[Total No. of Printed Pages:1] **CODE NO:- U-212**

FACULTY OF ENGINEERING

TE(EC/ECT/IE/E&C)Examination - DEC – 2014

DIGITAL SIGNAL PROCESSING(Revised)

[Time: THREE *Hours*] **[Max. Marks: 80]**

"Please check whether you have got the right question paper."

I) Q.no.1 and q no.6 are compulsory.
II) Solve any two questions from q. no.2 to Q.no.5
III) Solve any two questions from Q.No.7 to Q.No.10
IV) Figures to the right indicate full marks.
V) Assume suitable data, if necessary and state it clearly.

SECTION A

Q 1 Attempt any two from the following. (10)

a) Relation between Fourier transform and z-transform. b) Limitations of DSP
c) Linear convolution Vs circular convolution. d) Convolution property of z- transforms.

Q2 a) find the response of LTI system to the input $x(n)=\{1,-1,\underset{\uparrow}{-3},2\}$ and its impulse response is $h(n)=\{\underset{\uparrow}{1},-2,-3,4\}$ (07)

b) Derive an expression for Fourier transform of discrete time signal. Find the FT of the signal $x(n) = (1/4)^n u(n+4)$. (08)

Q3 a) Determine the z-transform of the signals. i) $x(n)=\propto^n u(n)$ ii) $x(n)=n\, a^n \sin(wn)$ (08)

b) Explain the properties of z- transform (07)

Q4 a) explain decimation in frequency (DIF) FFT algorithm. (08)

b) Find the inverse DFT of x(K)={1,0,1,0} (07)

Q5 a) Compute 8-point circular convolution of the following signals $x_1(n)=\{1,1,1,1,0,0,0,0\}$ $x_2(n)=\sin(3\pi n/8)$ $o \leq n \leq 7$ (08)

b) Explain test convolution techniques. (07)

SECTION-B

Q6 Answer any two from the following (10)

i) Frequency response of chebyshev type-I and type-II filter.
ii) Advantages of FIR over IIR filter.
iii) Quantization process.
iv) Approximation of derivatives method for the design of IIR digital filter.

Q7 a) Apply the impulse invariance method to obtain the digital filter from the second order amalog filter. $Ha(s)=\frac{s+a}{(s+a)^2+b^2}$ (07)

b) What is warping effect? What is its effect on magnitude and phase response (08)

Q8 a) Explain the rectangular window for FIR fitter design. (07)

b) Design a low pass FIR filter of length 11 for the following specifications (08)

i) Passband frequency=0.2 KHz
ii) sampling frequency= 1KHz by using Hamming window

Q9 a) Explain quantization effect- in the computation of DFT. (07)

b) Explain limit cycle in recursive system. (08)

Q10 a) Describe the steps in designing FIR filter by frequency sampling method. (07)

b) Realize the system given by difference equation (08)
Y (n) =-0.1y (n-1) +0.72 y (n-2) +0.7x (n)-0.25x (n-2) in IIR parallel form structure.