USN | | | | | | | | | | 10EC52

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.***

Important Note : 1. On completing your answers, compulsorily draw diagonal cross lines on the remaining blank pages.
2. Any revealing of identification, appeal to evaluator and /or equations written eg, 42+8 = 50, will be treated as malpractice.

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 $|H_a(j\Omega)|^2 = \dfrac{1}{(1+4\Omega^2)}$, determine the analog filter system function $H_a(s)$. **(08 Marks)**
b. Let $H(s) = \dfrac{1}{(s^2+\sqrt{2}s+1)}$ 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)**

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,

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

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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