# DEPARTMENT OF ELECTRONICS AND COMMUNICATION ENGINEERING

## EC6502 PRINCIPLES OF DIGITAL SIGNAL PROCESSING

### (2 Mark Questions and Answers)

## UNIT I

## DISCRETE FOURIER TRANSFORM

**1. Define DSP.**

DSP - Digital Signal Processing. It is defined as changing or analyzing information which is measured as discrete time sequences.

**2. List out the basic elements of DSP.**

- Signal in
- Analog to Digital converter
- Digital Signal processor
- Digital to Analog converter
- signal out

**3. Mention the advantages of DSP.**

(i)Veracity        (ii) Simplicity     (iii) Repeatability

**4. Give the applications of DSP.**

* Telecommunication – spread spectrum, data communication

* Biomedical – ECG analysis, Scanners.

* Speech/audio – speech recognition

* Military – SONAR, RADAR

**5. Define Signal.**

Signal is a physical quantity that varies with respect to time , space or any other independent variable.

(Or) It is a mathematical representation of the system

Eg.  $y(t) = t$ and $x(t)= \sin t$.

**6. Define system.**

A set of components that are connected together to perform the particular task. E.g. Filters.

**7. What are the major classifications of the signal?**

(i) Discrete time signal

(ii) Continuous time signal

**8. Define continuous time signals and classify them.**

Continuous time signals are defined for a continuous of values of the Independent variable. In the case of

continuous time signals the independent Variable is continuous. For example:

(i)    A speech signal as a function of time

(ii)   Atmospheric pressure  as a function of altitude

Classification of continuous time signal:  (i)Periodic and A periodic  signal  (ii)Even and Odd signal

**9. Define discrete time signals and classify them.**

Discrete time signals are defined only at discrete times, and for these signals, the independent variable takes on only a discrete set of values.

Classifications of discrete time signals are,

1. Periodic and A periodic signal

2. Even and Odd signal

**10. Define discrete time unit step &unit impulse.**

Discrete time Unit impulse is defined as

$\delta[n]$=    {1,  n=0; }

            {0, otherwise}

Unit impulse is also known as unit sample. Discrete time unit step signal is defined by

$U[n]$=   {0, n=0}

        {1, n>= 0}

**11. Define even and odd signal. (NOV/DEC-2010)**

A discrete time signal is said to be even when, $x[-n]=x[n]$. The continuous time signal is said to be even when, $x(-t)= x(t)$

For example, Cosine wave is an even signal. The discrete time signal is said to be odd when $x[-n]= -x[n]$

➢ The continuous time signal is said to be odd when $x(-t)= -x(t)$

➢ Odd signals are also known as non-symmetrical signal. Sine wave signal is an odd signal.

**12. Define Energy and power signal.**

A signal is said to be energy signal if it have finite energy and zero power.  A signal is said to be power signal if it have infinite energy and finite power. If the above two conditions are not satisfied then the signal is said to be neither energy nor power signal.

**13. What is analog signal?**

The analog signal is a continuous function of independent variables.  The analog Signal is defined for every instant of independent variable and so magnitude of Independent   variable is continuous in the specified range.   Here both the independent Variable and magnitude are continuous.

**14. What is digital signal?**

The digital signal is same as discrete signal except that the magnitude of signal is quantized.

**15. What are the different types of signal representations?**

A). Graphical representation            b). Functional representations

c). Tabular representation            d). Sequence representation.

**16. Define periodic and non periodic discrete time signals?**

If the discrete time signal repeated after equal samples of time then it is called periodic signal. When the discrete time signal $x[n]$ satisfies the condition $x[n+N]=x(n)$, then it is called periodic signal with fundamental period N samples. If $x(n) * x(n+N)$ then it is called non periodic signals.

**17. State the classification of discrete time signals.**

The types of discrete time signals are

* Energy and power signals

* Periodic and A periodic signals

* Symmetric (Even) and Ant symmetric (Odd) signals

**18. Define energy and power signal.**

If E is finite i.e. $0 < E < \alpha$, then x (n) is called energy signal.

If P is finite i.e. $0 < P < \alpha$, then the signal x(n) is called a power signal.

**19. What are all the blocks are used to represent the CT signals by its samples?**

(i) Sampler     (ii) Quantizer

**20. Define sampling process.**

Sampling is a process of converting Ct signal into Dt signal.

**21. Mention the types of sampling.**

(i) Up sampling      (ii) Down sampling

**22. What is meant by quantizer?**

It is a process of converting discrete time continuous amplitude into discrete time discrete amplitude.

**23. Define system function?**

The ratio between z transform of out put signal y(z) to z transform of input signal x(z) is called system function of the particular system.

**24. List out the types of quantization process.**

(i) Truncation          (ii) Rounding

**25. Define truncation.**

Truncating the sequence by multiplying with window function to get the finite value.

