

UNIT I**DISCRETE FOURIER TRANSFORM****Part A- 2marks****1. Compute the DFT of $x(n) = \delta(n-n_0)$? AU MAY/JUNE 11**

$$\begin{aligned} \text{Given } X(n) &= \delta(n-n_0) \\ \delta(n-n_0) &= 1 \text{ for } n=n_0 \\ &= 0 \text{ for } n \neq n_0 \\ X(k) &= \sum_{n=0}^{n=N-1} x(n) e^{-\frac{j2\pi nk}{N}} \\ &= \sum \delta(n-n_0) e^{-j2\pi nk/N} \\ &= e^{-j2\pi nk_0/N} \end{aligned}$$

2. Find the DFT of the following signals (i) $x(n) = \delta(n)$ (ii) $x(n) = a^n$ AU MAY 12

$$\begin{aligned} \text{(i) Given } x(n) &= \delta(n) \\ X(k) &= \sum x(n) e^{-j2\pi nk/N} \\ &= \sum \delta(n) e^{-j2\pi nk/N} \\ &= 1 \end{aligned}$$

$$\begin{aligned} \text{(ii) Given } x(n) &= a^n \\ X(k) &= \sum a^n e^{-j2\pi nk/N} \\ &= \sum (a e^{-j2\pi nk/N})^n \\ &= (1 - a^N e^{-j2\pi nk}) / (1 - a e^{-j2\pi nk/N}) \end{aligned}$$

3. Find the DFT of Non-causal 3-sample average? AU NOV/DEC 13

$$\begin{aligned} h(n) &= 1/3 \text{ for } -1 \leq n \leq 1 \\ &= 0 \text{ otherwise} \\ H(k) &= \sum h(n) e^{-j2\pi nk/N} \\ &= 1/3 [1 + e^{-j2\pi nk/N} + e^{-j2\pi nk(N-1)/N}] \\ &= (1 + 2 \cos(2\pi k/N))/3 \end{aligned}$$

4. Write the analysis and synthesis equations of DFT?

The analysis equation is the direct transform given by

$$X(K) = \sum_{n=0}^{N-1} x(n) e^{-\frac{j2\pi kn}{N}} \quad K=0,1,2,3,\dots,N-1$$

The synthesis equation is the direct transform given by

$$x(n) = \sum_{k=0}^{N-1} X(k) e^{\frac{j2\pi kn}{N}} \quad n=0,1,2,3,\dots,N-1$$

5. State Sampling theorem. AU MAY/JUNE 13

A band limited signals of finite energy which has no frequency component higher than f_m Hz is uniquely determined by its value at uniform interval less than $1/2f_m$ second apart.

$$f_s \geq 2f_m$$

where f_m =modulating frequency

f_s =sampling frequency

6. What is meant by aliasing. & What are the effects of aliasing AU NOV/DEC 12

When a band limited signal $x(t)$ is sampled with sampling frequency of $F_s < 2F_m$, then $X(j\Omega)$ is having spectral overlapping. i.e. high frequency interferes with low frequency and appears as low frequency then the phenomenon is called aliasing.

Since the high frequency interferes with low frequency then the distortion is generated.

The data is lost and it cannot be recovered.

7. Establish the relation between DFT and Z- transform.

$$X(z) = X(K) / z = e^{j2\pi nk/N}$$

8. What is Fast Fourier Transform(FFT)?

The Fast Fourier Transform(FFT) is an algorithm used to compute the DFT. It makes use of symmetry and periodicity properties of the twiddle factor W_N^k to effectively reduce the DFT computation time.

9. Why FFT is needed? AU NOV/DEC 13

Direct computation of DFT using formula needs more computation time

ie). DFT using the formula $X(k) = \sum x(n) e^{-j2\pi kn/N}$ requires N^2 complex multiplications and $N(N-1)$ complex additions.

Thus for eg: $N = 16$, complex multiplication = 256 & complex addition = 240 using DFT method

If we compute using radix 2 FFT algorithm

$N/2 \log_2 N$ complex multiplications and $N \log_2 N$ complex additions.

For : $N = 16$, complex multiplication = 32 & addition = 64 using FFT method

Therefore Reduces the computation complexity and calculate fast and efficient and removing the redundant parts of the computation, saving roughly a factor of two in time and memory

10. What is the speed improvement factor in calculating 64-point DFT of a sequence using direct computation and FFT algorithm? AU NOV/DEC 13 & AU MAY JUNE 13

The Number of complex multiplications required using DFT (direct computation) is

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

The number of complex multiplications required using FFT is

$$N/2 \log_2 N = 64/2 \log_2 64 = 192.$$

Speed improvement factor = $4096/192 = 21.33$

11. What is meant by radix-2 FET?

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

12. What are the difference and similarities between DIT and DIF algorithm? AU NOV/DEC 12

Differences

1. For DIT, the input is bit reversed while the output is in natural order, whereas for DIF, the input is in natural order while the output is bit reversed.
2. In DIT butterfly is complex multiplication of phase factor is before the add subtract operation. In DIF complex multiplication takes place after the add-subtract operation in DIF.

