Code No: R32043 **R10** **Set No: 1**

III B.Tech. II Semester Supplementary Examinations, January -2014

DIGITAL SIGNAL PROCESSING

(Common to Electronics and Communication Engineering & Electronics and Computer Engineering)

Time: 3 Hours **Max Marks: 75**

Answer any FIVE Questions

All Questions carry equal marks

*****

1. a) Check for the causality, stability, linearity and time-invariance of the system represented by

$$y(n) = \sum_{k=n-n_0}^{n+n_0} x[k]$$

b) The unit-sample response of a system is specified by $h(n) = 1 \quad 0 \le n \le N-1$

$= 0$ else where

Obtain and sketch the magnitude and phase response of the system. (8+7)

2. a) Explain the symmetry properties of DFT and verify them for the sequence $x[n] = a^n u[n]$.

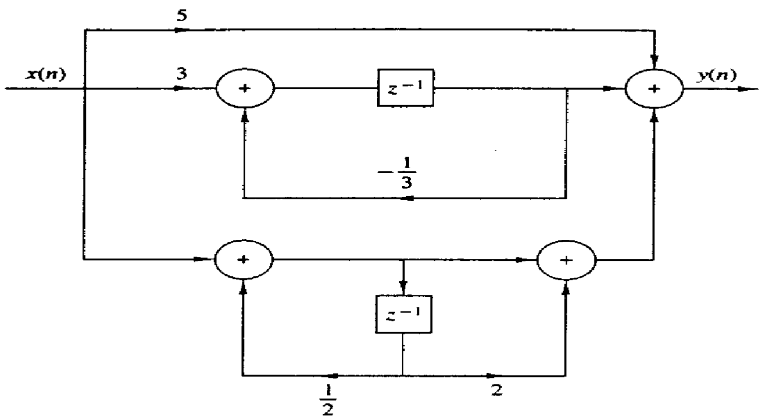
b) State the properties of discrete Fourier series and prove one of them. (10+5)

3. a) Explain radix-2 DIT FFT algorithm and draw the butter fly diagram for 8-point DIT FFT.

b) Compute the IDFT of X (k) = {15, 0, 0, 0, 5, 0, 0, 0} using DIF algorithm. (8+7)

4. a) Write the differences between FIR and IIR systems and derive their general transfer functions.

b) Determine the transposed structure of the system shown in figure below and verify that both the original and transposed system have same transfer function.



(7+8)

Code No: R32043 **R10** **Set No: 1**

5. Design a low-pass Butterworth filter using bilinear transformation method to satisfy the following constraints. Passband edge: 400 Hz, stopband edge:2.1 kHz, pass band ripple ≤2dB, stopband attenuation ≥ 20dB, sampling frequency :12kHz. (15)

6. a) Design an FIR digital filter and obtain H(z) to approximate an ideal low-pass filter with passband gain of unity, cutoff frequency of 1kHz and working at a sampling frequency of 4 kHz. Use Fourier series method. Consider the length of impulse response to be 11.
   b) Write the principle of design of FIR filter using frequency sampling method. (10+5)

7. a) Explain the process of decimation using a block schematic , waveforms and relevant expressions. (8+7)
   b) Discuss the implementation of a interpolation system using a polyphase structure.

8. a) Explain the difference between Von Neumann architecture and Harvard architecture for computer. Which architecture is preferred for DSP applications and why?
   b) Explain instruction pipelining with an example and show how it increases the throughput efficiency. (6+9)

*****

Code No: R32043 **R10** **Set No: 2**

III B.Tech. II Semester Supplementary Examinations, January -2014

DIGITAL SIGNAL PROCESSING

(Common to Electronics and Communication Engineering & Electronics and Computer Engineering)

Time: 3 Hours **Max Marks: 75**

Answer any FIVE Questions
All Questions carry equal marks
*****

1. a) Check for the causality, stability, linearity and time-invariance of the system represented by $T(x[n]) = ax[n] + b$

   b) Obtain the frequency response and impulse response of a causal LTI system is represented by the constant coefficient difference equation
   $$y[n] - \frac{5}{6}y[n-1] + \frac{1}{6}y[n-2] = \frac{1}{3}x[n-1]$$
   (8+7)

2. a) Derive the expression for discrete Fourier series representation of a periodic sequence. Also obtain the relationship between DFS and Z-Transform.
   b) Perform circular convolution of two sequences $x_1[n] - x_2[n] - \begin{cases} 1 & 0 < n < 4 \\ 0 & \text{else where} \end{cases}$ (10+5)

3. a) Draw and explain the basic butterfly diagrams for DIF FFT and DIT FFT and compare the two algorithms.
   b) An 8-point sequence is given by x[n]={ 1, 1, 1, 1, 2, 2, 2, 2 } compute 8-point DFT of x[n] using radix-2 DIF FFT algorithm.
   c) Calculate the percentage saving in complex additions in a 256-point radix-2 FFT compared to DFT. (5+7+3)

4. a) Compare direct form-I and direct form-II structures of realization of IIR filters.
   b) Obtain the direct form-I and cascade structures for the system represented by
   $y(n) = \frac{3}{4}y(n-1) - \frac{1}{8}y(n-2) + x(n) + \frac{1}{3}x(n-1)$
   c) Write the difference between recursive and non-recursive system using relevant expressions. (4+8+3)

5. Design a Chebyshev digital low pass filter and obtain its transfer function to meet the following specifications passband ripple ≤ 0.5dB, passband edge : 1.2 kHz, stopband attenuation ≥ 40 dB, stopband edge: 2 kHz, sampling rate 8kHz. (15)