**26. State sampling theorem. (APRIL/MAY 2011) (NOV/DEC-11)**

➢ A band limited signal of finite energy, which has no frequency components higher than the W hertz, is completely described by specifying the values of the signal at the instant of time separated by 1/2W seconds and

➢ A band limited signal of finite energy, which has no frequency components higher than the W hertz, is completely recovered from the knowledge of its samples taken at the rate of 2W samples per second.

➢ The sampling frequency must be at least twice the maximum frequency present in the signal. That is Fs = > 2fm    Where, Fs = sampling frequency

Fm = maximum frequency

**27. Define Nyquist rate. (MAY/JUNE-12)**

It is the minimum rate at which a signal can be sampled and still reconstructed from its samples. Nyquist rate is always equal to 2fm.

**28. Define aliasing or folding. (APRIL/MAY 2011) (NOV/DEC-2010)**

The superimposition of high frequency behaviour on to the low frequency behaviour is referred as aliasing. This effect is also referred as folding.

**29. What is the condition for avoid the aliasing effect?**

To avoid the aliasing effect the sampling frequency must be twice the maximum frequency present in the signal.

**30. What is meant by interpolation?**

It is also referred as up sampling. that is , increasing the sampling rate.

**31. What is an anti-aliasing filter?**

A filter that is used to reject high frequency signals before it is sampled to remove the aliasing of unwanted high frequency signals is called an ant aliasing filter.

**32. Mention the types of sample/hold?**

➢ Zero order hold

➢ First order hold

**33. What is meant by sampling rate?**

Sampling rate = number of samples / second.

**34. What is meant by step response of the DT system?**

The output of the system $y(n)$ is obtained for the unit step input $u(n)$ then it is said to be step response of the system.

**35. Define Transfer function of the DT system.**

The Transfer function of DT system is defined as the ratio of Z transform of the system output to the input. That is, $H(z)=Y(z)/X(z)$.

**36. Define impulse response of a DT system.**

The impulse response is the output produced by DT system when unit impulse is applied at the input. The impulse response is denoted by $h(n)$. The impulse response $h(n)$ is obtained by taking inverse Z transform from the transfer function $H(z)$.

**37. What are the properties of convolution?**

(I) Commutative          (II)Associative    (III) Distributive

**38. State the Commutative properties of convolution. (NOV/DEC-11)**

Commutative property of Convolution is,

$x(n)*h(n) = h(n)*x(n)$

**39. State the Associative properties of convolution**

Associative Property of convolution is, $[x(n)*h1n)]*h2(n)=x(n)*[h1(n) h2(n)]$

**40. State Distributive properties of convolution. (NOV/DEC-11)**

The Distributive Property of convolution is,

$\{x(n)*[h1(n)+ h2(n)]\}= [x(n)*h1(n) + x(n)*h2(n)]$

**41. Define causal LTI DT system.**

For a LTI system to be causal if h(n)=0, for n<0.

**42. What are the steps involved in calculating convolution sum?**

The steps involved in calculating sum are

1).Folding                    2). Shifting                    3).Multiplication

4).Summation

**43. What is the condition for stable LTI DT system?**

A LTI system is stable if, shere, the summation is absolutely sum able.

**44. Define a causal system.**

The causal system generates the output depending upon present &past inputs only. A causal system is non anticipatory.

**45. What is meant by linear system?**

A linear system should satisfy superposition principle. A linear system should satisfy   F[ax1(n)+bx2(n)]
= a y 1(n)+by2(n)              Where, y1 (n)=F[x1(n)]            y2(n)=F[x2(n)]

**46. Define linear time invariant system. (NOV/DEC-11)**

1 .A system is time invariant if the behaviour and characteristics of the system are fixed over time.

2. A system is time invariant if a time shift in the input signal results in an identical time shift in the output signal.

3. For example, a time invariant system should produce y(t-t0)as the output when x(n-no) is the input.

# PART-B

1).Explain the digital signal processing system with necessary sketches and gives its merits and demerits.

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

3).Check for following systems are linear, causal, time in variant, static.

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

 (v) y (n) = x(n) + n x (n+2).

4).compute linear and circular convolution of the two sequence $x_1(n)$ = {1, 2, 2, 2} and
 X2 (n) = {1, 2, 3, 4}.

5). Discuss whether the following are energy or power signals (i) x(n) = (3/2)n u(n) (ii) x (n) = Aejwn .

6).Describe in detail the process of sampling and quantization. Also determine the expression for quantization.

7).Check whether the following are periodic.  (i) x (n) = cos (3πn)  (ii) x(n) = sin(3n)

**1. Define z transform?**

The Z transform of a discrete time signal x(n) is defined as,

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

Where, z is a complex variable. In polar form $z = re-j\omega$

**2. What is meant by ROC? (APRIL/MAY 2011-14) (NOV/DEC-11)**

The region of convergence (ROC) is defined as the set of all values of z for which X(z) converges.