Similarities

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

13. What do you mean by the term "bit reversal" as applied to FFT

In DIT algorithm we can find that for the output sequence to be in a natural order (i.e., $X(k)$, $k=0,1,2,\dots,N-1$) the input sequence has to be stored in a shuffled order. For an 8-point DIT algorithm the input sequence is in the order $x(0), x(4), x(2), x(6), x(1), x(5), x(3)$ and $x(7)$. We can see that when N is a power of 2, the input sequence must be stored in bit-reversal order for the output to be computed in a natural order. For $N = 8$ the bit-reversal process is shown in table.

Input sample Index	Binary representation	Bit reversed binary	Bit reversed sample index
0	000	000	0
1	001	100	4
2	010	010	2
3	011	110	6
4	100	001	1
5	101	101	5
6	110	011	3
7	111	111	7

14. What are the applications of FFT algorithm? AU NOV/DEC 12

The applications of FFT algorithm includes

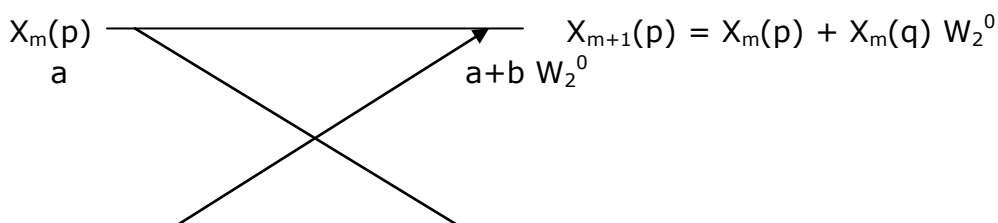
- (i) Linear filtering
- (ii) Correlation and pattern recognition
- (iii) Spectrum analysis
- (iv) Image analysis, image filtering, image reconstruction and image compression

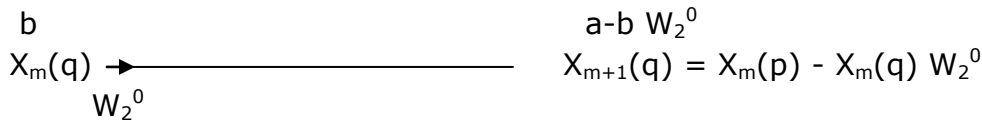
15. Why the computations in FFT algorithm is said to be in place? AU May/June 12

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.

16. Draw the flow graph of a two-point radix-2 DIT-FFT. AU MAY/JUNE 13

The Flow graph of a two-point DFT for a Decimation-in-time frequency algorithm is

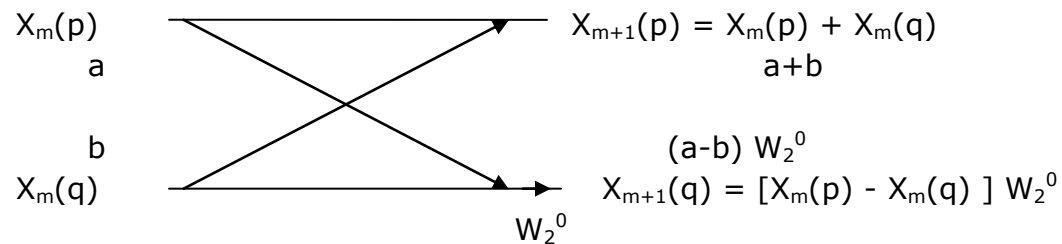




Where $X_m(p)$ and $X_m(q)$ are inputs, $X_{m+1}(p)$ and $X_{m+1}(q)$ are outputs of the butterfly. The nodes p and q represents memory locations.

17. Draw the basic butterfly diagram for DIF algorithm. AU NOV/DEC 12

The basic butterfly diagram for DIF algorithm is



Where $X_m(p)$ and $X_m(q)$ are inputs, $X_{m+1}(p)$ and $X_{m+1}(q)$ are outputs of the butterfly. The nodes p and q represents memory locations.

18. What is zero padding? What are its uses? AU NOV/DEC 13

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.

19. Distinguish between DFT and DTFT.

S.N	DFT	DTFT
1	Obtained by performing sampling operation in both the time and frequency domains.	Sampling is performed only in time domain.
2	Discrete frequency spectrum	Continuous function of ω
3	DFT gives only positive frequency values and gives less number of frequency components than DTFT	DTFT gives both positive and negative frequency values.

20. How will you perform linear convolution via circular convolution? (May 2006)

Let the length of $x(n)$ be L , length of $h(n)$ be M . then linear convolution of $x(n)$ and $h(n)$ can be obtained through following steps.

