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QUESTION BANK

Name of the Department : **Electrical and Electronics Engineering**

Subject Code & Name : **EE8591 & Digital Signal Processing**

Year & Semester : **III & V**

UNIT I INTRODUCTION

PART-A

1. What do you understand by the terms: Signal and Signal Processing?
A signal is defined as any physical quantity that varies with time, space, or any other independent variable. Signal processing is any operation that changes the characteristics of a signal. These characteristics include the amplitude, shape, phase and frequency content of a signal.

2. What are the classifications of signals?
There are five methods of classifying signals based on different features:

(a) Based on independent variable.
- (i) Continuous time signal
- (ii) Discrete time signal.

(b) Depending upon the number of independent variable.
- (i) One dimensional signal,
- (ii) Two dimensional signal.
- (iii)Multi dimensional signal.

(c) Depending upon the certainty by which the signal can be uniquely described as
- (i) Deterministic signal.
- (ii)Random signal.

(d) Based on repetition nature.
- (i)Periodic signal.
- (ii)Non – Periodic signal.

(e) Based on reflection
- (i)Even signal.
- (ii)Odd signal.

3. Define discrete system.
A discrete time system is a device or algorithm that operates on a discrete time input signal x(n) , according to some well defined rule , to produce another discrete – time signal y(n) called the output signal.

4. 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.

5. Differentiate Continuous time and Discrete time signal.

Continuous time signal: It is also referred as analog signal i.e., the signal is represented continuously in time.

Discrete time signal : Signals are represented as sequence at discrete time intervals .

6. Test whether the system governed by the relation $y(n)=\sum_{k=-\infty}^{n} x(k)$ is time –invariant or not?

If the output is delayed by k units in time

$$y(n)=x(-\infty)+............+x(0)+........x(n)$$

$$y(n-k)=x(-\infty)+............+x(0)+........x(n-k)----(1)$$

If we delay the input by k units in time

$$y(n)=x(-\infty)+............+x(0)+........x(n)$$

$$y(n,k)=x(-\infty)+............+x(0)+........x(n-k)----(2)$$

Equation (1)=(2). So, that the system is time invariant.

7. What is Deterministic signal and random signal? Give example.

A signal that can be uniquely determined by a well - defined process such as a mathematical expression or rule , or look-up table is called a deterministic signal.

Example : A sinusoidal signal $v(t)=V_m \sin \omega t$

A signal that is generated in a random fashion and cannot be predicted ahead of time is called a " randomsignal".

Example : Speech signal , ECG signal and EEG signal.

8. Define (a) Periodic signal (b) Non – periodic signal.

Periodic signal: A signal x(n) is periodic with period N if and only if x(n+N)=x(n)for all n.

Non – periodic signal: If there is no value of N that satisfies the above equation the signal is called non-periodic or aperiodic.

9. Define symmetric and antisymmetric signals.

Symmetric signal: A real valued signal x(n) is called symmetric if x(-n) = x(n).

Antisymmetric signal: A signal x(n) is called antisymmetric if x(-n) = -x(n).

10. Differentiate energy and power signals?

Energy signal:

- The energy of a discrete time signal x(n) is defined as $E=\sum_{n=-\infty}^{\infty}|x(n)|^2$
- A signal x(n) is called an energy signal if and only if the energy obeys the relation $0<E<\infty$ and an energy signal P=0.

Power signal :

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

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

- A signal x(n) is called power signal if and only if the average power P satisfies the condition $0 < P < \infty$ and E= ∞.

11. What are the different types of operations performed on discrete – time signals?

The different types of operations performed on discrete – time signals are

(1)Delay of a signal
(2) Advance of a signal
(3)Folding or Reflection of a signal
(4) Time scaling
(5) Amplitude scaling
(6)Addition of signals
(7)Multiplication

12. What is a static and dynamic system?

A discrete –time system is called static or "memory less" if its output at any instants 'n' depends on the input samples at the same time , but not an past or future samples of the input.

Ex.,y(n) = ax(n) Y(n)=ax^2(n)

In any other case, the system is said to be dynamic or to have memory.

Ex.,y(n) = ax(n-1)+x(n-2) y(n)=x(n)+x(n-1)

13. What is a time – invariant system?

A system is called time – invariant if its input – output characteristics do not change with time.

Ex.,y(n)=x(n)+x(n-1)

14. What is a causal system?

A system is said to be causal if the output of the system at any time n depends only on present and past inputs, but does not depend on future inputs.

This can be expressed mathematically as,

y(n)=F[x(n), x(n-1), x(n-2)………]

15. Define a 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, their exist some finite numbers, M_x and M_y such that,

$$|x(n)| \le Mx < \infty and |y(n)| \le My < \infty$$

16. 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.

17. 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.

18. Define Nyquist rate.

The Nyquist rate or frequency is the minimum rate at which a finite bandwidth signal needs to be sampled to retain all of the information. For a bandwidth of span fm Hz, the Nyquist frequency is $2f_m$ Hz.

19. What is aliasing effect? How can aliasing be avoided?

The superimposition of high frequency component on the low frequency is known as "frequency aliasing" or "aliasing effect".To avoid aliasing the sampling frequency must be greater than twice the highest frequency present in the signal.

20. What is a linear time invariant system?

An LTI system is one which possess 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 $T[a_1 x_1(n)+a_2 x_2(n)] = a_1 y_1(n)+ a_2 y_2(n)$

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

$\omega_1=600\pi$ $\quad\quad$ $\omega_2=1800\pi$
$2\pi f_1= 600\pi$ $\quad\quad$ $2\pi f_2= 1800\pi$
$f_1= 300$Hz $\quad\quad$ $f_2= 900$Hz
Nyquist rate $F_s=2f_m= 2x900= 1800$Hz.

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

Solution: Fundamental period, N= $\left(\frac{2\pi}{\omega_o}\right)m$,

Where $\omega_o = \frac{30\pi}{105} = \frac{105}{15}m,$
when m=1 & N = 7 periods.

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

$\omega=500\pi$, $2\pi f= 250\pi$
f= 250Hz
hence Nyquist rate $F_s=2f_m= 2x250= 500$Hz

24. 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

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

Here E= ∞ and P= Finite. Therefore the given signal is a power signal.

25.Given a Continuous signal x(t) =2 cos300πt.What is the nyquist rate and fundamental frequency of the signal.

$\omega = 300\pi$,

$2\pi f = 150\pi$

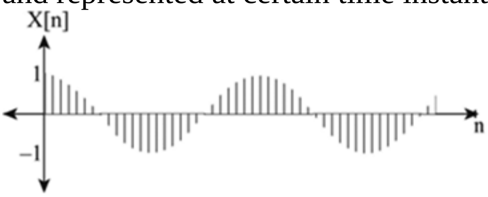
f= 150Hz hence Nyquist rate $F_s = 2f_m = 2\times150 = 300$Hz.

26.What is an Anti –Aliasing filter?

An anti-aliasing filter (AAF) is a filter used before a signal sampler to restrict the bandwidth of a signal to approximately or completely satisfy the sampling theorem over the band of interest.

27. Distinguish between discrete signal and discrete signal representation.

Discrete time signals: Discrete time signals are the signals or quantities that can be defined and represented at certain time instants of the sequence.



Discrete signal representation:A discrete time signal can be represented in (i)Tabular representation (ii) Graphical representation (iii) Sequence representation

28.If x(n) = x(n+1)+x(n-2), is the system causal?

When n=0; x(0)= x(1)+x(-2).

When n= -1, x(-1) = x(0)+x(-3).

The system is non causal, since the system depends on future.

29. List the sampling techniques.

There are three types of sampling techniques,

i) Impulse sampling,

ii) Natural sampling

iii) Flat top sampling.

30. Define spectral density.

The spectral density of the signal describes the energy or power present in the signal as a function of frequency, per unit frequency. The power spectral density has infinite energy and finite power.