**3. Explain about the roc of causal and anti-causal infinite sequences?**

For causal system the roc is exterior to the circle of radius r. For anti causal system it is interior to the

Circle of radius r.

**4. Explain about the roc of causal and anti causal finite sequences**

For causal system the roc is entire z plane except z=0. For anti causal system it is entire z plane except z=α.

**5. What are the properties of ROC?**

a). The ROC is a ring or disk in the z plane cantered at the origin.

b). The ROC cannot contain any pole.

c). The ROC must be a connected region

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

**6. Explain the linearity property of the z transform**

If   $z\{x_1(n)\}=X_1(z)$ and $z\{X_2(n)\}=x_2(z)$ then, $z\{ax_1(n)+bx_2(n)\}=aX_1(z)+bX_2(z)$

a&b are constants.

**7. State the time shifting property of the z transform**

If $z\{x(n)\}=X(z)$ then $z\{x(n-k)\}=z-kX(z)$

**8. State the scaling property of the z transform**

If $z\{x(n)\}=X(z)$ then $z\{an\ x(n)\}=X(a-1\ z)$

**9. State the time reversal property of the z transform.(MAY-12)**

If $z\{x(n)\}=X(z)$ then $z\{x(-n)\}=X(z-1)$

**10. Write the convolution property of the z transform**

If $z\{x(n)\}=X(z)$ & $z\{h(n)\}=H(z)$ then, $z\{x(n)*h(n)\}=X(z)H(z)$

**11. Write the multiplication property of z transform**

If $z\{x(n)\}=x(z)$ & $z\{h(n)\}=h(z)$ then,

**12. State Parseval's relation in z transform.(MAY/JUNE-2014)**

If  $x_1[n]$ and $x_2[n]$ are complex valued sequences, then the Parseval's relation states that

$x_1[n]\,x_2^*[n]= 1/2\Pi j\ \ X_1(v).\,X_2^*(1/v^*)\,v^{-1}dv.$

**13. State initial value theorem of z transforms.(MAY/JUNE-14)**

If $L[x(t)]=X(s)$, then initial value theorem states that

$x(0)=\lim(s\text{---}> )\ SX(S)$

**14. State final value theorem of z transforms. .(MAY/JUNE-14)**

If $L[x(t)]=X(s)$, then final value theorem states that $\lim(t\text{---}> )$
$x(t)=\lim(s\text{---}> 0)\ SX(S)$
If $x(n)$ is causal $z\{x(n)\}=X(z)$,

**15. Define system function.**

The ratio between z transform of out put signal $y(z)$ to z transform of

input signal $x(z)$ is called system function of the particular system.

**16. What are the conditions of stability of a causal system?**

All the poles of the system are with in the unit circle. The sum of impulse response for all values of n is

bounded.

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

It can be evaluated using several methods.

i). Long division method

ii). Partial fraction expansion method

iii). Residue method

iv). Convolution method

**18. Give the Z-transform pair.**

$$x(n) \quad \xleftarrow{\ z\ }\rightarrow X(z)$$

**19. What is the need for Z-transform?**

Z-transform is used for analysis the both periodic and a periodic signals.

**20. Give the Z-transform of unit sample sequence δ(n).**

$Z[\ \delta(n)\ ] =1$

**21. Define zeros**.

The zeros of the system $H(z)$ are the values of z for which $H(z) = 0$.

**22. Define poles.**

The poles of the system $H(z)$ are the values of z for which $H(z) = \alpha$.

**23. What is the z-transform of A δ (n-m)?**

$Z[\ A\ \delta(n\text{-}m)\ ] =1$.

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

$x(n)=\{1,2,3,4\}\ X(z)= x(n)z^{-n}$

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

$= 1+2/z+3/z2+4/z3$

**25. State the convolution properties of Z transform?**

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

**26. What z transform of (n-m)?**

By time shifting property

Z [A (n-m)]=AZ-m sinZ[ (n)] =1

**27. Obtain the inverse z transform of X(z)=1/z-a,|z|>|a|  ?**

Given X(z)=z-1/1-az-1

By time shifting property

X(n)=an.u(n-1)

**28. What is the relation between DFT and Z transform? (APRIL/MAY 2011)**

Both DFT and Z-transform work for discrete signal. I have read that "Z-transform is the general case of DFT, when we consider unit circle then, Z-transform becomes Discrete Fourier Transform (DFT)".

## PART-B

1).Explain in detail about overlap add method and overlap save method for filtering of long data sequences using DFT.

2).Develop a 8 point DIT-FFT algorithm. Draw the signal flow graph.

3).Explain Radix-2 DIT-FFT algorithm. Compare it with DIT-FFT algorithms.

4).Compute the linear convolution of finite duration sequences h(n) = { 1, 2 } and

x(n) = {1, 2, -1, 2,3, -2, -3, -1, 1, 1, 2, -1 } by overlap add method.