- i. Append $x(n)$ with $M-1$ zeros. Hence its length will be $L+M-1$
- ii. Append $h(n)$ with $L-1$ zeros. Hence its length will be $L+M-1$
- iii. Perform circular convolution of above sequences. The result is linear convolution of length $L+M-1$

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

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 contain $N = L + M - 1$ samples.	If $x(n)$ is a sequence of L number of samples and $h(n)$ with M number of samples, after convolution $y(n)$ will contain $N = \text{Max}(L, M)$ samples.
2. Linear convolution can be used to find the response of a linear filter.	Circular convolution cannot be used to find the response of a linear 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 linear filter.

22. What is meant by sectioned 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 into smaller sections. These sections are processed separately one at a time and combined later to get the output.

23. What are the two methods used for the sectioned convolution?

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

24. Compare Overlap Add and Overlap Save method

S.N	Overlap Add method	Overlap Save method
1	The size of the input data block is L	The size of the input data block is $N = L + M - 1$
2	Each data block is L point and we append $M - 1$ zeros to compute N point DFT	Each data block consists of the last $M - 1$ data points of previous data block followed by L new data point
3	In this no corruption due to aliasing, as linear convolution is performed using circular convolution.	In each output block $M - 1$ points are corrupted due to aliasing, as circular convolution is employed.
4	To get output sequence, the last $M - 1$ points from each output block is added to first $M - 1$ points of succeeding block.	To get output sequence, the first $M - 1$ points from each output block is data are fitted together.

25. What are the steps involved in calculating convolution sum?

The steps involved in calculating sum are

- Folding
- Shifting
- Multiplication
- Summation

PART B

1. State and prove the properties of DFT transform

2. Determine the response of FIR system with the impulse response $h(n) = \{-1, -1, 1, 1\}$ to the input sequence $x(n) = \{1, 2, -1, -3\}$ by means of DFT and IDFT. Verify the answer by linear convolution.
3. Compute DFT of the sequence whose values is $x(n) = \{1, 1, -2, -2\}$ and draw the spectrum of $x(k)$
4. Compute 8 point DFT the sequence $x(n) = \{2, 2, 2, 2, 1, 1, 1, 1\}$
5. Find the DIT FFT of the $x(n) = \{1, 2, 3, 4, 4, 3, 2, 1\}$ and also using DIF FFT
6. Compute the 8 point DFT of the sequence $x(n) = \begin{cases} 1 & 0 \leq n \leq 7 \\ 0 & \text{otherwise} \end{cases}$
7. Compute the 8 point DFT of the sequence $x(n) = \begin{cases} 1 & -3 \leq n \leq 3 \\ 0 & \text{otherwise} \end{cases}$
8. Derive the N point Radix 2 DIT FFT algorithm and also draw the flow or computation diagram?
9. Derive the N point Radix 2 DIF FFT algorithm and also draw the flow diagram?
10. Compute DFT for the basic discrete signals?
11. using linear convolution find $y(n) = x(n) * h(n)$ for the sequences $x(n) = \{1, 2, -1, 2, 3, -2, -3, -1, 1, 1, 2, -1\}$ and $h\{1, 2\}$. Compare the result by solving the problem using i). Overlap add ii). Overlap save method.

Unit 2- INFINITE IMPULSE RESPONSE FILTERS

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

1. IIR-infinite impulse response
2. FIR-finite impulse response

2. Distinguish between FIR & IIR?

AU Nov/Dec 13

FIR	IIR
i) Easily designed to have perfect linear phase.	These filter do not have linear phase.
ii) FIR filter can be realized in both recursive & non-recursive structure.	IIR filter are easily realized with recursive structure.
iii) Greater flexibility to control the shape of their magnitude response.	Less flexibility, limited to specific kind of filter.
iv) Error due to round off noise are less severe in FIR filters, mainly because feedback is not used.	Round off noise is more.

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

Based on frequency response the filters can be classified as

1. Lowpass filter
2. Highpass filter
3. Bandpass filter
4. Bandreject filter

4. Give any two properties of butterworth filters?

1. The magnitude response of the butterworth filter decreases monotonically as frequency Ω increases from 0 to ∞ .
2. The magnitude response is closely approximates the ideal response as the order N increases.

3. The poles of the filter lie on a circle.

5. What are the properties of chebyshev filter?

1. The magnitude response of the chebyshev filter exhibits ripples either in pass band or in stop band according to type. Ripples in pass band are type I filter. Ripples in stop band are type II filter.
2. The poles of the chebyshev filter lies on an ellipse.

