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Fifth Semester B.E. Degree Examination, June/July 2016
Digital Signal Processing

Time: 3 hrs. Max. Marks:100

Note: *1. Answer any FIVE full questions, selecting atleast TWO questions from each part.*
2. Use of prototype filter tables is 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. Find the N – point DFT of $x(n) = a^n$ for $0 < a < 1$. **(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 $x(n) = \{\delta(n) + 3\delta(n-1) + 2\delta(n-2) - \delta(n-3) + \delta(n-4)\}$ using circular convolution. **(06 Marks)**

c. Determine 8 – point DFT of the signal $x(n) = \{1, 1, 1, 1, 1, 1, 0, 0\}$. 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. **(07 Marks)**

b. Given $x(n) = \{1, 2, 3, 4\}$ and $h(n) = \{1, 2, 2\}$ compute i) circular convolution ii) linear convolution iii) linear convolution using circular convolution. **(08 Marks)**

c. Prove Parseval's relation as applied to DFT. **(05 Marks)**

3 a. Explain with necessary diagrams and equations the concept of overlap – save method for linear filtering. **(10 Marks)**

b. Write a note on Goertzel algorithm. **(05 Marks)**

c. What is in-place computation? What is the total number of complex additions and multiplications required for N = 64 point, if DFT is computed directly and if FFT is used? Also find the number of stages required and its memory requirement. **(05 Marks)**

4 a. First five points of the 8 – point DFT of a real valued sequence is given by $x(0) = 0$, $x(1) = 2 + 2j$, $x(2) = -4j$, $x(3) = 2 - 2j$, $x(4) = 0$. Determine the remaining points. Hence find the original sequence x(n) using DIT – FFT algorithm. **(10 Marks)**

b. Find the 4 – pt circular convolution of $x(n) = \{1, 1, 1, 1\}$ and $h(n) = \{1, 0, 1, 0\}$ using radix 2 DIF – FFT algorithm. **(10 Marks)**

PART – B

5 a. Design an analog Chebyshev filter with the following specifications :
Passband ripple : 1 dB for $0 \le \Omega \le 10$ rad/sec
Stopband attenuation : –60 dB for $\Omega \ge 50$ rad/sec. **(12 Marks)**

b. Derive the expressions of order and cutoff frequency of a analog butter worth filter. **(08 Marks)**

6 a. Realize the following difference equation using digital structures in all the forms :
$y(n) - \frac{3}{4}y(n-1) + \frac{1}{8}y(n-2) = x(n) + \frac{1}{3}x(n-1)$. **(16 Marks)**

b. Realize the FIR filter whose transfer function is given by :
$H(z) = 1 + \frac{3}{4}z^{-1} + \frac{17}{8}z^{-2} + \frac{3}{4}z^{-3} + z^{-4}$ in direct form. **(04 Marks)**

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

$$H_d(\omega) = \begin{cases} e^{-j\omega p} & \text{for } |\omega| \le \omega_c \\ 0 & \text{otherwise} \end{cases}$$

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

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

8 a. Let $H_a(S) = \dfrac{b}{(s+a)^2 + b^2}$ be a causal II order analog transfer function. Show that the causal II order digital transfer H(z) obtained from $H_a(s)$ through impulse invariance is given by :

$$H(z) = \frac{e^{-aT}\sin bTz^{-1}}{1 - 2e^{-aT}\cos bTz^{-1} + e^{-2aT}z^{-2}}.$$ **(10 Marks)**

b. Design an IIR digital butterworth filter that when used in the analog to digital with digital to analog will satisfy the following equivalent specification.

i) Lowpass filter with –1 dB cutoff 100 πrad/sec
ii) Stopband attenuation of 35 dB at 1000 π rad/sec
iii) Monotonic in stopband and passband
iv) Sampling rate of 2000 rad/sec
v) Use bilinear transformation. **(10 Marks)**

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