

VALLIAMMAI ENGINEERING COLLEGE

DEPARTMENT OF ELECTRONICS AND INSTRUMENTATION
ENGINEERING

EE6403 – DISCRETE TIME SYSTEMS AND SIGNAL PROCESSING

Year / Semester: II / IV (2014 – 2015)

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UNIT – I

2 Marks Q & A

1. Consider the analog signal $x(t) = 3\cos 50\pi t + 10 \sin 300\pi t - \cos 100\pi t$.
What is the nyquist rate for this signal?

$$\text{Here } \omega_{\max} = 300\pi$$

$$\text{So, } 2\pi f_m = 300\pi$$

$$\text{Hence, Nyquist rate } F_s = 2f_m = 300$$

2. State Shannon's sampling theorem.

A band limited continuous time signal with highest frequency (band width) f_m hertz, can be uniquely recovered from its samples provided that the sampling rate f_s is greater than or equal to $2f_m$ samples per second

$$F_s \geq 2f_m$$

3. Given a continuous time signal $x(t) = 2\cos 500\pi t$. What is the Nyquist rate and fundamental frequency of the signal?

$$\omega = 500\pi$$

$$2\pi f = 500\pi$$

$$f = 250\text{Hz}$$

$$\text{Nyquist rate } F_s = 2f_m = 2 \times 250 = 500\text{Hz}$$

4. Determine whether $x[n] = u[n]$ is a power signal or an energy signal.

The energy of a discrete time signal $x(n)$ is defined as

$$E = \sum_{n=-\infty}^{\infty} |x(n)|^2 = \infty$$

The average power of a discrete time signal $x(n)$ is defined as

$$P = \lim_{N \rightarrow \infty} \frac{1}{2N+1} \sum_{n=-N}^N |x(n)|^2 = 0.5$$

Here $E = \infty$ and $P = \text{Finite}$. Therefore the given signal is a power signal.

5. What is the Nyquist rate for the signal $x_a(t) = 3\cos 600\pi t + 2\cos 1800\pi t$?

$$\begin{aligned} \text{Solution: } \omega_1 &= 600\pi & \omega_2 &= 1800\pi \\ 2\pi f_1 &= 600\pi & 2\pi f_2 &= 1800\pi \\ f_1 &= 300\text{Hz} & f_2 &= 900\text{Hz} \\ \text{Nyquist rate } F_s &= 2f_m = 2 \times 900 = 1800\text{Hz.} \end{aligned}$$

6. Determine fundamental period of the signal $\cos\left(\frac{\pi 30n}{105}\right)$

$$\begin{aligned} \text{Solution: Fundamental period, } N &= \left(\frac{2\pi}{\omega_o}\right)m, \quad \text{Where } \omega_o = \frac{30\pi}{105} = \frac{10\pi}{35}m, \\ \text{when } m &= 1 \text{ \& } N = 7 \text{ periods.} \end{aligned}$$

7. What is a linear time invariant system?

An LTI system is one which possesses both Linearity and Time-invariance.

A system is linear if $y_1(n) = T[x_1(n)]$ and $y_2(n) = T[x_2(n)]$ then
 $T[a_1 x_1(n) + a_2 x_2(n)] = a_1 y_1(n) + a_2 y_2(n)$

8. What is meant by aliasing effect?

The superimposition of high frequency component on the low frequency is known as “frequency aliasing” or “aliasing effect”.

9. Define Nyquist rate.

The frequency $2f_m$, which, under sampling theorem, must be exceeded by the sampling frequency is known as the Nyquist rate.

10. What do you meant by sampling process?

Sampling is the conversion of a continuous –time signal (or analog signal) into a discrete – time signal obtained by taking samples of the continuous time signal (or analog signal) at discrete time instants.

11.How can aliasing be avoided?

To avoid aliasing the sampling frequency must be greater than twice the highest frequency present in the signal.

12.What is an anti aliasing filter?

A filter is used to reject frequency signals before it is sampled to reduce the aliasing is called an anti aliasing filter.

13.What is meant by Quantization error?

It is the difference between the quantized value and actual sample value.

$$e_q(n) = x_q(n) - x(n)$$

14. What is a quantization level?

The value allows in a digital signal are called the quantization level.

15.Define resolution or quantization step size.

The distance between two successive level is called quantization step size or resolution

16.Define unit sample response (impulse response) of a system and what is its significance?

The response or output signal designated as $h(n)$, obtained from a discrete – time system when the input signal is a unit sample sequence (unit impulse) , is known as the unit sample response(impulse response).

The output $y(n)$ of an LTI system for an input signal $x(n)$ can be obtained by convolving the impulse response $h(n)$ and the input signal $x(n)$.

$$\begin{aligned} y(n) &= x(n) * h(n) \\ &= \sum_{k=-\infty}^{\infty} x(k)h(n - k) \end{aligned}$$

17. What is the causality condition for an LTI system?

The necessary and sufficient condition for causality of an LTI system is, its unit sample response $h(n)=0$ for negative values of n i.e., $h(n)=0$ for $n<0$

18. What is the necessary and sufficient condition on the impulse response for stability?

The necessary and sufficient condition guaranteeing the stability of a linear time-invariant system is that its impulse response is absolutely summable. $\sum_{k=-\infty}^{\infty} |h(k)| < \infty$.