5).Compute the eight-point DFT of the sequence x(n) = 1    0≤ n ≤ 7

$\qquad\qquad\qquad\qquad\qquad\qquad\qquad$ = o   otherwise

6).By using DIT and DIF algorithm.

7).Summarize the Difference between overlap-save method and overlaps add method.

8).Compute an 8 point DFT of the sequence x (n) = ( 1, 0, 1, -1, 1, -1, 0, 1 ).

9).Find the output y(n) of a filter whose impulse response is h(n) = {1, 1, 1 } and input signal

 x(n) = {3, -1, 0,1,3,2,0,1,2,1} overlap save method.

## UNIT III

## FIR FILTER DESIGN

**1. Define DFT of a discrete time sequence.**

The DFT is used to convert a finite discrete time sequence x(n) to an N point frequency domain sequence X(k).The N point DFT of a finite sequence x(n) of length L,(L<N) is defined as,

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

## 2. Define Inverse DFT.

The Inverse DFT of the sequence of length N is defined as,

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

## 3. List any four properties of DFT

A). Periodicity           B).Linearity

C). Time reversal        D).Circular time shift

E). Duality               F).Circular convolution

G). Symmetry          H).Circular symmetry

## 4. State periodicity property with respect to DFT.

If x(k) is N-point DFT of a finite duration sequence x(n), then x(n+N) = x(n) for   all n.

X (k+N) = X(k) for all k.

## 5. State periodicity property with respect to DFT.

If   X1(k) and X2(k) are   N-point DFTs of finite duration sequences x1(n) and x2(n), then DFT [a X1(n) + b X2(n)] = a X1(k) + b X2(k), a, b are constants.

## 6. State time reversal property with respect to DFT.

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

DFT[x((-n))N] = DFT[x(N-n)] = X((-k))N = X(N-k)

## 7. Define circular convolution.

Let x1(n) and x2(n) are finite duration sequences both of length n with DFTs x1(k) and x2(k). If X3(k) = X1(k) X2(k), then the sequence X3(k) can be obtained by circular convolution.

## 8. What is the need for DFT?

DFT is used for analysis the both periodic and a periodic signals.

## 9. What is zero padding? What are its uses? (MAY/JUNE-2014)

Let the sequence x (n) has a length L. If we want to find the N-point DFT (N>L) of the sequence x(n), we have to add (N-L) zeros to the sequence x(n). This is known as Zero padding.

The uses of zero padding are

1) We can get better display of the frequency spectrum.

2) With zero padding the DFT can be used in linear filtering.

## 10 Why FFT is needed?

The direct evaluation of DFT requires complex multiplications and  N  - N complex additions. Thus for large values of N direct evaluation of the DFT is difficult. By using FFT algorithm the number of complex computations can be reduced.

**11. What is FFT?**

The Fast Fourier Transform is an algorithm used to compute the DFT. It makes use of the symmetry and periodicity properties of twiddle factor to effectively reduce the DFT computation time. It is based on the fundamental principle of decomposing the mutation of DFT of a sequence of length N into successively smaller DFTs.

**12. 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 log2 N and N/2 log2 N respectively.

**13. What is meant by radix-2 FFT?**

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

**14. What is DIT algorithm?**

Decimation-In-Time algorithm is used to calculate the DFT of a N point sequence. The idea is to break the N point sequence into two sequences, the DFTs of which can be combined to give the DFT of the original N point sequence. This algorithm is called DIT because the sequence x(n) is often spitted into smaller sub- sequences.

**15. What DIF algorithm?**

It is a popular form of the FFT algorithm. In this the output sequence X(k) is divided into smaller and smaller sub-sequences , So it is called the name is "Decimation In Frequency".

**6. What are the applications of FFT algorithm?**

The applications of FFT algorithm includes

1) Linear filtering

2) Correlation

3) Spectrum analysis

**17. Distinguish between linear convolution and circular convolution of two sequences.**

If $x(n)$ is a sequence of L number of samples and $h(n)$ with M number of samples, after convolution y(n). If Circular convolution $x(n)$ is a sequence of L number of samples.

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

**Differences:**

1) The input is bit reversed while the output is in natural order for DIT, whereas for DIF the output is bit reversed while the input is in natural order.

2) 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.

### 19. What is meant by radix-2 FFT?

If the number of output points N can be expressed as a power of 2, i.e., N = 2M Where M is an integer then this algorithm is known as radix-2 algorithm.

### 20. What is DIT radix-2 algorithm?

The radix 2 DIT FFT is an efficient algorithm for computing DFT. The idea is to break N point sequence in to two sequences, the DFT of which can be combined to give DFT of the original N-point sequence. Initially the N point sequence is divided in to two N/2 point sequences, on the basis of odd and even and the DFTs of them are evaluated and combined to give N-point sequence. Similarly the N/2 DFT s are divided and expressed in to the combination of N/4 point DFTs. This process is continued until we left with 2-point DFT's.

