DEPARTMENT OF ELECTRONICS AND COMMUNICATION ENGINEERING

IT6502 - DIGITAL SIGNAL PROCESSING (2013 regulation)

UNIT-1 SIGNALS AND SYSTEMS

PART-A

1. What is a continuous and discrete time signal?

Continuous time signal: A signal x(t) is said to be continuous if it is defined for all time t. Continuous time signal arise naturally when a physical waveform such as acoustics wave or light wave is converted into a electrical signal.

Discrete time signal: A discrete time signal is defined only at discrete instants of time. The independent variable has discrete values only, which are uniformly spaced. A discrete time signal is often derived from the continuous time signal by sampling it at a uniform rate.

2. Give the classification of signals?

Continuous-time and discrete time signals

Even and odd signals

Periodic signals and non-periodic signals

Deterministic signal and Random signal

Energy and Power signal

3. **What are the types of systems?**

Continuous time and discrete time systems

Linear and Non-linear systems

Causal and Non-causal systems

Static and Dynamic systems

Time varying and time in-varying systems

Distributive parameters and Lumped parameters systems

Stable and Un-stable systems.

4. What are even and odd signals?

Even signal: continuous time signal x(t) is said to be even if it satisfies the condition $x(t)=x(-t)$ for all values of t.

Odd signal: he signal $x(t)$ is said to be odd if it satisfies the condition $x(-t)=-x(t)$ for all t. In other words even signal is symmetric about the time origin or the vertical axis, but odd signals are anti-symmetric about the vertical axis.

5. What are deterministic and random signals?

Deterministic Signal: deterministic signal is a signal about which there is no certainty with respect to its value at any time. Accordingly, we find that deterministic signals may be modeled as completely specified functions of time.

Random signal: random signal is a signal about which there is uncertainty before its actual occurrence. (e.g.) The noise developed in a television or radio amplifier is an example for random signal.

6. What are energy and power signal?

Energy signal: Signal is referred as an energy signal, if and only if the total energy of the signal satisfies the condition $0<E<\infty$.

Power signal: Signal is said to be power signal if it satisfies the condition $0<P<\infty$.

7. What are elementary signals and name them?

The elementary signals serve as a building block for the construction of more complex signals. They are also important in their own right, in that they may be used to model many physical signals that occur in nature.

There are five elementary signals. They are as follows

Unit step function

Unit impulse function

Ramp function

Exponential function

Sinusoidal function

8.What are time invariant systems?

A system is said to be **time invariant** system if a time delay or advance of the input signal leads to an identical shift in the output signal. This implies that a time invariant system responds identically no matter when the input signal is applied. It also satisfies the condition R{x (n-k)} = y (n-k).

9. What do you mean by periodic and non-periodic signals?

A signal is said to be periodic if x (n + N) = x (n), Where N is the time period.

A signal is said to be non-periodic if x (n + N) = -x (n).

10. Define time variant and time invariant system.

A system is called **time invariant** if its output, input characteristics does not change with time. A system is called **time variant** if its input, output characteristics changes with time.

11. Define linear and non-linear system.

Linear system is one which satisfies superposition principle. Superposition principle:

The response of a system to a weighted sum of signals be equal to the corresponding weighted sum of responses of system to each of individual input signal.

i.e., T [a1x1(n)+a2x2(n)]=a1T[x1(n)]+a2T[x2(n)]

A system, which does not satisfy superposition principle, is known as non-linear system.

12. Define causal and non-causal system.

The 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 the future inputs.

e.g.:- y (n) =x (n)-x (n-1)

A system is said to be **non-causal** if a system does not satisfy the above definition.

13. Define causal LTI system.

The LTI system is said to be causal if h (n) = 0 for n < 0

14. Define stable LTI system.

The LTI system is said to be stable if its impulse response is absolutely summable.

