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15EC52

Fifth Semester B.E. Degree Examination, Feb./Mar. 2022 Digital Signal Processing

Time: 3 hrs.

Max. Marks: 80

Note: 1. Answer any FIVE full questions, choosing ONE full question from each module.

2. Use of Normalized filter tables not permitted.

Module-1

- a. Describe the process of frequency domain sampling and reconstruction of discrete time signals. (06 Marks)
 - b. Compute 8-point DFT of $x(n) = \{1, 1, 1\}$, also sketch magnitude and phase plot. (10 Marks)

OR

2 a. Derive the Relationship of DFT with Z-transform.

(04 Marks)

b. State and prove circular time shift property of DFT.

(04 Marks)

c. Compute circular convolution of $x(n) = \{1, 2, 3, 4\}$ and $h(n) = \{1, 2, 2\}$ using transform domain approach. (08 Marks)

Module-2

- 3 a. Find the output y(n) of a filter whose impulse response $h(n) = \{1, -2\}$ and input $x(n) = \{3, -2, 4, 1, 5, 7, 2, -9\}$ using overlap add method. Use 5-point circular convolution in your approach. (08 Marks)
 - b. Determine N-point circular correlation of $x(n) = \cos\left(\frac{2\pi n}{N}\right)$ and $y(n) = \sin\left(\frac{2\pi n}{N}\right)$ (08 Marks)

OR

4 a. State and prove Parseval's theorem of DFT.

(04 Marks)

b. Explain the linear filtering of long data sequence using overlap-save method.

(08 Marks)

c. State and prove properties of twiddle factor.

(04 Marks)

Module-3

- 5 a. Develop DIT-FFT algorithm to compute DFT of a sequence and obtain the signal flow diagram for N = 8. (12 Marks)
 - b. Compute 4-point IDFT of $X(K) = \{6, (-1 j), 0, (-1 + j)\}$ using DIT-FFT algorithm.

(04 Marks)

OR

- 6 a. Compute 8-point DFT of $x(n)=\{1, 2, 3, 4, 5, 6, 7, 8\}$ using DIF-FFT algorithm. (08 Marks)
 - b. Explain Geortzal algorithm for computation of DFT.

(08 Marks)

Module-4

- 7 a. Obtain DF-II and parallel realization of $H(z) = \frac{1+z^{-1}}{\left(1-\frac{1}{4}z^{-1}\right)\left(1-z^{-1}+\frac{1}{2}z^{-2}\right)}$. (08 Marks)
 - b. Derive the expression for order and cutoff frequency for a lowpass Butterworth filter.

(08 Marks)

OR

- Design a digital filter H(z) that when used in A/D H(z) D/A structure gives an equivalent analog filter with the following specifications: passband attenuation of 3 dB at 500 Hz, stopband attenuation of 15 dB at 750 Hz with sampling rate 2 kHz. The filter is to be designed by performing a BLT on an analog system function. Use Butterworth prototype. (10 Marks) Also obtain the difference equation.
 - b. Explain how an analog filter is mapped on to a digital filter using impulse invariance (06 Marks) method. What are the limitations of the method?

- (08 Marks)
 - A FIR filter is described by $y(n) = x(n) + \frac{2}{5}x(n-1) + \frac{3}{4}x(n-2) + \frac{1}{3}x(n-3)$. Draw its (08 Marks) Lattice structure.

Design a LPF with the frequency response

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

using rectangular window, also find its impulse response, frequency response and difference

b. Realize the linear phase FIR filter having the impulse response $h(n) = \left\{1, \frac{1}{4}, -\frac{1}{8}, \frac{1}{4}, 1\right\}$.