19. Define BIBO stable system?

Any relaxed system is said to be bounded input-bounded output (BIBO) stable if and only if every bounded input yields a bounded output. Mathematically, there exist some finite numbers, M_x and M_y such that, $|x(n)| \leq M_x < \infty$ and $|y(n)| \leq M_y < \infty$

20. What are the classifications of discrete – time systems?

1. Static and Dynamic system.
2. Time – variant and time – invariant system.
3. Linear and non – linear system.
4. Stable and Un-stable system.
5. Causal and non-causal system.
6. IIR and FIR system.

16marks

1. Discuss whether the following are energy or power signals.

(i) $x(n) = \left(\frac{3}{2}\right)^n u(n)$

(ii) $x(n) = Ae^{j\omega n}$

2. Explain the concept of quantization.

3. Check whether following are linear, time invariant, causal and stable.

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

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

(iii) $y(n) = x(-n-5)$

4. What is causality and stability of a system? Derive the necessary and sufficient condition on the impulse response of the system for causality and stability.

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

$$x_1(n) = \left(\frac{1}{2}\right)^n u(n) \quad x_2(n) = \sin\left(\frac{\pi}{6}n\right), \quad x_3(n) = e^{j\left(\frac{\pi}{3}n + \frac{\pi}{6}\right)}, \quad x_4(n) = e^{2n}u(n)$$

6. 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 system with respect to the properties above $y(n) = x(n) + nx(n+1)$

7. Given $y[n] = x[n^2]$. Determine whether the system is linear, time invariant, memoryless and causal.

8. Determine whether the following is an energy signal or power signal.

a. $x_1[n] = 6\cos\left(\frac{\pi}{2}n\right)$
 b. $x_2[n] = 3(0.5)^n u(n)$.

9. Explain the digital signal processing system with necessary sketches and give its merits and demerits

10. 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 system with respect to the properties above $y(n) = x(n) + nx(n+1)$

UNIT – II

2 Marks Q & A

1. What is meant by region of convergence?

The region of convergence (ROC) of $X(z)$ is the set of all values of z for which $X(z)$ attains a finite value.

2. What are the properties of region of convergence?

The ROC is a ring or disk in the Z – plane centered at the origin.

The ROC cannot contain any poles.

The ROC of an LTI stable system contains the unit circle.

The ROC must be a connected region.

3. What are the properties of z- transform ?

i. Linearity: $z [a_1 x_1(n) + a_2 x_2(n)] = a_1 X_1(z) + a_2 X_2(z)$

ii. Shifting: (a) $z[x(n+m)] = z^m \left[X(z) - \sum_{i=0}^{m-1} x(i)z^{m-i} \right]$

1. (b) $z[x(n-m)] = z^{-m} X(z)$

iii. Multiplication: $z[n^m x(n)] = \left(-z \frac{d}{dz} \right)^m X(z)$

iv. Scaling in z- domain: $z[a^n x(n)] = X(a^{-1}z)$

v. Time reversal : $z[x(-n)] = X(z^{-1})$

vi. Conjugation: $z[x^*(n)] = X^*(z^*)$

vii. Convolution: $z \left[\sum_{m=0}^n h(n-m)r(m) \right] = H(z)R(z)$

viii. Initial value: $z[x(0)] = \lim_{z \rightarrow \infty} X(z)$

ix. Final value: $z[x(\infty)] = \lim_{z \rightarrow 1} (1 - z^{-1})X(z)$

4. What is the relationship between z-transform and DTFT?

The z-transform of $x(n)$ is given by $X(z) = \sum_{n=-\infty}^{\infty} x(n)z^{-n}$ (1)

Where $z = re^{j\omega}$

Substituting z value in eqn (1) we get,

$$X(re^{j\omega}) = \sum x(n)r^{-n}e^{-j\omega n} \dots\dots\dots (2)$$

The Fourier transform of $x(n)$ is given by

$$X(e^{j\omega}) = \sum_{n=-\infty}^{\infty} x(n)e^{-j\omega n} \dots\dots\dots (3)$$

Eqn (2) and Eqn (3) are identical, when $r=1$. In the z -plane this corresponds to the locus of points on the unit circle $|z|=1$.

Hence $X(e^{j\omega})$ is equal to $X(z)$ evaluated along unit circle, or

$$X(e^{j\omega}) = X(z) \big|_{z=e^{j\omega}}$$

For $X(e^{j\omega})$ to exist, the ROC of $X(z)$ must include the unit circle.

5. Write the commutative and distributive properties of convolution.

Commutative Property: $x(n)*h(n)=h(n)*x(n)$

Distributive property: $x(n)*[h_1(n)+h_2(n)]=[x(n)*h_1(n)]+[x(n)*h_2(n)]$

6. Determine the Z-transform and ROC for the signal $x(n)=\delta(n-k)+\delta(n+k)$.

Solution: $X(Z)=Z^{-k} + Z^{+k}$, $X(Z)$ will converge for all the values of Z , except $Z=0$ and ∞ .

7. Given a difference equation $y(n)=x[n]+3x[n-1]+2y[n-1]$. Determine the system function $H(Z)$.

Solution: On taking Z- Transform, $Y(Z)=X(Z)+3Z^{-1}X(Z)+2Z^{-1}Y(Z)$

$$Y(Z)[1-2Z^{-1}]=X(Z)[1+3Z^{-1}]$$

$$H(Z)=\frac{Y(Z)}{X(Z)}=\frac{1+3Z^{-1}}{1-2Z^{-1}}$$

8. Find the stability of the system whose impulse response $h(n)=\left(\frac{1}{2}\right)^n u(n)$

$$\text{For Stable, } \sum_{-\infty}^{\infty} |h(k)| < \infty$$

$$\sum_{-\infty}^{\infty} |h(k)| = 2 < \infty$$

Therefore the system is stable.

