B.E. Sixth Semester (Electronics / Electronics Telecommunication / Electronics Communication Engineering) (C.B.S.)

Digital Signal Processing

P. Pages : 3 **KNT/KW/16/7380/7385**

Time : Three Hours Max. Marks : 80

Notes :
1. All questions carry marks as indicated.
2. Solve Question 1 OR Questions No. 2.
3. Solve Question 3 OR Questions No. 4.
4. Solve Question 5 OR Questions No. 6.
5. Solve Question 7 OR Questions No. 8.
6. Solve Question 9 OR Questions No. 10.
7. Solve Question 11 OR Questions No. 12.
8. Due credit will be given to neatness and adequate dimensions.
9. Assume suitable data whenever necessary.
10. Illustrate your answers whenever necessary with the help of neat sketches.
11. Use of non programmable calculator is permitted.

1. a) Consider the analog signal. **6**

$$xa(t)=3\cos 2000\pi t+5\sin 6000\pi t+10\cos 12{,}000\,\pi t$$

i) What is the Nyquist rate for this signal?

ii) Assume that we sample this signal using a sampling rate F_S = 5000 samples/sec. What is the discrete-time signal obtained after sampling?

iii) What is the analog signal $y_a(t)$ that can reconstruct from the samples if we use ideal interpolation?

b) The impulse response of linear time-invariant system is $h(n)=\{1,\underset{\uparrow}{2},1,-1\}$. Determine linear convolution to the input signal $x(n)=\{\underset{\uparrow}{1},2,3,1\}$ using graphical method. **7**

OR

2. a) Determine energy and power of the signal. **6**

$$x(n)=\left(\frac{1}{3}\right)^n \mu(n).$$

b) Determine whether the system y (n) = 2.x (-n) is **7**

i) Static or Dyamic

ii) Linear or Non-linear

iii) Time variant or time invariant

iv) Causal or Non-causal.

3. a) Determine the Z-transform of the signal **7**

$$x(n)=\left(\frac{1}{2}\right)^{n}\mu(n).$$

Also sketch region of convergence (ROC).

b) State and prove any three properties of the Z-transform. **6**

OR

4. a) Determine the inverse z-transform of **7**

$$x(z)=\frac{1}{1-1\cdot 5\,z^{-1}+0.5z^{-2}}$$

when

i) ROC: $|z|>1$

ii) ROC: $|z|<0\cdot 5$

b) Find Y(n) using unilateral z-transform **6**

$y(n)+\frac{1}{2}(y-1)=0$ Given that

y (-1) = 1

5. Find circular convolution of the following sequences using DFT-IDFT. **14**

$x_1(n)=\{2,1,2,1\},\ x_2(n)=\{1,2,3,4\}$.

OR

6. a) State any two properties of DFT. **6**

b) Compute 4 point DFT using DIT-FFT. **8**

x (n) = {0, 1, 2, 3}

7. The specification of LFF is **13**

$0.8\le\ |H(\omega)|\le 1 \quad \text{for } 0\le\omega\le 0\cdot 2\pi$

$|H(\omega)|\le 0\cdot 2 \quad \text{for } 0\cdot 32\pi\le\omega\le\pi$

Design Butterworth digital filter using impulse invariance method.

OR

8. Draw DF-I, DF-II, cascade and parallel structures for the system function. **13**

$$H(Z)=\frac{1+\frac{1}{3}z^{-1}}{\left(1-\frac{1}{2}z^{-1}+\frac{1}{3}z^{-2}\right)\left(1+\frac{1}{4}z^{-1}\right)}$$

9. Sketch the block diagram for frequency sampling realization of the M=32, $\alpha = 0$ linear phase (symmetric) FIR filter which has frequency samples. **14**

$$H\left(\frac{2\pi k}{32}\right) = \begin{cases} 1, & k = 0,1,2 \\ \frac{1}{2}, & k = 3 \\ 0, & k = 4,5,\ldots15. \end{cases}$$

OR

10. Determine the coefficient of a linear phase FIR filter of length M=15 which has a symmetric unit sample response and a frequency response that satisfies the conditions. **14**

$$Hr\left(\frac{2\pi k}{15}\right) = \begin{cases} 1, & k = 0,1,2,3 \\ 0\cdot 4, & k = 4 \\ 0, & k = 5,6,7 \end{cases}$$

11. a) Explain what do you mean by Multirate converter ? Also explain what do you mean by Decimation and Interpolation. **7**

b) Obtain the output signal Y(n) from the input signal x(n) as shown in fig Q. 11 (a). **6**

$$x(n) = \{\underset{\uparrow}{1}, 2, 3, 4, 5, 6, 7, 8, 9\}$$

$$x(n) \longrightarrow \boxed{\uparrow I = 2} \longrightarrow \boxed{\downarrow D = 3} \longrightarrow y(n)$$

fig. Q. 11 (b)

OR

12. a) Explain with block diagram the sampling rate conversion to change the sampling rate by rational factor I/D. **7**

b) Write the technical notes on "subband coding of speech signals"? **6**

****