15. What are the properties of convolution sum?

The properties of convolution sum are

Commutative property

The commutative law can be expressed as x (n) * h (n)=h (n) * x (n)

Associative law

The associative law can be expressed as [x (n) * h1 (n)] * h2 (n) = x (n) [h1 (n) * h2 (n)] Where x (n) – input, h1 (n), h2 (n) - impulse response

Distributive law

The distributive law can be expressed as

x (n) * [h1 (n) + h2 (n)] = x (n) * h1 (n) + x (n) * h2 (n)

16. Define Region of convergence

The region of convergence (ROC) of X (Z) is the set of all values of Z for which X (Z) attain final value.

17. State properties of ROC

The ROC does not contain any poles.

When x (n) is of finite duration then ROC is entire Z-plane except Z = 0 or Z = ∞

If X (Z) is causal, then ROC includes Z = ∞

If X (Z) is non-casual, then ROC includes Z = 0

18. State Sampling theorem.

A bandlimited continous time signal , with higher frequencies f_m Hz, can be uniquely recovered from its samples provided the sampling rate f>= 2fm samples per second.

19. What is an LTI System?

An LTI system is one who possesses two of the basic properties using linearity and time invariance.

20. What are the different types of signal representation?

a. Graphical representation
b. Functional representation
c. Tabular representation
d. Sequence representation.

<u>**PART-B**</u>

1. State and prove sampling theorem. How do you recover continuous signals from its samples? Discuss the parameters involved in sampling and reconstruction.
2. Compute the linear and circular convolution of h(n) = {**1,**2,1,1} and x(n) = { **1** , -3,0,2,2}
3. What is meant by energy and power signal? Determine whether the Signals are energy or power or neither energy nor power signals.

   $X(n)= (1/3)^n u(n)$

   $X(n)=\sin (\frac{\pi}{4}n)$

   $X(n)=e^{2n}u(n)$

   $X(n)=e^{j(\frac{\pi}{2}n+\frac{\pi}{4})}$
4. Check whether the following signals are linear, time invariant, Causal and stable

   Y(n)=x(n)+nx(n+1)

   Y(n)=cos x(n)
5. Discuss the different types of digital signal representation?
6. Find the cross correlation of two finite length sequences x(n) = {1, -1, 1, 2} and h(n) = {1, 2, 1, -1}
7. Find the impulse response given by difference equation. Y(n)-3y(n-1)-4y(n-2) = x(n)+2x(n-1)
8. Determine the system function and impulse response of the system described by the difference equation y(n) = x(n) +2x(n-1)- 4x(n-2) + x(n-3)

9. Find the impulse response of the system described by the difference equation y(n) = 0.7 y(n-1) -0.1 y(n-2) +2 x(n) – x(n-2)

10. Find inverse Z transform of X(Z) = $\frac{1}{1-1.5Z^{-1}+0.5\,Z^{\wedge}-2}$ using partial fraction or convolution method $|Z| > 1$, $|Z| < 1$

11. Find the inverse Z transform of X(Z) = $\frac{z2+z}{(z-1)(z-3)}$ ROC [z]>3 using Convolution method

UNIT-2 FREQUENCY TRANSFORMATIONS

PART-A

1.State the properties of DFT

Periodicity

Linearity and symmetry

Multiplication of two DFTs

Circular convolution

Time reversal

Circular time shift and frequency

shift Complex conjugate

Circular correlation

2. 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.

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

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

We can get better display of the frequency spectrum.

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

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

The two methods used for the sectional convolution are

1) The overlap-add method and 2) overlap-save method.

5. 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 cicular 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.

6. 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.

7. 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. Therefore, we use

FFT..

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

9. 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.

10. 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.

11. What is DIT algorithm?

Decimation-In-Time algorithm is used to calculate the DFT of an 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.

12. What is 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.

13. What are the applications of FFT algorithm?

The applications of FFT algorithm includes

Linear filtering

Correlation

Spectrum analysis

14. 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.

