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Question Bank

EE6403: DISCRETE TIME SYSTEMS AND SIGNAL PROCESSING

(Regulation 2013)

UNIT-I – INTRODUCTION

PART A

1. Define Signal.

A Signal is defined as any physical quantity that varies with time, space or any other independent variables.

2. Define a system.

A System is a physical device (i.e., hardware) or algorithm (i.e., software) that performs an operation on the signal.

3. What are the steps involved in digital signal processing?

- Converting the analog signal to digital signal, this is performed by A/D converter
- Processing Digital signal by digital system.
- Converting the digital signal to analog signal, this is performed by D/A converter.

4. Give some applications of DSP?

- Speech processing – Speech compression & decompression for voice storage system
- Communication – Elimination of noise by filtering and echo cancellation.
- Bio-Medical – Spectrum analysis of ECG, EEG etc.

5. Write the classifications of DT Signals.

- Energy & Power signals
- Periodic & A periodic signals
- Even & Odd signals.

6. What is an Energy and Power signal?

Energy signal:

A finite energy signal is periodic sequence, which has a finite energy but zero average power.

Power signal:

An Infinite energy signal with finite average power is called a power signal.

7. What is Discrete Time Systems?

The function of discrete time systems is to process a given input sequence to generate output sequence. In practical discrete time systems, all signals are digital signals, and operations on such signals also lead to digital signals. Such discrete time systems are called digital filter.

8. Write the Various classifications of Discrete-Time systems.

- Linear & Non linear system
- Causal & Non Causal system
- Stable & Un stable system
- Static & Dynamic systems

9. Define linear system

A system is said to be linear system if it satisfies Super position principle. Let us consider $x_1(n)$ & $x_2(n)$ be the two input sequences & $y_1(n)$ & $y_2(n)$ are the responses respectively

$[ax_1(n) + bx_2(n)] = a\, y_1(n) + by_2(n)$

10. Define Static & Dynamic systems

When the output of the system depends only upon the present input sample, then it is called static system, otherwise if the system depends past values of input then it is called dynamic system

11. Define causal system.

When the output of the system depends only upon the present and past input sample, then it is called causal system, otherwise if the system depends on future values of input then it is called non-causal system

12. Define Shift-Invariant system.

If y(n) is the response to an input x(n), then the response to an input

$X(n) = x(n-n_0)$ then $y(n) = y(n-n_0)$.

When the system satisfies above condition then it is said to shift invariant, otherwise. it is variant.

13. Define impulse and unit step signal.

Impulse signal δ(n):

The impulse signal is defined as a signal having unit magnitude at n = 0 and zero for other values of n.

$$\boldsymbol{\delta(n)} = \begin{cases} 1; n = 0 \\ 0; n \neq 0 \end{cases}$$

Unit step signal u(n):

The unit step signal is defined as a signal having unit magnitude for all values of $n \geq 0$

$$\mathbf{u(n)} = \begin{cases} \mathbf{1;\ n \geq 0} \\ \mathbf{0;\ n < 0} \end{cases}$$

14. What are FIR and IIR systems?

The impulse response of a system consist of infinite number of samples are called IIR system & the impulse response of a system consist of finite number of samples are called FIR system.

15. What are the basic elements used to construct the block diagram of discrete time system?

The basic elements used to construct the block diagram of discrete time Systems are Adder, Constant multiplier &Unit delay element.

16. What is ROC in Z-Transform?

The values of z for which z – transform converges is called region of convergence (ROC). The z-transform has an infinite power series; hence it is necessary to mention the ROC along with z-transform.

18. What are the different methods of evaluating inverse z- transform?

- Partial fraction expansion
- Power series expansion
- Contour integration (Residue method)

19. Define sampling theorem.

A continuous time signal can be re resented in its samples and recovered back if the sampling frequency $F_s \geq 2B$. Here 'F_s' is the sampling frequency and 'B' is the maximum frequency present in the signal.

20. What are the properties of convolution?

1. Commutative property $x(n) * h(n) = h(n) * x(n)$
2. Associative property $[x(n) * h_1(n)]*h_2(n) = x(n)*[h_1(n) * h_2(n)]$
3. Distributive property $x(n) *[h_1(n)+h_2(n)] = [x(n)*h_1(n)]+[x(n)* h_2(n)]$

PART-B

1. What is meant by energy and power signal? Determine whether the following signals are energy or power or neither energy nor power signals.