### 21. What is DIF radix-2 algorithm?

1. The radix 2 DIF FFT is an efficient algorithm for computing DFT in this the out put sequence x(k) is divided in to smaller and smaller.

2. The idea is to break N point sequence in to two sequences ,x1(n) and x2(n) consisting of the first N/2 points of x(n)and last N/2 points of x(n) respectively. Then we find N/2 point sequences f(n) and g(n Similarly).

3. The N/2 DFT s are divided and expressed in to the combination of N/4 point DFT' s. This process is continued until we left with 2-point DFT's.

### 22. What are the differences between DIT and DIF algorithms? (MAY/JUNE-14)(NOV/DEC-10)

* For DIT the input is bit reversed and the output is in natural order, and in DIF the input is in natural order and output is bit reversed.

* In butterfly the phase factor is multiplied before the add and subtract operation but in DIF it is multiplied after add-subtract operation.

    1).DIT – Time is decimated and input is bi reversed format output in natural order

    2).DIF – Frequency is decimated and input is natural order output is bit reversed Format

### 23. What is meant by in place in DIT and DFT algorithm?

An algorithm that uses the same location to store both the input and output sequence is called in-place algorithm.

### 25. How many stages are there for 8 point DFT

   3 stages

### 26 How many multiplication terms are required for doing DFT by expressional?

    Expression Method and FFT method

    Expression –N2    FFT - N /2 log N

### 27. What is DFT?

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

### 28. What is the DFT of unit impulse δ(n) ?

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

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

Circular convolution contains same number of samples as that of x(n) and h(n), while in linear convolution, number of samples in the result(N) are,N =L + M -1.

Where, L = Number of samples in x(n).

M = Number of samples in h(n). So the result of linear and circular convolution is not same.

**30. How to obtain same result from linear and circular convolution?**

* Calculate the value of 'N', that means number of samples contained in linear convolution.

* By doing zero padding make the length of every sequence equal to number of samples contained in linear convolution.

* Perform the circular convolution. The result of linear and circular convolution will be same.

**31. How will you perform linear convolution from circular convolution?**

* Calculate the value of 'N', that means number of samples contained in linear convolution.

* By doing zero padding make the length of every sequence equal to number of samples contained in linear convolution.

* Perform the circular convolution. The result of linear and circular convolution will be same.

**32. What methods are used to do linear filtering of long data sequences?**

* Overlap save method. * Overlap adds method.

**33. What is the disadvantage of direct computation of DFT?**

For the computation of N-point DFT, $N^2$ complex multiplication and $N^2 - N$ complex additions are required. If the value of N is large then the number of computations will go into lakhs. This proves inefficiency of direct DFT computation.

**34. What is the way to reduce number of arithmetic operations during DFT computation?**

Numbers of arithmetic operations involved in the computation of DFT are greatly reduced by using different FFT algorithms as follows,

1). Radix-2 FFT algorithm.

A).Radix-2 Decimation In Time (DIT) algorithm.

B).Radix-2 Decimation In Frequency (DIF) algorithm.

2).Radix-4 FFT algorithm.

**35. What are the properties of twiddle factor?**

* Twiddle factor is periodic.

* Twiddle factor is symmetric.

**36. What is up sampling process and what its effect?**

Addition of one zero after each sample in x(n) is called up sampling process. Due to this process, the entire DFT repeats one time.

**37. How linear filtering is done using FFT?**

Correlation is the basic process of doing linear filtering using FFT. The correlation is nothing but the convolution with one of the sequence, folded. Thus, by folding the sequence h(n), we can compute the linear filtering (convolution) using FFT.

## PART-B

1).Explain the designing of FIR filters using frequency sampling method.

2).State and explain the properties of FIR filters. State their importance. (8)

3).Explain linear phase FIR structures. What are the advantages of such structures?

4).Design an ideal high pass filter with a frequency response

$$Hd (ejw) = 1 \text{ for } \pi/4 \leq \left| w \right| \leq \pi$$

$$= 0 \text{ for } \left| w \right| \leq \pi/4 \text{ Find the values of h(n) for N = 11 using}$$

hamming window. Find H (z) and determine the magnitude response.

5).Explain the designing of FIR filters using Windows.

6).Obtain the direct form I, direct form II and Cascade form realization of the following system functions.

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

## UNIT IV

## FINITE WORD LENGTH EFFECTS

**1. What is a digital filter?**

A digital filter is a device that eliminates noise and extracts the signal of interest from other signals.

**2. Analog filters are composed of which parameters?**

1). pass band 2).stop band   3).Cut-off frequency

**3. Define pass band.**

It passes certain range of frequencies. In this, attenuation is zero.

**4. Define stop band.**

It suppresses certain range of frequencies. In this, attenuation is infinity.

**5. What is mean by cut-off frequency?**

This is the frequency which separates pass band and stop band.

**6. What is the difference between analog and digital filters?**

Analog filters are designed using analog components (R,L,C) while digital filters are implemented using difference equation and implemented using software.

