Code No.2356/N

FACULTY OF INFORMATICS

BE 3/4 (IT) I-Sem (New) (Main) Examinations, November / December 2012

Subject: Digital Signal Processing

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

Note: *1. Answer All questions from Part-A & Any Five questions From Part-B*
2. Assume suitable missing data of any

Part-A

1. Compare DFT and FFT. (3)
2. Show how FFT of a N-point sequence can be computed using two N/2 point DFT's. (3)
3. State the condition for a filter to have constant group delay interms of impulse response. (3)
4. A non casual FIR filter cannot be realized. Is it true? Justify. (2)
5. What is the transformation required to transform a LPF to BPF. (3)
6. Can you use Direct form –II to implement a 20th order filter. Justify. (2)
7. Significance of Barnel Shifter. (2)
8. List the on-chip peripherals. (2)
9. Properties of auto correlation. (3)
10. Huffman decoding. (2)

Part-B

11. a) A System is described by the difference equation (6)
    Y(n) = 3y(n-1) + 2y (n-2) + x(n).
    (i) Find the impulse response of the system
    (ii) Is it stable.
    b) Compute the DFT of x(n) = [4 3 2 1] (4)

12. a) Show that FIR filters have linear phase characteristics. (5)
    b) Design a FIR band pass filter of length 11 to approximate the ideal characteristics with pass band cut off frequencies at 500 Hz and 600 Hz use Hamming window. (5)

13. a) Compare FIR and IIR filters (4)
    b) Convert the analog filter $H_a(s)$ to a digital filter using impulse invariant method, when (6)

$$H_a(s) = \frac{s+2}{s2+3s+6}$$

14. a) Explain what is meant by precarping. (2)
    b) Design a digital Butter worth high pass filter to meet the following specifications. (8)
    pass band cutoff frequency : 1000 Hz .
    Stop band cutoff frequency : 200 Hz .
    Alternation in PB $\leq$ 3 dB
    Alternation in SB $\geq$ 10 dB

15. a) Draw the block diagram of 54 x X processor and explain its salient features (5)
    b) Explain the pipelining stages of 54 x X processor. (5)

16. a) Draw the model of speech generating system and explain (5)
    b) Write the assembly code to estimate the pitch period (5)

17. Write short notes on any two (10)
    (i) Equiripple FIR filters
    (ii) Round off errors in digital filters
    (iii) Data addressing modes

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Code No. 6313

FACULTY OF INFORMATICS

B.E. 3/4 (I.T.) I-Semester (Main) Examination, November/December, 2009

Subject : DIGITAL SIGNAL PROCESSING

Time : 3 Hours] [Max. Marks : 75

Note : *Answer* ***all*** *questions from* **Part-A**. *Answer* ***five*** *questions from* **Part-B**.

PART - A (25 Marks)

1. Determine whether or not each of the following signals is periodic, if so find fundamental period. 2

   (a) $x(n) = \cos\left(\frac{n}{8}\right)\cos\left(\frac{\pi n}{8}\right)$

   (b) $x(n) = \cos\left(\frac{\pi n}{2}\right) - \sin\left(\frac{\pi n}{8}\right) + 3\cos\left(\frac{\pi n}{4} + \frac{\pi}{3}\right)$

2. Show that the system $y(n) = n\, y(n-1) + x(n)$, is time variant or not. 2

3. Find IDFT of $x(k) = \{1, 1\}$. 2

4. State and Prove Persaval's theorem. 3

5. What are the advantages and disadvantages of FIR filters over IIR filters ? 3

6. Explain Gibb's phenomenon. 3

7. Distinguish between Butterworth and Chebyshev filters. 3

8. Why stable IIR filters can not have linear phase characteristics ? 2

9. Distinguish between voiced and unvoiced speech. 2

10. What are the model parameters to be estimated in the speech analysis system ? 3

Contd...2

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PART - B

(50 Marks)

11. (a) Determine the response $y(n)$, $n \geq 0$ of the system described by difference equation 5

$$y(n) - 3y(n-1) - 4y(n-2) = x(n) + 2x(n-1)$$

When the input sequence is

$$x(n) = \left(\frac{1}{4}\right)^n u(n)$$

(b) Show that the energy of real valued energy signal is equal to the sum of the energies of it's even and odd components. 5

12. (a) Determine the energy density spectrum $s_{xx}(w)$ of the signal 5

$$x(n) = a^n u(n) \quad -1 < a < 1$$

and sketch $s_{xx}(w)$ for $a = 0.5$

(b) Determine the linear convolution of the sequences where 5

$x(n)$ z $\{1, -1, 3, -1, 6\}$ and $h(n)$ z $\{1, 1, 2\}$.

