Exam.Code:1005
Sub. Code: 7694

1129

M. E. (Information Technology)

First Semester

MEIT-103/113: Advanced Digital Signal Processing

Time allowed: 3 Hours **Max. Marks: 50**

NOTE: *Attempt five questions in all, including Question No. I which is compulsory and selecting two questions from each Unit.*

x-x-x

I. Attempt the following:-

a) Discuss two applications of adaptive filter?

b) What is the role of Digital Filter Banks in DSP?

c) What is the significance of Nyquist Sampling criteria?

d) Name different types of windows used in FIR filter design?

e) Why sampling rate conversion is done in multirate DSP? (5x2)

UNIT – I

II. a) Describe the important properties of region of convergence in z transform covering finite and infinite duration signals? How is z transform obtained from Laplace transform?

b) Design a FIR low pass filter with a cut off frequency 1 Khz and sampling rate 4khz with eleven samples using Fourier series. (4,6)

III. a) Derive the DFT of the sample data sequence x(n)={1,2,3,4,5,6} and compute corresponding amplitude and phase spectrum.

b) Find IDFT of X(k)= {3,(2+j), 1 ,(2-j)} (6,4)

IV. a) Derive the expression for Frequency Response of Digital FIR Filter? Explain why FIR filter is more stable than IIR filter?

b) Find the response of an FIR filter with impulse response H(n)= {1,2,4} to the input sequence x(n)= {1,2} by using circular convolution method?

UNIT – II

V. a) Draw the structures of cascade and parallel realizations of $H(z) = \frac{(1-z^{-1})^3}{[(1-(1/2)z^{-1}][1-(1/8)z^{-1}]}$

b) Derive mathematically the expressions for Decimation and Interpolation in multirate digital signal processing? How aliasing effect is reduced by doing sampling rate conversion? (2x5)

P.T.O.

VI. a) Draw the Direct Form representation of M to 1 Decimator using polyphase decomposition using Type I and Direct Form representation of 1 to L Interpolator for Type II Poyphase Decomposition.

b) Write a short note on Uniform DFT filter banks in multirate signal processing. (6,4)

VII. a) Explain with the principle of adaptive Direct Form FIR filter ,how system modeling and adaptive equalization is possible?

b) Describe in detail the Widrow LMS Algorithm mathematically? (2x5)

x-x-x