   (i) $x_1(n)=(1/2)^n u(n)$

   (ii) $x_2(n)=\sin(\pi n/6)$

   (iii) $x_3(n)=e^{j(\pi n/3+\pi/6)}$

   (iv) $x_4(n)=e^{2n}u(n)$

2. Explain the concept of quantization.

3.Check for the following systems are linear,time invariant,causal,stable,static

   (i) $y(n)=x(1/2n)$

   (ii) $y(n)=\sin(x(n))$

4.Check whether the following are periodic

   (i) $x(n)=\cos(3\pi n)$

   (ii) $x(n)=\sin(3n)$

5. Check whether the following are energy or power signals

   (i) $x(n)=[1/2]^n u(n)$

   (ii) $x(n)=Ae^{jwon}$

6.What do you mean by Nyquist rate?Give its significance.

7.Explain the classification of discrete signal.

8.Starting from first principles,state and explain sampling theorem both in time domain and in frequency domain.

9.A.discrete time systems can be

(i) Static or dynamic

(ii) Linear or non linear

(iii) Time invariant or time varying

(iv) Stable or unstable

Examine the following systems with respect to the properties above

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

(ii) $y(n)=x(n)+\cos(x(n))$

10.Check the causality and stability of the systems $y(n)=x(-n)+x(n-2)+x(2n-1)$

UNIT-II

DISCRETE TIME SYSTEM ANALYSIS

1. Define DTFT.

Let us consider the discrete time signal x(n).Its DTFT is denoted as X(ω).It is given as $X(\omega)= x(n)e^{-j\omega n}$

2. State the condition for existence of DTFT?

The conditions are

- If x(n)is absolutely summable
- If x(n) is not absolutely summable then it should have finite energy

3. List the properties of DTFT.

- Periodicity
- Linearity
- Time shift
- Frequency shift
- Scaling
- Differentiation in frequency domain
- Time reversal
- Convolution
- Multiplication in time domain
- Parseval’s theorem

4. What is the DTFT of unit sample?

The DTFT of unit sample is 1 for all values of w.

5. Define DFT.

DFT is defined as $X(w) = x(n)e^{-jwn.}$.

Here x(n) is the discrete updates time sequence

X(w) is the Fourier transform of x(n).

6. Define Twiddle factor.

The Twiddle factor is defined as $W_N = e^{-j2\ /N}$

7. Define Zero padding.

The method of appending zero in the given sequence is called as Zero padding.

8. Define circularly even sequence.

A Sequence is said to be circularly even if it is symmetric about the point zero on the circle. x(N-n)=x(n),1<=n<=N-1.

9. Define circularly odd sequence.

A Sequence is said to be circularly odd if it is anti symmetric about point x(0) on the circle

10. Define circularly folded sequences.

A circularly folded sequence is represented as x((-n))N. It is obtained by plotting x(n) in clockwise direction along the circle.

11. State circular convolution.

This property states that multiplication of two DFT is equal to circular convolution of their sequence in time domain.

12. State parseval's theorem.

Consider the complex valued sequences x(n) and y(n).If $x(n)y^*(n)=1/N\ X(k)Y^*(k)$

13. Define Z transform.

The Z transform of a discrete time signal x(n) is denoted by X(z) and is given by $X(z)= x(n)Z^{-n}$.

14. Define ROC.

The value of Z for which the Z transform converged is called region of convergence.

15. Find Z transform of x(n)={1,2,3,4}

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

$X(z)= x(n)z^{-n}$

$= 1+2z^{-1} +3z^{-2} +4z^{-3}$.

$= 1+2/z+3/z^{2} +4/z^{3}$

16.Define time shifting property

The convolution property states that the convolution of two sequences in time domain is equivalent to multiplication of their Z transforms.

17. What z transform of (n-m)?

$Z[A\ (n-m)]=AZ^{-m}$

sin Z[(n)] =1

18. State initial value theorem.

If x(n) is causal sequence then its initial value is given by x(0)=lim X(z)

19. List the methods of obtaining inverse Z transform.

Inverse z transform can be obtained by using

- Partial fraction expansion.
- Contour integration
- Power series expansion

PART B

1.Find the Z Transform and its ROC of

$x(n)=[-1/5]^{n}u(n)+5[1/2]^{-n}u(-n-1)$

2.Find the linear convolution of x(n)={2,4,6,8,10} with h(n)={1,3,5,7,9}.

3.Determine the Z transform of the following

(i) $x(n)=a^{n}\cos w_{o}n\,u(n)$

(ii) $x(n)=3^{n}u(n)$.