13. (a) Use four point DFT and IDFT to determine the sequence $x_3(n) = x_1(n) Ⓝ x_2(n)$ 5

Where $x_1(n) = \{1, 2, 3, 1\}$

$x_2(n) = \{4, 3, 2, 2\}$

(b) Compute the 8-point DFT of the sequence 5

$$x(n) = \begin{cases} 1, & 0 \leq n \leq 7 \\ 0, & \text{otherwise} \end{cases}$$

by using the DIT-FET signal flow diagram.

14. (a) Using Rectangular window technique, design an ideal Low Pass Filter with unity passband gain, cut-off frequency of 1000 Hz and working at a sampling frequency of 5000 Hz. The length of impulse response should be 7. 5

(b) Realize the system using minimum number of multipliers 5

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

Contd...3

15. Design a digital Butterworth LPFilter to meet the specifications 10

Passband ripple : $\leq$ 1 dB

Passband edge : 4 kHz

Stopband attenuation : $\geq$ 40 dB

Stopband edge : 6 kHz

Sample rate : 24 kHz

using bilinear Transformation method.

16. (a) Explain the model of speech production. 5

(b) Explain the operation of channel vocoder with neat block diagram. 5

17. (a) Determine the Direct form-II realization for a 2^{nd} order IIR system. 5

$$H(z) = \frac{1+2^{-1}}{1-0.5\,z^{-1}+0.06\,z^{-2}}$$

(b) Find the output $y(n)$ of a filter whose impulse response is $h(n) = \{1, 1, 1\}$ and input signal $x(n) = \{2, 1, 0, 1, -1, 2, 0, 1, 2, 1\}$ using overlap-save method. 5

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Code No. : 3059

FACULTY OF INFORMATICS
B.E. 3/4 (IT) I Semester (Main) Examination, December 2010
DIGITAL SIGNAL PROCESSING

Time : 3 Hours] [Max. Marks : 75

*Note : Answer **all** questions from Part – **A**. Answer **any five** questions from Part – **B**.*

PART – A **(25 Marks)**

1. List the advantages of DSP. 3
2. Verify causality and invariance of a system
   y (n) – 10y (n – 1) = 5x(n) + 2x(n – 1). 2
3. Give the differences and similarities between DIT-FFT and DIF-FFI algorithm. 3
4. What is the significance of Z-transforms in DSP ? 3
5. Give the pole mapping property of impulse invariance. 3
6. What is Gibb's phenomenon ? 2
7. Define limit cycle oscillation. 2
8. Prove the symmetry property of DFT. 3
9. What is a canonical structure ? 2
10. Distinguish between voiced and unvoiced sounds. 2

PART – B **(50 Marks)**

11. a) The discrete time system y(n) = α x (– n) where α is a non-zero constant. Determine, whether or not the system defined by the given input-output relationship is
    a) linear b) Casual c) Shift invariant 5
    b) Derive energy density spectrum. 5

12. a) Find the linear convolution of the sequences
    x(n) = {1, 0, 1, 1, 2, 1, –1, 0, 1, 2, 1} and h(n) = {1, 2, 1} using overlap add method. 5
    b) Derive the radix-2 DIF-FFT algorithm. 5

13. Design an ideal bandpass filter with a frequency response

$H_d(e^{jw}) = 1$ for $\frac{\pi}{4} \leq |w| \leq \frac{3\pi}{4}$ $= 0$ otherwise

Using Hamming window for N = 11. **10**

14. Design a Butterworth low pass filter with the specifications $\alpha_p = I$ dB ripple on the pass band $0 \leq \omega \leq 0.2\pi$ $\alpha_s = 15$ dB in the stop band $0.3\pi \leq \omega \leq \pi$. Using Impulse Invariance method. **10**

15. a) Explain short-time spectrum analysis. **5**

b) Give the block diagram of channel vocoder. **5**

16. a) Compute 4-point DFT of a sequence x (n) = {0, 1, 2, 3} using DIT algorithm. **5**

b) Find the circular convolution of the two sequences $x_1(n) = \{1, 2, 2, 1\}$ and $x_2(n) = \{1, 2, 3, 1\}$. **5**