**7. What are the basic types of analog filters?**

1). Low pass filter – LPF 2).High pass filter – HPF  3).Band pass filter - BPF

4).Band stop filter - BSF

**8. What is the condition for digital filter to be realized?**

The impulse response of filter should be causal,  h(n) = 0 for n<0.

**9. Why ideal frequency selective filters are not realizable?**

Ideal frequency selective filters are not realizable because they are non- causal. That is, its impulse response is present for negative values of 'n' also.

**10. For IIR filter realization what is required?**

Present, past, future samples of input and past values of output are required.

**11. Why IIR systems are called recursive systems?**

Because the feedback connection is present from output side to input

**12. Which types of structures are used to realize IIR systems?**

**1).** Direct form structure   2).Cascade form structure   3).Parallel form structure

**13. Why direct form-II structure is preferred most and why?**

The numbers of delay elements are reduced    in direct form-II structure compared to direct form-I structure. That means the memory locations are reduced in direct form-II structure.

**14. Why direct form-I and direct form-II are called as direct form structures?**

The direct form-I and   direct form-II structures    are obtained directly from the corresponding transfer function without any   rearrangements.   So these structures are called as direct form structures.

**15. What is advantage of direct form structure?**

Implementation of direct form is very easy.

**16. Give the disadvantage of direct form structure?**

Both direct form structures are sensitive to the effects of quantization errors in the coefficients. So practically not preferred

**17. What is the use of transpose operation?**

If two digital structures have the same transfer function then they are called as equivalent   structures. By using   the transpose operation,   we can obtain equivalent structure from a given realization structure.

**18. What is transposition or flow graph reversal theorem?**

If we reverse the directions of all branch transmittances and interchange input and output in the flow graph then the system transfer function remains unchanged.

**19. How a transposed structure is obtained?**

1). Reverse all signal flow graph directions.

2). Change branching nodes into adders and vice-versa.

3). Interchange input and output.

**20. Why feed back is required in IIR systems?**

It is required to generate infinitely long impulse response in IIR systems.

**21. Write the expression for order of Butterworth filter?**

The expression is $N = \log (\lambda /\epsilon)^{1/2} / \log (1/k)^{1/2}$

**22. Write the expression for the order of chebyshev filter?**

$$N = \cosh^{-1}(\lambda/e)/\cosh^{-1}(1/k)$$

**23. Write the various frequency transformations in analog domain?**

LPF to LPF: $s = s/\Omega c$
LPF to HPF: $s = \Omega c/s$
LPF to BPF: $s = s2xlxu/s(xu-xl)$
LPF to BSF: $s = s(xu-xl)?s2 = xlxu$. $X = \Omega$

**24. Write the steps in designing chebyshev filter?**

1. Find the order of the filter.

2. Find the value of major and minor axis.

3. Calculate the poles.

4. Find the denominator function using the above poles.

5. The numerator polynomial value depends on the value of n.

If n is odd: put s=0 in the denominator polynomial. If n is even put s=0 and divide it by $(1+e2)1/2$

**25. Write down the steps for designing a Butterworth filter?**

1. From the given specifications find the order of the filter
2 find the transfer function from the value of N
3. Find $\Omega c$
4 find the transfer function ha(s) for the above value of $\Omega c$ by su s by that value.

**26. State the equation for finding the poles in chebyshev filter.**

$sk = acos¢k + jbsin¢k$, where $¢k = \prod/2 + (2k-1)/2n)\prod$

**27. State the steps to design digital IIR filter using bilinear method.**

Substitute s by $2/T (z-1/z+1)$, where $T = 2/\Omega$ (tan (w/2) in h(s) to get h (z)

**28. What is warping effect or frequency warping? (NOV/DEC-10)**

For smaller values of w there exist linear relationship between w and but for Larger values of w the relationship is nonlinear. This introduces distortion in the Frequency axis. This effect compresses the magnitude and phase response. This Effect is called warping effect.

**29. Write a note on pre warping or pre scaling. (APRIL/MAY 2011) (MAY/JUNE-2014)**

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 pre warping the critical frequencies.

**30. Give the bilinear transform equation between s plane and z plane.**

$s = 2/T (z-1/z+1)$

**31. Give hamming window function.(MAY/JUNE-14)**

The equation for Hamming window is given by

$$WH(n)= 0.54-0.46 \cos 2\pi n/M-1 \qquad 0 \le n \le M-1$$
$$0 \qquad\qquad\qquad otherwise$$

**32. What is meant by impulse invariant method?**

In this method of digitizing an analog filter, the impulse response of the resulting digital filter is a sampled version of the impulse response of the analog filter. For e.g. if the transfer function is of the form, 1/s-p, then

$$H (z) =1/1-e-pTz-1$$

**33. What do you understand by backward difference?**

One of the simplest methods of converting analog to digital filter is to approximate the differential equation by an equivalent difference equation.

