SNS COLLEGE OF ENGINEERING

(An Autonomous Institution)

DEPARTMENT OF ECE

Subject: Digital Signal Processing

Year/Sem/Branch: II/IV/ECE

2 Marks Questions & Answers

UNIT – I -Review of discrete time signals and systems

Overview of signals and systems – DFT–FFT using DIT and DIF algorithms – Inverse DFT– Applications – Circular convolution – MATLAB programs for DFT and FFT.

1. How many multiplication and additions are required to compute N point DFT using radix 2 FFT? (*NOV/DEC 2004)*

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

2. Define DFT of a sequence.

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

3. State Periodicity Property of DFT.

If X(n) 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

4. What is the difference between DFT and DTFT? (***MAY/JUNE 2009)***

DFT

* Obtained by performing sampling operation in both the time and frequency domain
* Discrete frequency spectrum

DTFT

* Sampling is performed only in time domain
* Continuous function of frequency spectrum

5. What is meant by radix 2 FFT?

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

6. What is the main advantage of FFT?

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

7. State the properties of DFT.

1) Periodicity
2) Linearity and symmetry
3) Multiplication of two DFTs
4) Circular convolution
5) Time reversal
6) Circular time shift and frequency shift
7) Complex conjugate
8) Circular correlation

8. How to obtain the output sequence of linear convolution through circular convolution?

Consider two finite duration sequences x(n) and h(n) of duration L samples and M samples. The linear convolution of these two sequences produces an output sequence of duration L+M-1 samples, whereas, the circular convolution of x(n) and h(n) give N samples where N=max(L,M).In order to obtain the number of samples in circular convolution equal to L+M-1, both x(n) and h(n) must be appended with appropriate number of zero valued samples. In other words by increasing the length of the sequences x (n) and h(n) to L+M-1 points and then circularly convolving the resulting sequences we obtain the same result as that of linear convolution.

9. What is zero padding? What are its uses? (*NOV 2006,DEC 2009)*

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. Define sectional convolution.

If the data sequence x(n) is of long duration it is very difficult to obtain the output sequence y(n) due to limited memory of a digital computer. Therefore, the data sequence is divided up into smaller sections. These sections are processed separately one at a time and controlled later to get the output.

11. What are the two methods used for the sectional convolution?

The two methods used for the sectional convolution are
1) overlap-add method
2)overlap-save method.

12. What is overlap-add method?

In this method the size of the input data block xi(n) is L. To each data block we append M-1 zeros and perform N point circular convolution of xi(n) and h(n). Since each data block is terminated with M-1 zeros the last M-1 points from each output block must be overlapped and added to first M-1 points of the succeeding blocks. This method is called overlap-add method.

13. What is overlap-save method?

In this method the data sequence is divided into N point sections xi(n).Each section contains the last M-1 data points of the previous section followed by L new data points to form a data sequence of length N=L+M-1.In circular convolution of xi(n) with h(n) the first M-1 points will not agree with the linear convolution of xi(n) and h(n) because of aliasing, the remaining points will agree with linear convolution. Hence we discard the first (M-1) points of filtered section xi(n) N h(n). This process is repeated for all sections and the filtered sections are abutted together.

14. Why FFT is needed?

The direct evaluation DFT requires N2 complex multiplications and N2 –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. So we use FFT.

15. What is FFT? *NOV/DEC 2006*

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 computation of DFT of a sequence of length N into successively smaller DFTs.

16. 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 splitted into smaller sub- sequences.

17. 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 , that is why the name Decimation In Frequency.

18. What are the applications of FFT algorithm?

The applications of FFT algorithm includes
1) Linear filtering
2) Correlation
3) Spectrum analysis

19. Why the computations in FFT algorithm is said to be in place?

Once the butterfly operation is performed on a pair of complex numbers (a,b) to produce (A,B), there is no need to save the input pair. We can store the result (A,B) in the same locations as (a,b). Since the same storage locations are used throughout the computation we say that the computations are done in place.

