8. Draw the architecture of TMS series of processors. Discuss its main features.

9. Design a digital filter using bilinear transformation for the following analog transfer function :

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

Obtain transfer function H (s) of digital filter assuming 3 db cut off frequency of 150 Hz and sampling frequency of 1.28 kHz.

Roll No..................

A

Total No. of Questions : 9]

[Total No. of Printed Pages : 4

EC–308

DIGITAL SIGNAL PROCESSING (NEW)

(B.Tech., 6th Semester, 2055)

Time : 3 Hours | Maximum Marks : 60

Note :– Section A is compulsory. Attempt any *Four* questions from Section B and any *Two* questions from Section C.

Section–A Marks : 2 Each

1. (a) List applications of Digital Signal Processing.

   (b) What are the features of TMS series of processor ?

   (c) What is the importance of ROC ?

   (d) Define z-transform.

   (e) What is the relationship between z-transform and Laplace transform ?

(f) Differentiate between FIR and IIR filters.

(g) List name of structure for FIR system.

(h) What are the advantages of Digital filter over analog filter ?

(i) Find the z-transform of $\delta(n) = 1$ at $t = 0$.

(j) What are various methods of finding inverse z-transform ?

Section–B Marks : 5 Each

2. Solve the following second order difference equation :

$$2\,x(n-2) - 3\,x(n-1) + x(n) = 3^{n-2} \quad n > 0.$$

3. Find inverse z-transform of

$$x(z) = \frac{5}{1-z^{-1}} - \frac{7}{1-\frac{8}{5}z^{-1}}.$$

If :

(a) ROC = $|z| > 1$

(b) $|z| < 3/5$

(c) ROC = $3/5 < |z| < 1$.

4. State 3 properties of z-transform and give examples.

5. State 3-properties of Fourier Transform and give examples.

6. What is decimation in frequency for computing DFT ? What are advantages of Radix 2FFT.

Section–C Marks : 10 Each

7. Fig. 1 shows direct–II form of realisation of IIR filter :

(a) Find Transform functions H (z) of filter.

(b) Find corresponding difference equation

(c) Realize filter using cascade form using first order module ?

(d) Find impulse response function for filter.

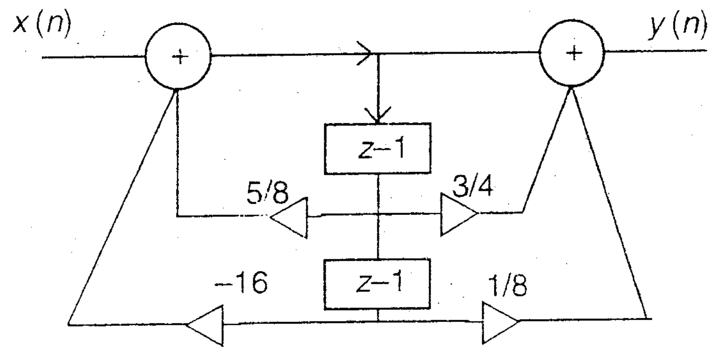

Fig. 1

Digital Signal Processing
(EC-308, Dec-2005)

Note: Section A is compulsory. Attempt any four questions from Section-B and any two from Section-C.

Section-A

1. a) What are the advantages of DSP over analog processing?
   b) State sampling theorem.
   c) Convolve [1, 3, 1] and [1, 2, 2]
   d) Define LTI system.
   e) Differentiate between linear-nonlinear systems.
   f) What is the difference between stable-astable systems?
   g) What is FIR?
   h) Define DFT.
   i) What is Region of convergence?
   j) State convolution theorem.

Section-B

2. Perform circular convolution of two sequences.
   $x_1(n) = \{0.2, 0.4, 0.6, 0.8, 1, 1.2, 1.4, 1.6\}$
   $x_2(n) = \{0.1, 0.3, 0.5, 0.7, 0.9, 1.1, 1, 3, 1\}$
3. Find z transforms of $\left\{\cos\left\{\frac{n\pi}{4}+\alpha\right\}\right\}_{n\geq 0}$

4. Find the inverse z transform of X(z) = $\frac{1+2z-1+z-2}{1-z-1+0.356z-2}$
5. Represent system function using linear phase FIR structure
   H(z) = $\frac{z}{2}+1+\frac{z-1}{2}$
6. What are the various realization techniques of linear time invariant systems? Mention.

