USN [ ][ ][ ][ ][ ][ ][ ][ ][ ][ ]

10MT74

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

Time: 3 hrs. Max. Marks:100

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

PART – A

1 a. Obtain N-point DFT of an N-point sequence x(n).
i) $x(n) = a^n u(n)$ ii) $x(n) = \delta(n)$ **(10 Marks)**

b. Compute 4-point DFT of the sequence x(n) = {1, 1, 1, 1}. Also plot $|X(K)|$ and $\angle X(K)$ **(06 Marks)**

c. Derive relation between N-point DFT and Z-transform. **(04 Marks)**

2 a. State and prove periodicity and linearity properties of DFT. **(08 Marks)**

b. A long sequence x(n) is filtered through a filter with impulse response h(n) to yield the output y(n) if, x(n) = {1, 2, 3, 3, 2, 1, -1, -2, -3, 5, 6, -1, 2, 0, 2, 1} and h(n) = {3, 2, 1, 1}. Compute y(n) using overlap add method. Allow block length as '7'. **(12 Marks)**

3 a. Tabulate the number of complex multiplications and complex additions required for direct computation of DFT and FFT algorithms for N = 8, 16, 32. **(08 Marks)**

b. Compute circular convolution using concentric circle method:
$x_1(n) = \{-1, 1, 1, 1, -1, -1, -1, -1\}$
$x_2(n) = \{0, 1, 2, 3, 4, 3, 2, 1\}$ **(12 Marks)**

4 a. Derive the radix-2 DIT, FFT algorithm for N = 8 and draw the signal flow graph. **(08 Marks)**

b. If x(n) = {1, 2, 3, 4, 1, 2, 2, 1}, compute DFT of x(n) using DIF-FFT algorithm. **(12 Marks)**

PART – B

5 a. Explain the design of IIR filter by solution of differential equations. [Use equivalent difference equation]. **(10 Marks)**

b. Use impulse invariance method to design a digital filter from an analog prototype that has a system function $H_a(s) = \dfrac{s+a}{(s+a)^2 + b^2}$. **(10 Marks)**

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

b. Design a lowpass filter with a cut-off frequency $\omega_c = \pi/4$, a transition width $\Delta\omega = 0.02\pi$ and a stopband ripple $\delta_s = 0.01$. Use Kaiser for your design. **(10 Marks)**

7 a. 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. **(12 Marks)**

b. The system function of the first order lowpass butter worth filter is given as:

$$H_a(s) = \frac{\Omega_c}{s+\Omega_c}$$

Here Ω_c is the 3-dB cutoff frequency of the analog filter. Apply bilinear transformation to this filter such that the digital filter will have 3-dB frequency of 0.2π. **(08 Marks)**

8 a. Explain the cascade form realization of FIR filters. **(10 Marks)**

b. Explain the cascade form realization of IIR filters. **(10 Marks)**

* * * * *