9. State the convolution property of Z transform.

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

10. What are the computational savings in evaluation of DFT using radix-2 FFT?

Multiplications: $N/2 \log N$

Additions: $N \log N$

11.State the meaning of bit reversal in FFT algorithm.

DIT FFT: I/P – bit reversal order & O/P – normal order

DIF FFT: O/P – bit reversal order & I/P – normal order

12.State initial value theorem.

If $x(n)$ and $X(z)$ are z-Transform pairs, then ,

$$x(0) = \lim_{z \rightarrow \infty} X(z)$$

provided that the first derivative of $x(t)$ should be laplace transformable.

13.State final value theorem.

If $x(n)$ and $X(z)$ are z-Transform pairs, then the final value of $x(z)$ is given as ,

$$x(\infty) = \lim_{z \rightarrow 1} (1-z^{-1})X(z) \text{ have no pole on or outside the unit circle.}$$

14.Define prewarping or prescaling.

For large frequency values the non linear compression that occurs in the mapping of Ω to ω is more apparent .This compression causes the transfer function at high frequency to be highly distorted when it is translate to the ω domain.This compression is being compensated by introducing a prescaling or prewarpping to frequency Ω scale.For bilinear transform Ω scale is converted into Ω^* scale (i.e)

$$\Omega^* = 2/T_s \tan (\Omega T_s/2) \text{ (prewarped frequency)}$$

15.Write the properties of frequency response of LTI system.

The frequency response is periodic function ω with a period of 2π .

- i) If $h(n)$ is real then $|H(\omega)|$ is symmetric and $\angle H(\omega)$ is antisymmetric.
- ii) If $h(n)$ is complex then the real part of $H(\omega)$ is antisymmetric over the interval $0 \leq \omega \leq 2\pi$.
- iii) The frequency response is a continuous function of ω .

16.Write short notes on the frequency response of first order system.

A first order system is characterized by the difference equation

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

The frequency response of first order system depends on the co efficient “a” in the difference equation governing the LTI system. When the value of “a|”

is in the range of $0 < a < 1$, the first order system behave as a low pass filter. When the value of “a” is in the range $-1 < a < 0$, the first order system behave as a high pass filter.

17. Write a short note on the frequency response of second order system.

A second order system is characterized by the difference equation

$$y(n) = 2r \cos \omega_0 y(n-1) - r^2 y(n-2) + x(n) - r \cos \omega_0 x(n-1)$$

The frequency response of second order system depends on the parameters “r” and “ ω_0 ” in the difference equation the LTI system. When the value of r is in the range of $0 < r < 1$, the second order system behave as a resonant filter with center frequency ω_0 . When the value of r is varied from 0 to 1, the sharpness of resonant peak increases.

18. Define discrete Fourier series.

Consider a sequence $x_p(n)$ with a period of N samples so that $x_p(n) = x_p(n/N)$; Then the discrete Fourier series of the sequence $x_p(n)$ is defined as

$$X_p(k) = \sum_{n=0}^{N-1} x_p(n) e^{-j2\pi kn/N}$$

19. What are the two basic differences between the Fourier transform of a discrete time signal with the Fourier transform of a continuous time signal?

For a continuous signal, the frequency range extends from $-\infty$ to $+\infty$.

On the other hand, the frequency range of a discrete – time signal extends from $-\pi$ to $+\pi$ (or 0 to 2π).

The Fourier transform of a continuous signal involves integration, whereas, the Fourier transform of a discrete – time signal involves summation process.

**20. Find the Fourier transform of a sequence $x(n) = 1$ for $-2 \leq n \leq 2$
= 0 otherwise.**

Solution:

$$\begin{aligned} X(\omega) &= \sum_{n=-\infty}^{\infty} x(n) e^{-j\omega n} = \sum_{n=-2}^2 e^{-j\omega n} \\ &= e^{j2\omega} + e^{j\omega} + 1 + e^{-j\omega} + e^{-j2\omega} \end{aligned}$$

16 Marks Questions

1. Find the Z- Transform of following :

$$(a)x(n) = \sin(n\omega_0)u(n)$$

$$(b)x(n) = \cos(\omega_0 n)u(n)$$

2. Determine the causal signal $x(n]$ having the Z- Transform

$$(a) X(Z) = \frac{1 + Z^{-1}}{1 - Z^{-1} + 0.5Z^{-2}}$$

$$(b) X(Z) = \frac{1}{1 - Z^{-1} + 0.5Z^{-2}}$$

$$(c) X(Z) = \frac{1 + 2Z^{-1} + Z^{-2}}{1 + 4Z^{-1} + 4Z^{-2}}$$

3. Obtain the linear convolution of $x(n) = \{3, 2, 1, 2\}$ & $\{1, 2, 1, 2\}$

4. A discrete time system is described by the following equations:

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

5. Determine $x(n]$ for the given $x(Z)$ with ROC

$$(1) |z| > 2$$

$$(2) |z| < 2$$

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

6. Find the Z- transform and its associated ROC for the following discrete time

$$\text{signal } x[n] = \left(\frac{1}{3}\right)^n u[n] + 5\left(\frac{1}{2}\right)^{-n} u[-n-1] \text{ (May/Jun 13)}$$

7. A system is described by the difference equation $y(n) - \frac{1}{2}y(n-1) = 5x(n)$. Determine the solution when the input $x(n) = (1/5)^n u(n)$ and the initial condition is given by $y(-1) = 1$ using Z transforms. (Nov/Dec'12)

8. Determine the impulse response of the system described by the difference equation $y(n) = y(n-1) - \frac{1}{2}y(n-2) + x(n) = x(n-1)$ using Z transforms and discuss its stability.