15. Distinguish between linear convolution and circular convolution of two sequences.

Linear convolution	Circular convolution
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.
It can be used to find the response of a linear filter	It cannot be used to find the response of a filter
Zero padding is not necessary to find the response of a linear filter.	Zero padding is necessary to find the response

16. 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.

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

Overlap-save method	Overlap-add method
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
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
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
To form the output sequence the first M-1 data points are discarded in each	To form the output sequence the last M-1 points from each output block is added

output block and the remaining data are fitted together	to the first M-1 points of the succeeding block

18. When the DFT X(K) of a sequence x(n) is imaginary?

If the sequence x(n) is real and odd (or) imaginary and even then X(K) is purely imaginary.

19. When the DFT X(K) of a sequence x(n) is real?

If the sequence x(n) is real and even (or) imaginary and odd then X(K) is purely Real.

20. Define DTFT.

Sampling is performed only in time domain. Continous function of w.

PART-B

1. Compute the DFT for the sequence {1, 2, 0, 0, 0, 2, 1, 1}. Using radix -2 DIF FFT and radix -2 DIT- FFT algorithm.
2. 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}. Using Overlap-add and overlap save method.
3. Obtain the 8 point DFT of sequence X(n) = $\begin{cases} 1, -3 \le n \le 3 \\ 0, \ otherwise \end{cases}$ Using DIT FFT algorithm.
4. In an LTI system the input x(n) = {1, 1, 1}and the impulse response h(n) = {-1, - 1 } Determine the response of LTI system by radix -2 DIT FFT
5. Derive and draw the radix -2 DIT algorithms for FFT of 8 points.
6. Prove if $X_3(k)=X_1(K)\,X_2(k)$ then $x_3(n)=\sum_{m=0}^{N-1} x_1(m)x_2((n-m))N$

7. Discuss the properties of DFT
8. Derive the expression for Decimation in time algorithm.
9. An 8 point sequence is given by x(n) ={0.5, 0.5, 0.5, 0.5, 0, 0, 0, 0} Compute 8 point DFT of x(n) by radix 2 DIT FFT algorithm.
10. Compute IDFT of the sequence by using DIT FFT algorithm.
    X(k) = {7, -0.707-j0.707, -j, 0.707-j0.707, 1, 0.707+j0.707, j, -0.707+j0.707}

UNIT-3 IIR FILTER DESIGN

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

Based on impulse response the filters are of two types

IIR filter

FIR filter

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

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

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

Based on frequency response the filters can be classified as

i. Lowpass filter

ii. Highpass filter

iii. Bandpass filter

iv. Bandreject filter

3. What are the advantages and disadvantages of FIR filters?

Advantages:

a. FIR filters have exact linear phase.

b. FIR filters are always stable.

c. FIR filters can be realized in both recursive and non recursive structure.

d. Filters with any arbitrary magnitude response can be tackled using FIR sequence.

Disadvantages:

For the same filter specifications the order of FIR filter design can be as high as 5 to 10 times that in an IIR design.

Large storage requirement is requirement

Powerful computational facilities required for the implementation.

4. How one can design digital filters from analog filters?

· Map the desired digital filter specifications into those for an equivalent analog filter.

· Derive the analog transfer function for the analog prototype.

· Transform the transfer function of the analog prototype into an equivalent digital filter transfer function.

5. Mention the procedures for digitizing the transfer function of an analog filter.

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

· Impulse invariance method.

· Bilinear transformation method.

. Approximation of derivatives

6. What do you understand by backward difference?

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

$$\frac{d}{dt}\ y(t) = y(nT) - y(nT-T)/T$$

The above equation is called backward difference equation.

7. What is the mapping procedure between S-plane & Z-plane in the method of mapping differentials? What are its characteristics?

The mapping procedure between S-plane & Z-plane in the method of mapping of differentials is given by

$$H(Z) = H(S)|S=(1-Z-1)/T$$

The above mapping has the following characteristics

· The left half of S-plane maps inside a circle of radius ½ centered at Z= ½ in the Zplane.