4.Explain the properties of Z transform.

5.Find the impulse response given by difference equation

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

6. Determine the Z transform of the signal $\cos w_{o}n\,u(n)$

7.Find the convolution

x(n)={-1,1,-2,2}, h(n)={0.5,1,-1,2,0.75}

8.Find inverse Z transform of $X(Z)=Z/[3Z^{2}-4Z+1]$,ROC |Z|>1

9.Determine the DTFT of the sequence x(n)=an(u(n)-u(n-8)),|a|<1

10.prove the linearity and frequency shifting theorems of DTFT.

UNIT-III

DISCRETE FOURIER TRANSFORM AND COMPUTATION

1. What is DFT?

It is a finite duration discrete frequency sequence, which is obtained by sampling one period of Fourier transform. Sampling is done at N equally spaced points over the period extending from w=0 to 2л.

2. Define N point DFT.

The DFT of discrete sequence x(n) is denoted by X(K). It is given by,

Here k=0,1,2…N-1Since this summation is taken for N points, it is called s N-point DFT.

3. What is DFT of unit impulse δ(n)?

The DFT of unit impulse δ(n) is unity.

4. List the properties of DFT.

Linearity, Periodicity, Circular symmetry, symmetry, Time shift, Frequency shift.

5.State Linearity property of DFT.

DFT of linear combination of two or more signals is equal to the sum of linear combination of DFT of individual signal.

6. When a sequence is called circularly even?

The N point discrete time sequence is circularly even if it is symmetric about the p on the circle.

7. What is the condition of a sequence to be circularly odd?

An N point sequence is called circularly odd it if is antisymmetric about point zero on the circle.

8. Why the result of circular and linear convolution is not same?

Circular convolution contains same number of samples as that of x (n) and h (n), while in linear convolution, number of samples in the result (N) are,

$$N=L+M-1$$

Where L=Number of samples in x (n) M=Number of samples in h (n)

9. What is circular time shift of sequence?

Shifting the sequence in time domain by '1' samples is equivalent to multiplying the sequence in frequency domain by W_N^{kl}

10. What is the disadvantage of direct computation of DFT?

For the computation of N-point DFT, N2 complex multiplications and N[N-1] Complex additions are required. If the value of N is large than the number of into lakhs. This proves inefficiency of direct DFT computation.

11. What is the way to reduce number of arithmetic operations during D computation?

Number of arithmetic operations involved in the computation of DFT is greatly reduced using different FFT algorithms as follows.

1. Radix-2 FFT algorithms.

   -Radix-2 Decimation in Time (DIT) algorithm.

   - Radix-2 Decimation in Frequency (DIF) algorithm.

2. Radix-4 FFT algorithm.

12. What is the computational complexity using FFT algorithm?

1. Complex multiplications = N/2 log 2 N
2. Complex additions = N log 2 N

13. How linear filtering is done sing FFT?

Thus, by folding the sequence h (n), e can compute the linear filtering using FFT. Correlation is the basic process of doing linear filtering using FFT. The correlation is the sequence x(n) has a length L. If we want to find the N point DFT (N>L) of the sequence x (n).

14. What is zero padding? What are its uses?

This is known as zero padding. The uses of padding a sequence with zeros are

(i) We can get 'better display' of the frequency spectrum.

(ii) With zero padding, the DFT can be used in linear filtering.

15. Why FFT is needed?

The direct evaluation of the DFT using the formula requires N^2 complex multiplications and N (N-1) complex additions. Thus for reasonably large values of N (in order of 1000) direct evaluation of the DFT requires an inordinate amount of computation. By using FFT algorithms the number of computations can be reduced. For example, for an N-point DFT,The number of complex multiplications required using FFT is $N/2\log_2 N$. If N=16, the number of complex multiplications required for direct evaluation of DFT is 256, whereas using DFT only 32 multiplications are required.

16. What is the speed of improvement factor in calculating 64-point DFT of a sequence using direct computation and computation and FFT algorithms?

Or Calculate the number of multiplications needed in the calculation of DFT and FFT with 64-point sequence.

The number of complex multiplications required using direct computation is

N^2 =642=4096.

The number of complex multiplications required using FFT

is $N/2 \log_2 N$ = 64/2log 2 64=192.

Speed improvement factor = 4096/192=21.33

17. What is the main advantage of FFT?

FFT reduces the computation time required to compute discrete Fourier transform.