Section-C

7. For a given analog filter system function H(S) = $\frac{S+0.1}{(S+0.1)2+16}$ into digital IPR filter by means of Bilencar Z transformation. Digital filter is to have resonant frequency $w_r = \frac{\pi}{2}$
8. (a) Compare different forms/structures of filter realization from the point of view of speed and memory requirement.
   (b) Explain with neat sketches, the cascade and parallel realization forms of digital filters.
9. (a) List various properties of z transforms.
   (b) What is decimation in frequency method for computing DFT? What are advantages of radix 2 FFT?

Roll No ....

Total No. of Questions : 09]

[Total No. of Pages : 02

J-303[6012A] [2126]

B.Tech. (Semester - 6th & 7th)

DIGITAL SIGNAL PROCESSING (EC - 308)

Time : 03 Hours

Maximum Marks : 60

Instruction to Candidates:

1) Section - A is **compulsory.**
2) Attempt any **Four** questions from Section - B.
3) Attempt any **Two** questions from Section - C.

Section - A

Q1) *(10 x 2 = 20)*

a) Define a discrete time unit sequence function.

b) Plot the derivative of the function $x(t) = \sin c(t)$.

c) Find the signal energy of the signal $x(t) = u(t) - u(10-t)$.

d) What are the applications of z-transform?

e) What are the conditions for the region of convergence of a noncausal LTI system?

f) What is the linearity property of DTFT?

g) Define transfer function of a system.

h) Define Radix-2 FFT algorithm.

i) What is the importance of windowing?

j) Compare the performance of FIR filter and IIR filter.

Section - B

(4 x 5 = 20)

Q2) Differentiate between a recursive and non-recursive system. Determine if the recursive system defined by the difference equation $y(n) = ay(n-1) + x(n)$ is linear.

P.T.O.

Q3) Define the stability conditions for a linear time invariant system. Determine the range of values of 'a' for which the LTI system with impulse response h(n) as defined below is stable.

$$h(n) = \begin{cases} a^n, & n \geq 0,\ n \text{ even} \\ 0, & \text{elsewhere.} \end{cases}$$

Q4) Determine the causal signal $x(n)$ if its z-transform X(z) is given by

$$X(z) = \frac{1+3z^{-1}}{1+3z^{-1}+2z^{-2}}$$

Q5) State the Goertzel algorithm and give its importance.

Q6) Describe the magnitude and phase response of FIR filters. How is linear phase FIR filter defined?

Section - C

(2 x 10 = 20)

Q7) Determine the cascade and parallel realizations for the system described by the system function

$$H(z) = \frac{10\left(1-\frac{1}{2}z^{-1}\right)\left(1-\frac{2}{3}z^{-1}\right)\left(1-2z^{-1}\right)}{\left(1-\frac{3}{4}z^{-1}\right)\left(1-\frac{1}{8}z^{-1}\right)\left[1-\left(\frac{1}{2}+j\frac{1}{2}\right)z^{-1}\right]\left[1-\left(\frac{1}{2}-j\frac{1}{2}\right)z^{-1}\right]}$$

Q8) (a) What are quantization errors in FFT algorithms?

(b) Define circular convolution. How can linear convolution be realized using circular convolution?

Q9) What are the limitations of IIR filter design by impulse invariance method? How are they overcome by bilinear transformation method? Convert the analog filter with system function

$$H(s) = \frac{s+0.1}{(s+0.1)^2+16}$$

into digital IIR filter by means of bilinear transformation.

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Digital Signal Processing
(EC-308, Dec-2007)

Note: Section A is compulsory. Attempt any four questions from Section-B and any two from Section-C.

Section-A

1. a) What are the constraints on the transfer function if it were to represent a casual LTI system?
   b) What is the relationship between the Z-transform and the discrete Fourier transform?
   c) In what respect does DFT differ from continuous Fourier transform?
   d) Explain the symmetry properties of DFTs which provide basis for fast algorithms.
   e) State the final value theorem of Z-transform.
   f) Mention two symmetry properties of FIR filters for obtaining linear phase.
   g) State the desirable characteristics of windows in the design of FIR digital filters.
   h) What is frequency warping in Bilinear transformation?
   i) What is the difference between Butterworth and chebyshev filters in terms of frequency response.
   j) Explain the concept of pipelining in DSP processor.

