Reg. No. :

Question Paper Code : 40495

B.E./B.Tech. DEGREE EXAMINATIONS, NOVEMBER/DECEMBER 2021.

Fifth/Eighth Semester

Electrical and Electronics Engineering

EE 8591 — DIGITAL SIGNAL PROCESSING

(Common to Electronics and Instrumentation Engineering/Instrumentation and Control Engineering)

(Regulations 2017)

Time : Three hours Maximum : 100 marks

Answer ALL questions.

PART A — ($10 \times 2 = 20$ marks)

1. What is meant by Nyquist rate? Comment on aliasing effect.

2. When is a system said to be causal and stable?

3. State convolution theorem wrt Z-transform.

4. List any two properties of DTFT, with relevant expression.

5. Compare DFT and FFT based on computational complexity.

6. Draw the basic DIT butterfuly diagram marking input and output with twiddle factor.

7. Write the windowing function due to Blackmann for FIR filter design.

8. What is meant by prewarping? Give remedial measure to prevent this.

9. Differentiate Von Numann and Harvard architecture

10. Give the different addressing formats of digital signal processors.

PART B — ($5 \times 13 = 65$ marks)

11. (a) For the system described by $y(n) = x(n)x(n-2)$, check whether it is static/dynamic, causal/non-casual, linear/non-linear and time invariant/time variant. (13)

Or

(b) (i) Represent the sequence $x(n) = (3,1,-2,1,4,2,5,1)$ for $n = -3,-2,-1,0,1,2,3,4$ as weighted sum of unit impulses. (6)

(ii) Check whether the signal $x(n) = u(n) - u(n-6)$ is an energy signal or power signal. (7)

12. (a) Find the linear convolution of the two signals represented by (13)

$$x(n) = \begin{cases} 2 & for \quad n = -2,\ 0,1 \\ -1, & for \quad n = -1 \\ 0 & elsewhere \end{cases}$$

and $h(n) = \delta(n) - 2\delta(n-1) + 3\delta(n-2) - \delta(n-3)$.

Or

(b) Find all possible inverse Z-transforms of $X(z) = [z(z^2 - 4z + 5)]/(z^3 - 6z^2 + 11z - 6)$. (13)

13. (a) (i) Find the frequency response, magnitude response and phase response of $y(n) = x(n) + 0.81x(n-1) + 0.81x(n-2) - 0.45x(n-3)$. (6)

(ii) Find the 4 point DFT using matrix method if $x(n) = \{1,-2,3,2\}$. (7)

Or

(b) Compute $X(k)$ of $x(n) = \{1,-1,-1,-1,1,1,1,-1\}$ using radix-2 DIT FFT. Also plot amplitude and frequency spectra. (13)

14. (a) Convert the analog filter with transfer function $H_a(S) = s + 0.1/(s+0.1)^2 + 9$ into digital filter by impulse invariant transformation. (13)

Or

(b) Design a low pass Butterworth digital filter to give response of 3dB or less for frequencies upto 2kHz and an attenuation of 20 dB or more beyond 4kHz. Use the bilinear transformation technique and obtain H(z) of the desired filter. (13)

15. (a) Explain in detail with a neat diagram the architecture of any one of the latest digital signal processors. (13)

Or

(b) (i) Write a brief technical note on commercial digital signal processors. (6)

(ii) Describe in detail any four addressing formats of digital signal processor. (7)

PART C — (1 × 15 = 15 marks)

16. (a) Design an FIR low pass filter satisfying the following specifications :

$(\alpha_p \leq 0.1dB,\ a_s \geq 38\ dB)$

$w_p = 15\ rad/\text{sec},\ w_s = 25\ rad/\text{sec},\ w_{sf} = 80\ rad/\text{sec}$. Use Kaiser window.

Or

(b) The desired frequency response of a low pass filter is

$$H_d\left(e^{jw}\right) = \begin{cases} 1; & -\pi/2 \leq w \leq \pi/2 \\ 0; & \pi/2 \leq w \leq \pi \end{cases}$$

Determine $h_d(n)$ for M=7 using rectangular window.

Reg. No. :

Question Paper Code : X10402

B.E./B.Tech. DEGREE EXAMINATIONS, NOVEMBER/DECEMBER 2020
Fifth Semester
Electrical and Electronics Engineering
EE 8591 – DIGITAL SIGNAL PROCESSING
(Common to Electronics and Instrumentation Engineering)
(Regulations 2017)

Time : Three Hours | Maximum : 100 Marks

Answer ALL questions

PART – A **(10×2=20 Marks)**

1. Show that $\delta(n) = u(n) - u(n-1)$.

2. Given that $x(n) = \begin{cases} 1+\frac{n}{3} & , -3 \le n \le -1 \\ 1 & , 0 \le n \le 3 \\ 0 & , \text{elsewhere} \end{cases}$, sketch the signal $x(-n+4)$.

