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# Sixth Semester B.E. Degree Examination, June 2012

## Digital Signal Processing

Time: 3 hrs. Max. Marks:100

**Note:** ***Answer FIVE full questions, selecting at least TWO questions from each part.***



### PART – A

1 a. Find the N-point DFT of $x(n) = a^n$ for $0 < a < 1$. — 4 **(04 Marks)**

b. A discrete time LTI system has impulse response $h(n) = 2\delta(n) - \delta(n-1)$. Determine the output of the system if the input is $x(n) = \{\delta(n) + 3\delta(n-1) + 2\delta(n-2) - \delta(n-3) + \delta(n-4)\}$, using circular convolution. — 4 **(06 Marks)**

c. Determine 8- point DFT of the signal: — 8
$x(n) = \{1, 1, 1, 1, 1, 1, 0, 0\}$. Also sketch its magnitude and phase. **(10 Marks)**

2 a. g(n) and h(n) are the two sequences of length 6 with 6-point DFT's G (k) and H (k) respectively. The sequence $g(n) = \{4, 3, 1, 5, 2, 6\}$. The DFT's are related by circular frequency shift as $H(k) = G((k-3))_6$. Determine h(n) without computing DFT and IDFT. **(05 Marks)**

b. Perform circular convolution of the sequences $x_1(n) = \{1, 1, 2, 2\}$ and $x_2(n) = \{1, 2, 3, 4\}$, using tabular arrays. 5 **(05 Marks)**

c. Explain with necessary diagrams and equations, the concept of overlap-save method for linear filtering. 8 **(10 Marks)**

3 a. First five points of the 8-point DFT of a real valued sequence is given by $x(0) = 0$, $x(1) = 2 + j2$, $x(2) = -j4$, $x(3) = 2 - j2$, $x(4) = 0$. Determine the remaining points. Hence find the original sequence x(n) using DIT-FFT algorithm. 2 **(12 Marks)**

b. Develop DIT-FFT algorithm for composite value of N = 6. Draw the corresponding signal flow graph. 2 **(08 Marks)**

4 a. Determine direct forms I and II for the second-order filter given by:

$$Y(n) = 2b\cos w_o\, y(n-1) - b^2 y(n-2) + x(n) - b\cos w_o\, x(n-1).$$ **(08 Marks)**

b. Given the system function:

$$H(z) = \frac{2 + 8_z^{\ -1} + 6z^{-2}}{1 + 8z^{-1} + 12z^{-2}}$$

Relaise using ladder structure. **(06 Marks)**

c. Obtain cascade relisation of the system function:

$$H(z) = \left(1 + \frac{1}{2}z^{-1} + z^{-2}\right)\left(1 + \frac{1}{4}z^{-1} + z^{-2}\right).$$ **(06 Marks)**

### PART – B

5 a. Design an analog chebyshev with following specifications: 8 8
Passband ripple : 1 db for $0 \le \Omega \le 10$ rad/sec.
Stopband attenuation: –60 db for $\Omega \ge 50$ rad/sec. **(10 Marks)**

b. The system function of the analog filter is given as

$$H_a(s) = \frac{s + 0.1}{(s + 0.1)^2 + 9}.$$

Obtain the system function of the IIR digital filter by using impulse invariance method. **(10 Marks)**

6 a. What is frequency transformation? Why is it required? **(04 Marks)**

b. Compare bilinear transformation with impulse invariance transformation. **(06 Marks)**

c. Design first high pass butter worth filter for cutoff frequency = 30 Hz and sampling frequency = 150 Hz by bilinear transformation. **(10 Marks)**

7 a. Design a symmetric FIR low pass filter whose desired frequency response is given as,

$$H_{\alpha}(w) = \begin{cases} e^{-jwp}, & \text{for } |w| \leq w_c \\ 0, & \text{otherwise} \end{cases}.$$

The length of the filter should be 7 and $w_c = 1$ rad/sample. Use rectangular window. **(10 Marks)**

b. Design a normalized linear phase FIR filter having the phase delay of T = 4 and at least 40 db attenuation in the stopband. Also obtain the magnitude/frequency response of the filter. **(10 Marks)**

8 a. Explain the design of an FIR filter based on frequency sampling approach. **(10 Marks)**

b. Draw the architecture of TMS 320 C5 x family DSP processors and explain. **(10 Marks)**

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