Section-B

2. What is the frequency response of a discrete LTI system? Derive the frequency response of a system whose impulse response is given by
   $h(n) = a^n U(n-1)$ for $|a| < 1$
3. Find the inverse of Z-transform of the function. $X(z) = \frac{(z-4)}{(z-1)(z-3)^2}$ for $|z| > 2$
4. draw a 8-point radix-2 FFT DIT flow graph and obtain DFT of the following sequence x(n) =(0, 1, -1, 0, 0, 2, -2, 0)
5. Design flow pass FIR filter using Hamming window to meet the following specifications.
   $H(w) = 1$ for $0 \le [w] \le \pi/6$
   $= 0$ for $\pi/6 \le [w] \le \pi$
   Use a 10 tap filter and obtain the impulse response of the desired filter.
6. Which is more sensitive network to finite word length?
   (a) Direct form-II
   (b) Cascade form
   Justify tour answer

Section-C

7. AN IIR low-pass filter is to be designed to meet the following specifications:
   (a) pass-band frequency: 0 to 1.2 k Hz
   (b) Stop band edge: 2 k Hz
   (c) pass-band attenuation ≤ 0.5 db
   (d) stop band attenuation ≥ 15 db

   Using butter worth approximation and bilinear transformation obtain the desired IIR digital filter.
8. A LTI system is described by $y(n) = y(n-1) - 0.24\, y(n-2) + x(n)$
   Find the response of this system fo an input of $x(n) = 10 \cos(0.05\pi n)$
9. With the help of a block diagram, explain the architecture of a TMS processor.

May-08
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Roll No. ....

Total No. of Questions : 09]

[Total No. of Pages : 02

Paper ID [EC308]

(Please fill this Paper ID in OMR Sheet)

B.Tech. (Sem. - 6th)

DIGITAL SIGNAL PROCESSING (EC - 308)

Time : 03 Hours

Maximum Marks : 60

Instruction to Candidates:

1) Section - A is **Compulsory**.
2) Attempt any **Four** questions from Section - B.
3) Attempt any **Two** questions from Section - C.

Section - A

Q1) *(10 x 2 = 20)*

a) Write two advantages of digital over analog signal processing.

b) What are symmetric and asymmetric signals?

c) Define circular convolution.

d) Define a Causal system.

e) Differentiate stable from a unstable system.

f) Write any two areas of applications of DSPs.

g) What is aperiodic discrete time sequence?

h) Define a symmetry property of DFT?

i) Write any two basic features of IIR filters.

j) Write any two applications of Z-Transforms in signal processing.

Section - B

(4 x 5 = 20)

Q2) Determine the output y(n) of a LTI system with impulse response

$h(n) = a^{nu}(n), |a| 1$

when input is a unit step sequence, that is $x(n) = u(n)$

Q3) What is the physical significance of ROC in Z transform.

Q4) Find out the Z-transform for the following discrete time sequence

$x(n) = kn, n \geq 0.$

Q5) Discuss FFT algorithm using decimation in time technique.

Q6) Discuss Linear filtering approach for the computation of DFT.

Section - C

(2 x 10 = 20)

Q7) Discuss signal flow graph representation and lattice structures for IIR systems.

Q8) Discuss various steps for the design of linear phase FIR filters using window method.

Q9) Discuss basic architecture of TMS series of digital signal processors.

Roll No. ....

Total No. of Questions : 09] [Total No. of Pages : 02

Paper ID [A0321]

(Please fill this Paper ID in OMR Sheet)

B.Tech. (Sem. - 6th)

DIGITAL SIGNAL PROCESSING (EC - 308)

Time : 03 Hours Maximum Marks : 60

Instruction to Candidates:

1) Section - A is **Compulsory**.
2) Attempt any **Four** questions from Section - B.
3) Attempt any **Two** questions from Section - C.

Section - A

Q1) *(10 × 2 = 20)*

a) Write any disadvantage of digital over analog signal processing.

b) Differentiate time variant from time invariant system?

c) Define sampling theorem.

d) Define a Causal system.

e) Differentiate stable from a non-stable system.

f) Write application of FFT algorithm?

g) What is zero padding in DFT?

h) What is linear convolution.

i) Write any two basic features of IIR filters.

j) Write any two applications of Z-Transforms in signal processing.

Section - B

(4 × 5 = 20)

Q2) Show that h(n) is equal to the convolution of the following signals

$h_1(n) = \delta(n) + \delta(n-1)$

$h_2(n) = (½)^n u(n)$.

Q3) What is the physical significance of ROC in Z transform.

Q4) Find out the Z-transform for the following discrete time sequence $x(n) = kn^2, n \geq 0$.

Q5) Discuss FFT algorithm using decimation in frequency technique.

Q6) Discuss various properties of DFT.

Section - C

(2 × 10 = 20)

Q7) Discuss signal flow graph representation and lattice form structures for FIR systems.

Q8) Discuss various steps for the design of linear phase IIR filters by impulse invariance technique.