20 .What are the differences and similarities between DIF and DIT algorithms? ***(NOV/DEC 2006)(MAY/JUNE 2009)***

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

21. Distinguish between linear convolution and circular convolution of two sequences ***MAY/JUNE 2006***

No.	Linear convolution	Circular convolution
1	If x(n) is a sequence of L number of samples and h(n) with M number of samples, after convolution y(n) will have N=L+M-1 samples.	If x(n) is a sequence of L number of samples and h(n) with M samples, after convolution y(n) will have N=max(L,M) samples.
2	It can be used to find the response of a linear filter.	It cannot be used to find the response of a filter.
3	Zero padding is not necessary to find the response of a linear filter.	Zero padding is necessary to find the response of a filter.

22. What are differences between overlap-save and overlap-add methods.

No	Overlap-save method	Overlap-add 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 block followed by L new data points	Each data block is L points and we append 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 data 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

23. Define twiddle factor of FFT. ***NOV/DEC 2009***

The complex number W_N is called phase factor or twiddle factor. The W_N represent a complex number $e^{-j2\Pi/N}$. It is used to reduce the computational complexity.

UNIT- II- Design and implementation of IIR filters

Design of analog filters using Butterworth and Chebyshev approximations – IIR digital filter design from analog filter using impulse invariance technique and bilinear transformations – Matlab programs for IIR filters.

1. What is filter?

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

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

- IIR (Infinite Impulse Response)filter
- FIR (Finite Impulse Response)filter.

3. Define IIR filter?

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

4. 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

5. What are the properties of chebyshev filter? ***NOV/DEC 2006***

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

2. The poles of this filter lies on the ellipse

6. Give the equation for the order N, major, minor axis of an ellipse in case of chebyshev filter? ***MAY/JUNE 2009***

The order is given by N=cosh-1(((10.1_p)-1/10.1_s-1)1/2))/cosh-1_s/_p

A= (µ1/N-µ-1/N)/2Ωp

B=Ωp(µ1/N+µ-1/N)/2

7. Write the various frequency transformations in analog domain?

LPF to LPF:s=s/c
LPF to HPF:s=c/s
LPF to BPF:s=s2xlxu/s(xu-xl)
LPF to BSF:s=s(xu-xl)/s2=xlxu.

8. 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

3. Bilinear transformation

9. Write down bilinear transformation.

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

10. 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)

11. Differentiate Butterworth and Chebyshev filter. ***MAY/JUNE 2006***

Butterworth damping factor 1.44 chebyshev 1.06

Butterworth flat response chebyshev damped response.

12. How phase distortion and delay distortion are introduced?

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

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

13. Distinguish IIR and FIR filters. ***NOV/DEC 2007***

- Impulse response is finite
- They have perfect linear phase
- Impulse Response is infinite
- They do not have perfect linear phase
- Non recursive.
- Greater flexibility to control the shape of magnitude response
- Less flexibility

14. Distinguish analog and digital filters

Constructed using active or passive components and it is described by a differential equation Consists of elements like adder, subtractor and delay units and it is described by a difference equation. Frequency response can be changed by changing the Components Frequency response can be changed by changing the filter Coefficients It processes and generates analog output Processes and generates digital output. Output varies due to external conditions Not influenced by external conditions

15. 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. 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

16. 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 c

4 find the transfer function ha(s) for the above value of c by by that value.

17. State the equation for finding the poles in chebyshev filter

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

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

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

19. What is warping effect? ***NOV/DEC 2003***

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.

20. Write a note on pre warping. ***NOV/DEC 2008, MAY/JUNE 2009***

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 prewarping the critical frequencies.

21. Give the bilinear transform equation between s plane and z plane

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

22. Why impulse invariant method is not preferred in the design of IIR filters other than low pass filter?

In this method the mapping from s plane to z plane is many to one. Thus there are an infinite number of poles that map to the same location in the z plane, producing an aliasing effect. It is inappropriate in designing high pass filters. Therefore this method is not much preferred.

23. What is meant by impulse invariant method? ***MAY/JUNE 2004***

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

24. Define FIR

This type of system has an impulse response which is zero outside a finite time interval

25. Define IIR

This type of system has an impulse response of infinite time interval

26. Mention the features of IIR filters

No linear phase
Have desired characteristics for magnitude response only

27. Mention the classification of filter based on frequency response

Low pass
High pass
Band pass
Band stop

28. Mention the advantages of digital filters

High thermal stability
Accurate
Easily programmable
Filtering is possible

UNIT – III- Design and implementation of FIR filters

Linear phase response- and -Design of Linear phase FIR filters - Fourier series method and frequency sampling method - Design of Linear phase FIR filters using windows: Rectangular, Hanning and Hamming windows- MATLAB program of FIR filters.