· The right half of S-plane maps into the region outside the circle of radius ½ in the Z-plane.

The j .-axis maps onto the perimeter of the circle of radius ½ in the Z-plane.

8. What is meant by impulse invariant method of designing IIR filter?

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. If the transfer function is of the form, 1/s-p, then

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

9. What is bilinear transformation?

The bilinear transformation is a mapping that transforms the left half of S-plane into the unit circle in the Z-plane only once, thus avoiding aliasing of frequency components. The mapping from the S-plane to the Z-plane is in bilinear transformation is

$$S=2/T(1-Z-1/1+Z-1)$$

10. What are the properties of bilinear transformation?

· The mapping for the bilinear transformation is a one-to-one mapping that is for every point Z, there is exactly one corresponding point S, and vice-versa.

· The j .-axis maps on to the unit circle |z|=1,the left half of the s-plane maps to the interior of the unit circle |z|=1 and the half of the s-plane maps on to the exterior of the unit circle |z|=1.

· **11. Define signal flow graph.**

· A signal flow graph is a graphical representation of the relationship between the variables of a set of linear difference equations.

12.Write a short note on pre-warping.

The effect of the non-linear compression at high frequencies can be compensated. When the desired magnitude response is piece-wise constant over frequency, this compression can be compensated by introducing a suitable pre-scaling, or pre-warping the critical frequencies by using the formula.

13. What are the advantages & disadvantages of bilinear transformation?

Advantages:

· The bilinear transformation provides one-to-one mapping.

· Stable continuous systems can be mapped into realizable, stable digital systems.

· There is no aliasing.

Disadvantage:

· The mapping is highly non-linear producing frequency, compression at high frequencies.

14.Distinguish analog and digital filters

Analog Filter	**Digital Filter**
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	Frequency response can be changed by

changing the components	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.What are the properties of chebyshev filter?

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

16.List the Butterworth polynomial for various orders.

N	**Denominator polynomial**
1	S+1
2	S^2+.707s+1
3	(s+1) (s2+s+1)
4	(s^2+.7653s+1) (s^2+1.84s+1)
5	(s+1 (s^2+.6183s+1) (s2+1.618s+1)
6	(s^2+1.93s+1) (s^2+.707s+1) (s^2+.5s+1)
7	(s+1 (s^2+1.809s+1) (s^2+1.24s+1) (s^2+.48s+1)

17.Differentiate Butterworth and Chebyshev filter.

Butterworth damping factor 1.44 chebyshev 1.06 Butterworth flat response damped response.

18.What is filter?

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

19. What is the main advantage of direct form II realization when compared to direct form I realization?

In direct form II realization the no of memory locations required is less than that of direct form I realization.

20. What is the advantage of cascade realization?

Quantization errors can be minimized if we realize an LTI system in cascade form.

PART-B

1. Apply impulse invariant transformation to H(S) =2/(S +1) (S + 2)
   With T =1sec and find H (Z).
2. For the constraints $0.8 \leq |H(\omega)| \leq 1.0$, $0 \leq \omega \leq 0.2\pi$

$$|H(\omega)| \leq 0.2, \quad 0.6\pi \leq \omega \leq \pi$$

With T= 1 sec determine the system function H(z) for a Butterworth filter using Bilinear transformation.

