Code No: RR320402 **Set No. 1**

III B.Tech II Semester Regular Examinations, Apr/May 2007
DIGITAL SIGNAL PROCESSING
(Common to Electronics & Communication Engineering, Electronics & Instrumentation Engineering, Electronics & Control Engineering, Electronics & Telematics and Instrumentation & Control Engineering)

Time: 3 hours **Max Marks: 80**

Answer any FIVE Questions
All Questions carry equal marks

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1. (a) Consider a LSI system with unit sample response $h(n)n = \alpha^n u(n)$ where α is real and $0 < \alpha < 1$. If the input is $x(n) = \beta^n\, u(n), 0 < |\beta| < 1$,determine the the output f(n) in the form $y(n) = (k_1\alpha^n + k_2\beta^n)u(n)$ by explicitly evaluating the convolution sum.
   (b) Define causality and stability of LSI system and state the conditions for stability.

   [12+4]

2. (a) Let x(n) and $X(e^{jw})$ represent a sequence and its transform. Determine, in terms of $X(e^{jw})$, the transform of each of the following sequences :
      i. k x(n), k = any constant
      ii. $x(n - n_0)$, n_0 = a real integer
   (b) By explicitly evaluating the transforms $X(e^{jw}), H(e^{jw}) and Y(e^{jw})$ corresponding to x(n), h(n) and y(n) specified in part (a) show that $Y(e^{jw}) = H(e^{jw})X(e^{jw})$

   [8+8]

3. (a) State and prove the circular time shifting and frequency shifting properties of the DFT.
   (b) Compute the circular convolution of the sequences
   $x_1(n) = \{1, 2, 0, 1\}$ and
   $x_2(n) = \{2, 2, 1, 1\}$ Using DFT approach. [8+8]

4. (a) Draw the butterfly line diagram for 8 - point FFT calculation and briefly explain. Use decimation -in-time algorithm.
   (b) What is FFT? Calculate the number of multiplications needed in the calculation of DFT using FFT algorithm with 32 point sequence. [8+8]

5. (a) With reference to Z-transform, state the initial and final value theorem.
   (b) Determine the causal signal x(n) having the Z-transform $X(Z) = \frac{Z^2+Z}{\left(Z-\frac{1}{2}\right)^2\left(Z-\frac{1}{4}\right)}$.

   [6+10]

6. (a) What is warping effect? Discuss influence of warping effect on amplitude response and phase response of a derived digital filter from a corresponding analog filter.

(b) Discuss impulse invariance method. [8+8]

7. (a) Design a low pass filter by the Fourier series method for a seven stage with cut-off frequency at 300 Hz if $t_s = 1msec$. Use hanning window.

   (b) Explain in detail, the linear phase response and frequency response properties of Finite Impulse Response filters. [8+8]

8. (a) What are the basic elements used to construct the block diagram of discrete time system?

   (b) Construct the block diagram and signal flow graph of the discrete time system whose input-output relations are described by following difference equation

   i. $y(n) = 0.5x(n) + 0.5x(n-1)$

   ii. $y(n) = 0.25y(n-1) + 0.5x(n) + 0.75x(n-1)$ [4+12]

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Code No: RR320402 **Set No. 2**

III B.Tech II Semester Regular Examinations, Apr/May 2007
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Time: 3 hours **Max Marks: 80**

Answer any FIVE Questions
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1. (a) Define a linear-time invariant system in discrete time. Define the terms causality and stability of such systems. Discuss the properties for the following sequences.

$$y(n) = \begin{cases} x(n), for\ n \geq 1 \\ 0,\ for\ n = 0 \\ x(n+1), for\ n \leq -1 \end{cases}$$

   (b) Find the:
   i. impulse response
   ii. output response for a step input applied at n=0 of a discrete time linear time invariant system whose difference equation is given by y(n) = y(n-1)+0.5 y(n-2)+x(n)+x(n-1). [8+8]

2. (a) Show that the frequency response of a discrete system is a periodic function of frequency.
   (b) Obtain the frequency response of the first order system with difference equation y(0) = x(n)+10y(n-1) with initial condition y(-1) = 0 and sketch it comment about its stability.
   (c) State and prove the frequency shifting property of Fourier transform. [5+6+5]

3. (a) State and prove the circular time shifting and frequency shifting properties of the DFT.
   (b) Compute the circular convolution of the sequences
   $x_1(n) = \{1, 2, 0, 1\}$ and
   $x_2(n) = \{2, 2, 1, 1\}$ Using DFT approach. [8+8]

4. (a) Implement the decimation in time FFT algorithm for N=16.
   (b) In the above Question how many non - trivial multiplications are required. [10+6]

5. (a) With reference to Z-transform, state the initial and final value theorem.
   (b) Determine the causal signal x(n) having the Z-transform $X(Z) = \frac{Z^2+Z}{\left(Z-\frac{1}{2}\right)^2\left(Z-\frac{1}{4}\right)}$. [6+10]

6. Design a low pass digital filter with the following specifications;
   Maximum pass band attenuation $\alpha max = -3db, for 0 \leq w \leq 2\pi/10$
   Minimum stop band attenuation $\alpha max = -15db, for 3\pi/10 \leq w \leq \pi$
   Use following two methods:
   (a) Impulse invariance method
   (b) Bilinear transformation method.
   Assume T = 0.2 msec. [16]

7. (a) Draw the frequency response of N-point rectangular window.
   (b) Design a fifth order band pass linear phase filter for the following specifications.
      i. Lower cut-off frequency = 0.4 $\pi rad/sec$
      ii. Upper cut-off frequency = 0.6 $\pi rad/sec$
      iii. Window type = Hamming
      Draw the filter structure. [4+12]