$$d/dt(y(t)/t=nT=(y(nT)-y(nT-T))/T$$

**34. What are the significance of chebyshev filter? (NOV/DEC-10)**

1. The magnitude response of the chebyshev filter exhibits ripple either in the stop band or the pass band.

2. The poles of this filter lies on the ellipse.

**35. Give the Butterworth filter transfer function and its magnitude characteristics for Different orders of filter.**

The transfer function of the Butterworth filter is given by
$$H (j\Omega) =1/1+j (\Omega/\Omega c) N$$

**36. Give the equation for the order N, major, minor axis of an ellipse in case of chebyshev filter?**

The order is given by $N=\cosh^{-1}(((10^{.1\alpha p})-1/10^{.1\alpha s}-1)1/2))/\cosh^{-1}\Omega s/\Omega p$

$A= (\mu^{1/N}-\mu^{-1/N})/2\Omega p$
$B=\Omega p (\mu^{1/N}+ \mu^{-1/N})/2$

**37. How can you design a digital filter from analog filter?**

Digital filter can de designed from analog filter using the following methods

1. Approximation of derivatives

2. Impulse invariant method (IIM)

3. Bilinear transformation (BLT)

**38. Write down bilinear transformation.**

$s=2/T (z-1/z+1)$

**39. What is a disadvantage of BLT method?**

The mapping is non-linear and because of this, frequency warping effect takes place.

## 40. List the Butterworth polynomial for various orders.

N Denominator polynomial

1).S+1

2).S2+.707s+1

3). (s+1)(s2+s+1)

4). (s2+.7653s+1)(s2+1.84s+1)

5). (s+1)(s2+.6183s+1)(s2+1.618s+1)

6). (s2+1.93s+1)(s2+.707s+1)(s2+.5s+1)

7). (s+1)(s2+1.809s+1)(s2+1.24s+1)(s2+.48s+1)

## 41. Differentiate Butterworth and Chebyshev filter.

Butterworth damping factor 1.44 and chebyshev is 1.06.Butterworth is flat response .but chebyshev is damped response.

## 42. What is filter?

Filter is frequency selective devices, which amplify particular range of frequencies and attenuate particular range of frequencies.

## 43. What are the types of digital filter according to their impulse response?

IIR (Infinite impulse response) filter

FIR (Finite Impulse Response) filter.

## 44. How phase distortion and delay distortion are introduced?

1. The phase distortion is introduced when the phase characteristics of a filter is Nonlinear with in the desired frequency band

2. The delay distortion is introduced when the delay is not constant with in the desired frequency band

## 45. Define IIR filter.

The filters designed by considering all the infinite samples of impulse response are called IIR filter.

## 46. What is the limitation of approximation of derivative method?

It is suitable only for designing of low pass and band pass IIR digital filters with relatively small resonant frequencies.

## 47. What is the condition for linear phase FIR filter? (APRIL/MAY 2011)

Linear phase is a property of a filter, where the phase response of the filter is a linear function of frequency. The result is that all frequency components of the input signal are shifted in time (usually delayed) by the same constant amount, which is referred to as the phase delay. And consequently, there is no phase distortion due to the time delay of frequencies relative to one another.

A filter with linear phase may be achieved by an FIR filter which is either symmetric or anti-symmetric.[1] A necessary but not sufficient condition is:

$$\sum_{n=-\infty}^{\infty} h[n] \cdot \sin(\omega \cdot (n - \alpha) + \beta) = 0$$

For some $\alpha, \beta$.[2]

**48.What is Gibb's Phenomenon?(MAY/JUNE-12)**

One possible way of finding an FIR filter that approximates H(ejw) would be to truncate the infinite Fourier series at n=±(N-1/2).Direct truncation of the series will lead to fixed percentage overshoots and undershoots before and after an approximated discontinuity in the frequency response.

## PART-B

1).Explain the characteristics of limit cycle oscillation with respect to the system described by the difference equation: y(n) = 0.95 y(n-1) + x(n); x(n) = 0 and y(-1) = 13. Determine the Dead band range of the system.

2).Explain the limit cycle oscillations due to product round off and overflow errors.

3).Discuss in detail the errors resulting from rounding and truncation.

4).Explain the quantization process and errors introduced due to quantization.

5).Describe the quantization in floating point realization of IIR digital filters.

6).Discuss about signal scaling and coefficient quantization.

## UNIT V

## DSP APPLICATIONS

**1. What are the factors that influence selection of DSPs?**

1). Architectural features

2).Execution speed

3).Type of arithmetic

4).Word length

**2. What are the classification digital signal processors?**

The digital signal processors are classified as

(i) General purpose digital signal processors.  (ii) Special purpose digital signal processors.

**3. 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.

**4. Give some examples for fixed point DSPs.**

TM32OC50, TMS320C54, TMS320C55, ADSP-219x, ADSP-219xx..