17. a) Obtain a cascade realization of the following system using first order sections whenever possible $y(n) - \frac{3}{4}y(n-1) + \frac{1}{8}y(n-2) = x(n) + \frac{1}{3}x(n-1)$. **5**

b) What are the desirable characteristics of a window ? Why is it necessary for FIR filter design ? **5**

Code No. : 5284

FACULTY OF INFORMATICS

B.E. 3/4 (IT) (I Semester) (Main) Examination, December 2011

DIGITAL SIGNAL PROCESSING

Time : 3 Hours] [Max. Marks : 75

Note : *Answer all questions from Part A.*
Answer any five questions from Part B.

PART – A (25 Marks)

1. Give the impulse response of a LTI system. 2
2. Check the signal $x(n) = \sin(n\pi/2)$ for energy, power. 3
3. Write the computation steps of IDFT through FFT and derive the formula. 3
4. Find the circular convolution of $x_1(n) = \{1, 2, 2, 1\}$ and $x_2(n) = \{0, 1, 2, 3\}$. Using matrix method. 2
5. Define and explain frequency wrapping. 3
6. Find H(z) using bilinear transform for a system whose $H(S) = \dfrac{1}{4S+3}$ with T = 1sec. 2
7. Explain Gibb's phenomenon. 2
8. Distinguish between FIR and IIR filters. 3
9. Give a model of speech production. 2
10. Explain short time spectrum analysis briefly. 3

PART – B (50 Marks)

11. a) Find the impulse response, magnitude and phase response of the given second order system. 6

$$y(n) - y(n-1) + \frac{3}{16}y(n-2) = x(n) - \frac{1}{2}x(n-1)$$

b) Check whether the following systems are causal, linear 4

i) $T[x(n)] = x(n-n_o)$ ii) $T[x(n)] = e^{x(n)}$

12. a) Outline steps in implementing DIF FFT algorithm for an N-point sequence. 4

b) Compute the FFT of $x(n) = \{1, 2, 3, 4, 4, 3, 2, 1\}$ using DIT FFT algorithm. 6

13. Design a digital butterworth lowpass filter using bilinear transformation for the following specification. Assume suitable value of T. 10

pass-band attenuation $= -3$ dB , $0 \le \omega \le 0.2\pi$

stop-band attenuation $= -15$ dB , $0.3\pi \le \omega \le \pi$

14. a) Why do we use windows in FIR filter design. 3

b) Design an ideal HPF filter with

$$H(e^{j\omega}) = \begin{cases} 1 & -\pi \le \omega \le -3\pi/4 \text{ and } 3\pi/4 \le \omega \le \pi \\ 0 & \text{Elsewhere} \end{cases}$$

Find h(n) for N = 7 using rectangular window. 7

15. a) Explain briefly speech analysis and synthesis. 5

b) What are the methods of exciation sources for pitch detection. Explain. 5

16. a) Explain the advantages of using FFT for computing DFT. Compare the number of multiplications and additions for N = 256 and N = 512. 5

b) Prove the properties of time convolution and frequency convolution of DFT. 5

17. a) Explain the method of designing FIR filters using Fourier series method. 5

b) Find the Fourier transform of $x(n) = \left(\frac{1}{4}\right)^n u(n+4)$. 5

08-S131

Code No. : 6313

FACULTY OF INFORMATICS

B.E. 3/4 (I.T.) I Semester (Supplementary) Examination, July 2010

DIGITAL SIGNAL PROCESSING

Time : 3 Hours] [Max. Marks : 75

*Note : Answer **all** questions from Part **A**, Answer **five** questions from Part **B**.*

PART – A 25

1. Give the classification of Discrete-Time Systems. 3
2. Sketch the response of ideal LPF, HPF, BPF. 2
3. Write the Symmetry Properties of DFT. 3
4. What is inplace computation in FFT ? 2
5. Write the characteristics of different windows used in the design of FIR filters. 3
6. What is Gibbs Phenomenon ? 2
7. Obtain H(z) by impulse invariance for a system whose transfer function is given by

   $H(S) = \frac{1}{S^2 - 1}$ and T = 1 sec. 3
8. Compare FIR and IIR filters. 2
9. What are voiced and unvoiced decisions and detections ? 3
10. Draw the model of Speech Production System. 2

PART – B **50**

11. a) Give the classification of Discrete-Time Signals. **4**

b) What is the impulse response of a LTI system ? Obtain the impulse response of the following causal systems. **6**

i) $y(n) - 3y(n - D - 4\,y(n-2) = x(n) + 2x(n-1)$

ii) $y(n) = 0.6\,y(n-1) - 0.08\,y(n-2) + x(n)$.

