Code No: RR410201 **Set No.1**

IV B.Tech. I Semester Regular Examinations, November -2005
DIGITAL SIGNAL PROCESSING
(Electrical & Electronic Engineering)

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

Answer any FIVE Questions
All Questions carry equal marks

⋆ ⋆ ⋆ ⋆ ⋆

1. (a) For each of the following systems, determine whether or not the system is
   i. stable
   ii. causal
   iii. linear
   iv. shift-invariant.

   A. $T[x(n)] = x(n - n_0)$
   B. $T[x(n)] = e^x(n)$
   C. T[x(n)] = a x(n) + b.
   Justify your answer.

   (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) Discuss the frequency-domain representation of discrete-time systems and signals. By deriving the necessary relation.

   (b) Draw the frequency response of LSI system with impulse response $h(n) = a^n u(-n)$ $(|a| < 1)$

   [8+8]

3. (a) Prove the following properties.
   i. $x^*(n) \rightarrow X^*((-K))_N R_N(K)$
   ii. $x^*((-n))_N R_N(n) \rightarrow X_{ep}(k) = 1/2[X((K))_N + X^*((-K))_N]R_N(K)$

   (b) Let X(K) denotes the N-point DFT of the N-point sequence x(n) show that if x(n) satisfies the relation $x(n) = -x(N - 1 - n) then X(0) = 0$. [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. Determine the system function H(Z) of the lowest order Chebyshev digital filter that meets the following specifications.
   (a) 1 db ripple in the passband $0 \leq |W| \leq 0.3\pi$
   (b) At least 60 db attenuation in the stopband $0.35\pi \leq |W| \leq \pi$. Use the bilinear transformation. [16]

7. (a) Design a low pass filter using rectangular window by taking samples of $\omega(n)$ and with a cut-off frequency of 1.2 radians/sec.
   (b) Compare the various window functions. [8+8]

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. [5+6+5]

⋆ ⋆ ⋆ ⋆ ⋆

Set No.2

IV B.Tech. I Semester Regular Examinations, November -2005
DIGITAL SIGNAL PROCESSING
(Electrical & Electronic Engineering)

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

Answer any FIVE Questions
All Questions carry equal marks

⋆ ⋆ ⋆ ⋆ ⋆

1. (a) The unit-sample response of a linear-shift-invariant system is known to be zero. Except in the interval $N_0 \leq n \leq N_1$. The input x(n) is known to be zero except in the interval $N_2 \leq n \leq N_3$. As a result, the output is constrained to be zero except in some interval $N_4 \leq n \leq N_5$. Determine $N_4 and N_5$ in terms of $N_0, N_1, N_2 and N_3$.

   (b) By direct evaluation of the convolution sum, determine the step response of a Linear shift-invariant system whose unit-sample response h(n) is given by $h(n) = a^{-n}u(-n)$, 0 < a < 1.

   [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) 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. (a) Let x(n) be a real valued sequence with N-points and Let X(K) represent its DFT , with real and imaginary parts denoted by $X_R(K)$ and $X_1(K)$ respectively. So that X(K) = $X_R(K) + JX_1(K)$. Now show that if x(n) is real, $X_R(K)$ is even and $X_1(K)$ is odd.

   (b) Compute the FFT of the sequence x(n) = { 1, 0, 0, 0, 0, 0, 0, 0 } [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. 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. Design a low pass Finite Impulse Response filter that approximate the following frequency response:
$H(f) = \begin{cases} 1\,; & 0 \leq f \leq 1000\,Hz \\ 0\,; & elsewhere\,\text{in}\,the\,range\,0 \leq f \leq f_s/2 \end{cases}$
when the sampling frequency is 8000 sps. The impulse response duration is to be limited to 2.5 msec. Draw the filter structure. [16]

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

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

⋆ ⋆ ⋆ ⋆ ⋆

Code No: RR410201 **Set No.3**

IV B.Tech. I Semester Regular Examinations, November -2005
DIGITAL SIGNAL PROCESSING
(Electrical & Electronic Engineering)

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

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

1. By explicitly evaluating the convolution sum, evaluate the convolution y(n)=(n)*h(n) of the sequences
$h(n) = \begin{cases} \alpha^n & 0 \le n < N \\ 0 & elsewhere \end{cases}$
$X(n) = \begin{cases} \beta^{n-n0}, & n_0 \le n \\ 0 \quad , & n \le n_0 \end{cases}$ [16]

2. (a) Let $x_1(n) = x_2(n) = \begin{cases} 1 & 0 \le n \le N-1 \\ 0 & otherwise \end{cases}$
Find out the circular convolution of these sequences.
(b) If $x_1(n)$ *and* $x_2(n)$ are two finite sequences, derive the linear convolution of these sequences. Using circular convolution.
[8+8]

3. (a) Define DFT. Guide tow properties of DFT.
(b) Discuss the effects of truncating a sequence x(n) of infinite duration.
(c) Compute the DFT of $X(n) = \{-1, 0, -1\}$ with T = 0.5. Plot the DFT sequence suggest a method for improving frequency resolution. [4+6+6]

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) 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. Determine the system function H(Z) of the lowest order Chebyshev and Butterworth digital filter with the following specification
(a) 3 db ripple in pass band $0 \le w \le 0.2\pi$
(b) 25 db attenuation in stop band $0.45\pi \le w \le \pi$ [16]

7. (a) Design a high pass filter using hamming window with a cut-off frequency of 1.2 radians/second and N=9
(b) Compare FIR and IIR filters. [10+6]

8. (a) Explain the parallel form realisation for IIR system and obtain the direct form I, direct form II realisation of the LTI systems governed by the equation.
$y(n) = -\frac{3}{8}y(n-1) + \frac{3}{32}y(n-2) + \frac{1}{64}y(n-3) + x(n) + 3x(n-1) + 2x(n-2)$

   (b) Compare cascade and parallel form relations. [12+4]

⋆⋆⋆⋆⋆

IV B.Tech. I Semester Regular Examinations, November -2005
DIGITAL SIGNAL PROCESSING
(Electrical & Electronic Engineering)

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

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

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) Prove the modulation and time shifting properties of distribute time Fourier transform.
   (b) A discrete system is given by following difference equation
   y(n)-5y(n-1) = x(n) + 4x(n-1)
   where x(n) is the input and y(n) is the out put. Determine its magnitude and phase response as a function of frequency.
   [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) 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. (a) Compare Butter worth and Chebydhev approximations.
   (b) Determine the order and transfer function of the Chebyshev filter for following specifications:
      i. Maximum pass band ripple is 1 db for $\Omega \leq 4$ radius/sec.
      ii. Stop band attenuation is 40 db for $\Omega \geq 4$ radius/sec. [8+8]

7. (a) Design a Finite Impulse Response low pass filter with a cut-off frequency of 1 kHz and sampling rate of 4 kHz with eleven samples using Fourier series method.
   (b) Show that an FIR filter is linear phase if h(n) = h(N-1-n). [8+8]
8. (a) Explain in detail the short time Fourier analysis for speech signals
   (b) What is a vocoder? Explain with a block diagram. [10+6]

⋆ ⋆ ⋆ ⋆ ⋆