Reg No :- Name:-

APJ ABDUL KALAM TECHNOLOGICAL UNIVERSITY

FIFTH SEMESTER B.TECH DEGREE EXAMINATION(S), MAY 2019

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) Find the 4-DFT and 8-DFT of the sequence {1, 1, 1, 0}. Plot |X(K)| and comment on the significance of N? (10)

b) State Parseval's property? (5)

DFT of a real valued signal X(K). = {j, 1+j, A, 1-j, -1, B, -1-j, C}. Find the energy of the signal?

2 a) Find the convolution of x(n) = {1, 2, 3, 4, 5, 6, 7, 8, 9} and h(n) = {2, 4, 6} using overlap add method? (6)

b) Find the response of an LTI system with impulse response h(n) = { 1, 2, 2, 1} for an input x(n) = {1 , -1 , 1, -1 } using circular convolution? (4)

c) If x(n) = {1, 2, 3, 4}. Find DFT[DFT(x(n))] without calculating DFT? (5)

3 a) Explain the radix-2 DIT FFT algorithm and draw the corresponding flow diagram for 16 DFT computation. (10)

b) Explain about the efficient computation of DFT of a 2N- point real sequence (5)

PART B

Answer any two full questions, each carries 15 marks.

4 a) Derive equations for magnitude and phase responses of FIR filter whose impulse response is symmetric and length N odd. (5)

b) Design an ideal 6th order linear phase lowpass filter with frequency response (6)

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

Use Hamming window.

c) Explain Gibb's phenomenon. (4)

5 a) Determine the filter coefficients of a linear phase FIR filter of length N = 15, which has a symmetric impulse response and a frequency response that satisfies the conditions, (10)

$$H\left(\frac{2\pi k}{15}\right) = \begin{cases} 1\,, & k = 0,1,2,3 \\ 0.4\,, & k = 4 \\ 0\,, & k = 5,6,7 \end{cases}$$

b) Prove that the zeros of FIR filter exists as reciprocals. (5)

6 Design a digital Butterworth filter that has -1dB pass band attenuation at 200 Hz and at least -15dB stop band attenuation at 540 Hz. Sampling frequency = 2000 Hz. Find the cut off frequency by matching pass band criterion. Use Bilinear transformation (T = 1 sec) (15)

PART C

Answer any two full questions, each carries 20 marks.

7 a) Explain the steps through which we obtained direct form II realization of recursive LTI system described by difference equation. (10)

$$y(n) = -\sum_{k=1}^{N} a_k\, y(n-k) + \sum_{k=0}^{M} b_k\, x(n-k)$$

b) Draw the architecture block diagram of TMS320C67XX processor (5)

c) Obtain the transposed direct form II structure for the system (5)

$y(n) = 2y(n-1) + 3y(n-2) + x(n) + 2x(n-1) + 3x(n-2)$

8 a) Find the impulse response h(n) of a FIR filter, if the reflection coefficients are $K_1 = 2/5$, $K_2 = 4/21$, $K_3 = 1/8$. (6)

b) What is transposition theorem and transposed structure? (6)

c) Obtain direct form II and cascade structure for the transfer function given below. (8)

$$H(z) = \frac{1 + 2z^{-1} + z^{-2}}{1 - \frac{3}{4}z^{-1} + \frac{1}{8}z^{-2}}$$

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

b) What are the main features of DSP processor? (5)

c) Explain the effect in the spectrum of a signal x(n) when it is (5)

(i) Decimated by a factor 3

(ii) Interpolated by a factor 2

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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)

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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)

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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)

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FIFTH SEMESTER B.TECH DEGREE EXAMINATION, DECEMBER 2017

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) Explain, how DFT and IDFT can be expressed as Linear Transformation (3)

b) Derive the relationship of DFT to Z-transform (3)

c) Find the circular convolution of x[n] = {1, 2, -1, 3, 4} and h[n] = {2, -1, 4, 1, 3} (5)

d) Explain overlap add method for filtering of long data sequences. (4)

2 a) Show that, if x[n] is a real and even sequence, then its DFT X[k] is also real and even (3)

b) Find linear convolution of x[n] = {2, 3, -1} and h[n] = {1, -1, 2}, using circular convolution. (5)

c) Find the number of complex multiplications involved in the calculation of a 1024 point DFT using (i) direct computation(ii) radix-2 FFT algorithm (3)

d) Explain, how N point DFTs of two real-valued sequences can be found by computing a single N point DFT. (4)

3 a) Find 8 point DFT of x[n] = {2, 1, -1, 3, 5, 2, 4, 1} using radix-2 decimation in time FFT algorithm (11)

b) Explain, how a 2N point DFT of a 2N point real-valued sequence can be found by computing a single N point DFT. (4)

PART B

Answer any two full questions, each carries 15 marks.

4 a) Prove that, if z_1 is a zero of a linear phase FIR filter, then $1/z_1$ is also a zero. (5)

b) Design a linear phase FIR low pass filter having length M = 15 and cut-off frequency $\omega_c = \pi/6$. Use Hamming window. (10)

5 a) Explain the design of linear phase FIR filters by the frequency sampling method. (9)

b) Explain the frequency transformations in the analog domain (6)

6 Design a digital Butterworth low pass filter with $\omega_p = \pi/6$, $\omega_s = \pi/4$, minimum pass band gain = -2dB and minimum stop band attenuation = 8dB. Use bilinear (15)

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transformation. (Take T = 1)

PART C

Answer any two full questions, each carries 20 marks.

7 a) Find the lattice structure implementation of FIR filter h[n] = {1, 0.5, 0.75, -0.6} (6)

b) Draw the direct form II structure and transposed direct form II structure of (5)

$$H(z) = \frac{1+0.5z^{-1}-0.75z^{-2}}{1+0.6z^{-1}+0.4z^{-2}-0.2z^{-3}}$$

c) Draw the block diagram of TMS320C67XX and briefly explain the function of each block. (9)

8 a) Draw the direct form realization of linear phase FIR filter (5)
h[n] = {1, 0.5, 0.25, -0.5, 0.8, -0.5, 0.25, 0.5, 1} using minimum multipliers.

b) Draw the signal flow graphs ofdirect form II and cascade form structures (5)
of $H(z) = \frac{(0.8+0.2z^{-1}+0.6z^{-2})(1-0.6z^{-1})}{(1-0.6z^{-1}+0.8z^{-2})(1+0.8z^{-1}-0.7z^{-2})}$

c) Explain the effects of coefficient quantization in IIR and FIR filters. (10)

9 a) Give the output of decimation by M system in time domain. Explain output frequency spectrum. What is the importance of low pass filtering prior to down-sampling? (10)

b) How does a floating-point number represented in a processor? Explain the operations of addition and multiplication of two floating point numbers with examples. (5)

c) Derive the variance of quantization noise in ADC with step size Δ. (Assume quantization noise has uniform distributed pdf with zero mean) (5)

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