3. State and prove the time shifting property of Z-transform.

4. State and prove Parseval's theorem in the context of DTFT.

5. Discuss the circular convolution property.

6. Draw the basic butterfly structure of a radix-4 DIT algorithm.

7. Differentiate IIR and FIR filters.

8. Give the basic structure of Direct form II structure for realizing an IIR filter.

9. Give the data formats of any one DSP processor.

10. How is pipelining effected in a DSP processor ?

PART – B **(5×13=65 Marks)**

11. a) Determine whether the following systems are static, linear, time invariant, causal and stable with proper justifications. **(4+4+5)**

   i) $y(n) = x(n) + nx(n + 1)$

   ii) $y(n) = x(-n)$

   iii) $y(n) = \text{sign}(x(n))$

(OR)

b) i) Determine the zero-input response of the difference equation given by the following :

$x(n) - 3y(n - 1) - 4y(n - 2) = 0$ **(6)**

   ii) Determine the impulse response of the following causal system. **(7)**

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

12. a) i) Determine the Z-transform and sketch the ROC of the following signal by applying the appropriate property of the Z-transform wherever necessary. **(7)**

$x(n) = n^2u(n)$

   ii) Determine the inverse Z-transform of **(6)**

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

If x(n) is causal, x(n) is anti-causal.

(OR)

b) i) Determine the magnitude and phase spectra for the following signal by computing its Fourier transform. **(7)**

$x(n) = u(n) - u(n - 6)$

   ii) Consider the following signal, determine its power density spectrum and evaluate the power of the signal. **(6)**

$$x(n) = 2 + 2\cos\frac{n\pi}{4} + \cos\frac{n\pi}{2} + \frac{1}{2}\cos\frac{3n\pi}{4}$$

13. a) i) Discuss the savings in time for a radix-2 DIT algorithm to compute FFT. **(4)**

   ii) Determine the eight point FFT using DIT algorithm. **(9)**

$x(n) = \{1, 1, 1, 1, 1, 1, 0, 0\}$

(OR)

b) Derive the butterfly structure for a radix-2 DIF algorithm that is used to compute FFT. Explain with an example. **(13)**

14. a) A digital low-pass filter is required to meet the following specifications : **(13)**

Pass band ripple : ≤ 1 dB

Pass band edge : 4 kHz

Stop band attenuation : ≥ 40 dB

Stop band edge : 6 kHz

Sample rate : 24 kHz

The filter is to be designed using bilinear transformation on an analog system function. Use Butterworth approximation.

(OR)

b) Determine the coefficients h(n) of a linear-phase FIR filter of length M = 15, which has a symmetric unit sample response and a frequency response that satisfies the following condition : **(13)**

$$H(k) = \frac{2\pi k}{15}\begin{cases} 1, & k = 0, 1, 2, 3 \\ 0.4, & k = 4 \\ 0, & k = 5, 6, 7 \end{cases}$$

15. a) Discuss the architecture of any one DSP processor and explain its features. **(13)**

(OR)

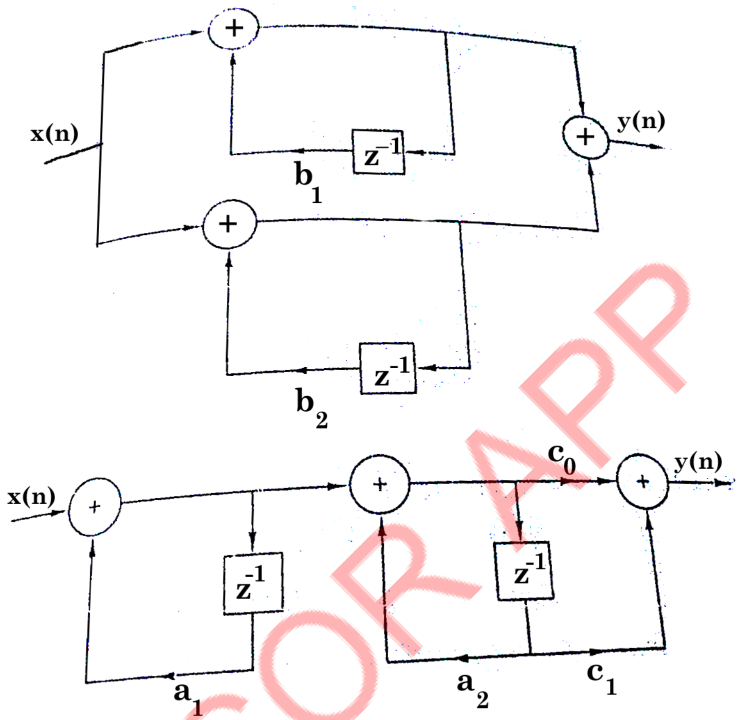
b) Discuss the addressing modes supported by a DSP processor and explain how each is used for various DSP operations. **(13)**

PART – C **(1×15=15 Marks)**

16. a) i) Obtain the cascade and parallel structures for the following system and realize it using Direct form II. **(8)**

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

ii) Determine the coefficients a_1, a_2, c_1, c_0 in terms of b_1 and b_2 so that the two systems in the given figure below are equivalent. **(7)**



(OR)

b) Discuss about implementation of FFT with any suitable digital signal processor. Also write a 'C' program to implement the FFT with the same processor.