3. Design a digital Chebyshev filter satisfying the following constraints with T= 1 sec, using Bilinear transformation.

$$0.707 \leq |H(\omega)| \leq 1.0, \quad 0 \leq \omega \leq \pi$$

$$|H(\omega)| \leq 0.2, \quad 3\pi/4 \leq \omega \leq \pi$$

4. Obtain the direct form I, direct form II, cascade and parallel Realization for the system

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

5. Design a digital Butterworth filter satisfying the following constraints using impulse invariant transformation.

$$0.8 \leq |H(\omega)| \leq 1.0, \quad 0 \leq \omega \leq 0.2\pi$$

$$|H(\omega)| \leq 0.2, \quad 0.32\pi \leq \omega \leq \pi$$

6. Design a chebyshev filter using bilinear transformation method for

$$0.8 \leq |H(e^{jw})| \leq 1 \quad \text{for} \quad 0 \leq w \leq 0.2\pi$$

$$|H(e^{jw})| \leq 0.2 \quad \text{For} \quad 0.6\pi \leq w \leq \pi$$

7. Using bilinear transform design high pass filter, monotonic in passband with cutoff frequency of 1000 HZ and down at 350 HZ. The sampling frequency is 5000 Hz.
8. Design a butterworth filter using impulse invariant method for

$$0.8 \le \left|H(e^{jw})\right| \le 1 \quad \text{for} \quad 0 \le w \le 0.2\pi$$

$$\left|H(e^{jw})\right| \le 0.2 \quad \text{For} \quad 0.6\pi \le w \le \pi$$

9. Apply bilinear transformation to transfer function

$$H(s) = \frac{2}{(s+1)(s+2)} \text{ when T=2 sec.}$$

10. Explain the procedure for designing analog filters using chebyshev approximations with suitable example.

UNIT-4 FINITE IMPULSE RESPONSE FILTER DESIGN

PART-A

1. How phase distortion and delay distortion are introduced?

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

2. 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

3. 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)

4. **Give the advantages of FIR filter?**

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.

5. List the disadvantages of FIR FILTER

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.

6. Define 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.

7.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

8.For what kind of application, the anti-symmetrical impulse response can be used?

The anti-symmetrical impulse response can be used to design Hilbert transforms and differentiators.

9. 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., lowpass, highpass, bandpass and band reject. The symmetric impulse response having even number of samples can be used to design lowpass and bandpass filter.

10. Justify that that FIR filter is always stable?

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

11. What condition on the FIR sequence h(n) are to be imposed in order that this filter can be called a linear phase filter?

The conditions are

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

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

12.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.

13. What are the properties of FIR filter?

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

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

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

15. 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 .

16. Define Gibbs phenomenon? OR What are Gibbs oscillations?

One possible way of finding an FIR filter that approximates $H(e^{jw})$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.

17. Give the desirable characteristics of the windows?

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 lobes of the frequency response should decrease in energy rapidly w tends to p.

18. What is the necessary and sufficient condition for linear phase characteristics in FIR filter?

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

19.What are the advantages of Kaiser Widow?

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

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.

20. 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.

21. For what type of filters frequency sampling method is suitable?

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.

22. Give the equation specifying Hanning and Blackmann window

<u>Hanning Window</u>

$WH(n) = 0.5 + 0.5\cos 2\pi n/N-1, \quad (N-1)/2 \le n \le (N-1)/2$

$= 0, \qquad\qquad$ otherwise

<u>Blackmann Window</u>

$WB(n) = 0.42 + 0.5\cos 2\pi n/N-1 + 0.08\cos 4\pi n/N-1, \quad -(N-1)/2 \le n \le (N-1)/2$

$= 0, \qquad\qquad$ else

23. What do you understand by Linear Phase Response in filters?

If phase response of the filter is nonlinear the output signal is distorted one. In many cases Linear Phase filter is required throughout the passband of the filter to preserve the shape of the given signal within the pass band. An IIR filter cannot produce a linear phase. The FIR filter can give linear phase, when the impulse response of the filter is symmetric about its midpoint.

24. State Frequency Warping

Because of the non-linear mapping: the amplitude response of digital IIR filter is expand at lower frequencies and compressed at higher frequencies in comparison to the analog filter.

25. What is the importance of poles in filter design?

The stability of a filter is related to the location of the poles. For a stable analog filter the poles should lie on the left half of s-plane. For a stable digital filter the poles should lie inside the unit circle in the z-plane.