9. Evaluate the frequency response of the system described by the system

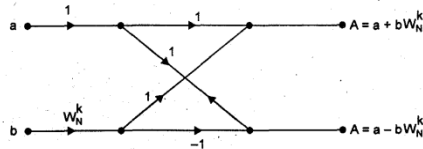
$$\text{function } H(Z) = \frac{1}{1 - 0.5Z^{-1}}$$

10. Using the Z- Transform determine the response $y[n]$ for $n \geq 0$ if $y[n] = \frac{1}{2} y[n-1] + x[n]$, $x[n] = \left(\frac{1}{3}\right)^n u[n]$, $y(-1) = 1$.

UNIT – III

2 Marks Q & A

1. Draw the basic butterfly diagram for Radix 2 DITFFT



2. Write the DTFT for

- $x(n) = a^n u(n)$
- $x(n) = 4\delta(n) + 3\delta(n-1)$

3. Define DTFT pair for a discrete sequence

$$\text{DTFT of } x(n) = X(\omega) = \sum_{n=-\infty}^{\infty} x(n) e^{-j\omega n}$$

4. Find the 4-point DFT of the sequence $x(n) = \{1, 1\}$
 $X(K) = \{2, 1-j, 0, 1+j\}$

5. In eight point decimation in time (DIT), what is the gain of the signal path that goes from $x(7)$ to $X(2)$?

Answer: $-W_8^0 W_8^2$

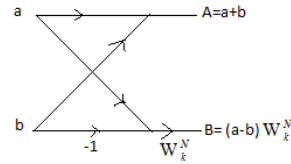
6. The first five points of the eight point DFT of a real valued sequence are $\{0.25, 0.125-j0.3018, 0, 0.125-j0.0518, 0\}$. Determine the remaining three points.

Answer: $X(5) = -0.125-j0.3018$, $X(6) = 0$, $X(7) = -0.125-j0.0518$

7. Find the discrete Fourier Transform for $\delta[n]$.

$$X(k) = \sum_{n=0}^{N-1} x(n) e^{-j2\pi kn/N} = \sum_{n=0}^{N-1} \delta(n) e^0 = 1.$$

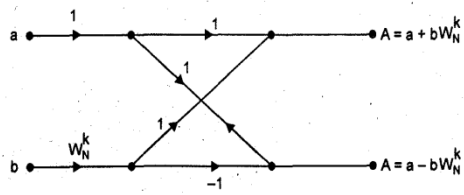
8. Draw the basic butterfly diagram for Radix 2 DIF FFT



9. What is decimation – in – time algorithm?

The computation of 8 – point DFT using radix-2 FFT , involves three stages of computations. Here $N=8=2^3$, therefore $r=2$ and $m=3$.

The given 8 – point sequence is decimated to 2- point sequences . For each 2 – point sequence, the 2-point DFT is computed . From the result of 2 – point DFT the 4 – point DFT can be computed. From the result of 4-point DFT , the 8 – point



DFT can be computed.

10.What is FFT?

The term Fast Fourier Transform (FFT) usually refers to a class of algorithms for efficiently computing the DFT.It makes use of the symmetry and periodicity properties of twiddle factor W_N^K to effectively reduce the DFT computation time.

It is based on the fundamental principle of decomposing the computation of DFT of a sequence of length N into successively smaller discrete Fourier transforms. The FFT algorithm provides speed increase factors , when compared with direct computation of the DFT, of approximately 64 and 205 for 256 points and 1024 – point transforms respectively.

11.Obtain the circular convolution the following sequences

$$x(n)=\{0,1,0,2 \}; h(n)=\{ 2,0,1 \}$$

Solution:

The circular convolution of the above sequences can be obtained by using matrix method.

$$\begin{bmatrix} h(0) & h(2) & h(1) \\ h(1) & h(0) & h(2) \\ h(2) & h(1) & h(0) \end{bmatrix} \begin{bmatrix} x(0) \\ x(1) \\ x(2) \end{bmatrix} = \begin{bmatrix} y(0) \\ y(1) \\ y(2) \end{bmatrix}$$

$$\begin{bmatrix} 1 & 2 & -2 \\ -2 & 1 & 2 \\ 2 & 2 & 1 \end{bmatrix} \begin{bmatrix} 1 \\ 2 \\ 1 \end{bmatrix} = \begin{bmatrix} 3 \\ 2 \\ -1 \end{bmatrix}$$

$$y(n) = \{3, 2, -1\}$$

12. What are the differences and similarities between DIF and DIT algorithms?

Differences:

For DIT, the input is bit reversal while the output is in natural order, whereas for DIF, the input is in natural order while the output is bit reversed.

The DIF butterfly is slightly different from the DIT butterfly, the difference being that the complex multiplication takes place after the add-subtract operation in DIF.