PART-B

1. Determine which of the following signals are periodic and determine the fundamental period also.

$(i) x(t) = 20\sin 25\pi t$ $\quad (ii) x(t) = 20\sin\sqrt{5}\,t$

$(iii) x(t) = 10\cos 10\pi t$ $\quad (iv) x(t) = 3\cos(5t + \frac{\pi}{6})$

2.Explain the digital signal processing system with necessary sketches and give its merits and demerits.State the advantages of convolution technique.

(or) Explain the process converting an analog signal to discrete time amplitude signal with necessary diagram.

3. i) Find the impulse response of a Discrete Time LTI system $y(n) = y(n-1) + 0.5y(n-2) + x(n) + x(n-1)$

(ii)What is meant by sampling? Explain sampling theorem? What is spectral density

(iii) Explain the properties of discrete time system?

4)i)A discrete time system is represented by difference equation verify whether it is linear? Shift Invariant? Causal? In each case justify your answer. $y(n) = 3y(n-1) - nx(n) + 4x(n-1) + 2x(n+1)$ *and* $n \geq 0$.

ii) What is meant by quantization and quantization error ?

5. Check whether following are linear, time invariant, causal and stable for the following system,

(i)y(n) = x(n)+nx(n+1) , (ii) y(n) = cos x(n), (iii) y(n)= x(-n=5), (iv) y(n)= x(2n)

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

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

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

$x_4(n) = e^{2n}u(n)$, $x_5(n) = \left(\frac{1}{5}\right)^n u(n)$, $x_6(n) = e^{j\left(\frac{\pi n}{3}+\frac{\pi}{7}\right)}$

8. 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)

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

10.(i) Check the causality and stability of the systems

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

(ii)Check the system for linearity and time variance

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

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

$(i) x_1(n) = \left(\frac{1}{2}\right)^n u(n)$

$(ii) x_2(n) = Sin\left(\frac{\pi}{6} n\right)$

State and prove sampling theorem.

12. (i) Determine if the signals $x_1(n)$ and $x_2(n)$ are power , energy or neither energy nor power signals. $x_1(n) = \left(\frac{1}{3}\right)^n u(n)$ and $x_2(n) = e^{2n} u(n)$

(ii)What is the input signal x(n) that will generate the output sequence Y(n)={1,5,10,11,8,4,1} for a system with impulse response h(n) {1,2,1}.

UNIT II DISCRETE TIME ANALYSIS

PART-A

1. Define Z – transform.

The Z –transform of a discrete time signal x(n) is denoted by X(z) and it is defined as,

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

Where z is a complex variable and n is the sequence interval. x(n) and X(z) is called z-transform pair.

2. 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. ROC for a discrete time signals x(n) is defined as a continuous region in z plane where the z-transform converges.

3. What are the properties of region of convergence?

(i) The ROC is a ring or disk in the Z – plane centered at the origin.
(ii) The ROC cannot contains any poles.
(iii)The ROC of an LTI stable system contains the unit circle.
(iv)The ROC must be a connected region.

4. Define system function.

Let x(n) and y(n) is the input and output sequences of an LTI system with impulse response h(n). Then the system function of the LTI system is defined as the ratio of Y(z) and X(z), i.e.,

$$H(z) = \frac{Y(z)}{X(z)}$$ where, Y(z) is the z – transform of the output signal y(n)

X(z) is the z – transform of the input signal x(n)

5. Why FT of a discrete time signal is called signal spectrum?

By taking Fourier transform of a discrete time signal x(n) , it is decomposed into its frequency components . Hence the Fourier transform is called signal spectrum.

6. What is the frequency response of LTI system?

The Fourier transform of the impulse response h(n) of the system is called frequency response of the system. It is denoted by H(ω).

7. What are the properties of z- transform ?

Linearity: $z\left[a_1 x_1(n) + a_2 x_2(n)\right] = a_1 X_1(z) + a_2 X_2(z)$

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

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

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

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

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

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

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

Initial value: $z[x(0)] = \underset{z\to\infty}{Lt}\, X(z)$

Final value: $z[x(\alpha)] = \underset{z\to 1}{Lt}(1-z^{-1})X(z)$

8. State Parseval's relation in z-transform.

If $x_1(n)$ and $x_2(n)$ are complex valued sequences, then

$$\sum_{n\to\infty} x_1(n)x_2^*(n) = \frac{1}{2\pi j}\oint_c X_1(v)X_2\left(\frac{1}{v^*}\right)v^{-1}dv$$

9. 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,

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

1. 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 $-\pi \text{ to } +\pi \quad (or\ 0 \text{ to } 2\pi)$.
2. The Fourier transform of a continuous signal involves integration , whereas , the Fourier transform of a discrete – time signal involves summation process.

11.Write the commutative and distributive properties of convolution.

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

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

12. Find the Fourier transform of a sequence x(n) = 1, for $-2 \le n \le 2$
= 0, otherwise.

Solution: $X(\omega) = \sum_{n=-\infty}^{\infty} x(n)e^{-j\omega n} = \sum_{n=-2}^{2} e^{-j\omega n}$

$= e^{j2\omega} + e^{j\omega} + 1 + e^{-j\omega} + e^{-j2\omega}$

$= 1 + 2\cos\omega + 2\cos 2\omega$

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

i) The frequency response is periodic function ωwith a period of 2π.
ii) If h(n) is real then $|H(\omega)|$ is symmetric and $\angle H(\omega)$ is antisymmetric.
iii) If h(n) is complex then the real part of $H(\omega)$ is antisymmteric over the interval $0 \le \omega \le 2\pi$.

14. Define Fourier transform of a discrete time signal.
The Fourier transform of a discrete time signal x(n) is defined as

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

15. 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, $Y(Z)= X(Z)+3Z^{-1} X(Z)+2Z^{-1} Y(Z)$

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

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

16. Define DTFT pair for a discrete sequence. (Dec 2012)
The discrete time fourier transform of a discrete time signal x(n) is defined as

$$F\{x(n)\}=X(\omega)= \sum_{n=-\infty}^{+\infty} x(n)\, e^{-j\omega n}$$

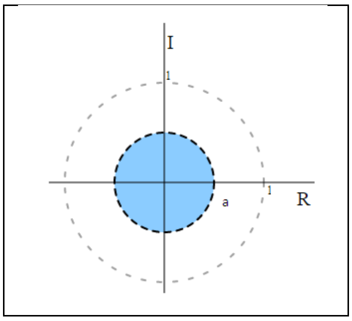
The discrete time fourier transform exists only if $\sum_{n=-\infty}^{+\infty} x(n) < \infty$

The inverse discrete time fourier transform of X(ω) is defined as

$$F^{-1}\{X(\omega)\}= x(n) = \frac{1}{2\pi}\int_{-\pi}^{\pi} X(\omega)\, e^{j\omega n} d\omega$$

17..What is the draw back in Discrete Time Fourier transform and how it is overcome?
The drawback in Discrete Time Fourier transform(DTFT) is that it is a continuous function of ω and so it cannot be processed by digital system. This drawback is overcome by using Discrete Fourier Transform(DFT). The DFT converts the continuous function of ω to a discrete function of ω.

18. Find the Z-transform and ROC of the discrete time signal
$x(n) = -a^n u(-n-1)$, a>0



$$X(Z)=\sum_{n=-\infty}^{\infty} x(n)Z^{-n}$$

$$=-\sum_{n=1}^{\infty}(a^{-1}Z)^{n}$$

$$=-\left[\frac{1}{1-a^{-1}Z}-1\right]$$

$$=\frac{-Z}{Z-a}$$

19.What is ROC of Z Transform? State its properties

The **region of convergence** (**ROC**) is the set of points in the complex plane for which the **Z-transform** summation converges.

Properties:

- The ROC for a finite duration sequence includes entire z-plane except z=0z=0 and/or |z|=∞.
- ROC does not contain any poles
- ROC is the ring in the z-plane centered about origin.

20.State initial and final value theorem of Z transform.