6. a) Design a digital FIR low-pass filter and obtain H(z)using rectangular window for N=9 with a cutoff frequency of 1.2 rad/sec.
   b) What is the advantage of windowing technique and write the characteristic features of Kaiser window. (10+5)

7. a) Explain the process of interpolation using a block schematic, waveforms and relevant expressions.
   b) Discuss the sampling rate converter implementation with polyphase filters. (8+7)

Code No: R32043 R10 Set No: 2

8. a) Explain bit reversed and circular addressing modes and mention their applications.
b) Discuss the relative merits and demerits of RISC and CISC processors. c) What is meant by memory mapped register? How is it different from memory? (6+6+3)

*****

Code No: R32043 **R10** **Set No: 3**

III B.Tech. II Semester Supplementary Examinations, January -2014

DIGITAL SIGNAL PROCESSING

(Common to Electronics and Communication Engineering & Electronics and Computer Engineering)

Time: 3 Hours **Max Marks: 75**

Answer any FIVE Questions

All Questions carry equal marks

*****

1. 1 a) Check for the causality, stability, linearity and time-invariance of the system represented by $T(x[n]) = x[n] + 3u[n+1]$
   b) Consider an LTI system represented by a first order difference equation y[n]= ay[n-1]+x[n]. Assuming zero initial conditions obtain unit-sample response. (8+7)

2. a) Perform linear convolution the following sequences.
   $x_1[n]$={ 3, 5, 4,6,7,0,2,4,1} and $x_2[n]$={1, 2, 3}
   b) State and prove circular convolution property of DFT. (8+7)

3. a) Explain radix-2 DIF FFT algorithm and draw the butter fly diagram for 8-point DIF FFT.
   b) Compute the IDFT of X(k) = {15, 0, 0, 0, 5, 0, 0, 0} using DIT algorithm. (8+7)

4. a) Obtain the direct form structure of FIR system represented by
   $H(z) = 1 + 0.2z^{-1} + 0.5z^{-2} + 0.33z^{-3} + 0.14z^{-4} + 0.15z^{-5}$.
   b) Obtain the direct form – I and parallel structures for the system represented by
   $y(n) = y(n-1) - \frac{1}{2}y(n-2) + x(n) - x(n-1) + x(n-2)$
   c) What are the factors that influence the choice of structure for realization of an LTIsystem? (4+8+3)

5. Design a low-pass Butterworth filter using bilinear transformation method and obtain its transfer function to satisfy the following constraints passband edge: 4kHz , stopband edge:6kHz, pass band ripple ≤1dB, stopband attenuation ≥ 40dB, sampling rate:24kHz. (15)

6. a) Design a FIR high pass filter using Hamming window with a cutoff frequency of 1.2 rad/s and N=9.
   b) What is Gibbs phenomenon? (12+3)

7. a) Explain the method of sampling rate conversion by a factor of I/D using relevant block schematics and expressions. (8+7)
   b) Discuss the implementation of a decimation system using a polyphase structure.

8. a) Explain the different techniques adopted for increasing the number of memory accesses/instruction cycle.
   b) Explain the operation of multiplier accumulator with neat diagram. .
   c) What is Harvard architecture? Explain its advantages and compare to Von Neumann architecture. (3+7+5)

*****

Code No: R32043 **R10** **Set No: 4**

III B.Tech. II Semester Supplementary Examinations, January -2014
DIGITAL SIGNAL PROCESSING
(Common to Electronics and Communication Engineering & Electronics and Computer Engineering)

Time: 3 Hours **Max Marks: 75**

Answer any FIVE Questions
All Questions carry equal marks
*****

1. a) Check for the causality, stability, linearity and time-invariance of the system represented by $T(x[n]) = e^{x[n]}$
b) Obtain the frequency response and impulse response of a causal LTI system is represented by the constant coefficient difference equation. Given ` a' is a constant
$y[n] + (1/a)y[n-1] = x[n-1]$ (8+7)

2. a) Perform linear convolution the following sequences
$x_1[n]=\{1, 2, 3, 4\}$ and $x_2[n]=\{1, 2, 3\}$
b) State and prove periodic convolution property of discrete Fourier series. (8+7)

3. a) Write the procedure to compute DFT using radix-2 FFT.
b) An 8-point sequence is given by x[n] = { 1, 1, 1, 1, 3, 3, 3, 3,} compute 8-point DFT of x[n] using radix-2 DIT FFT algorithm.
c) Calculate the percentage saving in complex multiplications in a 256-point radix-2 FFT compared to DFT. (4+8+3)

4. a) Realize the filter transfer function given by the expression below using the direct form $H(z) = (1-z^{-1})(1+2z^{-1}-5z^{-2})$
b) Obtain the direct form-II and parallel structures for the system represented by
$y(n) = \frac{3}{4}y(n-1) - \frac{1}{8}y(n-2) + x(n) + \frac{1}{3}x(n-1)$
c) Write the procedure to obtain transpose structure of a filter. (4+8+3)

5. Determine the system function H(z) of the lowest-order Chebyshev filter that meets the following specifications 1dB ripple in the pass band $0 \le |\omega| \le 0.3\pi$ and at least 60dB attenuation in the stopband $0.35\pi \le |\omega| \le \pi$. Use bilinear transformation. (15)

6. a) Write the advantages and disadvantages of FIR filters.
b) Design a band-pass FIR filter to pass frequencies in the range 1 to 2 rad/s using Hanning window with N=5. (5+10)

7. a) Explain down sampling operation with neat sketch.
b) Explain the method of sampling rate conversion by a factor of I/D using relevant block schematics and expressions. (7+8)

8. a) Explain special addressing modes in DSP processors.
b) Draw the diagram of internal architecture of TMS320C5X and describe its bus structure and ALU. (6+9)

*****