Reg. No. :

Question Paper Code : 90207

B.E./B.Tech. DEGREE EXAMINATIONS, NOVEMBER/DECEMBER 2019
Fifth Semester
Electrical and Electronics Engineering
EE 8591 – DIGITAL SIGNAL PROCESSING
(Common to : Electronics and Instrumentation Engineering)
(Regulations 2017)

Time : Three Hours | Maximum : 100 Marks

Answer ALL questions

PART – A **(10×2=20 Marks)**

1. Show that $\delta(n) = u(n) - u(n-1)$ graphically.

2. Show that the product of two even signals or of two odd signals is an even signal while the product of an even signal and an odd signal is an odd signal.

3. State and prove that convolution in the time domain is the same as multiplication in the Z-domain.

4. Determine the magnitude and phase representation for the following system : $y(n) + \frac{1}{4} y(n-1) = x(n) - x(n-1)$.

5. Draw the basic butterfly structure of radix-4 algorithm in the DIT algorithm.

6. Compute the x(n) for the following sequence using DIF FFT algorithm : $\{1, 1-j\sqrt{2}, 1, 1+j-j\sqrt{2}\}$.

7. Justify the usage of Hamming or Hanning window for FIR filter design as against Rectangular window.

8. Differentiate IIR and FIR filters.

9. Justify the usage of Branch, Call and Return instruction in Digital signal processor.

10. Name any 4 assembler directives and their usage in any Digital signal processor.

PART – B (5×13=65 Marks)

11. a) Check whether the following systems are static, linear, time invariant, causal or stable :

i) $y(n) = x(n) \cos w\, n$ (4)

ii) $y(n) = x(n) + n\, x(n + 1)$ (5)

iii) $y(n) =$ Truncation $[x(n)]$ (4)

(OR)

b) Two discrete time systems T1 and T2 are connected in cascade to form a new system T. Prove or disprove the following statements :

i) If T1 and T2 are causal, then T is causal. (3)

ii) If T1 and T2 are linear and time invariant, then interchanging the order doesn't change the system T. (4)

iii) If T1 and T2 are stable, then T is stable. (3)

iv) If T1 and T2 are non-causal, then T is non-causal. (3)

12. a) i) Determine the Z-transform and compute the Region of convergence : $x(n) = e^{-3n}u(n - 1)$ (7)

ii) Use convolution to find x(n), if X(z), is given by (6)

$$X(z) = \frac{1}{\left(1-\frac{1}{2}z^{-1}\right)\left(1+\frac{1}{4}z^{-1}\right)}.$$

(OR)

b) Compute and sketch the convolution and correlation of the two signals (7+6)

$x(n) = \{\underset{\uparrow}{1}, 2, 3, 4\}$ and $h(n) = \{1, 2, 3, \underset{\uparrow}{4}\}$.

13. a) Compute the FFT using DIT algorithm for the following sequence : (13)
$x(n) = \{1, -1, -1, -1, 1, 1, 1, -1\}$

(OR)

b) i) Compute the circular convolution of the following sequences : (7)

$x_1(n) = \{\delta(n) + \delta(n-1) - \delta(n-2) - \delta(n-3)$

$x_2(n) = \{\delta(n) - \delta(n-2) + \delta(n-4)$

ii) Establish the relationship between Fourier transform and Z-transform. (6)

14. a) Determine H(z) for a Butterworth filter satisfying the following constraints using impulse invariant transformation and T = 1 sec. (13)

Pass band edge magnitude = $\sqrt{0.5}$

Pass band frequency = $\frac{\pi}{2}$

Stop band magnitude = 0.2

Stop band frequency = $\frac{3\pi}{4}$.

(OR)

b) The desired frequency response of a Low pass filter is (13)

$$H_d\left(e^{j\omega}\right) = \begin{cases} e^{-j3\omega}, & \frac{-3\pi}{4} \le \omega \le \frac{3\pi}{4} \\ 0, & \frac{3\pi}{4} \le \omega \le \pi \end{cases}$$

Determine $H(e^{j\omega})$ for a rectangular window of width 7.

15. a) Discuss the architecture of any one commercial DSP and explain with necessary diagram. (13)

(OR)

b) Discuss in brief the addressing formats and functional modes of any one commercial DSP. (13)

PART – C (1×15=15 Marks)

16. a) i) Determine the response of the system defined by the equation

$$y(n) = \frac{5}{6}y(n-1) - \frac{1}{6}y(n-2) + x(n)$$

to the input signal $x(n) = \delta(n) - \frac{1}{3}\delta(n-1)$ assuming zero initial conditions. (10)

ii) Justify and explain the necessity for quantization and the Nyquist criterions influence on the quantization function. (5)

(OR)

b) Obtain the parallel and cascade realization of the following filter structure using Transpose form II. (7+8)

$$H(z) = \frac{\left(3+5z^{-1}\right)\left(0.6+3z^{-1}\right)}{\left(1-5z^{-1}+2z^{-2}\right)\left(1-z^{-1}\right)}.$$

Reg. No. :

Question Paper Code : 72054

02/06/2017 FN

B.E./B.Tech. DEGREE EXAMINATION, APRIL/MAY 2017.