8. (a) Describe how targets can be decided using RADAR
   (b) Give an expression for the following parameters relative to RADAR
      i. Beam width
      ii. Maximum unambiguous range
   (c) Discuss signal processing in a RADAR system. [4+6+4]

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Code No: RR320402 **Set No. 3**

III B.Tech II Semester Regular Examinations, Apr/May 2007
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Time: 3 hours **Max Marks: 80**

Answer any FIVE Questions
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1. (a) Let e(n) be an exponential sequence $e(n) = \alpha^n$ for all 'n' and let x(n) and y(n) be two arbitrary sequences. Show that [e(n)y(n)]* [e(n)x(n)]=e(n)[y(n)*x(n)]where '*' denotes convolution operation.
   (b) A system is described by the difference equation y(n)-y(n-1)-y(n-2) = x(n-1). Assuming that the system is initially relaxed, determine it s unit sample response h(n).

   [8+8]

2. (a) Prove that the convolution in time domain leads to multiplication in frequency domain for discrete time signals
   (b) The out put y(n) for a linear shit invariant system, with the input x(n) is given by
   Y(n) = x(n)-2x(n-1)+x(n-2)
   Compute and sketch the magnitude and phase response of the system $|w| \leq \pi$

   [8+8]

3. (a) Define DFT of a sequence. Compute the N - point DFT of the sequence. $X(n) = Cos(2\pi rn/N), 0 \leq n \leq N-1$ and $0 \leq r \leq N-1$
   (b) Explain how DFT can be obtained by sampling DFS for a given sequence. [8+8]

4. An 8 point sequence is given by x(n) = {2,2,2,2,1,1,1,1}. Compute 8 point DFT of x(n) by
   (a) radix - 2 D I T F F T
   (b) radix - 2 D I F FF T
   Also sketch magnitude and phase spectrum. [16]

5. (a) With reference to Z-transform, state the initial and final value theorem.
   (b) Determine the causal signal x(n) having the Z-transform $X(Z) = \frac{Z^2+Z}{\left(Z-\frac{1}{2}\right)^2\left(Z-\frac{1}{4}\right)}$.

   [6+10]

6. (a) What is an IIR digital filter?
   (b) How are IIR digital filter realized?

(c) What are the various realizability constraints imposed on transfer function of an IIR digital filter. [4+4+6]

7. (a) Outline the steps involved in the design of FIR filter using windows.

   (b) Determine the frequency response of FIR filter defined by y(n) = 0.25x(n)+ x(n-1)+ 0.25x(n-2). Calculate the phase delay and group delay. [8+8]

8. (a) List the different types of structures for realizing FIR system and determine the direct form-I, direct form II of the following LTI system y(n)=-0.5y(n-1)+0.25y(n-2)+0.125y(n-3)+x(n)+0.5x(n-1)+0.75x(n-2)

   (b) Write briefly about digital processing of speech. [6+10]

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Code No: RR320402 **Set No. 4**

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Time: 3 hours **Max Marks: 80**

Answer any FIVE Questions
All Questions carry equal marks

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1. (a) Consider a LSI system with unit sample response $h(n)n = \alpha^n u(n)$ where α is real and $0 < \alpha < 1$. If the input is $x(n) = \beta^n\, u(n), 0 < |\beta| < 1$,determine the the output f(n) in the form $y(n) =$ ($k_1\alpha^n + k_2\beta^n)u(n)$ by explicitly evaluating the convolution sum.
   (b) Define causality and stability of LSI system and state the conditions for stability.

   [12+4]

2. (a) If x(n) $\rightarrow x(e^{j\omega})$ Constitute a Fourier transform pair. Prove the following:

Sequence	Fourier Transform
i. x*(n)	$X*(e^{-j\omega}\,)$
ii. $x_o(n)$	$jIm[X(e^{j\omega})]$

   (b) Let x(n) and $X(e^{j\omega})$ represent a sequence and its transform. Determine, in terms of $X(e^{j\omega})$, the transform of each of the following sequences :
   i. $g(n) = \begin{cases} x(n/2) & n\ even \\ 0 & n\ odd \end{cases}$
   ii. $x^2(n)$

   [8+8]

3. (a) Prove the following properties
      i. $arg[X(K)] = -arg[X((-K)_N)R_N(K)]$
      ii. $Im[X(K)] = -Im[X((-K))_N R_N(K)]$
   (b) If X(K) denotes the N-point DFT of N-Point sequence x(n), show that with N even and if x(n) = x(N-1-n) then X(N/2)=0. [8+8]

4. An 8 point sequence is given by x(n) = {2,2,2,2,1,1,1,1}. Compute 8 point DFT of x(n) by
   (a) radix - 2 D I T F F T
   (b) radix - 2 D I F FF T
   Also sketch magnitude and phase spectrum. [16]

5. (a) Explain how the analysis of discrete time invariant system can be obtained using convolution properties of Z transform.

(b) Determine the impulse response of the system described by the difference equation y(n)-3y(n-1)-4y(n-2)=x(n)+2x(n-1) using Z transform. [8+8]

6. (a) Design a digital filter that will pass a 1 Hz signal with attenuation less than 2 db and suppress 4 Hz signal down to at least 42 db from the magnitude of the 1 Hz signal.

   (b) What are the limitations of Impulse invariance method? [12+4]

7. (a) Define Infinite Impulse Response & Finite Impulse Response filters and compare.

   (b) Design a low pass Finite Impulse Response filter with a rectangular window for a five stage filter given:
   Sampling time 1 msec; $f_c = 200Hz$
   Draw the filter structure with minimum number of multipliers. [6+10]

8. (a) How is a speech signal generated?

   (b) Give the model of human speech production system and explain. [6+10]

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