12. a) Compute the 8-point DFT of the sequence $x(n) = \{-1, -1, 2, 2, -1, -1, 2, 2\}$ using radix-2 DIF FFT. **7**

b) Find the circular convolution of the following sequences

$x(n) = \{3, 4, 5, 6\}$ and $y(n) = \{1, 2\}$. **3**

13. a) Design a Digital ideal HPF to meet the following specifications.

Cutoff frequency = 250 Hz ; sampling frequency = 1 KHz;

Filter length = 7; use rectangular window. **6**

b) Realise the following systems with minimum no. of multipliers. **4**

i) $H(z) = 1 + \frac{1}{2}z^{-1} + \frac{1}{2}z^{-2} + z^{-3}$ ii) $H(z) = \left(1 + \frac{1}{2}z^{-1} + z^{-2}\right)\left(1 + \frac{1}{4}z^{-1} + z^{-2}\right)$.

14. Design a digital IIR Butterworth LPF to meet the following specifications.

$W_c = 1$ KHz ; $W_s = 5$ KHz ; pass band ripple $\leq$ 3dB,

Stop band ripple $\geq$ 15 dB ; $F_s = 10$ KHz. Use Bilinear transformation. **10**

15. a) Give a brief account on speech analysis and synthesis. **5**

b) What is Vocoder ? Give the analysis of Vocoder. **5**

16. a) Show that FIR filters exhibit linear phase response ? **5**

b) Obtain the linear convolution using DFT for the sequences

$x(n) = \{1, \underset{\uparrow}{1}, 1\}$ and $h(n) = \{-1, \underset{\uparrow}{1}, -1\}$ **5**

17. Write short notes on :

i) Computation steps of IDFT using radix-2 FFT. **4**

ii) Comparison of Bilinear and Impulse Invariant Techniques. **3**

iii) Filtering of long data sequences. **3**

500

Code No. : 6313

FACULTY OF INFORMATICS

B.E. 3/4 (I.T.) I Semester (Supplementary) Examination, July 2010

DIGITAL SIGNAL PROCESSING

Time : 3 Hours] [Max. Marks : 75

*Note : Answer **all** questions from Part **A**, Answer **five** questions from Part **B**.*

PART – A 25

1. Give the classification of Discrete-Time Systems. 3
2. Sketch the response of ideal LPF, HPF, BPF. 2
3. Write the Symmetry Properties of DFT. 3
4. What is inplace computation in FFT ? 2
5. Write the characteristics of different windows used in the design of FIR filters. 3
6. What is Gibbs Phenomenon ? 2
7. Obtain H(z) by impulse invariance for a system whose transfer function is given by

   $H(S) = \frac{1}{S^2 - 1}$ and T = 1 sec. 3

8. Compare FIR and IIR filters. 2
9. What are voiced and unvoiced decisions and detections ? 3
10. Draw the model of Speech Production System. 2

PART – B

50

11. a) Give the classification of Discrete-Time Signals. 4

b) What is the impulse response of a LTI system ? Obtain the impulse response of the following causal systems. 6

i) $y(n) - 3y(n - D - 4\,y(n - 2) = x(n) + 2x(n - 1)$

ii) $y(n) = 0.6\,y(n - 1) - 0.08\,y(n - 2) + x(n)$.

12. a) Compute the 8-point DFT of the sequence $x(n) = \{-1, -1, 2, 2, -1, -1, 2, 2\}$ using radix-2 DIF FFT. 7

b) Find the circular convolution of the following sequences

$x(n) = \{3, 4, 5, 6\}$ and $y(n) = \{1, 2\}$. 3

13. a) Design a Digital ideal HPF to meet the following specifications.