1. What is mean by FIR filter?

The filter designed by selecting finite number of samples of impulse response h(n) obtained from inverse Fourier transform of desired frequency response. H (w)) are called FIR filters

2. Write the steps involved in FIR filter design

- Choose the desired frequency response Hd (w)
- Take the inverse Fourier transform and obtain Hd (n)
- Convert the infinite duration sequence Hd (n) to h (n)
- Take Z transform of h(n) to get H(Z)

3. What are advantages of FIR filter? (*NOV/DEC 2004)*

- Linear phase FIR filter can be easily designed.
- Efficient realization of FIR filter exists as both recursive and non-recursive structures.
- FIR filter realized non-recursively stable.
- The round off noise can be made small in non recursive realization of FIR filter.

4. What are the disadvantages of FIR FILTER?(*NOV/DEC 2006)*

The duration of impulse response should be large to realize sharp cutoff filters. The non integral delay can lead to problems in some signal processing applications.

5. What is the necessary and sufficient condition for the linear phase characteristic of a FIR filter?

The phase function should be a linear function of w, which in turn requires constant group delay and phase delay.

6. List the well known design technique for linear phase FIR filter design?

- Fourier series method and window method
- Frequency sampling method.
- Optimal filter design method.

7. For what kind of application, the symmetrical impulse response can be used?

The impulse response, which is symmetric having odd number of samples can be used to design all types of filters, i.e. low pass, high pass, band pass and band reject.

The symmetric impulse response having even number of samples can be used to design low pass and band pass filter.

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

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

9. What condition on the FIR sequence h(n) are to be imposed n order that this filter can be called a liner phase filter? ***NOV/DEC 2005***

The conditions are

(i) Symmetric condition h(n)=h(N-1-n)

(ii) Antisymmetric condition h(n)=-h(N-1-n)

10. Under what conditions a finite duration sequence h(n) will yield constant group delay in its frequency response characteristics and not the phase delay?

If the impulse response is anti symmetrical, satisfying the condition

H(n)=-h(N-1-n)

The frequency response of FIR filter will have constant group delay and not the phase delay.

11. What are the features of FIR filter? ***NOV/DEC 2009***

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

12. When cascade from realization is preferred in FIR filters?

The cascade from realization is preferred when complex zeros with absolute magnitude less than one.

13. What are the disadvantages of Fourier series method?

In designing FIR filter using Fourier series method the infinite duration impulse response is truncated 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 .

14. What is Gibbs phenomenon(*MAY 2007,2009, DEC 2009)*

One possible way of finding an FIR filter that approximates H(ej)would be to truncate the infinite Fourier series at n= (N-1/2).Abrupt truncation of the series will lead to oscillation both in pass band and is stop band .This phenomenon is known as Gibbs phenomenon.

15. What are the desirable characteristics of the windows?(*APRIL/MAY 2007)*

The desirable characteristics of the window are

1. The central lobe of the frequency response of the window should contain most of the energy and should be narrow.

2. The highest side lobe level of the frequency response should be small.

3. The side's lobes of the frequency response should decrease in energy rapidly as w tends to π.

16. Compare Hamming window with Kaiser Window.

Hamming window Kaiser Window

1. The main lobe width is equal to8π/N and the peak side lobe level is –41dB.
2. The low pass FIR filter designed will have first side lobe peak of –53 dB. The main lobe width, the peak side lobe level can be varied by varying the parameter π and N.
3. The side lobe peak can be varied by varying the parameter π.

17. What is the necessary and sufficient condition for linear phase characteristics in FIR filter?(*MAY/JUNE 2006)*

The necessary and sufficient condition for linear phase characteristics in FIR filter is the impulse response h(n) of the system should have the symmetry property,i.e,

$$H(n) = h(N-1-n)$$

where N is the duration of the sequence.

18. What is the advantage of Kaiser Widow?

1. It provides flexibility for the designer to select the side lobe level and N .

2. It has the attractive property that the side lobe level can be varied

Continuously from the low value in the Blackman window to the high value in the rectangle window.