Initial value theorem for causal signal u(0) = lim z→∞ U(z) if the limit exists.

The final value theorem for z-transforms states that if lim k→∞ x(k) exists, then lim k→∞ x(k) = limz→1 (z − 1)X(z).

21.Determine the Z Transform and ROC of the following finite duration signals

(i) x(n)= {3,2,2,3,5,0,1}

(ii)x(n)= Ƨ(n-k)

(i) x(n)= {3,2,2,3,5,0,1}

x(0) =3; x(1)=2; x(2)=2; x(3)=3; x(4)=5; x(5)=0; and x(6)=1

By the definition of Z-transform,

$$Z\{x(n)\}=X(Z)=\sum_{n=-\infty}^{\infty}x(n)Z^{-n}$$

The given sequence is a finite sequence defined in the range n=0 to 6, hence the limits of summation is changed to n=0 to 6.

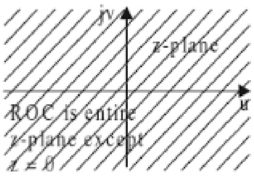
$$Z\{x(n)\}=X(Z)=\sum_{n=0}^{6}x(n)Z^{-n}$$

$= x(0)\, z^{0} + x(1)\, z^{-1}+ x(2)\, z^{-2}+ x(3)\, z^{-3}+ x(4)\, z^{-4}+ x(5)\, z^{-5}+ x(6)\, z^{-6}$

$= 3+2\, z^{-1}+2\, z^{-2} +3\, z^{-3} +5\, z^{-4} +0\, z^{-5}+1\, z^{-6}$

$= 3 + 2/Z+2/Z^{2} + 3/Z^{3}+5/Z^{4}+ 1/Z^{6}$

In X(z) , when z=0, except the first terms all other terms will become infinite. Hence X(z) will be finite for all values of z, except z=0. Therefore , the ROC is entire z-plane except z=0.

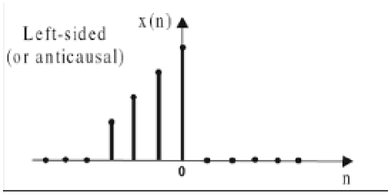


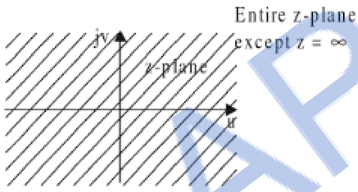
(ii)x(n)= Ƌ(n-k)

The above signal is a left sided or anticausalsequence and its Z- transform can be obtained with:

$$Z\{x(n)\} = X(Z) = \sum_{n=-k}^{0} x(n)Z^{-n}$$

The ROC of this signal is :





22.What is the relationship between s-plane and z-plane?

The s-plane and the z-plane are related by a conformal mapping specified by the analytic complex function $z = e^s = e^{\sigma+jw} = e^{\sigma} . e^{jw} = r\, e^{jw}$. Where, Re{s}= σ and Im{s}= jw and |z|= r = e^{σ} and <z= w.

23. Find the system transfer function H(Z)if y(n)= x(n)+y(n-1)

$Y(Z)=X(Z)+ Z^{-1}Y(Z)$
$Y(Z)- Z^{-1}Y(Z) = X(Z)$
$Y(Z)[1- Z^{-1}]=X(Z)$
$Y(Z)/X(Z)=1/[1- Z^{-1}]$
$H(Z)= 1/[1- Z^{-1}]$

24. List the methods to find inverse z transform.

i) Long division method
ii) Partial fraction method
iii) Residue method
iv) Convolution method

25. Write the conditions to define stability in ROC.

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

An LTI system is stable if and only if its impulse response is absolutely summable, i.e., the frequency response function $H(e^{j\omega})$ exits, i.e. the ROC of its transfer function $H(z)$ includes the unit circle $|z|=1$ z=1

PART-B

1.Find the Z- Transform of following

$(a)x(n) = \sin(\omega n)\, u(n)$

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

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

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

3. i) Determine the DTFT of the given sequence $x(n) = a^n(u(n) - u(n-8)), |a| < 1$

   ii) Prove the linearity and frequency shifting theorem of the DTFT.

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

5. Evaluate the frequency response of the system described by the system function H(Z)= $\frac{1}{1-0.5z^{-1}}$.

6. Find the Z- transform and its associated ROC for the following discrete time signal x[n]= $(\frac{1}{3})^n$ u[n]+5 $(\frac{1}{2})^{-n}$ u[-n-1]

7. i) Find the Z-transform and ROC of $x(n) = r^n \cos(n\theta)u(n)$.

   ii) Find the inverse Z-transform of $X(Z) = \frac{z}{3z^2 - 4z + 1}$, ROC $|z| > 1$

8. i) Find the Z- Transform of following x(n)=n^2u(n)

   ii) Obtain the linear convolution of $x_1[n] = [0,1,4,-2] \& x_2[n] = [1,2,2,2]$

   iii) Find the frequency response of LTI system governed by the equation

   $y(n) = a_1 y(n-1) - a_2 y(n-2) - x(n)$

9. (i) Find the Z transform and its ROC of x(n)=a^nu(n) $x(n) = \left(\frac{1}{2}\right)^{|n|} + \left(-\frac{1}{2}\right)^{|n|}$

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

10.(i) What is frequency response? Explain its properties.

   (ii) Find the inverse z –transform of X(Z)= $\frac{4Z}{(Z+1)^2(Z+3)}$ for all possible ROCs.

11. Find the inverse Z transform of $X(z) = \frac{z}{3z^2 - 4z + 1}$ ROC |Z| >1

12.(i) Find $x(n) if X(z)=\dfrac{1+\frac{1}{2}Z^{-1}}{1-\frac{1}{2}Z^{-2}}$

(ii) Find the response of the causal system y(n) – y(n-1) =x(n) + x(n-1) to the input x(n) = u(n) Test its stability

13. What is the need for frequency response analysis? Find the impulse response , frequency response ,magnitude response and phase response of the second order system
y(n) = 2x(n)+x(n-1)+y(n-2)

14.Find the circular convolution of the two sequences:
X(n)={1,3,5,7} h(n)={2,4,6,8}

UNIT-III DISCRETE FOURIER TRANSFORMATION & COMPUTATION

PART-A

1. Define the Discrete Fourier transformation of a given sequence x(n).
The N- point DFT of a sequence x(n) is

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

k= 0, 1, 2……..N-1.

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

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

n= 0, 1 , 2 …..N-1 .

3. List any four properties of DFT.

(a) Periodicity

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

$x(n+N)=x(n) \quad for\ all \quad n$

$X(k+N)=X(k)\ for\ all \quad k.$

(b)Linearity

If $X_1(k)$=DFT[$x_1(n)$] and
$X_2(k)$=DFT[$x_2(n)$]

then 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),
ThenDFT{$x((-n))_N$}=DFT{x(N-n)}=$X((-k))_N$=X(N-k)

(d)Circular time shifting of a sequence

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

4. 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:

$$(i)DFT\{x^*(n)\} = X^*(N-k)$$
$$(ii)DFT\{x^*(N-n)\} = X^*(k)$$
$$(iii)DFT\{x((n-l))_N\} = X(k)e^{-j2\pi kl/N}$$
$$(iv)DFT\{x(n)e^{j2\pi ln/N}\} = X((k-l))_N$$

5. Calculate the DFT of the sequence x(n) = $\left(\frac{1}{4}\right)^n$ $\quad for N = 16$

Solution:

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

$$= \sum_{n=0}^{15} \left(\frac{1}{4}\right)^n e^{-j2\pi kn/16}$$

$$= \sum_{n=0}^{15} \left(\frac{1}{4}e^{-j\pi k/8}\right)^n$$

$$= \frac{1-\left(\frac{1}{4}\right)^{16} e^{-j2\pi k}}{1-\frac{1}{4}e^{-j\pi k/8}}$$

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

Solution:

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

$$= \sum_{n=0}^{3} x(n)e^{-j2\pi kn/4} \qquad k = 0,1,2,3.$$

$$X(0) = \sum_{n=0}^{3} x(n) = \{1+1+0+0\} = 2$$

$$X(1) = \sum_{n=0}^{3} x(n)e^{-j\pi n/2} = \{1-j+0+0\} = 1-j$$

$$X(2) = \sum_{n=0}^{3} x(n)e^{-j\pi n} = \{1-1+0+0\} = 0$$

$$X(3) = \sum_{n=0}^{3} x(n)e^{-j3\pi n/2} = \{1+j+0+0\} = 1+j$$

$$X(k) = \{2, 1-j, 0, 1+j\}$$

7. State Circular frequency shifting property of DFT.

If DFT[x(n)]=X(k),

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

Thus shifting the frequency components of DFT circularly is equivalent to multiplying the time domain sequence by $e^{j2\pi ln/N}$

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

The process of lengthening the sequence by adding zero – valued samples is called appending with zeros or zero – padding.