Cutoff frequency = 250 Hz ; sampling frequency = 1 KHz;

Filter length = 7; use rectangular window. 6

b) Realise the following systems with minimum no. of multipliers. 4

i) $H(z) = 1 + \frac{1}{2}z^{-1} + \frac{1}{2}z^{-2} + z^{-3}$ ii) $H(z) = \left(1 + \frac{1}{2}z^{-1} + z^{-2}\right)\left(1 + \frac{1}{4}z^{-1} + z^{-2}\right)$.

14. Design a digital IIR Butterworth LPF to meet the following specifications.

$W_c = 1$ KHz ; $W_s = 5$ KHz ; pass band ripple $\leq$ 3dB,

Stop band ripple $\geq$ 15 dB ; $F_s = 10$ KHz. Use Bilinear transformation. 10

15. a) Give a brief account on speech analysis and synthesis. 5

b) What is Vocoder ? Give the analysis of Vocoder. 5

16. a) Show that FIR filters exhibit linear phase response ? **5**

b) Obtain the linear convolution using DFT for the sequences

$x(n) = \{1, \underset{\uparrow}{1}, 1\}$ and $h(n) = \{-1, \underset{\uparrow}{1}, -1\}$ **5**

17. Write short notes on :

i) Computation steps of IDFT using radix-2 FFT. **4**

ii) Comparison of Bilinear and Impulse Invariant Techniques. **3**

iii) Filtering of long data sequences. **3**

500

Code No. : 6316

FACULTY OF INFORMATICS

B.E. 3/4 (I.T.) I Semester (Suppl.)Examination, July 2010

THEORY OF AUTOMATA

Time: 3 Hours] [Max. Marks : 75

*Note : Answer **all** questions from Part A. Answer **five** questions from Part **B**.*

PART – A 25

1. Define E-Closure. 2
2. Write applications of FA. 2
3. Define left most and rightmost Derivations. 3
4. Give the properties of CFL's. 2
5. What is an useless symbol ? 3
6. What is restricted turing machine ? 2
7. What is Greibach Normal form ? 2
8. List the programming techniques for turing machines. 3
9. Construct E-NFA for the Regular expression 11 + 0*. 3
10. State the pumping lemma for CFL's. 3

PART – B (5×10=50)

11. Convert the following DFA to a regular expression using state elimination technique. 10

	0	1
→*P	S	P
q	P	S
r	r	q
s	q	r

(This paper contains 2 pages)

P.T

12. a) State the pumping lemma for regular languages and prove that the following language. 7
L = $\{O^n/n$ is a perfect square $\}$ is not regular.

b) Write the closure properties of regular languages. 3

13. a) Construct a PDA equivalent to the grammar 5

$S \rightarrow aAA$

$A \rightarrow aS/bS/a$

b) Convert the following Grammar to Chamsky Normal Form (CNF) : 5

$S \rightarrow bA / aB$

$A \rightarrow a / aS / bAA$

$B \rightarrow b / bS / aBB$

14. Explain different types of Turing Machines. 10

15. a) What is PCP and MPCP ? and test whether the following PCP instance has a solution or not. 8

A = (ab, a, bc, c)

B = (bc, ab, ca, a)

b) What is undecidle problem ? 2

16. Convert the following PDA to a CFG. 10

$P = (\{p, q\}, \{0, 1\}, \{x_1 Z_0\}, \delta, q, z_0)$ where δ is given by

1) $\delta(q, 1, z_0) = \{(q, XZ_0)\}$

2) $\delta(q, 1, x) = \{(q, xx)\}$

3) $\delta(q, 0, x) = \{(p, x)\}$

4) $\delta(q, E, x) = \{(q, E)\}$

5) $\delta(P, 1, X) = \{(P, E)\}$

6) $\delta(P, 0, Z_0) = \{(q, z_0)\}$

17. Write short notes on :

a) Decision properties of CFL's. 3

b) Recursive and Recursively enumerable languages. 3

c) P and NP classes of problems. 4

Code No. 3059

FACULTY OF INFORMATICS

B.E. 3/4 (IT) First Semester (Suppl.) Examination, June/July 2011

DIGITAL SIGNAL PROCESSING

Time : Three Hours] [Maximum Marks : 75

Note :— Answer **ALL** questions from Part A. Answer any **FIVE** questions from Part B.