**5. Give some example for floating point DSPs?**

TMS320C3x, TMS320C67x, ADSP-21xxx

**6. What is pipelining? (MAY/JUNE-2014) (NOV/DEC-11))**

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

**7. What is pipeline depth?**

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

**8. What are the advantages of VLIW architecture?**

Advantages of VLIW architecture

A). Increased performance

B).Better compiler targets

C).Potentially easier to program

D).Potentially scalable

E).Can adds more execution units to allow more instructions to be packed into the VLIW instruction.

**9. What are the disadvantages of VLIW architecture?**

Disadvantages of VLIW architecture

1).New kind of programmer/compiler complexity

2).Program must keep track of instruction scheduling

3).Increased memory use

4).High power consumption

**10. What is the pipeline depth of TMS320C50 and TMS320C54x?**

TMS320C50 – 4    TMS320C54x – 6

**11. What are the different buses of TMS320C5x? (MAY/JUNE-2014)**

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)

**12. Give the functions of program bus?**

The program bus carries the instruction code and immediate operands from program memory to the CPU.

**13. Give the functions of program address bus?**

The program address bus provides address to program memory space for both read and write.

**14. Give the functions of data read bus?**

The data read bus interconnects various elements of the CPU to data memory space.

**16. Give the functions of data read address bus?**

The data read address bus provides the address to access the data memory space.

**17. What are the different stages in pipelining?**

        1).The fetch phase

        2).The decode phase

        3).Memory read phase

        4).The execute phase

**18. List the various registers used with ARAU. (MAY/JUNE-2014)**

- ➢ Eight auxiliary registers (AR0 – AR7)
- ➢ Auxiliary register pointer (ARP)
- ➢ Unsigned 16-bit ALU

**19. What is the operation blocks involved in C5x processors? (May/june-2014)**

        The central processing unit consists of the following elements:

        1).Central arithmetic logic unit (CALU)

        2).Parallel logic unit (PLU)

        3).Auxiliary register arithmetic unit (ARAU)

        4).Memory mapped registers

        5).Program controller

**20. What is the function of parallel logic unit?**

        The parallel logic unit is a second logic unit that executes logic operations on data without affecting the contents of accumulator.

**21. List the on chip peripherals in 'C5x.**

        The on-chip peripherals interfaces connected to the 'C5x CPU include

- ➢ w. Clock generator
- ➢ x. Hardware timer
- ➢ y.  Software programmable wait state generators
- ➢ z.  General purpose I/O pins

        aa. Parallel I/O ports

        bb. Serial port interface

        cc. Buffered serial port

        dd. Time-division multiplexed (TDM)serial port

        ee. Host port interface ff.

User unmask able interrupts

**22. What are the arithmetic instructions of 'C5x?**

        ADD, ADDB, ADDC, SUB, SUBB, MPY, MPYU

**23. What are the shift instructions?**

        ROR, ROL, ROLB, RORB, BSAR.

**24. What are the general purpose I/O pins?**

Branch control input (BIO) External flag (XF)

**25. What are the logical instructions of 'C5x?**

AND, ANDB, OR, ORB, XOR, XORB

**26. What are load/store instructions?**

LACB, LACC, LACL, LAMM, LAR, SACB, SACH, SACL, SAR, SAMM.

**27. Mention the addressing modes available in TMS320C5X processor?**

1). Direct addressing mode

2). Indirect addressing mode

3). Circular addressing mode

4). Immediate addressing

5). Register addressing

6). Memory mapped register addressing

**28. Give the features of DSPs? (APRIL/MAY 2011) (NOV/DEC-11)**

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

**29. What is function of NOP instruction?**

1).NOP- No operation  2). Perform no operation.

**30. What is function of ZAC instruction?**

ZAC – Zero accumulator  Clear the contents of accumulator to zero.

**31. Give the function of BIT instruction.**

BIT – Test bit (Copy the specified bit of the data memory value to the TC bit in ST1).

**32. Mention the function of B instruction.**

B – Branch conditionally. Branch to the specified program memory address. Modify the current AR and ARP as specified.

**33. What is use of ADD instruction?**

ADD – Add to accumulator with shift.

Add the content of  addressed  data memory location  or an  immediate value  of accumulator,  if a  shift is  specified,  left-shift  the  data  before  the  add. During shifting,  low- order bits  are  Zero-filled,  and high-order  bits  are  sign extended if SXM=1.

**34. Give the advantages of DSPs?**

Architectural features, Execution speed, Type of arithmetic, Word length

**35. Give the applications of DSP Processors?**

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

## PART-B

1).Explain in detail about the polyphase implementation of FIR filters for interpolator and decimators.

2).Describe the procedure to implement digital filter bank using multirate signal processing.

3).Explain the application of multirate signal processing.

4).Explain how DSP can be used for speech processing.

5).Explain the efficient transversal structure for decimator and interpolator.

6).Explain the application of sampling rate conversion in sub-band coding and Narrow Band filter.