Uses:

1. We can get "better display" of the frequency spectrum.
2. With zero padding, the DFT can be used in linear filtering.

9. What are the steps involved in circular convolution.

The circular convolution involve basically four steps as the ordinary linear convolution. These are 1.Folding the sequence

2. Circular time shifting the folded sequence

3.Multiplying the two sequences to obtain the product sequence. 4. Summing the values of product sequence.

10. 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.

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

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

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

11. State the difference between (i) overlap-add method (ii) overlap – save method.

Sl.No	Overlap – add method	Overlap – save method
1.	In this method the size of the input data block is N=L+M-1	In this method the size of the input data block is L.
2.	Each data block consists of the last M-1 data points of the previous data followed by the L new data points.	Each data block is L points and we appended M-1 zeros to compute N-point DFT.
3.	In each output block M-1 points are corrupted due to aliasing , as circular convolution is employed.	In this no corruption due to aliasing as linear convolution is performed using circular convolution.
4.	To form the output sequence the first M-1 data points are discarded in each output block and the remaining datas are fitted together.	To form the output sequence, the last M-1 points from each output block is added to the first (M-1) points of the succeeding block.

12. 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.

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

The number of multiplications and additions required to compute N-point DFT using radix-2 FFT are $N\log_2 N \quad and \quad \frac{N}{2}\log_2 N$ respectively.

14. What is the speed improvement factor in calculating 64 – point DFT of a sequence using direct 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

$$N^2 = 64^2 = 4096$$

The number of complex multiplications required using FFT is

$$\frac{N}{2}\log_2 N = \frac{64}{2}\log_2 64 = 192$$

Speed improvement factor= $\frac{4096}{192} = 21.33$

15. 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-popint 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.

16.What are the difference between and similarities between DIT and DIF algorithms?

Difference between DIT and DIF:

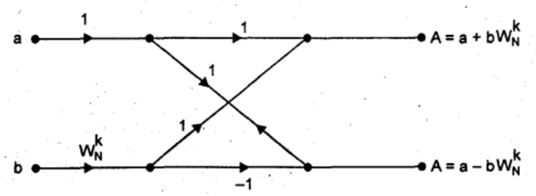
1. In DIT , the input is bit-reversed while the output is in natural order. For DIF , the reverse is true, i.e., input is normal order, while the output bit is reversed. However , both DIT and DIF can go from normal to shuffled data or vice versa.

2.Considering the butterfly diagram, in DIF , the complex multiplication takes place after the add – subtract operation.

Similarities:

1. Both algorithms require same number of operations to compute DFT.
2. Both algorithm require bit – reversal at some place during computation.

17. Draw the basic butterfly diagram for Radix 2 DITFFT.



18. In eight point decimation in time(DIT), what is the gain of the signal path that goes from x(7) to X(2)?
From the signal flow diagram of Radix -2 eight point DIT FFT, the signal path from x(7) to X(2) have gain as follows,

Gain from x(7) to X(2) = $-W_8^0 W_8^0 W_8^2 = -j$

19. Define circular convolution.
The convolution property of DFT says that , the multiplication of the DFTs of the two sequence is equivalent to the DFT of the circular convolution of the two sequences.

$$X_1(k)X_2(k) = DFT\{x_1(n) \quad x_2(n)\}$$

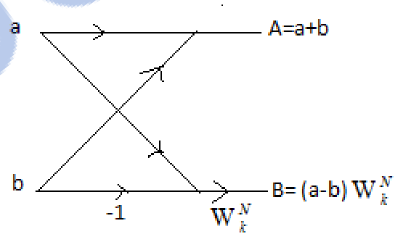
$$x_3(n) = \sum_{m=0}^{N-1} x_1(m)x_2((n-m))_N$$

20. Find the discrete Fourier Transform for δ[n].

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

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

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



22. What is the relation between Z-transform & DFT?
Let N-point DFT of x(n) be X(k) and the Z-transform of x(n) be X(z).The N-point sequence X(k) can be obtained from X(z) by evaluating X(z) at N equally spaced points around the unit circle.

$$i.e., X(k) = X(z)\Big|_{z=e^{j\frac{2\pi k}{N}}} ; for \ k = 0,1,2,......(N-1)$$

23. What are the steps involved in computing IDFT through FFT?

1. Take conjugate of x(k)
2. Compute N point DFT of complex conjugate x*(k) using FFT.

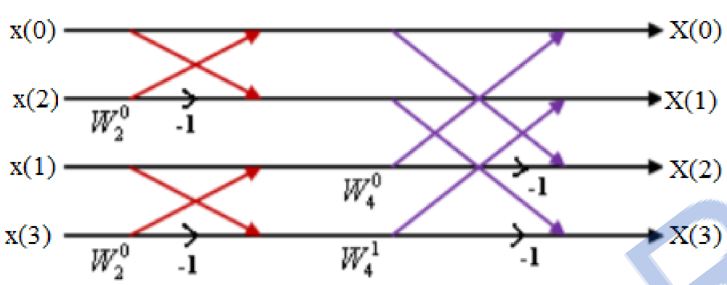
3. Take again the conjugate of the output sequence.
4. Then the resultant sequence is divided by N.

24. State parsavel's relation of DFT?

If x[k]x[k] and X[r]X[r] are the pair of discrete time Fourier sequences, where x[k]x[k] is the discrete time sequence and X[r]X[r] is its corresponding DFT. Prove that the energy of the aperiodic sequence x[k]x[k]of length NN can be expressed in terms of its N-point DFT as follows:

E_x=∑k=0N−1|x[k]|2=1N∑rN−1|X[r]|2.

25.Draw the flow graph of a 4 point radix-2 DIT FFT butterfly structure for DFT



26.What are the applications of FFT algorithm?

Some of the important applications of FFT includes

- Fast large integer and polynomial multiplication
- Efficient matrix-vector multiplication for Toeplitz, circulant and other structured matrices
- Filtering algorithms
- Fast algorithms for discrete cosine or sine transforms (example, Fast DCT used for JPEG, MP3/MPEG encoding)
- Fast Chebyshev approximation
- Fast discrete Hartley transform

PART-B

1. State and prove any four properties of DFT.
2.(i)Develop a Radix-2, 8-point DIF FFT algorithm with neat flow chart
(ii) Determine the DFT of the sequence $x(n)=\{1/4, \text{for } 0 \le n \le 2$.
0, otherwise
3. A finite duration sequence of length L is given as

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

Determine N−point DFT of this sequence for N ≥ L

4. By means of the DFT & IDFT, determine the sequence x_3 (n) corresponding to the circular convolution of the sequence x_1 (n) and x_2(n).

$x_1(n) = \{2,1,2,1\}$, $x_2(n) = \{0,1,2,3\}$

5.Find the DFT of the sequence x(n) = {1,2,3,4,4,3,2,1} using DIF-FFT algorithm.