PART—A (Marks : 25)

1. Define and give the different types of filters with waveforms. 3
2. Find the impulse response x(n) of the system

   $y(n) = 0{\cdot}5\ y(n-1) + 6\ y(n-2) + x(n).$ 2
3. Describe the bit-reversal order in DIF FFT and DIT FFT. 3
4. What is zero padding ? How do you convert a linear convolution to circular convolution ? 3
5. What is aliasing effect due to impulse invariant transform ? 2
6. Draw the structure of $H(z) = \dfrac{(0{\cdot}5+2z^{-1})(1+0{\cdot}6z^{-1})}{(z^{-1}+2)(4+z^{-1})}$ using Cascade method. 3
7. Mention the basic principle of FIR digital filters. 3
8. Distinguish between different types of windows. 2
9. What are voiced sounds ? Explain. 2
10. How is speech signal generated ? 2

PART—B (Marks : 50)

11. A second order discrete time system is characterized by the difference equation. Find the impulse response, magnitude and phase response of the given second order system

    $y(n) - 0{\cdot}1\ y(n-1) - 0{\cdot}02\ y(n-2) = 2\ x(n) - x(n-1).$

    Determine $y(n) \geq 0$ when $x(n) = u(n)$ and the initial conditions are $y(-1) = -10$ and $y(-1) = -5$. 10

12. (a) Determine the response of discrete LTI system if $x(n) = \{1, 2, 3, 4\}$ and $h(n) = \{1, 2, 1, 2\}$, using DFT approach. 7

(b) Prove that Parseval's relation for the DFT given by :

$$\sum_{n=0}^{N-1} |x(n)|^2 = \frac{1}{N}\sum_{K=0}^{N-1} |X(K)|^2 .$$ 3

13. (a) Explain the relation between analog and digital filter poles on impulse invariant transform. 4

(b) Convert the following analog filter with transfer function Ha(s) using bilinear transform. Draw the structure of IIR filter :

$$Ha(s) = \frac{0{\cdot}2}{(s+0{\cdot}2)^2+16}.$$ 6

14. Design a 5^{th} order band pass linear phase filter for the following specifications. Draw the filter structure :

Lower-cut-off frequency = $0{\cdot}4\ \pi$ rad/sec.

Upper-cut-off frequency = $0{\cdot}6\ \pi$ rad/sec.

Window type = Hamming. 10

15. (a) Write a brief note on channel vocoder. 5

(b) Explain the digital model of speech production with a block diagram. 5

16. (a) Perform the linear convolution of the sequences :

$x(n) = \{-1, 1, 2, -1, 1, 2, -1, 1\}$ and $h(n) = \{2, 3, -2\}$ using overlap-Add method. 7

(b) Define causal, linear and time-variant systems. 3

17. (a) Draw the butterfly structure of 8-point DFT using Radix-2 DIF FFT algorithm. 5

(b) Write a design note on linear face FIR filter using windows. 5

Code No. 3059

FACULTY OF INFORMATICS

B.E. 3/4 (IT) First Semester (Suppl.) Examination, June/July 2011

DIGITAL SIGNAL PROCESSING

Time : Three Hours] [Maximum Marks : 75

Note :— Answer **ALL** questions from Part A. Answer any **FIVE** questions from Part B.

PART—A (Marks : 25)

1. Define and give the different types of filters with waveforms. 3
2. Find the impulse response x(n) of the system

   $y(n) = 0{\cdot}5\ y(n-1) + 6\ y(n-2) + x(n).$ 2
3. Describe the bit-reversal order in DIF FFT and DIT FFT. 3
4. What is zero padding ? How do you convert a linear convolution to circular convolution ? 3
5. What is aliasing effect due to impulse invariant transform ? 2
6. Draw the structure of $H(z) = \dfrac{(0{\cdot}5+2z^{-1})(1+0{\cdot}6z^{-1})}{(z^{-1}+2)(4+z^{-1})}$ using Cascade method. 3
7. Mention the basic principle of FIR digital filters. 3
8. Distinguish between different types of windows. 2
9. What are voiced sounds ? Explain. 2
10. How is speech signal generated ? 2

PART—B (Marks : 50)

11. A second order discrete time system is characterized by the difference equation. Find the impulse response, magnitude and phase response of the given second order system

    $y(n) - 0{\cdot}1\ y(n-1) - 0{\cdot}02\ y(n-2) = 2\ x(n) - x(n-1).$

    Determine $y(n) \geq 0$ when $x(n) = u(n)$ and the initial conditions are $y(-1) = -10$ and $y(-1) = -5$. 10