19. What is the principle of designing FIR filter using frequency sampling method?

In frequency sampling method the desired magnitude response is sampled and a linear phase response is specified .The samples of desired frequency response are defined as DFT coefficients. The filter coefficients are then determined as the IDFT of this set of samples.

20. For what type of filters frequency sampling method is suitable? (*NOV/DEC 2005)*

Frequency sampling method is attractive for narrow band frequency selective filters where only a few of the samples of the frequency response are non-zero.

21. What are the requirements of analog filters.

Transfer function should be rational function
Coefficient should be real
Poles in the left half

22. Mention the disadvantages of digital filters

Bandwidth is limited
Performance depend on hardware.

23. 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

24. Mention the techniques for digitizing analog filter

Bilinear transformation
Impulse invariant

25. Describe the features of IIR filters.

Physically realizable
Desired characteristics for magnitude

26. Compare Rectangular and Hanning windows.

Rectangular window	Hanning window
(a) The width of the mainlobe in window spectrum is 4Π/N.	(a) The width of the mainlobe in window spectrum is 8Π/N.
(b) The maximum sidelobe magnitude in window spectrum is -13dB.	(b) The maximum sidelobe magnitude in window spectrum is -31dB.
(c) In window spectrum the sidelobe magnitude slightly decreases with increasing ω.	(c) In window spectrum the sidelobe magnitude decreases with increasing ω.
(d) The minimum stop band attenuation is 22 dB.	(d) The minimum stop band attenuation is 44 dB.

27. How phase and delay distortions are introduced?

- ❖ The phase distortion is introduced when the phase characteristics of a filter is not linear within the desired frequency band.
- ❖ The delay distortion is introduced when the delay is not a constant within the desired frequency.

28. Compare Hanning and Hamming windows.

Hanning window	Hamming window
(a) The width of the mainlobe in window spectrum is 8Π/N.	(a) The width of the mainlobe in window spectrum is 8Π/N.
(b) The maximum sidelobe magnitude in window spectrum is -31dB.	(b) The maximum sidelobe magnitude in window spectrum is -41dB.
(c) In window spectrum the sidelobe magnitude slightly decreases with increasing ω.	(c) In window spectrum the sidelobe magnitude remains constant.
(d) The minimum stop band attenuation is 44 dB.	(d) The minimum stop band attenuation is 51 dB.

29. Compare Hamming and Blackman windows. ***APRIL/MAY 2007***

Hamming window	Blackman window
(a) The width of the mainlobe in window spectrum is $8\Pi/N$.	(a) The width of the mainlobe in window spectrum is $12\Pi/N$.
(b) The maximum sidelobe magnitude in window spectrum is -41dB.	(b) The maximum sidelobe magnitude in window spectrum is -58dB.
(c) In window spectrum the sidelobe magnitude remains constant.	(c) In window spectrum the sidelobe magnitude slightly decreases with increasing ω.
(d) The minimum stop band attenuation is 51 dB.	(d) The minimum stop band attenuation is 78 dB.
(e) The higher value of sidelobe attenuation is achieved at the expense of constant attenuation at higher frequencies.	(e) The higher value of sidelobe attenuation is achieved at the expense of increased mainlobe width.

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

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

A digital filter is stable if its impulse response is absolutely summable ,i.e,

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

UNIT IV Finite word length effects in digital filters

Fixed point arithmetic -effect of quantization of the input data due to Finite word length. - Product round off - need for scaling- Zero input limit cycle oscillations- Limit cycle oscillations due to overflow of adders - Table look up implementation to avoid multiplications.

1. What do you understand by a fixed-point number? *MAY/JUNE 2004*

In fixed point arithmetic the position of the binary point is fixed. The bits to the right represent the fractional part of the number & those to the left represent the integer part.

For example, the binary number 01.1100 has the value 1.75 in decimal.

2. What is meant by block floating point representation? What are its advantages?

In block point arithmetic the set of signals to be handled is divided into blocks. Each block has the same value for the exponent. The arithmetic operations with in the block uses fixed point arithmetic & only one exponent per block is stored thus saving memory. This representation of numbers is more suitable in certain FFT flow graph & in digital audio applications.