6.Develop a Radix-2, 8-point DIT- FFT algorithm with neat flow chart

7. An 8-point sequence is given by x(n); x(n) = (1, 1, 1, 1, 1, 1, 1, 1).Compute 8 point DFT of x(n) by radix -2 DIF-FFT

8. i) Find the X(K) for $x(n)=n+1, for\ N=8$ using DIF FFT algorithm.

ii) Use four point inverse FFT for the DFT result $\{6,-2+j2,-2,-2-j2\}$ and determine the input sequence.

9.Obtain 8-point DFT of the sequence x(n) = (1, 1, 1, 1).

10.(i)The first five points of the eight point DFT of a real valued sequence are{0.25,0,0.125-j0.3018,0,0.125-j0.0518}.Determine the remaining three points.
(ii)Compute the eight point DFT of the sequence x= {0,1,2,3,4,5,6,7} using DIF FFT algorithm.

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

(ii)Using FFT algorithm compute the DFT of x(n)={2,2,2,2}

12.(i)Summarize the steps of radix -2 DIT FFT algorithm.

(ii)Compute the 4 point DFT of the sequence x(n)={0,1,2,3} using DIT and DIF algorithm.

13.Find the IDFT of the sequence
X(K)={4,1-j2.414,0,1-j0.414,0,1+j0.414,0,1+j2.414} using DIF algorithm.

14.Describe the need for bit reversal and butterfly structure. For a sequence x(n)={4,3,2,1,-1,2,3,4} obtain 8-point FFT by using DIT method.

15.Compute 8 point DFT of the sequence {1,1,1,1,1,1,0,0}.

16. Compute 8 point DFT of the given sequence using DIT algorithm.

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

17. The analog signal has a bandwidth of 4 KHz. If we use N point DFT with N=2m (m is an integer) to compute the spectrum of the signal with resolution less than or equal to 25 Hz.

Determine the minimum sampling rate, minimum number of required samples and minimum length of the analog signal. What is the step size required for quantize the signal

18. Determine the DFT of the following sequence x(n) = {5,-1,1,-1,2}

19. Find the DFT of the sequence x(n) = {1,2,3,4,4,3,2,1} using DIT-FFT algorithm.

20. Determine the DFT of the given sequence x(n) = {1,-1,-1,-1,1,1,1,-1} using DIT-FFT algorithm.

21. Determine the DFT of the given sequence x(n) = {1,1,1,1,1,1,1,0} using DIT-FFT algorithm.

UNIT IV DESIGN OF DIGITAL FILTERS

PART-A

1. What are the different types of structures for realization of IIR systems?

The different types of structures for realization of IIR system are
(i) Direct form I structure (ii) Direct form II structure
(iii) Cascade form structure (iv) Parallel form structure
(v) Lattice – ladder form structure.

2. What are the different types of filters based on impulse response?

Based on impulse response the filters are of two types
1. IIR filter
2. FIR filter

The IIR filters are of recursive type, whereby the present output sample depends on the present input, past inputs samples and output samples.

The FIR filters are of non recursive type whereby the present output sample is depends on the present input sample and previous input samples.

3. What is the general form of IIR filter?

The most general form of IIR filter can be written as

$$H(z) = \frac{\sum_{k=0}^{M} b_k z^{-k}}{1 + \sum_{k=1}^{N} a_k}$$

4. Give the magnitude of Butterworth filter. What is the effect of varying order of N on magnitude and phase response?

The magnitude function of the Butterworth filter is given by

$$|H(j\Omega)| = \frac{1}{\left[1+\left(\frac{\Omega}{\Omega_c}\right)^{2N}\right]^{\frac{1}{2}}} \quad N = 1,2,3.............$$

Where N is the order of the filter and Ω_c is the cut off frequency. The magnitude response of the Butterworth filter closely approximates the ideal response as the order N increases . The phase response becomes more non-linear as N increases.

5. What is Type –1 and Type –2 Chebyshev approximation?

(i)In type –1 Chebyshev approximation, the error function is selected such that, the magnitude response is equiripple in the pass band and monotonic in the stop band.

(ii)In type –2 Chebyshev approximation, the error function is selected such that, the magnitude response is monotonic in pass band and equiripple in the stop band. The Type -2 magnitude response is called inverse Chebyshev response.

6. Write the magnitude function of Chebyshevlowpass filter?

The magnitude response of Type -1 lowpassChebyshev filter is given by

$$|H_a(\Omega)| = \frac{1}{\sqrt{1+\varepsilon^2 C_N^2\left(\frac{\Omega}{\Omega_c}\right)}}$$

where ε is attenuation constant and

$C_N\left(\frac{\Omega}{\Omega_c}\right)$ is the Chebyshev polynomial of the first kind of degree N.

7. 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.

1. The magnitude response approaches the ideal response as the value of N increases.
2. The Chebyshev type – 1 filters are all pole designs.
3. The poles of Chebyshev filter lies on an ellipse.

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

Sl.No	Butterworth filter	Chebyshev filter
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 of Ω.	The magnitude response is equiripple in pass band and monotonically decreasing in the stop band.
4.	The normalized magnitude response has a value of $\frac{1}{\sqrt{2}}$ at the cut off frequency Ω_c.	The normalized magnitude response has a value of $\frac{1}{\sqrt{1+\varepsilon^2}}$ at the cut off frequency Ω_c.
5.	Only few parameters has to be	A large number of parameter has to be

	calculated to determine the transfer function.	calculated to determine the transfer function.

9. 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.

10. What are the properties of FIR filter?(or) What are the advantages of FIR filter.

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

1. A realizable filter can always be obtained.
2. FIR filter has a linear phase response.

11. What are the steps involved in the FIR filter design?

1. Choose the desired (ideal) frequency response $H_d(\omega)$.
2. Take inverse Fourier transform of $H_d(\omega)$ to get $h_d(n)$.
3. Convert the infinite duration $h_d(n)$ to a finite duration sequence $h(n)$.
4. Take Z – transform of $h(n)$ to get the transfer function $H(z)$ of the FIR filter.

12. How the constant group delay and phase delay is achieved in linear phase FIR filters.

Frequency response of FIR filter with constant group and phase delay

$$H(\omega) = \pm|H(\omega)|e^{j(\beta-\alpha\omega)}$$

The following conditions have to be satisfied to achieve constant group and phase delay.

Phase delay, $\alpha = \dfrac{N-1}{2}$ (i.e., phase delay is constant)

Group delay, $\beta = \pm\dfrac{\pi}{2}$ (i.e., group delay is constant)

Impulse response, h(n) = - h(N -1 – n) (i.e., impulse response is anti symmetric)

13. Write the condition for stability of digital filter.

1. Choose the desired (ideal) frequency response $H_d(\omega)$ of the filter.

2. Evaluate the Fourier series co-efficient of $H_d(\omega)$ which gives the desired impulse response $h_d(n)$. Where $h_d(n) = \dfrac{1}{2\pi}\displaystyle\int_{-\pi}^{\pi} H_d(\omega)e^{j\omega n}d\omega$

Truncate the infinite sequence $h_d(n)$ to a finite duration sequence $h(n)$.
3.Take Z – transform of $h(n)$ to get a non causal filter transfer function $H(z)$ of the FIR filter.

4.Multiply $H(z)$ by $z^{-\left(\frac{N-1}{2}\right)}$ to convert noncausal transfer function to a realizable causal FIR filter transfer function.

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

14. 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.

15. 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.

16.Give the equation specifying Hamming windows.

The equation for Hamming window is given by

$$w_H(n) = 0.54 + 0.46\cos\frac{2\pi n}{N-1} \quad \text{for} \quad -\frac{(N-1)}{2} \le n \le \frac{(N-1)}{2}$$

$$= 0 \qquad \text{Otherwise.}$$

17. Compare the Rectangular and Hanning window.

Sl.No	Rectangular window	Hanning window
1.	The width of mainlobe in window spectrum is $\frac{4\pi}{N}$	The width of mainlobe in window spectrum is $\frac{8\pi}{N}$
2.	The maximum sidelobe magnitude in window spectrum is -13dB.	The maximum sidelobe magnitude in window spectrum is -31dB.
3.	In window spectrum the sidelobe magnitude slightly decreases with increasing ω	In window spectrum the sidelobe magnitude decreases with increasing ω
4.	In FIR filter designed using rectangular window the minimum stopband attenuation is 22dB.	In FIR filter designed using Hanning window the minimum stopband attenuation is 44dB.