6. Distinguish between butterworth & chebyshev filter?

Butterworth filter	Chebyshev filter
The magnitude response decreases monotonically as frequency Ω increases	The magnitude response exhibits ripples in pass band & monotonically decreases in stop band.(type I)
Transition band is more	Transition band is less
Poles lies on a circle	Poles lies on ellipse.
Order is more for same specification	Order is less .

7. Mention any two procedures for digitizing the transfer function of a analog filter?

AU NOV/DEC 13

1. Impulse invariance method.
2. Bilinear transformation.

8. Define bilinear transformation?

Bilinear transformation is a conformal mapping that transforms the $j\Omega$ axis into the unit circle in the z plane only once, thus avoiding aliasing of frequency components.

The transformation for the s plane to z plane is $s = \frac{2}{T} \left[\frac{1 - z^{-1}}{1 + z^{-1}} \right]$

9. What are the requirements for converting a stable analog filter into a stable digital filter?

1. The $j\Omega$ axis in the s plane should map into the unit circle in the z plane. This gives a direct relationship between the two frequency variables in the two domains.
2. The left half plane of the S plane should map into the inside of the circle.

10. What is Warping effect? What is prewarping AU NOV/DEC 12 AU May/June

The relation between the analog & digital frequency in bilinear transformation is given by

$$\Omega = 2/T \tan(\omega/2)$$

for smaller values of ω there exist linear relationship between ω & Ω . But nonlinear relation for larger values. This nonlinear introduces distortion in frequency axis. This called as Warping effect. This compresses the magnitude & phase response at high frequency.

Prewarping: 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.

11. Define ripples in a filter?

The limits of the tolerance in the magnitude of passband and stopband are called ripples. The tolerance in passband is denoted as δ_p and that in stopband is denoted as δ_s .

12. 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.
- Neither the impulse response nor the phase response of the analog filter is preserved in a digital filter obtained by bilinear transformation.

13. What are desirable & undesirable features/ Characteristics of FIR filter? Or what are advantages & disadvantages of FIR filter? AU May/June 13

Advantage:

- i) FIR filter have exact linear phase.
- ii) FIR filter are always stable. since all its poles of the filter are at the origin.
- iii) FIR filter can be realized in both recursive & non-recursive structure.
- iv) Filters with any arbitrary magnitude response can be tackled using FIR. sequency.

Disadvantage:

- i) For the same filter specification the order of FIR filter design can be as high as 5 to 10 times that in a IIR design.
- ii) Large storage requirements needed.
- iii) Powerful computational facilities required for the implementation.

14. What are the design techniques of FIR filter?

Fourier series method (not used frequently)

Window method

Frequency sampling method

Optimal filter design

15. What is meant by FIR filter and why is it stable?

The filters whose impulse response is finite are referred as FIR filters. These are usually implemented using nonrecursive structures.

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

16. What are the conditions to be satisfied for constant phase delay in linear phase FIR filter? AU NOV/DEC 13

The conditions for constant phase delay are

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

Impulse response, $h(n) = h(N-1-n)$ (ie, impulse response is symmetric).

17. What do you mean by linear phase response? AU May/June 12

For linear phase filter $\theta(\omega) \propto \omega$. The linear phase filter did not alter the shape of the original signal. If the phase response of the filter is not linear, the output signal may be distorted one. Linear phase characteristic is required throughout the pass band of the filter to preserve the shape of a given signal within the pass band. (IIR cannot produce linear phase filter) .

18. Write the procedure for designing FIR filter using windows?

- i) Choose the desired frequency response of the filter $h_d(\omega)$
- ii) Take inverse Fourier transform of $h_d(\omega)$ to obtain the desired impulse response $h_d(n)$.
- iii) Choose a window sequence $w(n)$ and multiply $h_d(n)$ with $w(n)$ to convert the infinite duration impulse response to finite duration impulse response $h(n)$.

PART – B

- 1. a) What are the main features of IIR filter. (4)
- b) Determine the order of the chebyshev filter to meet the following specifications. (8)

$$\begin{aligned} 0.85 \leq |H(e^{j\omega})| \leq 1 & \quad 0 \leq \omega \leq 0.2\pi \\ |H(e^{j\omega})| \leq 0.3 & \quad 0.5 \leq \omega \leq \pi \end{aligned}$$

- 2. The specification of the desired filter is

$$\begin{aligned} 0.707 \leq |H(e^{j\omega})| \leq 1.0 & \quad 0 \leq \omega \leq \pi/2 \\ |H(e^{j\omega})| \leq 0.2 & \quad 3\pi/4 \leq \omega \leq \pi \end{aligned}$$

Design an butterworth filter using Bilinear transformation $T=1$ sec.