Fifth/Sixth Semester

Information Technology

IT 6502 – DIGITAL SIGNAL PROCESSING

(Common to Computer Science and Engineering/Mechatronics Engineering)

(Regulations 2013)

Time : Three hours Maximum : 100 marks

Answer ALL questions.

PART A — (10 × 2 = 20 marks)

1. What is meant by aliasing? How can it be avoided?

2. Find the energy of $(1/4)^n u(n)$.

3. The first 5 DFT coefficients of a sequence x(n) are X(0) = 2, X(1) = 0.5 – j1.206, X(2) = 0, X(3) = 0.5-j0.206, X(4) = 0.Determine the remaining DFT coefficients.

4. Calculate % saving in computing through radix -2, DFT algorithm of DFT coefficients. Assume N = 512.

5. What does "frequency warping" mean? What is the effect on magnitude and phase response?

6. Given the Transfer function of LPF, $H(s)=\frac{1}{s+1}$, find the Transfer function of HPF having a cutoff frequency of 10 rad/sec.

7. State the advantages and disadvantages of FIR filter over IIR filter.

8. Define Gibbs phenomenon.

9. What is zero input limit cycle oscillation?

10. Define truncation error for sign magnitude representation and for 2's complement representation?

PART B — (5 × 16 = 80 marks)

11. (a) (i) Check whether the systems described by the following equations are (1) $y(n) = x(n)\cos\omega n$ (2) $y(n) = |x(n)|$

Static or Dynamic, Causal or non causal, Linear or nonlinear, Time variant or invariant, Stable or Unstable. (8)

(ii) Find the response of the system for the input signal, $x(n) = \{1,2,2,3\}$ and $h(n) = \{1,0,3,2\}$. (8)

Or

(b) Determine the inverse Z-transform of $X(z) = 1 / (1-1.5z^{-1}+0.5z^{-2})$ if

(i) ROC : $|Z|>1$ (ii) ROC : $|Z|<0.5$ (iii) ROC : $0.5 < |Z|<1$. (16)

12. (a) Explain the filtering methods based on DFT and FFT.

Or

(b) Determine the response of LTI system when input sequence $x(n) = \{-1, 1, 2, 1\}$ and impulse response $h(n) = \{-1, 1, -1, 1\}$ by radix-2 DIT FFT. (16)

13. (a) The specification of the desired low pass digital filter is

$0.8 \le |H(e^{j\omega})| \le 1.0;\ 0 \le \omega \le 0.2\pi$

$|H(e^{j\omega})| \le 0.2;\ 0.6\pi \le \omega \le \pi$. Design a Chebyshev digital filter using impulse Invariant Transformation. (16)

Or

(b) (i) Determine the system function of the IIR digital filter for the analog transfer function $H(s) = \dfrac{10}{s^2+7s+10}$ with T = 0.2 sec using impulse invariance method. (8)

(ii) Obtain the direct form-I and direct form-II realization for the system

$y(n) = -0.1\,y(n-1) + 0.2\,y(n-2) + 3x(n) + 3.6x(n-1) + 0.6x(n-2)$ (8)

14. (a) Design an FIR filter for the ideal frequency response using Hamming window with N=7.

$H_d(\omega) = e^{-j3\omega}$ for $-\dfrac{\pi}{8} \le \omega \le \dfrac{\pi}{8}$,

0 for $\dfrac{\pi}{8} \le |\omega| \le \pi$. (16)

Or

(b) Determine the filter coefficient $h(n)$ of length $M=15$ obtained by sampling and its frequency response as

$H\left(\dfrac{2\pi k}{15}\right) = 1$; K=0, 1, 2, 3, 4

$= 0.4$; K = 5

$= 0$; K = 6, 7. (16)

15. (a) (i) Explain the characteristics of a limit cycle oscillation w.r.to the system described by the equation $y(n) = 0.95y(n-1)+x(n)$. Determine the dead band of the filter. (12)

(ii) Bring out the differences between fixed-point and floating-point arithmetic. (4)

Or

(b) (i) Explain in detail about finite word length effects in Digital filter. (8)

(ii) Determine the variance of the round of noise power at the output of cascade realization of the filter is as described by the transfer function H(z) = $H_1(z)$ $H_2(z)$. Where $H_1(z) = \dfrac{1}{1-0.5z^{-1}}$ and $H_2(z) = \dfrac{1}{1-0.25z^{-1}}$. (8)

Reg. No. :

Question Paper Code : 41296

30/04/18

AN

B.E./B.Tech. DEGREE EXAMINATION, APRIL/MAY 2018
Sixth Semester
Information Technology
IT 6502 – DIGITAL SIGNAL PROCESSING
(Common to Computer Science and Engineering/Mechatronics Engineering)
(Regulations 2013)

Time : Three Hours Maximum : 100 Marks

Answer ALL questions

PART – A (10×2=20 Marks)

1. Define ROC.
2. Find the convolution of x(n) = {1, 2, 3, 1, 2, 1, 1} and h(n) = {1, 2, 1}.
3. Is DFT of a finite duration sequence is periodic ? If so, state the theorem.
4. Why FFT is needed ?
5. What is warping effect ?
6. Mention the methods for converting analog into digital IIR filter.
7. Compare Hanning and Hamming window.
8. What is Linear phase FIR filter ?
9. Mention the types of quantization errors.
10. What is zero input limit cycle oscillations ?

