Reg. No. :

Question Paper Code : X 60396

B.E./B.Tech. DEGREE EXAMINATIONS, NOVEMBER/DECEMBER 2020
Seventh Semester
Computer Science and Engineering
CS 2403 – DIGITAL SIGNAL PROCESSING
(Common to Fifth Semester – Information Technology)
(Regulations 2008)
(Also Common to PTCS 2403 – Digital Signal Processing for B.E. (Part-Time)
Sixth Semester – Computer Science and Engineering – Regulations 2009)

Time : Three Hours | Maximum : 100 Marks

Answer ALL questions

PART – A **(10×2=20 Marks)**

1. A discrete-time signal $x(n) = \left\{-2, -1, 0, \underset{\uparrow}{1}, -1, 1\right\}$ is multiplied by u(–n – 2). What is the resulting signal ?

2. What is a shift-invariant system ? Give an example.

3. What is meant by radix 2 FFT ?

4. Give transform pair equation of DFT.

5. Compare bilinear and impulse invariant transformation.

6. What is aliasing ?

7. What are Gibbs oscillations ?

8. Distinguish between FIR and IIR filters.

9. Write the main application areas of speech coding.

10. What is adaptive filter ?

PART – B **(5×16=80 Marks)**

11. a) i) Consider the analog signal
$x_a(t) = 3 \cos 2000 \pi t + 5 \sin 6000 \pi t + 10 \cos 12000 \pi t$.
1) What is the Nyquist rate for this signal ?
2) Assume now that we sample this signal using a sampling rate $F_s = 5000$ samples/s. What is the discrete time signal obtained after sampling.
3) What is the analog signal $y_a(t)$ that we can reconstruct from the samples if we use ideal interpolation ? **(8)**

ii) Derive the equation for convolution sum and summarize the steps involved in computing convolution. **(8)**

(OR)

b) i) Determine the z transform and ROC of the signal $x(n) = -\alpha^n u(-n-1)$. **(6)**

ii) Check whether the discrete time system $y(n) = \cos[x(n)]$ is
1) Static or dynamic
2) Linear or nonlinear
3) Time invariant or time varying
4) Causal or non-causal
5) Stable or unstable. **(10)**

12. a) i) The input x(n) and impulse response h(n) of a system are given by
$x(n) = \left\{-\underset{\uparrow}{1}, 1, 2, -2\right\}; h(n) = \left\{0.5, \underset{\uparrow}{1}, -1, 2, 0.75\right\}$
Determine the response of the system using DFT. **(10)**

ii) State and prove convolution property of DFT. **(6)**

(OR)

b) Compute the FFT of the sequence $x(n) = n^2 + 1$ for $0 \le n \le N-1$, where $N = 8$ using DIT algorithm. **(16)**

13. a) i) Design an analog Butterworth filter that has a –2dB passband attenuation at a frequency of 20 rad/sec and atleast –10dB stopband attenuation at 30 rad/sec. **(10)**

ii) Explain the steps of design of digital filters from analog filters. **(6)**

(OR)

b) i) Using the bilinear transform, design a high pass filter, monotonic in passband with cutoff frequency of 1000 Hz and down 10 dB at 350 Hz. The sampling frequency is 5000 Hz. **(10)**

ii) Explain the methods of realization of digital filters. **(6)**

14. a) Prove that an FIR filter has linear phase if the unit sample response satisfies the condition $h(n) = h(N - 1 - n)$. Also discuss symmetric and anti symmetric cases of FIR filter when N is even. **(16)**

(OR)

b) Explain in detail about Finite word length effects in digital filters. **(16)**

15. a) i) Discuss about multi rate signal processing. **(8)**

ii) Write short note on speech compression. **(8)**

(OR)

b) Discuss the role of DSP in image enhancement with an example. **(16)**