

B.Tech IV Year I Semester (R13) Supplementary Examinations June 2017

DIGITAL SIGNAL PROCESSING
(Electrical and Electronics Engineering)

Time: 3 hours

Max. Marks: 70

PART – A
(Compulsory Question)

1 Answer the following: (10 X 02 = 20 Marks)

- (a) Define unit impulse and unit step signals.
- (b) Sketch the following signals: $x(t) = r(-t+2)$.
- (c) Write down DFT pair of equations.
- (d) What are the computational saving (both complex multiplication and complex addition) using N point FFT algorithm.
- (e) Draw the Direct form I structure of the FIR filter.
- (f) State the condition for a digital filter to be causal and stable.
- (g) What are the limitations of impulse invariance method?
- (h) Explain the advantage and drawback of Bilinear transformation.
- (i) What is Multirate digital signal processing?
- (j) Write the different applications of Multirate DSP.

PART – B

(Answer all five units, 5 X 10 = 50 Marks)

UNIT – I

- 2 (a) Explain, how linear convolution of two finite sequences are obtained via DFT.
- (b) Find the Z-transform of the following sequences : $x(n) = (0.5)^n u(n) + u(n-1)$.

OR

- 3 (a) By means of the DFT and IDFT, determine the response of the FIR filter with impulse response $h(n) = \{1, 2, 3\}$ to the input sequence $x(n) = \{1, 2, 2, 1\}$.
- (b) State and prove any two properties of DFT.

UNIT – II

- 4 Compute the eight point DFT of the given sequence $x(n) = \{ \frac{1}{2}, \frac{1}{2}, \frac{1}{2}, \frac{1}{2}, 0, 0, 0, 0 \}$ using radix-2 DIT-DFT algorithm.

OR

- 5 Find DFT for $\{1, 1, 2, 0, 1, 2, 0, 1\}$ using FFT DIT butterfly algorithm and plot the spectrum.

UNIT – III

- 6 Draw the direct form implementation of the FIR system having difference equation:
 $y(n) = x(n) - 2x(n-1) + 3x(n-2) - 10x(n-6)$

OR

- 7 Draw two different FIR structures for the $H(z)$ given below: $H(z) = (1 + 5z^{-1} + 6z^{-2})(1 + z^{-1})$.

UNIT – IV

- 8 Using a rectangular window technique, design LPF with band pass gain of unity, cutoff frequency 1 kHz and sampling frequency 5 kHz. The length of the impulse response is 7.

OR

- 9 Convert the analog filter into a digital filter whose system function is $H(S) = \frac{S+0.2}{(S+0.2)^2+9}$. Use the impulse invariant technique. Assume $T = 1s$.

UNIT – V

- 10 Explain the poly phase decomposition for:

- (a) FIR filter structure
- (b) IIR filter structure

OR

- 11 Explain in detail about Interpolation and decimation with examples.