Similarities: Both algorithms require same number of operations to compute the DFT. Both algorithms can be done in place and both need to perform bit reversal at some place during the computation.

13. What is FFT?

The fast Fourier transforms (FFT) is an algorithm used to compute the DFT. It makes use of the Symmetry and periodicity properties of twiddle factor W_N^k to effectively reduce the DFT computation time. It is based on the fundamental principle of decomposing the computation of the DFT of a sequence of length N into successively smaller discrete Fourier transforms. The FFT algorithm provides speed-increase factors, when compared with direct computation of the DFT, of approximately 64 and 205 for 256-point and 1024-point transforms, respectively.

14. What is the main advantage of FFT?

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

15. Relationship between DTFT and DFT and Z transform

DTFT output is continuous in time where as DFT output is Discrete in time. Whereas Z transform is order of discretization in z-domain

16.Distinguish between FIR and IIR filters.

S.No.	FIR filter	IIR filter
	These filters can be easily designed to have perfectly linear phase.	These filters do not have linear phase.
	FIR filters can be realized recursively and non-recursively.	IIR filters can be realized recursively.
	Greater flexibility to control the shape of their magnitude response.	Less flexibility, usually limited to kind of filters.
	Errors due to roundoff noise are less severe in FIR filters, mainly because feedback is not used.	The roundoff noise in IIR filters are more.

17.Define the Discrete Fourier transformation of a given sequence x(n).

The N- point DFT of a sequence x(n) is

$$X(k) = \sum_{n=0}^{N-1} x(n)e^{-j2\pi kn/N} \quad k=0, 1, 2, \dots, N-1.$$

18.Write the formula for N- point IDFT of a sequence X(k).

The N-point IDFT of a sequence X(k) is

$$x(n) = \frac{1}{N} \sum_{k=0}^{N-1} X(k)e^{j2\pi kn/N} \quad n=0, 1, 2, \dots, N-1.$$

19.List any four proerties of DFT.

(a) Periodicity

If X(k) is N- point DFT of a finite duration sequence x(n)
then

$$x(n+N) = x(n) \text{ for all } n$$

$$X(k+N) = X(k) \text{ for all } k.$$

(b)Linearity

If $X_1(k) = \text{DFT}[x_1(n)]$ and $X_2(k) = \text{DFT}[x_2(n)]$

then

$$\text{DFT}[a_1x_1(n)+a_2x_2(n)]=a_1X_1(k)+a_2X_2(k)$$

(c) Time reversal of a sequence

If DFT $\{x(n)\}=X(k)$,

then

$$\text{DFT}\{x((-n))_N\}=\text{DFT}\{x(N-n)\}=X((-k))_N=X(N-k)$$

(d)Circular time shifting of a sequence

If DFT $\{x(n)\}=X(k)$,

then

$$\text{DFT}\{x((n-l))_N\}=X(k)e^{-j2\pi kl/N}$$

20. IF N-point sequence $x(n)$ has N- point DFT $X(k)$ then what is the DFT of the following?

$$(i)x^*(n) \quad (ii)x^*(N-n) \quad (iii)x((n-l))_N \quad (iv)x(n)e^{j2\pi ln/N}$$

Solution:

$$(i)\text{DFT}\{x^*(n)\}=X^*(N-k)$$

$$(ii)\text{DFT}\{x^*(N-n)\}=X^*(k)$$

$$(iii)\text{DFT}\{x((n-l))_N\}=X(k)e^{-j2\pi kl/N}$$

$$(iv)\text{DFT}\{x(n)e^{j2\pi ln/N}\}=X((k-l))_N$$

16 Marks Questions

1. Discuss various properties of DFT.
2. Perform circular convolution of two sequences
 $x_1(n) = \{0.2, 0.4, 0.6, 0.8, 1, 1.2, 1.4, 1.6\}$
 $x_2(n) = \{0.1, 0.3, 0.5, 0.7, 0.9, 1.1, 1, 3, 1\}$ (Dec. 2006)
3. Develop a Radix-2, 8-point DIF FFT algorithm with neat flow chart.
4. Find the inverse DFT of $X(k) = \{1, 2, 3, 4\}$
5. An 8-point sequence is given by $x(n) = \{2, 2, 2, 2, 1, 1, 1, 1\}$. Compute 8-point DFT of $X(n)$ by radix -2 DIT FFT. Also sketch the magnitude and phase spectrum.
6. Compute the DFT of $x(n)=\{1, 1, 0, 0\}$
7. Describe the following properties of DFT.

(i) Time reversal (ii) Circular convolution

8. Obtain the circular convolution of $x_1(n)=\{1, 2, 2, 1\}$, $x_2(n)=\{1, 2, 3, 1\}$

9. Describe the following properties of DFT.

(i) Time reversal (ii) Circular convolution

10. Find the inverse DFT of $X(k)=\{7, -\sqrt{2}-j\sqrt{2}, -j, \sqrt{2}-j\sqrt{2}, 1, \sqrt{2}+j\sqrt{2}, j, -\sqrt{2}+j\sqrt{2}\}$

UNIT – IV

2 Marks Q & A

1. What are the properties of Chebyshev filter?

- The magnitude response of the Chebyshev filter exhibits in ripple either in pass band or in the stop band according to the type.
- The magnitude response approaches the ideal response as the value of N increases.
- The Chebyshev type – 1 filters are all pole designs.
- The poles of Chebyshev filter lies on an ellipse.
- The normalized magnitude function has a value of $\frac{1}{\sqrt{1+\epsilon^2}}$ at the cutoff frequency Ω_c .