18. How many multiplications and additions are required to compute N-point DFT using redix-2 FFT?

The number of multiplications and additions required to compute N-point DFT using redix-2 FFT are $N \log_2 N$ and N/2 log 2 N respectively.

19. What is meant by radix-2 FFT?

The FFT algorithm is most efficient in calculating N-point DFT. If the number of output points N can be expressed as a power of 2, that is, N=2 M , where M is an integer, Then this algorithm is known as radix-s FFT algorithm.

20. What are the applications of FFT algorithms?

1. Linear filtering
2. Correlation
3. Spectrum analysis

21. What is a decimation-in-frequency algorithm?

In this the output sequence X (K) is divided into two N/2 point sequences and each N/2 point sequences are in turn divided into two N/4 point sequences.

Part-B

1.Compute the DFT of the sequence x(n) = {1,1,1,1,1,1,0,0}

2. From the first principles obtain the signal flow graph for computing 8 – point DFT using radix-2 DIT FFT algorithm. Using the above compute the DFT of sequence

$x(n)=\{0.5,0.5,0.5,0.5,0,0,0,0\}$

3.Explain any five properties of DFT.

4.Derive DIF – FFT algorithm. Draw its basic butterfly structure and compute the DFT $x(n) = (-1)^n$ using radix 2 algorithm.

5.Find the circular convolution of x(n) = 1,2,3,4} and h(n) ={4,3,2,1}

6.Determine the 8 point DFT of the signal $x(n)=\{1,1,1,1,1,1,0,0\}$.

Sketch its magnitude and phase.

UNIT IV

DESIGN OF DIGITAL FILTERS

1. Define IIR filter?

IIR filter has Infinite Impulse Response.

2. What are the various methods to design IIR filters?

- Approximation of derivatives
- Impulse invariance
- Bilinear transformation.

3. What is the main problem of bilinear transformation?

Frequency warping or nonlinear relationship is the main problem of bilinear transformation.

4. Where the jω axis of s-plane is mapped in z-plane in bilinear transformation?

The jω axis of s-plane is mapped on the unit circle in z-plane in bilinear transformation

5. Where left hand side and right hand side are mapped in z-plane in bilinear transformation?

Left hand side -- Inside unit circle

Right hand side – Outside unit circle

6. What is the frequency response of Butterworth filter?

Butterworth filter has monotonically reducing frequency response.

7. What is impulse invariant transformation?

The transformation of analog filter to digital filter without modifying the impulse response of the filter is called impulse invariant transformation.

8. Compare the Butterworth and Chebyshev Type-1 filters.

#	Butterworth	Chebyshev Type - 1
1	All pole design.	All pole design.
2	The poles lie on a circle in s-plane	The poles lie on a ellipse in s-plane
3	The magnitude response is maximally flat at the origin and monotonically decreasing function	The magnitude response is equiripple in pass band and monotonically decreasing in the stop band.
4	iv. The normalized magnitude response has a value of 1 / √2 at the cutoff frequency	The normalized magnitude response has a value of 1 / √(1+_ 2) at the cutoff frequency
5	v. Only few parameters has to be calculated to determine the transfer function.	A large number of parameters has to be calculated to determine the transfer function

9. Distinguish between FIR and IIR filters.

#	FIR filter	IIR filter
1	These filters can be easily designed to have perfectly linear phase.	These filters do not have linear phase.
2	FIR filters can be realized recursively and non-recursively	IIR filters can be realized recursively.
3	Greater flexibility to control the shape of their magnitude response.	Less flexibility, usually limited to kind of filters.
4	Errors due to round off noise are less severe in FIR filters, mainly because feedback is not used.	The round off noise in IIR filters are more.

10. What are the techniques of designing FIR filters?

There are three well-known methods for designing FIR filters with linear phase. These are 1) windows method 2) Frequency sampling method 3) Optimal or minimax design.

11. State the condition for a digital filter to be causal and stable.

A digital filter is causal if its impulse response $h(n) = 0$ for $n<0$

A digital filter is stable if its impulse response is absolutely summable

12. What is the reason that FIR filter is always stable?

FIR filter is always stable because all its poles are at origin.

13. Write the steps involved in FIR filter design.

- Choose the desired (ideal) frequency response $H_d(w)$.
- Take inverse Fourier transform of $H_d(w)$ to get $h_d(n)$.
- Convert the infinite duration $h_d(n)$ to finite duration $h(n)$.
- Take Z-transform of $h(n)$ to get the transfer function $H(z)$ of the FIR filter.