PART-B

1. Design and FIR LPF using frequency sampling method for the following specifications.
   Pass band frequency = 400 Hz
   Stop band frequency = 800 Hz
   Sampling frequency = 2000 Hz
   Order of the filter = 11
2. Realize the system function H(Z)=(2/3)Z+1+(2/3)Z─1 by linear phase FIR structure.
3. Design an ideal high pass filter with $Hd(e^{jw}) = 1$; $\pi/4 \le |w| \le \pi$ and 0 ; $|w| \le \pi/4$ Using Hamming window and Hanning window with N =11
4. Design a low pass filter using rectangular window by taking 9 samples of w(n) and with a cutoff frequency of 1.2 radians/sec.
5. i.Design an ideal differentiator with frequency response
   $H(e^{jw}) = jw \quad -\pi \le w \, \pi \le \pi$ and N=8 Using Rectangular window.
   ii. Write the Need and choice of windows
6. Using frequency sampling method, design a band pass FIR filter with the following specification. Sampling frequency Fs =8000 Hz, Cutoff frequency fc1 =1000Hz, fc2 =3000Hz.Determine the filter coefficients for N =7.
7. Determine the coefficients $\{h(n)\}$ of a linear phase FIR filter of length M =15 which has a symmetric unit sample response and a frequency response that satisfies the condition $H(\frac{2\pi K}{15}) =$ 1 for K =0,1,2,3
   0 for K = 4,5,6,7

8. Realize the system using for using linear phase realization

$$H(z) = 1 + \frac{1}{3} Z^{-1} + \frac{1}{4}Z^{-2} + \frac{1}{4} Z^{-3} + + \frac{1}{3} Z^{-4} + Z^{-5}$$

9. Design an ideal high pass filter with

$H_d(e^{jw}) = 1$; $\pi/4 \le |w| \le \pi$ and

0 ; $|w| \le \pi/4$ Using Hamming window with

N =11 and plot the magnitude response.

10. Design a low pass filter using rectangular window by taking 1 samples of w(*n*) with cut off sequence of 1-2 radians/sec , N =7.

UNIT-5 FINITE WORD LENGTH EFFECTS IN DIGITAL FILTERS

PART-A

1. Define 1's complement form?

In 1,s complement form the positive number is represented as in the sign magnitude form. To obtain the negative of the positive number, complement all the bits of the positive number.

2. What is meant by 2's complement form?

In 2's complement form the positive number is represented as in the sign magnitude form. To obtain the negative of the positive number, complement all the bits of the positive number and add 1 to the LSB.

3. Define floating pint representation?

In floating point form the positive number is represented as F =2CM,where is mantissa, is a fraction such that1/2<M<1and C the exponent can be either positive or negative.

4. Give the different quantization errors occurdue to finite word length registers in digital filters?

Input quantization errors

Coefficient quantization errors

Product quantization errors

5.What do you understand by input quantization error?

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

6 .Define product quantization error?

The product quantization errors arise at the output of the multiplier. Multiplication of a b bit data with a b bit coefficient results a product having 2b bits. Since a b bit register is used the multiplier output will be rounded or truncated to b bits which produce the error.

7.Mention the different quantization methods available for Finite Word Length Effects?

Truncation

Rounding

8.State truncation?

Truncation is a process of discarding all bits less significant than LSB that is retained

9. Define Rounding?

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

10. List the two types of limit cycle behavior of DSP?

Zero limit cycle behavior

Over flow limit cycle behavior

11.Mention the methods to prevent overflow?

Saturation arithmetic and

Scaling

12. Give the different types of arithmetic in digital systems

There are three types of arithmetic used in digital systems. They are fixed point arithmetic, floating point, block floating point arithmetic.

13. What is meant by fixed point number?

In fixed point number the position of a binary point is fixed. The bit to the right represent the fractional part and those to the left is integer part.