PART – B (5×13=65 Marks)

11. a) i) Find the Z transform and ROC of
a) $x(n) = \delta(n)$
b) $x(n) = [3(3)^n - 4(2)^n]\, u(n)$ (6)

ii) Check whether the system $y(n) = nx^2(n)$ is static or dynamic, linear or non-linear, time variant or invariant, causal or non-causal. (7)

(OR)

b) Determine the response of the system described by the difference equation $y(n) = 0.7\,y(n-1) - 0.12\,y(n-2) + x(n-1) + x(n-2)$ to the input $x(n) = nu(n)$. (13)

12. a) Starting from the key equation of DFT, with necessary equations explain DIT – FFT algorithm. (13)

(OR)

b) Find the 8 point DFT of $x(n) = \{0, 1, 2, 3, 4, 5, 6, 7\}$ using DIF-FFT algorithm. (13)

13. a) Convert the analog filter with transfer function $H(s) = \frac{2}{(s+1)(s+2)}$ into digital filter using impulse invariant method. (13)

(OR)

b) Design a digital filter which exhibits equiripple behaviour only either in pass band or stop band and monotonic characteristics either in pass band or stop band and satisfying the constraints.

$0.8 \leq |H(e^{j\omega})| \leq 1$ for $0 \leq \omega \leq 0.2\pi$

$|H(e^{j\omega})| \leq 0.2$ for $0.6\pi \leq \omega \leq \pi$

using Bilinear transformation. (13)

14. a) Explain the procedure of designing FIR filters by windows. (13)

(OR)

b) Explain frequency sampling method of designing FIR filters. (13)

15. a) Explain the various quantization errors in detail. (13)

(OR)

b) Explain limit cycle oscillations in detail. (13)

PART – C (1×15=15 Marks)

16. a) Find the 8 point DFT of $x(n) = \{1, 2, 3, 4, 4, 3, 2, 1\}$ using DIT-FFT algorithm. (15)

(OR)

b) Explain the characteristics of limit cycle oscillation represented to the system described by $y(n) = 0.95\,y(n-1) + x(n)$. Determine the dead band of the filter. (15)

PART C — (1 × 15 = 15 marks)

16. (a) Compute the characteristics of a limit cycle oscillation with respect to the system described the difference equation $x(n) = 0.95\, y(n-1) + x(n)$. Determine the dead band of the filter. Assume 4 bit sign magnitude representation including sign bit and the input as $x(n) = \begin{cases} 0.875, & \text{for } n = 0 \\ 0, & \text{otherwise} \end{cases}$. (15)

Or

(b) (i) Perform Circular convolution of the two sequences: (7)

$x_1(n) = \{2, 1, 2, 1\}$ $x_2(n) = \{1, 2, 3, 4\}$

(ii) Find the 4 point DFT of the sequence $x(n) = \cos\left(\frac{\pi}{4} n\right)$ using Decimation in Frequency algorithm. (8)

Reg. No. :

Question Paper Code : 53236

16-4-19

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B.E./B.Tech. DEGREE EXAMINATIONS, APRIL/MAY 2019.

Fifth/Sixth Semester

Information Technology

IT 6502 — DIGITAL SIGNAL PROCESSING

(Common to Computer Science and Engineering/Mechatronics Engineering)

(Regulation 2013)

Time : Three hours Maximum : 100 marks

Answer ALL questions.

PART A — (10 × 2 = 20 marks)

1. For the discrete time signal $x(n)$ shown in the Fig. 1 below, sketch the signal $x(n-3)$ and $x(n+2)$.

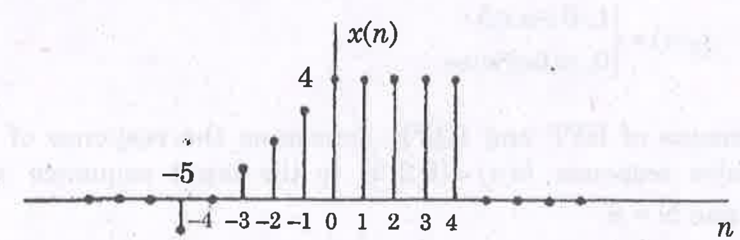

Fig. 1

2. Define correlation of two different signals.

3. Mention the number of computations involved in direct computation of DFT.

4. State the circular frequency shift property of DFT.

5. Mention the characteristics of the Butterworth and Chebychev analog filters.

6. Mention two advantages and disadvantages of IIR filters.

7. Define the Hamming and Hanning window functions.

8. Sketch the direct form structure for the FIR filter with the difference equation:

$$y(n) = x(n) + \frac{1}{2}x(n-1) + \frac{1}{4}x(n-2) + \frac{1}{8}x(n-3).$$

9. Mention the three ways of representing negative numbers. Express –7/8 in the three forms.

10. What is the advantage of scaling compared to saturation arithmetic?

PART B — (5 × 13 = 65 marks)

11. (a) (i) Consider the periodic sampling of a continuous time signal, establish the relation between analog and digital signal frequencies. (7)

(ii) Consider the analog signal $x_a(t) = 3\cos 100\pi t$.

