M.E. First Semester (Digital Electronics) (Part Time / Full Time) (C.G.S.- New)

13202 : Advanced Digital Signal Processing : 1 UMEF 2

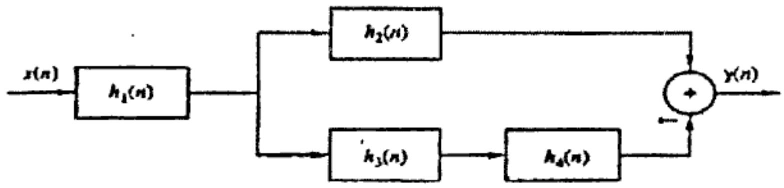
P. Pages : 2 | **AW - 3756**

Time : Three Hours | Max. Marks : 80

Notes : 1. Due credit will be given to neatness and adequate dimensions.
2. Assume suitable data wherever necessary.
3. Illustrate your answer necessary with the help of neat sketches.
4. Use of non programmable calculators is permitted.

1. a) Express the overall impulse response in terms of $h_1(n)$, $h_2(n)$, $h_3(n)$ and $h_4(n)$. **8**

b) Determine h(n) when of $h_1(n) = \{1/2, 1/4, 1/2\}$, $h_2(n) = h_3(n) = (n+1)u(n)$ and $h_4(n) = \delta(n-2)$ **6**



b) State and prove any four properties of DFT.

OR

2. a) Compute the eight-point DFT of the sequence x(n) = {½, ½, ½, ½, 0, 0, 0, 0) using the in-place radix-2 decimation-in-frequency algorithms. Follow exactly the corresponding signal flow graphs and keep track of all the intermediate quantities by putting them on the diagrams. **7**

b) Determine the response of filter by the method of DFT x(n) = {2, 1) and h(n) = {1, 2}. **6**

3. a) Determine the coefficients h(n) of a linear phase FIR filter of length M = 15 which has a symmetric unit sample response and frequency response that satisfies the conditions. **7**

$$H(2\pi k/15) = \begin{cases} 1, & \text{for } k = 0, 1, 2, 3 \\ 0.4, & \text{for } k = 4 \\ 0, & \text{for } k = 5, 6, 7 \end{cases}$$

b) What is finite word length effect in digital filters? What are the different types of finite word length effects that occur in digital filters? Explain how it can be minimized. **7**

OR

4. a) Explain Fourier series method of designing an FIR Filter. Hence explain Gibbs phenomenon. Also lay down the step wise procedure for designing FIR filter using Fourier Series Method. **6**

b) Find the pole and Zero location of an analog Chebyshev type-II filter for the following digital filter specifications. Use bilinear transformation. **7**

$$-1 \le \left|H(e^{j\omega})\right| dB \le 0; \qquad 0 \le |\omega| \le 0.2\pi$$

$$\left|H(e^{j\omega})\right| dB \le -20; \qquad |\omega| \ge 0.3\pi$$

5.	a)	A sequence x(n) is up-sampled by $I = 2$, it passes through an LTI system $H_1(z)$, and then it is down-sampled by D=2. Can we replace this process with a single LTI system with $H_2(z)$? If the answer is positive, determine the system function of this system.	7
	b)	Explain Decimation and Interpolation process with example. Also obtain the spectrum of down sampled and up sampled signal.	6

OR

6.	a)	The prototype filter in a four channel uniform DFT filter bank is characterized by the transfer function $H_0(z) = 1 + z^{-1} + 3z^{-2} + 4z^{-4}$ i) Determine the transfer function of the filters $H_1(z), H_2(z)$ & $H_3(z)$ in the analysis section. ii) Determine the transfer function of the filters in the synthesis section. iii) Sketch the analysis and synthesis sections of the uniform DFT filter bank	7
	b)	Explain polyphase decomposition process. Hence obtain FIR & IIR polyphase filter structures for decimator and interpolator.	6
7.	a)	Explain the basic idea of Steepest Descent method and hence explain how it is used to minimize the cost function. Also discuss the stability of Steepest Descent algorithm with appropriate signal flow graph.	7
	b)	State matrix inversion lemma and hence derive the recursive least square adaptive filtering algorithm. Also give the summary of RLS algorithm.	7

OR

8.	a)	Explain in detail the principle of orthogonality applied to the adaptive filtering, hence derive Wiener-Hopf equation from the principle of orthogonality.	7
	b)	Explain Least Mean Square adaptive filtering algorithm in detail. Also, discuss the stability constraint, convergence speed and excess MSE related to Least Mean Square adaptive filtering algorithm.	7
9.	a)	What is STFT? what are its advantages over Fourier transform. State and prove properties of STFT.	6
	b)	What is discrete Hilbert transform? State and prove the different properties of Hilbert transform.	7

OR

10.	a)	Show that DCT is an orthogonal transform.	6
	b)	What is Gabor transform? List the advantages and applications of Gabor transform. Explain any one application of Gabor transform in detail.	7
11.	a)	Draw & explain the Internal memory Architecture of DSP Processor TMS320C67XX.	7
	b)	Explain the application of digital signal processor in biomedical signal processing with neat block diagram and associated waveforms.	6

OR

12.	a)	Explain the various registers, pointers and interrupts of TMS 320C6713 DSP.	6
	b)	Discuss various issues involved in DSP Processor design. Also state basic requirement of implantation of smallest DFT using DSP processor as an instruction.	7

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