USN

Sixth Semester B.E. Degree Examination, June/July 2019
Digital Signal Processing

Time: 3 hrs. Max. Marks:100

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

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. Determine the 8 point DFT of the Signal x(n) = {1, 1, 1, 0, 0, 0, 0, 0}. Also sketch its magnitude and phase. **(10 Marks)**

b. Given the periodic sequences x(n) = {1, 3, 5, 7}, h(n) = {2, 4, 6, 8}. Find their convolution using Stockman's method. **(10 Marks)**

2 a. The 5 samples of the 8 point DFT X(K) are X(0) = 0.25, X(1) = 0.125 – j 0.3018, X(6) = 0, X(4) = 0, X(5) = 0.125 – j 0.0518. Determine the remaining samples. **(04 Marks)**

b. Compute the N point DFT of $x(n) = a^n$ where $0 \le n \le N-1$. Hence determine the value of DFT for $x(n) = 0.5^n u(n)\ 0 \le n \le 3$. **(06 Marks)**

c. Determine the output of a linear FIR filter whose impulse response is h(n) = {1, 2, 3} and input signal x(n) = {1, 2, 3, 4, 5, 6, 7, 8, 9} using overlap save method. Use 5 point circular convolution in your approach. **(10 Marks)**

3 a. Develop decimation in Time FFT algorithm with all necessary steps and neat signal flow diagram used in computing N point DFT X(K) of a N point sequence x(n). **(10 Marks)**

b. Determine the DFT of the given sequence x(n) = {2, 1, 4, 6, 5, 8, 3, 9} by FFT algorithm which gives the output in bit reversal order. **(10 Marks)**

4 a. Find the Four point circular convolution of x(n) and h(n) using Radix x-2 DITFFT algorithm $x_1(n)$ = {2, 1, 1, 2} h(n) = {1, –1, –1, 1}. **(10 Marks)**

b. Develop a radix – 3 DITFFT algorithm for evaluating the DFT for N = 9 which accepts bit reversal I/P. **(10 Marks)**

PART – B

5 a. Design a Digital Butterworth digital high pass filter with the following specifications.

$$|H(e^{j\omega})| \le 0.2 \qquad 0 \le \omega \le 0.2\pi \quad T = 1\text{Sec}$$

$$0.8 \le |H(e^{j\omega})| \le 1 \qquad 0.6\pi \le \omega \le \pi$$

Using Impulse Invariant method. **(10 Marks)**

b. Convert the given analog transfer function into equivalent digital transfer function using bilinear transformation technique.

$$H(s) = \frac{1}{(s+1)(s+2)}.\ \text{Take } T = 1 \text{ sample/Sec.}$$ **(04 Marks)**

c. Determine the order of low pass filter if it has pass band attenuation of –3dB at 500Hz and stop band attenuation of –40dB at 1000 Hz. **(06 Marks)**

6 a. Explain Impulse Invariant transformation technique. **(10 Marks)**

b. Design Chebyshev high pass filter using bilinear transformation for the following specifications

$\omega_p = 0.2\pi \quad \omega_s = 0.01\pi$

$\alpha_p = -1dB \quad \alpha_s = -10dB$ **(10 Marks)**

7 a. Design an ideal low pass FIR filter whose desired frequency response is

$$H_d(e^{j\omega}) = \begin{cases} 1 & \frac{\pi}{3} \ge \omega \ge \frac{-\pi}{3} \\ 0 & \pi \ge |\omega| \ge \frac{\pi}{3} \end{cases}$$

Using Hamming window.

Determine the impulse response for N = 9. **(10 Marks)**

b. Design a high pass FIR filter whose desired frequency response is given by

$$H_d(e^{j\omega}) = \begin{cases} e^{-j\left(\frac{N-1}{2}\right)^{\omega}} & \pi \ge |\omega| \ge \frac{\pi}{2} \\ 0 & \pi > |\omega| \ge 0 \end{cases}$$

For N = 9 using Frequency Sampling technique. **(10 Marks)**

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8 a. Obtain the direct form I, direct form II and parallel form realization for the transfer function. $H(z) = \dfrac{8z^3 - 4z^2 + 11z - 2}{\left(z - \frac{1}{4}\right)\left(z^2 - z + \frac{1}{2}\right)}$ **(10 Marks)**

b. Realize the ladder structure for

$H(z) = \dfrac{2z^{-2} + 3z^{-1} + 1}{z^{-2} + z^{-1} + 1}$. **(05 Marks)**

c. Realize the FIR filter having impulse response

$h(n) = \delta(n) - \frac{1}{4}\delta(n-1) + \frac{1}{2}\delta(n-2) + \frac{1}{2}\delta(n-3) - \frac{1}{4}\delta(n-4) + \delta(n-5)$

Use minimum number of multipliers. **(05 Marks)**

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