18. Compare the Hamming and Blackman window.

Sl.No	Hamming window	Blackman window
1.	The width of mainlobe in window spectrum is $\frac{8\pi}{N}$	The width of mainlobe in window spectrum is $\frac{12\pi}{N}$
2.	The maximum sidelobe magnitude in window spectrum is -41dB.	The maximum sidelobe magnitude in window spectrum is -58dB.
3.	In window spectrum the sidelobe magnitude remains constant with increasing ω	In window spectrum the sidelobe magnitude decreases rapidly with increasing ω
4.	In FIR filter designed using Hamming window the minimum stopband attenuation is 51dB.	In FIR filter designed using Blackman window the minimum stopband attenuation is 78dB.
5.	The higher value of sidelobe attenuation is achieved at the expense of constant attenuation at high frequencies.	The higher value of sidelobe attenuation is achieved at the expense of increasedmainlobe width.

19 .Compare the Hamming and Kaiser window.

Sl.No	Hamming window	Kaiser window
1.	The width of mainlobe in window spectrum is $\frac{8\pi}{N}$	The width of mainlobe in window spectrum depends on the values of 'a' and N. Here 'a' is a variable parameter introduced to modify the characteristics of window and N is the order of the filter.
2.	The maximum sidelobe magnitude in window spectrum is -41dB.	The maximum sidelobe magnitude with respect to peak of mainlobe is variable using the parameter 'a'.
3.	In window spectrum the sidelobe magnitude remains constant with increasing ω	In window spectrum the sidelobe magnitude decreases with increasing ω
4.	In FIR filter designed using Hamming window the minimum stopband attenuation is 51dB.	In FIR filter designed using Kaiser window the minimum stopband attenuation is variable and depends on the value of 'a'.

20.Mention any two procedures for digitizing the transfer function of an analog filte

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

1. Impulse invariance method.
2. Bilinear transformation method.

21. 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.

22. 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 prewarpfrequencies, the analog filter transfer function is designed and then it is transformed to digital filter transfer function.

23. Compare the impulse invariant and bilinear transformations.

Sl.No	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 linear.	The relation between analog and digital frequency is nonlinear.
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.

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

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

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

25.Comment on the pass band and stop band characteristics of butter worth filter.

(i)The frequency response of the Butterworth filter is maximally flat (i.e. has no ripples) in the passband and rolls off towards zero in the stopband.

(ii)When viewed on a logarithmic Bode plot, the response slopes off linearly towards negative infinity.

(iii)Butterworth filters have a monotonically changing magnitude function with ω, unlike other filter types that have non-monotonic ripple in the passband and/or the stopband.

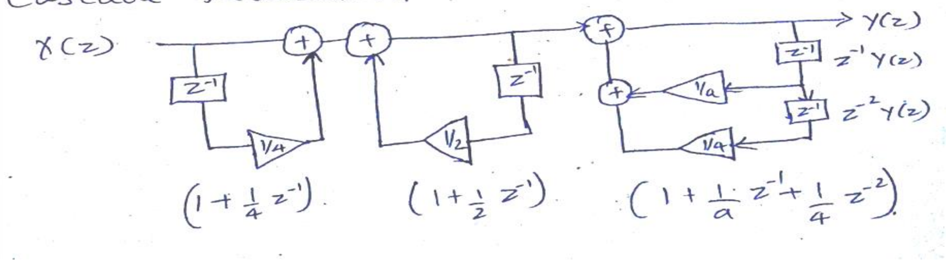
26.Define pre-wraping effect? Why it is employed?

The effect of the non linear compression at high frequencies can be compensated. When the desired magnitude response is piecewise constant over frequency, this compression can be compensated by introducing a suitable rescaling or prewar ping the critical frequencies.

27.Obtain the cascade realization for the system function.

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

Cascade Realisation.



28. Mention the advantages of FIR filters over IIR filters.
FIR filters are more powerful than IIR filters, but also require more processing power and more work to set up the filters. They are also less easy to change "on the fly" as you can by tweaking (say) the frequency setting of a parametric (IIR) filter. However, their greater power means more flexibility and ability to finely adjust the response of your active loudspeaker.

29.Why are digital filters more useful than analog filters?
Digital filters have the following advantages compared to analog filters:

- Digital filters are software programmable, which makes them easy to build and test.
- Digital filters require only the arithmetic operations of addition, subtraction, and multiplication.
- Digital filters do not drift with temperature or humidity or require precision components.
- Digital filters have a superior performance-to-cost ratio.
- Digital filters do not suffer from manufacturing variations or aging.

30. Write the advantages and disadvantages of digital filters.
Advantages:

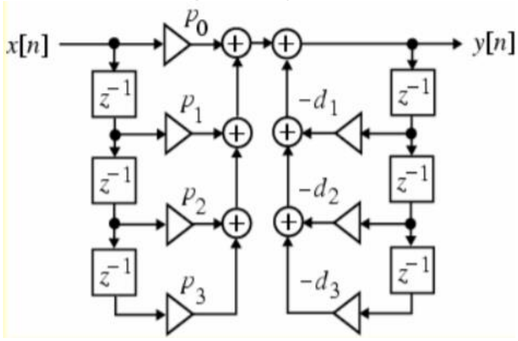
- Digital filters are software programmable, which makes them easy to build and test.
- Digital filters require only the arithmetic operations of addition, subtraction, and multiplication.
- Digital filters do not drift with temperature or humidity or require precision components.
- Digital filters have a superior performance-to-cost ratio.
- Digital filters do not suffer from manufacturing variations or aging.

Disadvantages

- In reality, the signal bandwidth of the digital sequence is much lower than the analog sequence.
- Finite word-length effect, which results quantizing noise and round-off noise, is another major drawback during computation.
- It needs much longer time to design and develop the digital sequence.

31. Draw the direct form I structure for 3rd order system.
Consider a 3rd order system transfer function,

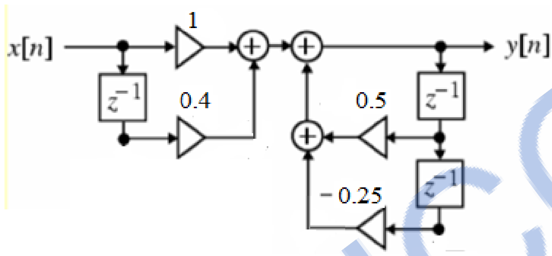
The direct form I structure of IIR filter is given by,



32. What is meant by radix-4 FFT?

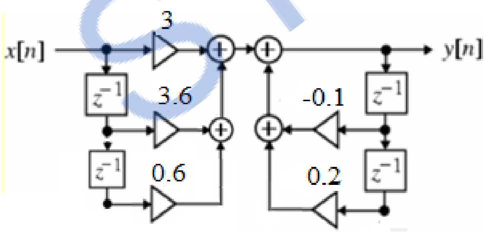
For a radix-4 FFT, the value of sequence N should be such that, $N=2^m$, so that the N-point sequence is decimated into 4-point sequences and 4-point DFT of each decimated sequence is computed. Here 'm' is the number of stage.

33. Obtain the direct form-I realization for the given difference equation y(n) = 0.5y(n-1)-0.25y(n-2)+x(n)+0.4x(n-1)



34. Obtain the direct form-I realization for the given difference equation

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



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

The convolution spreads the true frequency to frequency bins around it. By using a window function other than the rectangular one there is less of the sidelobes compared to the main lobe, so less artificial long-distance spread making the result cleaner and better suited for frequency-selective analysis.