3. What are the advantages of floating point arithmetic?

1. Large dynamic range
2. Over flow in floating point representation is unlike.

4. What are the three-quantization errors to finite word length registers in digital filters?NOV/DEC 2004

1. Input quantization error
2. Coefficient quantization error
3. Product quantization error

5. What do you understand by input quantization error?

In digital signal processing, the continuous time input signals are converted into digital using a b-bit ACD. The representation of continuous signal amplitude by a fixed digit produce an error, which is known as input quantization error.

6. How the multiplication & addition are carried out in floating point arithmetic?

In floating point arithmetic, multiplication are carried out as follows,

Let f1 = M1*2c1 and f2 = M2*2c2. Then f3 = f1*f2 = (M1*M2) 2(c1+c2)

That is, mantissa is multiplied using fixed-point arithmetic and the exponents are added. The sum of two floating-point number is carried out by shifting the bits of the mantissa of the smaller number to the right until the exponents of the two numbers are equal and then adding the mantissas.

7. What is the relationship between truncation error e and the bits b for representing a decimal into binary?

For a 2's complement representation, the error due to truncation for both positive and negative values of x is 0>=xt-x >-2-b

Where b is the number of bits and xt is the truncated value of x. The equation holds good for both sign magnitude, 1's complement if x>0 If x<0, then for sign magnitude and for 1's complement the truncation error satisfies.

8. What is meant rounding? NOV/DEC 2006

Rounding a number to b bits is accomplished by choosing the rounded result as the b bit number closest to the original number unrounded.

9. What is meant by A/D conversion noise?MAY/JUNE 2009

A DSP contains a device, A/D converter that operates on the analog input x(t) to produce xq(t) which is binary sequence of 0s and 1s.

At first the signal x(t) is sampled at regular intervals to produce a sequence x(n) is of infinite precision. Each sample x(n) is expressed in terms of a finite number of bits given the sequence xq(n). The difference signal e(n)=xq(n)-x(n) is called A/D conversion noise.

10. What is the effect of quantization on pole location?NOV/DEC 2004

Quantization of coefficients in digital filters lead to slight changes in their value. This change in value of filter coefficients modifies the pole-zero locations. Some times the pole locations will be changed in such a way that the system may drive into instability.

11. Which realization is less sensitive to the process of quantization?

Cascade form.

12. What is meant by quantization step size? ***NOV/DEC 2006, 2008***

Let us assume a sinusoidal signal varying between +1 and -1 having a dynamic range 2. If the ADC used to convert the sinusoidal signal employs b+1 bits including sign bit, the number of levels available for quantizing x(n) is 2b+1. Thus the interval between successive levels q= 2 =2-b -------- 2b+1

Where q is known as quantization step size.

13. How would you relate the steady-state noise power due to quantization and the b bits representing the binary sequence?

Steady state noise power

Where b is the number of bits excluding sign bit.

14. What is overflow oscillation? ***MAY/JUNE 2005,NOV/DEC 2009***

The addition of two fixed-point arithmetic numbers cause over flow the sum exceeds the word size available to store the sum. This overflow caused by adder make the filter output to oscillate between maximum amplitude limits. Such limit cycles have been referred to as over flow oscillations.

15. What are the methods used to prevent overflow? NOV/DEC 2009

There are two methods used to prevent overflow

1. Saturation arithmetic
2. Scaling

16. What are the two kinds of limit cycle behavior in DSP?

1. Zero input limit cycle oscillations
2. Overflow limit cycle oscillations

17. What is meant by "dead band" of the filter? NOV/DEC 2009

The limit cycle occur as a result of quantization effect in multiplication. The amplitudes of the output during a limit cycle are confined to a range of values called the dead band of the filter.

18. Explain briefly the need for scaling in the digital filter implementation. (APR/MAY 2004)

To prevent overflow, the signal level at certain points in the digital filter must be scaled so that no overflow occurs in the adder.

19. What is meant by autocorrelation?

The autocorrelation of a sequence is the correlation of a sequence with its shifted version, and this indicates how fast the signal changes.

20. Define noise transfer function (NTF)?

The NTF is defined as the transfer function from the noise source to the filter output. The NTF depends on the structure of the digital network.

21. What are the two types of quantization employed in digital system?

The two types of quantization in digital system are truncation and rounding.

24. What is truncation? ***MAY/JUNE 2005***

The process of reducing the size of binary number by discarding all bits less significant than the least significant bit that is retained.

