G 1199 (Pages : 3) Reg. No.................................

Name.................................

B.TECH. DEGREE EXAMINATION, MAY 2015

Sixth Semester

Branch : Applied Electronics and Instrumentation/Electronics and Communication/ Electronics and Instrumentation Engineering

AI 010 602/EC 010 602/EI 010 602—DIGITAL SIGNAL PROCESSING (AI, EC, EI)

(New Scheme—2010 Admission onwards)

[Regular/Improvement/Supplementary]

Time : Three Hours Maximum : 100 Marks

Part A

Answer **all** *questions.*
Each question carries 3 *marks.*

1. Explain the key benefits of Digital Signal Processing.
2. Determine the systems described by the following equations are linear or non-linear
   (a) $y(n) = nx(n)$
   (b) $y(n) = x^2(n)$.
3. Explain two methods in IIR filter design for mapping of transfer function from s domain to z domain.
4. Determine the direct form II of the LTI system described by the difference equation
   $y(n) = \frac{1}{4} y(n-2) + x(n)$.
5. Let X (k) be N point DFT of a N-point sequence $x(n)$, show that $x^*(n) = X^*(N-k)$.

(5 × 3 = 15 marks)

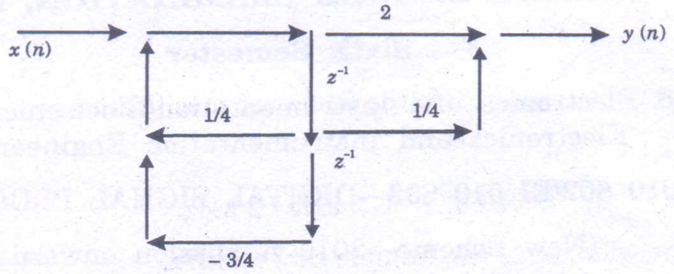
Part B

Answer **all** *questions.*
Each question carries 5 *marks.*

6. State the sampling theorem. Also write down the reconstruction formula. For the analog signal $x(t) = 3\cos 50\pi t + 10\sin 300\pi t - \cos 200\pi t$ find the minimum rate of sampling.

Turn over

7. Obtain the system function H (z) of the discrete system shown :



8. Compare IIR and FIR filters. State an application where FIR filters are preferred ?

9. Explain quantization and round off effects in digital filters.

10. Find the 4 point DFT of the sequence $x[n] = [2, 1, 2, 1]$.

(5 × 5 = 25 marks)

Part C

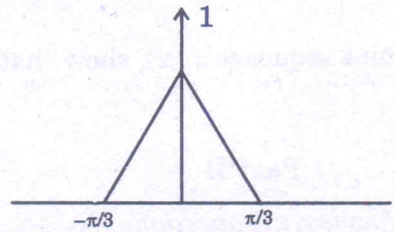
Answer **all** *questions.*
Each question carries 12 *marks.*

11. (a) State and prove Parseval's relation in the case of DTFT.

(b) Obtain DTFT of the sequence $y[n] = n\alpha^n u[n], |\alpha| < 1$.

Or

12. For a sequence $x[n]$ with spectrum shown below down sampled by a factor 3. Express the resultant sequence in terms of $x(n)$ and draw its spectrum.



13. The transfer function of a LTI system is given by $H(z) = \dfrac{1}{1 - \frac{11}{2}z^{-1} + \frac{17}{2}z^{-2} - 3z^{-3}}$.

Find the impulse response if the system is causal and stable.

Or

14. Explain four types of linear phase systems.

15. Obtain the direct form structure for the following linear phase filter with system function

$H(z) = 1 + 2.88\,z^{-1} + 3.4048z^{-2} + 1.7\,z^{-3} + 0.4\,z^{-4}$.

Or

16. Design a digital Butterworth filter with the following specifications :

Pass band ripple : ≤ 0.5 dB.

Pass Band Edge / Stop Band Edge : 1. 2 kHz / 2 kHz.

Stop band Attenuation : ≥ 40 dB, Sample rate : 8 kHz.

17. Convert the analog filter with system function $H(s) = \dfrac{s + 0.1}{(s + 0.1)^2 + 9}$ into digital IIR filter by impulse invariance method.

Or

18. Design a FIR filter with Kaiser window to meet the following specifications :

$\left|H\left(e^{j\omega}\right)\right| \geq 1, \qquad 0 \leq w \leq 0.25\pi$

$\left|H\left(e^{j\omega}\right)\right| \leq 0.01, \qquad 0.35\,\pi \leq w \leq \pi.$

(a) Determine the minimum length M + 1 of the impulse response and the value of the Kaiser window parameter for the filter.

(b) Find the delay of the filter.

19. Explain the DIT-FFT algorithm. Find the DFT of the sequence $x(n) = \{0, 1, 2, 0, 1, 2, 3, 1\}$ using radix 2 FFT algorithm.

Or

20. Given the sequences $x_1(n) = \{1, 2, 1, 2\}$ and $x_2(n) = \{1, -1, 1, -1\}$. Compute :

(a) The circular convolution $x_1(n) \circledcirc x_2(n)$

(b) Linear convolution using DFT.

(5 × 12 = 60 marks)