14. What are the advantages of FIR filters?

- .Linear phase FIR filter can be easily designed.
- Efficient realization of FIR filter exist as both recursive and non recursive structures.
- FIR filters realized nonrecursively are always stable.
- The round off noise can be made small in nonrecursive realization of FIR filters.

15. Compare the rectangular window and hanning window.

#	Rectangular window	Hanning Window
1	The width of main lobe in window spectrum is $4\pi/N$	The width of main lobe in window spectrum is $8\pi/N$
2	The maximum side lobe magnitude in window spectrum is –13dB.	The maximum side lobe magnitude in window spectrum is –31dB.

16. Compare the rectangular window and hamming window.

#	Rectangular window	Hamming Window
1	The width of main lobe in window spectrum is $4\pi/N$	The width of main lobe in window spectrum is $8\pi/N$
2	The maximum side lobe magnitude in window spectrum is –13dB.	The maximum side lobe magnitude in window spectrum is –41dB.

Part-B

1.With suitable examples, describe the realization of linear phase FIR filters.

2.Discuss about frequency transformations in detail.

3.Explain the quantization effects in design of digital filters

4.Draw the structure for IIR filter in direct form – I and II for the following
transfer Function $H(z) = (2 + 3z^{-1})(4 + 2z^{-1} + 3z^{-2}) / (1 + 0.6z^{-1})(1 + z^{-1} + 0.5z^{-2})$

5.Obtain the cascade and parallel realization of system described by difference equation
$y(n) = -0.1\,y(n-1) + 0.2\,y(n-2) + 3x(n) + 3.6\,x(n-1) + 0.6\,x(n-2)$

6. Design a high pass filter of length 7 samples with cut off frequency of 2 rad / sec
using Hamming window. Plot its magnitude and phase response.

UNIT V
DIGITAL SIGNAL PROCESSOR

1. Write short notes on general purpose DSP processors

General-purpose digital signal processors are basically high speed microprocessors with hard ware architecture and instruction set optimized for DSP operations. These processors make extensive use of parallelism, Harvard architecture, pipelining and dedicated hardware whenever possible to perform time consuming operations

2. Write notes on special purpose DSP processors.

There are two types of special; purpose hardware.

(i) Hardware designed for efficient execution of specific DSP algorithms such as digital filter, FFT.

(ii) Hardware designed for specific applications, for example telecommunication, digital audio.

3.What about of Harvard architecture?

The principal feature of Harvard architecture is that the program and the data memories lie in to separate spaces, permitting full overlap of instruction fetch and execution.

Typically these types of instructions would involve their distinct type.

1. Instruction fetch
2. Instruction decode
3. Instruction execute

4. What are the types of MAC is available?

There are two types MAC'S available

1. Dedicated & integrated
2. Separate multiplier and integrated unit

5. What is meant by pipeline technique?

The pipeline technique is used to allow overall instruction executions to overlap. That is where all four phases operate in parallel. By adapting this technique, executi n speed is increased.

6. What are four phases available in pipeline technique?

The four phases are

(i) Fetch

(ii) Decode

(iii)Read

(iv) Execution

7.Write down the name of the addressing modes.

- Direct addressing.
- Indirect addressing.
- Bit-reversed addressing.
- Immediate addressing.
- Short immediate addressing.

- Long immediate addressing.
- Circular addressing

8.What are the instructions used for block transfer in C5X Processors?

The BLDD, BLDP and BLPD instructions use the BMAR to point at the source or destination space of a block move. The MADD and MADS also use the BMAR to address an operand in program memory for a multiply accumulator operation

9. What is meant by auxiliary register file?

The auxiliary register file contains eight memory-mapped auxiliary registers (AR0-AR7), which can be used for indirect addressing of the data memory or for temporary data storage.

10. Write the name of various part of C5X hardware.

1. Central arithmetic logic unit (CALU)
2. Parallel logic unit (PLU)
3. Auxiliary register arithmetic unit (ARAU)
4. Memory-mapped registers.
5. Program controller.

Part-B

1.Describe the function of on chip peripherals of TMS 320 C54 processor.

2.What are the different buses of TMS 320 C54 processor? Give their functions.

3.Explain the function of auxiliary registers in the indirect addressing mode to point the power.

4. Give a detailed note on Direct memory Access controller in TMS 320 C54x processor.

5.Explain the statistical characterization of quantization effects in fixed point realization of digital filter.