14. What are the different types of fixed point arithmetic?

Depending on the negative numbers representation, there are three forms of fixed point arithmetic. They are sign magnitude, 1's complement, 2's complement

15.Distinguish between fixed point and floating point arithmetic

S.No	Fixed Point Arithmetic	Floating Point Arithmetic
1.	Fast Operation	Slow Operation
2.	Relatively Economical	More expensive because of costlier hardware
3.	Overflow occurs in addition	Overflow does not arise
4.	Used in small computers	Used in large general purpose computers
5.	Small Dynamic Range	Increased dynamic range

16. Express the fraction 7/8 and -7/8 in sign magnitude, 1's and 2's complement.

<u>7/8</u>

Sign magnitude----------$(0.111)_2$

1's complement----------$(0.111)_2$

2's complement-----------$(0.111)_2$

<u>-7/8</u>

Sign magnitude-------$(1.111)_2$

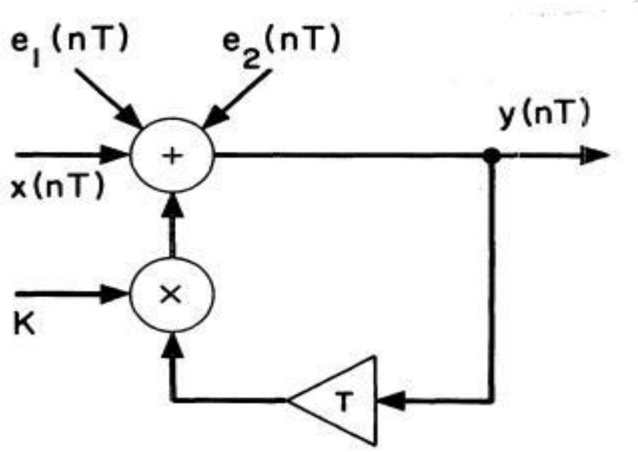
1's complement-------

--- $(1.000)_2$

2's complement--------(0.001)

--- 2

17. Draw the quantization noise model for a I order System



18. What do you understand by (Zero input) Limit cycle oscillations?

When a stable IIR filter digital filter is excited by a finite sequence, that is constant, the output will ideally decay to zero. However, the non-linearity due to finite precision arithmetic operations often causes periodic oscillations to occur in the output. Such oscillations occur in the recursive systems are called Zero input Limit Cycle Oscillation. Normally oscillations in the absence of output u (k) =0 by equation given below is called Limit cycle oscillations

Y[k] = -0.625.y [k-1] +u[k]

19. Determine Dead Band of the Filter.

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

20.Why Rounding is preferred to truncation in realizing digital filter?

i The quantization error due to rounding is independent of type arithmetic

ii The mean of rounding error is zero

iii The variance of rounding error is low

21.Define overflow oscillations

The overflow caused by adder makes the filter output to oscillate between maximum amplitude limits and such oscillations is referred as overflow oscillations

22. What is meant by sign magnitude representation?

For sign magnitude representation the leading binary digit is used to represent the sign. If it is equal to 1 the number is negative, otherwise it is positive.

PART-B

1. Explain the quantization process and errors introduced due to Truncation and Rounding?
2. With respect to finite word length effects in digital filter with examples discuss about
   i. Overflow limit cycle oscillations
   ii. Signal Scaling
3. Explain in detail about finite word length effects in the filter design.
4. Draw the Quantization model for 2^{nd} order system

   $H(Z) = \frac{1}{1-2r\cos\theta\, z^{\wedge -1} + r^{\wedge}2\, z^{\wedge -2}}$ and Find steady state output Noise Variance.

5. Explain in detail about Number Representations?
6. Discuss about Coefficient Quantization Error
7. Explain in detail about product quantization error.
8. Explain in detail about input quantization error with its steady state input and output noise power.
9. A digital filter is characterized by the difference equation y (n) = 0.85 y(n-1)+x(n). Determine the dead band system when x (0) = 0 and y (-1) = 13

10. The input to the system $y(n) = 0.999y(n-1)+x(n)$ is applied to an ADC. What is the power produced by quantization noise at the output of the filter if the input is quantized to a) 8 bits b) 16 bits.

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****ALL THE BEST****