(Following Paper ID and Roll No. to be filled in your Answer Book)

PAPER ID : 2488 Roll No. | | | | | | | | | | |

B.Tech.

(SEMESTER-VI) THEORY EXAMINATION, 2012-13

DIGITAL SIGNAL PROCESSING

Time : 3 Hours] *[Total Marks : 100*

SECTION – A

1. Attempt **all** question parts : $10 \times 2 = 20$

(a) Determine the direct form realization of the linear phase filter given by $h(n) = \{\underset{\uparrow}{1}, 2, 3, 4, 3, 2, 1\}$.

(b) Find whether the digital filter given by the impulse response $h(n) = \{1\ -2\ 0\ 2\ -1\}$ is linear phase or not. Prove your answer.

(c) "A stable system is always causal". Justify the statement.

(d) Compute the N-point DFT of the signal $x(n) = a^n$ for $0 \le n \le N-1$

(e) Compare between linear convolution and circular convolution.

(f) If $x(n) = \delta(n) + \delta(n-1) + \delta(n-2)$, determine $X(e^{j\omega})$.

(g) Discuss the properties of Chebyshev polynomial.

(h) Explain the term "Group delay" with respect to filters.

(i) Draw the butterfly diagram for N = 8.

(j) State the condition for design of stable digital filter from stable analog filter.

SECTION – B

2. Attempt any **three** question parts : $3 \times 10 = 30$

(a) Consider an FIR filter described by the system function

$$H(z) = 1 + 2.88z^{-1} + 3.4048z^{-2} + 1.74z^{-3} + 0.4z^{-4}$$

(i) Sketch the lattice realization of the filter.

(ii) Is the system minimum phase

(b) Find the linear convolution of sequences using the overlap add method $x(n) = \{1, 2, 0, 3, 4, 1, 0, 1, 2, 3, 2, 1\}$ and $h(n) = \{4, 1, 2\}$

(c) Convert the analog filter with the system function

$$H_a(s) = \frac{s + 0.2}{(s + 0.2)^2 + 9}$$

into a digital IIR filter using impulse invariant technique. Assume T = 1s.

(d) Determine the cascade and parallel form realization for an LTI system described by the difference equation

$$y(n) = \frac{1}{4} y(n-2) + x(n)$$

(e) Given that $x(n) = \{2, 2, 2, 2, 1, 1, 1, 1\}$. Compute 8 point FFT using decimation-in-time method.

SECTION – C

Attempt **all** questions : **5 × 10 = 50**

3. Attempt any **two** parts : **2 × 5 = 10**

(a) An FIR filter is described by the difference equation $y(n) = x(n) - x(n-10)$. Compute and sketch its Fourier Transform magnitude and phase spectrum.

(b) Define Discrete Time Fourier Transform (DTFT) of a sequence $x(n)$. Determine DTFT for $\cos\omega_0 n$ and $\sin \omega_0 n$.

(c) Obtain the direct form II structure for the system given by the transfer function

$$y(n) = \frac{3}{4} y(n-1) - \frac{1}{8} y(n-2) + x(n) + \frac{1}{3} x(n-1)$$

4. Attempt any **one** part : **1 × 10 = 10**

(a) The desired frequency response of a low pass filter is given by

$$H_d(e^{j\omega}) = \begin{cases} e^{-j\omega}, & |\omega| \le \frac{\pi}{4} \\ 0, & \text{otherwise} \end{cases}$$

Design the filter using hamming window.

(b) Given an 8 point sequence $x(n) = n + 1$ where n = 0.1,2,…,7. Develop an FFT algorithm using decimation-in-frequency (DIF) approach. Discuss its advantages in terms of savings in complex additions and multiplication.

5. Attempt any **one** part : **1 × 10 = 10**

(a) The frequency response of an ideal bandpass filter is given by

$$H(\omega) = \begin{cases} 0 & |\omega| \le \frac{\pi}{8} \\ 1 & \frac{\pi}{8} < |\omega| < \frac{3\pi}{8} \\ 0 & \frac{3\pi}{8} \le |\omega| \le \pi \end{cases}$$

Determine its impulse response.

(b) Perform the convolution of the following two sequences h(n) and x(n)

$$h(n) = \begin{cases} \left(\frac{1}{2}\right)^n & 0 \le n \le 2 \\ 0 & \text{otherwise} \end{cases} \quad \text{and}$$

$$x(n) = \delta(n) + \delta(n-1) + 4\delta(n-2)$$

6. Attempt any **one** part : **1 × 10 = 10**

(a) Derive the condition of an FIR filter to give linear phase response. Also, find the frequency response of a given FIR filter, if the number of samples, N, in its impulse response h(n) is odd.

(b) Using bilinear transformation, design a Butterworth low pass filter to meet the following specifications :

$$0.8 \le |H(e^{j\omega})| \le 1 \qquad 0 \le |\omega| \le 0.2\pi$$

$$|H(e^{j\omega})| \le 0.2 \qquad 0.6\pi \le |\omega| \le \pi$$

7. Attempt any **two** parts : **2 × 5 = 10**

(a) Derive the frequency transformation for converting a prototype digital low pass filter into a digital highpass filter.

(b) Show that the zeros of a linear phase FIR filter occurs at reciprocal location.

(c) Show that the output data is in bit reversed order for the decimation in frequency algorithm for N = 8.