2. Distinguish between FIR and IIR filter.

Sl No	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 are easily realized recursively.

3.	Greater flexibility to control the shape of their magnitude response.	Less flexibility, usually limited to specific kind of filters.
4.	Error due to round off noise are less severe in FIR filters, mainly because feedback is not used.	The rounds off noise in IIR filters are more.

3. Write the condition for stability of digital filter

- Choose the desired (ideal) frequency response $H_d(\omega)$ of the filter.
- Evaluate the Fourier series co-efficient of $H_d(\omega)$ which gives the desired impulse response $h_d(n)$. Where $h_d(n) = \frac{1}{2\pi} \int_{-\pi}^{\pi} H_d(\omega) e^{j\omega n} d\omega$
- Truncate the infinite sequence $h_d(n)$ to a finite duration sequence $h(n)$.
- Take Z – transform of $h(n)$ to get a non causal filter transfer function $H(z)$ of the FIR filter.
- Multiply $H(z)$ by $z^{-\left(\frac{N-1}{2}\right)}$ to convert noncausal transfer function to a realizable causal FIR filter transfer function.

$$f. H(z) = z^{-\left(\frac{N-1}{2}\right)} \left[h(0) + \sum_{n=1}^{\frac{N-1}{2}} h(n)(z^n + z^{-n}) \right]$$

4. What is Gibbs phenomenon?

One possible way of finding an FIR filter that approximates $H(e^{j\omega})$ would be to truncate the infinite Fourier series at $n = \pm \left[\frac{N-1}{2} \right]$. The abrupt truncation of the series will lead to oscillation both in passband and in stopband. This phenomenon is known as Gibbs phenomenon.

5. What is window and why it is necessary?

One possible way of finding an FIR filter that approximates $H(e^{j\omega})$ would be to truncate the infinite Fourier series at $n = \pm \left[\frac{N-1}{2} \right]$. The

abrupt truncation of the series will lead to oscillation both in passband and in stopband. These oscillations can be reduced through the use of less abrupt truncation of the Fourier series. This can be achieved by multiplying the infinite impulse response with a finite weighing $w(n)$, called a window.

6. Mention any two procedures for digitizing the transfer function of an analog filter.

The two important procedures for digitizing the transfer function of an analog filter are

(i) Impulse invariance method. (ii) Bilinear transformation method.

7. What is frequency warping?

In bilinear transformation the relation between analog and digital frequencies is nonlinear. When the s-plane is mapped into z-plane using bilinear transformation, this nonlinear relationship introduces distortion in frequency axis, which is called frequency warping.

8. Explain the technique of prewarping.

In IIR filter design using bilinear transformation the specified digital frequencies are converted to analog equivalent frequencies, which are called prewarp frequencies. Using the prewarp frequencies, the analog filter transfer function is designed and then it is transformed to digital filter transfer function.

9. Compare the impulse invariant and bilinear transformations.

S l. N o	Impulse Invariant transformation	Bilinear transformation
1	It is many – to – one mapping	It is one – to – one mapping.
2	The relation between analog and digital frequency is	The relation between analog and digital frequency is nonlinear.

	linear.	
3	To prevent the problem of aliasing the analog filters should be band limited.	There is no problem of aliasing and so the analog filter need not be band limited.
4	The magnitude and phase response of analog filter can be preserved by choosing low sampling time or high sampling frequency.	Due to the effect of warping, the phase response of analog filter cannot be preserved . But the magnitude response can be preserved by prewarping.

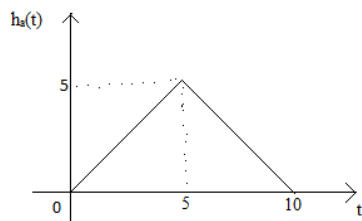
10.What is meant by linear phase response of a filter?

The maintenance of a constant gain between the cut-off frequencies.

11.Is the given transfer function $H(z) = \frac{1 + 0.8z^{-1}}{1 - 0.9z^{-1}}$ represents low pass filter or high pass filter?

Answer: Low pass filter

12.The impulse response of an analog filter is shown in below figure. Let $h(n) = h_a(nT)$. Where $T=1\text{sec}$. Determine the System function.



Solution : $h_a(nT) = \begin{cases} nt, & 0 < t < 5 \\ 10 - nT, & 5 < t < 10 \end{cases}$

$$H(Z) = \sum_{n=0}^5 nZ^{-n} + \sum_{n=5}^{10} (10 - n)Z^{-n}$$

13.What is prewarping? Why it is employed?

In IIR filter design using bilinear transformation , the conversion of the specified digital frequencies to analog frequencies is called prewarping.

The prewarping is necessary to eliminate the effect of warping on amplitude response.

14. What are the advantages and disadvantages of digital filters?

Advantages:

- a) High thermal stability due to absence of resistors, inductors and capacitors.
- b) The performance characteristics like accuracy, dynamic range, stability and tolerance can be enhanced by increasing the length of the registers.
- c) The digital filters are programmable.
- d) Multiplexing and adaptive filtering are possible.

Disadvantages:

- a) The bandwidth of the discrete signal is limited by the sampling frequency.
- b) The performance of the digital filter depends on the hardware used to implement the filter.