- 3. Derive the equation for calculating the order of the butterworth filter.& using mpulse invariant method convert analog to digital($T=0.2$ sec) $H(s) = 2/(s+1)(s+2)$
- 4. Explain the design of chebyshev filter of type I & type II. Plot the frequency response of both filter.
- 5. Design a butterworth HPF satisfying $\alpha_p \leq 0.5\text{dB}$ $\alpha_s \geq 30\text{ dB}$ $f_p = 0.32\text{Hz}$ $f_s = 0.16\text{Hz}$ $F=1\text{ Hz}$.
- 6. Derive the expression for a linear phase FIR filter with symmetric impulse response?

7. Design an FIR digital filter to approximate an ideal LPF with pass band gain of unity, cut off frequency of 850 Hz & working at a sampling frequency of $F_s = 5\text{KHZ}$. The length of the impulse response should be 5. use rectangular window

- 8. The desired response of a LPF is

$$H_d(e^{j\omega}) = \begin{cases} e^{-j3\omega} & -3\pi/4 \leq \omega \leq 3\pi/4 \\ 0 & 3\pi/4 < |\omega| \leq \pi \end{cases}$$

Determine the frequency response of the filter for $M=7$ using hamming window.

- 9. Design an ideal LPF with frequency response is

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

Using hanning window with $N=11$.

UNIT – III FINITE IMPULSE RESPONSE FILTERS

1. What is Gibb’s phenomenon? AU NOV/DEC 12

One possible way of finding an FIR filter that approximates $H_d(e^{j\omega})$ would be to truncate the infinite Fourier series at $n=\pm(N-1)/2$. Abrupt truncation of the series will lead to oscillation both in pass band & in stop band. This phenomenon is called Gibb's phenomenon

2. Give the steps in the design of a digital filter from analog filter? AU NOV/DEC 13

1. Map the desired digital filter specifications into those for an equivalent analog filter.
2. Derive the analog transfer function for the analog prototype
3. Transform the analog transfer function into an equivalent digital filter transfer function

3. What are the disadvantages of Fourier series method and why we use windowing method for filter design?

In designing FIR filter using Fourier series method the infinite duration impulse response is truncated at $n = \pm(N-1)/2$. Direct truncation of the series will lead to fixed percentage overshoots and undershoots before and after an approximated discontinuity in the frequency response.

While designing using window methods the oscillation is reduced.

4. 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 identified as DFT coefficients. The filter coefficients are then determined as IDFT of this set of samples.

5. For what type of filter 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.

6. What is type I and Type II Chebyshev approximation?

In chebyshev approximation, the approximation function is selected such that the error is minimized over a prescribed band of frequencies.

In type-I chebyshev approximation, the error function is selected such that, the magnitude response is equiripple in the passband and monotonic in the stopband.

In type-II chebyshev approximation, the error function is selected such that, the magnitude response is monotonic in the passband and equiripple in the stopband. The type-II magnitude response is called inverse Chebyshev response.

PART B

1. Determine the coefficients of a linear phase filter of length 13 which has a symmetric unit sample response and a frequency response that satisfies the condition.

$$\begin{aligned} H(k) &= 1, k=0 \text{ to } 2 \\ &= 0, k=3 \text{ to } 5 \end{aligned}$$

2. Determine the coefficients of a linear phase filter $M= 15$ which has a symmetric unit sample response and a frequency response that satisfies the condition.

$$\begin{aligned} H(2\pi k/15) &= 1, k=0,1,2,3 \\ &= 0, k=4,5,6,7 \end{aligned}$$

3. Using frequency sampling method, design a BPF with the specifications

Sampling frequency $F= 8000\text{Hz}$

Cutoff frequencies $f_{c1} = 1000 \text{ Hz}$, $f_{c2} = 3000\text{Hz}$. Determine the filter coefficients for $N=7$.

4. The desired response of a BPF is

$$H_d(e^{j\omega}) = \begin{cases} 1 & \pi/4 \leq \omega \leq 3\pi/4 \\ 0 & \text{otherwise} \end{cases}$$

Determine the frequency response of the filter for M=11 using hamming window

5. Explain the type 1 design and type 2 design of FIR filter using frequency sampling techniques?

6. Compare FIR and IIR filters.

7. Discuss the steps in the design of IIR filter using Bilinear transformation for any one type of filter?

8. Discuss the steps in the design of IIR filter using impulse invariant method for any one type of filter?

UNIT- 4 FINITE WORD LENGTH EFF

Part A- 2 MARKS

1. What is meant by finite word length effects in digital filter mean?

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The fundamental operation in digital filters is multiplication and addition. When these operations are performed in digital system the input data as well as the sum product (o/p data) have to be represented in finite word length, which depends on the size (length) of the register to store the data. In digital computation the i/p & o/p data (sum & product) are quantized by rounding or truncation to convert them to a finite word size. This creates error (noise) in the o/p or creates oscillations (limit cycles) in the o/p. These effects due to finite precision representation of numbers in a digital system are called finite word length effects.

