

7E7072

Roll No.

Total No of Pages: 2

## 7E7072

B. Tech. VII Sem. (Main/Back) Exam., Nov.-Dec.-2016 Electronics & Communication Engineering 7EI2A Digital Signal Processing

Time: 3 Hours

Maximum Marks: 80

Min. Passing Marks Main: 26

Min. Passing Marks Back: 24

#### Instructions to Candidates:

Attempt any five questions, selecting one question from each unit. All questions carry equal marks. Schematic diagrams must be shown wherever necessary. Any data you feel missing suitably be assumed and stated clearly.

Units of quantities used/calculated must be stated clearly.

Use of following supporting material is permitted during examination. (Mentioned in form No. 205)

1. NIL

2. NIL

## UNIT-I

Q.1 (a) Explain Energy & Power Signals with help of suitable examples.

[8]

(b) Explain Sampling & different types of Sampling Techniques.

[8]

...

<u>OR</u>

- Q.1 (a) Explain in detail the concept of continuous time processing of discrete time signals. [8]
  - (b) Explain the properties & applications of discrete time signals.

[8]

# <u>UNIT - II</u>

Q.2 (a) For the Transform analysis of LTI System, explain the phenomena of all pass System. [8]

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(b) Find the magnitude & phase responses for the system characterized by difference

equation - 
$$y(n) = \frac{1}{6}x(n) + \frac{1}{3}x(n-1) + \frac{1}{6}x(n-2)$$

[8]

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[7060].



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### OR [8] Explain Linear System with Linear phase. Q.2 (a) [8] Discuss briefly the frequency response of LTI system. UNIT - III Q.3 Obtain the structures of cascade & parallel realization of following transfer $H(z) = \frac{(1-z^{-1})^3}{\left(1-\frac{1}{2}z^{-1}\right)\left(1-\frac{1}{8}z^{-1}\right)}$ [16] function -OR [8] Explain filter & their use in DSP. Q.3 (a) [8] Compare the structures for IIR & FIR System. (b) UNIT - IV For design of FIR filters by windowing, explain Hamming & Kaiser. [8] Q.4 (a) Convert the analog filter into a digital filter whose system function is-(b) $H(S) = \frac{S + 0.2}{(S + 0.2)^2 + 9}$ Use the impulse invariant technique. Assume T= 1s. [8] Convert the analog filter with system function $H(S) = \frac{S + 0.1}{(S + 0.1)^2 + 9}$ into a digital Q.4 (a) IIR filter using bilinear transformation. The digital filter should have a resonant [8] frequency of $Wr = \frac{\pi}{n}$ ? [8] Explain briefly the Filter Design Techniques. UNIT - V Explain how DFT can be used as a linear transformation tool in Digital Signal Q.5 (a) [8] Processing. [8] Explain various properties of the DFT in brief. OR Q.5 Explain & find N-point DFT of following sequence $h(n) = \begin{cases} \frac{1}{3} & \text{for } 0 \le n \le 2\\ 0 & \text{else where} \end{cases}$ [7060]

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