Exam.Code:0925
Sub. Code: 6541

2122

B. E. (Information Technology)
Seventh Semester
IT-701: Digital Signal Processing

Time allowed: 3 Hours Max. Marks: 50

***NOTE**: Attempt <u>five</u> 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) What are energy and power signals?

b) An audio signal is in the range of 15- 30 kHz. The signal is to be digitized for the purpose of transmission. What will be the sampling rate used?

c) State two applications of Digital Signal Processing.

d) State two differences between FIR and IIR filters.

e) Write the condition for the digital filter to be causal.

f) The two poles of a digital filter are located on the imaginary axis. Is that digital filter stable, unstable or critically stable?

g) Calculate Fourier transform of an impulse signal located at origin.

h) Fourier series can be calculated for a periodic signal. True/False?

i) What are the basic elements required for realizing a digital filters?

j) Invariance technique (10x1)

UNIT - I

II. Given an input sequence x(n) = {1,2,3} is passed through a filter given by h(n) = {1,2}. Find the output y(n). (10)

III. Sketch the lattice ladder structure for H (z)

$$H(z)=\frac{1-0.8z^{-1}+0.15z^{-2}}{1+0.1z^{-1}-0.72z^{-2}}$$ (10)

IV. Given input x(n)= (1,2,3,0) and system function h(n)= (1,2,0,0). Use FFT method to calculate output y(n), using DIT algorithm for FFT. (10)

P.T.O.

UNIT - II

V. Design a low pass filter for the following specifications.

$$H_d(w) = \begin{cases} e - j2\omega, & |\omega| \le \frac{\pi}{4} \\ 0, & \frac{\pi}{4} \le |\omega| \le \pi \end{cases} \quad (10)$$

VI. Explain impulse invariant method of IIR filter design. An analog filter has the following system function. Convert this filter into a digital filter using backward difference for the derivative.

$$H(s) = \frac{1}{(s+0.1)^2 + 0.9} \quad (10)$$

VII. Explain the architecture and addressing modes of ADSP 21XX processor. (10)

x-x-x