Q9) Discuss basic architecture of ADSP series of digital signal processors.

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Roll No. ....

Total No. of Questions : 09] [Total No. of Pages : 02

B.Tech. (Sem. - 6th)

DIGITAL SIGNAL PROCESSING

SUBJECT CODE : EC - 308

Paper ID : [A0321]

[Note : Please fill subject code and paper ID on OMR]

Time : 03 Hours Maximum Marks : 60

Instruction to Candidates:

1) Section - A is **Compulsory**.
2) Attempt any **Four** questions from Section - B.
3) Attempt any **Two** questions from Section - C.

Section - A

Q1) *(10 x 2 = 20)*

a) What is Gibb's phenomenon?

b) Differentiate floating point and fixed point number system?

c) What is the function of MAC in DSP processors?

d) State scaling property of z transform?

e) Determine the z-transform of the following signal and sketch the pole-zero pattern:

$$x(n) = (-1)^n.(2)^{-n}.u(n).$$

f) What is a linear phase filter?

g) Explain causal and non-causal LTI systems. Give examples of each?

h) Explain sampling function or sinc function.

i) What is the relation between z transform and laplace transform?

j) What are the various methods to find out inverse z transform?

Section - B

(4 x 5 = 20)

Q2) What are the advantages of FIR filters over IIR filters?

Q3) State and prove convolution property of z transform?

Q4) Obtain the cascade realization of the system characterized by transfer function

$$H(z) = 2(z+2) / z(z-0.1)(z+0.5)(z+0.4)$$

Q5) State five properties of Discrete Fourier Transform (DFT)?

Q6) Given an analog transfer function as

$$H(s) = 1 / (s+1)(s+2)$$

Obtain H(z) using impulse invariant method. Take T=1s.

Section - C

(2 x 10 = 20)

Q7) Find the inverse DFT of $X(k) = \{1,2,3,4\}$

Q8) Draw the architecture of TMS 320C5x?

Q9) (a) Show that for LTI discrete-time system to stable, all the poles should lie within the unit circle?

(b) Write a short note on Bilinear transformation method?

Roll No. ....

Total No. of Questions : 09] [Total No. of Pages : 02

B.Tech. (Sem. - 6^{th})

DIGITAL SIGNAL PROCESSING

SUBJECT CODE : EC-308

Paper ID : [A0321]

[Note: Please fill subject code and paper ID on OMR]

Time : 03 Hours **Maximum Marks : 60**

Instruction to Candidates:

1) Section - A is **Compulsory**.
2) Attempt any **Four** questions from Section - B.
3) Attempt any **Two** questions from Section - C.

Section - A

(10 × 2 = 20)

Q1) a) What is the disadvantage of using DSPs if signal with large bandwidth are involved?

b) What is scaling of discrete time signals?

c) What is the difference between static and dynamic discrete time signals?

d) Describe linearity property of Z transform.

e) Define Circular symmetric of a sequence in DFT.

f) What is the computational advantage of FFT?

g) What is the basic difference between cascade form and direct form structures for FIR systems?

h) In what cases FIR filters will be preferred over IIR filters?

i) What will happen if length of windows is increased in design of FIR filters?

j) Write the basic difference between ADSP and TMS series of processors.

Section - B

(4 × 5 = 20)

Q2) Describe basic elements of DSP systems with block diagram.

Q3) Compute convolution of $y(n)$ of the signals

$$X(n) = \begin{cases} \alpha^n, -3n \le n \le 5 \\ 0, \text{ elsewhere} \end{cases}$$

$$h(n) = \begin{cases} 1, 0 \le n \le 4 \\ 0, \text{ elsewhere} \end{cases}$$

Q4) Find Z-transform of the following discrete time sequences

$X(n) = \sin(nwT)$, $n = 0,1,$ ....

Q5) A finite duration sequence of length L is given as

$$X(n) = \begin{cases} 1, 0 \le n \le L-1 \\ 0, \text{ otherwise} \end{cases}$$

Determine N-point DFT of this sequence for $N \ge L$.

Q6) Discuss DIT and DIF algorithms and also compare the two algorithms.

Section - C

(2 × 10 = 20)

Q7) (a) Describe Cascade form structure for FIR system.

(b) Discuss quantization of filter coefficients in design of IIR and FIR filter.

Q8) Discuss design of FIR filter using window method. Also compare design using Kaiser and Hanning Windows.

Q9) (a) Why frequency transformation in analog domain is done? Discuss in detail.

(b) Discuss architecture of ADSP processor using a block diagram.

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