12. (a) Determine the response of discrete LTI system if x(n) = {1, 2, 3, 4} and h(n) = {1, 2, 1, 2}, using DFT approach. 7

(b) Prove that Parseval's relation for the DFT given by :

$$\sum_{n=0}^{N-1} |x(n)|^2 = \frac{1}{N}\sum_{K=0}^{N-1} |X(K)|^2 .$$ 3

13. (a) Explain the relation between analog and digital filter poles on impulse invariant transform. 4

(b) Convert the following analog filter with transfer function Ha(s) using bilinear transform. Draw the structure of IIR filter :

$$Ha(s) = \frac{0{\cdot}2}{(s+0{\cdot}2)^2+16}.$$ 6

14. Design a 5^{th} order band pass linear phase filter for the following specifications. Draw the filter structure :

Lower-cut-off frequency = 0·4 π rad/sec.

Upper-cut-off frequency = 0·6 π rad/sec.

Window type = Hamming. 10

15. (a) Write a brief note on channel vocoder. 5

(b) Explain the digital model of speech production with a block diagram. 5

16. (a) Perform the linear convolution of the sequences :

x(n) = {–1, 1, 2, –1, 1, 2, –1, 1} and h(n) = {2, 3, –2} using overlap-Add method. 7

(b) Define causal, linear and time-variant systems. 3

17. (a) Draw the butterfly structure of 8-point DFT using Radix-2 DIF FFT algorithm. 5

(b) Write a design note on linear face FIR filter using windows. 5

Code No.2206/O

FACULTY OF INFORMATICS

BE 3/4 (IT) I-Sem (Old) Examinations, November / December 2012

Subject: Digital Signal Processing

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

Note: *Answer all questions from Part-A & any five questions from Part-B*

Part-A

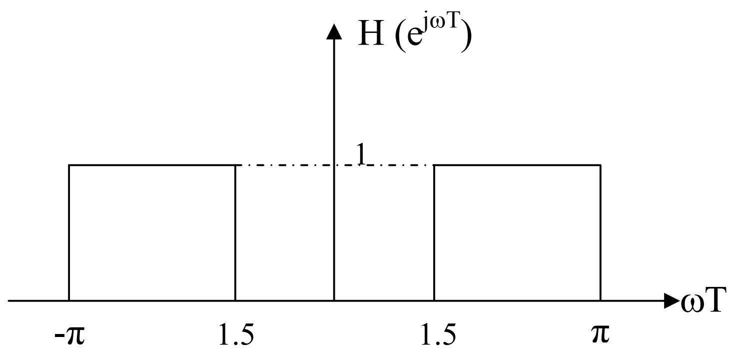
1. Distinguish between Analog and digital frequencies (2)
2. Draw the block diagram of digital signal processing system. (2)
3. A digital system is specified by y(n) = ny(n-1)+x(n) is it a LTI system? (3)
4. Give the properties of DFT (3)
5. Compare DFT & FFT (3)
6. What is meant by 'in-place' computation (2)
7. Distinguish between IIR & FIR filters (3)
8. Explain Gibb's Phenomenon (3)
9. Differentiate symmetric and antisymmetric FIR filters (2)
10. What is vocoder? Why it is needed? (2)

Part-B

11. What is meanth by stability, casuality, and time invariance?
    a) Give example systems? (5)
    b) Find and sketch the magnitude spectrum of a signal defined as the convolution of x(n) = u(n)-u(n-5) and h(n) = u(n)-u(n-5) (5)

12. a) Explain overleaf save method of performing linear convolution with an example (3)
    b) Illustrate the development of radix –2 DIT-FFT algorithm (7)

13. Design a FIR digital HPF using Hamming window of length 9 having the ideal characteristics shown in figure. (10)



14. Design a digital chebysher low pass filter for the following specifications. Use Belinear Transformation method. Passband frequency:1 KHZ, Passband ripple $\leq$3db, stop band frequency:5 KHZ, stop band attenuation $\geq$ 15dB Sampling frequency = 10 KHZ (10)

15. a) Explain about speech analysis and synthesis (5)
    b) Explain the speech production mechanism and develop the model for the same (5)

