR filter by means of bilinear transformation (BLT). The digital filter is to have a resonant frequency of  $\omega_r = \frac{\pi}{2}$  **(10 Marks)**

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