36. The most straightforward approach to FIR filter design is to truncate the impulse response of an ideal IIR filter. Why this is usually an undesirable approach.

The undesirable features of FIR filters are

- There are large storage requirements
- Powerful computational facilities needed for the implementation

37. Obtain the transfer function for a normalized Butterworth filter of order 2.

The transfer function of normalized Butterworth filter is given by

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

PART-B

1. Obtain the Direct form –I, Direct form – II, Cascade form and Parallel form structure for the system described by

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

$$(b)\ y(n) = 0.5y(n-1) + 0.25y(n-2) + x(n) + x(n-1)$$

(c) y(n) - 2y(n-1) + y(n+2) = x(n) +0.5 x(n-1)

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

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

$|H(e^{j\omega})| \le 1 \qquad 0.6\pi \le \omega \le \pi$

3. Design (a) a Butterworth and (b) a Chebyshev analog high pass filter that will pass all signals of radian frequencies greater than200rad/sec with no more than 2 dB attenuation and have a stop band attenuation of greater than 20 dB for all Ω less than 100rad/sec.

4. Design an FIR linear phase, digital filter approximating the ideal frequency response

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

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

5. A low pass filter is to be designed with the following desired frequency response.

$$H_d(e^{jw}) = \begin{cases} e^{-j2w}, & for\ |\omega| \le \frac{\pi}{4} \\ 0, & \frac{\pi}{4} \le |\omega| \le \pi \end{cases}$$

Determine the filter coefficients $h_d(n)$ if the window function is defined as
$$w(n) = \begin{cases} 1, & for\ 0 \le n \le 4 \\ 0, & otherwise \end{cases}$$

6.Convert the analog filter with system function $H_{a(S)} = \frac{1}{(s+0.2)^2+2}$ into digital IIR filter by means of Impulse invariant transformation and Bilinear Transformation method.

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

8. Determine H(Z) for a butterworth filter Satisfying following constraints
$$\sqrt{0.5} \le |H(j\Omega)| \le 1 \quad for \quad 0 \le w \le \frac{\pi}{2}$$
$$|H(j\Omega)| \le 0.2 \quad for \quad \frac{3\pi}{4} \le w \le \pi$$
With T=1 s .Apply Impulse invariant transformation.

9. (i) Convert the following analog transfer function into digital using impulse invariant technique with sampling period T=1sec. $H(s) = \frac{s+1}{(s+3)(s+5)}$

10.Design a butterworth filter using the impulse invariance method for the following specifications
$$0.8 \le |H(e^{jw})| \le 1 \qquad 0 \le w \le 0.2\pi$$
$$|H(e^{jw})| \le 0.2 \qquad 0.6\pi \le w \le \pi$$
Realize the designed filter using direct form II structure.

11.Design a filter with desired frequency response
$$Hd(e^{jw}) = e^{-j3w} \quad for \frac{-3\pi}{4} \le w \le \frac{3\pi}{4}$$
$$= 0 \quad for \frac{3\pi}{4} \le |w| \le \pi$$
Using a hanning window for N=7

12.Design an ideal low pass filter with a frequency response
$$for \frac{-\pi}{2} \le w \le \frac{\pi}{2}$$
$H_d(e^{jw}) = 1$

=0 $for \frac{\pi}{2} \leq |w| \leq \pi$

Find the values of h(n) for N=11. Find H(z) and the filter coefficients**. Ass**ume rectangular window

13. (i) Given the specifications α_p=3dB.α_s=10dB.f_p=1 kHz and f_s=2 kHz. Determine the order of the filter using chebyshev approximation Find H(s

(ii) Apply bilinear transformation to

$H(s) = \frac{2}{(s+1)(s+2))}$ with T=1sec and find H(z)

14.(i) Explain the role of windowing to realize a FIR filter

(ii) Compare and explain on the choice and type of windows selection for signal analysis.

(iii)Compute numerically the effect of Hamming windows and design the filter if cut –off frequency=100Hz, Sampling frequency=1000Hz, Order of the filter =2, Filter length required=5.

15. Using bilinear transformation, design a high pass filter, monotonic in passband with cutoff frequency of 1000 Hz and down 10 dB at 350Hz. The sampling frequency is 5000Hz.

16. Convert the signal pole low pass filter with system function $H(z) = \frac{0.5(1+z^{-1})}{(1-0.302z^{-3})}$ into band pass filter with upper and lower cutoff frequencies ωu and ωl respectively. The lowpass filter has 3dB bandwidth and ωp =π/6 and ωu=3π/4 ,ωl=π/4 and draw its realization in direct form II.

17. Obtain the analog transfer Chebyshev filter transfer fuction that satisfies the given constraints

$$\frac{1}{\sqrt{2}} \leq |H(j\Omega)| \leq 1 \quad for \quad 0 \leq \Omega \leq 2$$

$$|H(j\Omega)| \leq 0.1 \quad for \quad \Omega \geq 4$$

18. Design a filter using hamming window with the specifications N=7 of the system

$Hd(e^{jw}) = e^{-j3w}$ $for \frac{-\pi}{4} \leq w \leq \frac{\pi}{4}$

=0 $for \frac{-\pi}{4} \leq |w| \leq \pi$

19. Determine an ideal high pass filter using hanning window with the specification N=11 of the system

$Hd(e^{jw}) = 1$ $for \frac{\pi}{4} \leq |w| \leq \pi$

=0 $for |w| \leq \frac{\pi}{4}$

20. Design a ideal bandpass filter with a frequency response.

$$H_d(\omega)=\begin{cases}1, & for \ \frac{\pi}{4}|\omega| \le \frac{3\pi}{4} \\ 0, & otherwise\end{cases}$$

Find the value of h(n) for N=11 and plot the frequency response.

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

Pass band edge magnitude = √0.5
Pass band frequency = π/2
Stop band magnitude = 0.2
Stop band frequency = 3π/4

22.The desired frequency response of a Low pass filter is

$$H_d(e^{jw}) = e^{-j3w} \quad for \ \frac{-3\pi}{4} \le w \le \frac{3\pi}{4}$$

$$= \mathbf{0} \quad for \ \frac{3\pi}{4} \le |w| \le \pi$$

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

UNIT V DIGITAL SIGNAL PROCESSOR

PART-A

1. What are the classification digital signal processors?

The digital signal processors are classified as

- General purpose digital signal processors.
- Special purpose digital signal processors.
-

2. Give some examples for fixed point and floating point DSPs.

Fixed point DSPs are
TMS320C50, TM 320C54, TM 320C55, ADSP-219x, ADSP-219xx.
Floating point DSPs are
TMS320C3x, TMS320C67x, ADSP-21xxx.

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

The main factors that influence selection of DSPs are,
1. Architectural features 2.Execution speed 3.Type of arithmetic 4.Word length

4. What are the applications of PDSPs?

Digital cell phones, automated inspection, voicemail, motor control, video conferencing, Noise cancellation, Medical imaging, speech synthesis, satellite communication etc.

5. What is pipelining?

Pipelining a processor means breaking down its instruction into a series of discrete pipeline stages which can be completed in sequence by specialized hardware.

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

TMS320C50 – 4 TM 320C54x – 6

7.What are the different stages in pipelining?

i. The Fetch phase ii. The Decode phase iii. Memory read phase iv. The Execute phase

8.What are the different buses of TM 320C5x and their functions?
The 'C5x architecture has four buses
1. Program bus (PB) 2.Program address bus (PAB) 3.Data read bus (DB) 4.Data read address bus (DAB)

- The program bus carriers the instruction code and immediate operands from program memory to the CPU.
- The program address bus provides address to program memory space for both read and write.
- The data read bus interconnects various elements of the CPU to data memory spaces.
- The data read address bus provides the address to access the data memory spaces.

9. List the various registers used with ARAU of DSP processor?
i) Eight auxiliary registers (AR0 –AR7). They are used for indirect addressing. ii) Index (INDX) and auxiliary register compare register (ARCR) are used to calculate indirect address.

