USN

10EC52

Fifth Semester B.E. Degree Examination, Aug./Sept. 2020
Digital Signal Processing

Time: 3 hrs. Max. Marks:100

Note: *1. Answer any FIVE full questions, selecting at least TWO questions from each part.*
2. Use of Filter Tables are not permitted.

PART – A

1 a. Find the N – point DFT of the sequence x(n) interms of Cos function

$$x(n) = \begin{cases} \frac{1}{5}, & 0 \le n \le 2 \\ 0, & \text{otherwise} \end{cases}$$ **(06 Marks)**

b. Compute the 10-point DFT of the sequence

$$x(n) = \cos\left(\frac{2\pi n}{10}\right), 0 \le n \le 9.$$ **(06 Marks)**

c. Let a sequence $x(n) = \{\underset{\uparrow}{2}, 3, 2, 1\}$ and its DFT $x(k) = \{\underset{\uparrow}{8}, -j2, 0, j2\}$. Compute :

i) DFT of the 12-point signal described by $x_1(n) = \{x(n).x(n).x(n)\}$

ii) 12-point zero interpolated signal $h(n) = x\left(\frac{n}{3}\right)$. **(08 Marks)**

2 a. Let X(k) denotes a 6-point DFT of a sequence $x(n) = \{\underset{\uparrow}{1}, -1, 2, 3, 0, 0\}$ without computing the IDFT, determine the 6-point sequence g(n) whose 6-point DFT is given by $G(k) = W_3^{2k} X(k)$ **(06 Marks)**

b. Evaluate $y(n) = x(n) \circledast_8 h(n)$ for the sequences

$x(n) = e^{j\pi n}, 0 \le n \le 7$

$h(n) = u(n) - u(n-5)$. **(06 Marks)**

c. Give the 8-point sequence x(n) is $x(n) = \begin{cases} 1, & 0 \le n \le 3 \\ 0, & 4 \le n \le 7 \end{cases}$. Compute the DFT to the sequence

$$x_1(n) = \begin{cases} 1, & n = 0 \\ 0, & 1 \le n \le 4 \\ 1, & 5 \le n \le 7 \end{cases}$$. Use the suitable property of DFT. **(08 Marks)**

3 a. Find the output y(n) of a filter whose impulse response h(n) = { 1, –2, 1} and input signal x(n) = {3, 1, –2, 1, –1, 2, 4, 3, 6 }. Use a 8 - point circular convolution and also use over Lap-add method. **(08 Marks)**

b. Calculate the percentage saving in calculations in a 512-point radix – 2FFT, when compared to direct DFT. **(05 Marks)**

c. What is signal segmentation? Explain the procedure used for over Lap – save method. **(07 Marks)**

4 a. Develop DIF – FFT algorithm for N = 8 and draw the complete signal graph. Using this signal flow graph, compute the DFT of the sequence.
$x(n) = \{ 1, -1, 1, -1, 1, 0, 0, 0 \}$. **(14 Marks)**

b. Consider a finite length sequence $x(n) = \{\underset{\uparrow}{1}, 2, 3, 4, 5, 6\}$ find X(3) using Goertzel algorithm. Assume initial conditions are zero. **(06 Marks)**

PART – B

5 a. Explain Analog to Analog Frequency Transformation. **(05 Marks)**

b. What is Chebyshev polynomials and mention its properties. **(05 Marks)**

c. Find the order of a Low pass Butterworth filter to meet the following specifications.
$\delta_P = 0.001, \quad \delta_S = 0.001$
$\Omega_P = 1$ rad/sec, $\Omega_S = 2$ rad/sec **(05 Marks)**

d. What are the advantages and disadvantages of IIR Filters? **(05 Marks)**

6 a. Obtain Parallel form Realization of system Transfer function

$$H(z) = \frac{1 + \frac{1}{2}z^{-1}}{\left(1 - z^{-1} + \frac{1}{4}z^{-2}\right)\left(1 - z^{-1} + \frac{1}{2}z^{-2}\right)}.$$ **(10 Marks)**

b. What are the features of a FIR Lattice structure? **(05 Marks)**

c. Realize the following FIR system with minimum number of multipliers
$h(n) = \{ -0.5, 0.8, -0.5 \}$ **(05 Marks)**

7 a. A filter is to be designed with the following desired frequency response

$$H_d(e^{jw}) = \begin{cases} 0, & \frac{-\pi}{4} \le w \le \frac{\pi}{4} \\ e^{-j2w}, & \frac{\pi}{4} < |w| \le \pi \end{cases}$$

Determine the filter coefficient $h_d(n)$ if the window function is defined as

$$w(n) = \begin{cases} 1, & 0 \le n \le 4 \\ 0, & \text{otherwise} \end{cases}$$ **(10 Marks)**

b. Find the impulse response h(n) of a linear phase FIR filter of length = 4 for which the frequency response at w = 0 and $w = \frac{\pi}{2}$ is specified as

$H_r(0) = 1$ and $H_r\left(\frac{\pi}{2}\right) = \frac{1}{2}$ **(07 Marks)**

c. Mention the advantages of Window Technique. **(03 Marks)**

8 a. Design an IIR digital filter that when used in a prefilter A/D – H(z) – D/A structure, will satisfy the following analog specification of Chebyshev filter.
i) LPF with – 2dB cutoff at 100Hz
ii) Stopband attenuation of 20DdB or greater at 500Hz
iii) Sampling rate 4000 samples/sec **(14 Marks)**

b. Obtain the digital filter, equivalent of the analog filter shown in Fig Q8(b). Using impulse invariance method. Assume $f_s = 8f_c$, where f_c – cutoff frequencies of the filter.

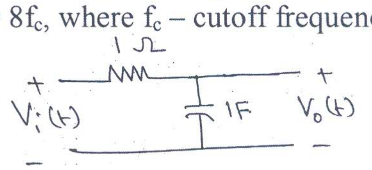

Fig Q8(b) **(06 Marks)**