(1) Determine the minimum sampling rate required to avoid aliasing. (2)

(2) Suppose that the signal is sampled at the rate $F_s = 300$ Hz and 75 Hz. What is the discrete time signal obtained after sampling? (4)

Or

(b) (i) Determine the power and energy of the unit step signal. (3)

(ii) Determine the Z-transform of the signal $x(n) = -a^n u(-n-1)$. Sketch its ROC. (5)

(iii) Compute the convolution of the two signals $x_1(n) = \{1, -2, 1\}$ and $x_2(n) = \begin{cases} 1, & 0 \le n \le 5 \\ 0, & \text{otherwise} \end{cases}$ (5)

12. (a) By means of DFT and IDFT, determine the response of the filter with impulse response $h(n) = \{1, 2, 3\}$ to the input sequence $x(n) = \{1, 2, 2, 1\}$. Assume N = 8. (13)

Or

(b) (i) Sketch the flow graphs of the basic butterfly computation and the 8 point Decimation in time FFT. (6)

(ii) Using the flow graph, determine the 8 point DFT of the sequence $x(n) = \{1, 2, 2, 2, 1, 0, 0, 0\}$. (7)

13. (a) A digital IIR low pass filter is required to meet the following frequency domain specifications :

3 dB ripple (maximum) in the passband $0 \le \omega \le 0.3\pi$ rad.

At least 20 dB (minimum) attenuation in the stopband $0.6\pi \le \omega \le \pi$

The digital filter is to be designed by applying bilinear transformation. (13)

Or

(b) A digital low pass filter is to be designed to have a maximally flat frequency response with the following specifications.

$$20\log|H(\omega)|_{\omega=0.2\pi} \ge -1.9328\ dB$$
$$20\log|H(\omega)|_{\omega=0.6\pi} \le -13.9794\ dB$$

Find the transfer function of the filter to meet the above specifications using impulse invariant transformation method. (13)

14. (a) The desired frequency response of a low pass filter is given by

$$H_d(\omega) = \begin{cases} e^{-j3\omega}, & |\omega| < \dfrac{3\pi}{4} \\ 0, & \dfrac{3\pi}{4} < |\omega| < \pi \end{cases}$$

. Determine the frequency response of the FIR filter if Hamming window is used with N = 7. (13)

Or

(b) Design a 17 tap linear phase FIR low pass filter with cut off frequency $\omega_c = \dfrac{\pi}{2}$. The design is to be done using frequency sampling technique. (13)

15. (a) Consider the recursive filter shown in the Fig. 2 below. The input $x(n)$ has a range of values ± 100 V. represented by 8 bits. Compute the variance of the output of the A/D conversion process. (13)

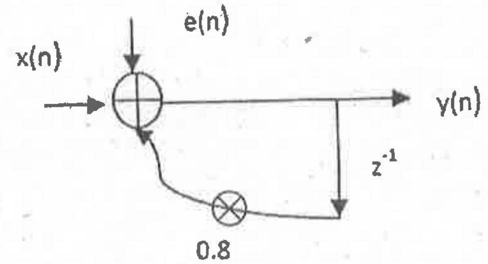

Fig. 2

Or

(b) Find the effect of coefficient quantization on pole locations of the given second order IIR system, when it is realized in direct form I and in cascade form. Assume a word length of 4 bits through truncation. $H(z) = \dfrac{1}{1 - 0.9z^{-1} + 0.2z^{-2}}$. (13)

23/11/16

Reg. No. :

Question Paper Code : 80596

B.E./B.Tech. DEGREE EXAMINATION, NOVEMBER/DECEMBER 2016.

Fifth Semester

Information Technology

IT 6502 — DIGITAL SIGNAL PROCESSING

(Common to Sixth Semester Computer Science and Engineering and Mechatronics Engineering)

(Regulations 2013)

Time : Three hours — Maximum : 100 marks

Answer ALL questions.

PART A — (10 × 2 = 20 marks)

1. What do you mean by Signal and Signal Processing?

2. What do you mean by convolution?

3. Write N-point DFT for, $x(n)$ and IDFT of $X(k)$.

4. What is meant by radix-2 FFT?

5. Distinguish analog and digital filters.

6. What is meant by impulse invariant method?

7. What are advantages of FIR filter over IIR filter?

8. What condition on the FIR sequence $h(n)$ are to be imposed n order that this filter can be called a linear phase filter? Write the necessary and sufficient condition for the FIR filter to have linear phase.

9. Compare fixed point and floating point representations.

10. Define dead band.

PART B — (5 × 16 = 80 marks)

(a) (i) Determine the power and energy of the signal $x(n) = \sin\left(\frac{\pi}{4}\right)n$. (8)

(ii) Determine whether the system described by the input output relation is time invariant or not

(1) $y(n) = x(n-1)$

(2) $y(n) = x(-n)$. (8)

Or

(b) (i) Determine the z transform and ROC of the signal $x(n) = (1/3)^n u(n)$. (8)

(ii) Find the cross correlation of $x(n) = \{1,2,1,1\}$ and $y(n) = \{1,1,2,1\}$. (8)

(a) Find the 8 point DFT of the sequence $x(n) = \{1, 1, 1, 1, 1, 1, 0, 0\}$. (16)

Or

(b) Compute the DFT for the sequence {2, 2, 2, 2, 1, 1, 1, 1}. Using radix -2 DIT - FFT algorithm. (16)

(a) Design a Butterworth low pass filter satisfying the following constraints.