15. 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 (i.e., in this transformation the impulse response of the digital filter will be sampled version of the impulse response of the analog filter.)

16. What is the main objective of impulse invariant transformation?

The objective of this method is to develop an IIR filter transfer function whose impulse is the sampled version of the impulse response of the analog filter. Therefore the frequency response characteristics of the analog filter are preserved.

17. Write the impulse invariant transformation used to transform real poles with and without multiplicity.

The impulse invariant transformation used to transform real poles (at $s = -p_i$) without multiplicity is

$$\frac{1}{s + p_i} \text{ is transformed to } \rightarrow \frac{1}{1 - e^{-p_i T} z^{-1}}$$

The impulse invariant transformation used to transform multiple real pole (at $s = -p_i$) is

$$\frac{1}{(s + p_i)^m} \text{ is transformed to } \rightarrow \frac{(-1)^{m-1}}{(m-1)!} \frac{d^{m-1}}{dp_i^{m-1}} \frac{1}{1 - e^{-p_i T} z^{-1}}$$

18.What is the relation between digital and analog frequency in impulse invariant transformation.

The relation between analog and digital frequency in impulse invariant transformation is given by

Digital frequency, $\omega = \Omega T$

Where, Ω - Analog frequency and T - Sampling time period

19.What is Bilinear transformation?

The Bilinear transformation is a conformal mapping that transforms the s-plane to z-plane . In this mapping the imaginary axis of s-plane is mapped into the unit circle in z-plane, the left half of s-plane is mapped into interior of unit circle in z-plane and the right half of s-plane is mapped into exterior of unit circle in z-plane . The Bilinear mapping is a one – to-one mapping and it is accomplished when

$$s = \frac{2}{T} \frac{1 - z^{-1}}{1 + z^{-1}}$$

20.What is the relation between digital and analog frequency in Bilinear transformation?

In Bilinear transformation , the digital frequency and analog frequency are related by the equation,

$$\text{Digital frequency, } \omega = 2 \tan^{-1} \frac{\Omega T}{2} \quad \text{or}$$

$$\text{Analog frequency } \Omega = \frac{2}{T} \tan \frac{\omega}{2}$$

Where, Ω - Analog frequency & T - Sampling time period

16 Marks Questions

1. Obtain the Direct form –I, Direct form – II, Cascade form and Parallel form structure for the system described by
 - (a) $y(n) = -0.1y(n-1) + 0.2y(n-2) + 3x(n) + 3.6x(n-1) + 0.6x(n-2)$
 - (b) $y(n) = 0.5y(n-1) + 0.25y(n-2) + x(n) + x(n-1)$
2. Design an FIR linear phase, digital filter approximating the ideal frequency response

$$H_d(\omega) = \begin{cases} 1, & \text{for } |\omega| \leq \frac{\pi}{6} \\ 0, & \text{for } \frac{\pi}{6} \leq |\omega| \leq \pi \end{cases}$$

3. Determine the coefficients of a 11- tap filter based on the window method with a Hamming window

4. Design a FIR filter having following specification

$$H_d(e^{j\omega}) = \begin{cases} 1, & \text{for } \frac{\pi}{4} \leq |\omega| \leq \pi \\ 0, & \text{otherwise} \end{cases}$$

and given that N=11. Using (a) Hanning Window (b) Hamming Window

5. Convert analog filter $H_{a(s)} = \frac{2}{(s+1)(s+2)}$ into digital filter by means of bilinear transformation when T = 1 sec

6. Convert the analog filter with system function

$H_{a(s)} = \frac{s+0.1}{(s+0.1)^2 + 9}$ into digital IIR filter by means of Impulse invariant transformation

7. Design (a) Butterworth and (b) Chebyshev analog high pass filter that will pass all signals of radian frequencies greater than 200 rad/sec with no more than 2 dB attenuation and have a stop band attenuation of greater than 20 dB for all Ω less than 100 rad/sec.

8. Obtain the cascade realization of

$$H(z) = \frac{2 + Z^{-1} + Z^{-2}}{\left(1 + \frac{1}{2}Z^{-1}\right)\left(1 - \frac{1}{4}Z^{-1}\right)\left(1 + \frac{1}{8}Z^{-1}\right)}$$

9. Design a chebyshev filter for the following specification using (a) bilinear transformation (b) Impulse invariance method.

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

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

10. Realize the following system functions using a minimum number of multipliers

$$(a) H(z) = 1 + \frac{1}{2}z^{-1} + \frac{3}{4}z^{-2} + \frac{1}{2}z^{-3} + z^{-4}$$

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

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

UNIT – V

2 Marks Q & A

1. Give the special features of DSP processors. (Nov/Dec'11)

- (i) Harmonics can be analyzed using Fourier analysis.
- (ii) Generation of pulses
- (iii) Discretizing the waveform

2. What is the function of parallel logic unit DSP processor? (Nov/Dec'12)

The parallel logic unit is a second logic unit, that execute logic operations on data Without affecting contents of accumulator.

3. Define Period gram. (May/Jun'12)

Periodic analysis of the waveform can be analyzed.

4. What is meant by bit reversed addressing mode? What is the application for which this addressing mode is preferred? (Nov/Dec'13)

Bit-reverse addressing is a special type of indirect addressing. It uses one of the auxiliary registers (AR0–AR7) as a base pointer of an array and uses temporary register 0 (T0) as an index register. When you add T0 to the auxiliary register using bit-reverse addressing, the address is generated in a

bit-reversed fashion, with the carry propagating from left to right instead of from right to left.