2. List some of the finite word length effects in digital filter.

- 1.Errors due to quantization of input data.
- 2.Errors due to quantization of filter coefficient.
- 3.Errors due to rounding the product in multiplications.
- 4.limit cycles due to product quantization and over flow in addition.

3. What are the three quantization errors due to finite word length registers in digital filters?

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

4. What are the different quantization methods?

The common methods of quantization are 1.Truncation 2. Rounding.

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 ADC. The representation of continuous signal amplitude by a fixed digit produce an error, which is known as input quantization error. The input quantization errors arise due to A/D conversion. Ie. In digital computation the filter coefficients are represented in binary and are stored in registers. If a b bit register is used the filter coefficients must be rounded or truncated to b bits ,which produces an error.

6. What is coefficient inaccuracy? (Or) What is coefficient quantization error? What is its effect?

The filter coefficients are computed to infinite precision in theory. But, in digital computation the filter coefficients are represented in binary and are stored in registers.

If a b bit register is used, the filter coefficients must be rounded or truncated to b bits, which produces an error.

Due to quantization of coefficients, the frequency response of the filter may differ appreciably from the desired response and sometimes the filter may actually fail to meet the desired specifications. If the poles of desired filter are close to the unit circle, then those of the filter with quantized coefficients may lie just outside the unit circle, leading to instability.

7. What is product quantization error? Or what is product round off error in digital signal processing?

Product quantization errors arise at the output of a 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 must be rounded or truncated to b bits, which produces an error. This error is known as product quantization error.

8. What is steady state noise power due to quantization if the number of bits is b? AN.U 2005

$$\sigma_e = 2^{-2b} / 12$$

9. What is truncation? AN.U.MAY 2005

The truncation is the process of reducing the size of binary numbers by discarding all bits less significant than the least significant bit that is retained. (In truncation of a binary number to b bits all the less significant bits beyond b^{th} bit are discarded)

10. What is Rounding? What is upward rounding?

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

In upward rounding of a number to b-bit s by retaining the most significant b-bits. If the bit next to the least significant bit that is retained is zero, then zero is added to the least significant bit of the truncated number. If the bit next to the least significant bit that is retained is one then one is added to the least significant bit of the truncated number.

11. What are the errors generated by A/D process?

The A/D process generates two types of errors. They are **quantization error and saturation error**. The quantization error is due to representation of sampled signal by a fixed number of digital levels (quantization levels). The saturation errors occur when analog signal exceeds the dynamic range of A/D converter.

12. Why errors are created in A/D process?

In A/D process the analog signals are sampled and converted to binary. The sampled analog signals will have infinite precision. In binary representation of b-bit we have 2^b different values with finite precision. The 2^b binary values are called quantization levels. Hence the samples of analog signals are quantized in order to fit into any of the quantized levels. This quantization process introduces errors in the signal.

13. How the digital filters are affected by quantization of filter co-efficient? Or what is the effect of quantization on pole locations? AU dec 13

The quantization of the filter co-efficient will modify the values of poles & zeros and so the location of poles and zeros will be shifted from the desired location. This will create deviations in the frequency response of the system. Hence the resultant filter will have a frequency response different from that of the filter with unquantized coefficient.

14. How the sensitivity of the filter frequency response to the filter coefficient is minimized?

The sensitivity of the filter frequency response to the filter coefficient is minimized by realizing the filter having a large number of poles and zeros as an interconnection of second order sections. Hence the filter can be realized in which the basic building block are first order and second order sections.

15. What is the drawback in saturation arithmetic?

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

16. Why rounding are preferred than truncation for quantizing the product?

In digital system the product quantization is performed by rounding due to the following desirable characteristics of rounding.

- i) The rounding error is independent of the type of arithmetic.
- ii) The mean value of rounding error signal is zero.
- iii) The variance of the rounding error signal is least.

17. Define noise transfer function (NTF)?

The noise transfer function (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.

18. What are the assumption made by regarding the statistical independence of the various noise source s in the digital filter?

The assumption made by regarding the statistical independence of the various noise sources in the digital filter are,

- i) Any two different samples from the same noise source are uncorrelated.
- ii) Any two different noise source, when considered, as random processes are uncorrelated.
- iii) Each noise source is uncorrelated with the input sequence.

19. What are limit cycles? AN.U. 2002

In recursive system when input is zero or some non-zero constants value, the non-linearities due to finite arithmetic operation may cause periodic oscillation in the output. These oscillations are called limit cycles.

20. What are the two types of limit cycles?

- i) Zero input limit cycles
- ii) Overflow limit cycles.

21. What is zero input limit cycles? AN.U. 2003

In recursive system, the product quantization may create periodic oscillation in the output. These oscillations are called limit cycles. If the system output enters a limit cycles, it will continue to remain in limit cycles even when the input is made zero. Hence these input limit cycles are also called as zero input limit cycles.