25. What is the drawback in saturation arithmetic? ***APR/MAY 2007***

The saturation arithmetic introduces non-linearity in the adder which creates signal distortion.

26. What are the types of arithmetic used in digital computers?

The floating point arithmetic and two's complement arithmetic are the two types.

UNIT-V Processor Fundamentals

Features of DSP processors - DSP processor packaging(Embodiments)- Fixed point Vs floating point DSP processor data paths - Fixed point Vs floating point DSP processor data paths – pipelining - TMS320 family of DSPs (architecture of C5x)- Memory architecture of a DSP processor (Von Neumann - Harvard) - Addressing modes.

1. What are the key features of DSP processors?

DSP processors have the following features:

- Specialized hardware for fast arithmetic operations
- Harvard architecture for parallel processing
- Dedicated multiply-accumulate (MAC) units
- Hardware looping and pipelining for efficiency
- High-speed memory access

2. Why is Harvard architecture preferred in DSP processors?

Harvard architecture allows separate data and instruction memory, enabling simultaneous fetching of data and instructions, improving processing speed.

3. What are the common packaging types of DSP processors?

DSP processors are commonly packaged as:

- DIP (Dual In-line Package)
- QFP (Quad Flat Package)
- BGA (Ball Grid Array)

4. Why is BGA packaging preferred for modern DSPs?

BGA packaging provides better heat dissipation, improved electrical performance, and higher pin density, making it suitable for high-performance DSP applications.

5. What is the difference between fixed-point and floating-point DSP processors?

Fixed-point processors represent numbers with fixed precision, whereas floating-point processors use a more flexible format that allows for a wider dynamic range and greater precision.

6. Which DSP processors are more power-efficient: Fixed-point or Floating-point?

Fixed-point DSP processors are more power-efficient because they require simpler hardware and consume less power compared to floating-point processors.

7. What is pipelining in DSP processors?

Pipelining is a technique where multiple instruction stages (fetch, decode, execute) are overlapped to improve execution speed and efficiency.

8. How does pipelining improve DSP performance?

Pipelining reduces execution time by allowing multiple instructions to be processed simultaneously, leading to higher throughput.

9. What are the main features of the TMS320C5x DSP processor?

Features of TMS320C5x:

- 16-bit fixed-point architecture
- Modified Harvard architecture
- Multiple addressing modes
- On-chip RAM and ROM

- High-speed MAC unit

10. What type of architecture does TMS320C5x use?
The TMS320C5x uses a Modified Harvard Architecture for efficient data and instruction access.

11. What is the difference between Von Neumann and Harvard architectures?
- Von Neumann: Uses a single memory for both data and instructions, leading to a bottleneck.
- Harvard: Uses separate memories for data and instructions, allowing parallel access and higher speed.

12.Why is the Harvard architecture used in DSP processors?
Harvard architecture enables faster data access and instruction execution by using separate buses for data and instructions, improving overall performance.

13. What are the common addressing modes in DSP processors?
Common addressing modes include:
- Immediate
- Direct
- Indirect
- Register
- Indexed
- Circular

14. What is circular addressing in DSP processors?
Circular addressing is a special addressing mode used in DSP processors for efficient handling of circular buffers, commonly used in signal processing applications.

15. What is the function of a Multiply-Accumulate (MAC) unit in DSP?
The MAC unit performs multiplication and accumulation in a single cycle, essential for signal processing tasks like filtering and convolution.

16. How does hardware looping benefit DSP performance?
Hardware looping reduces the overhead of loop control instructions, enabling faster execution of repetitive computations.

17. What is the significance of zero-overhead looping in DSP processors?
Zero-overhead looping eliminates the need for extra clock cycles to manage loop counters, improving efficiency in executing iterative operations.

18. What are the advantages of using a DSP processor over a general-purpose processor for signal processing?
DSP processors offer optimized performance for mathematical computations, real-time processing, energy efficiency, and specialized instruction sets designed for digital signal processing applications.

19. What is the role of on-chip memory in DSP processors?
On-chip memory provides fast access to frequently used data and instructions, reducing latency and improving processing speed.

20. How do interrupt controllers enhance DSP processor performance?
Interrupt controllers manage multiple interrupt sources efficiently, allowing DSP processors to respond quickly to high-priority tasks without significant processing delays.