ED – 882

VII Semester B.E. (E&E) Degree Examination, December 2014/January 2015
(2K6 Scheme)
(2K6EE704) : DIGITAL SIGNAL PROCESSING

Time : 3 Hours Max. Marks : 100

***Instruction** : Answer **any five full** questions.*

1. a) State and prove the following properties of DFT
   i) Convolution in Time Domain
   ii) Circular Frequency shift. **10**

   b) Compute N point DFT of the sequences
   i) $x(n) = a^n \; 0 \leq n \leq N-1$
   ii) $x(n) = 1$ for n even
   $\quad = 0$ for n odd **10**

2. a) Obtain Linear convolution of the following sequences using overlap add and overlap save method with N = 6 and verify the Results.
   $x(n) = \{1, 2, 0, -3, 4, 2, -1, 1, -2, 3, 2, 1, -3\}$
   $h(n) = \{1, 1, 1\}$ **10**

   b) Determine DFT for a continuous time signal $x(t) = \sin 2\pi f t$; with f = 50 Hz. Use 8 point DIT-FFT algorithm. **10**

3. a) The even samples of the 11-point DFT of length 11 real sequences are given by
   $X(0) = 2$, $X(2) = -1 - j3$, $X(4) = 1 + j4$
   $X(6) = 9 + j3$, $X(8) = 5$, $X(10) = 2 + j2$
   Determine the missing odd samples. **6**

   b) Compare the number of complex multiplications required to compute N = 64 point sequence using Direct computations of DFT versus FFT algorithms. Also obtain speed improvement factor for the above case. **4**

   c) Realize the following systems using direct form and cascade form with

P.T.O.

minimum number of multipliers

i) $H(z) = \left(1 + \frac{1}{2}z^{-1} + \frac{1}{2}z^{-2} + z^{-3}\right)(1 + z^{-1})$

ii) $H(z) = \left(\frac{1}{2} + z^{-1} + \frac{1}{2}z^{-2}\right)\left(1 + \frac{1}{3}z^{-1} + z^{-2}\right).$ **10**

4. a) Obtain cascade and parallel realization of the system function

$$H(z) = \frac{(1 + z^{-1})}{\left(1 - \frac{1}{4}z^{-1}\right)\left(1 - \frac{3}{2}z^{-1} + \frac{1}{2}z^{-2}\right)}.$$ **6**

b) Sketch the ladder structure for the system $H(z) = \dfrac{1 - 0.6\, z^{-1} + 1.2\, z^{-1}}{1 + 0.15\, z^{-1} - 0.64\, z^{-2}}.$ **6**

c) Determine order and poles of low pass Butterworth filter that has a 3 dB attenuation at 500 Hz and an attenuation of 40 dB at 1000 Hz. **8**

5. a) Design a digital Chebyshev filter to satisfy the constraints, T = 1 sec.

$0.8 \le \left|H(e^{j\omega})\right| \le 1, \quad 0 \le \omega \le 0.2\pi$

$\left|H(e^{j\omega})\right| \le 0.2, \quad 0.6\pi \le \omega \le \pi$

using Bilinear transformations. **12**

b) An Analog filter has a transfer function $H(s) = \dfrac{10}{s^2 + 7s + 10}$. Design a digital filter to realize this using Impulse Invariant Method. Take T = 1 Sec. **8**

6. a) List the advantages and disadvantages of Bilinear transformations. **5**

b) Derive the transformations of IIR filter using approximation of derivatives by backward difference and verify whether it satisfies the sufficient and necessary conditions of mapping. **5**

c) Convert the following low pass Butterworth filter with system function $H(s) = \dfrac{s + 0.3}{s^2 + 3s + 5}$ into

i) High Pass filter

ii) Band stop filter. **10**

7. a) Compare FIR and IIR filters. **5**

   b) Discuss the merits and demerits of window function based filter design. **5**

   c) Design a FIR filter with $H_d(e^{j\omega}) = \begin{cases} e^{-j3\omega} & -\pi/4 \le \omega \le \pi/4 \\ 0 & \pi/4 \le |\omega| \le \pi \end{cases}$. use Hanning window. **10**

8. a) Explain different addressing modes with examples as related to a DSP processor. **8**

   b) With a Block diagram, describe the architecture of TMS – 320 DSP processor. **12**

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