Reg No :_______________ Name:___________________________

APJ ABDUL KALAM TECHNOLOGICAL UNIVERSITY

FIFTH SEMESTER B.TECH DEGREE EXAMINATION, DECEMBER 2018

Course Code: EC301

Course Name: DIGITAL SIGNAL PROCESSING

Max. Marks: 100 Duration: 3 Hours

PART A

Answer any two full questions, each carries 15 marks. Marks

1 a) Given x(n) = { 1, -2, 3, -4, 5, -6 } without calculating DFT find the following quantities? (5)

a) X(0) b) $\sum_{K=0}^{5} X(K)$ c) X(3) d)$\sum_{K=0}^{5} |X(K)|^2$ e) $\sum_{K=0}^{5} -1^K X(K)$

b) Find the convolution of x(n) = {1, 2, 3, 4, 5} and h(n) = {1, 1, 1} using overlap save method? (5)

c) State Circular frequency shift property of DFT? (5)

4 –point DFT of the signal x(n) ={a, b, c, d} is X(K). Find the IDFT of X(K-2) ?

2 a) Find the number of complex multiplications and additions involved in the calculation of 1024 DFT using direct computation and radix2 FFT algorithm? (4)

b) How will you obtain linear convolution from circular convolution? For x(n) ={1, 2, 3} and h(n) = {-1, -2}, obtain linear convolution x(n)*h(n) using circular convolution? (5)

c) Given at g(n) ={1, 0, 1, 0} and h(n) = {1, 2, 2, 1} find the 4 point DFTs of these sequences using a single 4 point DFT. (6)

3 a) Describe the steps involved in radix 2 DIT FFT algorithm (5)

b) Find the DFT of the sequence {1, 2, 3, 4, 4, 3, 2, 1} using DIT algorithm (7)

c) What do you mean by in place computation of DFT? (3)

PART B

Answer any two full questions, each carries 15 marks.

4 a) Explain the significance of linear phase FIR filter and comment on its impulse response? (4)

b) Design an ideal lowpass filter with frequency response (6)

$H(e^{j\omega})$ = 1 for $-0.5\pi \le \omega \le 0.5\pi$ and $H(e^{j\omega})$ = 0 for $0.5\pi \le |\omega| \le \pi$.

Find h(n) for N = 11.(use rectangular window)

c) Determine the frequency response of FIR filter defined by (5)

y(n) = 0.25x(n) + x(n - 1) + 0.25x(n - 2). Calculate the phase delay and group delay?

5 a) Convert the analog filter *H(s)* given below in to a second order Butterworth digital filter using impulse invariance technique. (6)

$$H(s) = \frac{1}{s^2 + \sqrt{2}\,s + 1}$$

b) Why can't we use impulse invariance technique for implementing digital highpass filter? (4)

c) Describe the steps involved in the design of digital Butterworth bandpass filter? (5)

6 a) Derive the equation for cutoff frequency in Butterworth filter? (5)

b) Apply bilinear transformation to $H(s) = \frac{2}{(s+1)(s+2)}$ with T = 1 sec and find H(z)? (5)

c) What is warping effect in bilinear transformation method and how can we eliminate it? (5)

PART C

Answer any two full questions, each carries 20 marks.

7 a) Draw the block diagram of TMS320C67XX and explain functions of each block? (10)

b) Realize the system function using minimum number of multipliers (5)
$H(z) = (1 + z^{-1})\,(1 + 0.5z^{-1} + 0.5z^{-2} + z^{-3})$

c) Obtain the transposed directform II structure for the system (5)
$y(n) = 0.5y(n-1) - 0.25y(n-2) + x(n) + x(n-1)$

8 a) Realize the system given by difference equation $y(n) = -0.1y(n-1) + 0.2y(n-2) + 3x(n) + 3.6x(n-1) + 0.6x(n-2)$ in cascade form? (6)

b) Obtain the parallel form realization for above system (6)

c) Find the lattice structure implementation of FIR filter h(n) = {1, 13/24, 5/8, 1/3} (8)

9 a) Explain the effect of coefficient quantization in IIR and FIR filters? (10)

b) If quantization noise has uniform distribution with zero mean, find the quantization noise in ADC with step size Δ? (5)

c) A signal x(n) is obtained by sampling analog signal x(t) at twice the Nyquist rate. If we wish to down sample x(n) by a factor 4, obtain the bandwidth of the decimation filter required for supressing aliasing distortion. (5)

****