Use Bilinear transformation

$$\sqrt{0.5} \le \left|H(e^{iw})\right| \le 1, \quad 0 \le w \le \frac{\pi}{2}$$

$$\left|H(e^{jw})\right| \le 0.2, \quad \frac{3\pi}{4} \le w \le \pi$$

Or

(b) Design an analog Chebyshev filter for the following specifications. Passband gain 0.89. Stop band attenuation 0.2, passband edge frequency 30Hz and stop band edge frequency 75Hz.

(a) Design a HPF with cut off frequency 1.2 radians of length N = 9 using Hamming window. (16)

Or

(b) Using frequency sampling method design a lowpass filter with the following specifications cut off frequency, $\omega_c = \pi/4$ and $N = 15$ and plot the magnitude response. (16)

(a) Derive the steady state output noise power and Find the steady state variance of the noise in the output due to quantization of input for the first order filter $y(n) = ay(n-1) + x(n)$. (16)

Or

(b) State the need for Scaling and derive the scaling factor for a second order IIR filter.. (16)

10/11/19 - FN

Reg. No. :

Question Paper Code : 50765

B.E./B.Tech. DEGREE EXAMINATION, NOVEMBER/DECEMBER 2017
Fifth/Sixth Semester
Information Technology
IT 6502 – DIGITAL SIGNAL PROCESSING
(Common to Computer Science and Engineering/Mechatronics Engineering)
(Regulations 2013)

Time : Three Hours Maximum : 100 Marks

Answer ALL questions.

PART – A (10×2=20 Marks)

1. Find the equivalent digital frequency 'w' given the analog frequency $\Omega = 20\pi$ rad/sec and sampling frequency $F_s = 30$ Hz.

2. Given $X(Z) = Z^2 + 2Z + 1 - 2Z^{-2}$. Find the equivalent time domain signal x(n).

3. Given x(n) = {1, 2, 3, 4} and h(n) = {2, 1, 3}. Circularly convolve x(n) and h(n).

4. State the need for using FFT algorithms for computing Discrete Fourier Transform (DFT).

5. Find the equivalent digital filter H(z) given the analog filter $H(S) = \frac{A}{S+a}$ using impulse invariant transformation.

6. What is the transfer function H(S) for a normalised Butterworth filter of order 3 ?

7. What are the various windows used for designing FIR filters ?

8. Given the impulse response of an FIR filter, h(n) = {1, 2, 3, 1, 3, 2, 1}. Is it a linear phase FIR filter ? Justify your answer.

9. Distinguish between truncation and rounding of binary digits using relevant examples.

10. What is meant by signal scaling ?

PART – B (5×13=65 Marks)

11. a) A discrete time system is represented by the following difference equation $y(n) = 3y^2(n-1) - nx(n) + 4x(n-1) - 2x(n+1)$ for $n \geq 0$. Determine whether the system is memoryless, causal, linear and shift invariant. Justify your answers.

(OR)

b) A causal system is represented by the following difference equation

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

i) Find the system function H(Z) and its Corresponding Region of Convergence (ROC).

ii) Find the unit sample response h(n) of the system.

12. a) Given x(n) = {1, 2, –1, 2, 2, –1, 2, 1}. Compute 8-point DFT using Decimation-in-time algorithm.

(OR)

b) Given x(n) = {1, 2, 3, 4} = h(n). Circularly convolve x(n) and h(n) using DFT and IDFT computations.

13. a) Explain with necessary equations the approximation of derivatives method for converting an analog filter into a digital filter.

(OR)

b) Using bilinear transformation design a low pass filter monotonic in passband with –3.01 dB cutoff frequency of 0.4π rad and magnitude down atleast by 15 dB at 0.75π rad.

14. a) Design an FIR filter with N = 7 using Hanning window, given

$$H_d(e^{j\omega}) = \begin{cases} e^{-j3\omega}, & -\frac{\pi}{4} \leq \omega \leq \frac{\pi}{4} \\ 0, & \frac{\pi}{4} < |\omega| \leq \pi \end{cases}$$

(OR)

b) Given $H(Z) = 0.5 + 0.25Z^{-1} + 0.75Z^{-2} + Z^{-3} + 0.75Z^{-4} + 0.25Z^{-5} + 0.5Z^{-6}$. Draw the linear phase realization and direct form realization and compare both the structures.

15. a) Explain in detail the input quantization error and coefficient quantization error and its effect on digital filter design, with an example.

(OR)

b) Explain the characteristics of limit cycle oscillation with respect to the system described by the difference equation $y(n) = 0.95\, y(n-1) + x(n)$. Determine the deadband of the system when $x(n) = 0.875$ for $n = 0$ and $y(-1) = 0$. Assume 4-bit sign magnitude representation (excluding sign bit).