10. What are the elements that the control processing unit of 'c5X consists of
1. Central arithmetic logic unit (CALU).2 Parallel logic units (PLU)
3. Auxiliary register arithmetic unit (ARAU) 4. Memory mapped registers
5. Program controller

11. What is the function of parallel logic unit?
The parallel logic unit is second logic units, which execute logic operations on data without affecting the contents of accumulator.

12. List the on chip peripherals in 'c5x.
The on-chip peripherals interfaces connected to the 'c5x CPU include
1. Clock generator 2. Hardware timer 3. Software programmable wait state generators, 4.General purpose I/O pins 5. Parallel I/O ports, 6. Serial port interface, 7. Buffered serial port
8. Time-divisions multiplexed (TDM) serial port, 9. Host port interface10. User unmask able interrupts

13.Give the special features of DSP processors.
1. Harmonics can be analyzed using Fourier analysis.
2. Generation of pulses
3. Discretizing the waveform

14. What is the function of parallel logic unit DSP processor?
The parallel logic unit is a second logic unit, that execute logic operations on data Without affecting contents of accumulator.

15. Define Period gram.
Periodic analysis of the waveform can be analyzed.

16. What is meant by bit reversed addressing mode? What is the application for which this addressing mode is preferred?

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

17. Compare the RISC and CISC processors.

CISC	RISC
Emphasis on hardware	Emphasis on software
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.e.* as CISC computer), it means a computer that has a **C**omplex **I**nstruction **S**et **C**hip as its cpu.	If it reads as above (*i.e.* as RISC computer), it means a computer that has a **R**educed **I**nstruction **S**et **C**hip as its cpu

18. Mention one important feature of Harvard architecture.

(i) Harvard architecture has separate memories for program and data. It also has separate address and data buses for program and data. Because of these separate on chip memories and internal buses, the speed of execution in Harvard architecture is high.

19. What is the advantage of Pipelining?

It provides sequential flow of execution with one after the other process without any interruption. This concept of pipelining increases computational efficiency of the processor.

20.How many buses does C54X processor have & what are they?

i) There are eight major 16 bit buses (four program/data bus and four address buses).
ii) Program bus (PB) carries instruction code and immediate operands from program memory.
iii) Three address buses (CB, DB and EB) interconnect CPU, data address generation logic, program address generation logic, on chip peripherals and data memory.
iv) Four address buses (PAB, CAB, DAB and EAB) carry the addresses needed for instruction execution.

21.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

22.What is IMR and IFR?

IMR (Interrupt mask register) individually masks off specific interrupts at required times.
IFR (Interrupt flag register) indicates the current status of the interrupts.

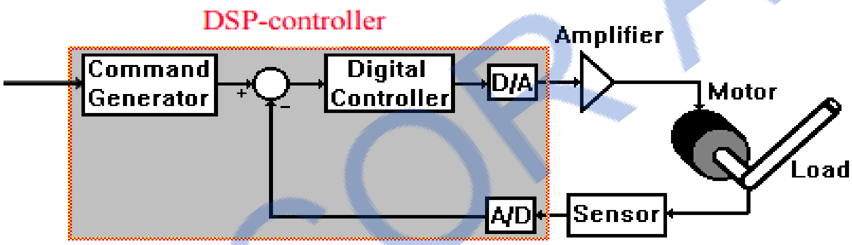
23.State the intended applications of DaVinci Digital Media processors.

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

24.What is the advantage of Harvard architecture in a DS processor?

In Harvard architecture, memory of data and memory of instruction are separated. Its advantages includes, faster execution time and it allows concurrent access of data and instruction.

25.How is a DS processor applicable for motor control applications.



26. What are the merits and demerits of VLIW architecture?

Advantages:Increased performance , Better compiler targets, Potentially easier to program ,
Disadvantages:Increased memory use, High program memory bandwidth requirements ,High power consumption,Misleading MIPS ratings

27. What are the factors that influence the selection of DSP processor for an application?

The right DSP processor for a job depends heavily on the application. One processor may perform well for some applications, but be a poor choice for others. With this in mind, one can consider a number of features that vary from one DSP to another in selecting a processor.
These features are: Ease of Development,Multiprocessor Support,Power Consumption and Management, Cost,Memory Organization.

28.State how spectrum meter application can be designed with DS processor

The FFT or Fast Fourier Transform spectrum analyser uses digital signal processing techniques to analyse a waveform with Fourier transforms to provide in depth analysis of signal waveform spectra.With the FFT analyse, its able to provide facilities that cannot be provided by swept frequency analyzers, enabling fast capture and forms of analysis that are not possible with sweep / superheterodyne techniques alone.

29. What is pipelining and how do define its depth?

Pipelining a processor means breaking down its instruction into a series of discrete pipeline stages which can be completed in sequence by specialized hardware. The number of pipeline stages is referred to as the pipeline depth.

30. Write some commercial DSP Processors.

TMS320C50, TM 320C54, TM 320C55, ADSP-219x, ADSP-219xx, TMS320C3x, TMS320C67x, ADSP-21xxx.

31. Justify the usage of Branch, Call and Return instruction in digital signal processor.

In P-DSPs branch, call and return instructions are called as delayed branch/call/return instructions used for pipelining. When a delayed branch/call/return instructions are executed there is no need for flushing the pipeline and maximum throughput is obtained. For example when delayed branch instruction is executed, the program branches to the new program address only after two 1-word instruction or single 2-word instruction following the branch instruction are completely executed.

32. Name some assembler directives and its usage in any Digital signal processor.

.mmregs – Permit the memory map registers to be referred using the names such as AR0, SP etc.
.end – The end of assembly language program
.data – Assemble into data memory area
.text – Assemble into program memory area

33. State how a Digital Signal Processor is different from other processors.

DSP processor has features to support high-performance, repetitive, numerically intensive tasks. DSP processors are designed specifically to perform large number of complex arithmetic calculations and as quickly as possible.

PART-B

1. Describe in detail the architecture of TMS 320C 54 DSP processor and state the main features of this processor

2. Explain the following with reference to DSP processors.
(i) MAC (ii) Pipelining

3.Explain the addressing modes of a DSP processor with suitable examples (OR) What are the special addressing modes of TMS 320C54 chip

4. Explain Von Neumann, Harvard architecture and modified Harvard architecture for the computer

5. Explain the advantages and disadvantages of VLIW architecture

6.(i)Write short notes on memory mapped register addressing
(ii)Explain about pipelining in DSP

7.(i)Write short notes on circular addressing mode
(ii)Write short notes on auxiliary registers

8. Explain how convolution is performed using a single MAC unit. Discuss the addressing modes used in programmable DSPs.

9. Explain the types of operations performed by L functional mode.

10.(i)Discuss on the addressing modes supported by a DSP processor
(ii)Design a DSP based system for the process of audio signals in an audio recorder system.

11. (i) Explain the data path architecture and the bus structure in a DSP processor with suitable diagram
(ii) Elaborate on radar signal processing using a DSP processor

12.Compute the following if x_1 =[-1,-1,-1,2]; x= [-2,-1,-1,-2]
(i) Linear and circular convolution of a sequence
(ii)x_1:x_2 subject to addition and multiplication

13.Write briefly on any two of the following:
(i) Quantisation and errors in DS processor
(ii) With neat figure explain the architecture of any one type of a DS processor
(iii)The addressing modes of one type of DS processor.

14.Elaborate one application of digital signal processing with a DS processor

15. Discuss the features and architecture of TMS 32050 processor.

16. Explain the addressing mode and registers of DSP Processors.

17. Explain the classification of instructions in DSP processor with suitable examples.

18.Draw the structure of central processing unit and explain the each units with its functions.

19. Discuss the architecture of any one commercial DSP and explain with necessary diagram,

20.Discuss in brief the addressing format and functional modes of a any one commercial DSP.