22. How the system o/p can be brought out of the limit cycle?

The system o/p can be brought out of the limit cycle by applying an input of large magnitude, which is sufficient to drive the system out of the limit cycles

23. What is saturation arithmetic?

In saturation arithmetic when the result of the arithmetic operations exceeds the dynamic range of number system, then the result is set to maximum or minimum possible value. If the upper limit is exceeded then the result is set to maximum possible values. If the lower limit is exceeded then the result is set to the minimum possible values.

24. What is over flow limit cycles? AN.U MAY2005

In fixed-point addition the over flow occurs when the sum exceeds the finite word length of the register to store the sum. The over flow in addition may lead to oscillations in o/p which is called over flow limit cycles.

(Or) overflow caused by the adder make the filter output to oscillate between maximum amplitude limits.

25. How over flow limit cycles are eliminated? AN.U MAY2004

The over flow cycles are eliminated either by using saturation arithmetic or by scaling the input cycle to the adder.

26. Define "dead band" of the filter.

The quantization effect in multiplication makes filter to lock in unstable mode in which limit cycle occur. The amplitudes of the output during a limit cycle are confined to a range of values called the dead band of the filter.

$$K = \frac{2^{-b}/2}{1 - |a|}$$
 Here $[-k,+k]$ is known as dead band , $b =$ number of quantization bits.

27. What are the different types of arithmetic in digital systems?

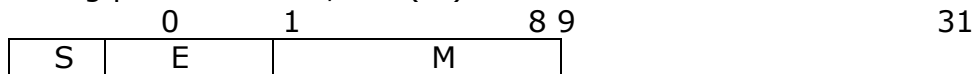
There are three types of arithmetic used in digital systems

- Fixed point arithmetic
- Floating point arithmetic
- Block Floating arithmetic

28. Give the IEEE-754 standard format for 32-bit floating-point numbers.

The IEEE-754 standard for 32-bit single precision floating point number is given by

Floating point numbers, $N_f = (-1)^s * 2^{E-127} * M$



S=1-bit field for sign of number.

E=8-bit field for exponent.

M=23-bit field for mantissa.

29. What do you understand by a Fixed-point number?

In Fixed point arithmetic the position of the binary point is fixed. The bit to the right represent the fractional part of the number and those to the left represent the integer part. For example, the binary number 01.1100 has the value 1.75 in decimal.

30. What are the different types in fixed-point number representation?

Depending on the way negative numbers are represented, there are three different forms of fixed-point arithmetic. They are 1. Sign magnitude 2. 1's complement 3. 2's complement.

31. What are the advantages of floating point arithmetic?

1. Larger dynamic range.
2. Overflow in floating point representation is unlikely.

32. Compare the fixed point and floating-point arithmetic.

S.N	Fixed Point Arithmetic	Floating point arithmetic
1.	Fast Operation	Slow Operation
2.	Small dynamic range.	Increased dynamic range.
3.	Round off errors occur only for addition.	Round off errors can occur with both addition & multiplication.
4.	Overflow occur in addition	Overflow does not arise.

16 MARKS

1. Explain briefly the process of limit cycle oscillation.
2. The output of an A/D converter is applied to a digital filter with the system function $H(Z) = 0.5z / (z-0.5)$. Find the output noise power from the digital filter when the input signal is quantized to have 8 bits.
3. Derive the steady state noise power at the output of an LTI system due to the Quantization at the input.
4. Explain about the fixed & floating representation?
5. Explain the coefficient quantization effect in direct form realization of IIR filter?
6. A digital system is characteristic by difference equation $y(n) = 0.9y(n-1) + x(n)$
Determine the dead band of the system. When $x(n) = 0$ $y(-1) = 12$
7. Draw the quantization noise model for II order function
 $H(Z) = 1 / (1 - 2r\cos\theta \cdot z^{-1} + r^2 z^{-2})$. Find the steady state output noise variance.
8. Derive the expression for signal scaling or Get the expression of a scalar, which is used in digital systems to avoid over flow?
9. For a digital network shown in fig. Find $H(Z)$ & scaling S_0 to avoid over flow in the register A_1 .
10. Consider the transfer function $H(Z) = H_1(Z) \cdot H_2(Z)$ where
 $H_1(Z) = 1 / (1 - a_1 z^{-1})$ $H_2(Z) = 1 / (1 - a_2 z^{-1})$.
Find the output round off noise power. Assume $a_1 = 0.5$ & $a_2 = 0.6$.
11. Explain briefly the different types of quantization errors that occur in digital systems?
12. Two first order LPF whose system functions are given below are connected in cascade. Determine the overall o/p noise power $H_1(Z) = 1 / (1 - 0.9 z^{-1})$ $H_2(Z) = 1 / (1 - 0.8 z^{-1})$ with $b=3$.
13. Describe the quantization errors that occur in rounding and truncation in two's complement?
14. What is meant by truncation? Explain the error that arises due to truncation in floating point number.