16. a) Relate circular convolution and linear convolution (5)
    b) Find the DFI of a sequence x(n) = {1,1,0,0} for N=4 (5)

17. Write short notes on:
    a) Round off effects (5)
    b) Structures for IIR & FIR systems (5)

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08/05/18 FN

Reg. No. :

Question Paper Code : 40962

B.E./B.Tech. DEGREE EXAMINATION, APRIL/MAY 2018

Fifth/Sixth Semester

Electronics and Communication Engineering

EC 6502 – PRINCIPLES OF DIGITAL SIGNAL PROCESSING

(Common to Biomedical Engineering/Medical Electronics)

(Regulations 2013)

Time : Three Hours | Maximum : 100 Marks

Answer ALL questions

PART – A | (10×2=20 Marks)

1. Calculate the 4-point DFT of the sequence $x(n) = \{\underset{\uparrow}{1} \quad 0 \quad -1 \quad 0\}$.

2. What is the relationship between Fourier transform and DFT ?

3. What are the methods used for digitizing the analog filter into a digital filter ?

4. What is meant by frequency warping ?

5. Draw the direct form realization of FIR system.

6. How the zeros in FIR filter is located ?

7. Distinguish between fixed point arithmetic and floating point arithmetic.

8. Why is rounding preferred over truncation in realizing a digital filter ?

9. Show that the up sampler and down sampler are time invariant system.

10. Write the expression for the output y(n) as a function of the input x(n) for the given multirate system as in Figure 1.

$$x(n) \rightarrow \boxed{\uparrow 5} \rightarrow \boxed{\downarrow 10} \rightarrow \boxed{\uparrow 2} \rightarrow y(n)$$

Figure 1

PART – B (5×13=65 Marks)

11. a) i) State and prove any four properties of DFT. (8)

ii) Perform circular convolution of the following sequences $x_1(n) = \{1\ \ 1\ \ 2\ \ 1\}$; $x_2(n) = \{1\ \ 2\ \ 3\ \ 4\}$. (5)

(OR)

b) i) Mention the differences and similarities between DIT and DIF algorithms. (5)

ii) Compute 4 point DFT of a sequence $x(n) = \{0\ \ 1\ \ 2\ \ 3\}$ using DIF and DIT algorithms. (8)

12. a) i) Design an analog Butterworth filter for a given specifications. (7)

$0.9 \le |H(j\Omega)| \le 1$ for $0 \le \Omega \le 0.2\pi$.

$|H(j\Omega)| \le 0.2$ for $0.4\pi \le \Omega \le \pi$.

ii) Apply impulse invariant method and find H(z) for $H(s) = \dfrac{s+a}{(s+a)^2 + b^2}$. (6)

(OR)

b) i) Apply bilinear transformation to $H(s) = \dfrac{2}{(s+1)(s+2)}$ with T = 1 sec and find H(z). (6)

ii) Explain the Lattice-Ladder structure with neat diagram. (7)

13. a) Write the expression for the frequency response of Rectangular window and Hamming window and explain. (7+6)

(OR)

b) Determine the filter coefficients h(n) obtained by sampling

$$H_d(e^{j\omega}) = e^{-j(N-1)\omega/2} \qquad 0 \le |\omega| \le \frac{\pi}{2}$$

$$= 0 \qquad \frac{\pi}{2} \le |\omega| \le \pi$$

for N = 7. (13)

14. a) The output signal of an A/D convertor is passed through a first order low pass filter, with transfer function given by $H(z) = \dfrac{(1-a)z}{z-a}$ for $0 \le a \le 1$. Find the steady state output noise power due to quantization at the output of the digital filter. (13)

(OR)

b) Briefly explain the following :

i) Coefficient quantization error. (4)

ii) Product quantization error. (4)

iii) Truncation and Rounding. (5)

15. a) Explain sampling rate conversion by a rational factor and derive input-output relation in both time and frequency domain. (13)

(OR)

b) With neat required diagrams explain any two applications of adaptive filtering. (6+7)

PART – C (1×15=15 Marks)

16. a) An FIR Filter is given by the difference equation

$$y(n) = 2x(n) + \frac{4}{5}x(n-1) + \frac{3}{2}x(n-2) + \frac{2}{3}x(n-3)$$

Determine its lattice form. (15)

(OR)

b) How is signal scaling used to prevent overflow limit cycle in the digital filter implementation ? Explain with an example. (15)