PART – C (1×15=15 Marks)

16. a) A system with input x(n) and output y(n) is characterized by the following equation $y(n) = x(n+1) + x(n-1)$.

i) Find the impulse response of the system.

ii) Is the system causal ? Justify.

iii) Is this an FIR or IIR filter ?

iv) Find the frequency response of the system.

v) Is this system BIBO stable ? Justify.

(OR)

b) The impulse response of a discrete time system LTI system is given by

$$h(n) = \left(\frac{1}{2}\right)^n u(n) + \left(-\frac{1}{3}\right)^n u(n)$$

i) Is this a causal system ? Justify.

ii) Is this an FIR of IIR filter ?

iii) Find the frequency response $H(e^{j\omega})$ of the system ?

iv) Give a difference equation realization of the system.

9/11/19 AN Reg. No. :

Question Paper Code : 91771

B.E./B.Tech. DEGREE EXAMINATIONS, NOVEMBER/DECEMBER 2019
Fifth/Sixth Semester
Information Technology
IT 6502 – DIGITAL SIGNAL PROCESSING
(Common to Sixth Semester Computer Science and Engineering and Mechatronics Engineering)
(Regulations 2013)

Time : Three Hours Maximum : 100 Marks

Answer ALL questions

PART – A **(10×2=20 Marks)**

1. State sampling theorem.
2. What is meant by correlation and mention its need in signal processing ?
3. The first 5 DFT coefficients of a 8 sample sequence, x(n) are X(0) = 2, X(1) = 0.5 – j1.206, X(2) = 0, X(3) = 0.5 – j0.206, X(4) = 0. Determine the remaining DFT coefficients.
4. Define DCT pair.
5. Discuss the need for prewarping.
6. What are the properties of Butterworth filter ?
7. What is the necessary and sufficient condition for the impulse response, h(n) of an FIR filter to be linear phase ?
8. Mention the desirable characteristics for choosing a window function for designing FIR filters.
9. Perform the addition of the decimal numbers (0.5 and 0.25) using binary fixed point representation.
10. Define deadband. How to compute the deadband of a recursive system ?

PART – B **(5×13=65 Marks)**

11. a) i) Determine the power and energy of the signal $x(n) = \left(\frac{1}{3}\right)^n u(n)$. **(5)**

ii) Determine whether the system described by the input-output relation y(n) = nx(n) linear, shift invariant, dynamic and causal. **(8)**

(OR)

b) i) Determine the z transform and plot its ROC for the signal, $x(n) = (a)^n u(n)$; $0 \le a \le 1$. **(5)**

ii) Perform the convolution given $h(n) = b^n u(n); 0 \le b \le 1$ and $x(n) = a^n u(n); 0 \le a \le 1$. **(8)**

12. a) Use DFT-IDFT method to compute the response of the system with impulse response $h(n) = \{1, 2, 1\}$ for the input $x(n) = \{3, 2\}$. **(13)**

(OR)

b) Compute the DFT of given sequence using DIF-FFT algorithm.

$x(n) = \{1, 2, 3, 4, 4, 3, 2, 1\}$. **(13)**

13. a) The specification of the desired low pass digital filter is $0.8 \le |H(e^{j\omega})| \le 1.0; 0 \le \omega \le 0.2\pi$. $|H(e^{j\omega})| \le 0.2; 0.6\pi \le \omega \le \pi$. Design a Butterworth digital filter using Bilinear Transformation technique. **(13)**

(OR)

b) i) Determine the system function of the IIR digital filter for the analog transfer function $H(s) = \dfrac{10}{s^2 + 7s + 10}$ with T = 0.2 sec using impulse invariance method. **(8)**

ii) Obtain the direct form-II realization for the system $y(n) = -0.1\, y(n-1) + 0.2\, y(n-2) + 3x(n) + 3.6x(n-1) + 0.6x(n-2)$. **(5)**

14. a) Design a non-recursive HPF with cutoff frequency 1.2 radians of length N = 9 using Hamming window. **(13)**

(OR)

b) Using frequency sampling method, design a lowpass filter with the following specifications: cut off frequency, $\omega_c = \pi/4$ radians and order, N = 15 and plot the magnitude response of the designed filter. **(13)**

15. a) Find the steady state variance of the noise in the output, due to quantization of input for the first order filter $y(n) = ay(n-1) + x(n)$; $0 < a < 1$. **(13)**

(OR)

b) With relevant example, briefly discuss about the effect of coefficient quantization on the location of poles and zeros in the z-plane. **(13)**

PART – C **(1×15=15 Marks)**

16. a) Using Decimation in Time FFT algorithm, compute the circular convolution between the two given sequences, $x_1(n) = \{1, 2, 2, 1\}$ and $x_2(n) = \{4, 5\}$. **(15)**

(OR)

b) Use hanning window to design a 9 tap non-recursive LPF with cutoff frequency, $f_c = 2$ kHz. The filter should operate at the rate of 8000 samples/sec. Draw the linear phase realization structure for the designed filter. **(15)**