UNIT V INTRODUCTION TO DIGITAL SIGNAL PROCESSORS

Part A- 2 Marks

1) Give some example of DSP

[Audio signal processing](#), [audio compression](#), [digital image processing](#), [video compression](#), [speech processing](#), [speech recognition](#), [digital communications](#), [RADAR](#), [SONAR](#), [Financial signal processing](#), [seismology](#) and [biomedicine](#)

2) What is the "interpolation factor"?

The interpolation factor is simply the ratio of the output rate to the input rate. It is usually symbolized by "L", so output rate / input rate=L.

3) Why interpolate?

The primary reason to interpolate is simply to increase the sampling rate at the output of one system so that another system operating at a higher sampling rate can input the signal.

4) What is multirate signal processing?

Digital signal processing system handles processing at multiple sampling rates and then it is called multi-rate signal processing.

5) Define Downsampling or Decimation?MAY 2012

Down sampling a sequence $x(n)$ by a factor M is the process of picking every Mth sample and discarding the rest.

6) Define Up sampling or Interpolation?MAY 2013

Up sampling by a factor L is the process of inserting L-1 zeros between two consecutive samples.

7) What is know to be subband coding?

The input signal is first split into number of non-overlapping frequency bands by band pass filters. The output of each band pass filter is decimated or down sampled by a factor M. this can be achieved by retaining every Mth sample of the filter output and discarding M-1.

8) Define sampling rate conversion

Sample rate conversion is the process of converting a (usually digital) signal from one sampling rate to another, while changing the information carried by the signal as little as possible. When applied to an image, this process is sometimes called image scaling.

9) What are the two techniques of sampling rate conversion? NOV2012

i) D/A conversion and resampling at required rate.

ii) Sampling rate conversion in digital domain (multi-rate processing)

10) What are the applications of sampling rate conversion? MAY 2013

i) Narrow band filters.

ii) Quadrature mirror filters.

iii) Digital filter banks

11) Define speech compression and decompression. MAY2012

Speech analysis by a vocoder becomes the compression and synthesis by a vocoder becomes decompression. Vocoder extracts the spectral envelope of speech and information regarding voicing and pitch. This data is coded and transmitted. The synthesizer generates speech from the received data.

12) Writ the principle of adaptive filters. NOV 2013

The coefficients of the filter are changed automatically according to the changes in input

signal. This means the filtering characteristics of the adaptive filter are changed or adapted

according to the changes in input signal.

13. How the image enhancement is achieved using DSP? NOV 2013

- i) Local neighbourhood operations as in convolution.
- ii) Transform operations as in DFT.
- iii) Mapping operations as in pseudo coloring and gray level mapping

14) Define image enhancement ?

Enhancement is an image processing filter that enhances the edge contrast of an image or video in an attempt to improve its acutance (apparent sharpness).

15) Define vocoders

A synthesizer that produces sounds from an analysis of speech input.

16) Explain adaptive filtering

An **adaptive filter** is a filter that *self-adjusts* its transfer function according to an optimizing algorithm. Because of the complexity of the optimizing algorithms, most adaptive filters are digital filters that perform digital signal processing and adapt their performance based on the input signal. By way of contrast, a non-adaptive filter has static filter coefficients (which collectively form the transfer function).

17) Application of adaptive filter

- Noise cancellation
- Signal prediction
- Adaptive feedback cancellation
- Echo cancellation

18) Define ANC?

Active noise control (ANC) (also known as **noise cancellation**, **active noise reduction (ANR)** or **antinoise**) is a method for reducing unwanted sound.

19) Define Linear prediction?

Linear prediction is a mathematical operation where future values of a discrete-time signal are estimated as a linear function of previous samples.

20) Prove that up sampling by a factor M is time varying system.

When a system is upsampled by a factor M is the process of inserting M-1 zeros between the two consecutive samples. Hence the system will always be a time varying systems.

21) Give any two image enhancement methods.

Spatial domain method: The techniques that operate directly on the pixels are known as the spatial domain methods. The methods are contrast sketching, thresholding, frequency domain method.

Frequency domain method: Frequency domain refers to the fourier transform and there techniques are based on the modifying the fourier transform of an image.

16 MARKS

- 1) Explain the concept of decimation by a factor D and interpolation by factor I
- 2) With help of equation explain sampling rate conversion by a rational factor I/D
- 3) Explain the following application

- i) speech compression
- ii) sound processing
- 4) With help diagram explain adaptive filtering process
- 5) Explain speech vocoders and sub band coding
- 6) Explain how image enhancement was achieved
- 7) For the system $H(z)=1/(1-az^{-1})$. Determine $P_0(z)$ and $P_1(z)$ for the two component decomposition
- 8) Explain musical sound processing operation?