Total No. of Questions : 8] [Total No. of Printed Pages : 3

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MEDC–103

M. Tech. (First Semester)
EXAMINATION, Dec., 2010

DSP APPLICATION

(MEDC–103)

Time : Three Hours

Maximum Marks : 100

Minimum Pass Marks : 40

Note : Attempt any *five* questions. All questions carry equal marks.

1. (a) The accumulator described by equation $y(n) = \sum_{k=-\infty}^{n} x(k)$ is excited by the sequence $x(n) = n\,u(n)$. Determine its output under the following condition :

   (i) It is initially relaxed [*i.e.* $y(-1) = 0$]

   (ii) Initially $y(-1) = 1$.

   (b) Determine the output $y(n)$ of a relaxed linear time invariant system with impulse response :

   $$h(n) = a^n u(n) \quad |a| < 1$$

   when the input is a unit step sequence that is :

   $$x(n) = u(n)$$

P. T. O.

2. (a) Determine the impulse response for the cascade of two linear time invariant systems having impulse responses :

   $$h_1(n) = \left(\frac{1}{2}\right)^n u(n) \text{ and } h_2(n) = \left(\frac{1}{4}\right)^n u(n)$$

   (b) Differentiate between a recursive and a non-recursive system.

3. (a) Determine the z-transform and ROC of the signal :

   (i) $x(n) = [3(2^n) - 4(3^n)]\,u(n)$

   (ii) $x(n) = a^n (\cos \omega_0 n)\,u(n)$

   (b) Determine the Z-transform and ROC of the signal :

   (i) $x(n) = n\,a^n\,u(n)$

   (ii) $x(n) = -n\,a^n\,u(-n-1)$

4. (a) Prove the convolution property of Z-transform.

   (b) Determine the signal $x(n)$ whose Z-transform is given by :

   (i) $X(z) = \log(1 + az^4) \quad |z| > |a|$

   (ii) $X(z) = \dfrac{1}{1 - 1{\cdot}5z^{-1} + 0{\cdot}5z^{-2}} \quad |z| > 1$

5. (a) Compute the N-point DFT of the signal :

   $$x(n) = \sin \frac{2n}{N} K_0 n \qquad 0 \le n \le N-1$$

   (b) Compute the eight point DFT of the sequence :

   $$x(n) = \begin{cases} 1 & \le n \le 7 \\ 0 & \text{otherwise} \end{cases}$$

   by using the decimation in frequency FFT algorithm.

6. (a) Discuss the designing of Butterworth IIR filter.

   (b) Design a single pole low pass digital filter with a 3 db bandwidth of $0 \cdot 2\pi$ using the bilinear trransformation applied to the analog filter :

$$H(s) = \frac{\Omega_c}{s + \Omega_c}$$

   where Ω_c is the 3 db bandwidth of the analog filter.

7. (a) Discuss the designing of FIR filter using Kaiser window. Show its relationship to other window.

   (b) Discuss the round off effects in Digital Filters.

8. Write short notes on any *two* of the following :

   (a) Wavelet transform

   (b) Haar transform

   (c) Quantization of filter coefficients

3