Application: Bit-reversed addressing, a special addressing mode useful for calculating FFTs

5. Compare the RISC and CISC processors. (Nov/Dec'13)

CISC	RISC
Emphasis on hardware	Emphasis on software
Includes multi-clock complex instructions	Single-clock, reduced instruction only
Memory-to-memory: "LOAD" and "STORE" incorporated in instructions	Register to register: "LOAD" and "STORE" are independent instructions
Small code sizes, high cycles per second	Low cycles per second, large code sizes
Transistors used for storing complex instructions	Spends more transistors on memory registers
If it reads as above (<i>i.e.</i> as CISC computer), it means a computer that has a <u>C</u> omplex <u>I</u> nstruction <u>S</u> et <u>C</u> hip as its cpu.	If it reads as above (<i>i.e.</i> as RISC computer), it means a computer that has a <u>R</u> educed <u>I</u> nstruction <u>S</u> et <u>C</u> hip as its cpu

6. Mention one important feature of Harvard architecture. (May/Jun'13)

(i) High Compatibility, (ii) RISC Processor

7. What is the advantage of Pipelining? (May/Jun'13)

It provides sequential flow of execution with one after the other process without any interruption.

8. What are the different buses of TMS320C54X and their functions?

The C5X architecture has four buses and their functions are as follows:

- a. Program bus (PB):
It carries the instruction code and immediate operands from program memory Space to the CPU.
- b. Program address bus (PAB):
It provides addresses to program memory space for both reads and writes.
- c. Data read bus (DB):
It interconnects various elements of the CPU to data memory space.
- d. Data read address bus (DAB):
It provides the address to access the data memory space.

9. Briefly explain about Harvard architecture.

The principal feature of Harvard architecture is that the program and the data memories lie in two separate spaces, permitting full overlap of instruction fetch and execution called pipelining. Thus reducing the operation time. Typically these types of instructions would involve their distinct type.

- Instruction fetch
- Instruction decode
- Instruction execute

10. How DSP processor works faster than general purpose processors?

Digital signal processors are basically high speed microprocessors with hardware 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

11. What is meant by pipeline technique? What are the different stages in pipelining?

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, execution speed is increased.

The four stages are

- (i) Fetch
- (ii) Decode
- (iii) Read
- (iv) Execution

12. What is the function of parallel logic unit in DSP processor

The parallel logic unit (PLU) can directly set, clear, test, or toggle multiple bits in control/status register or any data memory location. The PLU provides a direct logic operation path to data memory values without affecting the contents of the ACC or the PREG.

13. State the intended applications of DaVinci Digital Media processors

Image compression, Image coding, speech compression, multirate signal filters

14. List out the typical features of Digital Signal Processor

- a. Fixed-point processor (TMS320C5000, 56000...) or floating point processor (TMS320C67, 96000...)
- b. Architecture optimized for intensive computation. For instance the TMS320C67 can do 1000 Million floating point operations a second (1 GIGA Flop).
- c. Narrow address bus supporting a only limited amounts of memory.
- d. Specialized addressing modes to efficiently support signal processing operations (circular addressing for filters, bit-reverse addressing for Fast Fourier Transforms...etc.)
- e. Narrow data formats (16 bits or 32 bits typical).
- f. Many specialized peripherals integrated on the chip (serial ports, memory,
- g. Low power consumption & Low cost.

15. What is Barrel Shifter?

The Barrel Shifter is used for scaling operations such as prescaling an input data-memory operand or the accumulator value before an ALU operation;

performing a logical or arithmetic shift of the accumulator value;
normalising the accumulator; postscaling the accumulator before storing the accumulator value into data memory.

16.What are the elements present in CPU of 54X processor?

- (i) 40-Bit Arithmetic Logic Unit (ALU)
- (ii) Two 40- Bit Accumulator Registers
- (iii) Barrel Shifter
- (iv) Multiply / Accumulate Block
- (v) 16-Bit Temporary Register (T)
- (vi) 16-Bit Transition Register
- (vii) Compare, select and store unit
- (viii) Exponent Encoder

17.List the various registers used with ARAU.

Eight auxiliary registers (AR0 –AR7)
Auxiliary register pointer (ARP)

18.What are the factors that influence selection of DSPs?

- 1. Architectural features
- 2.Execution speed
- 3.Type of arithmetic
- 4.Word length

19.What is pipeline depth?

The number of pipeline stages is referred to as the pipeline depth.

20.What is the pipeline depth of TMS320C50, TM 320C54x?

TMS320C50 – 4 TM 320C54x – 6

16 Marks Questions

- 1. Explain Von Neumann, Harvard architecture and modified Harvard architecture for the computer.
- 2. Write short notes on auxiliary registers
- 3. Write short notes on circular addressing mode
- 4. Write short notes on memory mapped register addressing
- 5. Explain the advantages and disadvantages of VLIW architecture
- 6. Draw the block diagram of Harvard architecture and explain

7. Explain about pipelining in DSP
8. Explain the addressing modes of a DSP processor.
9. Describe in detail the architecture of TMS 320C 54 DSP processor and state the main features of this processor.
10. Enumerate various applications of DSP processors.

-----ALL THE BEST-----